UNIVERGE SV9500

FP95-115 V5

Data Programming Manual - Business

NWD-165800-001
ISSUE 10.0

LIABILITY DISCLAIMER

NEC Corporation reserves the right to change the specifications, functions, or features, at any time, without notice.

NEC Corporation has prepared this document for use by its employees and customers. The information contained herein is the property of NEC Corporation, and shall not be reproduced without prior written approval from NEC Corporation.

All brand names and product names on this document are trademarks or registered trademarks of their respective companies.

Copyright 2013 - 2017

NEC Corporation

UNIVERGE SV9500 BUSINESS - FEATURE LIST (1 / 25)

LEGEND

*** - There are some restrictions.**
X - Applicable
— - Not Applicable
N - Not Available

A - Administration Station for Hotel System
G - Guest Station for Hotel System
XX - Feature developed by NEC Corporation of America
Countries - Asia, Latin America, Middle East, Russia
SP Controlled Std.SIP - Connection test is required in advance.
SIP Handler-Controlled Std.SIP (Single/Multi Line) - Connection test is required in advance.
SIP Handler-Controlled Std.SIP (Multi Line) - North America ONLY

INDEX	FEATURE NAME	PIR/ Peripheral Box	MARKET INFORMATION				Analog			Dterm / DT				PHS	PCS	Vo WLAN	SP Controlled Std.SIP	SIP Handler-Controlled Std.SIP (Single Line)	SIP Handler-Controlled Std.SIP (Multi Line)	Hotel	CCIS	FCCS	Location Diversity	SR-MGC(E)	REMARKS
			North America	Australia	Countries	EMEA	PIR	UG50	MC	Digital Terminal	UG50	IP Terminal *1	Soft Phone												***1**: This list does not include availability of the cradle type of SIP Multiple Line terminal. Features available on the cradle type of SIP Multiple Line terminal differ depending on the application installed in it.
[A-1]	Attendant Camp-On with Tone Indication	PIR	X	X	X	X	-	-	-	-	-	-	-	N	N	N	N	-	-	X	-	X	-	N (ATT)	
[A-2]	Attendant-Controlled Conference	PIR & CFT or VS32	X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	-	X	-	N (ATT)	
[A-3]	Attendant Console	PIR	X	X	X	X	-	-	-	-	-	-	X	-	-	-	-	-	-	-	X	-	-	-	N (ATT)	Refer to the Attendant Console section of the manual for further details on conditions.
[A-4]	Attendant Keypad	PIR	X	X	X	X	-	-	-	-	-	-	X	-	-	-	-	-	-	-	X	-	-	-	N (ATT)	
[A-5]	Attendant Lockout	PIR	X	X	X	X	-	-	-	-	-	-	X	-	-	-	-	-	-	-	X	-	-	-	N (ATT)	
[A-6]	Attendant Loop Release	PIR	X	X	X	X	-	-	-	-	-	-	-	-	-	-	-	-	-	-	X	-	-	-	N (ATT)	
[A-7]	Attendant Override	PIR	X	X	X	X	-	-	-	-	-	-	-	X	X	-	-	-	-	-	X	-	X	-	N (ATT)	
[A-8]	Automatic Recall	PIR	X	X	X	X	X	X	X	X	X	X	-	X	X	X	X	X	X	X	X	-	-	-	N (ATT)	
[A-8D]	Automatic Recall -D	PIR	X	X	X	X	-	-	-	X	X	X	-	-	-	-	-	X	X	X	X	-	-	-	X	
[A-15]	Announcement Service	PIR VS32	X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	-	X	X	X	
[A-16]	Alternate Routing		X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	
[A-17]	Audible Indication Control	PIR	X	X	X	X	-	-	-	-	-	-	-	-	-	-	-	-	-	-	A	-	-	-	N (ATT)	
[A-18]	Account Code		X	X	X	X	X	X	X	X	X	X	X	N	N	N	N	X	X	A	-	X	X	X		
[A-19]	Attendant Night Transfer	PIR	X	X	X	X	-	-	-	-	-	-	-	X	X	X	X	-	-	A	-	X	-	N (ATT)	
[A-20]	Authorization Code		X	X	X	X	X	X	X	X	X	X	X	N	N	N	N	X	X	A	-	X	X	X		
[A-29]	Authorization Code - Tie Line Incoming Routes		X	N	N	N	-	-	-	-	-	-	-	-	-	-	-	-	-	A	-	X	X	N		
[A-30]	Automatic/Manual Intercom		X	X	X	X	-	-	-	X	X	X	X	-	-	-	-	N	N	A	-	N	N	N	Available only within the same system	
[A-43]	AC Clearing		N	X	N	N	-	-	-	-	-	-	-	-	-	-	-	-	-	X	-	-	-	X		
[A-52]	Account Code - Attendant	PIR	X	X	X	X	-	-	-	-	-	-	-	N	N	-	-	-	-	A	-	X	X	N (ATT)		

UNIVERGE SV9500 BUSINESS - FEATURE LIST (2 / 25)

LEGEND

*** - There are some restrictions.**
X - Applicable
— - Not Applicable
N - Not Available

A - Administration Station for Hotel System
G - Guest Station for Hotel System
XX - Feature developed by NEC Corporation of America
Countries - Asia, Latin America, Middle East, Russia
SP Controlled Std.SIP - Connection test is required in advance.
SIP Handler-Controlled Std.SIP (Single/Multi Line) - Connection test is required in advance.
SIP Handler-Controlled Std.SIP (Multi Line) - North America ONLY

INDEX	FEATURE NAME	PIR/ Peripheral Box	MARKET INFORMATION				Analog			Dterm / DT				PHS	PCS	Vo WLAN	SP Controlled Std.SIP	SIP Handler-Controlled Std.SIP (Single Line)	SIP Handler-Controlled Std.SIP (Multi Line)	Hotel	CCIS	FCCS	Location Diversity	SR-MGC(E)	REMARKS
			North America	Australia	Countries	EMEA	PIR	UG50	MC	Digital Terminal	UG50	IP Terminal *1	Soft Phone												*1: This list does not include availability of the cradle type of SIP Multiple Line terminal. Features available on the cradle type of SIP Multiple Line terminal differ depending on the application installed in it.
[A-53]	All Zone Paging	PIR 6COT 2COT4 LC UG50	X	X	X	X	X	X	X	X	X	X	X	X	X	N	N	X	X	X	-	X	N	-	6COT and 2COT4LC can be used as an interface to the Paging Trunk. A COT card (GCD-4COTA/B/C) installed in UG50 cannot be used as an interface to the Paging Trunk. From UG50 software version 2.00, UG50 with GCD-PGTA card can be used as a paging trunk interface.
[A-55]	Automatic Change of Night Service	PIR	N	X	X	X	X	N	X	X	N	X	-	X	X	X	X	X	X	A	-	X	X	N (ATT)	
[A-59]	Announcement Service - Attendant	PIR & DAT or VS32	X	X	X	X	-	-	-	-	-	-	-	-	-	-	-	-	-	A	-	X	X	N (ATT)	
[A-60]	Attendant-to-Attendant Calling	PIR	X	X	X	X	-	-	-	-	-	-	-	-	-	-	-	-	-	X	-	N	-	N (ATT)	
[A-68]	Automatic Hold		X	X	X	X	-	-	-	X	X	X	X	-	-	-	-	-	-	-	-	-	-	X	
[A-69]	Automatic Idle Return		X	X	X	X	-	-	-	X	X	X	X	-	-	-	-	X*1	X*1	X	-	X	X	X	*1: This terminal may hear the ROT Tone or become idle automatically after a predetermined time when the line is released by the opposite Station on-hook, etc.,
[A-78]	Automatic Number Identification (ANI)		X	N	N	N	X	X	X	X	X	X	X	X	X	X	X	X	X	A	-	N	X	X	
[A-82]	Automated Attendant	PIR VS32	X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	-	X	X	X	
[A-98]	Answer Hold - Attendant	PIR	X	X	X	X	-	-	-	-	-	-	-	N	N	N	N	-	-	X	-	X	-	N (ATT)	
[A-99]	Authorization Code Display Elimination		X	X	X	X	-	-	-	X	X	X	X	-	-	-	-	N	N	A	-	-	X	X	
[A-105]	Attendant Overflow	PIR	X	N	N	N	-	-	-	-	-	-	-	N	N	N	N	-	-	A	-	-	-	N (ATT)	
[A-121]	Add-On Conference - 8 Party	PIR VSS32	X	X	X	X	X	X	X	X	X	X	X	N	N	N	N	X	X	A	X	X	X	X	
[A-125]	Analog Caller ID (Class SM)		X	X	X	X	X*	-	X*	X*	X*	X*	X*	X	X	X	X*	X*	X*	A	X	X	X	X	*: LCD is required.
[A-127]	Analog Caller ID - Station		X	X	X	X	X*1	X*2	X*1	-	-	-	-	-	-	-	-	-	-	A	X	X	X	X	*1: LCD is required. *2: This feature is available in Retrofit mode (UG50-8LC) and Multi-slot mode (UG50-LC(SIP)).
[A-139]	Account Code Override		X	N	N	N	X	X	X	X	X	X	X	N	N	N	N	X	N	A	-	X	X	X	
[A-142]	Anonymous Call Block		X	N	N	N	X	X	X	X	X	X	X	X	X	X	X	X	X	A	-	X	N	N	

UNIVERGE SV9500 BUSINESS - FEATURE LIST (3 / 25)

LEGEND

*** - There are some restrictions.**
X - Applicable
— - Not Applicable
N - Not Available

A - Administration Station for Hotel System
G - Guest Station for Hotel System
XX - Feature developed by NEC Corporation of America
Countries - Asia, Latin America, Middle East, Russia
SP Controlled Std.SIP - Connection test is required in advance.
SIP Handler-Controlled Std.SIP (Single/Multi Line) - Connection test is required in advance.
SIP Handler-Controlled Std.SIP (Multi Line) - North America ONLY

INDEX	FEATURE NAME	PIR/ Peripheral Box	MARKET INFORMATION				Analog			Dterm / DT				PHS	PCS	Vo WLAN	SP Controlled Std.SIP	SIP Handler-Controlled Std.SIP (Single Line)	SIP Handler-Controlled Std.SIP (Multi Line)	Hotel	CCIS	FCCS	Location Diversity	SR-MGC(E)	REMARKS
			North America	Australia	Countries	EMEA	PIR	UG50	MC	Digital Terminal	UG50	IP Terminal *1	Soft Phone												
[A-143]	Assisted Conference 8 Party	PIR VS32	XX	N	N	N	X	X	X	X	X	X	X	N	N	N	N	X	X	A	X	X	X	X	
[B-2]	Busy Lamp Field - Flexible	PIR	X	X	X	X	-	-	-	-	-	-	-	X	X	X	X	-	-	X	-	X	-	N (ATT)	
[B-3]	Busy Verification	PIR	X	X	X	X	-	-	-	-	-	-	-	N	N	N	N	-	-	X	-	X	-	N (ATT)	
[B-5D]	Boss - Secretary Override -D		X	X	X	X	X*1	X*1	X*1	X	X	X	X	-	-	-	-	N	N	A	-	N	X	N	*1: This terminal can be used only as a Boss station.
[B-6]	Brokerage - Hotline		X	X	X	X	-	-	-	X	X	X	X	-	-	-	-	-	-	A	-	-	X	N	
[B-8]	Boss - Secretary - Message Waiting Lamp Control		X	X	X	X	X	X	X	X	X	X	X	N	N	N	N	N	N	X	-	N	X	N	Available only within the same system.
[B-11]	Boss - Secretary Transfer		X	X	X	X	X	X	X	X	X	X	X	N	N	N	N	N	N	A	-	N	X	N	
[B-12]	Brokerage Hotline - Outside		X	X	X	X	-	-	-	X	X	X	X	-	-	-	-	-	-	A	-	X	N	N	
[B-13]	Blocking of Trunk Usage		N	X	N	N	-	-	-	-	-	-	-	-	-	-	-	-	-	A	X	X	-	-	
[B-17]	Blind Transfer To Attendant	PIR	X	X	X	X	X	X	X	X	X	X	-	X	X	X	X*	X	X	A	X	X	-	N (ATT)	*: Depends on terminal
[B-18]	Blind Transfer To Station		X	X	X	X	X	X	X	X	X	X	X	X	X	X	X*	X	X	A	X	X	X	X	*: Depends on terminal
[B-29]	BLF over FCCS		X	X	X	X	-	-	-	-	-	-	-	-	-	-	-	-	-	X	-	X	N	N (ATT)	
[B-30]	Blind Transfer with Recall		XX	N	N	N	X	X	X	X	X	X	X	X	X	N	N	X	X	X	-	X	N	X	
[C-1]	Call Back		X	X	X	X	X	X	X	X	X	X	X	X	X	X	N	X	X	A	-	X	X	X	Depends on terminal (Need a hooking function.)
[C-2]	Call Forwarding - Busy Line		X	X	X	X	X	X	X	X	X	X	X	X	X	X	X*	X	X	A	-	X	X	X	*: Depends on terminal
[C-3]	Call Forwarding - Don't Answer		X	X	X	X	X	X	X	X	X	X	X	X	X	X	X*	X	X	A	-	X	X	X	*: Depends on terminal
[C-5]	Call Forwarding - All Calls		X	X	X	X	X	X	X	X	X	X	X	X	X	X	X*1	X*2	X*2	A	-	X	X	X	*1: Depends on terminal *2: Special Dial Tone cannot be provided instead of Dial Tone for the terminal when the Station user setting CALL FORWARDING - ALL CALLS goes off-hook.
[C-6]	Call Hold		X	X	X	X	X	X	X	X	X	X	X	N	N	N	X	N	N	A	-	X	X	X	
[C-7]	Call Pickup - Group		X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	A	-	X	X	X	
[C-8]	Call Processing Indication	PIR	X	X	X	X	-	-	-	-	-	-	-	X	X	X	X	-	-	A	X	X	-	N (ATT)	
[C-9]	Call Queueing	PIR	X	X	X	X	-	-	-	-	-	-	-	X	X	X	X	-	-	X	-	-	-	N (ATT)	

REMARKS

***1**: This list does not include availability of the cradle type of SIP Multiple Line terminal. Features available on the cradle type of SIP Multiple Line terminal differ depending on the application installed in it.

UNIVERGE SV9500 BUSINESS - FEATURE LIST (4 / 25)

LEGEND

*** - There are some restrictions.**
X - Applicable
— - Not Applicable
N - Not Available

A - Administration Station for Hotel System
G - Guest Station for Hotel System
XX - Feature developed by NEC Corporation of America
Countries - Asia, Latin America, Middle East, Russia
SP Controlled Std.SIP - Connection test is required in advance.
SIP Handler-Controlled Std.SIP (Single/Multi Line) - Connection test is required in advance.
SIP Handler-Controlled Std.SIP (Multi Line) - North America ONLY

INDEX	FEATURE NAME	PIR/ Peripheral Box	MARKET INFORMATION				Analog			Dterm / DT				PHS	PCS	Vo WLAN	SP Controlled Std.SIP	SIP Handler-Controlled Std.SIP (Single Line)	SIP Handler-Controlled Std.SIP (Multi Line)	Hotel	CCIS	FCCS	Location Diversity	SR-MGC(E)	REMARKS
			North America	Australia	Countries	EMEA	PIR	UG50	MC	Digital Terminal	UG50	IP Terminal *1	Soft Phone												*1: This list does not include availability of the cradle type of SIP Multiple Line terminal. Features available on the cradle type of SIP Multiple Line terminal differ depending on the application installed in it.
[C-10]	Call Transfer - Attendant	PIR	X	X	X	X	X	X	X	X	X	X	-	X	X	X	X	X	N	X	-	X	-	N (ATT)	
[C-11]	Call Transfer - All Calls		X	X	X	X	X	X	X	X	X	X	X	X*1	X*1	X	X*2	X	N	X	-	X	X	X	*1: Camp-On by station is unavailable. *2: Depends on terminal
[C-12]	Call Waiting - Terminating		X	X	X	X	X	X	X	X	X	X	X	X	X	X	N	N	N	X	-	X	X	X	
[C-13]	Call Waiting Lamp	PIR	X	X	X	X	-	-	-	-	-	-	-	X	X	X	X	-	-	X	-	-	-	N (ATT)	
[C-14]	CCSA Access	PIR	X	N	N	N	-	-	-	-	X	-	-	-	-	-	-	-	-	A	-	-	X	N	
[C-15]	Class of Service - Individual		X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	-	N	X	X	
[C-17]	Consultation Hold - All Calls		X	X	X	X	X	X	X	X	X	X	X	X	X	X	X*	X	N	X	-	X	X	X	*: Depends on terminal
[C-20]	Centralized Attendant Service (CAS)	PIR	X	X	X	X	-	-	-	-	-	-	-	-	-	-	-	-	-	A	-	N	-	N (ATT)	
[C-21D]	Code Calling Access -D		X	N	N	N	-	-	-	X	X	X	X	-	-	-	-	X	X	A	X	X	X	X	
[C-22D]	Called Station Status Display -D		X	X	X	X	-	-	-	X	X	X	X	-	-	-	-	N	N	X	-	X	X	X	
[C-24D]	Calling Number Display -D		X	X	X	X	-	-	-	X	X	X	X	-	-	-	-	X	X	X	-	X	X	X	
[C-25]	Call Forwarding - Intercept / Announcement		X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	A	-	X	X	X	
[C-26]	Call Forwarding - Override		X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	A	-	-	X	X	
[C-27D]	Call Waiting - Answer -D		X	X	X	X	-	-	-	X	X	X	X	-	-	-	-	X	-	X	-	X	X	X	
[C-28]	Call Forwarding - All Calls - Outside		X	X	X	X	X	X	X	X	X	X	X	X	X	X	X*	X	X	A	-	X	X	X	*: Depends on terminal Std. SIP terminal can't hear Tone(ex. SST)
[C-29]	Call Park		X	X	X	X	X	X	X	X	X	X	X	X	X	X	N	X	X*1	A	-	X	X	X	*1: This terminal cannot park the call and can retrieve a parked call only
[C-30]	Call Pickup - Direct		X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	X*1	A	-	X	X	N (ATT)	*1: This terminal cannot park the call and can retrieve a parked call only
[C-31]	Call Waiting - Originating		X	X	X	X	X	X	X	X	X	X	X	X	X	X	N	X	N	A	-	X	X	X	
[C-33]	Consecutive Dialing - Attendant	PIR	X	X	X	X	-	-	-	-	-	-	-	-	-	-	-	-	-	A	-	-	-	N (ATT)	
[C-43]	Call Metering		N	X	X	X	X	N	X	X	N	X	X	X	X	X	X	X	X	X	X	X	X	N	
[C-59]	Called Number Display - Attendant	PIR	X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	-	-	-	N (ATT)	
[C-60]	Call Forwarding - Busy Line - Outside		X	X	X	X	X	X	X	X	X	X	X	X	X	X	X*	X	X	A	-	X	X	X	*: Depends on terminal Std. SIP terminal can't hear Tone(ex. SST)

UNIVERGE SV9500 BUSINESS - FEATURE LIST (5 / 25)

LEGEND

*** - There are some restrictions.**
X - Applicable
— - Not Applicable
N - Not Available

A - Administration Station for Hotel System
G - Guest Station for Hotel System
XX - Feature developed by NEC Corporation of America
Countries - Asia, Latin America, Middle East, Russia
SP Controlled Std.SIP - Connection test is required in advance.
SIP Handler-Controlled Std.SIP (Single/Multi Line) - Connection test is required in advance.
SIP Handler-Controlled Std.SIP (Multi Line) - North America ONLY

INDEX	FEATURE NAME	PIR/ Peripheral Box	MARKET INFORMATION				Analog			Dterm / DT				PHS	PCS	Vo WLAN	SP Controlled Std.SIP	SIP Handler-Controlled Std.SIP (Single Line)	SIP Handler-Controlled Std.SIP (Multi Line)	Hotel	CCIS	FCCS	Location Diversity	SR-MGC(E)	REMARKS
			North America	Australia	Countries	EMEA	PIR	UG50	MC	Digital Terminal	UG50	IP Terminal *1	Soft Phone												
[C-62]	Consecutive Speed Calling - System		X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	-	-	X	X	
[C-63]	Camp-On Busy	PIR	N	X	N	N	-	-	-	-	-	-	-	-	-	-	-	-	-	A	-	X	-	N (ATT)	
[C-69]	Call Forwarding - All Calls - Announcement		X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	A	-	X	X	X	
[C-73]	Call Back - Delayed		X	X	X	X	X	X	X	X	X	X	X	N	N	N	N	X	X	A	-	X	X	X	
[C-74]	Call Forwarding - Don't Answer - Outside		X	X	X	X	X	X	X	X	X	X	X	X	X	X	X*	X	X	A	-	X	X	X	*: Depends on terminal Std. SIP terminal can't hear Tone(ex. SST)
[C-75]	Call Forwarding - Intercept / Announcement - Attendant		X	X	X	X	X	X	X	X	X	X	X	-	-	-	-	X	X	X	-	X	X	N (ATT)	
[C-76]	Call Waiting Lamp - UCD		X	X	X	X	-	-	-	X	X	X	X	N	N	-	-	N	N	A	-	X	N	X	
[C-81]	Call Forwarding - I'm Here		X	X	X	X	X	X	X	X	X	X	X	X	X	X*1	X*1	X	X	A	-	X	X	X*2	*1: Reconfiguration and User Authentication cannot be performed. *2: In SR-MGC mode, user can not change the destination to transfer.
[C-109]	Centrex Compatibility		X	X	X	X	X	X	X	X	X	X	X	X	N	N	N	N	N	X	-	-	X	X	Available only with the Analog C.O
[C-131]	Call Forwarding - Toll Restriction		N	X	N	N	X	N	X	X	N	X	X	X	X	X	X	X	X	A	-	X	X	X	
[C-133]	Call Forwarding - Don't Answer To UCD Queue		X	N	N	N	X	X	X	X	X	X	X	X	X	X	X	X	X	A	-	X	X	X	
[C-146]	Call Hold - Conference	PIR VS32	X	X	X	X	-	-	-	X	X	X	X	-	-	-	-	N	X	A	-	X	X	X	
[C-149]	CS Report		N	X	N	N	-	-	-	-	-	-	-	-	-	-	-	-	-	-	-	N	-	-	
[C-150]	Call Block		X	N	X	X	X	X	X	X	X	X	X	X	X	N	N	X	X	X	-	X	X	X	
[C-151]	Call Return		X	X	X	X	X	-	-	X	X*1	X*1	X*1	N	N	N	N	-	-	X*2	-	X	X*3	N	*1: Use a terminal function. *2: Only an incoming trunk call can be returned, but not an incoming station call. *3: Depends on Data setting of Backup node.
[C-154]	Call Hold Pick Up		X	N	N	N	X	X	X	X	X	X	X	X	X	N	N	X	X	A	-	N	X	X	
[C-157]	Calling Party Number - Name Assignment and Display		X	X	X	X	-	-	-	X	X	X	X	N	N	-	-	X	X	A	-	N	X	X	

REMARKS
*1: This list does not include availability of the cradle type of SIP Multiple Line terminal. Features available on the cradle type of SIP Multiple Line terminal differ depending on the application installed in it.

UNIVERGE SV9500 BUSINESS - FEATURE LIST (6 / 25)

LEGEND

*** - There are some restrictions.**
X - Applicable
— - Not Applicable
N - Not Available

A - Administration Station for Hotel System
G - Guest Station for Hotel System
XX - Feature developed by NEC Corporation of America
Countries - Asia, Latin America, Middle East, Russia
SP Controlled Std.SIP - Connection test is required in advance.
SIP Handler-Controlled Std.SIP (Single/Multi Line) - Connection test is required in advance.
SIP Handler-Controlled Std.SIP (Multi Line) - North America ONLY

INDEX	FEATURE NAME	PIR/ Peripheral Box	MARKET INFORMATION				Analog			Dterm / DT				PHS	PCS	Vo WLAN	SP Controlled Std.SIP	SIP Handler-Controlled Std.SIP (Single Line)	SIP Handler-Controlled Std.SIP (Multi Line)	Hotel	CCIS	FCCS	Location Diversity	SR-MGC(E)	REMARKS
			North America	Australia	Countries	EMEA	PIR	UG50	MC	Digital Terminal	UG50	IP Terminal *1	Soft Phone												***1**: This list does not include availability of the cradle type of SIP Multiple Line terminal. Features available on the cradle type of SIP Multiple Line terminal differ depending on the application installed in it.
[C-160]	Call Trace	PIR 6COT 2COT4 LC UG50	X	N	X	X	X	X	X	X	X	X	X	N	N	-	-	X	X	X	-	-	X*	-	Not available with ISDN terminal, PS or AT-TCON. *: Depends on Data setting of Backup node for SFC in User profile data. The physical number output is a physical number in a Backup node.
[C-169]	Consultation Hold Release		X	X	X	X	X	X	X	X	X	X	X	X	X	X	X*	X	N	X	-	X	X	X	*: Depends on terminal
[C-172]	Call Forward by Service Feature Class		X	N	N	N	X	X	X	X	X	X	X	X	X	X	X*	X	X	A	-	X	X	X	*: Depends on terminal
[C-173]	Call Redirect		X	N	N	X	-	-	-	X	X	X	X	-	-	-	-	X	X	A	-	X	X	-	
[C-174]	Calling Station Number Display Elimination		X	X	X	X	-	-	-	X	X	X	X	-	-	-	-	-	-	A	-	X	X	X	
[C-176D]	Call Park Group -D		X	X	X	X	-	-	-	X	X	X	X	-	-	-	-	-	-	A	-	N	N	X	Only available under the same system *: The call in progress is retained even when user presses idle CALL PARK GROUP -D key by mistake.
[C-177]	Caller ID on Call Waiting		XX	N	N	N	X*	N	N	-	-	-	-	-	-	-	-	-	-	X	-	X	-	-	*: Only available with the terminal supporting caller number (ID) display
[C-192]	Called Station Name Display on CF		X	N	N	N	-	-	-	X	X	X	X	-	-	-	-	N	N	X	-	X	X	X	
[C-193]	Call Forwarding - Don't Answer while Call Waiting - Termination		XX	N	N	N	X	X	X	X	X	X	X	N	N	N	N	N	N	X	-	X	X	X	
[C-194]	Call Park Group		X	X	X	X	X	X	X	X	X	X	X	N	N	N	N	X	X	X	-	N	X	X	
[C-195]	Call History		X	X	X	X	N	N	N	X*1 *2	X*3 *4	X*5	N	-	-	-	-	-	-	X*6	N	X	X	X	*1: System Call History is supported only for DT400/DT300 Series with LCD. These terminals need to be accommodated in CH-16ELCA. *2: Terminal Call History is supported only for DT430/DT330 Series. *3: This feature is not available for Dterm Series i. *4: System Call History is supported only for DT400/DT300 Series with LCD. Terminal Call History is not supported. *5: Terminal Call History is supported only for DT800/DT700 Series. *6: Call History data (both System side and Terminal side) are not cleared by Guest Station Call Record Automatic Clear [G-27]. However, when you use DT700 series with firmware versions earlier than 2.2.0.0, only Terminal Call History data (terminal side) can be cleared.

UNIVERGE SV9500 BUSINESS - FEATURE LIST (7 / 25)

LEGEND

*** - There are some restrictions.**
X - Applicable
— - Not Applicable
N - Not Available

A - Administration Station for Hotel System
G - Guest Station for Hotel System
XX - Feature developed by NEC Corporation of America
Countries - Asia, Latin America, Middle East, Russia
SP Controlled Std.SIP - Connection test is required in advance.
SIP Handler-Controlled Std.SIP (Single/Multi Line) - Connection test is required in advance.
SIP Handler-Controlled Std.SIP (Multi Line) - North America ONLY

INDEX	FEATURE NAME	PIR/ Peripheral Box	MARKET INFORMATION				Analog			Dterm / DT				PHS	PCS	Vo WLAN	SP Controlled Std.SIP	SIP Handler-Controlled Std.SIP (Single Line)	SIP Handler-Controlled Std.SIP (Multi Line)	Hotel	CCIS	FCCS	Location Diversity	SR-MGC(E)	REMARKS
			North America	Australia	Countries	EMEA	PIR	UG50	MC	Digital Terminal	UG50	IP Terminal *1	Soft Phone												***1**: This list does not include availability of the cradle type of SIP Multiple Line terminal. Features available on the cradle type of SIP Multiple Line terminal differ depending on the application installed in it.
[C-203]	Call Protected Re-Registration	24PRI/ 30PRI (Proprietary) UG50	X	X	X	X	-	-	-	-	-	-	-	-	-	-	-	-	-	-	-	-	X	X	
[C-204]	Caller ID Display - Sub Line		X	X	X	X	-	-	-	X	X	X	X	-	-	-	-	-	X	X	X	X	X	X	Line keys on DSS Console are not supported.
[C-206]	Call Forward Line Lamp Indication		X	X	X	X	-	-	-	X*1	X*1	X*2	X*3	-	-	-	-	-	-	X	-	-	X	X	*1: This feature is available for Dterm Series i and DT400/DT300 series. *2: This feature is available for DtermIP (NEC Proprietary), DtermIP (SIP), Dterm Series i/ DT400/DT300 with IP and DT800/DT700 series. *3: This feature is available for Soft Client SP350 only.
[C-207]	Call Pickup - One Touch Key		X	X	X	X	X	X	X	X	X*1	X	X	X*1	X*1	X*1	X*1	-	-	-	-	-	X	X	*1: The station can be assigned to Call Pickup Group.
[C-208]	Call Forwarding (CF) Button - One Touch		X	X	X	X	-	-	-	X*1	N	X*2	N	-	-	-	-	-	-	N	-	X	N	X*3	*: This feature is available since FP95-114 V4 Issue 2.0. *1: Available for DT400/DT300 series. *2: Available for DT800/DT700 series. *3: SR-MGCs other than SR-MGC(E) do NOT support this feature.
[C-209]	Call Forwarding Destination Per Trunk Route - Incoming Call		X	X	X	X	-	-	-	-	-	-	-	-	-	-	-	-	-	X	X	X	X	X*1	*: This feature is available since FP95-114 V4 Issue 2.0. *1: SR-MGCs other than SR-MGC(E) do NOT support this feature.
[C-210]	Call Recording		X	X	X	X	-	-	-	-	-	X*1	-	-	-	-	-	-	-	A	X	X	N	X	*1: This feature is available for DT800 series.
[D-2]	Dial Access To Attendant	PIR	X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	N (ATT)	
[D-3]	Dictation Access		X	X	X	X	X	X	X	X	X	X	X	X	X	N	N	X	X	X	-	-	X	N	
[D-5]	Digital Display - Station	PIR	X	X	X	X	-	-	-	X	X	X	X	X	-	-	-	X	X	X	-	X	-	N (ATT)	
[D-6]	Digital Display - Trunk	PIR	X	X	X	X	-	-	-	-	-	-	-	-	-	-	-	-	-	X	-	X	-	N (ATT)	
[D-7]	Direct-In Termination (DIT)		X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	A	-	X	X	X	
[D-8]	Direct Inward Dialing (DID)		X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	-	X	X	X	
[D-9]	Direct Outward Dialing (DOD)		X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	-	X	X	X	
[D-10]	Distinctive Ringing		X	X	X	X	X	X	X	X	X	X	X	X*	X*	X	N	N	N	X	-	-	X	X	*: Depends on terminal

UNIVERGE SV9500 BUSINESS - FEATURE LIST (8 / 25)

LEGEND

*** - There are some restrictions.**
X - Applicable
— - Not Applicable
N - Not Available

A - Administration Station for Hotel System
G - Guest Station for Hotel System
XX - Feature developed by NEC Corporation of America
Countries - Asia, Latin America, Middle East, Russia
SP Controlled Std.SIP - Connection test is required in advance.
SIP Handler-Controlled Std.SIP (Single/Multi Line) - Connection test is required in advance.
SIP Handler-Controlled Std.SIP (Multi Line) - North America ONLY

INDEX	FEATURE NAME	PIR/ Peripheral Box	MARKET INFORMATION				Analog			Dterm / DT				PHS	PCS	Vo WLAN	SP Controlled Std.SIP	SIP Handler-Controlled Std.SIP (Single Line)	SIP Handler-Controlled Std.SIP (Multi Line)	Hotel	CCIS	FCCS	Location Diversity	SR-MGC(E)	REMARKS
			North America	Australia	Countries	EMEA	PIR	UG50	MC	Digital Terminal	UG50	IP Terminal *1	Soft Phone												***1**: This list does not include availability of the cradle type of SIP Multiple Line terminal. Features available on the cradle type of SIP Multiple Line terminal differ depending on the application installed in it.
[D-11D]	Do Not Disturb -D		X	X	X	X	-	-	-	X*	X	X*	X*	-	-	X	-	N	N	X	-	X	X	X	*: When the terminal performs re-registration while DO NOT DISTURB is set, the associated LED of the DND key will light and the Soft key will flash. However, this is only available for IP terminals and SIP Multiple Line terminals and will not be available for Digital terminals.
[D-13D]	Dial Monitor -D		X	X	X	X	-	-	-	X*	X*	X*	X*	-	-	-	-	-	-	X	-	-	X	X	*: LCD is required.
[D-15]	Day/Night Class of Service		X	X	X	X	X	X	X	X	X	X	X	X	X	X	N	X	X	X	-	X	X	N (ATT)	
[D-16]	Direct Digital Interface		X	X	X	X	-	-	-	-	-	-	-	-	-	-	-	-	-	X	X	X	-	-	
[D-31]	Delay Announcement - UCD		X	X	X	X	X	X	X	X	X	X	X	X	X	X	X*	X	X	A	-	X	N	X	*: Holding sound which terminal machine originally owns is played when UCD group station gets available.
[D-32D]	Dual Hold -D		X	X	X	X	-	-	-	X	X	X	X	-	-	-	-	X	X	X	-	N	-	X	
[D-63]	Day/Night Change - Outside Key Operation	PIR	N	N	X	X	-	-	-	-	-	-	-	-	-	-	-	-	-	A	-	-	-	N	
[D-80]	Denied Tone Classification		N	X	N	N	X	N	X	X	N	X	X	X	X	N	N	X	X	A	-	-	X	X	
[D-81]	Double Seizure Protection		N	X	N	N	-	-	-	-	-	-	-	-	-	-	-	-	-	X	-	-	-	-	
[D-87]	Dial Intercom		X	X	X	X	-	-	-	X	X	X	X	-	-	-	-	N	N	A	-	N	N	N	
[D-90D]	Delayed Ringing -D		X	X	X	X	-	-	-	X	X	X	X	-	-	-	-	N	N	A	-	X	X	X	
[D-114]	Delay Announcement - Attendant	PIR	X	X	X	X	-	-	-	-	-	-	-	-	-	-	-	-	-	X	-	X	-	N (ATT)	
[D-149]	Direct Station Selection (DSS) Console		X	X	X	X	-	-	-	X	N	X*1	N	N	N	-	-	-	-	A	-	N	N	X	*1: DSS/BLF mode is available since FP95-114 V4 Issue 2.0 or later.
[D-153]	Distinctive Ringing - Caller ID		X	N	N	N	X	-	-	X	X	X	X	N	N	N	N	-	-	X	-	X	X*	-	*: Depends on Data setting of Backup node
[D-156]	Dual Station Call		X	X	X	X	N	N	N	X	X	X	X	N	N	N	N	X	X	A	-	X	X	X	Suite Room Service is equivalent feature for Hotel system
[D-157]	Dialed Number Primary Restriction		X	N	N	X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	-	X	X	-	
[D-158]	Digital Announcement Trunk Recording via Remote Access	PIR	X	N	N	N	-	-	-	X	-	X	X	-	-	-	-	X	X	-	-	X	X	N	This feature is not supported by the VS32
[D-159]	DtermIP		X	X	X	X	-	-	-	-	N	X	-	-	-	-	-	-	-	X	X	X	X	X	
[D-162]	Distinctive Ringing List		XX	N	N	N	X	X	X	X	X	X	X	N	N	N	N	N	N	N	-	X	X	-	
[D-163]	Distinctive Ringing for Toll Calls		X	N	N	N	X	N	X	X	N	X	X	X	X	X	N	N	N	N	X	X	X	-	*: Depends on terminal
[D-164D]	Day-Night Mode Status -D - II		X	N	N	N	-	-	-	X	X	X	N	-	-	-	-	N	N	A	-	X	X	N	

UNIVERGE SV9500 BUSINESS - FEATURE LIST (9 / 25)

LEGEND

*** - There are some restrictions.**
X - Applicable
— - Not Applicable
N - Not Available

A - Administration Station for Hotel System
G - Guest Station for Hotel System
XX - Feature developed by NEC Corporation of America
Countries - Asia, Latin America, Middle East, Russia
SP Controlled Std.SIP - Connection test is required in advance.
SIP Handler-Controlled Std.SIP (Single/Multi Line) - Connection test is required in advance.
SIP Handler-Controlled Std.SIP (Multi Line) - North America ONLY

INDEX	FEATURE NAME	PIR/ Peripheral Box	MARKET INFORMATION				Analog			Dterm / DT				PHS	PCS	Vo WLAN	SP Controlled Std.SIP	SIP Handler-Controlled Std.SIP (Single Line)	SIP Handler-Controlled Std.SIP (Multi Line)	Hotel	CCIS	FCCS	Location Diversity	SR-MGC(E)	REMARKS
			North America	Australia	Countries	EMEA	PIR	UG50	MC	Digital Terminal	UG50	IP Terminal *1	Soft Phone												***1**: This list does not include availability of the cradle type of SIP Multiple Line terminal. Features available on the cradle type of SIP Multiple Line terminal differ depending on the application installed in it.
D-164D-II	Day-Night Mode Status -D - II		N	X	X	X	N	N	N	X	X	X	N	-	-	-	-	-	-	-	X	X	-	X	
[D-165]	Dialed Number Privacy Mode		X	X	X	X	-	-	-	X	X	X	N	-	-	-	-	-	-	X	-	X	X	X	
[D-166]	Dial Plan to support IP Centrex Functionality	PIR	X	X	X	X	-	-	-	-	X	X	X	X	X	X	N	X	X	N	-	X	-	X	
[D-168]	Distinctive Ringing - PBC		N	N	N	X	X	N	X	X	N	X*	X*	N	N	N	N	N	N	-	-	-	-	X	*: Supported DtermIP (Protims), SP30(Protims) only.
[D-169]	DESI-Less Terminal		X	X	X	X	-	-	-	X	N	X	-	-	-	-	-	-	-	X	X	X	X	X*	*: Supported only IP terminal.
[D-170]	Display Reversing		X	X	X	X	-	-	-	X*1	X*1	X*2	-	-	-	-	-	-	-	-	-	-	X*3	X*3	*1: Available for DT430/DT330 *2: Available for DT830/DT730, DT710, but not available 2-button type without LCD *3: Saved data for this feature returns to default.
[D-172]	Display Enlarged Characters		X	X	X	X	-	-	-	N	N	X*	-	-	-	-	-	-	-	-	-	-	X	X	*: Available for DT800/DT700 Series
[D-173]	DNIS without CTI Server		X	X	X	X	-	-	-	-	-	-	-	-	-	-	-	X	X	A	-	X	N	N	
[E-1]	Executive Right-of-Way	PIR VS32	X	X	X	X	X	X	X	X	X	X	X	X	X	X*1	N	X	X	A	-	X	X	X	3 Party conference trunk is required. *1: Unavailable from VoWLAN.
[E-3D]	Elapsed Time Display -D		X	X	X	X	-	-	-	X*	X*	X*	X*	-	-	-	-	-	-	X	-	X	X	X	*: LCD is required
[E-4D]	Exclusive Hold -D		X	X	X	X	-	-	-	X	X	X	X	N	N	N	N	X	N	X	-	-	X	X	
[E-11]	Emergency Call		X	X	X	X	X	X	X*	X	X	X*	X	N	N	N	N	N	N	X	-	X	X	N	*: IP station cannot be assigned to terminate the emergency call, though call origination is available.
[E-18]	E911-ANI Unified Number of Digits		X	N	N	N	N	N	N	X	X	X	X	N	N	N	N	X	X	A	-	N	X	-	
[E-22]	Emergency System Control		N	X	N	N	X	N	X	X	N	X	X	X	X	X	X	X	X	X	X	N	N	N	
[E-26D]	Expanded Multiple Line Operation -D		X	X	X	X	-	-	-	X	X	X	X	-	-	-	-	X	X	X	-	N	X	-	
[E-27]	Expanded Individual Speed Dial		XX	N	N	N	X	X	X	X	X	X	X	N	X	N	N	X	X	X	-	X	N	X	This is integrated into "Speed Calling-Station [S-21]".
[E-28]	Expanded Individual Speed Dial - Group		XX	N	N	N	X	X	X	X	X	X	X	N	X	N	N	X	X	X	-	X	N	X	This is integrated into "Speed Calling-Station [S-21]".
[E-29]	Expanded Account Code		X	N	N	N	X	X	X	X	X	X	X	N	N	N	N	X	X	A	-	X	X	X	
[F-1]	Flexible Numbering of Stations		X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	-	X	-	X	
[F-2]	FX Access		X	N	N	N	-	-	-	-	-	-	-	-	-	-	-	-	-	N	-	N	X	N (ATT)	
[F-3D]	Flash Button -D		X	X	X	X	-	-	-	X	X	X	X	-	-	-	-	N	N	X	-	-	X	X	
[F-4D]	Flash Entry -D		X	X	X	X	-	-	-	X	X	X	X	-	-	-	-	X	X	X	-	-	X	X	

UNIVERGE SV9500 BUSINESS - FEATURE LIST (10 / 25)

LEGEND

*** - There are some restrictions.**
X - Applicable
— - Not Applicable
N - Not Available

A - Administration Station for Hotel System
G - Guest Station for Hotel System
XX - Feature developed by NEC Corporation of America
Countries - Asia, Latin America, Middle East, Russia
SP Controlled Std.SIP - Connection test is required in advance.
SIP Handler-Controlled Std.SIP (Single/Multi Line) - Connection test is required in advance.
SIP Handler-Controlled Std.SIP (Multi Line) - North America ONLY

INDEX	FEATURE NAME	PIR/ Peripheral Box	MARKET INFORMATION				Analog			Dterm / DT				PHS	PCS	Vo WLAN	SP Controlled Std.SIP	SIP Handler-Controlled Std.SIP (Single Line)	SIP Handler-Controlled Std.SIP (Multi Line)	Hotel	CCIS	FCCS	Location Diversity	SR-MGC(E)	REMARKS
			North America	Australia	Countries	EMEA	PIR	UG50	MC	Digital Terminal	UG50	IP Terminal *1	Soft Phone												***1**: This list does not include availability of the cradle type of SIP Multiple Line terminal. Features available on the cradle type of SIP Multiple Line terminal differ depending on the application installed in it.
[F-5D]	Flexible Ringing Assignment -D		X	X	X	X	-	-	-	X	X	X	X	-	-	-	-	-	-	X	-	-	X	X	
[F-6]	Faulty Trunk Report	PIR	X	X	X	X	-	-	-	-	-	-	X	X	-	-	-	-	-	X	-	-	X	-	
[F-7]	Forced Account Code	PIR	X	X	X	X	X	X	X	X	X	X	X	N	N	N	N	X	X	X	-	-	X	X	
[F-17]	Flexible Assignment of Function Buttons		X	X	X	X	-	-	-	X	X	X	X	-	-	-	-	X	X	X	-	-	X	X	
[F-24]	Forced Release - Attendant	PIR	N	N	X	X	-	-	-	-	-	-	-	-	-	-	-	-	-	A	-	N	N	N (ATT)	
[F-31]	Follow Phone		X	X	X	X	X	N	N	X	N	N	X	N	N	N	N	N	N	N	-	N	-	-	
[F-35]	Flexible Routing - FCCS	PIR	X	X	X	X	-	-	-	-	-	-	-	-	-	-	-	X	X	A	-	X*	N	N	*: Only TDM based FCCS is supported
[F-36]	FCCS Networking via IP		X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	-	-	-	N	FCCS over IP through IPTRK is available only for a migrated system from IPX, SV7000 and SV8500.
[F-37]	FCCS Alternate Routing - Backup	PIR	X	X	X	X	-	-	-	-	-	-	-	-	-	-	-	-	-	A	-	X*1	X	N	*1: Only IP based FCCS is supported *: This feature is available only for a migrated system from IPX, SV7000 and SV8500.
[F-38]	Flexible ROT Time		X	X	X	X	X	X	X	X	X	X	X	N	N	N	N	N	N	A	-	-	X	X	
[F-41]	Flexible Network Re-routing	ISDN	X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	-	X	-	X	
[G-2]	Group Calling		X	X	X	X	X	X	X	X	X	X	X	X	X	N	N	X	X	A	-	X	X	X	
[G-3]	Ground Button Access		N	X	N	N	X	N	N	-	-	-	-	-	-	-	-	-	-	X	-	-	-	-	
[G-29]	Group Listening		X	X	X	X	-	-	-	X*1	N	X*2	N	-	-	-	-	-	-	-	-	-	N	X	*1: Available only for Dterm series i and DT series. *2: Available for DT800/DT700 series.
[H-1]	Hotline		X	X	X	X	X	X	X	X	X	X	X	X	X	N*	N*	N*	N*	X	-	X	X	N	*: This terminal can not be a calling party; however, this terminal can be called by another party.
[H-2]	House Phone		X	X	X	X	X	X	X	X	X	X	X	-	-	-	-	X	X	X	-	X	-	N (ATT)	
[H-4D]	Hands-free Answer Back -D		X	X	X	X	-	-	-	X	X	X	N	-	-	-	-	-	-	X	-	X	X	X	
[H-5D]	Hands-free Dialing/Monitoring -D		X	X	X	X	-	-	-	X	X	X	X	-	-	-	-	-	-	X	-	-	X	X	
[H-9]	Hotline - Outside		X	X	X	X	X	X	X	X	X	X	X	-	-	-	-	N	N	X	-	X	N	N	
[I-1]	Immediate Ringing		X	X	X	X	X	X	-	X	-	-	-	-	-	-	-	-	-	X	-	-	-	X	IP terminal is not supported
[I-2]	Incoming Call Identification	PIR	X	X	X	X	-	-	-	-	-	-	-	N	N	X	X	-	-	N	X	-	-	N (ATT)	

UNIVERGE SV9500 BUSINESS - FEATURE LIST (11 / 25)

LEGEND

*** - There are some restrictions.**
X - Applicable
— - Not Applicable
N - Not Available

A - Administration Station for Hotel System
G - Guest Station for Hotel System
XX - Feature developed by NEC Corporation of America
Countries - Asia, Latin America, Middle East, Russia
SP Controlled Std.SIP - Connection test is required in advance.
SIP Handler-Controlled Std.SIP (Single/Multi Line) - Connection test is required in advance.
SIP Handler-Controlled Std.SIP (Multi Line) - North America ONLY

INDEX	FEATURE NAME	PIR/ Peripheral Box	MARKET INFORMATION				Analog			Dterm / DT				PHS	PCS	Vo WLAN	SP Controlled Std.SIP	SIP Handler-Controlled Std.SIP (Single Line)	SIP Handler-Controlled Std.SIP (Multi Line)	Hotel	CCIS	FCCS	Location Diversity	SR-MGC(E)	REMARKS
			North America	Australia	Countries	EMEA	PIR	UG50	MC	Digital Terminal	UG50	IP Terminal *1	Soft Phone												***1**: This list does not include availability of the cradle type of SIP Multiple Line terminal. Features available on the cradle type of SIP Multiple Line terminal differ depending on the application installed in it.
[I-3]	Incoming Central Office Call to TIE-Line Connection	PIR	X	X	X	X	-	-	-	-	-	-	-	X	X	-	-	-	-	A	X	X	X	N (ATT)	
[I-4]	Individual Trunk Access	PIR	X	X	X	X	-	-	-	-	-	-	-	-	-	-	-	-	-	X	-	X	-	N	
[I-5]	Inter Position Transfer	PIR	X	X	X	X	-	-	-	-	-	-	-	N	N	N	N	-	-	X	-	X	-	N (ATT)	
[I-6]	Individual Attendant Access	PIR	X	X	X	X	-	-	-	-	-	-	-	X	X	X	X	-	-	X	X	X	-	N (ATT)	
[I-7D]	I-Hold Indication -D		X	X	X	X	-	-	-	X	X	X	X	-	-	-	-	-	-	X	-	-	X	X	*: Flashing interval of the LED differs from it of Dterm
[I-8D]	I-Use Indication -D		X	X	X	X	-	-	-	X	X	X	X	-	-	-	-	-	-	X	-	-	X	X	*: Flashing interval of the LED differs from it of Dterm
[I-9]	Intercom Calling		X	N	N	N	-	-	-	X	X	X	N	-	-	-	-	N	N	A	-	-	N	N	
[I-10D]	Intermediate Station Number Display -D		X	X	X	X	-	-	-	X*	X*	X*	X*	-	-	-	-	-	-	X	-	X	X	X	*: LCD is required.
[I-11]	Inter-Office Off-Hook Queuing		X	X	X	X	-	-	-	X	X	X	X	-	-	-	-	-	-	A	-	X	-	N	
[I-20]	Immediate Ringback Tone		X	X	X	X	X	X	X	X	X	X	X	X	X	X	N	X	X	X	-	-	X	X	
[I-21]	Internal Zone Paging		X	X	X	X	-	-	-	X	N	X*2	N	-	-	-	-	-	-	A	-	X*1	X	N	*1: This feature is not available for Peer-to-Peer FCCS Network. *2: IP terminals cannot be the call initiator except for DT830G/DT730G.
[I-43]	IP Enabled Dterm		X	X	X	X	-	-	-	X*	N	-	-	-	-	N	N	-	-	X	X	X	X	X	*: DT400/DT300 Series is not applicable.
[I-44]	IP Control-FAX	PIR	X	X	X	X	-	X*2	X*1	-	-	-	-	-	-	-	-	X*3	X*3	X	X	X	-	X	*1: Available only within the same system. *2: The data transmission as voice data (G.711), G. 726 and T.38 UDP are available on FAX communications. *3: This feature is available since V4. This feature is not available in Hotel system and cannot be used via CCIS over IP
[I-45]	Individual Call Hold Recall Timer		X	N	N	N	-	-	-	X	X	X	X	-	-	-	-	X	X	X	-	X	N	X	
[I-46]	Individual Call Block		XX	N	N	N	X	X	X	X	X	X	X	-	-	-	-	X	X	X	-	X	X	X	
[I-47]	IP Device Firmware Remote Download		X	X	X	X	-	N	X	-	N	X	-	-	-	-	-	-	-	X	-	-	-	-	
[I-48]	IP Terminal Failover - Alternate System		X	X	X	X	N	N	N	N	-	X	X	N	N	-	-	N	N	X	X	-	X	X	
[I-49]	Internal Zone Paging - Multi-Zone		X	N	N	N	-	-	-	X	X	X	X	-	-	N	N	N	N	X	-	X	X	-	

UNIVERGE SV9500 BUSINESS - FEATURE LIST (12 / 25)

LEGEND

*** - There are some restrictions.**
X - Applicable
— - Not Applicable
N - Not Available

A - Administration Station for Hotel System
G - Guest Station for Hotel System
XX - Feature developed by NEC Corporation of America
Countries - Asia, Latin America, Middle East, Russia
SP Controlled Std.SIP - Connection test is required in advance.
SIP Handler-Controlled Std.SIP (Single/Multi Line) - Connection test is required in advance.
SIP Handler-Controlled Std.SIP (Multi Line) - North America ONLY

INDEX	FEATURE NAME	PIR/ Peripheral Box	MARKET INFORMATION				Analog			Dterm / DT				PHS	PCS	Vo WLAN	SP Controlled Std.SIP	SIP Handler-Controlled Std.SIP (Single Line)	SIP Handler-Controlled Std.SIP (Multi Line)	Hotel	CCIS	FCCS	Location Diversity	SR-MGC(E)	REMARKS
			North America	Australia	Countries	EMEA	PIR	UG50	MC	Digital Terminal	UG50	IP Terminal *1	Soft Phone												***1**: This list does not include availability of the cradle type of SIP Multiple Line terminal. Features available on the cradle type of SIP Multiple Line terminal differ depending on the application installed in it.
[L-1]	Lamp Check	PIR	X	X	X	X	-	-	-	-	-	-	-	-	-	-	-	-	-	X	-	-	-	N (ATT)	
[L-3]	Line Lockout		X	X	X	X	X	X	X	X	X	X*	X	N	N	N	N	N	N	X	-	N	-	X	*: This feature is not available, but automatic call release can be used.
[L-5]	Least Cost Routing - 3/6-Digit		X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	-	X	
[L-6]	Last Number Called - Single-Line Station		X	X	X	X	X	X	X	X	X	X	X	N*	N*	N	N	X	X	A	-	X	X	X	*: The redial function of a terminal is substituted.
[L-6D]	Last Number Called -D		X	X	X	X	-	-	-	X	X	X	X	-	-	-	-	-	-	X	-	X	X	X	
[L-7D]	Line Reconnect - Same-Line -D		X	X	X	X	-	-	-	X	X	X	X	-	-	-	-	-	-	X	-	X	X	X	
[L-8D]	Line Reconnect - Other-Line -D		X	X	X	X	-	-	-	X	X	X	X	-	-	-	-	-	-	X	-	X	X	X	
[L-9D]	Line Pre-selection -D		X	X	X	X	-	-	-	X	X	X	X	-	-	-	-	-	-	X	-	X	X	X	
[L-10]	LCR - Time Of Day Routing		X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	-	X	
[L-11]	Line Load Control - Attendant		X	X	X	X	X	X	X	X	X	X	X	X	X	N	N	X	X	X	-	-	-	N (ATT)	
[L-18]	Long Line Trunk		N	N	X	X	X	-	-	-	-	-	-	-	-	-	-	-	-	A	-	-	-	-	
[L-20]	Line Load Control - Automatic		X	X	X	X	X	X	X	X	X	X	X	X	X	N	N	X	X	X	-	-	X*	X	*: Depends on Data setting of Backup node for SFC in User profile data
[L-21]	Line Load Control - Manual		X	X	X	X	X	X	X	X	X	X	X	X	X	N	N	X	X	X	-	-	-	-	
[L-22]	Line Fault Detection	PIR	N	X	N	N	-	-	-	-	-	-	-	-	-	-	-	-	-	X	-	-	-	-	
[L-24]	Listed Directory Number	PIR	X	X	X	X	-	-	-	-	-	-	-	-	-	-	-	-	-	X	-	X	X	N (ATT)	
[L-28]	LDN Night Connection		X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	N (ATT)	
[L-30]	LDN Night Connection - Outside		X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	N (ATT)	
[L-52]	Last Number Called - Attendant	PIR	N	N	X	X	-	-	-	-	-	-	-	-	-	-	-	-	-	X	-	N	-	N (ATT)	
[L-53]	Logged Out IP Station - Call Destination		X	X	X	X	-	X	-	-	X	X	X	-	-	-*1	-*1	X	X	X	-	X	X*2	X	*1: When the terminal is logged out, Call Forwarding-Incoming Incomplete will be set. *2: A logout transmission setup by a backup node is regulated.
[L-54]	Least Cost Routing for MG-PRI	MG UG50	X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	X*	X	*: Depends on Data setting of Backup node for SFC in User profile data

UNIVERGE SV9500 BUSINESS - FEATURE LIST (13 / 25)

LEGEND

*** - There are some restrictions.**
X - Applicable
— - Not Applicable
N - Not Available

A - Administration Station for Hotel System
G - Guest Station for Hotel System
XX - Feature developed by NEC Corporation of America
Countries - Asia, Latin America, Middle East, Russia
SP Controlled Std.SIP - Connection test is required in advance.
SIP Handler-Controlled Std.SIP (Single/Multi Line) - Connection test is required in advance.
SIP Handler-Controlled Std.SIP (Multi Line) - North America ONLY

INDEX	FEATURE NAME	PIR/ Peripheral Box	MARKET INFORMATION				Analog			Dterm / DT				PHS	PCS	Vo WLAN	SP Controlled Std.SIP	SIP Handler-Controlled Std.SIP (Single Line)	SIP Handler-Controlled Std.SIP (Multi Line)	Hotel	CCIS	FCCS	Location Diversity	SR-MGC(E)	REMARKS
			North America	Australia	Countries	EMEA	PIR	UG50	MC	Digital Terminal	UG50	IP Terminal *1	Soft Phone												***1**: This list does not include availability of the cradle type of SIP Multiple Line terminal. Features available on the cradle type of SIP Multiple Line terminal differ depending on the application installed in it.
[L-55]	Location Diversity		X	X	X	X	-	X*3	X	-	X*3	X	X*1	X	-	N	N	N	N	X	-	-	-	X*2	*: This feature does not support UMGi/RNIP system. *1: SP30(Proprietary Protocol) of software version 16 or later can be rescued. For prior to version 16, recovery for SP30(Proprietary Protocol) is only available for TP failure. *2: Terminals protected by this service are limited. *3: For Retrofit mode, available from initial software version. For multi-slot mode, available from software version 2.00.
[L-56]	Login Code		X	X	X	X	-	-	-	-	-	X	X	-	-	-	-	-	-	-	-	X	X	X	
[L-57]	Last Login Info Retention		X	X	X	X	-	-	-	-	-	X	X	-	-	-	-	N	N	-	-	X	X	X	
[M-1]	Meet-Me Paging	PIR 6COT 2COT4LC UG50	X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	-	X	N	-	6COT and 2COT4LC can be used as an interface to the Paging Trunk. A COT card (GCD-4COTA/B/C) installed in UG50 cannot be used as an interface to the Paging Trunk. From UG50 software version 2.00, UG50 with GCD-PGTA card can be used as a paging trunk interface.
[M-2]	Miscellaneous Trunk Access		X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	N	
[M-3]	Miscellaneous Trunk Restriction		X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	
[M-4]	Multiple Console Operation		X	X	X	X	-	-	-	-	-	-	-	-	-	-	-	-	-	-	A	X	X	-	N	
[M-7]	Music On Hold	PIR VS32	X	X	X	X	X	X*1	X	X	X*1	X	X	X	X	X	X	X	X	X	X	-	X	X	X	*External music is not supported. *1: Music on Hold for the terminal accommodated in UG50 differs from SV9500's one. MOH is set on a IPG-wide basis and can be select one of the following: 1.La Chanson de L'adieu (Frédéric François Chopin) / 2.Trout Quintet (Franz Peter Schubert) /3.Plaisir d'Amour (Jean Paul Egide Martini) /4. Menuet (Johann Sebastian Bach)
[M-11]	Meet-Me Paging - Attendant	PIR 6COT 2COT4LC UG50	X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	-	X	N	N	6COT and 2COT4LC can be used as an interface to the Paging Trunk. A COT card (GCD-4COTA/B/C) installed in UG50 cannot be used as an interface to the Paging Trunk. From UG50 software version 2.00, UG50 with GCD-PGTA card can be used as a paging trunk interface.
[M-15]	Maintenance Printout	PIR	X	X	X	X	-	-	-	-	-	-	-	-	-	-	-	-	-	X	-	-	-	-	

UNIVERGE SV9500 BUSINESS - FEATURE LIST (14 / 25)

LEGEND

*** - There are some restrictions.**
X - Applicable
— - Not Applicable
N - Not Available

A - Administration Station for Hotel System
G - Guest Station for Hotel System
XX - Feature developed by NEC Corporation of America
Countries - Asia, Latin America, Middle East, Russia
SP Controlled Std.SIP - Connection test is required in advance.
SIP Handler-Controlled Std.SIP (Single/Multi Line) - Connection test is required in advance.
SIP Handler-Controlled Std.SIP (Multi Line) - North America ONLY

INDEX	FEATURE NAME	PIR/ Peripheral Box	MARKET INFORMATION				Analog			Dterm / DT				PHS	PCS	Vo WLAN	SP Controlled Std.SIP	SIP Handler-Controlled Std.SIP (Single Line)	SIP Handler-Controlled Std.SIP (Multi Line)	Hotel	CCIS	FCCS	Location Diversity	SR-MGC(E)	REMARKS
			North America	Australia	Countries	EMEA	PIR	UG50	MC	Digital Terminal	UG50	IP Terminal *1	Soft Phone												
[M-16]	MFC - Incoming Call	PIR	N	N*	X	X	X	N	X	X	N	X	X	X	X	X	X	X	X	X	-	X	X	N	*: Refer to [M-53] for equivalent feature for Australian market
[M-17]	MFC - Outgoing Call	PIR	N	N	X	X	X	N	X	X	N	X	X	X	X	X	X	X	X	X	-	X	X	N	
[M-19]	Message Reminder		X	X	X	X	X	X	X	X	X	X	X	X	X	N	N	N	N	A	-	X	X	X	
[M-20D]	Multiple Line Operation -D		X	X	X	X	-	-	-	X*	X*	X*	X	-	-	-	-	-	X	X	-	N	X	X	*: Available only under the same system
[M-23]	Message Waiting Lamp Setting - Attendant		X	X	X	X	X*	X*	X*	X	X	X	X	-	-	-	-	X	X	A	-	X	-	N (ATT)	*: Available only with the terminal equipped with the message waiting lamp.
[M-24]	Multiple Call Forwarding - Busy Line		X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	A	X	X	X	X	
[M-25]	Multiple Call Forwarding - Don't Answer		X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	A	X	X	X	X	
[M-26]	Message Center Interface		X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	-	X	X	N	
[M-30]	Message Waiting Lamp Setting - Station		X	X	X	X	X	X	X	X	X	X	X	-	-	-	-	X	X	A	X	X	X	X	
[M-44]	Multiple Call Forwarding - All Calls		X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	A	X	X	X	X	
[M-47]	Manual Signaling		X	X	X	X	-	-	-	X	X	X	-	-	-	-	-	-	-	A	-	N	X	X	
[M-48]	Multi Channel Recording - DAT	PIR	X	X	X	X	-	-	-	X*	X*	X*	X*	-	-	-	-	X*	X*	A	-	X	X	-	VS32 is not supported. *: LCD is required.
[M-53]	MFC - Incoming Call	PIR	N	X	N	N	X	N	X	X	N	X	X	X	X	X	X	X	X	X	-	X	X	N	
[M-59]	MFC - Incoming Call To Tie Line	PIR	N	X	N	N	-	-	-	-	-	-	-	-	-	-	-	-	-	X	-	X	X	N	
[M-84]	MF Signaling - DID		X	N	N	N	X*	X*	X*	X	X	X	X	X	X	X	X	X	X	X	-	X	-	-	*: Terminal need to be able to send the DTMF
[M-85D]	Message Waiting Indication -D		X	X	X	X	-	-	-	X*	X*	X*	X*	-	-	-	-	X*	X*	X	-	X	X	X	*: LCD is required.
[M-92]	MFC - Tandem Switching		N	N	X	X	-	-	-	-	-	-	-	-	-	-	-	-	-	X	X	X	X	N	
[M-99]	Multiple Music-on Hold		X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	A	-	X	X	N	
[M-101]	MF ANI to SMDR		X	N	N	N	X	X	X	X	X	X	X	X	X	X	X	X	X	X	A	-	X	X	
[M-104]	Message Waiting-Special Dial Tone		N	X	N	N	X	N	X	X	N	X	X	X	X	X	N	N	N	A	-	-	X	N	
[M-105]	Multiple Music on Hold Source	PIR	N	X	X	X	X	N	N	X	N	X	N	N	N	N	N	X	X	A	-	X	X	N	
[M-107]	Message Audible Alert		X	N	N	N	X	X	X	X	X	X	X	N	N	N	N	N	N	X	-	X	X	-	
[M-111]	Media Gateway SIP	MG	X	X	X*	X*	X	X	X	X	X	X	X	X	-	X	X	-	-	X	-	X	X	X	*: Connection test with common carrier is required.
[M-113]	Multiple Number Service (MG-SIP)	MG	X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	-	X	-	X	

REMARKS

***1**: This list does not include availability of the cradle type of SIP Multiple Line terminal. Features available on the cradle type of SIP Multiple Line terminal differ depending on the application installed in it.

UNIVERGE SV9500 BUSINESS - FEATURE LIST (15 / 25)

LEGEND

*** - There are some restrictions.**
X - Applicable
— - Not Applicable
N - Not Available

A - Administration Station for Hotel System
G - Guest Station for Hotel System
XX - Feature developed by NEC Corporation of America
Countries - Asia, Latin America, Middle East, Russia
SP Controlled Std.SIP - Connection test is required in advance.
SIP Handler-Controlled Std.SIP (Single/Multi Line) - Connection test is required in advance.
SIP Handler-Controlled Std.SIP (Multi Line) - North America ONLY

INDEX	FEATURE NAME	PIR/ Peripheral Box	MARKET INFORMATION				Analog			Dterm / DT				PHS	PCS	Vo WLAN	SP Controlled Std.SIP	SIP Handler-Controlled Std.SIP (Single Line)	SIP Handler-Controlled Std.SIP (Multi Line)	Hotel	CCIS	FCCS	Location Diversity	SR-MGC(E)	REMARKS
			North America	Australia	Countries	EMEA	PIR	UG50	MC	Digital Terminal	UG50	IP Terminal *1	Soft Phone												***1**: This list does not include availability of the cradle type of SIP Multiple Line terminal. Features available on the cradle type of SIP Multiple Line terminal differ depending on the application installed in it.
[M-114]	Multiple Time Zone		X	X	X	X	-	X*1	X	-	X*1	X	X	-	-	N	N	-	-	X	-	-	-	X	*1: The date and time can be changed on an IPG basis. The different date and time information cannot be set within the same IPG.
[M-115]	Multi Group Station - UCD		X	X	X	X	X	N	X	X	X	X	X	X	X	X	N	X	X	N	-	X	X	X	
[M-117]	Multilingual Display		X	X	X*1	X	N	N	N	X*2	N	X	X	N	N	N	N	N	N	-	-	-	-	X	*1: Russian is supported with Digital Terminal only (Russia only). *2: Multilingual menu display is only available for DT430/DT330.
[M-118]	Mobility Access		N	N	N	X	-	-	-	-	-	-	-	-	-	-	-	-	-	-	N	X	N	X	Public ISDN with Caller ID is required.
[M-119D]	Multiple Tone -D		N	N	N	X	-	-	-	-	-	X	-	-	-	-	-	X	X	X	-	-	-	N	
[M-120]	Multi-Boss/Multi-Secretary		X	X	X	X	X	X	X	X	X	X	X	N	N	N	N	X	X	N	-	X*1	N	X	*1: No restriction for incoming call. Boss' Station and Secretary's Station must be in the same node.
[M-122]	MG-SIP TIE Line	MG	X	X	X	X	X	X	X	X	X	X	X	X	-	X	X	-	-	-	-	X	N	X	
[M-123]	MG-SIP Shared Route	MG	X	X	X	N	X	X	X	X	X	X	X	X	-	X	X	-	-	N	-	X	N	X	
[M-124]	My Line Number Display -D		X	X	X	X	-	-	-	X*1	X*2	X*3	N	-	-	-	-	-	-	-	-	-	X	X	*1: This feature is available only for DT400/ DT300 (CH-16ELCA is required). *2: This feature is available only for DT400/ DT300. *3: This feature is available only for DT800/ DT700.
[M-125]	MG External Music-On-Hold	MG	X	X	X	N	-	-	-	-	-	-	-	-	-	-	-	-	-	X	-	X	X	X	
[M-126]	MG-SIP TIE Line Point-to-Multipoint Connection	MG-SIP(Tie line)	X	X	X	X	-	-	-	-	-	-	-	-	-	-	-	-	-	-	-	-	-	X	*: This feature is available since FP95-113 V3.
[M-127]	MG-SIP TIE Line Path Splitting	MG-SIP(Tie line)	X	X	X	X	-	-	-	-	-	-	-	-	-	-	-	-	-	-	-	-	-	X	*: This feature is available since FP95-113 V3.
[M-128]	MG-SIP TIE Line Failure Detection Per Path	MG-SIP(Tie line)	X	X	X	X	-	-	-	-	-	-	-	-	-	-	-	-	-	-	-	-	-	X	*: This feature is available since FP95-113 V3.
[M-129]	MG-SIP TIE Line Loop Detection	MG-SIP(Tie line)	X	X	X	X	-	-	-	-	-	-	-	-	-	-	-	-	-	-	-	-	-	X	*: This feature is available since FP95-113 V3.
[M-130]	MG-SIP TIE Line In-band Tone	MG-SIP(Tie line)	X	X	X	X	-	-	-	-	-	-	-	-	-	-	-	-	-	-	-	-	-	X	*: This feature is available since FP95-113 V3.

UNIVERGE SV9500 BUSINESS - FEATURE LIST (16 / 25)

LEGEND

*** - There are some restrictions.**
X - Applicable
— - Not Applicable
N - Not Available

A - Administration Station for Hotel System
G - Guest Station for Hotel System
XX - Feature developed by NEC Corporation of America
Countries - Asia, Latin America, Middle East, Russia
SP Controlled Std.SIP - Connection test is required in advance.
SIP Handler-Controlled Std.SIP (Single/Multi Line) - Connection test is required in advance.
SIP Handler-Controlled Std.SIP (Multi Line) - North America ONLY

INDEX	FEATURE NAME	PIR/ Peripheral Box	MARKET INFORMATION				Analog			Dterm / DT				PHS	PCS	Vo WLAN	SP Controlled Std.SIP	SIP Handler-Controlled Std.SIP (Single Line)	SIP Handler-Controlled Std.SIP (Multi Line)	Hotel	CCIS	FCCS	Location Diversity	SR-MGC(E)	REMARKS
			North America	Australia	Countries	EMEA	PIR	UG50	MC	Digital Terminal	UG50	IP Terminal *1	Soft Phone												***1**: This list does not include availability of the cradle type of SIP Multiple Line terminal. Features available on the cradle type of SIP Multiple Line terminal differ depending on the application installed in it.
[M-131]	MG-SIP Call Forward - Moved Temporarily Redirect	MG	N	X*1	N	N	X	X	X	X	X	X	X	X	X	-	-	-	-	X	N	X	X	X	*: This feature is available since FP95-114 V4. *1: For New Zealand only.
[M-132]	MG-SIP REFER Based Transfer	MG	X	X*1	N	N	X	X	X	X	X	X	X	X	X	X	X	X	X	-	-	X	X	X	*: This feature is available since FP95-114 V4. *1: For New Zealand only.
[N-1]	Night Connection - Fixed		X	X	X	X	X	-	-	X	X	X	-	X	X	-	-	X	X	A	-	N	X	N (ATT)	
[N-2]	Night Connection - Flexible		X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	A	-	N	-	N (ATT)	
[N-3]	Non-delay Operation	PIR	X	X	X	X	X	X	X	X	X	X	-	X	X	X	X	-	-	X	-	X	-	N (ATT)	
[N-7D]	Non-exclusive Hold -D		X	X	X	X	-	-	-	X	X	X	X	-	-	-	-	-	-	X	-	-	X	X	
[N-8D]	Non-square Line Assignment -D		X	X	X	X	-	-	-	X	X	X	X	-	-	-	-	X	X	X	-	-	X	X	
[N-17]	Night Connection Outside - System		X	X	X	X	-	-	-	-	-	-	-	-	-	-	-	-	-	A	-	-	X	-	
[N-28]	Name Display - System		X	X	X	X	-	-	-	X*	X*	X*	X*	-	-	-	-	X	X	A	-	X	X	X	*: LCD is required.
[N-41]	Network Night Switching	PIR	N	X	N	N	-	-	-	-	-	-	-	-	-	-	-	-	-	X	-	N	N	N	
[O-1]	On-line Maintenance		X	X	X	X	-	-	-	-	-	-	-	-	-	-	-	-	-	A	-	-	-	-	
[O-2]	Outgoing Trunk Queuing		X	X	X	X	X	X	X	X	X	X	X	N	N	N	N	X	X	X	X	X	X	X	
[O-6]	Off-Hook Alarm		X	X	X	X	X	X	X	X	X	X	X	N	N	N	N	N	N	X	X	X	X	X	
[O-7]	Off-Hook Queuing		X	X	X	X	X	X	X	X	X	X	X	N	N	N	N	X	X	X	X	X	X	X*	*: In the FCCS network, this feature is NOT available.
[O-13]	Overflow - UCD		X	X	X	X	X	X	X	X	X	X	X	N	N	N	N	X	X	A	X	X	X	X	
[O-14]	Operator Monitoring	PIR	N	X	N	N	-	-	-	-	-	-	-	-	-	-	-	-	-	A	-	X	X	N (ATT)	
[O-15]	Overlapping	PIR	N	X	N	N	-	-	-	-	-	-	-	-	-	-	-	-	-	A	-	X	-	N (ATT)	
[O-16]	Outgoing Trunk Queuing - Attendant		X	X	X	X	-	-	-	-	-	-	-	-	-	-	-	-	-	X	X	X	X	N (ATT)	
[O-21]	Outgoing Trunk Busy Announcements	PIR VS32	X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	-	N	X	N	
[O-37D]	Off-Hook Line Number Display -D		X	X	X	X	-	-	-	X*	X*	X*	X*	-	-	-	-	N	N	X	-	-	X	X	*: LCD is required.
[O-41D]	Off-Hook Line Number Display -D (US)		XX	N	N	N	-	-	-	X*	X*	X*	X*	-	-	-	-	N	N	-	-	-	X	X	*: LCD is required.

UNIVERGE SV9500 BUSINESS - FEATURE LIST (17 / 25)

LEGEND

*** - There are some restrictions.**
X - Applicable
— - Not Applicable
N - Not Available

A - Administration Station for Hotel System
G - Guest Station for Hotel System
XX - Feature developed by NEC Corporation of America
Countries - Asia, Latin America, Middle East, Russia
SP Controlled Std.SIP - Connection test is required in advance.
SIP Handler-Controlled Std.SIP (Single/Multi Line) - Connection test is required in advance.
SIP Handler-Controlled Std.SIP (Multi Line) - North America ONLY

INDEX	FEATURE NAME	PIR/ Peripheral Box	MARKET INFORMATION				Analog			Dterm / DT				PHS	PCS	Vo WLAN	SP Controlled Std.SIP	SIP Handler-Controlled Std.SIP (Single Line)	SIP Handler-Controlled Std.SIP (Multi Line)	Hotel	CCIS	FCCS	Location Diversity	SR-MGC(E)	REMARKS
			North America	Australia	Countries	EMEA	PIR	UG50	MC	Digital Terminal	UG50	IP Terminal *1	Soft Phone												*1: This list does not include availability of the cradle type of SIP Multiple Line terminal. Features available on the cradle type of SIP Multiple Line terminal differ depending on the application installed in it.
[O-42]	Optional Language Pack Loading		X	X	X	X	-	-	-	X	N	X	N	-	-	-	-	N	N	-	-	X*1	X*1	X*2	*1: All nodes must have the same Extension Language data and settings. *2: The Extension Languages configured on the Telephony Server are automatically copied only when the automatic program update feature is enabled.
[P-1]	Paging Access	PIR 6COT 2COT4 LC UG50	X	X	X	X	X	X	X	X	X	X	X	N	N	N	N	X	X	X	-	X	N	X	6COT and 2COT4LC can be used as an interface to the Paging Trunk. A COT card (GCD-4COTA/B/C) installed in UG50 cannot be used as an interface to the Paging Trunk. From UG50 software version 2.00, UG50 with GCD-PGTA card can be used as a paging trunk interface.
[P-2]	Passing Dial Tone	PIR	X	X	X	X	X	X	X	X	X	X	X	X	X	N	N	X	X	X	-	X	-	N (ATT)	
[P-3]	Power Failure Transfer		X	X	X	X	X*	N	N	X*	N	X*	-	N	N	N	N	-	-	X	-	-	-	-	*: Available with the combination of Analog COT and Analog terminal, or Desktop terminal with PWR failure adapter.
[P-4]	Pushbutton Calling		X	X	X	X	X	X	X	X	-	-	-	-	-	-	-	-	-	X	-	-	-	X	
[P-5]	Pushbutton Calling - Attendant Only	PIR	X	X	X	X	-	-	-	-	-	-	-	-	-	-	-	-	-	A	-	-	-	N (ATT)	
[P-6]	Pushbutton to Rotary Conversion	PIR	X	X	X	X	X	N	N	-	-	-	X	-	-	-	-	-	-	X	-	-	-	-	
[P-7]	Peg Count		X	X	X	X	-	-	-	X*	X*	X*	X*	X*	-	X*	X*	X*	X*	X	X	X	-	-	*: Supported features differ from type of terminal.
[P-9]	Priority Call	PIR	X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	-	X	X	N (ATT)	
[P-10]	Paging Transfer	PIR 6COT 2COT4 LC UG50	X	X	X	X	X	X	X	X	X	X	X	N	N	N	N	X	N	X	-	X	N	-	6COT and 2COT4LC can be used as an interface to the Paging Trunk. A COT card (GCD-4COTA/B/C) installed in UG50 cannot be used as an interface to the Paging Trunk. From UG50 software version 2.00, UG50 with GCD-PGTA card can be used as a paging trunk interface.
[P-11]	Periodic Time Indication Tone		X	X	X	X	X	X	X	X	X	X	X	X	X	X	N	N	N	X	-	-	X	X	
[P-13D]	Prime Line Pickup -D		X	X	X	X	-	-	-	X	X	X	X	-	-	-	-	N	N	X	-	-	X	X	
[P-14D]	Privacy -D		X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	N	A	-	X	X	X	
[P-15D]	Privacy On All Lines -D		X	X	X	X	-	-	-	X	X	X	X	-	-	-	-	X	N	X	-	-	X	X	
[P-18]	Privacy Release		X	X	X	X	-	-	-	X	X	X	X	-	-	-	-	N	N	A	-	N	X	X	

UNIVERGE SV9500 BUSINESS - FEATURE LIST (18 / 25)

LEGEND

*** - There are some restrictions.**
X - Applicable
— - Not Applicable
N - Not Available

A - Administration Station for Hotel System
G - Guest Station for Hotel System
XX - Feature developed by NEC Corporation of America
Countries - Asia, Latin America, Middle East, Russia
SP Controlled Std.SIP - Connection test is required in advance.
SIP Handler-Controlled Std.SIP (Single/Multi Line) - Connection test is required in advance.
SIP Handler-Controlled Std.SIP (Multi Line) - North America ONLY

INDEX	FEATURE NAME	PIR/ Peripheral Box	MARKET INFORMATION				Analog			Dterm / DT				PHS	PCS	Vo WLAN	SP Controlled Std.SIP	SIP Handler-Controlled Std.SIP (Single Line)	SIP Handler-Controlled Std.SIP (Multi Line)	Hotel	CCIS	FCCS	Location Diversity	SR-MGC(E)	REMARKS
			North America	Australia	Countries	EMEA	PIR	UG50	MC	Digital Terminal	UG50	IP Terminal *1	Soft Phone												
[P-23]	Periodic Warning Tone		N	N	X	X	X	N	X	X	N	X	X	X	X	N	N	N	N	A	-	-	-	X	
[P-24]	Preset Conference		N	N	X	X	X	N	X	X	N	X	X	X	X	X	X	X	X	A	-	N	-	N	
[P-25]	Protection from Abnormal Dialing		N	N	X	X	X	N	X	X	N	X	X	X	X	X	X	X	X	A	-	-	-	X	
[P-26]	Protection from C.O. Dial Tone Delay		N	N	X	X	X	N	X	X	N	X	X	X	X	X	X	X	X	A	-	N	N	X	
[P-30]	Priority Paging	PIR 6COT 2COT4 LC UG50	X	X	X	X	X	X	X	X	X	X	X	N	N	N	N	X	X	X	-	X	N	X	6COT and 2COT4LC can be used as an interface to the Paging Trunk. A COT card (GCD-4COTA/B/C) installed in UG50 cannot be used as an interface to the Paging Trunk. From UG50 software version 2.00, UG50 with GCD-PGTA card can be used as a paging trunk interface.
[P-57]	Pad Lock		X	N	X	X	X	X	X	X	X	X	X	-	-	-	-	-	-	X	-	N	X	-	
[P-59]	Personal Ringer		X	X	X	X	X	X*1	X	X	X*1	X	X	X*	X*	N	N	N	N	A	-	N	X	X	*: Depends on terminal *1: Ringer patterns are different from the default SV9500 pattern
[P-60]	Privacy Release - Single Line		N	N	X	X	-	-	-	X	N	X	X	-	-	-	-	X	X	X	-	N	N	X	
[P-75]	Peer to Peer Bandwidth Control		X	X	X	X	-	X	X	X	X	X	X	-	-	X*	X*	X	X	X	-	N	-	X	*: Depends on the terminal's Codec.
[P-77]	Phantom Number Distinctive Ringing - Single Line		X	N	N	N	X	X	X	X	-	X	X	X*	X*	N	N	N	N	A	X	X	-	-	*: Depends on terminal
[P-78]	Presence		X	X	X	X	-	-	-	-	-	X*1	X*2	-	-	X	N	N	N	N	N	X	X	N	*1: Only available DT800/DT700 Series *2: Not available SP30(NEC proprietary)
[P-83]	Phonebook		X	X	X	X	-	-	-	X*2	X*5	X*2	N	-	-	-	-	-	-	X	X	X*3	X*1*4	X*1	*1: Not supported Local Phonebook. Supported System Phonebook only (DT400/300/800/700 Series only). *2: Depends on terminal (Need a Soft key function). *3: P2P FCCS only. *4: FCCS Cluster is supported when data is assigned to Backup node in advance. *5: This feature is available only for DT400/300 Series equipped with Soft Key.
[P-84]	Preventative Maintenance Notification		X	X	X	X	-	-	-	-	-	-	-	-	-	-	-	-	-	-	-	-	-	-	In data assignment, non-supported equipment can be input as a subject to Preventative Maintenance Notification.
[P-87]	Physical Station Number Calling Restriction		X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	-	X	X	X	

REMARKS

*1: This list does not include availability of the cradle type of SIP Multiple Line terminal. Features available on the cradle type of SIP Multiple Line terminal differ depending on the application installed in it.

UNIVERGE SV9500 BUSINESS - FEATURE LIST (19 / 25)

LEGEND

*** - There are some restrictions.**
X - Applicable
— - Not Applicable
N - Not Available

A - Administration Station for Hotel System
G - Guest Station for Hotel System
XX - Feature developed by NEC Corporation of America
Countries - Asia, Latin America, Middle East, Russia
SP Controlled Std.SIP - Connection test is required in advance.
SIP Handler-Controlled Std.SIP (Single/Multi Line) - Connection test is required in advance.
SIP Handler-Controlled Std.SIP (Multi Line) - North America ONLY

INDEX	FEATURE NAME	PIR/ Peripheral Box	MARKET INFORMATION				Analog			Dterm / DT				PHS	PCS	Vo WLAN	SP Controlled Std.SIP	SIP Handler-Controlled Std.SIP (Single Line)	SIP Handler-Controlled Std.SIP (Multi Line)	Hotel	CCIS	FCCS	Location Diversity	SR-MGC(E)	REMARKS
			North America	Australia	Countries	EMEA	PIR	UG50	MC	Digital Terminal	UG50	IP Terminal *1	Soft Phone												
[P-89]	Progressive Paging - VS32	VS32	X	X	X	X	-	-	-	X	N	X	X	-	-	-	-	-	-	A	-	X*	X	N	*: This feature is not available for Peer-to-Peer FCCS Network.
[P-90]	Preference Setting - Multi-Line Telephone with PCPro		X	X	X	X	-	-	-	N	X	X*1	N	-	-	-	-	-	-	N	-	-	N	X*2	*: This feature is available since FP95-114 V4 Issue 2.0. *1: Available for DT800/DT700 Series. *2: SR-MGCs other than SR-MGC(E) do NOT support this feature.
[Q-9]	Quick Transfer VMM		X	N	N	N	-	-	-	X	X	X	X	-	-	-	-	-	-	-	-	X	X	X	
[R-2]	Remote Access to System		X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	-	X	X	X	
[R-3]	Reserve Power	PIR	X	X	X	X	-	-	-	-	-	-	-	-	-	-	-	-	-	X	-	-	-	-	Battery backup feature for when commercial power failure
[R-4]	Restriction from Outgoing Calls		X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	
[R-5]	Rotary Dial Calling		X	X	X	X	X	X*	X*	-	-	-	-	-	-	-	-	-	-	X	-	-	-	-	*: Analog MC does not support DP telephones
[R-6]	Route Advance		X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	-	X	X	N	
[R-16]	Radio Paging	PIR	X	X	X	X	X	X	X	X	X	X	X	X	X	N	N	X	X	X	-	X	-	-	The following card cannot be used as an interface to the Paging Trunk: • 6COT • 2COT4LC • COT card installed in UG50
[R-20]	Restriction from Outgoing Calls - Attendant	PIR	N	N	X	X	-	-	-	-	-	-	-	-	-	-	-	-	-	X	-	X	X	N (ATT)	
[R-23]	Reverted Call Metering	PIR	N	X	N	N	-	-	-	-	-	-	-	-	-	-	-	-	-	X	-	X	-	N	
[R-27]	Route Restriction - Announcement		X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	-	-	X	X	
[R-37]	RS-464A PAD	PIR	X	N	N	N	-	-	-	-	-	-	-	-	-	-	-	-	-	X	-	-	-	-	PAD control feature for TRK circuit cards
[R-49]	Ringer Mute		X	N	X	X	-	-	-	X	X	X	X	-	-	-	-	-	-	X	-	-	X	X	Ringer Mute button needs to be assigned to a Desktop terminal Feature key.
[R-51]	Route Name Display		X	X	X	X	-	-	-	X*	X*	X*	X*	-	-	-	-	-	-	X	-	X	-	-	*: LCD is required.
[R-54]	Remote Call Forwarding Control		X	X	X	X	X	X	X	X	X	X	X	N	N	N	N	X	X	A	-	X	N	N	
[R-55]	Remote Call Hold		X	N	N	N	-	-	-	X	X	X	X	-	-	-	-	X	X	A	-	X	X	-	
[R-57D]	Ringing Line Pickup -D		X	X	X	X	-	-	-	X	X	X	X	-	-	-	-	-	-	N	-	N	X	X	

REMARKS

***1**: This list does not include availability of the cradle type of SIP Multiple Line terminal. Features available on the cradle type of SIP Multiple Line terminal differ depending on the application installed in it.

UNIVERGE SV9500 BUSINESS - FEATURE LIST (20 / 25)

LEGEND

*** - There are some restrictions.**
X - Applicable
— - Not Applicable
N - Not Available

A - Administration Station for Hotel System
G - Guest Station for Hotel System
XX - Feature developed by NEC Corporation of America
Countries - Asia, Latin America, Middle East, Russia
SP Controlled Std.SIP - Connection test is required in advance.
SIP Handler-Controlled Std.SIP (Single/Multi Line) - Connection test is required in advance.
SIP Handler-Controlled Std.SIP (Multi Line) - North America ONLY

INDEX	FEATURE NAME	PIR/ Peripheral Box	MARKET INFORMATION				Analog			Dterm / DT				PHS	PCS	Vo WLAN	SP Controlled Std.SIP	SIP Handler-Controlled Std.SIP (Single Line)	SIP Handler-Controlled Std.SIP (Multi Line)	Hotel	CCIS	FCCS	Location Diversity	SR-MGC(E)	REMARKS
			North America	Australia	Countries	EMEA	PIR	UG50	MC	Digital Terminal	UG50	IP Terminal *1	Soft Phone												***1**: This list does not include availability of the cradle type of SIP Multiple Line terminal. Features available on the cradle type of SIP Multiple Line terminal differ depending on the application installed in it.
[R-58]	RTP Information Output		X	X	X	X	-	X*1	X	-	X*1	X	X	-	-	-	-	-	-	A	-	X*2	X	N	*1: Communication Log Collection Function, Fault Log Collection Function and Playback and Cancel of warning tone for RTP packet communication error Function are supported. *2: Not applicable for Peer-to-Peer FCCS Network.
[R-60]	Redial		X	X	X	X	-	-	-	-	X*2	X*1	-	-	-	-	-	-	-	X	-	-	-	-	*1: This feature is available for DT800/DT700 series only. *2: This feature is available for DT400/DT300 series only.
[R-61]	Retention of Customized Terminal Settings		X	X	X	X	-	-	-	-	N	X*1	-	-	-	-	-	-	-	N	-	-	-	X	*1: This feature is available for DtermIP (NEC Proprietary), Dterm Series i with IP adapter and DT800/DT700 series only.
[R-62]	Ring Volume in Headset Mode		X	X	X	X	-	-	-	N	N	X	N	-	-	-	-	-	-	-	-	-	X	X	
[R-63]	Real-Time Call Information on Softphone		X	X	X	X	-	-	-	N	N	N	X*1	-	-	-	-	-	-	-	-	-	-	-	*1: This feature is available for SP350 R6.0.0.0 or later is required.
[S-1]	Single-Digit Station Calling		X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	-	X	-	X	
[S-2]	Special Dial Tone		X	X	X	X	X	X*	X	X	X*	X	X	X	X	X	N	X	X	X	-	-	X	X	*: Tones for terminals accommodated in IPG may differ from the tones for terminals directly accommodated in SV9500.
[S-3]	Speed Calling - System		X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	-	X	X	X	
[S-4]	Splitting	PIR	X	X	X	X	-	-	-	-	-	-	-	X	X	-	-	-	-	X	-	-	-	N (ATT)	
[S-4D]	Splitting -D		X	X	X	X	-	-	-	X	X	X	X	-	-	-	-	-	-	X	-	-	-	N	
[S-6]	Station-Controlled Conference	PIR VS32	X	X	X	X	X	X	X	X	X	X	X	X	X	X	X*	X	X*	X	-	-	X	X	*: Std.SIP terminal cannot be the conference leader.
[S-7]	Station Hunting - Circular		X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	-	X	X	X	
[S-8]	Station Hunting - Secretarial		X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	-	X	X	X	
[S-9]	Station Hunting - Terminal		X	X	X	X	X	X	X	X	X	X	X	X*1	X*1	X*1	X*1	X	X	X	-	X	X	X	*1: When using the transfer of unanswered calls in a Hunt Group feature, do not include in the Hunt Group PS or SP Controlled SIP terminals except as the Pilot Station.
[S-10]	Station Message Detail Recording (SMDR)		X	X	X	X	X	X	X	X	X	X	-	X	X	X	X	X	X	X	X	X	X	X	
[S-11]	Station-to-Station Calling		X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	-	X	X	X	

UNIVERGE SV9500 BUSINESS - FEATURE LIST (21 / 25)

LEGEND

*** - There are some restrictions.**
X - Applicable
— - Not Applicable
N - Not Available

A - Administration Station for Hotel System
G - Guest Station for Hotel System
XX - Feature developed by NEC Corporation of America
Countries - Asia, Latin America, Middle East, Russia
SP Controlled Std.SIP - Connection test is required in advance.
SIP Handler-Controlled Std.SIP (Single/Multi Line) - Connection test is required in advance.
SIP Handler-Controlled Std.SIP (Multi Line) - North America ONLY

INDEX	FEATURE NAME	PIR/ Peripheral Box	MARKET INFORMATION				Analog			Dterm / DT				PHS	PCS	Vo WLAN	SP Controlled Std.SIP	SIP Handler-Controlled Std.SIP (Single Line)	SIP Handler-Controlled Std.SIP (Multi Line)	Hotel	CCIS	FCCS	Location Diversity	SR-MGC(E)	REMARKS
			North America	Australia	Countries	EMEA	PIR	UG50	MC	Digital Terminal	UG50	IP Terminal *1	Soft Phone												***1**: This list does not include availability of the cradle type of SIP Multiple Line terminal. Features available on the cradle type of SIP Multiple Line terminal differ depending on the application installed in it.
[S-12]	Station-to-Station Calling - Operator Assistance	PIR	X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	-	X	X	N (ATT)	
[S-13]	Step Call		X	X	X	X	X	X	X	X	X	X	X	X	X	X	N	X	X	X	-	X	X	X	
[S-15]	Serial Call	PIR	X	X	X	X	X	X	X	X	X	X	-	X	X	X	X	-	-	X	X*	X	-	N (ATT)	*: Only available P2P CCIS.
[S-19]	Single-Digit Feature Code		X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	-	X	X	X	
[S-21]	Speed Calling - Station		X	X	X	X	X	X	X	X	X	X	X	N	X	N	N	X	X	X	-	X	X	X	The E-27/E-28 feature are included in this feature.
[S-23]	Speed Calling - Group		X	X	X	X	X	X	X	X	X	X	X	-	-	-	-	X	X	X	-	X	X	X	
[S-24D]	Save And Repeat -D		X	X	X	X	-	-	-	X	X	X	X	-	-	-	-	-	-	X	-	X	X	X	
[S-25D]	Service Display -D		X	X	X	X	-	-	-	X*	X*	X*	X*	-	-	-	-	-	-	X	-	X	X	X	*: LCD is required.
[S-26D]	Speed Calling - One Touch -D		X	X	X	X	-	-	-	X	X	X	X*	-	-	-	-	-	-	X	-	X	X	N	*: Although there is no exclusive button, the same service can be used by registering a one-touch button into a variable functional button.
[S-27]	Service Feature Peg Count		X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	-	-	-	-	
[S-31]	Supervisory Call		X	X	X	X	-	-	-	-	-	-	-	X*	X*	X*	X*	-	-	X	-	X	-	N	*: Supervisory Call cannot be set before the called terminal answers.
[S-33]	Software Line Appearance		X	X	X	X	X	X	X	X	X	X	X	-	-	-	-	-	X	X	-	N	X	X	
[S-46]	Sender/Mark - Out Signal		N	N	X	X	X	-	-	X	N	-	-	-	-	-	-	-	-	X	-	N	N	X	
[S-48]	Station Message Detail Recording - Printout		N	N	X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	-	-	-	N	Printing out the SMDR information
[S-61]	Speed Calling Override - System		X	X	X	X	X	X	X	X	X	X	X	X	X	X	X*	X	X	X	X	X	X	X	*: Depends on terminal
[S-62]	Station 5db PAD		X	X	X	X	X	X*1 *2	X*1 *2	-	-	-	X	X	X	X*3	X*3	N	N	X	-	X	X	X*2	*1: Always assigned data “1” *2: Using PAD/EC Control *3: Depends on the volume control of the terminal for the SIP terminal.
[S-64]	Serial Call - Loop Release	PIR	X	X	X	X	-	-	-	-	-	-	-	-	-	-	-	-	-	A	-	X	-	N (ATT)	
[S-65]	Short Break Hooking		N	X	N	N	X	N	X	X	N	X	X	X	X	X	N	-	-	X	-	-	-	-	*: Depends on terminal
[S-99]	Split Call Forwarding		X	X	X	X	X	X	X	X	X	X	X	X	X	X	N	X	X	X	-	X	X	X	
[S-106]	Slumber Time - Do Not Disturb		X	X	X	X	X	X	X	X	X	X	X	N	N	N	N	X	X	A	-	X	X	N	
[S-127D]	Serial Call -D		X	N	X	X	-	-	-	X	X	X	X	-	-	-	-	-	-	X	-	X	X	N	
[S-129]	SMDR Output Expansion - ANI/CPN	PIR	X	N	N	N	X	X	X	X	X	X	X	X	X	X	X	X	X	X	-	-	X	X	Out put of SMDR information

UNIVERGE SV9500 BUSINESS - FEATURE LIST (22 / 25)

LEGEND

*** - There are some restrictions.**
X - Applicable
— - Not Applicable
N - Not Available

A - Administration Station for Hotel System
G - Guest Station for Hotel System
XX - Feature developed by NEC Corporation of America
Countries - Asia, Latin America, Middle East, Russia
SP Controlled Std.SIP - Connection test is required in advance.
SIP Handler-Controlled Std.SIP (Single/Multi Line) - Connection test is required in advance.
SIP Handler-Controlled Std.SIP (Multi Line) - North America ONLY

INDEX	FEATURE NAME	PIR/ Peripheral Box	MARKET INFORMATION				Analog			Dterm / DT				PHS	PCS	Vo WLAN	SP Controlled Std.SIP	SIP Handler-Controlled Std.SIP (Single Line)	SIP Handler-Controlled Std.SIP (Multi Line)	Hotel	CCIS	FCCS	Location Diversity	SR-MGC(E)	REMARKS
			North America	Australia	Countries	EMEA	PIR	UG50	MC	Digital Terminal	UG50	IP Terminal *1	Soft Phone												*1: This list does not include availability of the cradle type of SIP Multiple Line terminal. Features available on the cradle type of SIP Multiple Line terminal differ depending on the application installed in it.
[S-137]	Selectable Station Call Pickup		X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	A	-	X	X	X	
[S-138]	Station Hunting - Priority		X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	-	X	X	X	
[S-139]	Station Hunting - Priority with Switch-back		X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	-	X	X	X	
[S-140D]	Side Tone On/Off -D		X	X	X	X	-	-	-	X	X	X*	N	-	-	-	-	-	-	X	-	X	X	X	*: Depends on the terminal
[S-141]	Survivable Remote MGC		X	X	X	X	-	X	X	-	X	X	X	-	-	X	X*1	N	N	N	N	N	X	-	*:Flexible Network Re-routing is available. *1: Some types of IP components can not be retrieved.
[S-143]	Software Multiple My Line Appearance		X	X	X	X	-	-	-	X	X	X	X	-	-	-	-	-	X	A	-	X	X	X	
[S-144]	SMDR Multi Area ID via IP	PIR	X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	-	X	X	X	MA-ID need to be assigned
[S-147]	Soft Phone Wired-Phone Handset		X	X	X	X	X	X	X	X	X	X	-	-	-	-	N	N	N	-	-	-	N	X	Need Fver(Ver.7) or later of SP30.
[S-148]	SNMP		X	X	X	X	-	-	-	-	-	-	-	-	-	-	-	-	-	A	-	-	-	N	*: SNMP manager can be connected to the system even when IP address of the SNMP manager is undefined under certain circumstances, such as when the SNMP manager uses DHCP.
[S-150]	System Message Report		X	X	X	X	-	-	-	-	-	-	-	-	-	-	-	-	-	-	-	-	X*	N	*: Depends on Data setting of Backup node
[S-151]	SIP Instant Message		X	X	X	X	-	-	-	-	-	-	X*	-	-	X	N	N	N	-	X	X	X	N	*: SP30(NEC proprietary) is Not applicable
[S-152]	SIP Terminal Authentication		X	X	X	X	-	X*4	X*1	-	-	X*2	N*2	-	-	N	N	N	N	-	-	X	X	X*3	*1: Only 8MC(SIP) is supported. *2: SP350 does not support this feature but [V-27] VoIP Encryption is available. *3: Some types of IP components can not be retrieved. *4: This feature is available for UG50-LC (SIP) and UG50-8LC only.
[S-154]	SIP Wireless Terminal Backup		X	X	X	X	-	-	-	-	-	-	-	-	-	X	X*1	-	-	X*2	X	X	X	-	*1: Conformance testing is required. *2: Suite Room secondary terminals only. *: Some types of IP components cannot be retrieved.
[S-155]	Soft Phone		X	X	X	X	-	-	-	-	-	-	X	-	-	-	-	-	-	A	X	X	X*	X	*: SP30(Proprietary Protocol) of software version 16 or later can be rescued. For prior to version 16, recovery for SP30(Proprietary Protocol) is only available for TP failure.
[S-157]	Selective Disconnect 8 Party Conference		XX	N	N	N	X	X	X	X	X	X	X	N	N	N	N	X	X	A	X	X	X	X	*: Depends on terminal

UNIVERGE SV9500 BUSINESS - FEATURE LIST (23 / 25)

LEGEND

*** - There are some restrictions.**
X - Applicable
— - Not Applicable
N - Not Available

A - Administration Station for Hotel System
G - Guest Station for Hotel System
XX - Feature developed by NEC Corporation of America
Countries - Asia, Latin America, Middle East, Russia
SP Controlled Std.SIP - Connection test is required in advance.
SIP Handler-Controlled Std.SIP (Single/Multi Line) - Connection test is required in advance.
SIP Handler-Controlled Std.SIP (Multi Line) - North America ONLY

INDEX	FEATURE NAME	PIR/ Peripheral Box	MARKET INFORMATION				Analog			Dterm / DT				PHS	PCS	Vo WLAN	SP Controlled Std.SIP	SIP Handler-Controlled Std.SIP (Single Line)	SIP Handler-Controlled Std.SIP (Multi Line)	Hotel	CCIS	FCCS	Location Diversity	SR-MGC(E)	REMARKS
			North America	Australia	Countries	EMEA	PIR	UG50	MC	Digital Terminal	UG50	IP Terminal *1	Soft Phone												***1**: This list does not include availability of the cradle type of SIP Multiple Line terminal. Features available on the cradle type of SIP Multiple Line terminal differ depending on the application installed in it.
[S-159]	Soft Phone WLAN Handset		X	X	X	X	-	-	-	-	-	-	X*1 *2	-	-	X	-	-	-	A	X	X	N	X	*1: Available with DtermSP30(Proprietary) Cver(Ver.4) or later. *2: Available with DtermSP30(SIP) Fver(Ver.7) or later.
[S-160]	SIP Multiple Line Operation		X	X	X	X	-	-	-	-	-	X*	-	-	-	-	-	-	-	X	X	X	X	X	*: The terminal of NEC proprietary protocol is not applicable
[S-162]	Selective Dialing		X	X	X	X	-	-	-	X*	X*	X*	N	-	-	-	-	-	-	-	-	-	X	X	*: Depends on terminal (Need a Soft key function).
[S-165]	Security Control Setup		X	X	X	X	-	-	-	-	-	X*1	-	-	-	-	-	-	-	X	-	-	X	X	*1: DT750, DT830/DT730 only.
[S-166]	Software Digital Tone Generator		X	X	X	X	-	-	-	-	-	-	-	-	-	-	-	X	X	X	-	-	X	N	
[S-167]	SIP Handler - SIP Terminal		X	X	X	X	-	-	-	-	-	-	-	-	-	-	-	X	X	X	-	-	X	N	
[S-168]	SP Controlled SIP Terminal		X	X	X	X	-	-	-	-	-	-	-	-	-	X	X *1 *4	-	-	X*2	-	-	N	X*3	*1: Conformance testing is required. *2: Suit Room Slave terminals only in R20.5 *3: Some types of IP components can not be retrieved. *4: Terminal supporting RFC3264 a=sendonly method on call hold feature can be accommodated.
[S-169]	SIP Handler - SIP Terminal Support for TLS Encryption		X	X	X	X	-	-	-	-	-	-	-	-	-	-	-	X	X	-	-	X	N	N	*: This feature is available since FP95-114 V4.
[T-1]	Tandem Switching of TIE Trunks - 2/4-Wire		X	X	X	X	-	-	-	X	X	X	X	-	-	-	-	-	-	A	X	X	-	-	
[T-2]	Three-Way Calling	PIR VS32	X	X	X	X	X	X	X	X	X	X	X	X	X	X	X*1	X	-*3	X	-	X	X	X*2	*1: Std.SIP terminal cannot be the conference leader and availability of this feature depends on the terminal. *2: VS32 is required. *3: This service can be used by assigning the Call Hold Conference key on the terminal by AKYD.
[T-3]	TIE Line Access		X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	-	X	X	N	
[T-5]	TIE Line Connection With Pad Control		X	X	X	X	-	-	-	-	-	-	-	-	-	-	-	-	-	X	X	X	-	-	
[T-6]	Toll Denial/Toll Diversion		X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	X*	*: Under the usage of tenant basis route restriction, need to set the same number between UGN number and tenant number.
[T-7]	Toll Restriction - 3/6-Digit		X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	
[T-8]	Trunk Answer from Any Station (TAS)		X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	A	X	X	N	N (ATT)	

UNIVERGE SV9500 BUSINESS - FEATURE LIST (24 / 25)

LEGEND

*** - There are some restrictions.**
X - Applicable
— - Not Applicable
N - Not Available

A - Administration Station for Hotel System
G - Guest Station for Hotel System
XX - Feature developed by NEC Corporation of America
Countries - Asia, Latin America, Middle East, Russia
SP Controlled Std.SIP - Connection test is required in advance.
SIP Handler-Controlled Std.SIP (Single/Multi Line) - Connection test is required in advance.
SIP Handler-Controlled Std.SIP (Multi Line) - North America ONLY

INDEX	FEATURE NAME	PIR/ Peripheral Box	MARKET INFORMATION				Analog			Dterm / DT				PHS	PCS	Vo WLAN	SP Controlled Std.SIP	SIP Handler-Controlled Std.SIP (Single Line)	SIP Handler-Controlled Std.SIP (Multi Line)	Hotel	CCIS	FCCS	Location Diversity	SR-MGC(E)	REMARKS
			North America	Australia	Countries	EMEA	PIR	UG50	MC	Digital Terminal	UG50	IP Terminal *1	Soft Phone												***1**: This list does not include availability of the cradle type of SIP Multiple Line terminal. Features available on the cradle type of SIP Multiple Line terminal differ depending on the application installed in it.
[T-9]	Trunk Group Busy Lamp	PIR	X	X	X	X	-	-	-	-	-	-	-	-	-	-	-	-	-	X	-	X	-	N (ATT)	
[T-10]	Trunk-to-Trunk Connection		X	X	X	X	X	X	X	X	X	X	X	X	X	X	X*	X	X	A	X	X	X	X	*: Depends on terminal
[T-12]	Tenant Service		X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	-	X	X	X	
[T-18]	Time Display	PIR	X	X	X	X	-	-	-	-	-	-	-	-	-	-	-	-	-	X	-	X	-	N (ATT)	
[T-18D]	Time Display -D		X	X	X	X	-	-	-	X*	X*	X*	X	-	-	-	-	-	-	X	-	-	X	X	*: LCD is required.
[T-23]	Trunk Line Appearance		X	X	X	X	-	-	-	X	X	X	X	-	-	-	-	-	-	X	-	N	X	X	Feature key Assignment is required
[T-23D]	Trunk Line Appearance -D	6COT/ 2COT4 LC UG50	X	X	X	X	-	-	-	-	X	X	X	-	-	-	-	-	-	-	-	X	X	X	
[T-28]	Trunk-to-Trunk Third-Party Cancellation		X	X	X	X	X	X	X	X	X	X	X	X	X	X	X*	X	X	X	-	X	X	X	*: Depends on terminal
[T-62]	Timed Reminder		X	X	X	X	X	X	X	X	X	X	X	N	N	N	N	X	X	X	-	X	X	X	
[T-63]	Tone Block		X	N	X	X	X	X	X	X	N	-	-	N	N	N	N	-	-	X	-	X	X	X	
[T-64]	Terminal Login via NAT		X	X	X	X	-	-	-	-	N	X*	-	-	-	-	-	-	-	A	-	-	-	X	*: DtermIP (Protims) only
[T-65]	Terminal Claim Handling		X	X	X	X	-	-	-	X	N	X	X	-	-	-	-	-	-	-	-	-	-	-	
[T-66]	Terminal Remote Login		X	X	X	X	-	-	-	-	N	X	X	-	-	-	-	N	N	A	X	X	-	N	
[T-67]	Timed Forced Disconnect Trunk		XX	N	X	N	-	-	-	-	-	-	-	-	-	-	-	-	-	-	-	-	-	X	
[T-68]	Time-based Routing for Multiple Time Zones		X	X	X	X	-	X	X	-	X	X	X	N	N	N	N	X	X	X	-	-	-	X	
[T-69]	Terminal Power Save Mode	7U-PIR	X	X	X	X	-	-	-	X	N	X*	-	-	-	-	-	-	-	-	-	-	N	X	*: Available for DT800/DT700 series
[T-86]	Timing Start		X	X	X	X	X	X	X	X	X	X	X	-	-	N	N	N	N	X*	X	X	X	X	*: To use this features for the Hotel system, see Timing Start for Hotel [T-21].
[U-1]	Uniform Call Distribution (UCD)		X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	-	X	X*	X	*: Controlling stations are not available in Location Diversity [L-55] (only non-controlling stations are available).
[U-3]	Universal Sender	PIR	X	X	X	X	X	-	-	X	N	-	-	-	-	-	-	-	-	X	-	-	-	-	
[V-1]	Variable Timing Parameters		X	X	X	X	-	-	-	-	-	-	-	-	-	-	-	-	-	X	-	-	X	X	

UNIVERGE SV9500 BUSINESS - FEATURE LIST (25 / 25)

LEGEND

*** - There are some restrictions.**
X - Applicable
— - Not Applicable
N - Not Available

A - Administration Station for Hotel System
G - Guest Station for Hotel System
XX - Feature developed by NEC Corporation of America
Countries - Asia, Latin America, Middle East, Russia
SP Controlled Std.SIP - Connection test is required in advance.
SIP Handler-Controlled Std.SIP (Single/Multi Line) - Connection test is required in advance.
SIP Handler-Controlled Std.SIP (Multi Line) - North America ONLY

INDEX	FEATURE NAME	PIR/ Peripheral Box	MARKET INFORMATION				Analog			Dterm / DT				PHS	PCS	Vo WLAN	SP Controlled Std.SIP	SIP Handler-Controlled Std.SIP (Single Line)	SIP Handler-Controlled Std.SIP (Multi Line)	Hotel	CCIS	FCCS	Location Diversity	SR-MGC(E)	REMARKS
			North America	Australia	Countries	EMEA	PIR	UG50	MC	Digital Terminal	UG50	IP Terminal *1	Soft Phone												***1**: This list does not include availability of the cradle type of SIP Multiple Line terminal. Features available on the cradle type of SIP Multiple Line terminal differ depending on the application installed in it.
[V-2]	Voice Call		X	X	X	X	X	X	X	X*1 *2	X	X*1 *2	X	X	X	N	N	-	-	X	-	X*2	X* 2 *3	X*2	Depends on terminal. *1: Cannot be the called party in Hotel System. *2: Voice call can be made to the called Desktop terminal which uses headset. *3: Depends on Data setting of Backup node for SFC in User profile data
[V-3D]	Volume Control -D		X	X	X	X	-	-	-	X	X	X	X	-	-	-	-	-	-	X	-	-	X	X	
[V-19]	Voice Mail Password Display Elimination		XX	N	N	N	-	-	-	X	X	X	X	-	-	-	-	N	N	X	X	X	X	X	
[V-26]	VS32 Conference Server	VS32	X	X	X	X	X	X	X	X	X	X	X	X	X	X	X*	X	X	X	-	X	N	X	*: Std.SIP terminal cannot be the conference leader and availability of this feature depends on the terminal
[V-27]	VoIP Encryption		X	X	X*1	X	-*2	N	X	-*2	N	X	X*3	-	-	N	N	N	N	X	N	X	X	X	*1: Not available in China. *2: Need to arrange appropriate IPPADA. *3: This feature is available for SP350 ONLY.
[V-29]	Voice Mail Indication - SIP		X	X	X	X	-	-	-	-	-	-	-	-	-	X	X	-	-	-	-	-	-	N	
[V-30]	Volume Set - Receiver		X	X	X	X	-	-	-	X	X	X	-	-	-	-	-	-	-	X	-	-	N	X	
[V-31]	Video Call	MG	X	X	X	X	-	-	-	-	-	-	-	-	-	-	-	X	X	X	N	X	N	N	*: This feature is available since FP95-114 V4.
[W-1]	WATS Access	PIR	X	N	N	N	X	X	X	X	X	X	X	X	X	X	X	X	X	X	-	-	N	-	
[W-11]	WLAN Handset Name Display		X	X	X	X	-	-	-	-	-	-	-	-	-	X*1	-	-	-	N	N	X*2	-	X	*1: Only MH250 is supported. *2: P2P Only

NWD-165800-001
ISSUE 10.0
OCTOBER, 2017

UNIVERGE SV9500
Data Programming Manual - Business

TABLE OF CONTENTS

Page

CHAPTER 1 INTRODUCTION 1
1. General 2
2. How to Use This Manual 2
3. Abbreviations for System Types 2

CHAPTER 2 ASSIGNMENT OF BASIC OFFICE DATA 4
1. Tenant Data 6
1.1 Use of Tenant Service 6
1.2 Example of Tenant Development 6
1.3 Office Data related to Tenant Service 8
1.4 Examples of Data Programming 10
2. Trunk Data 12
2.1 Trunk 12
2.2 Internal Trunk 12
2.3 External Trunk 15
3. Attendant Console (ATTCON) Data 16
3.1 Attendant Console 16
3.2 Assignment of ATTCON Data 16
4. Station Data 19
4.1 Assignment of Station Data 19
4.2 Assignment of Station-to-Station Connection Data 23
5. Desktop Terminal Data 24
5.1 Multiple Lines 25
5.2 Assignment of Desktop Terminal Station Data 26
5.3 Display (LCD) Indications 28
5.4 Line Pickup Preference 30
5.5 Restriction of Call Termination on My Line 31
5.6 Soft Key Assignment 32
5.7 Expanded Ringer Pattern for Digital Terminal 37
5.7.1 To change the ringer pattern for SIP Multiple Line terminal 37
5.8 Headset Data 38
5.9 Terminal Operations with Feature and Digit Keys for Digital Terminals 39
5.10 Terminal Call History Data Clear 40
6. Telephone Number 42
7. OGC/OGCA 46
7.1 OGC 46
7.2 OGCA 47
7.3 Network by OGC 48
8. LCR/LCRS 49

TABLE OF CONTENTS (CONTINUED)

Page

8.1 LCR/LCRS 49
8.2 Difference between LCR and LCRS 49
8.3 Network by LCR/LCRS 54
9. Uniform Numbering 55
10. PH Data Setting 59
10.1 MPH (Multi Protocol Handler) 59
10.2 Internal PHE 59
10.3 SP-PHD 60
10.4 SP-PHI 61
10.5 Internal PHI 61
10.6 Internal PHF 62
10.7 Internal PHC 62
10.8 Internal SIP Handler 62
10.9 Consideration for SP-PHI Data Assignment 62
11. Tone Setting 68
12. Data Setting for Switch Redundancy Configuration 71
13. Day/Night Data Retention 74
14. IP Related Data Setting 80
14.1 IP Station Data 80
14.2 Consideration for Connecting an IP Station to a PSTN (Analog and ISDN) 88
14.3 Virtual PIR Data 92
14.4 IPPAD Data 96
14.5 IP Service PAD/Echo Canceler Data 102
14.5.1 PAD/EC Control on a Node Basis 102
14.5.2 PAD/EC Control on a Location ID Basis 109
14.6 Tag VLAN Data 113
14.7 ToS Control Data 116
14.8 PAD/EC Control IP/PHC Support 119
14.8.1 Initial Value 121
14.8.2 Initial Value Modification 121
14.8.3 Designated Station Assignment 121
14.9 MG Based Call Retention 169
15. UG50 Related Data Setting 177
15.1 How to Use the ARAI Command to Assign or Delete Office Data for a UG50 177
15.1.1 Office Data Assignment 179
15.1.2 Office Data Deletion 183

CHAPTER 3 BUSINESS FEATURES 184
A-1 Attendant Camp-On with Tone Indication 186
A-2 Attendant-Controlled Conference 191
A-3 Attendant Console 196
A-4 Attendant Keypad 199
A-5 Attendant Lockout 200
A-6 Attendant Loop Release 201
A-7 Attendant Override 203
A-8 Automatic Recall 206

TABLE OF CONTENTS (CONTINUED)

		Page
A-8D	Automatic Recall -D	208
A-15	Announcement Service	209
A-16	Alternate Routing	212
A-17	Audible Indication Control	213
A-18	Account Code	214
A-19	Attendant Night Transfer	217
A-20	Authorization Code	220
A-29	Authorization Code - Tie Line Incoming Routes	227
A-30	Automatic/Manual Intercom	232
A-43	AC Clearing	236
A-52	Account Code - Attendant	237
A-53	All Zone Paging	239
A-55	Automatic Change of Night Service	242
A-59	Announcement Service - Attendant	246
A-60	Attendant-to-Attendant Calling	250
A-68	Automatic Hold	252
A-69	Automatic Idle Return	257
A-78	Automatic Number Identification (ANI)	260
A-82	Automated Attendant	262
A-98	Answer Hold - Attendant	266
A-99	Authorization Code Display Elimination	268
A-105	Attendant Overflow	270
A-121	Add-On Conference - 8 Party	274
A-125	Analog Caller ID (Class SM)	293
A-127	Analog Caller ID - Station	297
A-139	Account Code Override	308
A-142	Anonymous Call Block	309
A-143	Assisted Conference 8 Party	312
B-2	Busy Lamp Field - Flexible	315
B-3	Busy Verification	318
B-5D	Boss - Secretary Override -D	323
B-6	Brokerage - Hotline	326
B-8	Boss - Secretary - Message Waiting Lamp Control	330
B-11	Boss - Secretary Transfer	334
B-12	Brokerage Hotline - Outside	336
B-13	Blocking of Trunk Usage	339
B-17	Blind Transfer To Attendant	340
B-18	Blind Transfer To Station	343
B-29	BLF over FCCS	346
B-30	Blind Transfer with Recall	347
C-1	Call Back	348
C-2	Call Forwarding - Busy Line	352
C-3	Call Forwarding - Don’t Answer	363
C-5	Call Forwarding - All Calls	372
C-6	Call Hold	382
C-7	Call Pickup - Group	388
C-8	Call Processing Indication	398
C-9	Call Queueing	399
C-10	Call Transfer - Attendant	400

TABLE OF CONTENTS (CONTINUED)

		Page
C-11	Call Transfer - All Calls	402
C-12	Call Waiting - Terminating	412
C-13	Call Waiting Lamp	416
C-14	CCSA Access	419
C-15	Class of Service - Individual	420
C-17	Consultation Hold - All Calls	423
C-20	Centralized Attendant Service (CAS)	426
C-21D	Code Calling Access -D	432
C-22D	Called Station Status Display -D	434
C-24D	Calling Number Display -D	436
C-25	Call Forwarding - Intercept / Announcement	437
C-26	Call Forwarding - Override	442
C-27D	Call Waiting - Answer -D	443
C-28	Call Forwarding - All Calls - Outside	445
C-29	Call Park	452
C-30	Call Pickup - Direct	457
C-31	Call Waiting - Originating	460
C-33	Consecutive Dialing - Attendant	465
C-43	Call Metering	467
C-59	Called Number Display - Attendant	468
C-60	Call Forwarding - Busy Line - Outside	469
C-62	Consecutive Speed Calling - System	474
C-63	Camp-On Busy	476
C-69	Call Forwarding - All Calls - Announcement	477
C-73	Call Back - Delayed	481
C-74	Call Forwarding - Don’t Answer - Outside	482
C-75	Call Forwarding - Intercept / Announcement - Attendant	486
C-76	Call Waiting Lamp - UCD	489
C-81	Call Forwarding - I’m Here	492
C-109	Centrex Compatibility	499
C-131	Call Forwarding - Toll Restriction	501
C-133	Call Forwarding - Don’t Answer To UCD Queue	502
C-146	Call Hold - Conference	503
C-149	CS Report	505
C-150	Call Block	523
C-151	Call Return	530
C-154	Call Hold Pick Up	536
C-157	Calling Party Number - Name Assignment and Display	537
C-160	Call Trace	540
C-169	Consultation Hold Release	544
C-172	Call Forward by Service Feature Class	550
C-173	Call Redirect	553
C-174	Calling Station Number Display Elimination	556
C-176D	Call Park Group -D	560
C-177	Caller ID on Call Waiting	567
C-192	Called Station Name Display on CF	570
C-193	Call Forwarding - Don’t Answer while Call Waiting - Termination	571
C-194	Call Park Group	573
C-195	Call History	578

TABLE OF CONTENTS (CONTINUED)

		Page
C-203	Call Protected Re-Registration	597
C-204	Caller ID Display - Sub Line	602
C-206	Call Forward Line Lamp Indication	613
C-207	Call Pickup - One Touch Key	617
C-208	Call Forwarding (CF) Button - One Touch	626
C-209	Call Forwarding Destination Per Trunk Route - Incoming Call	632
C-210	Call Recording	641
D-2	Dial Access To Attendant	650
D-3	Dictation Access	652
D-5	Digital Display - Station	654
D-6	Digital Display - Trunk	657
D-7	Direct-In Termination (DIT)	660
D-8	Direct Inward Dialing (DID)	664
D-9	Direct Outward Dialing (DOD)	667
D-10	Distinctive Ringing	670
D-11D	Do Not Disturb -D	676
D-13D	Dial Monitor -D	678
D-15	Day/Night Class of Service	679
D-16	Direct Digital Interface	682
D-31	Delay Announcement - UCD	687
D-32D	Dual Hold -D	694
D-63	Day/Night Change - Outside Key Operation	696
D-80	Denied Tone Classification	697
D-81	Double Seizure Protection	698
D-87	Dial Intercom	699
D-90D	Delayed Ringing -D	704
D-114	Delay Announcement - Attendant	708
D-149	Direct Station Selection (DSS) Console	713
D-153	Distinctive Ringing - Caller ID	757
D-156	Dual Station Call	760
D-157	Dialed Number Primary Restriction	766
D-158	Digital Announcement Trunk Recording via Remote Access	768
D-159	DtermIP	777
D-162	Distinctive Ringing List	801
D-163	Distinctive Ringing for Toll Calls	803
D-164D	Day-Night Mode Status -D	805
D-164D - II	Day-Night Mode Status -D - II	809
D-165	Dialed Number Privacy Mode	817
D-166	Dial Plan to support IP Centrex Functionality	823
D-168	Distinctive Ringing - PBC	857
D-169	DESI-Less Terminal	859
D-170	Display Reversing	865
D-172	Display Enlarged Characters	869
D-173	DNIS without CTI Server	872
E-1	Executive Right-of-Way	876
E-3D	Elapsed Time Display -D	880
E-4D	Exclusive Hold -D	881
E-11	Emergency Call	883
E-18	E911-ANI Unified Number of Digits	886

TABLE OF CONTENTS (CONTINUED)

		Page
E-22	Emergency System Control	889
E-26D	Expanded Multiple Line Operation -D	891
E-27	Expanded Individual Speed Dial	894
E-28	Expanded Individual Speed Dial - Group	897
E-29	Expanded Account Code	900
F-1	Flexible Numbering of Stations	904
F-2	FX Access	906
F-3D	Flash Button -D	909
F-4D	Flash Entry -D	910
F-5D	Flexible Ringing Assignment -D	912
F-6	Faulty Trunk Report	914
F-7	Forced Account Code	916
F-17	Flexible Assignment of Function Buttons	921
F-24	Forced Release - Attendant	922
F-31	Follow Phone	923
F-35	Flexible Routing - FCCS	927
F-36	FCCS Networking via IP	928
F-37	FCCS Alternate Routing - Backup	953
F-38	Flexible ROT Time	961
F-41	Flexible Network Re-routing	963
G-2	Group Calling	990
G-3	Ground Button Access	992
G-29	Group Listening	993
H-1	Hotline	997
H-2	House Phone	999
H-4D	Hands-free Answer Back -D	1000
H-5D	Hands-free Dialing/Monitoring -D	1001
H-9	Hotline - Outside	1002
I-1	Immediate Ringing	1005
I-2	Incoming Call Identification	1006
I-3	Incoming Central Office Call to TIE-Line Connection	1009
I-4	Individual Trunk Access	1012
I-5	Inter Position Transfer	1016
I-6	Individual Attendant Access	1018
I-7D	I-Hold Indication -D	1019
I-8D	I-Use Indication -D	1020
I-9	Intercom Calling	1021
I-10D	Intermediate Station Number Display -D	1023
I-11	Inter-Office Off-Hook Queuing	1025
I-20	Immediate Ringback Tone	1030
I-21	Internal Zone Paging	1031
I-43	IP Enabled Dterm	1048
I-44	IP Control-FAX	1051
I-45	Individual Call Hold Recall Timer	1072
I-46	Individual Call Block	1077
I-47	IP Device Firmware Remote Download	1082
I-48	IP Terminal Failover - Alternate System	1093
I-49	Internal Zone Paging - Multi-Zone	1106
L-1	Lamp Check	1109

TABLE OF CONTENTS (CONTINUED)

		Page
L-3	Line Lockout	1110
L-5	Least Cost Routing - 3/6-Digit	1112
L-6	Last Number Called - Single-Line Station	1123
L-6D	Last Number Called -D	1125
L-7D	Line Reconnect - Same-Line -D	1131
L-8D	Line Reconnect - Other-Line -D	1132
L-9D	Line Pre-selection -D	1133
L-10	LCR - Time Of Day Routing	1137
L-11	Line Load Control - Attendant	1140
L-18	Long Line Trunk	1143
L-20	Line Load Control - Automatic	1144
L-21	Line Load Control - Manual	1145
L-22	Line Fault Detection	1146
L-24	Listed Directory Number	1147
L-28	LDN Night Connection	1150
L-30	LDN Night Connection - Outside	1157
L-52	Last Number Called - Attendant	1163
L-53	Logged Out IP Station - Call Destination	1164
L-54	Least Cost Routing for MG-PRI	1172
L-55	Location Diversity	1186
L-56	Login Code	1224
L-57	Last Login Info Retention	1228
M-1	Meet-Me Paging	1233
M-2	Miscellaneous Trunk Access	1236
M-3	Miscellaneous Trunk Restriction	1237
M-4	Multiple Console Operation	1239
M-7	Music On Hold	1241
M-11	Meet-Me Paging - Attendant	1245
M-15	Maintenance Printout	1249
M-16	MFC - Incoming Call	1250
M-17	MFC - Outgoing Call	1274
M-19	Message Reminder	1292
M-20D	Multiple Line Operation -D	1296
M-23	Message Waiting Lamp Setting - Attendant	1299
M-24	Multiple Call Forwarding - Busy Line	1302
M-25	Multiple Call Forwarding - Don’t Answer	1309
M-26	Message Center Interface	1314
M-30	Message Waiting Lamp Setting - Station	1324
M-44	Multiple Call Forwarding - All Calls	1328
M-47	Manual Signaling	1335
M-48	Multi Channel Recording - DAT	1337
M-53	MFC - Incoming Call	1346
M-59	MFC - Incoming Call To Tie Line	1352
M-84	MF Signaling - DID	1356
M-85D	Message Waiting Indication -D	1360
M-92	MFC - Tandem Switching	1361
M-99	Multiple Music-on Hold	1383
M-101	MF ANI to SMDR	1388
M-104	Message Waiting-Special Dial Tone	1391

		Page
M-105	Multiple Music on Hold Source	1392
M-107	Message Audible Alert	1397
M-111	Media Gateway SIP	1399
M-113	Multiple Number Service (MG-SIP)	1426
M-114	Multiple Time Zone	1432
M-115	Multi Group Station - UCD	1447
M-117	Multilingual Display	1450
M-118	Mobility Access	1461
M-119D	Multiple Tone -D	1508
M-120	Multi-Boss/Multi-Secretary	1512
M-122	MG-SIP TIE Line	1574
M-123	MG-SIP Shared Route	1601
M-124	My Line Number Display -D	1609
M-125	MG External Music-On-Hold	1616
M-126	MG-SIP TIE Line Point-to-Multipoint Connection	1623
M-127	MG-SIP TIE Line Path Splitting	1626
M-128	MG-SIP TIE Line Failure Detection Per Path	1685
M-129	MG-SIP TIE Line Loop Detection	1688
M-130	MG-SIP TIE Line In-band Tone	1691
M-131	MG-SIP Call Forward - Moved Temporarily Redirect	1694
M-132	MG-SIP REFER Based Transfer	1697
N-1	Night Connection - Fixed	1700
N-2	Night Connection - Flexible	1703
N-3	Non-delay Operation	1705
N-7D	Non-exclusive Hold -D	1707
N-8D	Non-square Line Assignment -D	1711
N-17	Night Connection Outside - System	1713
N-28	Name Display - System	1715
N-41	Network Night Switching	1728
O-1	On-line Maintenance	1729
O-2	Outgoing Trunk Queuing	1730
O-6	Off-Hook Alarm	1736
O-7	Off-Hook Queuing	1742
O-13	Overflow - UCD	1745
O-14	Operator Monitoring	1748
O-15	Overlapping	1749
O-16	Outgoing Trunk Queuing - Attendant	1751
O-21	Outgoing Trunk Busy Announcements	1753
O-37D	Off-Hook Line Number Display -D	1754
O-41D	Off-Hook Line Number Display -D (US)	1756
O-42	Optional Language Pack Loading	1759
P-1	Paging Access	1782
P-2	Passing Dial Tone	1793
P-3	Power Failure Transfer	1796
P-4	Pushbutton Calling	1798
P-5	Pushbutton Calling - Attendant Only	1800
P-6	Pushbutton to Rotary Conversion	1802
P-7	Peg Count	1804

		Page
P-9	Priority Call	1809
P-10	Paging Transfer	1814
P-11	Periodic Time Indication Tone	1825
P-13D	Prime Line Pickup -D	1827
P-14D	Privacy -D	1829
P-15D	Privacy On All Lines -D	1831
P-18	Privacy Release	1832
P-23	Periodic Warning Tone	1841
P-24	Preset Conference	1842
P-25	Protection from Abnormal Dialing	1846
P-26	Protection from C.O. Dial Tone Delay	1847
P-30	Priority Paging	1848
P-57	Pad Lock	1856
P-59	Personal Ringer	1860
P-60	Privacy Release - Single Line	1864
P-75	Peer to Peer Bandwidth Control	1865
P-77	Phantom Number Distinctive Ringing - Single Line	1888
P-78	Presence	1892
P-83	Phonebook	1895
P-84	Preventative Maintenance Notification	1913
P-87	Physical Station Number Calling Restriction	1914
P-89	Progressive Paging - VS32	1917
P-90	Preference Setting - Multi-Line Telephone with PCPro	1922
Q-9	Quick Transfer VMM	1928
R-2	Remote Access to System	1930
R-3	Reserve Power	1934
R-4	Restriction from Outgoing Calls	1935
R-5	Rotary Dial Calling	1938
R-6	Route Advance	1940
R-16	Radio Paging	1943
R-20	Restriction from Outgoing Calls - Attendant	1947
R-23	Reverted Call Metering	1948
R-27	Route Restriction - Announcement	1950
R-37	RS-464A PAD	1951
R-49	Ringer Mute	1955
R-51	Route Name Display	1956
R-54	Remote Call Forwarding Control	1958
R-55	Remote Call Hold	1963
R-57D	Ringing Line Pickup -D	1967
R-58	RTP Information Output	1969
R-60	Redial	1985
R-61	Retention of Customized Terminal Settings	1989
R-62	Ring Volume in Headset Mode	1997
R-63	Real-Time Call Information on Softphone	1999
S-1	Single-Digit Station Calling	2002
S-2	Special Dial Tone	2004
S-3	Speed Calling - System	2005
S-4	Splitting	2008
S-4D	Splitting -D	2009

TABLE OF CONTENTS (CONTINUED)

		Page
S-6	Station-Controlled Conference	2010
S-7	Station Hunting - Circular	2014
S-8	Station Hunting - Secretarial	2028
S-9	Station Hunting - Terminal	2030
S-10	Station Message Detail Recording (SMDR)	2044
S-11	Station-to-Station Calling	2090
S-12	Station-to-Station Calling - Operator Assistance	2092
S-13	Step Call	2094
S-15	Serial Call	2096
S-19	Single-Digit Feature Code	2099
S-21	Speed Calling - Station	2100
S-23	Speed Calling - Group	2105
S-24D	Save And Repeat -D	2109
S-25D	Service Display -D	2111
S-26D	Speed Calling - One Touch -D	2112
S-27	Service Feature Peg Count	2117
S-31	Supervisory Call	2119
S-33	Software Line Appearance	2121
S-46	Sender/Mark - Out Signal	2123
S-48	Station Message Detail Recording - Printout	2125
S-61	Speed Calling Override - System	2129
S-62	Station 5db PAD	2131
S-64	Serial Call - Loop Release	2132
S-65	Short Break Hooking	2133
S-99	Split Call Forwarding	2134
S-106	Slumber Time - Do Not Disturb	2143
S-127D	Serial Call -D	2146
S-129	SMDR Output Expansion - ANI/CPN	2150
S-137	Selectable Station Call Pickup	2155
S-138	Station Hunting - Priority	2162
S-139	Station Hunting - Priority with Switchback	2175
S-140D	Side Tone On/Off -D	2182
S-141	Survivable Remote MGC	2185
S-143	Software Multiple My Line Appearance	2251
S-144	SMDR Multi Area ID via IP	2259
S-147	Soft Phone Wired-Phone Handset	2267
S-148	SNMP	2282
S-150	System Message Report	2353
S-151	SIP Instant Message	2356
S-152	SIP Terminal Authentication	2359
S-154	SIP Wireless Terminal Backup	2365
S-155	Soft Phone	2368
S-157	Selective Disconnect 8 Party Conference	2378
S-159	Soft Phone WLAN Handset	2382
S-160	SIP Multiple Line Operation	2400
S-162	Selective Dialing	2405
S-165	Security Control Setup	2410
S-166	Software Digital Tone Generator	2414
S-167	SIP Handler - SIP Terminal	2419

TABLE OF CONTENTS (CONTINUED)

Page

S-168 SP Controlled SIP Terminal 2447
S-169 SIP Handler - SIP Terminal Support for TLS Encryption 2460
T-1 Tandem Switching of TIE Trunks - 2/4-Wire 2469
T-2 Three-Way Calling 2472
T-3 TIE Line Access 2479
T-5 TIE Line Connection With Pad Control 2481
T-6 Toll Denial/Toll Diversion 2483
T-7 Toll Restriction - 3/6-Digit 2501
T-8 Trunk Answer from Any Station (TAS) 2509
T-9 Trunk Group Busy Lamp 2512
T-10 Trunk-to-Trunk Connection 2513
T-12 Tenant Service 2516
T-18 Time Display 2520
T-18D Time Display -D 2521
T-23 Trunk Line Appearance 2523
T-23D Trunk Line Appearance -D 2526
T-28 Trunk-to-Trunk Third-Party Cancellation 2531
T-62 Timed Reminder 2533
T-63 Tone Block 2536
T-64 Terminal Login via NAT 2538
T-65 Terminal Claim Handling 2540
T-66 Terminal Remote Login 2552
T-67 Timed Forced Disconnect Trunk 2557
T-68 Time-based Routing for Multiple Time Zones 2559
T-69 Terminal Power Save Mode 2563
T-86 Timing Start 2572
U-1 Uniform Call Distribution (UCD) 2573
U-3 Universal Sender 2587
V-1 Variable Timing Parameters 2588
V-2 Voice Call 2591
V-3D Volume Control -D 2595
V-19 Voice Mail Password Display Elimination 2596
V-26 VS32 Conference Server 2597
V-27 VoIP Encryption 2652
V-29 Voice Mail Indication - SIP 2662
V-30 Volume Set - Receiver 2666
V-31 Video Call 2668
W-1 WATS Access 2675
W-11 WLAN Handset Name Display 2677

APPENDIX APP-1

APPENDIX A Tone Specifications for Each Country APP-2

APPENDIX B Tone Setting Differences between EMEA and Basic Tone Setting for EMEA APP-14

APPENDIX C Reference Information for Customizing/Editing Your Language Files through Optional Language Pack Loading APP-26

CHAPTER 1
INTRODUCTION

1. General

This manual covers data assignment for UNIVERGE SV9500 system (in the remainder of this manual, denoted as "the system") under the following headings:

CHAPTER 1 INTRODUCTION
Explains the configuration of this manual.
CHAPTER 2 ASSIGNMENT OF BASIC OFFICE DATA
Explains basic data assignment (minimum requirements) for the system.
CHAPTER 3 BUSINESS FEATURES
Gives details on each Business Feature.

2. How to Use This Manual

This manual may be used when assigning each service data available for the system. First, assign the basic office data by referring to Chapter 2. Then assign each service feature data as shown in Chapter 3.

3. Abbreviations for System Types

In this manual, SV9500, SV8500, SV7000, NEAX IPX, NEAX IMX, SV9300, SV8300, NEAX IPS, NEAX IVS^2, or NEAX IVS are used as abbreviations for each System type instead of the actual product name according to market country. The correspondences between the abbreviation of each System type and actual product name are described in the table shown below.

SYSTEM TYPE		REMARKS
SV9500	UNIVERGE SV9500	
SV8500	UNIVERGE SV8500	
SV7000	UNIVERGE SV7000	
NEAX IPX	UNIVERGE NEAX2400 IPX/NEAX2400 IPX	
NEAX IMX	<North America> NEAX2400 IMX	
	<Australia> NEAX7400 IMX	
	<Asia, Latin America, Middle East, Russia> NEAX7400 ICS (IMX)	
SV9300	UNIVERGE SV9300	
SV8300	UNIVERGE SV8300	
NEAX IPS	UNIVERGE NEAX2000 IPS/NEAX2000 IPS	

SYSTEM TYPE		REMARKS
NEAX IVS^2	<North America & Australia> NEAX2000 IVS^2	
	<UK> NEAX2000 INTEGRATED VOICE SERVER	
	<Asia, Latin America, Middle East, Russia> NEAX7400 ICS M100MX	
NEAX IVS	<Australia> NEAX7400 ICS Model120 NEAX7400 ICS Model110	
	<Asia, Latin America, Middle East, Russia> NEAX7400 ICS Model100 NEAX7400 ICS Model80VS	

CHAPTER 2
ASSIGNMENT OF BASIC OFFICE DATA

This CHAPTER outlines the following basic Office Data assignment.

1. Tenant Data
2. Trunk Data
3. Attendant Console (ATTCON) Data
4. Station Data
5. Desktop Terminal Data
6. Telephone Number
7. OGC/OGCA
8. LCR/LCRS
9. Uniform Numbering
10. PH Data Setting
11. Tone Setting
12. Data Setting for Switch Redundancy Configuration
13. Day/Night Data Retention
14. IP Related Data Setting
15. UG50 Related Data Setting

1. Tenant Data

1.1 Use of Tenant Service

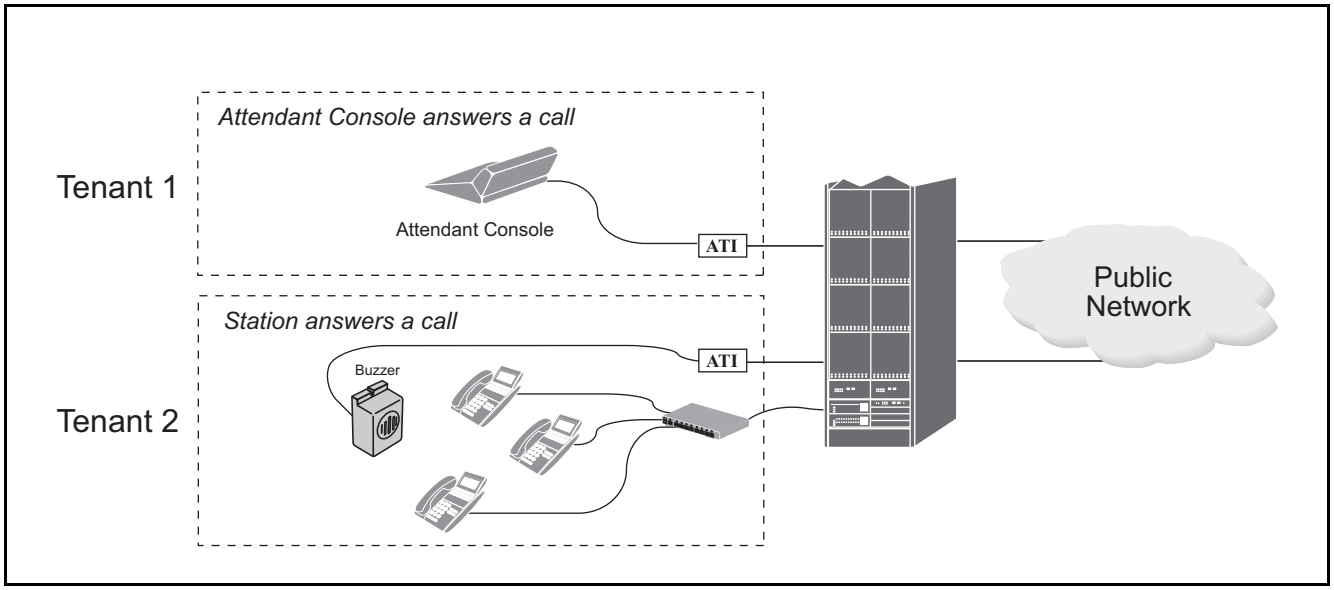
A system can be used with multiple tenants (TN) such as companies, branches, or sections. The C.O. line, the TIE line, or an attendant console can be used for different purpose for each tenant, and also some of the Office Data (such as Numbering Plan, Call Connection Restriction, and External Trunk Routing) can be configured separately for each tenant. This section explains the basic concept of Tenant service by using examples.

1.2 Example of Tenant Development

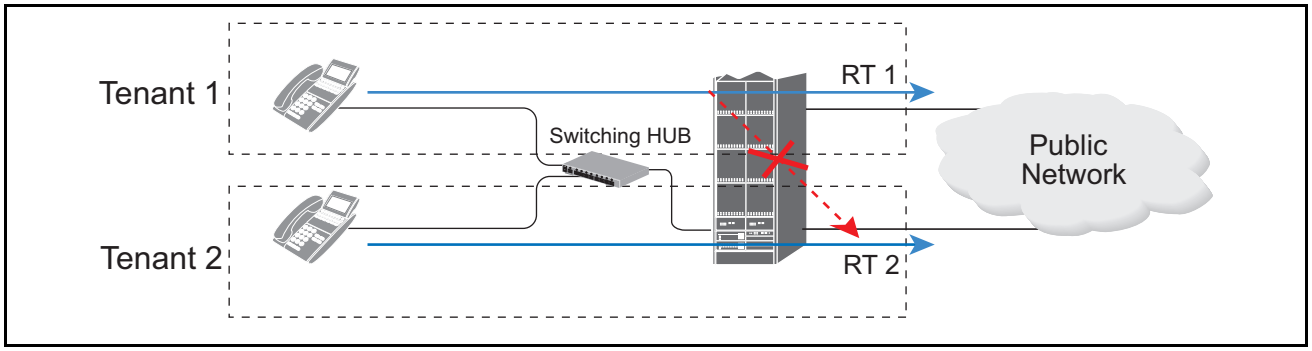
A System can have a maximum of 63 Tenants (TN 1 - 63), except Tenant 0 for ID code registration. The following are some examples of tenant development.

Note: System Initialization is required after changing the maximum number of Tenants.

Example 1: Tenant 1, answering Incoming Central Office Line Call via Attendant Console
Tenant 2, answering Incoming Central Office Line Call via Trunk Answer from Any Station (TAS) indicator

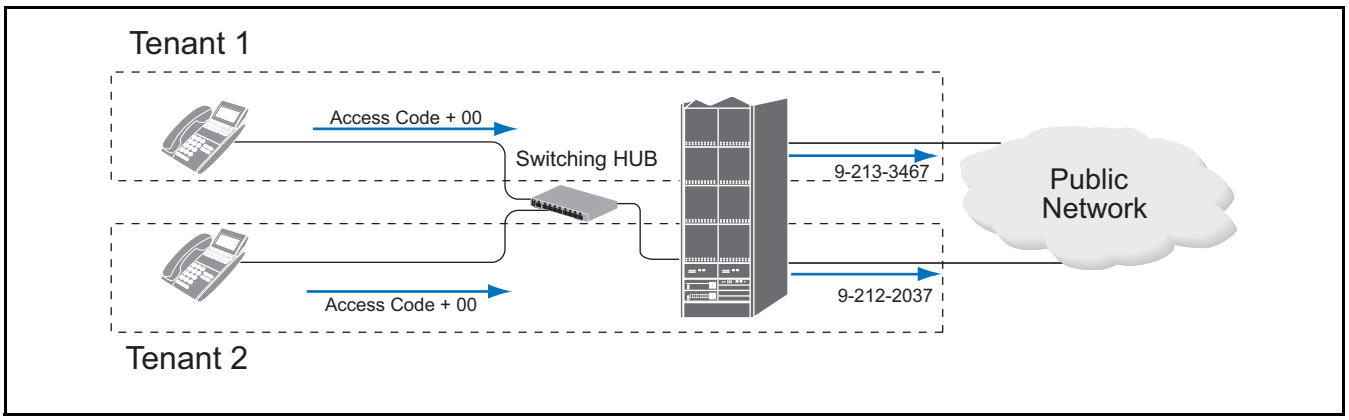


Example 2: Tenant 1, placing Central Office Outgoing Call via RT 1
Tenant 2, placing Central Office Outgoing Call via RT 2

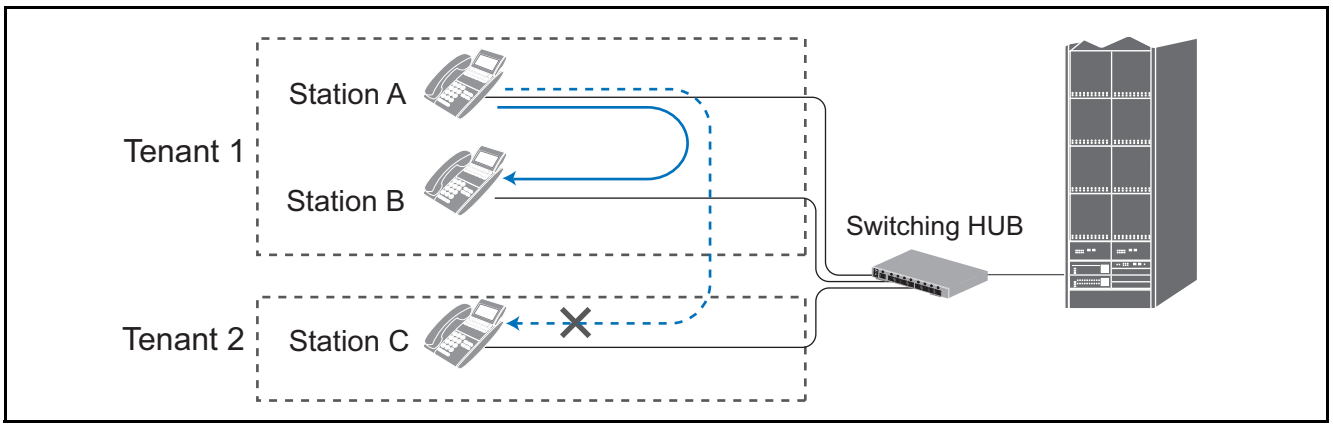


Example 3: Tenant 1, "Speed Calling Access Code (ACC) + 00" → "9-213-3467"
Tenant 2, "Speed Calling Access Code (ACC) + 00" → "9-212-2037"

Note: Speed Calling data can be developed differently for each Tenant.



Example 4: Inter-Tenant Call (between Station A and Station C) is restricted.



1.3 Office Data related to Tenant Service

A list of Office Data, related to Tenant Service, is shown below.
Assign whether to separate Tenants for these commands by ASYD, SYS1, Index 92, Index 93, Index 94 and ASYDL/ASYDN, SYS1, Index 800, Index 801, Index 802.

When changing these Office Data from common to separate, and vice versa, first delete all the currently assigned data by using appropriate commands, change the system data, and then assign new data.

Note: The data assigned by ASPAL command can be deleted collectively by using CSPAL command. Use CSPAN command to delete the data assigned by ASPAN command collectively.

Command Name		Setting of ASYD/ ASYDL/ ASYDN	Description
ASYD, SYS2	Assignment of System Data-2	Index 92, Bit 0=0	ASYD, SYS2, can be developed independently for each Tenant (TN 1-63). SYS2 is associated with the following service: • Busy service • Attendant service • Call Forwarding Service • SMDR (Station Message Detail Recording) service etc.
		Index 92, Bit 0=1	ASYD, SYS2, is assigned for the common Tenant (TN 1).
ANPD/ ANPDL/ ANPDN	Assignment of Numbering Plan Data	Index 92, Bit 2=0	"1stDC (first digit of Access Code)" for each feature access and cancel can be developed independently for each Tenant (TN 1-63).
		Index 92, Bit 2=1	"1stDC" for each feature access and cancel is assigned for the common Tenant (TN 1).
ASPA/ ASPAL/ ASPAN	Assignment of Special Access Code	Index 92, Bit 1=0	"ACC (Access Code)" for each feature access and cancel can be developed independently for each Tenant (TN 1-63).
		Index 92, Bit 1=1	"ACC" for each feature access and cancel is assigned for the common Tenant (TN 1).
ATRK	Assignment of Trunk Data		While a RT (the same Route Number) can be shared by multiple Tenants, the respective Trunk Number of the RT cannot be duplicated for multiple Tenants.
ARSC	Assignment of Route Restriction Class	Index 92, Bit 4=0	RSC (Route Restriction Class) tables, determining Incoming/Outgoing Call restriction for Stations or Attendant Consoles, can be developed independently for each Tenant (TN 1-63).
		Index 92, Bit 4=1	RSC table is assigned for the common Tenant (TN 1).
ACFR/ ACFRN	Assignment of Call Forwarding Restriction	Index 92, Bit 5=0	Various kinds of Call Forwarding Restriction, such as DIT (Direct-In Termination), Night Attendant, Remote Access to System and Call Forwarding - All Calls/Busy Line/Don't Answer, etc., can be developed independently for each Tenant (TN 1-63).
		Index 92, Bit 5=1	Various kinds of Call Forwarding Restriction are assigned for the common Tenant (TN 1).
ASFC	Assignment of Service Feature Restriction Class	Index 93, Bit 0=0	SFC (Service Feature Restriction Class) table, determining permission/ restriction of each service feature on a Service Feature Restriction Class basis, can be developed independently for each Tenant (TN 1-63).
		Index 93, Bit 0=1	SFC table is assigned for the common Tenant (TN 1).

Command Name		Setting of ASYD/ ASYDL/ ASYDN	Description
ACFO	Assignment of Call Forwarding Data	Index 93, Bit 1=0	Various Call Forwarding Destination Table can be developed independently for each Tenant (TN 1-63).
		Index 93, Bit 1=1	Various Call Forwarding Destination Table is assigned for the common Tenant (TN 1).
ASPD/ ASPDN	Assignment of Speed Calling	Index 93, Bit 3=0	Abbreviated Digit Code data table (for Speed Calling-System feature) can be developed independently for each Tenant (TN 1-63).
		Index 93, Bit 3=1	Abbreviated Digit Code Data Table is assigned for the common Tenant (TN 1).
ASTP/ ASTPL/ ASTPN	Assignment of Signal Translation Pattern	Index 93, Bit 4=0	PNI/PNO tables, to be elaborated by AISP/AOSP command, can be developed independently for each Tenant (TN 1-63). PNI: Incoming Selection Translation Pattern Number PNO: Outgoing Selection Translation Pattern Number
		Index 93, Bit 4=1	PNI/PNO table is assigned for the common Tenant (TN 1).
AFRS/ AFRSL/ AFRSN	Assignment of Flexible Route Selection Data	Index 93, Bit 4=0	OPR (Outgoing Route Selection Pattern Number) tables, to be programmed by AOPR command, can be developed independently for each Tenant (TN 1-63).
		Index 93, Bit 4=1	OPR tables are specified for the common Tenant (TN 1).
AAED/ AAEDL/ AAEDN	Assignment of Announcement Equipment Data	Index 93, Bit 7=0	DAT (Digital Announcement Trunk) data table can be developed independently for each Tenant (TN 1-63).
		Index 93, Bit 7=1	DAT data table is assigned for the common Tenant (TN 1).
ATNR	Assignment of Tenant Restriction Class Data	Index 94, Bit 0=0	Allow/Deny table for Inter- and Intra-Tenant events, such as Station-to-Station Call between Tenants or service activation (Call Forwarding Service) across Tenants, can be developed independently for each Tenant (TN 1-63).
		Index 94, Bit 0=1	Allow/Deny table for Inter- and Intra-Tenant events is assigned for the common Tenant (TN 1).
AEFR/ AEFRN	Assignment of EPN Facility Restriction Data	Index 94, Bit 1=0	Allow/Deny tables for Call establishment between RSC1 (Calling Party's Restriction Class) and RSC2 (Called Party Restriction Class) can be developed independently for each Tenant (TN 1-63).
		Index 94, Bit 1=1	Allow/Deny tables for Call establishment between RSC1 and RSC2 are assigned for the common Tenant (TN 1).
APCR/ APCRN	Assignment of Primary Call Restriction Data	Index 94, Bit 2=0	Allow/Deny table for Call Origination to specific Area Code(s) can be developed independently for each Tenant (TN 1-63).
		Index 94, Bit 2=1	Allow/Deny table for Call Origination to specific Area Code(s) is assigned for the common Tenant (TN 1).
AMND/ AMNDL/ AMNDN	Assignment of Maximum Necessary Digits Data	Index 93, Bit 6=0	Tables of MND (maximum necessary digits) data, to be received by each Register (ORT/IRT) Circuit, can be developed independently for each Tenant (TN 1-63).
		Index 93, Bit 6=1	Tables of MND data are assigned for the common Tenant (TN 1).
AATC/ AATCN	Assignment of Authorization Code Data	Index 94, Bit 3=0	Authorization Code Table, to be used by Authorization Code, Forced Account Code, Pad Lock and Remote Access to System features, can be developed independently for each Tenant (TN 1-63).
		Index 94, Bit 3=1	Authorization Code Table is assigned for the common Tenant (TN 1).
APCNL/ APCNN	PHS community information (LDM)	Index 800, Bit 3=1	This command is used to assign PHS community number for each corresponding tenant.

Command Name		Setting of ASYD/ ASYDL/ ASYDN	Description
AUNE/ AUNEL/ AUNEN	Uniform Numbering (LDM)	Index 800, Bit 4=1	When a self office number is received in each tenant, this command is used for deleting its office number. It can also add number after deleting it.
AUGNN	UGN Information (NDM)	Index 801, Bit 2=1	This command is used to assign/delete UGN (User Group Number) for NPC (Number Pattern Code).
APBKN	System Phone Book (NDM)	Index 802, Bit 3=1	This command assign/modify/delete name and telephone number for the System Phone Book.
ATD-PEL/ ATDPEN	Extended Toll Code Restriction Data (LDM)	Index 802, Bit 1=1	This command restricts /allows toll call by developing Tenant Number (TN) and Destination Code (DC) for PSTN connection (including tandem connection via intermediate node).

The following data cannot be developed for multiple Tenants.

Command Name		Description
APSD	Assignment of PS Operation Data	A Station Number cannot be duplicated in a single system. (Tenants cannot be developed.)
ASDT	Assignment of Station Data	A Station Number cannot be duplicated in a single system. (Tenants cannot be developed.)
ASAT	Assignment of specific Attendant Number Data	The same Individual Attendant Number cannot be duplicated in a single system. (Tenants cannot be developed.)
ALDN	Assignment of Listed Directory Number	The same LDN Number cannot be duplicated in a single system. (Tenants cannot be developed.)

Note: When reassigning data from separate tenants to common tenants, or vice versa, preassigned data must be completely deleted first.

Note: When changing the maximum number of tenants, System Initialization is necessary.

1.4 Examples of Data Programming

The following are the data programming for Examples 1, 2, 4 of Section 2.2.

STEP 1: **ASYD** - SYS1, Index 8: Number of Tenants (01 - 3F hexadecimal)
Index 92, Bit 0: Tenant data table development for System Data 2 (SYS2)
(0/1: Multiple Tenants are used/Common Tenant (TN 1) is used.)
Index 92, Bit 1: Tenant data table development for Special Access Codes
(0/1: Multiple Tenants are used/Common Tenant (TN 1) is used.)
Index 92, Bit 2: Tenant data table development for Numbering Plan
(0/1: Multiple Tenants are used/Common Tenant (TN 1) is used.)
Index 92, Bit 4: Tenant data table development for Route Restriction Class (RSC)
(0/1: Multiple Tenants are used/Common Tenant (TN 1) is used.)
Index 94, Bit 0: Tenant data table development for Tenant Restriction Class
(0/1: Multiple Tenants are used/Common Tenant (TN 1) is used.)

<Necessary data for Example 1, ATT answer for Central Office Call>
SYS2, Index 0, 2, 4, 5, 6, 7, 8, 9, 10, 11, 14

STEP 2: **ANPD** - Assign 1st Digit Code (1stDC) and Number of Necessary Digits (NND) for Central Office Trunk Access. Then, also assign 1stDC and NND for ATT Answer (Operator Call)/TAS Answer.
When multiple Tenants are to be used (ASYD, SYS1, Index 92, Bit 2=0), develop respective "1stDC" and "NND" for each Tenant. If not (Index 92, Bit 2=1), develop only for Tenant 1.

STEP 3: **ASPA** -Assign Access Code (ACC) for Central Office Trunk Access, "ACC" for ATT Answer (Operator Call) and "ACC" for TAS Answer.
When multiple Tenants are to be used (ASYD, SYS1, Index 92, Bit 1=0), develop respective "1st DC" and "NND" for each Tenant. If not (Index 92, Bit 1=1), develop only for Tenant 1. In case of Operator Call, the ACC should be assigned only to the Tenant that was listed first in the Tenant data table of ATRK.

STEP 4: **ARSC** - Assign permission/restriction tables for Incoming/Outgoing Calls on a Route Restriction Class (RSC) basis.
When multiple Tenants are to be used (ASYD, SYS1, Index 92, Bit 4=0), develop respective tables for each Tenant. If not (Index 92, Bit 4=1), develop only for Tenant 1.

Note: RSC may be broken down into a maximum of 16 groups (RSC 0 for ATT, RSC1 - 15 for Stations), by using ASDT/ATRK command.

STEP 5: **ASFC** - Assign a Service Feature Restriction Class that allows SFI 13, to the Stations used for "TAS Answer". Data must be "1".

STEP 6: **ATNR** - Assign connection restriction data between Originating Tenant (OGTN) and Terminating Tenant (TMTN).
TRI 0:Restriction/permission of Station-to-Station connection between Tenants.
TRI 4:Restriction/permission of the Individual Attendant Access/Inter-Position Transfer feature (Calling a specific ATT from Station/ATT in an different Tenant, by dialing individual ATT Number).
When multiple Tenants are to be used (ASYD, SYS1, Index 94, Bit 0=0), develop each data for each OGTN and TMTN. If not (Index 94, Bit 0=1), develop only for OGTN 1/TMTN 1.

STEP 7: **ASDT** - Assign Tenant Number and Route Restriction Class (RSC) Number to which each Station belong.
TN:Tenant Number (1-63)
RSC:Route Restriction Class (1-15)

STEP 8: **ARTD** - Assign Route Class Data to all the Route Numbers used for Central Office Call.

Note: To change the data for existing route (RT), initialize or reboot the trunk device such as circuit card, MG, or VS32.

STEP 9: **ATRK** - "RT (External or Internal Route Number)" and "TK (Number Trunk)" Number and its respective "TN (Tenant)" number, for all the Routes assigned by ARTD command. For each Trunk of ATT, specify TK, RSC (RSC 0) and all the TN that may use the ATT.

STEP 10: **MBTK** - Set each Trunk Port idle (serviceable) by using this command.

2. Trunk Data

2.1 Trunk

In addition to basic two-way connection between Stations, the System may handle various types of other connections, such as between a Station (Self Office) and a party in distant office (TIE Line/Public Network). When these connections are required, the system uses any of the following "Trunk" interfaces:

- External Trunk - A link channel between Self Office and other office (such as COT, ODT, DTI, CCT, PRT, etc.)
- Internal Trunk - Other function channel, such as Register (ORT, IRT), Sender (SND), Attendant Console Interface (ATI), Three-Party Conference Trunk (CFT), etc.

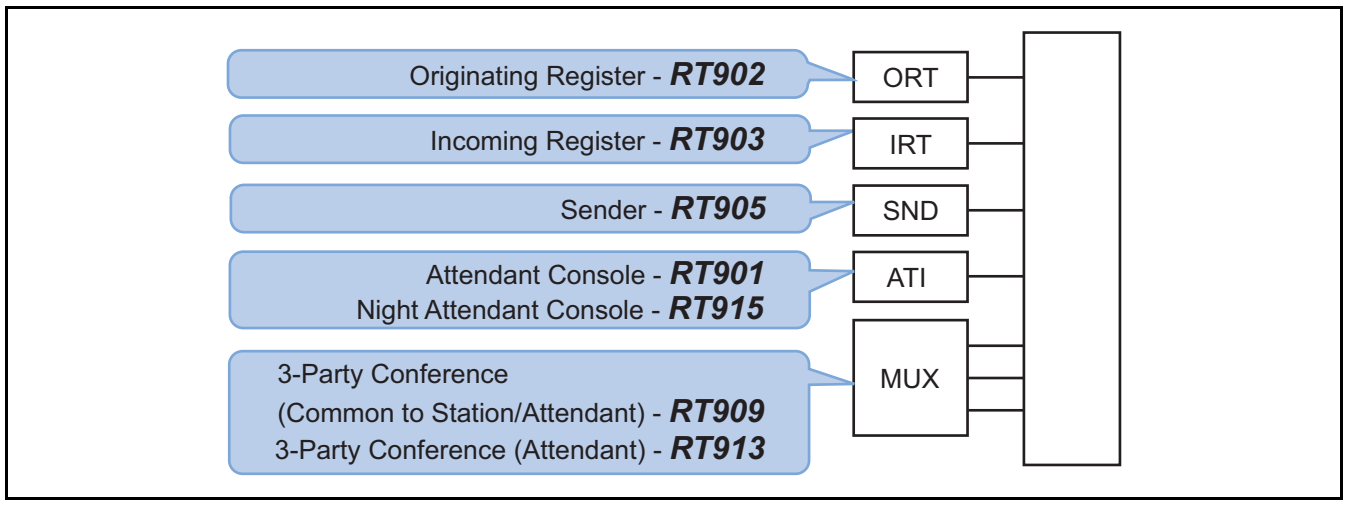
This section explains basic data assignment of these Trunks.

2.2 Internal Trunk

Each set of data for Internal Trunks is programmed by ATRK. Assign a Route Number (defined as "9xx" according to the Trunk Type), its Trunk Numbers and accommodated LENS (Line Equipment Numbers), following the command display.

Programming

STEP 1: **ATRK** - RT: Route Number (9xx)
Enter the defined Route Number, according to the Internal Trunk type. Refer to the following figure and table:



<Main Internal Trunk>

Definition of Route Number	Kind of Internal Trunk	Related Circuit Card (Symbol)	Maximum Capacity Note 1
901	ATT	ATI	60/System Note 2
902	ORT	RST	64/4 PIRs Note 3 Note 4
903	IRT	RST	64/4 PIRs Note 3 Note 4
905	SND	RST	64/4 PIRs Note 3 Note 5
906	PBR for Automated Attendant Service	RST	64/4 PIRs Note 3 Note 4
907	AMP	AMP	Limitless
909	DCFT	MUX TSW (PIR 0)	16/MG Note 6
913	DCFT for ATT	MUX TSW (PIR 0)	16/MG Note 6
915	Night ATT	ATI	60/System Note 2
933	CRT	RST	64/4 PIRs Note 3 Note 4
937	Caller ID Modem Sender Trunk Note 7	RST	64/4 PIRs Note 3 Note 5
940	Call Waiting Caller ID Register Trunk	RST	64/4 PIRs

Note: For other Internal Trunks, refer to the Command Manual.

Note 1: This field specifies the Physical Trunk capacity the system can register. No consideration is given to the circuit card mounting condition.

Note 2: Specifies total capacity of RT 901 and 915.

Note 3: For incoming/originating calls via ISDN lines, virtual registers/senders are used.

Note 4: The sum total of virtual registers and other registers (RT=902, RT=903, RT=906, and RT=933) per LP cannot exceed 64 channels. If all the channels are occupied by RT=902, RT=903, RT=906, and RT=933, there is no channel left for virtual registers to use. Be careful because this can cause incoming calls to ISDN lines unavailable.

Note 5: The sum total of virtual senders and other senders (T=905 and RT=937) per LP cannot exceed 64 channels. If all the channels are occupied by RT=905 and RT=937, there is no channel left for virtual senders to use. Be careful because this can cause originating calls from ISDN lines unavailable.

Note 6: Specifies total capacity of RT 909 and 913.

Note 7: This data is assignable only to Sender (SND) Trunk for Analog Caller ID-Station by Modem Sender.

TK: Trunk Number (1-255)
Enter the Trunk Number ranging from 1 to 255.
**When the "TK" is for Attendant Console (ATT), the number, assigned here, is used as the Attendant Number (ATN) for the AMAT/ASAT command.*

LENS: Line Equipment Numbers
Enter the LENS for each assigned Trunk. The LENS specifies accommodated location of the respective Trunk, by a six-digit (MG, U, G, Lv) number. For details, refer to the Operations and Maintenance Manual.

< For ATT or Night ATT Interface>
For ATT or Night ATT Interface, the following data must also be assigned:

CNT: Count (1-10)
Assign the total number of Tenants that may use the respective ATT. Note that allowed Tenants per ATT is a maximum of ten in total.

TN: Tenant Number
Assign all the Tenant Number, one by one, that may use the respective ATT.

RSC: Route Restriction Class
Assign "0" for this parameter (RSC for ATT is 0 only)

SFC: Service Feature Restriction Class
Assign "0" for this parameter (SFC for ATT is 0 only)

STEP 2: **MBTK** - Set each assigned Trunk Port Idle (in service) by using this command.

Attention: *When deleting the once-assigned Trunk Data for an ATT, be sure to place the corresponding ATI Card into a make-busy state, then perform the command operation.*

2.3 External Trunk

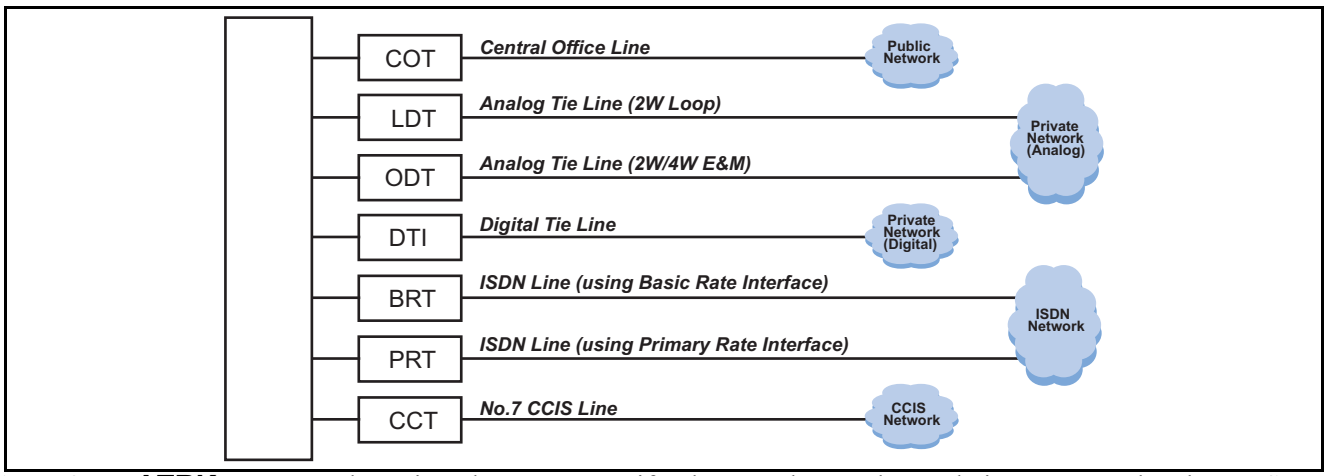
Data assignment of External Trunk is performed by using ASYD, ARTD, and ATRK. The main flow of data setting is shown below.

Programming

STEP 1: **ASYD** - SYS1, Index 65.
Assign the maximum number of Routes (External Routes) that may be used in the system in hexadecimal (01-FF). This data is necessary for the system to determine the range of Traffic Measurement Target Routes.

STEP 2: **ARTD** - Assign Route Numbers and their respective attribute data, following the command display. Any Route, within the range specified by ASYD, SYS1, Index 65, can be entered uniquely. For detailed parameters and each input data definition, refer to the Command Manual.

Note: To change the data for existing route (RT), initialize or reboot the trunk device such as circuit card or MG.



STEP 3: **ATRK** - For each assigned Route, specify the Trunk Numbers, their accommodated LENS and belonging Tenant.
RT: Route Number
Enter the Route Numbers that were specified by ARTD. For each Route Number, detailed Trunks are to be assigned in the parameters, TK, LENS and TN below.
TK: Trunk Number (1-255)
For each Route assigned in the RT parameter, set the Trunk Numbers ranging from 1 to 255.
LENS: Line Equipment Numbers
Enter the LENS for each assigned Trunk. The LENS specifies accommodated location of the respective Trunk, by 6-digit (MG, U, G, Lv) number. For details, refer to the Operations and Maintenance Manual.
TN: Tenant Number (1-63)
For each Trunk, assign the Tenant Number to belong to. Note that, although a Route can be used for multiple Tenants, each Trunk of the Route cannot be duplicated at multiple Tenants.

Ex.)

RT	TK	TN	RT	TK	TN
10	1	1	10	31	2
	⋮			⋮	
10	30	1	10	120	2

STEP 4: **MBTK** - Set each assigned Trunk Port Idle (in service) by using this command.

3. Attendant Console (ATTCON) Data

3.1 Attendant Console

The Attendant Console (ATTCON) is used by the operator to answer an Incoming Call, then facilitate the connection by inward transfer, or place Outgoing (External Trunk) Calls, through each switched loop (generally, each ATTCON has six loops). This section explains basic data assignment of the ATTCON.

3.2 Assignment of ATTCON Data

Programming

STEP 1: **ASYD** - SYS1, Index 6, Bit 4 to Bit 6

Assign the Key Pattern for Attendant Console.

Bit 6	Bit 5	Bit 4
0	0	0=Pattern 1
0	0	1=Pattern 2
0	1	0=Pattern 3
1	0	0=Pattern 4

Note: For details on each pattern face layout, refer to the Command Manual.

SYS1, Index 9
Assign the number of Attendant Consoles to be used, including the Night ATT.

SYS1, Index 70, Bit 5
Assign priority to the call to be answered by ATTCON "ANSWER" Key.
0: Depends on type of call (Central Office Call has priority)
1: By call termination order

SYS1, Index 73, Bit 7
Assign whether the ATT uses the loop-to-loop connection function for the Meet-Me Paging - Attendant feature.
0: Loop-to-loop connection is not used.
1: Loop-to-loop connection is used.

* *Meet-Me Paging-Attendant is a feature that allows ATT to hold an Incoming Call, page the Called Party, and connect the two after the Called Party has answered the page. Of these operations, the last two-party connection is loop-to-loop connection.*

SYS1, Index 82
Bit 0 to Bit 3: Assign the Station Number Display Pattern on the ATTCON.
Bit 4 to Bit 7: Assign the Station Number Display Pattern for CCIS on the ATTCON.

Note: For details on each pattern, please refer to the Command Manual.

SYS1, Index 141
Assign the timer for recalling ATTCON, after the ATTCON activates Blind Transfer (ATT transfers a call, and then hangs up without response) to a Station that is setting Call Forwarding - Don't Answer.

* *The default value (00 hexadecimal) is ten seconds. Refer to the Command Manual.*

SYS1, Index 155
Assign the Sender prepause timer.
* *The default value (00 hexadecimal) is four seconds. Refer to the Command Manual.*

SYS1, Index 229, Bit 2
Specify the necessity for sending DTMF (PB) signal as a result of Key Pad Operation on the ATTCON, while speaking.
0: Not necessary
1: Necessary

SYS1, Index 246, Bit 0
Assign whether ATTCON can answer the next call, by placing the current call on hold by pressing either the ANSWER Key or Incoming Call Identification (ICI) Key, while in two-party connection with Station/Trunk.
0: Unable to answer
1: Able to answer

SYS1, Index 246, Bit 1
Assign whether ATTCON can answer the next call, by placing the current call on hold by pressing either ANSWER Key or Incoming Call Identification (ICI) Key, while in three-party connection with Station/Trunks.
0: Unable to answer
1: Able to answer

SYS1, Index 257, Bit 0
Assign the mounting condition of DLKC circuit card. Data must be “1 (fixed)”.

SYS2, Index 2, Bit 1
Specify the Sender signal (ATTCON) for DP/PB Route assigned by ONSG (CDN: 2) of ARTD.
0: DP
1: PB

SYS2, Index 2, Bit 2 to Bit 3
ATTCON Day/Night mode change is:

Bit 3	Bit 2
0	0=Not required (External Switch effective)
0	1=by Primary Attendant Console
1	0=by all Attendant Console
1	1=–

SYS2, Index 6, Bit 7
Specify either recalling ATTCON or activating Call Forwarding Don’t Answer, when a call is transferred (by ATT) to the Station provided with Call Forwarding Don’t Answer.
0: ATTCON is recalled
1: Call Forwarding Don’t Answer is activated

SYS2, Index 8 and Index 9

Assign detailed kinds of call to be counted as waiting calls on the display of the ATTCON. Refer to the Command Manual for details.

STEP 2: **ANPD** -Assign 1st Digit Code (1stDC) and Number of Necessary Digits (NND) for ATTCON access.
1stDC: 0
NND:1

STEP 3: **ASPA** -Elaborate full Access Code (ACC) for ATTCON access SRV (Kind of Service) = "SSC: Service Code", SID (Service Index) = 2 [Operator Call].
A sample data assignment for "ACC=0" is shown below:
TN = 1
ACC = 0
CI = N: Normal, H: Hooking
SRV = SSC: Service Code
SID = 2 (Operator Call)

Note: For Connection Index (CI), specify what state a Station may dial the designated Access Code (here, Operator Access Code "0"). When "N" is assigned, the Station can access ATTCON by dialing the Access Code in a normal state. When "H" is assigned, the Station can do so after making switch hook flash (SHF).

Attention: *The following shows a condition when using "#" as one of the dial numbers:*

- *After dialing ACC assigned by ASPA, the system recognizes "#" code as the identifier for Dialed Number completion. Therefore, for example, when "8# (ACC) + 2#2500 (Calling Number)" is dialed, only "2" before the "#" code is recognized as the Calling Number ("2500" after "#" code is not recognized by the system.).*

STEP 4: **ATRK** - Assign the Trunk (accommodated location) for each ATT
RT: Assign the defined Route Number out of the following:
RT = 901: ATT
RT = 915: Night ATT
RT = 913: Digital Conference Trunk (DCFT) for ATT
TK: Assign Trunk Numbers for designated "RT".
LENS:Assign the LENS (Line Equipment Numbers) for each "TK". The LENS specifies accommodated location of the respective Trunk, by 6-digit (MG, U, G, Lv) number. For details, refer to the Operation and Maintenance Manual.
CNT:Assign the total number of Tenants that may use the respective ATT.
Note that allowed Tenants per ATT is a maximum of 20.
TN:Assign all the Tenant Numbers, one by one, that may use the respective ATT.
RSC: Assign "0" for Route Restriction Class (RSC). (RSC for ATT is 0 only)
SFC:Assign "0" for Service Feature Restriction Class (SFC). (SFC for ATT is 0 only)

STEP 5: **MBTK** - Set each assigned Trunk Port Idle (in service) by using this command.

STEP 6: **AMAT** - When Day/Night mode of ATTCON is to be controlled by Primary Console (ASYD, SYS2, Index 2, Bit 2 and Bit 3), the Primary Console should be designated by this command.
MODE: Assign "1" for ATT (RT = 901), or "2" for Night ATT (RT = 915).
ATN:Assign Attendant No. of Master Console. (Assign the same number assigned in TK of ATRK.)

Attention: *When deleting already-assigned Trunk Data, be sure to place the corresponding ATI Card into make-busy state first, and then perform the command operation. When Attendant Number (ATN) is assigned on the ASAT command, be sure to delete this command data, too.*

4. Station Data

This section explains the basic data assignment of a station. For data assignment of a Desktop terminal station, refer to 5. Desktop Terminal Data in this chapter.

4.1 Assignment of Station Data

Main flow of data assignment for station is shown below:

Programming

<System Data>
Assign necessary System Data (ASYD Command), according to the pre-determined Station Numbering Plan. Refer to the following for the prerequisite minimum data:

STEP 1: **ASYD** - SYS1, Index 4, Bit 2 to Bit 3
Assign the Releasing Method for Station to Station Calling Service
00 = Calling Party Release
01 = Called Party Release
10 = First Party Release
11 = Both Party Release
10 = First Party Release" is usually assigned.

SYS1, Index 16
Assign the Number Level on Station Number composition
Bit 0:0/1 = one-digit Station Number is not used/used
Bit 1:0/1 = two-digit Station Number is not used/used
Bit 2:0/1 = three-digit Station Number is not used/used
Bit 3:0/1 = four-digit Station Number is not used/used
Bit 4:0/1 = five-digit Station Number is not used/used
Bit 5:0/1 = six-digit Station Number is not used/used (for Hotel system)

Note: A station number can be one to five digits. A six-digit number is only for the Hotel system.

SYS3, Index 0
Configure the Ring Cycle when a call from a Central Office Line terminates to a station. Normally, "21" should be assigned in hexadecimal (one-second ringing period, two-second silent period). Refer to Command Manual for details.

SYS3, Index 1
Configure the Ring Cycle when a station is called from another station/ATTCON (Ringer Pattern 1). Normally, "42" should be assigned in hexadecimal (two-second ringing period, four-second silent period). Refer to Command Manual for details.

SYS3, Index 2
Configure Station Release Timer (LT: Long Timer, ST: Short Timer). Normally, in North America, "91" should be assigned in hexadecimal (LT = 1,080 milliseconds, ST = 120 milliseconds). Refer to the Command Manual for details.

<Numbering Plan - Access Code Setting>

Program the Access Codes (ACC) for station access, by using the combination of ANPD and ASPA.
As described below, there are two patterns for these commands entry.
- Case 1: Assignment of 1stDC (1st digit code) for fixed-digit station numbers

When the number of total digits is assured for every station number that begins with the same 1stDC, the necessary data assignment is as follows:

STEP 1: **ANPD** -1stDC: Assign the 1stDC of the Station Numbers
NND: Assign the total Station Number digits

STEP 2: **ASPA** -ACC: Assign the Access Code using the "1stDC" specified by ANPD.

**For Case 1 here, assignment of "NND" for ASPA command is not required.*

- Case 2:Assignment of 1stDC (1st digit code) for mixed-digit pattern Station Numbers

When the digit composition of Station Numbers having the same 1stDC is not stable (Station Numbers of different digit compositions are mixed), necessary data assignment is as follows:

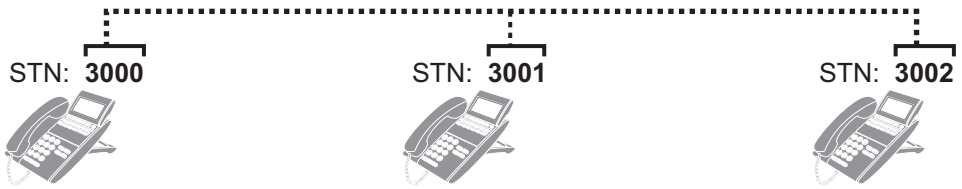
STEP 3: **ANPD** -1stDC: Assign the 1stDC of the Station Numbers
NND: Assign the total digit number of ACC to be specified by ASPA.

STEP 4: **ASPA** -ACC: Assign the Access Code using the 1stDC specified by ANPD. As the mixed-digit numbering for Case 2 uses several ACCs for a 1st C, assign one of the ACC and its corresponding NND (below) first, and then proceed with the next one repeating this command entry.
SRV: Assign "STN: Station Number" for Station Number Access
NND: Assign the total Station Number digits, associated with each ACC.

Note: For details, refer to the following figure.

Case 1: A fixed number of digits is used for every station number beginning with the same "1stDC"

A fixed number of digits (4 digits) is used for every station number beginning with "1stDC = 3"

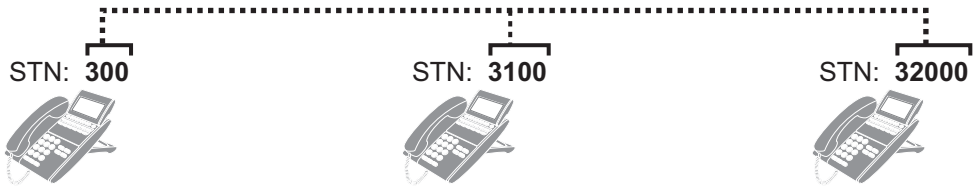


< Data Assignment >

ANPD	**1stDC = 3**	**NND = 4**
ASPA	**ACC = 3**	**SRV = STN**

Case 2: Different digit numbers are mixed for station numbers beginning with the same "1stDC"

3-digit, 4-digit and 5-digit numbers are mixed for station numbers beginning with "1stDC = 3"



< Data Assignment >

ANPD	**1stDC = 3**	**NND = 2 (the number of digits of "ACC" for ASPA)**	
ASPA	**ACC = 30**	**SRV = STN**	**NND = 3**
	ACC = 31	**SRV = STN**	**NND = 4**
	ACC = 32	**SRV = STN**	**NND = 5**

STN: Station Number
1stDC: 1st Digit Code

< Station Accommodation Data >
According to the specified Numbering Plan (in the previous step), assign the Station Numbers, their accommodated location, telephone equipment class, and belonging Tenant, etc. For this data, ASDT is used.

Note: For Station Accommodation data programming of IP Stations (such as IP Enabled Dterm or analog/Fax Stations connected to Analog MC.) the AISTL (IP Station Data) command must be used instead of the ASDT (Station Data).
- For data assignment of IP Enabled Dterm, refer to IP ENABLED Dterm [I-43] of Chapter 3.
- For data assignment of the stations (such as Analog/Fax Stations) connected to Analog MC, refer to Peripheral Equipment Description (IP Devices).
- For data assignment of stations accommodated in UG50, refer to Peripheral Equipment Description (UG50).

ASDT - For each Station, assign the following data.
TN: Belonging Tenant Number
STN: Station Number
LENS: Line Equipment Numbers (accommodated location of the Station)
TEC: Telephone Equipment Class (DP or PB may be chosen. Refer to Command Manual for details)
RSC: Route Restriction Class, 1- 15 (class grouping on Outgoing/Incoming Call Restriction, specified by ARSC command)
SFC: Service Feature Restriction Class, 1 - 15 (class grouping on allowed/denied service features, specified by ASFC)

4.2 Assignment of Station-to-Station Connection Data

This section explains basic data assignment for Station-to-Station connection.

Programming

STEP 1: **ATNR** - Assign the connection restriction data between Originating Tenant (OGTN) and Terminating Tenant (TMTN). When ASYD, SYS1, Index 94, Bit 0=0 is assigned (i.e. multiple Tenant service is in service), assign the respective Tenant Number for OGTN and TMTN. If not (when the data is "1"), assign the fixed Tenant Number of "1" for both "OGTN" and "TMTN".
TRI: Assign "0" for Station-to-Station Calling.
OGTN: Assign the number of Originating Tenant.
TMTN: Assign the number of Terminating Tenant.
RES: Assign "0" for restriction, or "1" for permission.

STEP 2: **ATRK, MBTK**

- In addition to the above data, assignment of Originating Register Trunk (ORT) is also necessary. For details, refer to 2. Trunk Data.

Note: For the Station-to-Outgoing Trunk Connection, allow the Route restriction for each Trunk by the ARSC command.

Note: When using ISDN/CCIS service, the number of 8RST circuit cards that are accommodated in a LP must be seven or less (for the usual system condition, the maximum number of "8RST circuit cards" that can be accommodated per LP is eight) for the allocation of the ISDN/CCIS Register/Sender memory. When a maximum of eight 8RST circuit cards in an LP is assigned, the ISDN/CCIS call cannot be originated or terminated normally.

5. Desktop Terminal Data

This section explains the following data related to the Desktop terminal stations:

- Multiple Lines
- Assignment of Desktop Terminal Station Data
- Display (LCD) Indications
- Line Pickup Preference
- Restriction of Call Termination on My Line
- Soft Key Assignment
- Expanded Ringer Pattern for Digital Terminal
- Headset Data
- Terminal Operations with Feature and Digit Keys for Digital Terminals
- Terminal Call History Data Clear

Note: For DT800/DT700 Series, only the call termination on My line enables illumination and displaying phonebook for incoming call. Information, such as number and internal/external call, is managed by each terminal according to notifications from the system.

5.1 Multiple Lines

A Desktop terminal is capable of registering two or more lines (My Line, Sub Lines) on the programmable Line/ Feature Keys. These lines (details are shown below) are generically called “Multiple Lines”.

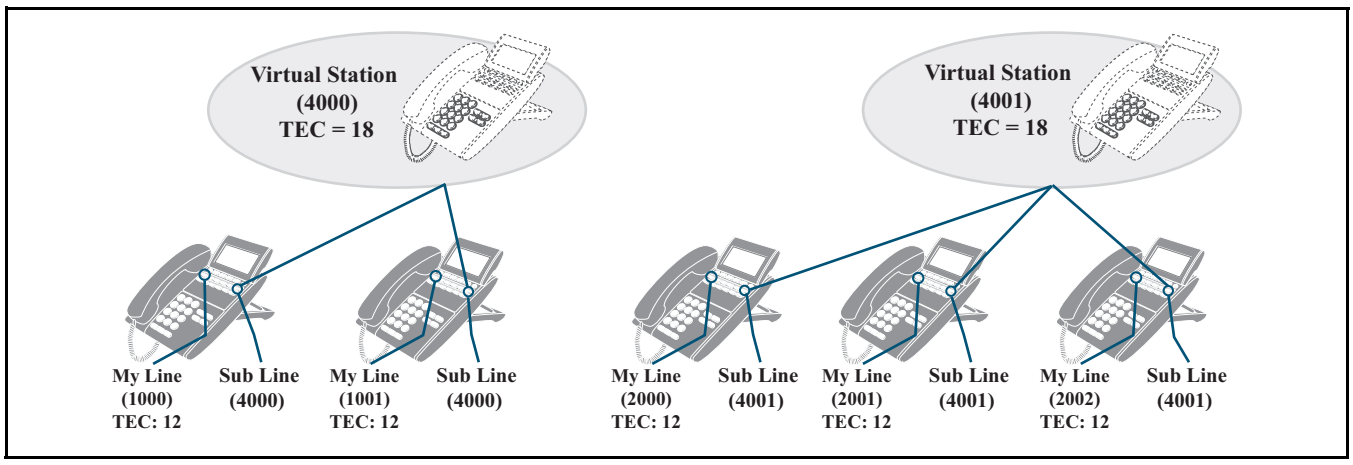
a.) My Line
Refers to the actual line that is directly associated with the extension number of a specific Station (Desktop terminal). This line is connected to the Station’s physically accommodated port of ELC (at the system side), and the Station can make/answer a call only via this line, even though such events are available by using other Sub Lines.

b.) Prime Line
Refers to the line that is seized automatically only by lifting the handset or pressing the Speaker Key. Normally, the same number as My Line is assigned for Prime Line. (See Prime Line Pickup -D [P-13D] of Chapter 3 for more information.)

c.) Sub Line
Refers generically to the lines, other than My Line, that are accommodated on programmable Line/ Feature Keys.The “Sub Line” can consist of other Desktop terminal’s My Line, the Station Number of Single Line Telephone, and also Software (Virtual) Line that may be assigned on a physically non-existent port of Group No. 24 - 31. **Note 1**
Usually, every Sub Line uses Software Line, and the line can be assigned on a group of Desktop terminals as a common Sub Line, as shown in the following figure.

d.) Software Line
Refers to the line that is logically registered on the memory, by using the physically non-existent Group No.24 - 31. While the Group Numbers 00 - 23 are assigned physically to the universal Slots of a PIR, the Group Numbers 24 - 31 for the Software Line are programmed/managed on the memory. **Note 1**
Thus, the theoretically assigned number for a Software Line can be used as a virtual Station Number (having no actual telephone set to its end), and a call to that number can be answered by any Desktop terminal that is registering the line. For more information, refer to Software Line Appearance [S-33] of Chapter 3.

Note 1: Group Number is one of the elements specifying the accommodated location of a line -- generally referred to as Line Equipment Numbers (LENS). A LENS is configured by a six-digit number (two-digit Module Group Number, one-digit Unit number, two-digit Group Number and one-digit Level Number). For more information, refer to the Operations and Maintenance Manual.



5.2 Assignment of Desktop Terminal Station Data

To use the Desktop terminal stations, the following must be assigned, in addition to the data explained in 4. Station Data.

Programming

< Necessary (minimum) data for Desktop terminals, in addition to the basic station data (4. Station Data) >

STEP 1: **ASYD** -SYS1, Index 78
Assign the following data for LCD display.
Bit 0: Assign "1" (fixed)...Calling and Intermediate Station Numbers are indicated.
Bit 1: Assign "1" (fixed)...Kind of Service Class is indicated.

SYS1, Index 321
Assign the Multiple Line Operation data.
Bit 0: Assign "1" to enable the Inter-Module/Inter-IMG Multiple Line Operation

Note: For details on Multiple Line Operation, refer to the Multiple Line Operation-D [M-20D] and Expanded Multiple Line Operation-D [E-26] of CHAPTER 3.

SYS1, Index 390
Assign the following data for Caller ID display.
Bit 2: Assign "1" to enable the Caller ID display on the bottom line of Desktop terminal LCD

SYS2, Index 12
Assign Automatic Recall Ringer data for Non-Exclusive/Exclusive Hold
Bit 0: 0/1 = Automatic Recall Ringer is required/not required for Non-Exclusive Hold
(Non-Exclusive Hold: When a call is held, the held call can be retrieved by other Stations)
Bit 1: 0/1 = Automatic Recall Ringer is required/not required for Exclusive Hold
(Exclusive Hold: When a call is held, the held call cannot be retrieved by other Stations)

STEP 2: **ASYDL** - SYS1, Index 672, Bit 1
Flag this bit when using Desktop terminal. If not, assign "0".

STEP 3: **ASDT** - Assign Telephone Equipment Class (TEC) for Desktop terminal designation.
TEC: 12=Desktop terminal (voice)/13=Connection to Data Terminal via Desktop terminal (using adapter)
TEC: 18=Software (Virtual) Line (for configuring Multiple Lines)

STEP 4: **AKYD** - Assign My Line, other Multiple Line or desired function to each Line/Feature Key. When no modification is necessary in regard to the following default data, the related key assignment is not required.

Default Data for Line/Feature Keys

KEY NO.	DEFAULT DATA	KEY NO.	DEFAULT DATA
1	CALL PICKUP	9	PRIME LINE
2	CALL FORWARDING - BUSY LINE	10	SPEED CALLING ONE - TOUCH
3	CALL FORWARDING - ALL CALLS	11	SPEED CALLING ONE - TOUCH
4	EXECUTIVE RIGHT OF WAY	12	SPEED CALLING ONE - TOUCH
5	VOICE CALL	13	SPEED CALLING ONE - TOUCH
6	MESSAGE REMINDER	14	SPEED CALLING ONE - TOUCH

KEY NO.	DEFAULT DATA	KEY NO.	DEFAULT DATA
7	SAVE AND REPEAT	15	SPEED CALLING ONE - TOUCH
8	LAST NUMBER CALL	16	SPEED CALLING - SYSTEM

Note: KYN 13-16 do not exist for DT800 Series: DT830 (12-button type)/DT830G (12-button type (12DG/12CG)/DT700 Series: DT730 (12-button type)/DT730G (12-button type (12DG/12CG)/DT400 Series: DT430 (12-button type) and DT300 Series: DT330 (12-button type).

Note: Assignment of key data is required for Desktop terminal with 8 keys or less.

When the Desktop terminal is used for a Data Terminal, the following must be assigned.

- FKY = 29 (DATA)
- FKY = 30 (Manual/Auto changeover)
- FKY = 31 (Data transfer)

When using Speed Calling - One Touch with SPD setting key, the following data assignment is required.

KYN: Key number
KYI = 1 (function key)
FKY = 199 (SPD setting)

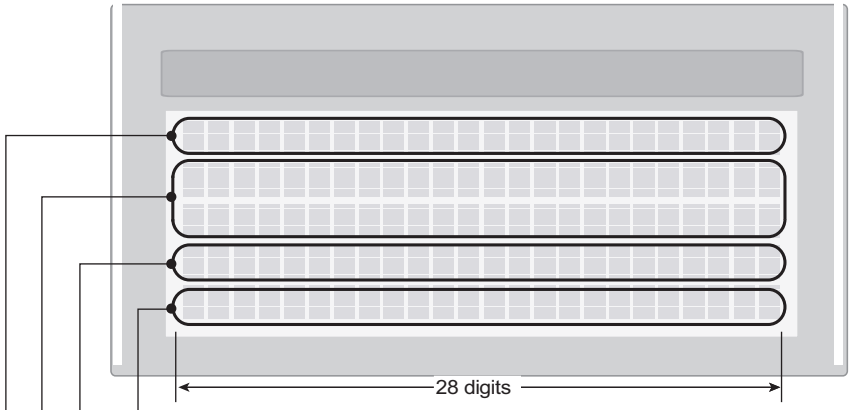
STEP 5: **ADSL** - Assign the following data to a Soft Key when using a Soft Key as a Speed Calling - One Touch with SPD setting key.

SN = 0 (idle)
FKY = 199 (SPD setting)

5.3 Display (LCD) Indications

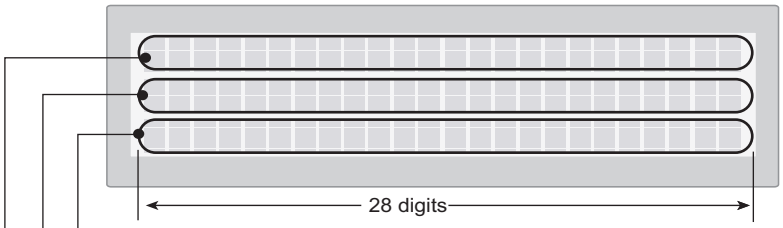
The following shows the patterns of the Liquid Crystal Display (LCD) Indications on DT800 Series/DT700 Series/DT400 Series/DT300 Series, and DtermIP and Dterm Series *i*.

- DT830/DT730DT430/DT330



- Function name of Soft Key **Note 3**
- Time display
- Caller information
- Message Waiting data
  (Displayed, when "MWD = 1 (middle line)" is assigned by AKYD command) **Note 1, Note 4**
- Feature name
- Intermediate station information
- Station number of call originator
- Message Waiting data
  (Displayed, when "MWD = 0 (top line)" is assigned by AKYD command.) **Note 2, Note 4**
- Icon indication

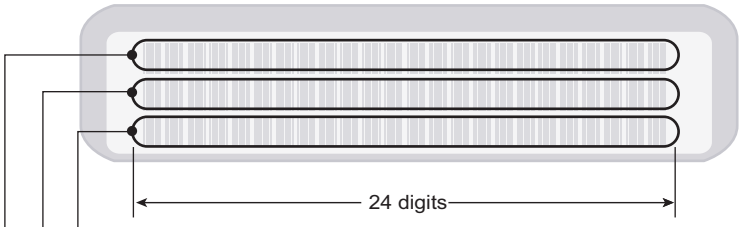
- DT820/DT710/DT410/DT310



- Function name of Soft Key **Note 3**
- Time display
- Caller information
- Message Waiting data
  (Displayed, when "MWD = 1 (middle line)" is assigned by AKYD command) **Note 1, Note 4**
- Feature Name
- Intermediate station information
- Station number of call originator
- Message Waiting data
  (Displayed, when "MWD = 0 (top line)" is assigned by AKYD command.) **Note 2, Note 4**

Note: For DT800 Series/DT700 Series/DT400 Series/DT300 Series, time display pattern can be changed by system data assignment. For details, see "Time Display [T-18D]".

- **DtermIP/Dterm Series *i***



- Function name of Soft Key **Note 3**
- Time display
- Caller information
- Message Waiting data
  (Displayed, when "MWD = 1 (middle line)" is assigned by AKYD command) **Note 1, Note 4**
- Service Name
- Intermediate station information
- Station number of call originator
- Message Waiting data
  (Displayed, when "MWD = 0 (top line)" is assigned by AKYD command.) **Note 2, Note 4**

Note 1: When Message Waiting is OFF, date and time are displayed. When Message Waiting is ON, the related Message Waiting information is displayed. When OAI service is set, the OAI feature information is displayed in stead of Message Waiting information. (Message Waiting information and OAI information cannot be displayed at the same time.)

Note 2: When display of Message Waiting data and service name coincides, service name data is displayed. Then, after service data disappears, Message Waiting data is displayed.

Note 3: For details on programming of Software Key data, refer to 5.6 Soft Key Assignment.

Note 4: Whether to display the Message Waiting data on top line or bottom line can be specified by the "MWD" parameter of AKYD.

Programming

STEP 1: **AKYD** - Select the line to display MW data
MWD: Message Waiting Data Display Selection [0/1]
0 = MW data display on the top line
1 = MW data display on the bottom line

STEP 2: **ASYD** -When you assign the service function to Desktop terminal, the assigned function name of Soft Key is displayed at the beginning of Soft Key display area. If you do not want to display the function name, assign the following System Data.
SYS1, Index 390, Bit 3 to Bit 4
Assign the display of Soft Key when the station is in an Idle status or is in Dial Tone connection.

Bit 4	Bit 3	
0	0	: Blink
0	1	: Steady-lights
1	0	: Displays the assigned code on first column **Note 5**
1	1	: No indication

Note 5: Displays the code assigned by SYS1, Index 391.

5.4 Line Pickup Preference

Assign line pickup preference for the Speaker, Answer and Originate Keys, by using the AKYD command.

Programming

AKYD - For each Desktop terminal, specify the line pickup preference when the Speaker, Answer, or Originate Key is pressed:
LN PRE: 0 = The following standard values are used:

BUTTON	LINE PREFERENCE
SPEAKER Button	Prime Line Preference
ANSWER Button	Incoming Line Preference (Button Location Ascending Order)
ORIGINATING Button	Idle Line Preference (Button Location Ascending Order)

1 = Line pickup preference is to be customized. Proceed with the SPK, ANS and ORG parameters below.

Note: After changing data for LN PRE parameter, ELC card initialization is required.

SPK: Assign the line pickup preference when the Speaker Key is pressed. **Note 1**
0 = Prime Line preference
1 = Idle Line preference
2 = Incoming Line preference
3 = Ringing Line preference
ANS: Assign the line pickup preference when the Answer Key is pressed. **Note 1**
0 = Prime Line preference
1 = Incoming Line preference
2 = Ringing Line preference
ORG: Assign the line pickup preference when the Originate Key is pressed. **Note 1**
0 = Prime Line preference
1 = Idle Line preference
2 = Prime Line, Idle Line (ascending order)
FKY: 27 (Originate [ORG])
28 (Answer [ANS])
52 (Speaker [SPKR])

Note 1: These parameters appear when "LN PRE" is set as "1". Each preference stands for:

- Prime Line preference: Prime Line is picked up when the corresponding key is pressed.
- Idle Line preference: Idle line is picked up when the corresponding key is pressed.
- Incoming line preference: Regardless of the ringing sound, the currently Incoming Call is picked up, when the corresponding key is pressed.
- Ringing Line preference: The line with the ringing sound is picked up when the corresponding key is pressed.(The line with no ringing sound is not picked up.)

5.5 Restriction of Call Termination on My Line

A Desktop terminal can deny any Incoming Call to its My Line, when a call is made/received via one of its Sub Lines.

<Conditions>
1. Restricts a call to My Line only. (A call to Sub Line is not restricted.)
2. Available when the caller is a Station, TIE Line, Central Office Line or Attendant Console.
3. Available for a call as the destination of Call Forwarding - All Calls.
4. When the Original Call on the Sub Line is placed on hold:
   a.) If the terminal is off–hook after placing the call to Sub Line on hold, a call to My Line is restricted.
   b.) If the terminal goes on–hook while the call to Sub Line is still on hold, a call to My Line is not restricted.

Programming

ASYD - SYS1, Index 321, Bit 2=1: Termination on My Line is restricted.

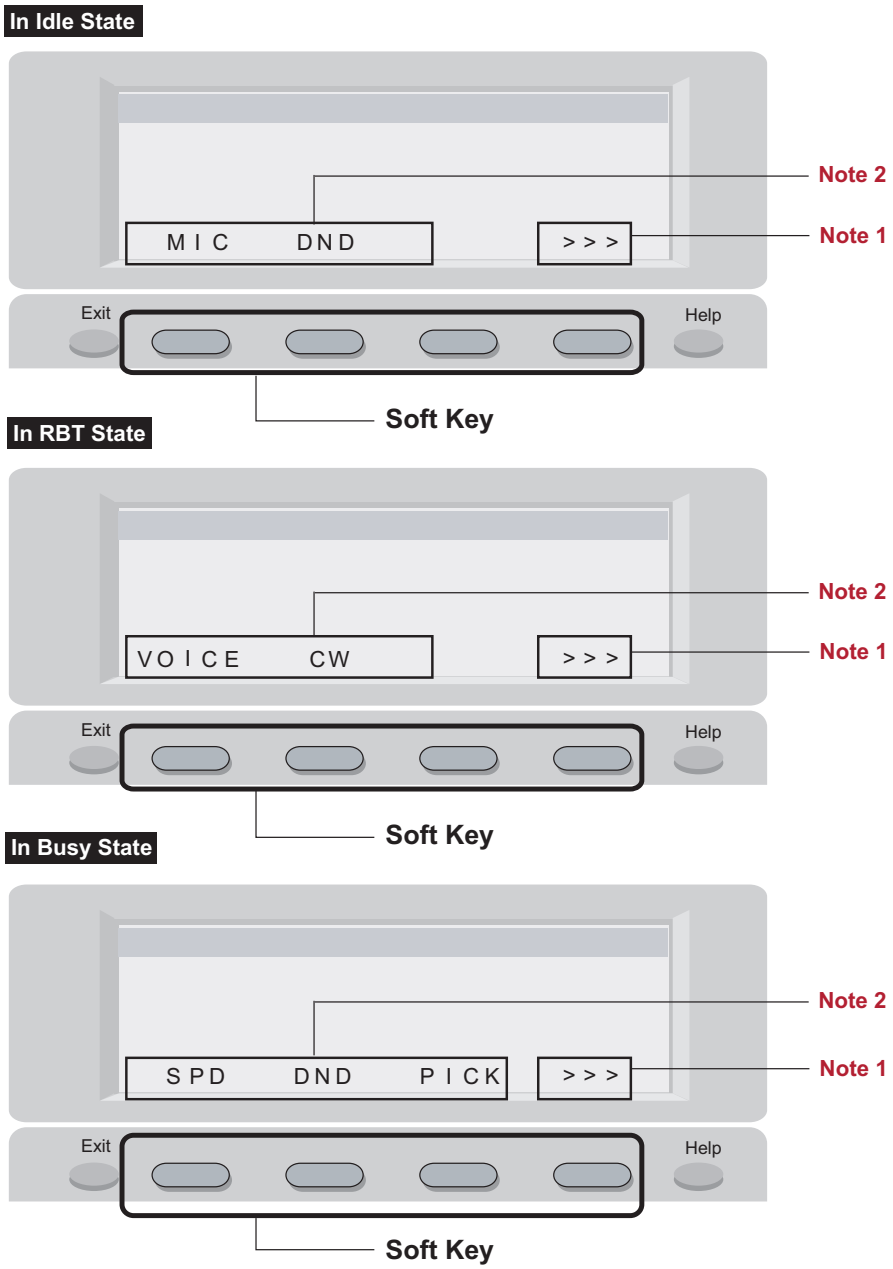
5.6 Soft Key Assignment

In addition to the programmable Line/Feature Key, DT800 Series, DT700 Series, DtermIP, DT400 Series, DT300 Series, Dterm Series *i* can use four Soft keys. Each key works variably according to the current state of station. For example, a Soft key provides a function of VOICE (Voice Call) when hearing Ringback Tone (after dialing the Called Station Number), etc.

Note: For adding a new node to the system in which the soft key data is already assigned by the AD-SLN command (*1), SKP (soft key pattern) and SN (status number) must be reassigned by the ADSLN command. (To reassign them, press [Get], [Del] and then [Set] buttons in that order.)

*(*1) It corresponds to the following situations:*
- *a new node is added to the existing standalone system and that system is upgraded to FCCS system.*
- *a new node is added to the existing FCCS system.*

An example of using Soft keys is shown below:



Note 1: ">>>" key is used to go to the next page. (Soft key data can be programmed across four pages to be turned over manually).

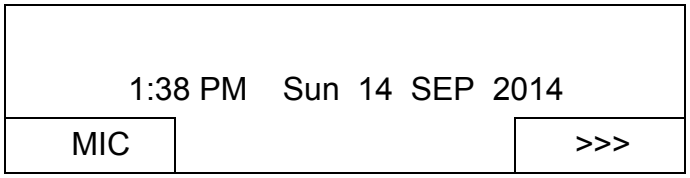
Note 2: For DT830G/DT830, eight digits characters with a frame are displayed on LCD in default settings. When you assign the system data (ASYDL/ASYDN, SYS1, Index 1258, Bit 2=1), six digits characters without a frame will be displayed. For other terminals other than DT830G/DT830, the soft key indication characters are displayed as six digits without a frame. For details, see Command Manual.

Programming

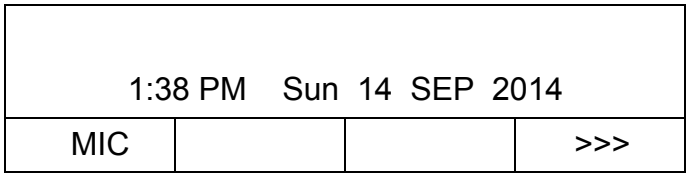
The Soft key data can be assigned by taking the following procedure:

Note: The data assignment of ADSLN must be all deleted before assigning ADSL. If the data of ADSLN is assigned, an “Input error” might be caused.

STEP 1: **ASYDL** - Assign Telephone Number Display Pattern
SYS1, Index 513=01 (LDM)
SYS1, Index 672, Bit 0
0 = Telephone Number is displayed on the top line of LCD
1 = Telephone Number is displayed on the middle line of LCD
SYS1, Index 1248, Bit 5 (Setting of Soft key frame display on the LCD of DT830G/DT830) **Note 2**, **Note 3**
0 = Displays a soft key with a frame.



1 = Displays the soft key indication area with a frame.



Note 2: Available for FP95-112 V2 or later.

Note 3: The changes made to the system data above will take effect after about 30 seconds.

SYS1, Index 1258, Bit 2 (Display the Soft key on the LCD as eight characters with a frame)
Note 4, **Note 5**
0 = Enable
1 = Disable

Note 4: For DT830G/DT830 only. When the data is changed by AFDD command, LCD displays the Soft key as six characters without a frame.

Note 5: For the language settings except for Japanese, NEC English, or Extension Languages, LCD displays the Soft key as six characters without a frame.

STEP 2: **AFDD** - Assign the characters to be displayed on the LCD, corresponding to each Soft Key function. For each function, default data can be used for each display, therefore use this command only when customizing the display characters.
FKY: Specify the service function by using the corresponding Function Key Code (1 - 320). (For details on Function Key Code allocation, refer to the Command Manual)
DISP: Assign the desired characters (a maximum of six letters or numbers) for the service specified by FKY above.

STEP 3: **ADSL** - Assign the Soft Key Pattern tables that determine each case of indication on the LCD bottom line (dedicated Soft Key display area). The function of each Soft Key is variable, according to the state of the Desktop terminal (idle or originating, busy, in speech, etc.). Therefore, taking this into consideration, create the desired number of tables for Soft Key Pattern. **Note 6**

SKP: Assign the Soft Key Pattern number (a maximum of 16 patterns [0 - 15] are assignable).

Note that SKP 0 is reserved for the Key Pattern to be commonly used on a system basis. For details on SKP 0 allocation, refer to Command Manual.

SN: Assign each Status Number for "SKP".

0 = Idle
1 = Originating (Dial Tone Connection)
2 = Originating (Ringback Tone Connection)
3 = Called Station Busy (Busy Tone Connection)
4 = Communication in progress
5 = Consultation Hold
6 = Trunk Busy (Busy Tone Connection)
8 = Consultation Hold when Called Station is busy
9 = While in Three-party Conference
15 = Others

SKN: Enter Soft Key Number (0 - 15 **Note 7**)

FKY: Assign the service function for each SKN by using the Function Key Code. (For details on Function Key Code allocation, refer to the Command Manual)

Note 6: ADSL and ADSLN cannot be assigned at the same time. When both ADSL and ADSLN are assigned, a disagreement might be caused.

Note 7: A set of four Soft Keys can be developed over four pages. Soft Key Number 0/4/8/12 is for the first key, 1/5/9/13 for second key, 2/6/10/14 for third key, and 3/7/11/15 for fourth key.

DISP: Display Message on LCD

The soft key indication is displayed when DISP has been assigned by the AFDD command.

Note: The soft key indication characters are displayed as six digits when using AFDD command.

Note: When the soft key indication is assigned by AFDD command, the data assignment of AFDD command has priority.

STEP 4: **ADKS** - Register each SKP (Soft Key Pattern) data to the desired Desktop terminal(s).

TN: Assign the Tenant Number of the Desktop terminal

STN: Assign the Station Number of the Desktop terminal

SKP: Specify the desired Soft Key Pattern, referring to SKP of ADSL.

LKP: Assign the Key Pattern of the Desktop terminal.

1 = 16 Line/Feature Buttons + DSS Keys
2 = 24 Line/Feature Buttons + DSS Keys
3 = ALL Line/Feature Buttons

PG: Specify the page scroll key allocation

0 = Page change key appears on a software key (>>>)
1 = Page change key disappears on a software key

STEP 5: **AKYD** - Assign "Soft Key Assignment" Key to Line/Feature Key. **Note 8**
KYN: Key Number [1-40]
KYI: 1 = Function Key
FKY:122 = Soft Key Assignment

Note 8: This data assignment enables the Desktop terminal users to change the Soft Key data assignment.

5.7 Expanded Ringer Pattern for Digital Terminal

The variety of ringer pattern for Digital terminals has been expanded from four to ten. The ringer pattern changes and rotates by pressing the Feature + 3 key. The rotation pattern of the ringer pattern is as follows.

TONE NUMBER	TONE KIND	1st/2nd FREQUENCY
Tone 1	16 Hz Modulation	520/660 Hz
Tone 2	8 Hz Modulation	520/660 Hz
Tone 3	16 Hz Modulation	1100/1400 Hz
Tone 4	16 Hz Modulation	660/760 Hz
Tone 5	Musical scale ringing of door bell	here
Tone 6	Musical scale ringing 1	-
Tone 7	Musical scale ringing 2	-
Tone 8	Musical scale ringing 3	-
Tone 9	Musical scale ringing 4	-
Tone 10	Musical scale ringing 5	-

Note: Tones shown in this list can be specified by pressing the Feature + 3.

Note: This feature is available with DT400/DT300 Series.

5.7.1 To change the ringer pattern for SIP Multiple Line terminal

- To change the ringer pattern for DtermIP (SIP), refer to “Ring Settings” in Peripheral Equipment Description (Multi-Line Telephone).
- To change the ringer pattern for DT800/DT700 Series, refer to “Ring Tone” Peripheral Equipment Description (Multi-Line Telephone).

5.8 Headset Data

Headset can be used by pressing the Headset/Handset Key programmed to the Desktop terminal.
To set Headset/Handset Key to the Soft Key, all of the data for Status Number (SN=0-15) need to be assigned by ADSL command.

Programming

STEP 1: **AKYD** - Assign FKY=125 (Headset/Handset Key) to a Feature Key.
STEP 2: **ADSL** - To assign the Headset/Handset Key to a Soft Key, assign the following data.
SN = 0-15
FKY = 125 (Headset/Handset Key)

5.9 Terminal Operations with Feature and Digit Keys for Digital Terminals

Following shows the functions of digital terminals by the operation with Feature and Digit Keys. For the operations of SIP Multiple Line terminals, refer to "SIP Multiple Line Operation [S-160]" in Chapter 3. **Note 4**

Digit Key	Function
0	Sidetone ON ⇔ OFF
1	Microphone ON ⇔ OFF
2	Handset receiving volume setting
3	Ringing Tone setting
4	Sending/Receiving volume control
5	Hands-Free ON **Note 1**
6	Hands-Free OFF
7	Call Indicator Lamp ON ⇔ OFF
8	Dynamic Dial Pad/Preset Dial ON ⇔ OFF **Note 2**, **Note 3**

Note 1: To operate Hands-Free call, be sure to set the microphone as "ON" pressing Feature Key + Digit Key "1".

Note 2: Dynamic Dial Pad and Preset Dial ON cannot be activated simultaneously.

Note 3: For using this function, system data assignment is required. For the details, refer to "Selective Dialing [S-162]" in Chapter 3.

Note 4: From FP95-114 V4 Issue 2.0, these functions can be set with the ATIFD command instead of using the Feature key and the digit keys. For details, refer to PREFERENCE SETTING - MULTI-LINE TELEPHONE WITH PCPRO [P-90] in Chapter 3.

5.10 Terminal Call History Data Clear

For DT800/DT700 Series, the terminal call history (outgoing call history/incoming call history) data can be deleted with the following operations.

- Resetting the terminal with the DISD command
- Logging out of the terminal using the LOGOUT key

Note: Available for FP95-112 V2 or later.

Note: For DT400 Series/DT300 Series, this feature is not available. To delete the terminal call history data of DT400 Series/DT300 Series, insert and extract the Modular Cable, or clear the terminal call history data from the [History] menu.

1. When you delete the terminal call history data by resetting the terminal, the method of terminal reset depends on the PH Fixed Hunting method (ASYDL, SYS1, Index 824, Bit 0).

   - When PH Fixed Hunting method is invalid (ASYDL, SYS1, Index 824, Bit 0=0)

   Available: × Not available: N

	When Terminal Call History Data Clear is enabled (ASYDL/ASYDN, SYS1, Index 1248, Bit 6=1)		When Terminal Call History Data Clear is disabled (ASYDL/ASYDN, SYS1, Index 1248, Bit 6=0)
The method of terminal reset	**Terminal reset by DISD command/Terminal reset with LOGOUT key**	**Other methods of terminal reset**	**All methods of terminal reset**
Terminal Call History (outgoing call history/ incoming call history)	×	N **Note 1**	N **Note 1**
Call History	×	×	×
Last Number Called	×	×	×
Redial	×	×	×

- When PH Fixed Hunting method is valid (ASYDL, SYS1, Index 824, Bit 0=1)

Available: × Not available: N

	When Terminal Call History Data Clear is enabled (ASYDL/ASYDN, SYS1, Index 1248, Bit 6=1)			When Terminal Call History Data Clear is disabled (ASYDL/ASYDN, SYS1, Index 1248, Bit 6=0)	
The methods of terminal reset	**PH information clear and terminal reset by DISD command/ Terminal reset with LOGOUT key**	**Terminal reset by DISD command**	**Other methods of terminal reset**	**PH information clear and terminal reset by DISD command**	**Other methods of terminal reset**
Terminal Call History (outgoing call history/ incoming call history)	×	×	N **Note 1**	N	N **Note 1**
Call History	×	N	N	×	N
Last Number Called	×	N	N	×	N
Redial	×	N	N	×	N

Note 1: When you turn off the power of the terminal or make the configuration setting, the terminal call history data will be deleted.

2. The terminal call history data cannot be deleted by system initialization or terminal reset by AKYD command.
3. When this feature is available in the FCCS cluster network, the terminal call history data will be deleted when:
   - the terminal is reset with the CBKS command.
   - the terminal is brought back under the control of Primary node by means of the Visitor key.
4. When the terminal is reset, the ringing volume is not saved and returns to the default.

Programming

STEP 1: **ASYDL/ASYDN** - SYS1, Index 1248, Bit 6 (LDM/NDM)
0 = Do not delete the terminal call history data for DT800/DT700 Series
1 = Delete the terminal call history data for DT800/DT700 Series

Note: When the System Data is changed, restart the terminal.

STEP 2: **AKYD/AKYD_T** - Assign a LOGOUT key to the feature key of terminal
KYN: Key Number
KYI: 1 (Feature Key)
FKY: 142 (LOGOUT Key)

STEP 3: **ADSL/ADSLN** - Assign a LOGOUT key to the Soft key of terminal
SN: 0 (Line is idle)
FKY: 142 (LOGOUT Key)

6. Telephone Number

This section explains how to program Telephone Numbers. To program Telephone Numbers, use the ALGSL/ ALGSN command. The Telephone Number is assigned to a Station by using the LENS or by using the physical Station Number depending on programming. For more information about Telephone Numbers in the FCCS network, refer to Data Programming Manual - FCCS.

Note: Although the physical Station Number is assigned to a terminal with the ASDT command, it is also possible to assign at the same time the physical Station Number (ASDT command) and the Telephone Number (ALGSN command) with the ATSTN command.

Service Conditions

1. Each Telephone Number must be unique in the network.
2. If Telephone Numbers are used in the network, they are assigned in the Network Control Node (NCN).
3. Telephone Numbers assigned by the ALGSL command can only be used inside its own node.
4. Telephone Numbers assigned on ALGSN command can be shared in the network.
5. A maximum of 16 digits are used as a Telephone Number.

   Note: The maximum of digits may vary depending on features or systems. For details, refer to the related manuals.
6. Station Hunting Group or Call Pickup Group can be created with Telephone Numbers (LDM).
7. Telephone Number can be assigned to Phantom Number for a Station.
8. Attendant Console displays Telephone Numbers (two to five digits only) as Station Numbers.
9. Telephone Numbers used only inside its own node (assigned by the ALGSL command) and Telephone numbers used in the network (assigned by the ALGSN command) cannot coexist.

Programming

<Telephone Number for Self Node (Setting at Local Nodes)>

STEP 1: **ASYDL** (Assignment of System Data (LDM))
SYS1, Index 513: Assign 01 (hexadecimal) as the LDM's Memory Block Number.
SYS1, Index 514: Assign 01 (hexadecimal) as the NDM's Memory Block Number.
SYS1, Index 677, Bit 3=1 Display the Call Forwarding Information to a maximum of 16 digits on LCD **Note 1**
SYS1, Index 800.
Bit 1: ASPAL Tenant development is Separate / Common = 0/1
Bit 2: ANPDL Tenant development is Separate / Common = 0/1
SYS1, Index 1131, Bit 5=1 Non-Display of Telephone Number when the service is assigned by Feature Codes or Feature Keys. **Note 1**

Note 1: Assign this data for Telephone Numbers that have seven or more digits.

STEP 2: **ANPDL** (Assignment of Numbering Plan Data for Local Data Memory)
Assign numbering plan data for Telephone Number.

STEP 3: **ASPAL** (Assignment of Special Access Code for LDM)
CI (Connection Index) = N (Normal) or H (Hooking)
SRV (Kind of Service) = TELN (Telephone Number)

Note: When assigning the special access code data for telephone numbers using more than one tenant with the ASPAL/ASPAN command, the valid value for NND will be the one of the last assigned tenant.
Example:
1) TN: 1, ACC: 21, CI: N, SRV: TELN and NND: 5 are assigned with ASPAL/ASPAN command.
2) TELN: 21000 is assigned with ALGSL/ALGSN command.
3) TN: 2, ACC: 2, CI: N, SRV: TELN and NND: 4 are assigned with ASPAL/ASPAN command.
4) TELN: 2000 is assigned with ALGSL/ALGSN command.
5) Try to get the data for TELN: 21000 with ALGSL/ALGSN command.
Because the valid NND value is 4 (the last assigned) and the NND value for 21000 is 5, TELN: 21000 data cannot be obtained and an error message will be displayed.

STEP 4: **ALGSL** (Assignment of Telephone Station Data for LDM)
Assign telephone numbers to physical Station Number (Type 2) or to physical Line Equipment Numbers/Phantom Index (Type 1) by using the ALGSL command.

- When assigning TYPE=1 (Line Equipment Numbers/Phantom Index is used to specify the Telephone Number)

  TYPE = 1 (Line Equipment Numbers/Phantom Index is used to specify the Telephone Number)
  UGN: User Group Number
  TELN: Telephone Number (Up to 16 digits)
  LENS/PHIN: Line Equipment Numbers/Phantom Index of the physical Station

- When assigning TYPE=2 (Station Number is used to specify the Telephone Number)

  TYPE = 2 (Station Number is used to specify the Telephone Number)
  UGN: User Group Number
  TELN: Telephone Number (Up to 16 digits)
  TN: Tenant number of the physical Station
  STN: Station Number of the physical Station

<Telephone Number for Network (Setting at Network Control Nodes)>

STEP 1: **ASYDL** (Assignment of System Data (LDM))
SYS1, Index 513: Assign 01 (hexadecimal) as the LDM's Memory Block Number.
SYS1, Index 514: Assign 01 (hexadecimal) as the NDM's Memory Block Number.
SYS1, Index 677, Bit 3=1 Display the Call Forwarding Information to a maximum of 16 digits on LCD **Note 2**
SYS1, Index 1131, Bit 5=1 Non-Display of Telephone Number when the service is assigned by Feature Codes or Feature Keys) **Note 2**

Note 2: Assign this data for Telephone Numbers that have seven or more digits.

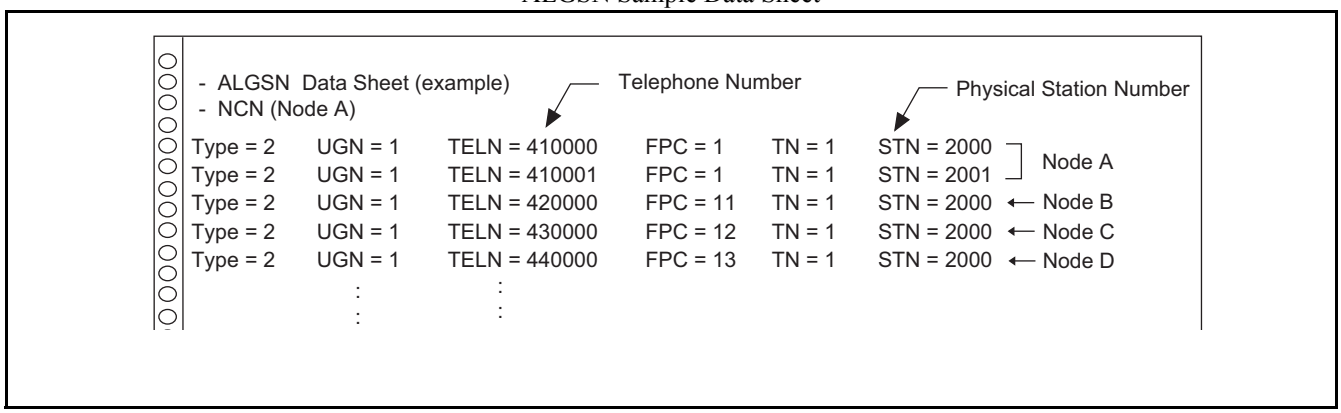
STEP 2: **ASYDN** (Assignment of System Data (NDM))
SYS1, Index 800.
Bit 1: ASPAN command tenant data table development is Separate / Common = 0/1
Bit 2: ANPDN command tenant data table development is Separate / Common = 0/1

STEP 3: **ANPDN** (Assignment of Numbering Plan Data for Network Data Memory)
Assign numbering plan data for Telephone Number.

STEP 4: **ASPAN** (Assignment of Special Access Code for NDM)
CI (Connection Index) = "N: Normal" or "H: Hooking"
SRV (Kind of Service) = "TELN: Telephone Number"

STEP 5: **ALGSN** (Assignment of Telephone Station Data for NDM)
Assign telephone numbers to physical Station Number (Type 2) or to physical Line Equipment Numbers/Phantom Index (Type 1) by using the ALGSN command. An example data sheet-Type 2 is shown as follows.

ALGSN Sample Data Sheet



- When assigning TYPE=1 (Line Equipment Numbers/Phantom Index is used to specify the Telephone Number)

  TYPE = 1 (Line Equipment Numbers/Phantom Index is used to specify the Telephone Number)
  UGN: User Group Number
  TELN: Telephone Number (Up to 16 digits)
  FPC: Fusion Point Code (FPC) of the designated LENS.
  LENS/PHIN: Line Equipment Numbers/Phantom Index of the physical Station

- When assigning TYPE=2 (Station Number is used to specify the Telephone Number)

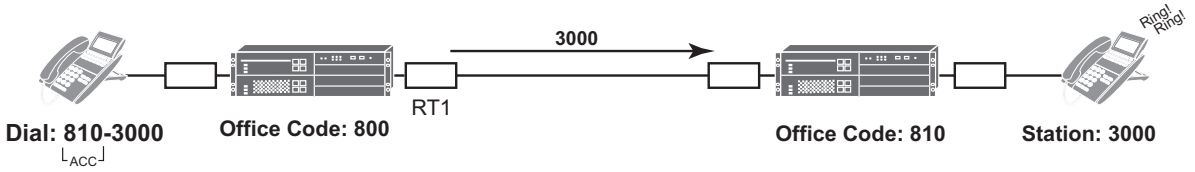
  TYPE = 2 (Station Number is used to specify the Telephone Number)
  UGN: User Group Number
  TELN: Telephone Number (Up to 16 digits)
  TN: Tenant number of the physical Station.
  STN: Station Number of the physical Station.

7. OGC/OGCA

7.1 OGC

OGC Access Code is only used for seizing a Trunk. Therefore, after seizing, the Station Number or subscriber line number is sent (the Access Code is not sent). Basically, the originating method does not add or delete any number. However, when senderized line is used, a maximum of 24 digits can be deleted, and four digits can be added by using ASTP/AOSP command. When the Route is busy, the Calling Party hears Busy Tone, because the OGC method can seize only one Route. Details on the OGC are shown in the following figure.

Example of OGC: When dialing 810-3000, RT1 is seized by Access Code (810). After seizing, Station No. 3000 is sent.



Note: MC&MG-COT/MG-COT is not available for OGC/OGCA. For MC&MG-COT/MG-COT, use LCR Access Code.

Programming

The following describe the data assignment of the example in above figure.

<OGC Access Code Setting>

STEP 1: **ANPD** - Assign the first digit of Access Code for OGC and digit number of the Access Code.
1stDC (First Digit of Access Code) = 8
CI (Connection Index) = Select N (Normal Service) or H (Hooking Service) from the combo box.
NND (Number of Necessary Digits for Access Code) = 3

STEP 2: **ASPA** - Assign the Access Code for OGC
ACC (Access Code) = 810
CI (Connection Index) = "N: Normal" or "H: Hooking"
SRV (Kind of Service) = "OGC: Outgoing Call"
RT (External Route Number) = 1

<Assignment of Necessary Digits to be received by Register>

- When acknowledgment signal interface is Second Dial Tone (ARTD command, OSGS = 2)
  AMND - Assign maximum necessary digits to be received by Register.
  DC (Destination Code) = 810
  MND (Maximum Necessary Digits) = 7

- When acknowledgment signal interface is Sender (ARTD command, OSGS =4/6/7)

  ANND - Assign the number of necessary digits to be received by Register.
  RT (Route Number) = 1
  IC/OG (Incoming/Outgoing Call) = O (Outgoing)
  NND (Number of Necessary Digits) = 7

  Note: When the digits, determined by the parameter, are dialed, Sender starts to transmit the number information to the Route.

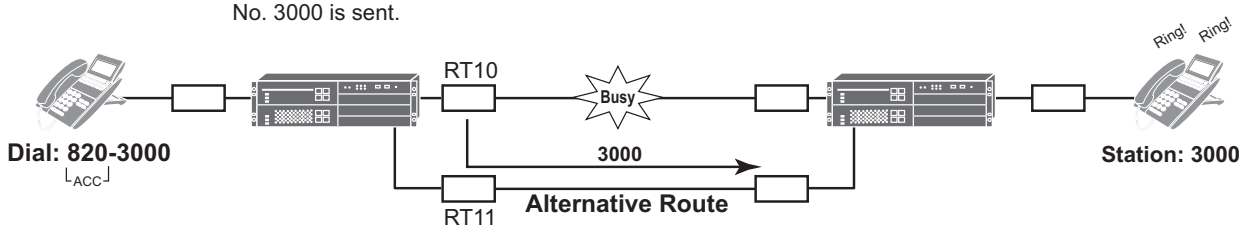
<Assignment of Access Code for each Route>

ARNP - Assign Access Code for each Route access. In the example here, assign the OGC Access Code for RT 1.
RT (Route) = 1
ACC (Access Code) = 810

7.2 OGCA

Same as OGC, OGCA Access Code is only used for seizing a Trunk. Therefore, after seizing, the Station Number or Subscriber Line Number is actually sent (the Access Code is not sent). When the first Route is busy or restricted by Office Data, a call, originated by OGCA, can be routed to other Trunk Group which goes in the same direction. Thus, it is possible to select multiple routes by only one Access Code. The maximum number of route, seized by one Access Code, is 15. When these 15 routes are all busy, Busy Tone is sent to the Calling Party.

Example of OGCA: When dialing 820-3000, RT11(alternative route) is seized by Access Code (810). After seizing, Station No. 3000 is sent.



Programming

The following describe the data assignment of the example in figure above.

<OGCA Access Code Setting>

STEP 1: **ANPD** - Assign the first digit of Access Code for OGCA and digit number of the Access Code.
1stDC (First Digit of Access Code) = 8
CI (Connection Index) = Select N (Normal Service) or H (Hooking Service) from the combo box.
NND (Number of Necessary Digits for Access Code) = 3

STEP 2: **ASPA** - Assign the Access Code for OGCA
ACC (Access Code) = 820
CI (Connection Index) = “N: Normal” or “H: Hooking”
SRV (Kind of Service) = “OGCA: Outgoing Call with route advance”
1st RT (First External Route Number) = 10
2nd RT (Second External Route Number) = 11

<Assignment of Necessary Digits to be received by Register>

- When acknowledgment signal interface is Second Dial Tone (ARTD command, OSGS = 2)

  AMND - Assign maximum necessary digits to be received by Register.
  DC (Destination Code) = 820
  MND (Maximum Necessary Digits) = 7

- When acknowledgment signal interface is Sender (ARTD command, OSGS =4/6/7)

  ANND - Assign the number of necessary digits to be received by Register.

  For RT 10
  RT (Route Number) = 10
  IC/OG (Incoming/Outgoing Call) = O (Outgoing)
  NND (Number of Necessary Digits) = 7

  For RT 11
  RT (Route Number) = 11
  IC/OG (Incoming/Outgoing Call) = O (Outgoing)
  NND (Number of Necessary Digits) = 7

  Note: When the digits, determined by the parameter, is dialed, Sender starts to transmit the number information to the route.

<Assignment of Access Code for each route>

ARNP - Assign Access Code for each route access. In the example here, assign the OGCA Access Code for RT 10 and 11.
RT (Route) = 10
ACC (Access Code) = 820
RT (Route) = 11
ACC (Access Code) = 820

7.3 Network by OGC

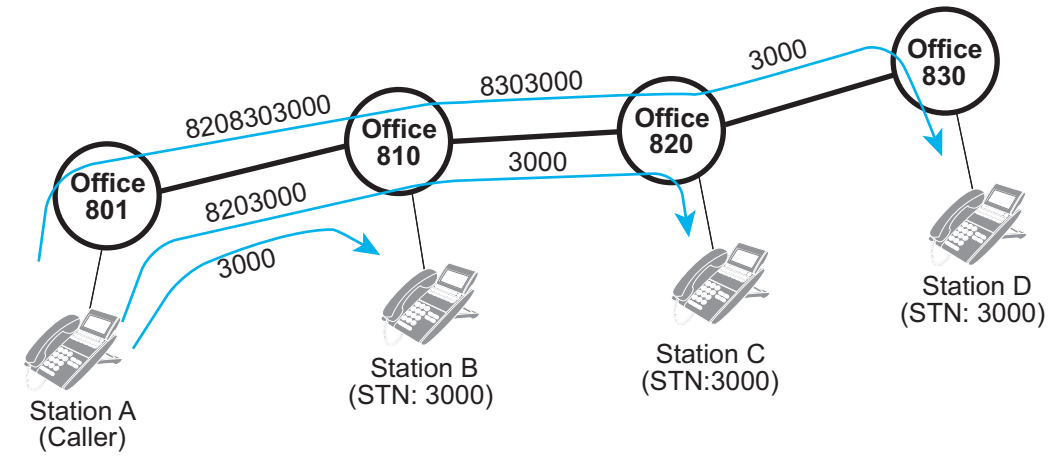
This section describes an example of a network using the OGC originating method.
If the user does not know the network configuration, the user may not be able to ring the Destination Station. Therefore, the OGC is most the suitable originating method for a small size network.

8. LCR/LCRS

8.1 LCR/LCRS

LCR/LCRS is the most suitable originating method for medium or large-sized network. It has many functions, for example, route selection and number transformation. If a call encounters busy, the system selects the next route according to the pre-programmed route selection table. Some number(s) may be added or deleted, depending on the selected route. The difference between OGC/OGCA and LCR/LCRS, that an Access Code for the latter is not necessarily to be deleted upon seizing a Trunk. Whether the system deletes/adds some numbers or not can be programmed by Office Data.

Example : To make a call, the caller should know the network configuration.



Number to be dialed
To Station B: 810-3000
To Station C: 810-820-3000
To Station D: 810-820-830-3000

8.2 Difference between LCR and LCRS

LCR: The system selects a Trunk after Register receives all dialed numbers. Therefore, the Trunk is not seized when dialing is not completed.
Use this method when:
- making a call to ISDN line or CCIS line.
- making a call with “Reduction of Data Consumption of the AFRSN command” feature. (**Note 1**)

Note 1: For details, see 2.3 Reduction of Data Consumption of the AFRSN command, CHAPTER 5 DATA PROGRAMMING of Data Programming Manual - FCCS.

LCRS: The system selects a Trunk immediately after Register receives Number Pattern Code (NPC) assigned by AFRS. Therefore, the time for seizing the Trunk is faster than the LCR method. However, even if dialing is not completed (when NPC is dialed), the Trunk is seized until time-out. When the user dials numbers that refer to data, assigned by AADC/ATDP/APCR, the call is originated by the LCR method even though the LCRS method is set.

When the LCRS method is used, the system starts connecting and communication between the Calling Party and the Called Party without completing the maximum digits (MND) of Destination Number which is set by the AMND command from the Calling Party.

- Assign the parameter DCANS (CDN: 54) of ARTI command as "1" to the objected route.
- Assign the parameter OSGS (CDN: 1) of ARTD as "2" (Second Dial Tone) and LSG (CDN: 15) of ARTD as other than "12" (CCIS/ISDN) to the objected Route Class of Outgoing Trunk.
- In Tandem Connection, when LSG (CDN: 15) of ARTD command is set as "0" and ANS (CDN: 28) of ARTD command as "0" to terminating route of tandem office, the feature is not available.
- This feature is available when a call is directly originated from a Station via LCRS.
- This feature is available when a Station/Trunk Call, once placed on hold, is originated via LCRS.
- This feature is available when a Trunk Call is originated via LCRS (Tandem Connection).
- The enhancement is not available when a call is placed from an Attendant Console via LCRS or a call is made via LCRS by the Individual Trunk Access feature.

Note: The originating method for ISDN/CCIS/Dp Channel TIE Line is LCR only. The following tables shows conditions for Alternate Routing.

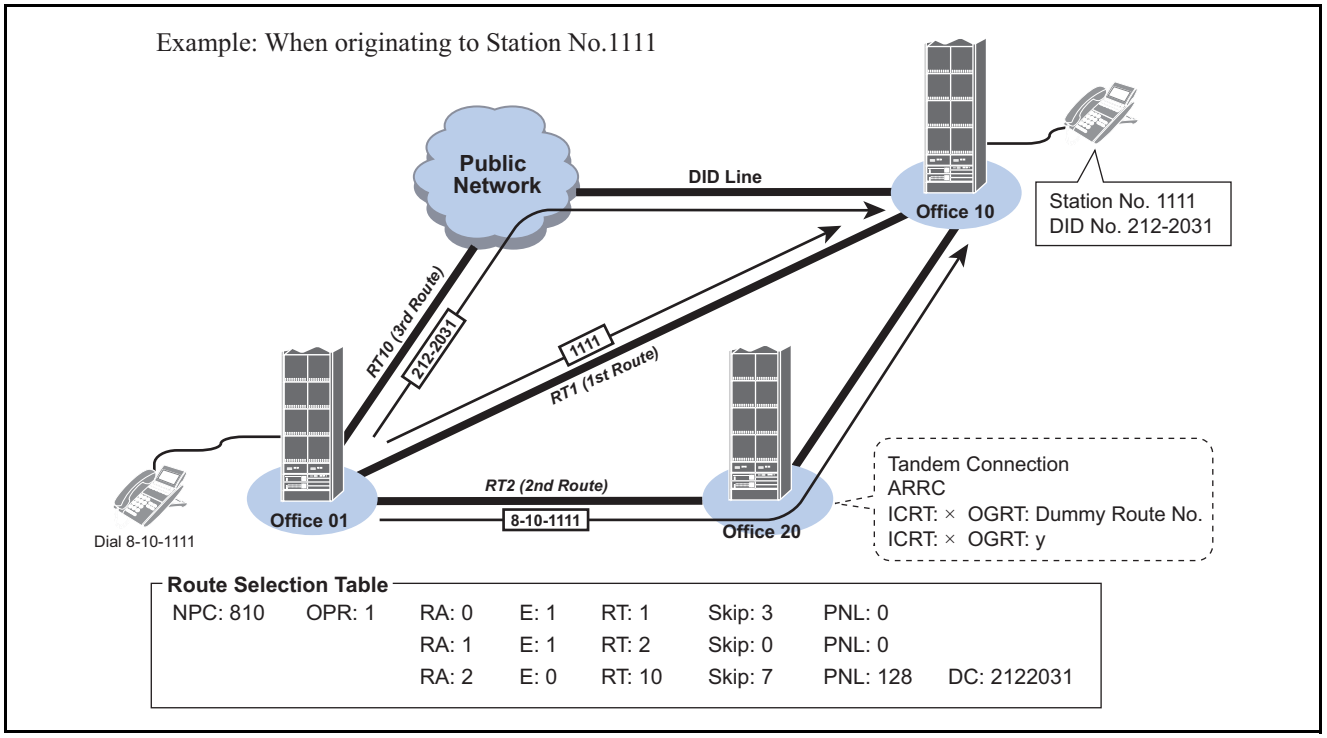
X: Available -: Not Available

	Alternate Routing						
	Analog Line ↓ ISDN Line	ISDN Line ↓ Analog Line	Analog TIE Line ↓ ISDN	CCIS Line ↓ ISDN Line	CCIS Line ↓ Analog Line	CCIS Line ↓ ACIS Line	ACIS Line ↓ CCIS Line
LCR	×	×	×	×	×	×	×
LCRS	× Note 2	–	–	–	–	–	× Note 2

Note 2: When alternating, the originating method is automatically changed to LCR.

Programming

The following describes an example of LCR/LCRS data assignment.



STEP 1: **ANPD** - Assign the first digit of an Access Code for LCR/LCRS and digit number of the Access Code.
1stDC (First Digit of Access Code for LCR/LCRS) = 8
CI (Connection Index) = Select N (Normal Service) or H (Hooking Service) from the combo box.
NND (Number of Necessary Digits for the Access Code) = 3

STEP 2: **ASPA** - Assign the Access Code for LCR/LCRS.
ACC (Access Code) = 810
CI (Connection Index) = "N: Normal" or "H: Hooking"
SRV (Kind of Service) = "LCR: Least Cost Routing" or "LCRS: Register Sender LCR"
RT (External Route Number) = Dummy Route Number (External Route Number that is not used in the system)

STEP 3: **ARTD** - Assign Route Class Data for dummy route.
RT (Route): Dummy Route Number
TCL = 4
L/T = 1
TQ = 1
AC = 1

STEP 4: **AFRS** - Assign Number Pattern Code (including Access Code) and route selection table used when dialing NPC (810).

RT (Route Number) = Dummy Route Number by ASPA.
NPC (Number Pattern Code) = 810
OPR (Outgoing Selection Pattern Number) = 1

STEP 5: **AOPR** - Develop the routing table associated with OPR of AFRS command.

TDPTN (Time-of-Day Change Pattern) = 0 (default) **Note 3**

Note 3: This parameter is used to select the Time-of-Day Change Pattern assigned by ATCP.
For details, refer to Command Manual.

OPR (Outgoing Route Selection Pattern) = 1 (the OPR table number assigned in OPR of AFRS)

1st Route Data
RA (Order of Route Advancing) = 0 (First Route)
E (Route Advancing End Indication) = 1
RT (Route Number) = 1
Skip (Number of Skipped Digits) = 3 (the number of digit to be deleted after RT 1 is selected)
PNL (Pattern Number Location) = 0
OVFT (Overflow Tone) = 0 (Overflow Tone is not required.)
PRSC (Priority Restriction Class) = 0 (Normally assign “0”)

2nd Route Data
RA (Order of Route Advancing) = 1 (Second Route)
E (Route Advancing End Indication) = 1
RT (Route Number) = 2
Skip (Number of Skipped Digits) = 0
PNL (Pattern Number Location) = 0
OVFT (Overflow Tone) = 0 (Overflow Tone is not required.)
PRSC (Priority Restriction Class) = 0 (Normally assign “0”)

3rd Route Data
RA (Order of Route Advancing) = 2 (3rd route)
E (Route Advancing End Indication) = 0 (Last route) **Note 4**
RT (Route Number) = 10
Skip (Number of Skipped Digits) = 7 (8101111 is deleted)
PNL (Pattern Number Location) = 128
OVFT (Overflow Tone) = 0 (Overflow Tone is not required.)
PRSC (Priority Restriction Class) = 0 (Normally assign “0”)

Note 4: Be sure to assign 0 at the last RA.

STEP 6: **AADC** - Assign an additional digit information to PNL (128) assigned by the AOPR command.

PNL (Pattern Number Location) = 128
DC (Additional Digit Information) = 2122031

STEP 7: **AMND** - Assign number of necessary digits to be received by Register.
Do not assign this information to Additional Digits Information specified by the AADC command.

DC (Destination Code) = 810
MND (Maximum Necessary Digits) = 7

Note: In addition to the data above, Route Class Data (ARTD), Trunk Data (ATRK) and Route Restriction Class data [ARSC, (RRI = 3)] are also required. Also, at Office 20, Tandem Connection data (ARRC) is necessary.

In the similar way, also perform the data programming for LCR access to Office 20 (ACC=820) and Public Network (ACC=9).

STEP 8: **ARNP** - Assign LCR Access Code for each Route, except Dummy Route.

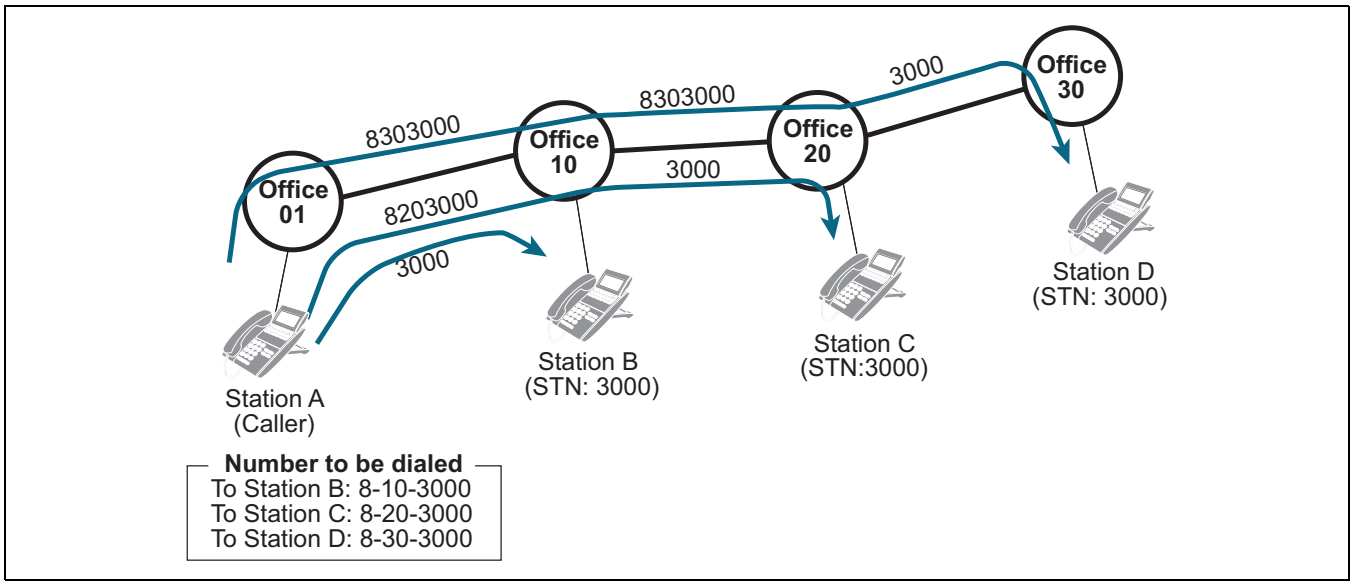
In this example, the following data is required:
RT (Route) = 1
ACC (Access Code) = 810
RT (Route) = 2
ACC (Access Code) = 820 **Note 5**
RT (Route) = 10
ACC (Access Code) = 9 **Note 5**

Note 5: When LCR data for RT 2/RT 10 is not programmed, the respective ACC assignment is not required for this command.

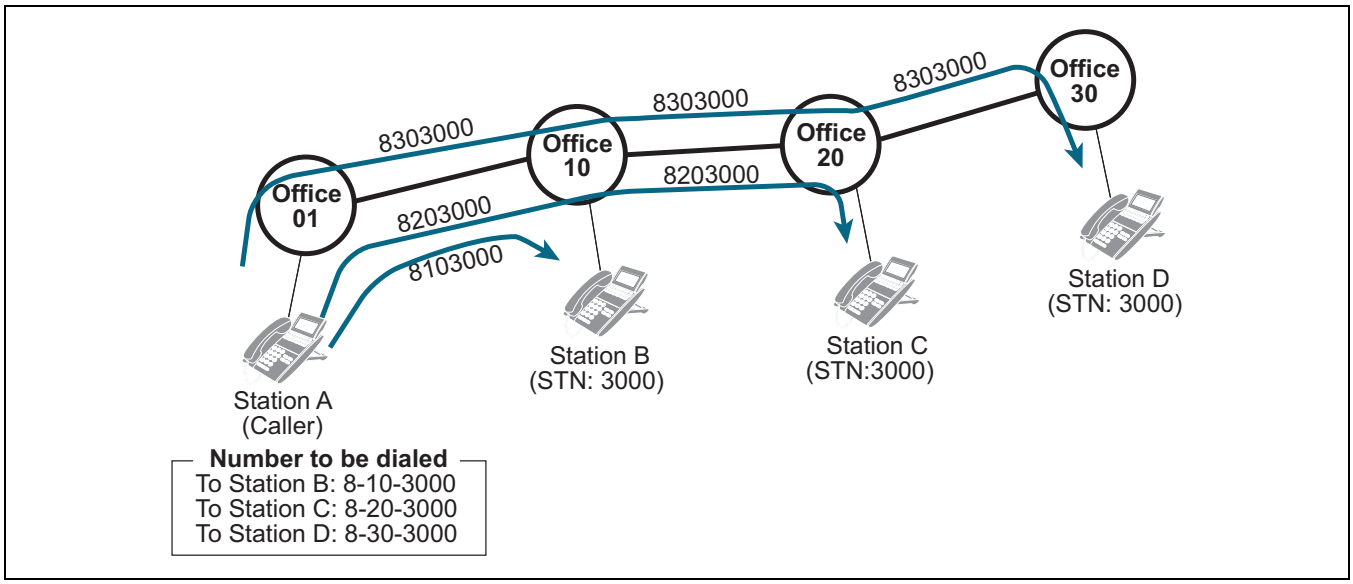
8.3 Network by LCR/LCRS

This section describes some examples of a network using LCR/LCRS originating method.

Example 1: When placing a call to a Station in the next office, only the Destination Station Number is sent. When originating to Station via tandem office, the Access Code, Office Code and Station Numbers are all sent. See the following figure.



Example 2: When placing a call to a Station in another office, Access Code, Office Code and Station Numbers are all sent at any time. In this case, data assignment by ASPA (SRV = "UNIF: Office Termination") or AUNE is necessary in each office. See the following figure.



Thus, on the network using LCR/LCRS, the user can easily place a call by dialing the "Access Code + Office Code + Station Number" (even if the user does not understand the network configuration). Addition or deletion of some number(s) is possible when the related Office Data has been assigned to do so.

9. Uniform Numbering

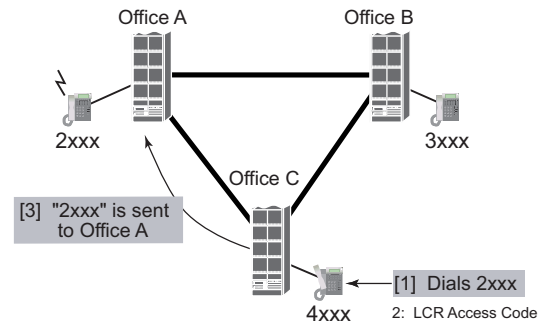
This section explains the data assignment procedure related to Uniform Numbering. The data is required when:

- Your system is one of the Nodes configuring an Open Numbering Network and,
- Inter Office Call via the network is set to be connected with the last destination Office deleting its Self Office Code.

If this is the case, perform the data setting as shown on the next pages. On some networks, the Office Code of the Call Destination Node is set to be deleted at the originating Node (by the data setting). However, if the network requests the Destination Office to delete the Code, this section data (Uniform Numbering) is necessary.

As shown in this figure, Uniform Numbering data is required when Systems achieve an Open Numbering Plan network, and the last destination office is set to receive its self-Office Code to allow inter-office calls bound for it.

< Closed Numbering >



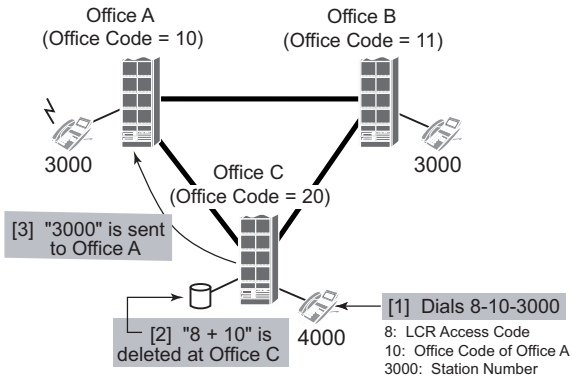


No station number is set to be duplicated on the network. An inter-office call can be made by dialing the intended station number only.
(i.e. there is no concept for "Office Code".)

Accordingly,

Uniform Numbering data ... NOT REQUIRED

< Open Numbering - Destination Office Code Is Deleted at Call Originating/Tandem Office >

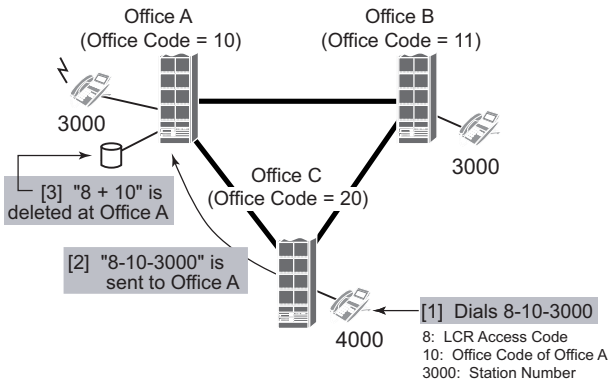


Station numbers are assigned on a node basis, therefore there can be duplicated number(s) between the nodes. On this network, the caller dials the trunk access code (LCR access code, etc.) + destination Office Code + station number. This example shows "trunk access code + Office Code" is deleted at the call originating office.

If this is the case,

Uniform Numbering data ... NOT REQUIRED

< Open Numbering - Destination Office Code Is Deleted at Call Destination Office >



Station numbers are assigned on a node basis, therefore there can be duplicated number(s) between the nodes. On this network, the caller dials the trunk access code (LCR access code, etc.) + destination Office Code + station number. This example shows "trunk access code + Office Code" is deleted at the call destination office.

If this is the case,

Uniform Numbering data ... REQUIRED !

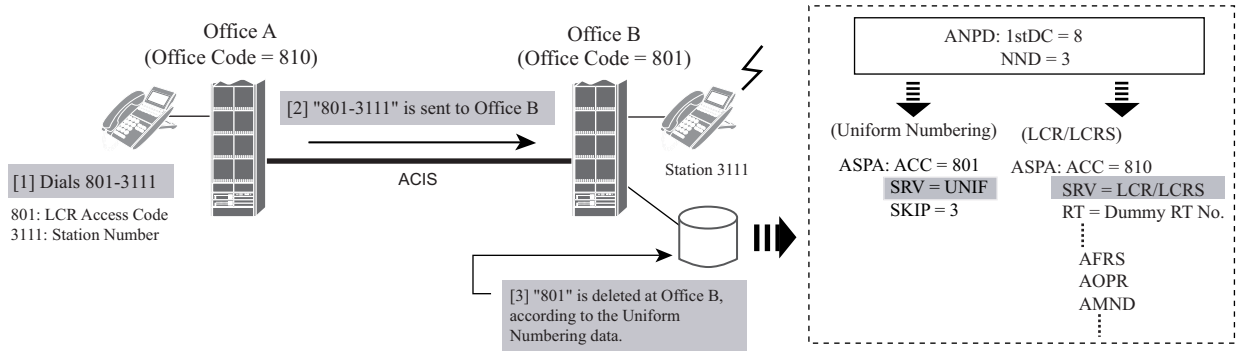
Programming

The following shows the Uniform Numbering data assignment procedure using examples for ACIS and CCIS network. By this data programming, you must allow your system to delete its Self Office Code when a call arrives with the number information inclusive of the Office Code. Note that you can use ASPA command or AUNE command, according to the Numbering Plan.

STEP 1: **ASPA** - Available when developing the Self Office Code itself directly as the Access Code for Uniform Numbering. You are to assign the same number as the Office Code in the ACC parameter, and to develop the Access Code (ACC) as SRV = "UNIF: Office Termination". In the SKIP parameter, specify the digit number for the Office Code. By this data, when your system receives a call with the number information of Self Office Code + Station Number, the call is connected after the Self Office Code is deleted. Note that ASPA cannot be used for Uniform Numbering in the case of a CCIS network.

[Example 1]

1. A station at Office A dials "801-3111".
2. The number information "801-3111" is sent to Office B.
3. "801" (self-Office Code) is deleted at Office B, and the call is connected to Station (3111).



The following is the data assignment (at Office B) of Example 1 in the preceding figure.

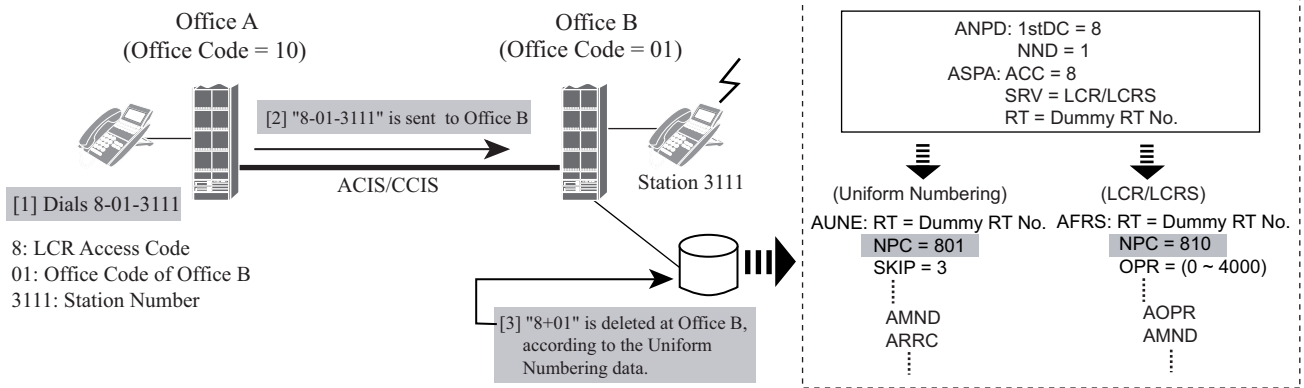
STEP 1: **ANPD** - Assign the first digit and digit number of Self Office Code.
1stDC (First Digit of Self Office Code) = 8
CI (Connection Index) = Select N (Normal Service) or H (Hooking Service) from the combo box.
NND (Number of Necessary Digits for Self Office Code) = 3

STEP 2: **ASPA** - Assign Self Office Code to be deleted.
ACC (Access Code) = 801 (Self Office Code)
CI (Connection Index) = "N: Normal"/"H: Hooking"
SRV = "UNIF: Office termination"
SKIP = 3 (Number of digits to be deleted)

STEP 3: **AUNE** - Available basically when first specifying the LCR/LCRS Access Code by the ANPD/ASPA, and then developing the Uniform Numbering data as its additional Number Pattern Code (NPC) table. When the NPC table is developed by the AFRS (and AOPR, AMND...) command, the table works for LCR/LCRS service. However, when the AUNE is used, the table works for Uniform Numbering. By this data, when a call arrives with the number information of LCR/LCRS Access Code + Self Office Code + Station Number, the system deletes the LCR/LCRS Access Code + Self Office Code to connect the call. Meanwhile, when the Node-to-Node Line is CCIS, you cannot use ASPA command. At this time, you must use AUNE for Uniform Numbering. When this command is used for a CCIS network, assign the LCR Access Code, assigned by ASPA, including the Self Office Code.

[Example 2]

1. A station at Office A dials "8-01-3111".
2. The number information "8-01-3111" is sent to Office B.
3. "8 + 01" (LCR Access Code + self-Office Code) is deleted at Office B, and the call is connected to Station (3111).



The following is the data assignment (at Office B) of Example 2 in the preceding figure.

STEP 1: **ANPD** - Assign the first digit and digit number of LCR/LCRS Access Code.
1stDC (First Digit of Access Code for LCR/LCRS) = 8
CI (Connection Index) = Select N (Normal Service) or H (Hooking Service) from the combo box.
NND (Number of Necessary Digits for the Access Code) = 1

STEP 2: **ASPA** - Assign the Access Code for LCR/LCRS.
ACC (Access Code) = 8 **Note 1**
CI (Connection Index) = "N: Normal"/"H: Hooking"
SRV = "LCR: Least Cost Routing"/"LCRS: Register Sender LCR"
RT (External Route Number) = Dummy Route Number

Note 1: When the network is CCIS, assign either "8" or "801".

STEP 3: **AUNE** - To delete Self Office Code, assign the Uniform Numbering data (including LCR/LCRS Access Code) for the dummy route assigned by the ASPA.
RT (External Route Number) = Dummy Route Number
NPC = 801 (LCR/LCRS Access Code + Self Office Code)
SKIP = 3 (The number of digits to be deleted)

STEP 4: **AMND** -Assign the MND (Maximum Necessary Digits) to be received.
DC = 801 (LCR/LCRS Access Code + Self Office Code)
MND = 7 (Maximum Necessary Digits)

STEP 5: **ARRC** -Allow the restriction between Incoming Route and Dummy Route.
ICRT = 1 (Incoming Route Number)
OGRT = Dummy Route Number
ARI-D_RES =1 (Tandem Connection without an operator/Station assistance is allowed)

10. PH Data Setting

This section gives the explanation of each PH and reference within each manual.
The DMOS command displays the status of the Telephony Server including TCP/IP Module, PH Module, SIP Server, and Standard SIP-GW.

10.1 MPH (Multi Protocol Handler)

MPH is an interface built-in the Telephony Server's SP which is used to connect TP and SP. MPH also has a function to control and monitor SP-PHD and SP-PHI.

<Reference>
Refer to the Appliance Model Installation Manual - Chapter 6 System Startup - 2. Additional information on System Startup - 2.1. Initial Data Assignment.

10.2 Internal PHE

Internal PHE is a type of PH embedded in the Telephony Server. Internal PHE enables the operation of SP Controlled SIP Terminals, MPH, MC functions, and VS-32 (Proprietary Protocol mode).

<Reference>

1. SP Controlled SIP Terminals

   a.) Basic Data Assignment

   Refer to the Appliance Model Installation Manual, Chapter 6 System Startup, 2. Additional information on System Startup, 2.1. Initial Data Assignment.

   b.) Assignment for Station Data

   Refer to the Data Programming Manual - Business, Chapter 3 Business Features, SP Controlled SIP Terminal [S-168].

2. MPH

   Refer to "10.1 MPH (Multi Protocol Handler)".

3. MC Functions

   a.) 2MC

   Refer to the Peripheral Equipment Description (IP Devices), Chapter 2 IP Devices, Analog 2MC (Analog Media Converter).

   For UG50-2MC, refer to the Peripheral Equipment Description (UG50).

   b.) UG50 (LC)/UG50-IPG (Analog)/UG50-4LC2COTA(COT)/UG50-4LC2COTA(PGT).

   Refer to the Peripheral Equipment Description (UG50).

c.) MC&MG-COT

Refer to the Peripheral Equipment Description (IP Devices), Chapter 2 IP Devices, MCMG [SCA-4LC2COTA/MG-4LC2COTA].

d.) IPG Analog

Refer to the Peripheral Equipment Description (IP Devices), Chapter 2 IP Devices, IP Gateway.

4. VS-32 (Proprietary Protocol mode)

Refer to the Data Programming Manual - Business, Chapter 3 Business Features, VS32 Conference Server [V-26].

10.3 SP-PHD

SP-PHD is a type of PHD accommodated in MPH. SP-PHD enables the operation of IP Multiple Line Terminal, SIP Multiple Line Terminal, Soft Phone, IPG Digital, UG50-IPG (Digital), and UG50 (DLC).

<Reference>

1. IP Multiple Line Terminal and Soft Phone

a.) Basic Data Assignment

Refer to the Appliance Model Installation Manual, Chapter 6 System Startup, 2. Additional information on System Startup, 2.1. Initial Data Assignment.

b.) SIP Multiple Line Terminal

Refer to the Data Programming Manual - Business, Chapter 3 Business Features, SIP Multiple Line Operation [S-160].

c.) IP Multiple Line Terminal

Refer to “14.1 IP Station Data” in this chapter.

d.) Soft Phone

Refer to “14.1 IP Station Data” in this chapter. and the Data Programming Manual - Business, Chapter 3 Business Features, Soft Phone [S-155].

2. SIP Multiple Line Terminals

a.) Basic Data Assignment

Refer to the Appliance Model Installation Manual, Chapter 6 System Startup, 2. Additional information on System Startup, 2.1. Initial Data Assignment.

b.) Station Data Assignment

Refer to the Data Programming Manual - Business, Chapter 3 Business Features, SIP Multiple Line Operation [S-160].

3. IPG Digital

   Refer to the Peripheral Equipment Description (IP Devices), Chapter 2 IP Devices, IP Gateway.

4. UG50-IPG (Digital)/UG50 (DLC)

   Refer to the Peripheral Equipment Description (UG50).

10.4 SP-PHI

SP-PHI is a type of PHI accommodated in MPH. SP-PHD enables the operation of MG(BRI), MG(PRI), MC&MG-COT, 8LC, UG50 (LC) [SIP] and VS-32 (SIP mode).

<Reference>

1. MG Data Assignment

   Refer to each section of Media Gateway in the Peripheral Equipment Description (IP Devices).

   For UG50 (PRT 1.5M) [SIP], UG50 (PRT 2M) [SIP], UG50-24PRIA [SIP], and UG50-30PRIA [SIP], refer to the Peripheral Equipment Description (UG50).

2. 8LC Data Assignment

   Refer to the Peripheral Equipment Description (IP Devices), Chapter 2 IP Devices, 8LC Card.

   For UG50-8LC, refer to the Peripheral Equipment Description (UG50).

3. UG50 (LC) [SIP], UG50 (COT-TYPE2) and UG50 (PGT-TYPE2) Data Assignment

   Refer to the Peripheral Equipment Description (UG50).

4. VS-32 (SIP mode) Data Assignment

   Refer to the Data Programming Manual - Business, Chapter 3 Business Features, VS32 Conference Server [V-26].

10.5 Internal PHI

Internal PHI is a type of PHI accommodated in the Telephony Server. There are two types of Internal PHI. Internal PHI (PRI) enables the operation of MG (PRI) (Proprietary Protocol mode) with TCP connection and Internal PHI (BRI) enables the operation of MG (BRI) (Proprietary Protocol mode), MC&MG-COT, and MG (SIP).

<Reference>

1. MG Data Assignment

   Refer to each section of Media Gateway in the Peripheral Equipment Description (IP Devices).

   For UG50 (COT-TYPE1), UG50 (COT-TYPE2), UG50-4LC2COTA (COT), UG50-6COT, UG50 (PGT-TYPE1), UG50 (PGT-TYPE2) and UG50-4LC2COTA (PGT) refer to the Peripheral Equipment Description (UG50).

10.6 Internal PHF

Internal PHF is an interface built-in the Telephony Server which is for transmitting the link information between nodes when the system is using FCCS networking. Only LAN1 can be used for transmitting the link information.

<Reference>
Refer to the Data Programming Manual - FCCS and the Data Programming Manual - Business, Chapter 3 Business Features, FCCS Networking via IP [F-36].

10.7 Internal PHC

Internal PHC is an interface built-in the Telephony Server which is for transmitting the link information between nodes when the system is using CCIS networking. LAN2 is used as a default setting for transmitting the link information, The LAN interface can be modified using the ADTM command.

<Reference>
Refer to the Data Programming Manual - CCIS and the Data Programming Manual - Business, Chapter 3 Business Features, CCIS Networking via IP [C-163].

10.8 Internal SIP Handler

Internal SIP Handler is an interface built-in the Telephony Server which is for controlling the Standard SIP Terminals without using SP. LAN1 (IP Address2 in ACT-side) is recommended for transmitting the signal packets.

<Reference>

a.) Basic Data Assignment

Refer to the Appliance Model Installation Manual, Chapter 6 System Startup, 2. Additional information on System Startup, 2.1. Initial Data Assignment.

b.) Assignment for Station Data

Refer to the Data Programming Manual - Business, Chapter 3 Business Features, SIP Handler - SIP Terminal [S-167].

10.9 Consideration for SP-PHI Data Assignment

When SP-PHI data is assigned, if MG registered to Internal PHI-BRI of TP is reset, MG automatically registers to SP-PHI of SP. Because MG control packet is sent through TP and SP-PHI, failure in SP might affect communication for MG. This automatic registration is regulated through System Data (ASYDL, SYS1, Index 1017, Bit 0). Refer to the below information and configure as needed.

Affected MG

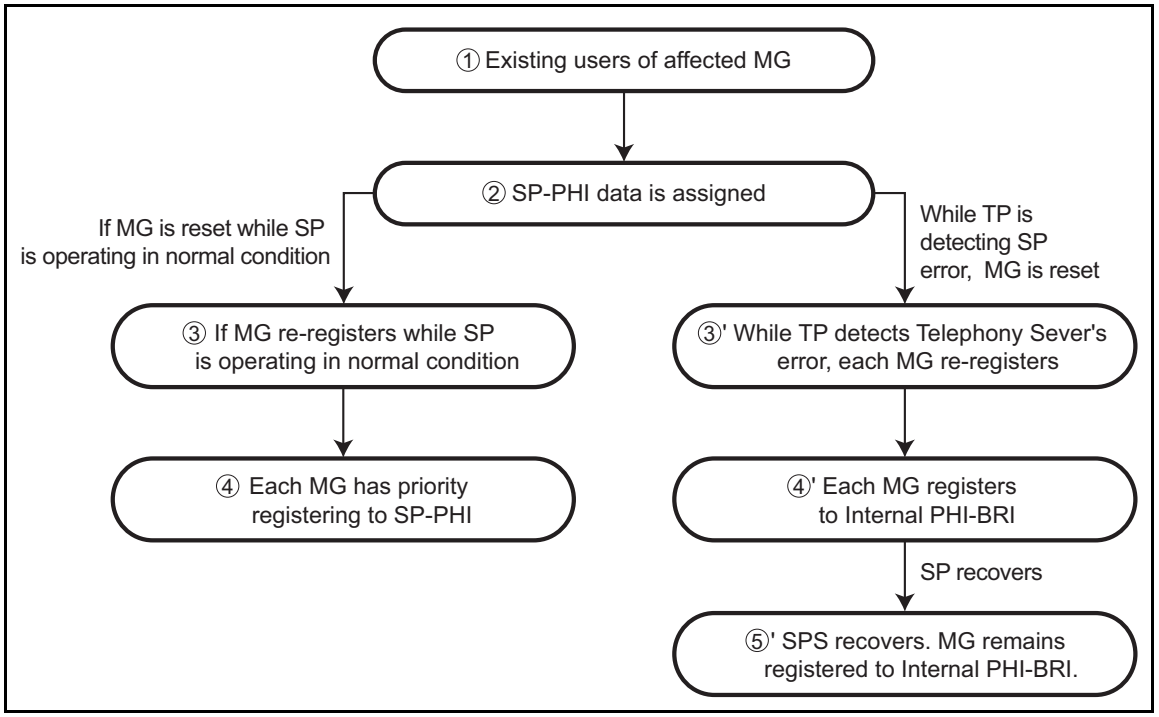
MG Types	Product Name
MC&MG-COT	MCMG [SCA-4LC2COTA/MG-4LC2COTA]
MG-COT (Proprietary Protocol)	MG-COT [SCA-6COTA]
	MG-COT [SCA-6COTB]
	MG-COT [SCA-6COTC]

The table below describes the registration destination for each MG. The shaded area indicates MG to be considered regarding the registration destination.

MG Types		Protocol	PH
IP equipment which registers only to TP	Analog MC	Proprietary Protocol	Internal PHE
	MG(PRI)		Internal PHI-PRI
	MG (SIP)		Internal PHI-BRI
	MC&MG-COT (MC Part)		Internal PHE
	VS32		Internal PHE
IP equipment which registers only to SP	Analog MC	SIP	SP-PHI
	MG(PRI)		
	MG(BRI)		
	VS32		
	MG-T1		
	MG-COT		
IP equipment which registers to TP or SP	MC&MG-COT (MG Part)	Proprietary Protocol	SP-PHI
	MG-COT		

Overview of Action

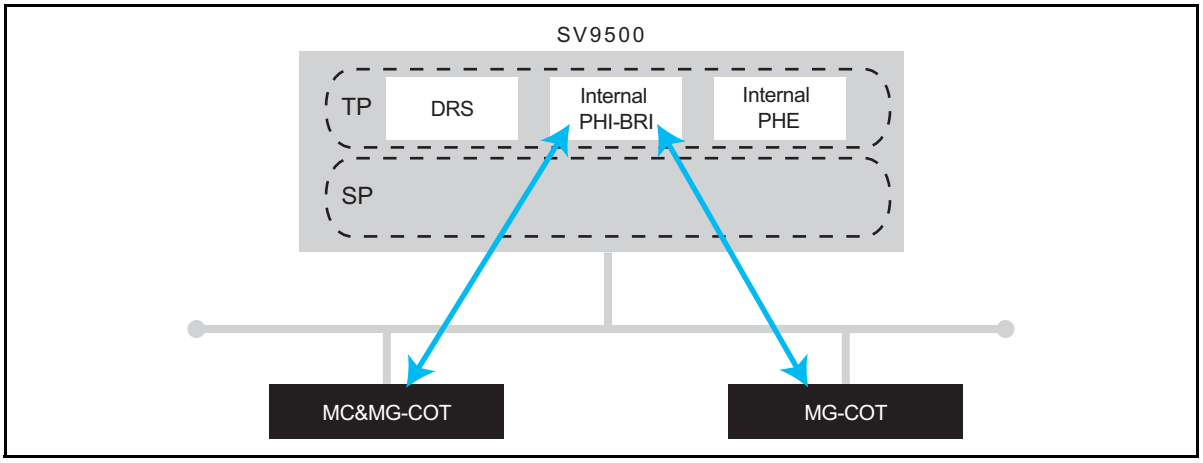
The following is the overview of action for MG when System Data (ASYDL, SYS1, Index, 1017, Bit 0=0) is in default.



(1) Existing users of affected MG

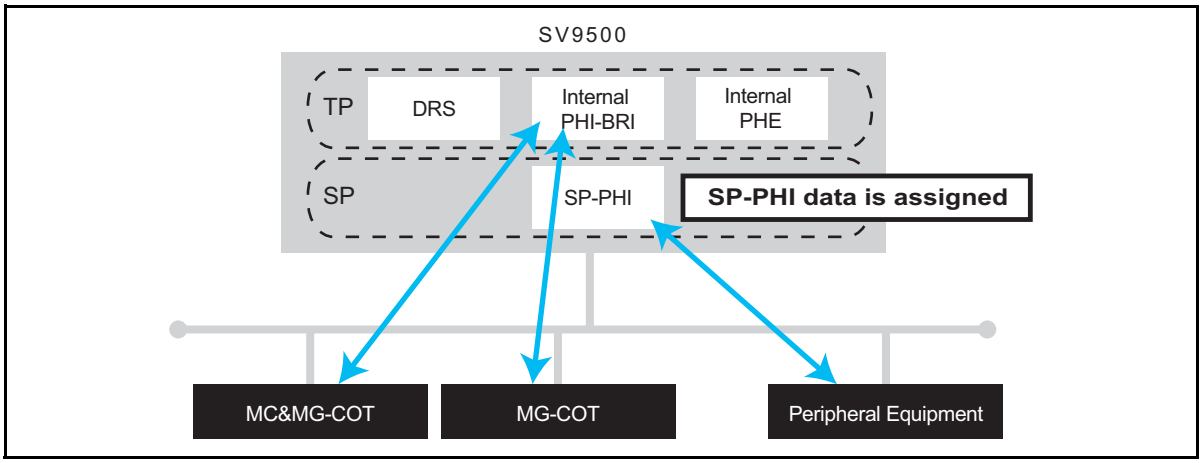
(a) Each affected MG registers to Internal PHI-BRI (MC registers to Internal PHE).

(b) Control packet is sent directly between MG and TP (including DRS). Even if error occurs in SP, there is no direct influence, such as resetting, for the MG.



(2) SP-PHI data is assigned

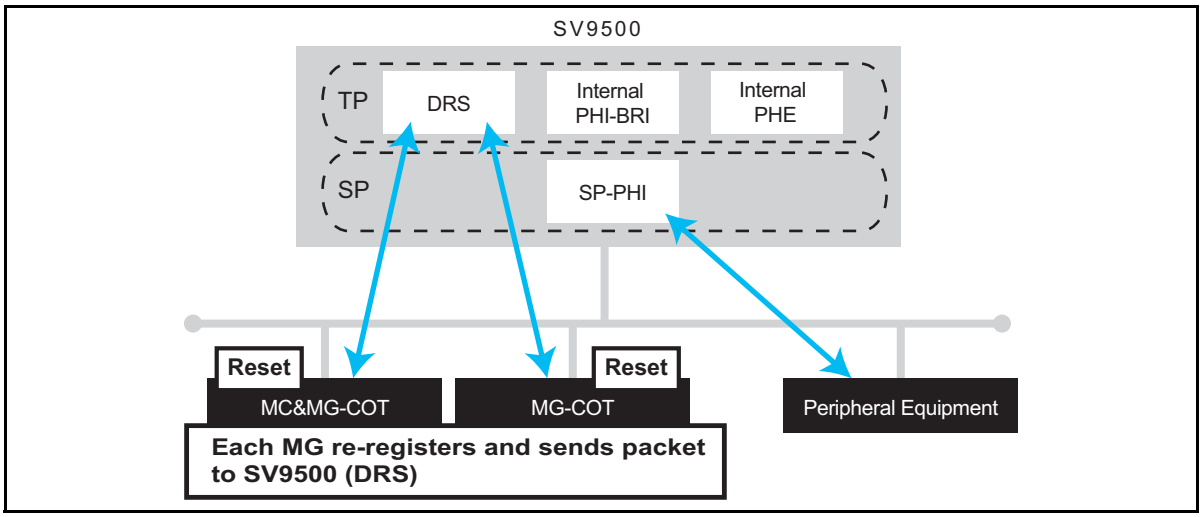
(a) MG remains registered to Internal PHI-BRI when SP-PHI data is assigned (MG has not yet been reset).



<If MG is reset while SP is operating in normal condition>

(3) If MG re-registers while SP is operating in normal condition

(a) When MG is reset, it sends packets to TP(DRS) to request registration.

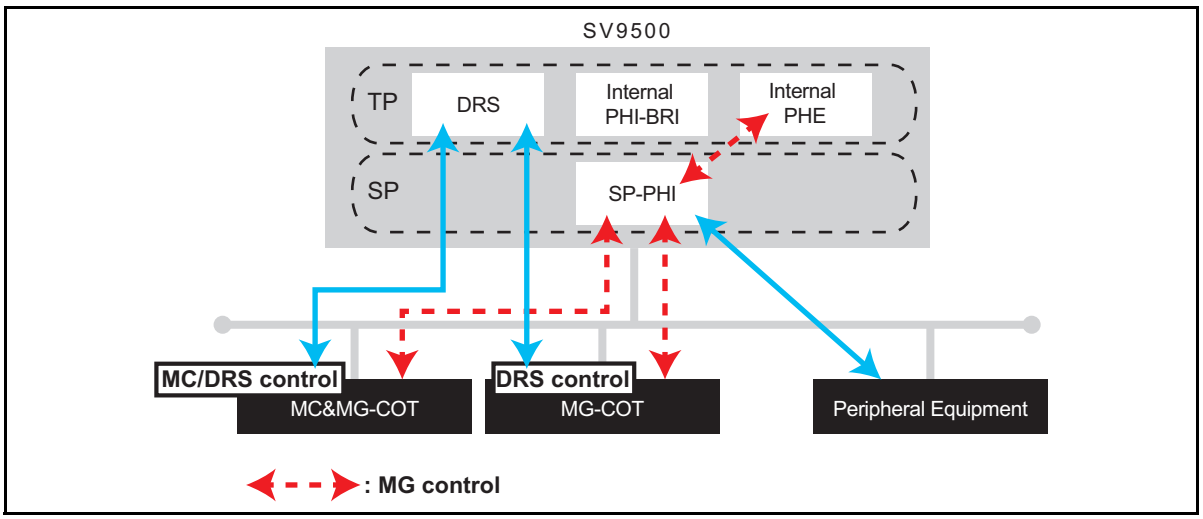


(4) MG has priority registering to SP-PHI (current default action)

(a) MC&MG-COT(MG) has priority registering to SP-PHI (MC registers to Internal PHE).

(b) MG control packet is sent through TP and SP-PHI. MC/DRS control packet is directly sent to TP (Internal PHE/DRS).

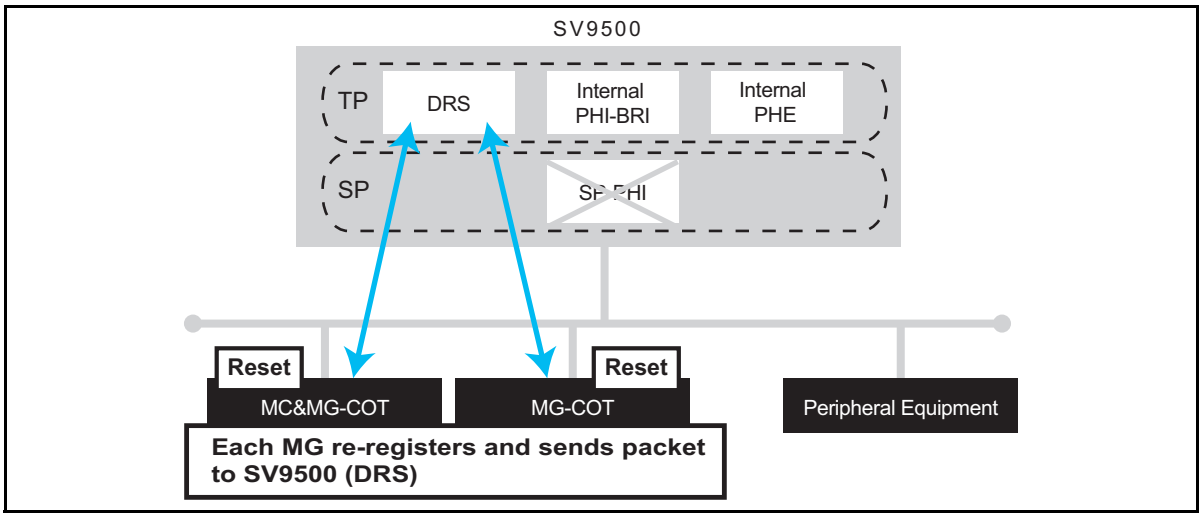
(c) Failure in SP might affect communication for MG and may cause it to reset. MG not affected by SP's error will be affected when SP-PHI data is assigned.



<While TP is detecting error in SP, affected MG is reset>

(3)' While TP is detecting error in SP, MG will re-register

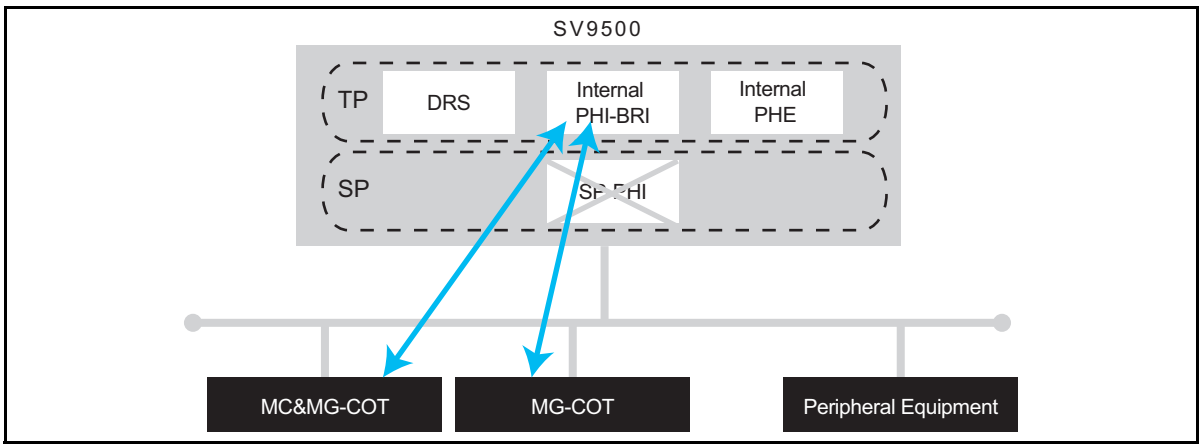
(a) When MG is reset, it sends packets to TP(DRS) to demand registration.



(4)' Each MG registers to Internal PHI-BRI

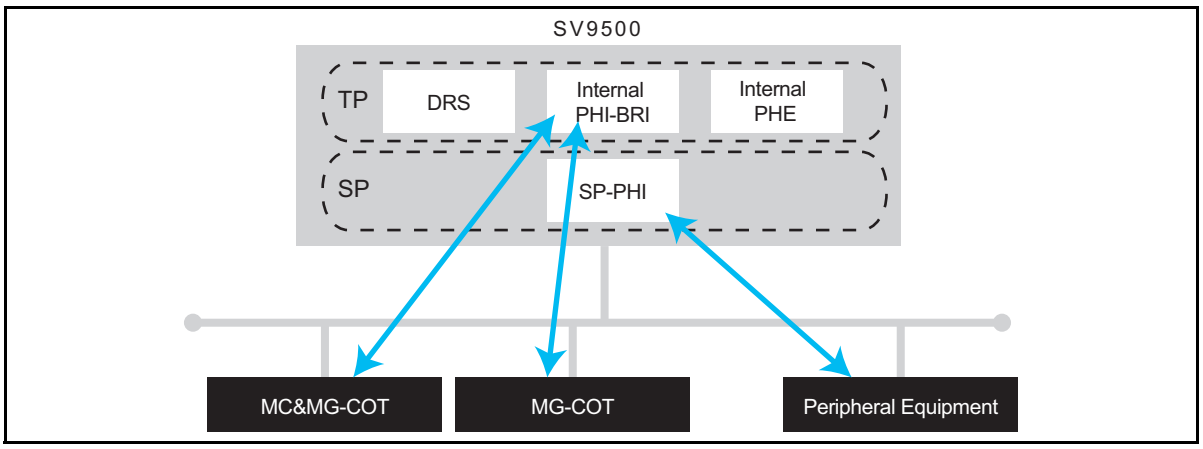
(a) During registration, if TP detects SP error, MG registers to Internal PHI-BRI (MC registers to Internal PHE).

(b) Control packet is sent directly between MG and TP (including DRS). Even if error occurs in SP, there is no direct influence, such as resetting, for the MG. Assigning data for SP-PHI does not necessarily cause MG to be influenced by SP error.



(5)' SP recovers. Affected MG remains registered to Internal PHI-BRI.

(a) After the recovery of SP, MG will remain registered to Internal PHI-BRI if it is not reset.



Programming

ASYDL, SYS1, Index 1017, Bit 0

<When Bit 0=0 (default)>
If MG registers while SP-PHI is in operation, MG will have priority registering to SP-PHI. If SP-PHI is not operating during registration, MG will register to TP's Internal PHI-BRI.

Effect: If affected MG is registered to SP-PHI, it will be influenced by SP error. If MG is registered to PHI, it is not influenced by SP error.

<When Bit 0=1>
MG registers to Internal PHI-BRI.

Effect: Assigning SP-PHI data will not cause MG to register to SP-PHI. Furthermore, pre-existing MG actions will not change, nor will MG be influenced by SP error.

11. Tone Setting

Here describes how to set tones for each countries. There are two type of tones, Basic Tones and EMEA Tones.

Note: For setting the tones for each terminals, refer to Multiple Tone - D [M-119D].

Note: When using the EMEA tones, refer to “Appendix A” or “Appendix B” in this manual for Tone Specification.

[Conditions]

1. The Basic Tones and EMEA Tones require the following DTG.

Basic Tones	DTG	EMEA Tones	DTG
North America	SM-025 N24 DTG DATA	Germany	SM-613 DTG DE DATA-A
Australia	SM-243 N24 DTG DATA-AUS SM-027 N24 DTG DATA-AUS	Italy	SM-623 DTG IT DATA-A
Hong Kong	SP-593 SW05-B PROG-A	Netherlands	SM-610 DTG NL DATA-A
Malaysia	SP-591 SW05-B PROG-A	Denmark	SM-619 DTG DK DATA-A
Singapore	SP-594 SW05-B PROG-A	Sweden	SM-612 DTG SE DATA-A
England	SM-560 DTG UK DATA-A	Spain	SM-621 DTG ES DATA-A
Mexico	SP-599 SW05-B PROG-A	Austria	SM-618 DTG AT DATA-A
Taiwan	SP-590 SW05-B PROG-A SM-349 N24 DTG DATA-A	Belgium	SM-611 DTG BE DATA-A
New Zealand	SM-328 N74 DTG DATA	England	SM-603 DTG UK DATA-A
Korea	SM-309 N24 DTG DATA-A SP-588 SW05-B PROG-A	Greece	SM-622 DTG GR DATA-A
Brazil	SM-283 N74 DTG DATA	Switzerland	SM-620 DTG CH DATA-A
China	SM-303 N74 DTG DATA-B	South Africa	SM-649 DTG SA DATA-A
Thailand	SM-201 N24 DTG DATA SM-346 N24 DTG DATA-A		
Thailand (EGAT)	SP-592 SW05-B PROG-A		
Sri Lanka	SP-594 SW05-B PROG-A		

Note: Be sure to select the same countries on the system data [ASYDL, SYS1, Index 811].

2. Tone assigned by the system data [ASYDL, SYS1, Index 811] will be unavailable if other country has been selected by the ASLG command (Multiple Tone - D [M-119D]).
3. If tone assigned by the system data [ASYDL, SYS1, Index 811] is not supported on each terminal, the default tone will be applied.
4. MG(PRI), 8LC card, UG50, VS32, and IP Terminal are required to install the firmware that is compatible with this feature.

Note: The South Africa tone is supported by the following firmware for the each equipment.
- MG(PRI): SP-3884 MG PRI PROG-A Issue 16.0 or later
- 8LC card: SP-3893 8MC PROG-A Issue 7.0 or later/SP-3983 8MC PROG-B Issue 1.0 or later
- UG50: No firmware condition
- VS32: SP-3891 CA-VS32 PROG-A Issue 6.0 or later

5. The following shows a listing of EMEA tones that are not supported by this feature.
   - Second External Dial Tone
   - Delayed Hotline Dial Tone
   - Paging Absent Tone
   - Paging In-Progress Tone
   - Bypass Tone
   - Through Connect Ring Tone
   - ARB Destination Ring Back Tone
   - Password Tone
   - Call Waiting Ticker
   - Dial Up Automatic Break-In Ticker
6. The EMEA tones take precedence over the Australia tone.

Programming

STEP 1: **ASYD** - SYS1, Index 59, Bit 6
Tone sending for Attendant and Override services (0/1 = Yes/No). When using EMEA tones, assign "1".

STEP 2: **ASYDL** - Assign the following system data.
SYS1
Index 811: Tone Code by Countries

ASYDL, SYS1, Index 811			
TONE CODE	**Basic Tones**	**TONE CODE**	**EMEA Tones**
00 H	Not used	C8 H	Germany
01 H	Japan	C9 H	Italy
02 H	North America	CA H	Netherlands
03 H	Australia	CC H	Denmark
04 H	Other countries	CD H	Sweden

ASYDL, SYS1, Index 811			
TONE CODE	**Basic Tones**	**TONE CODE**	**EMEA Tones**
05 H	Hong Kong	D3 H	Spain
06 H	Malaysia	D4 H	Austria
07 H	Singapore	D5 H	Belgium
08 H	England	D6 H	England [For EMEA]
09 H	Mexico	D7 H	Greece
0A H	Taiwan	D8 H	Switzerland
0B H	New Zealand	D9 H	South Africa
0C H	Korea	Other Values	Not used
0D H	Brazil		
0E H	China		
0F H	Thailand		
10 H	Thailand (EGAT)		
11 H	Sri Lanka		

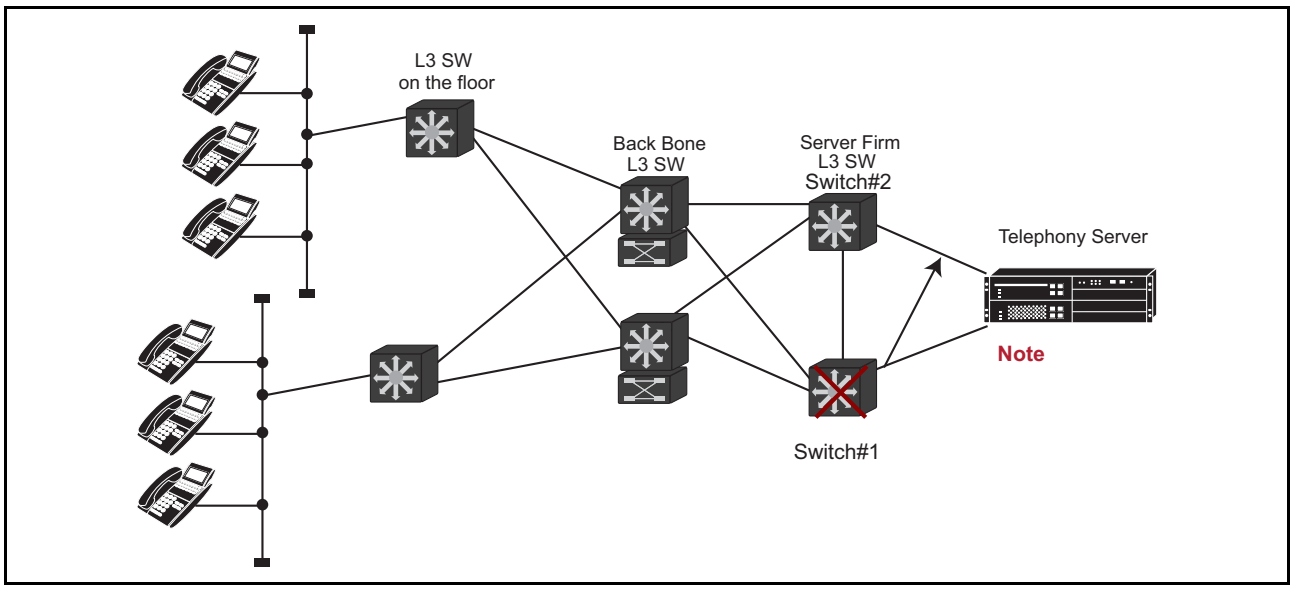
Index 830, Bit 7=0/1: Hold Tone can be changed when one of the countries for EMEA has been set in ASYDL, SYS1, Index 811.

ASYDL, SYS1, Index 830	
Bit 7	**Hold Tone**
1	Music **Note 1**
0	Tone for EMEA

Note 1: When customer-provided Music is used, music source must be connected to TSW-BOX or PLO card. For more details, please refer to Appliance Model Installation Manual - CHAPTER 4 MODULES MOUNTING AND CONNECTION (7UPIR CONFIGURATION) - “Cable Connections” - “External Music-On-Hold Cable Connection”.

12. Data Setting for Switch Redundancy Configuration

This section explains the data settings required when placing the Telephony Server in the redundant network.
As shown in the figure below, if the Telephony Server is configured redundantly by using the network equipment such as Layer 2 Switch and Layer 3 Switch, the particular data settings are required.



As shown in the above figure, when Switch#1 goes down while CPU#0 operates in Active mode, CPU#1 starts operating in Active mode instead of CPU#0.
(Only when CPU#0 detects the Layer 1 link disconnection after the Switch#1 failure.*)*

Note: Required to assign the same network address to CPU#0 and TP#1.

Programming

Perform System Change over Test associated with detection of the Layer 1 Link disconnection after the following data programming. For System Change over Test, refer to Chapter 7 Installation Test of the Appliance Model Installation Manual.

• For LAN1

STEP 1: **ASYDL** - SYS1, Index 1344, Bit 0 to Bit 2: LAN Cable Layer 1 Fault Detection Monitor Timer of LAN1 connector.

Bit 3=1: System changeover is executed when detecting the Layer 1 link disconnection.

Bit 3	Bit 2	Bit 1	Bit 0	Changeover Operation Time
1	0	0	0	Starts immediately, and ends about 12 seconds later.
1	0	0	1	Starts after 6-8 seconds, and ends about 12 seconds later.
1	0	1	0	Starts after 10-12 seconds, and ends about 12 seconds later. (Standard setting)
1	0	1	1	Starts after 16-17 seconds, and ends about 12 seconds later.
1	1	0	0	Starts after 20-22 seconds, and ends about 12 seconds later.
1	1	0	1	Starts after 26-27 seconds, and ends about 12 seconds later.
1	1	1	0	Starts after 30-32 seconds, and ends about 12 seconds later.
1	1	1	1	Starts after 37-38 seconds, and ends about 12 seconds later.

STEP 2: **ADTM** - TCP/IP Module Data.
NIC
ARP Broadcasting Period
Choose “30 sec”, “60 sec”, or “120 sec”.

Note: For the dual system configuration, it is recommended to enable the periodic transmission of G-ARP on both LAN1 and LAN2 with the ADTM command. The parameter of ARP Broadcasting Period should be preferably selected as “60 sec”. As for the offices that frequently execute the CPU system changeover, select “30 sec” for the parameter of ARP Broadcasting Period.

• For LAN2

STEP 1: **ASYDL** - SYS1, Index 598, Bit 4 to Bit 6: LAN Cable Layer 1 Fault Detection Monitor Timer of LAN2 connector.

Bit 7=1: System changeover is executed when detecting the Layer 1 link disconnection.

Bit 7	Bit 6	Bit 5	Bit 4	Changeover Operation Time
1	0	0	0	Starts immediately, and ends about 12 seconds later.

Bit 7	Bit 6	Bit 5	Bit 4	Changeover Operation Time
1	0	0	1	Starts after 6-8 seconds, and ends about 12 seconds later.
1	0	1	0	Starts after 10-12 seconds, and ends about 12 seconds later. (Standard setting)
1	0	1	1	Starts after 16-17 seconds, and ends about 12 seconds later.
1	1	0	0	Starts after 20-22 seconds, and ends about 12 seconds later.
1	1	0	1	Starts after 26-27 seconds, and ends about 12 seconds later.
1	1	1	0	Starts after 30-32 seconds, and ends about 12 seconds later.
1	1	1	1	Starts after 37-38 seconds, and ends about 12 seconds later.

STEP 2: **ADTM** - TCP/IP Module Data.
NIC
ARP Broadcasting Period
Choose “30 sec”, “60 sec”, or “120 sec”.

Note: For the dual system configuration, it is recommended to enable the periodic transmission of G-ARP on both LAN1 and LAN2 with the ADTM command. The parameter of ARP Broadcasting Period should be preferably selected as "60 sec". As for the offices that frequently execute the CPU system changeover, select “30 sec” for the parameter of ARP Broadcasting Period.

• For LAN3/LAN4

ASYDL - SYS1, Index 927, Bit 0 to Bit 2: LAN Cable Layer 1 Fault Detection Monitor Timer of LAN3/LAN4 connector.

Bit 3=1: System changeover is executed when detecting the Layer 1 link disconnection at LAN3/LAN4 connector.

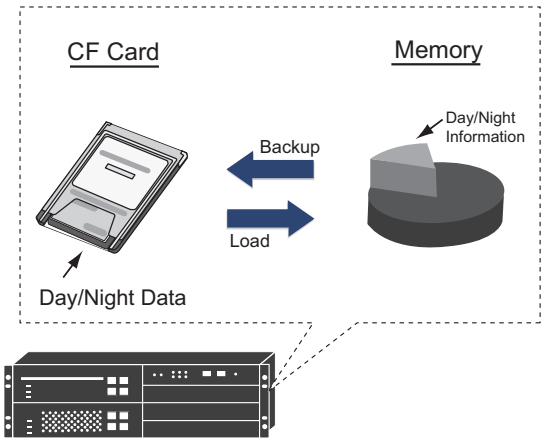
Bit 3	Bit 2	Bit 1	Bit 0	Changeover Operation Time
1	0	0	0	Starts immediately, and ends about 12 seconds later.
1	0	0	1	Starts after 6-8 seconds, and ends about 12 seconds later.
1	0	1	0	Starts after 10-12 seconds, and ends about 12 seconds later. (Standard setting)
1	0	1	1	Starts after 16-17 seconds, and ends about 12 seconds later.
1	1	0	0	Starts after 20-22 seconds, and ends about 12 seconds later.
1	1	0	1	Starts after 26-27 seconds, and ends about 12 seconds later.
1	1	1	0	Starts after 30-32 seconds, and ends about 12 seconds later.
1	1	1	1	Starts after 37-38 seconds, and ends about 12 seconds later.

13. Day/Night Data Retention

When system is initialized, this feature allows the system to be used in the day/night status before the system initialization by loading the backed up day/night data without setting the day/night mode again.

Service Conditions

1. For using this feature, the backup for day/night data is required. The backup method for day/night data, loading method for the backed up day/night data, and the day/night data verification is as follows.



a.) Automatic/manual backup for day/night data into CF card when changing the day/night mode.

- Manual Backup (by MEM_HDD/MEM_HDD_N command)

  From PCPro, execute **MEM_HDD/MEM_HDD_N command**, select [**Memory to Hard Disk**] → [**Day/Night data**], and press [**Start**]. For the detail, refer to Operations and Maintenance Manual - CHAPTER 2 Operations of Telephony Server - Office Data Maintenance - Office Data Backup. Use Telephony Server Maintenance Menu to back it up to PCPro.

- Automatic Backup (by assigning system data - ASYDL, SYS1, Index 1120, Bit 0 to Bit 5)

  Day/night data can automatically be backed up in a certain cycle. For enabling the Automatic Backup, the following data setting is needed. For the setting data of periodical backup activation time, refer to [Example for the setting of day/night data automatic backup cycle] in Programming. The automatic backup into CF card is not executed unless the Day/Night information is changed.

  [ASYDL, SYS1, Index 1120]

  Bit 0: Day/Night Data Automatic Backup, 0/1 = Out of service/In service
  Bit 1 to Bit 3: Backup Cycle,

Bit 3	Bit 2	Bit 1	Backup Cycle
0	0	0	every 2 hours (default)
0	0	1	every 1 hour

Bit 4: Standard Time for Starting Backup, 0/1 = 0/1 (hour)
Bit 5: Standard Time for Starting Backup, 0/1 = 0/30 (minutes)

[Conditions for Automatic Backup]

- If the time of day/night data automatic backup and routine diagnosis program start time (ASYD, SYS1, Index 87 and Index 88) are overlapped, the automatic backup will be executed while the routine diagnosis program is being executed. The time for starting automatic backup will be changed depending on routine diagnosis item (ASYD, SYS1, Index 89 and Index 90).
- If the time for starting automatic backup is within 5 minutes before the diagnosis program start time, the automatic backup will not be executed.
- The automatic backup will not be executed while the routine diagnosis program is being executed.
- The automatic backup will not be executed while office data is being copied to SR-MGC.
- If the Telephony Server (TP (CPU)) is configured redundant, the automatic backup will not be executed during the changeover of the CPU. Also, the changeover cannot be executed during the automatic backup.
- The automatic backup will not be executed while saving/loading/verifying office data by PCPro.

b.) Day/night data load after system initialization involving office data load (by Telephony Server Maintenance Menu)

Note: Use Telephony Server Maintenance Menu to load the backup data to PCPro. For detailed information, refer to Operations and Maintenance Manual.

Develop the backup data in Flash Card on system memory. For the detail, refer to Operations and Maintenance Manual - Chapter 7 Maintenance Commands.

1. In the case of system initial setting involving office data load:
The backup data is valid. **Note 1**

2.In the case of system initial setting not involving office data load:
The day/night data before the initialization is used.

3.In the case of system restart from while the power is off:
The backup data is used. **Note 1**

Note 1: Night mode 3 (N3) is used if the day/night data load failed when initialized.

c.) Day/night data verification (by Telephony Server Maintenance Menu and MEM_HDD/ MEM_HDD_N command) **Note 2**

Note 2: Use Telephony Server Maintenance Menu to verify the backup data in PCPro. For detailed information, refer to Operations and Maintenance Manual.

Verify the Day/Night data after the backup is finished. The verifiable data is as follows. For the details, refer to Operations and Maintenance Manual - Chapter 7 Maintenance Commands.

- Between HDD (CF card) and MEM (Memory on system)

d.) Display/deletion of the day/night backup data (by DHDF command)

From PCPro, execute **DHDF command**, select [**Day/Night data**], and press [**Get**] or [**Del**] button. For the datal, refer to Operations and Maintenance Manual - CHAPTER 2 Operations of Telephony Server - Office Data Maintenance - Office Data Backup.

2. Day/night data cannot be copied to SR-MGC. Therefore, the day/night data cannot be retained using this feature.
3. If using this feature in an environment where attendant console is accommodated, the day/night status after system initialization will be updated by the mode that is set at switch of ATI circuit cards (CH-CS00: SW12-8 (software switch), PA-CS33-C/PA-CS33: SW17-8). For the detail, refer to Circuit Card Description.
4. If using this feature in an environment where external key is accommodated, the day/night status after system initialization will be updated by the setting of external key.
5. If using this feature in FCCS network, the day/night status will be updated by the setting of NCN.

Programming

ASYDL - SYS1, Index 1120:
Bit 0: Day/Night Data Automatic Backup, 0/1 = Out of service/In service
Bit 1 to Bit 3: Backup Cycle

Bit 3	Bit 2	Bit 1	Backup Cycle
0	0	0	every 2 hours (default)
0	0	1	every 1 hour

Bit 4: Standard Time for Starting Backup, 0/1 = 0/1 (hour)
Bit 5: Standard Time for Starting Backup, 0/1 = 0/30 (minutes)

[Example for the setting of day/night data automatic backup cycle]

"AM 3:25" as current time:

if setting the standard time for starting backup as "AM 1:00" and backup cycle as "every 1 hour":

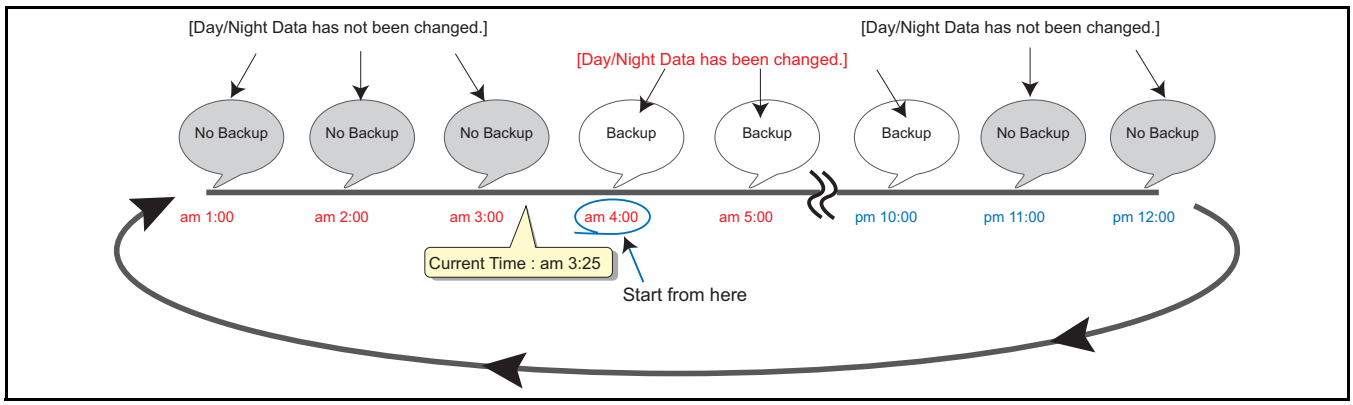
<Data Setting>

ASYDL, SYS1, Index 1120, Bit 4=0, Bit 5=1: Standard Time for Starting Backup is "AM 1:00"

ASYDL, SYS1, Index 1120, Bit 1=1: Day/night data is backed up every 1 hour from "AM 1:00" above.

<Backup Execution Time>

Because current time is "AM 3:25", the first backup will be executed at "AM 4:00". The backup will continues to be executed every 1 hour from now on. The automatic backup into CF card is not executed unless the Day/Night information is changed.



Ex: Automatic Backup Activation Time Table for Day/Night Data

Time	Standard Time for Starting Backup (ASYDL, SYS1, Index 1120)							
	Bit 5=0 (0 h) Bit 4=0 (0 min)		Bit 5=0 (0 h) Bit 4=1 (30 min)		Bit 5=1 (1 h) Bit 4=0 (0 min)		Bit 5=1 (1 h) Bit 4=1 (30 min)	
	0:00		0:30		1:00		1:30	
	Bit 1,2,3=0 (Every 2 hours)	Bit 1=1, Bit 2,3=0 (Every 1 hour)	Bit 1,2,3=0 (Every 2 hours)	Bit 1=1, Bit 2,3=0 (Every 1 hour)	Bit 1,2,3=0 (Every 2 hours)	Bit 1=1, Bit 2,3=0 (Every 1 hour)	Bit 1,2,3=0 (Every 2 hours)	Bit 1=1, Bit 2,3=0 (Every 1 hour)
00:00	×	×				×		
00:30			×	×				×
01:00		×			×	×		
01:30				×			×	×
02:00	×	×				×		

Ex: Automatic Backup Activation Time Table for Day/Night Data

Time	Standard Time for Starting Backup (ASYDL, SYS1, Index 1120)							
	Bit 5=0 (0 h) Bit 4=0 (0 min)		Bit 5=0 (0 h) Bit 4=1 (30 min)		Bit 5=1 (1 h) Bit 4=0 (0 min)		Bit 5=1 (1 h) Bit 4=1 (30 min)	
	0:00		0:30		1:00		1:30	
	Bit 1,2,3=0 (Every 2 hours)	Bit 1=1, Bit 2,3=0 (Every 1 hour)	Bit 1,2,3=0 (Every 2 hours)	Bit 1=1, Bit 2,3=0 (Every 1 hour)	Bit 1,2,3=0 (Every 2 hours)	Bit 1=1, Bit 2,3=0 (Every 1 hour)	Bit 1,2,3=0 (Every 2 hours)	Bit 1=1, Bit 2,3=0 (Every 1 hour)
02:30			×	×				×
03:00		×			×	×		
03:30				×			×	×
04:00	×	×				×		
04:30			×	×				×
05:00		×			×	×		
05:30				×			×	×
06:00	×	×				×		
06:30			×	×				×
07:00		×			×	×		
07:30				×			×	×
08:00	×	×				×		
08:30			×	×				×
09:00		×			×	×		
09:30				×			×	×
10:00	×	×				×		
10:30			×	×				×
11:00		×			×	×		
11:30				×			×	×
12:00	×	×				×		
12:30			×	×				×
13:00		×			×	×		
13:30				×			×	×
14:00	×	×				×		

Ex: Automatic Backup Activation Time Table for Day/Night Data

Time	Standard Time for Starting Backup (ASYDL, SYS1, Index 1120)							
	Bit 5=0 (0 h) Bit 4=0 (0 min)		Bit 5=0 (0 h) Bit 4=1 (30 min)		Bit 5=1 (1 h) Bit 4=0 (0 min)		Bit 5=1 (1 h) Bit 4=1 (30 min)	
	0:00		0:30		1:00		1:30	
	Bit 1,2,3=0 (Every 2 hours)	Bit 1=1, Bit 2,3=0 (Every 1 hour)	Bit 1,2,3=0 (Every 2 hours)	Bit 1=1, Bit 2,3=0 (Every 1 hour)	Bit 1,2,3=0 (Every 2 hours)	Bit 1=1, Bit 2,3=0 (Every 1 hour)	Bit 1,2,3=0 (Every 2 hours)	Bit 1=1, Bit 2,3=0 (Every 1 hour)
14:30			×	×				×
15:00		×			×	×		
15:30				×			×	×
16:00	×	×				×		
16:30			×	×				×
17:00		×			×	×		
17:30				×			×	×
18:00	×	×				×		
18:30			×	×				×
19:00		×			×	×		
19:30				×			×	×
20:00	×	×				×		
20:30			×	×				×
21:00		×			×	×		
21:30				×			×	×
22:00	×	×				×		
22:30			×	×				×
23:00		×			×	×		
23:30				×			×	×

×: Activation time for automatic backup

14. IP Related Data Setting

14.1 IP Station Data

This section describes basic data assignment of IP station such as DT800/DT700 Series, DtermIP and Soft Phone.

Note: For the data assignment of SP Controlled SIP terminals, refer to SP CONTROLLED SIP TERMINAL [S-168] in Chapter 3.
For data assignment of stations accommodated in Analog MC, refer to the Peripheral Equipment Description (IP Devices).
For data assignment of stations accommodated in UG50, refer to the Peripheral Equipment Description (UG50).

Note: IP Terminal Automatic Assignment Function is not supported by SIP Multiple Line Terminal.

Note: In the UMGi system, all of Internal PHE, MPH, SP-PHD and IP stations must be accommodated in the same node (CCN/RN).

Programming

STEP 1: **ASYDL** - SYS1, Index 811, Bit 0 to Bit 7. Assign Tone Code by countries.
For the data assignment of Tone Code by countries, refer to page 124.

SYS1, Index 959, Bit 0 to Bit 7. Assign the cycle of periodic Health Check
00 = 60 seconds (Default)
01-FE = 1-254 seconds
FF = Health Check is not required.
* between PHD and IP stations: Values assigned by the above System Data.
between System and PHD: (Values assigned by the above System Data) x 1.5

Note: This data is required only when using SR-MGC.

SYS1, Index 880, Bit 3=1 (Internal PHE is valid)

SYS1, Index 885, Bit 7=1 [SP (Signaling Processor) implementation]

STEP 2: **ANPD** - Assign 1st Digit Code of Station Number and the total digit number of Station Number or "ACC" to be specified by ASPA. For details, see "4. Station Data" in this Chapter.

STEP 3: **ASPA** - Assign the Access Code of Station Number. For details, see "4. Station Data" in this Chapter.

STEP 4: **ALIDL** - Assign the Internal PHE and SP-PHD accommodated locations (LENS) data. Refer to "14.8 PAD/EC Control IP/PHC Support".

STEP 5: **AISTL** - Assign the IP Station data of each IP Station.

TN: Tenant Number [1-63]

IP STN: Station Number of each IP Station (a maximum of 5 digits)

KIND: Select either item in combo box, referring to the following table.

TERMINAL	ITEM
DT800 Series/DT700 Series, DtermIP (SIP)	SIP MultiLine TEL
DtermIP (Proprietary Protocol), IP Enabled Dterm	DtermIP
Soft Phone (SIP)	SIP SoftPhone
Soft Phone (Proprietary Protocol)	Soft Dterm
DtermIP INASET	INASET

TEC: "TEC (Telephone Equipment Class)" for IP Enabled Dterm designation.
TEC 12 = Dterm (voice)/13 Connection to data terminal via Dterm (using adapter)
TEC 16 = Data Terminal via Data Module

RSC: Route Restriction Class [0-15]

SFC: Service Feature Restriction Class [0-15]

LENS: IP Station Equipment Number. Allocate the IP Enabled Dterm to the LENS (Physical PIR or Virtual PIR **Note 1**) of vacant Slot on which the Station/Trunk is not mounted. (IP Stations cannot be assigned to the already registered Station in duplicate.)

Note 1: IP Stations can be allocated on a physically non-existent PIR (called Virtual PIR). When using Virtual PIR, the particular data assignment is required. For more details, refer to "14.8 PAD/EC Control IP/PHC Support" in this Chapter.

STEP 6: **ALOCL**- Assign the Location ID data. Each IP Station is given a "Location ID" that identifies it's location within the network, per a network address basis. By using this Location ID, IP Services such as PEER TO PEER BANDWIDTH CONTROL [P-75] and ToS Control can be provided for each IP Station.

TYPE: Select "Network Address".

IP ADDRESS: Assign the IP Address. For the System (IPPAD), assign "0.0.0.0."

MASK BIT: Assign the MASK BIT in the range 1-31 except when IP ADDRESS 0.0.0.0 is determined. When IP address = 0.0.0.0, MASK BIT = 32 is automatically assigned.

LOC-ID: Assign any number in the range 0-4095, associated with the IP address assigned in the IP ADDRESS parameter. When 0.0.0.0 is assigned in the IP ADDRESS parameter, LOC-ID is automatically set to "0" (the value cannot be changed).

Note: LOC-ID=0 is predetermined to the location ID of System (IPPAD).

- Check the Setting Type of Signaling Packet Service check box to assign the following ToS for Signaling Packet data.

Note: If the check box is not selected, the default settings (IP PRECEDENCE: Priority = 6, Low Delay = 0, Through put = 0, Reliability = 0, Min Cost = 0) will be provided.

Select either “IP PRECEDENCE” or “DIFFSERV”.
When IP PRECEDENCE is selected:
Priority (PRECEDENCE / DSCP): Specify the precedence in the range 0-7 (low-high).
Low Delay: Place a checkmark in the checkbox to set Low Delay to ON.
Through put: Place a checkmark in the checkbox to set High Throughput to ON.
Reliability: Place a checkmark in the checkbox to set High Reliability to ON.
Min Cost: Place a checkmark in the checkbox to set Cost Precedence to ON.
When DIFFSERV is selected:
Priority (PRECEDENCE / DSCP): Specify the DSCP in the range 0-63.

STEP 7: **AIVCL**- Assign the IP service Voice Control data.
A LOC-ID: Location ID of the self-location. (0-4095)
B LOC-ID: Location ID of the target location. (0-4095)

Note: LOC-ID=0 is used for the Location ID of System (IPPAD).

Note: Only the Location ID corresponding to the network address can be assigned on the AIVCL/AIVCN when assigning settings for locally programmed terminals. The Location IDs corresponding to FPC or STN cannot be assigned.

[Setting of Jitter Buffer]
Select the Setting of Jitter Buffer check box to assign the following Jitter Buffer data.

Note: If the check box is not selected, the default setting (JB MIN = 1 (10ms), JB MAX = 30 (300ms)) will be provided.

JB MIN: Jitter Buffer minimum value
JB MAX: Jitter Buffer maximum value

[Setting Type of Voice Packet Service]
Select the Setting Type of Voice Packet Service check box to assign the following ToS for Voice Packet data.

Note: If the check box is not selected, the default settings (IP PRECEDENCE, Priority (PRECEDENCE/DSCP) = 5, Low Delay = 0, Through Put = 0, Reliability = 0, Min Cost = 0) will be provided.

Choose either IP PRECEDENCE or DIFFSERV from the drop-down list box.
When IP PRECEDENCE is chosen:
Priority (PRECEDENCE / DSCP): Specify the precedence in the range 0-7 (low-high).
Low Delay: Place a checkmark in the checkbox to set Low Delay to ON.
Through Put: Place a checkmark in the checkbox to set High Throughput to ON.
Reliability: Place a checkmark in the checkbox to set High Reliability to ON.
Min Cost: Place a checkmark in the checkbox to set Cost Precedence to ON.
When DIFFSERV is chosen:
Priority (PRECEDENCE / DSCP): Specify the DSCP in the range 0-63.

[Setting of Payload]

Select the Setting of Payload check box to assign the following Payload data.

PRI1-PRI4: Select a payload type (TYPE) and payload size (SIZE) in order of precedence (PRI1(high) → PRI4 (low))

TYPE (Payload Type):

- •G.711
- •G.729a
- •G.723.1 (5.3K → 6.3K)
- •G.723.1 (6.3K → 5.3K)
- •G.722 **Note 1**, **Note 2**
- •AMR-NB **Note 1**
- •AMR-WB **Note 1**
- •G.711 μ law (FEC) **Note 1**, **Note 3**
- •G.711 A law (FEC) **Note 1**, **Note 3**
- •G.729a (FEC) **Note 1**, **Note 3**
- •G.722 (FEC) **Note 1**, **Note 3**
- •AMR-NB (FEC) **Note 1**, **Note 3**
- •AMR-WB (FEC) **Note 1**, **Note 3**

Note 1: These Payload Types is available for DtermSP30 Version 15. These data are effective only for the connection between DtermSP30s Version 15.

Note 2: Payload Type G.722 has been effective for the following connections in addition to the connection between DtermSP30s Version 15:

- The connection between DT800 Series (DT830/DT830G)
- The connection between DT700 Series (DT730/DT730G/DT750)
- The connection between DtermSP30 Version 15 and DT700 Series (DT730/DT730G/DT750)

Note 3: When Payload Type with “FEC” is selected, the data that is not transmitted correctly because of packet loss or bit error can be corrected according to the redundant data by the method of Forward Error Correction (FEC).

SIZE (Payload Size):

- •10ms
- •20ms
- •30ms
- •40ms
- •60ms

Note: For available payload type/payload size, refer to “Bandwidth Required for Voice Traffic” in CHAPTER 3 IP EQUIPMENT of System Description.”

Note: Soft Client SP350 in R3 or earlier cannot be used as ACD terminal.

Note: When no data related to Payload is set, default value (Type: G.711 (A/u), Size: 40 milliseconds) is used.

Note: If payload type G.711 is specified to more than two priorities in PRI 1 through PRI 4, assign different payload size to each priority.
- When payload types G.723.1 (5.3K → 6.3K) and G.723.1 (6.3K → 5.3K) are to be assigned on the same terminal, specify the same value to the payload size of both types.
- It is not necessary to assign the payload type/size to all priorities. Specifying the first priority only is allowed.

[Setting of PAD]

Select the "Setting of PAD" check box to assign the following PAD data.

KIND1: PAD value for the IP terminal (0-15)

INPUT DATA	0	1	2	3	4	5	6	7	8	9-15
dB	No PAD Control	2 dB	4 dB	6 dB	8 dB	10 dB	12 dB	14 dB	16 dB	PAD OFF (0 dB)

KIND2: PAD value for Analog/Digital MC (0-3) Refer to Peripheral Equipment Description.
KIND3: PAD value for IPPAD card (0-15)
For details on IPPAD PAD data assignment, refer to the Appliance Model Installation Manual.

[Setting of Echo Canceller]

Select the "Setting of Echo Canceller" check box to assign the following Echo Canceler data.

Select the Echo Canceler value by clicking one of the following radio buttons:
- A EC-ON/B EC-ON
- A EC-ON/B EC-OFF
- A EC-OFF/B EC-ON
- A EC-OFF/B EC-OFF

Note: When the same value is assigned to both the A LOC-ID and B LOC-ID parameters, select either A EC-ON/B EC-ON or A EC-OFF/B EC-OFF.

STEP 8: **AKYD** - Assign My Line, Multi Line or desired feature to each Line/Feature Key on IP station. **Note 4**, **Note 5**

Note 4: For Soft Phone, be sure to assign the following key data to a Programmable Feature key/Flex key of the Soft Phone.
- SPEAKER (When the Flex key data has been changed)
- LOGOUT
- CALL FORWARDING - BUSY LINE (When using the Call Forward Settings Panel of the Soft Phone)
- CALL FORWARDING - ALL CALLS (When using the Call Forward Settings Panel of the Soft Phone)
- CALL FORWARDING - DON'T ANSWER (When using the Call Forward Settings Panel of the Soft Phone)
- CALL FORWARDING - LOGOUT (When using the Call Forward Settings Panel of the Soft Phone)
- that you do.

Note 5: For DtermIP INASET, same number must be assigned to the parameters STN and PL STN.

STEP 9: **ADSL** - Assign desired feature to each Soft Key of IP station. For DtermIP INASET, assign the default URL (within 191 characters) to the "Default URL" parameter of ADTM as necessary.

STEP 10: **ADKS** - Assign the Soft key Pattern (which is assigned by the ADSL command) onto IP station. For Soft Phone, be sure to assign "3 (ALL Line/Feature buttons)" to the "LKP" parameter.

<IP Terminal Automatic Assignment Function>

IP station data (Station/Telephone Number, MAC address, LENS) of IP stations can be easily registered by key operation on each IP station, without PCPro commands AISTL, ALGSL, and AMAC manual assignment for every station by maintenance personnel.

By connecting IP station (whose station number and MAC address have not been assigned yet) to LAN, IP station data can be registered/deleted automatically using the specified password and Station/Telephone Number as login ID.

Operating Procedure

To register new IP station data

1. Connect a not-registered IP station to LAN. (Or on any registered IP station in logout state, turn the LCD to password entry indication of Login Mode).
2. Enter the following.
   Login ID: Station Number/Telephone Number to be allocated to the connected IP station (Use a Station/Telephone Number complying with the assigned numbering plan.)
   Password: Password for Registration (assigned by the AEAAL command)
3. The IP station turns login state, and the IP station data is automatically assigned. When MAC address automatic registration function is available, the MAC address is also automatically registered.

To delete the registered IP station data

1. On any IP station in logout state, turn the LCD to password entry indication of Login Mode.
2. Enter the following.
   Login ID: Station Number/Telephone Number to be deleted
   Password: Password for Deletion (assigned by the AEAAL command)
3. The IP station turns logout state, and the specified station data is automatically deleted. (MAC address must be manually deleted by the AMAC command.)

Service Conditions

1. Different number of digit cannot be assigned as Station Number/Telephone Number.
2. A maximum 16 digits of Telephone Number can be assigned in LDM data. NDM data is not available.
3. IP station data cannot be registered to the LENS whose mounting status has not been assigned by system data (ASYD, SYS1, Index 198 to Index 205).
4. MAC address automatic registration function is available when the check box "MAC address is registered" is checked on "Password for registration class instruction" tab of the AEAAL command, depending on the selected Template ID.

5. The result of IP station data registration can be output as history (system message), depending on the setting of the selected Template ID. System message is always output for the execution of IP station data deletion.
6. The password for registration/deletion must be assigned to fixed 10 digits (0-9, *, #).
7. Because DRS reads the password data periodically, there is a case to take a while to make the changed data effective.
8. IP station key data assignment is not covered by this function.
9. On "Station LENS range" tab of AEAAL command, vacant LENS can be searched up to 64 times.
10. On "Physical station number range" tab of AEAAL command, vacant Station Number can be searched up to 100 times per range.
11. MAC address cannot be deleted in the operation of IP station data deletion by this function. Delete the MAC address of deleted IP stations by the AMAC command.
12. When "Physical station number range" tab is used, login ID on IP station is considered as Telephone Number.
13. The IP station turns Login Mode operation after the IP station data is registered by this function.
14. The message "Station Deleted" is displayed on the LCD when succeeded in deleting the IP station data.
15. When multiple Station data sets (the MAC Address is registered checkbox on AEAAL is checked) are assigned from a terminal, only the first assigned station data set includes MAC address. The other station data sets will not have MAC address. (All data except MAC address is assigned, and "Normal" is displayed in the System Message.)
16. The MAC Address is registered checkbox on AEAAL is basically checked when Station data is assigned from each terminal.

Note: When the MAC Address is registered (checkbox on AEAAL is checked) and multiple Station data sets are assigned from a terminal, assign LOGOUT (FKY: 142) to a Soft Key on the IP station. (ADSL, except SKP=0)

Programming

STEP 1: **ANPD/ANPDL** - Assign 1st Digit Code and the total digit number (NND) of Station/Telephone Numbers.

STEP 2: **ASPA/ASPAL** - Assign the Access Code of Station/Telephone Number, using the 1st DC specified by ANPD.

STEP 3: **AEAAL** - Assign the IP Terminal Automatic Data.

<Password for registration class instruction tab>

ID: Template ID [1-30]
A maximum of 30 patterns of template can be registered. Click the Get button to read out the assigned data of entered Template ID.
Password: Password for Registration [10 digits fixed (0-9, *, #)]
TN: Tenant Number [1-63]
RSC: Route Restriction Class [0-15]
SFC: Service Feature Restriction Class [0-15]
MAC Address is registered: Click the check box ON when MAC address registration is required.
Log is displayed: Click the check box ON when system message output is required.

<Password for deletion tab>

Password: Password for Deletion [10 digits fixed (0-9, *, #)]

<Station LENS range tab>

Start/End LENS: Specify the range of LENS (virtual accommodate location of IP station).

<Physical station number range tab>

Start/End STN: Specify the range of Station Number to be allocated to the IP station.

Note: This tab is available only when Telephone Number is used as Login ID on IP station.

14.2 Consideration for Connecting an IP Station to a PSTN (Analog and ISDN)

Voice quality for connecting an IP Station to a PSTN is explained below.
IP Station may be connected to a PSTN through the following methods:

- Connecting to an analog network (using MC&MG-COT / COT circuit card)
- Connecting to an ISDN network (using MG(PRI) / MG(BRI) / PRT circuit card)

When comparing voice quality between connections to an analog network and an ISDN network, the following voice quality deterioration may occur in an analog network:

(a) High possibility for echo to occur at beginning of speech (due to hybrid circuit wraparound)

(b) Audio breakup may occur during double talk (when both parties talk at once) since there is huge loss.

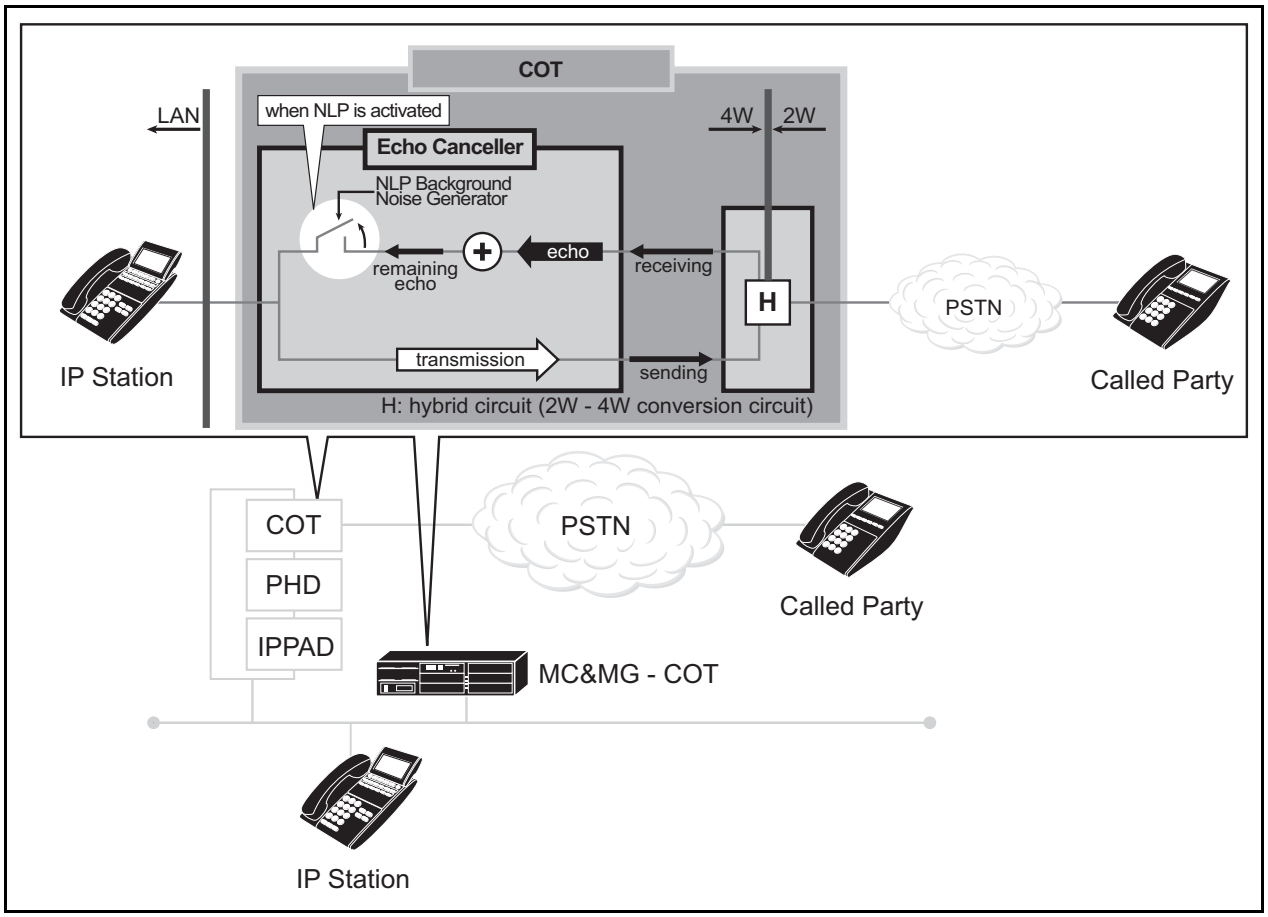
Hence, when voice quality equivalent to a legacy station is desired for an IP station connected to a PSTN, ISDN connection is recommended.

(1) Reasons for voice quality deterioration in an analog network (mechanisms for counter party's audio break-up)

When echo occurs during speech, echo is canceled by Echo Canceller. However, complete cancellation of echo is impossible since non-linear distortion cannot be completely avoided in the echo path.

Therefore to remove the echo that was not canceled by Echo Canceller, remaining echo is made inaudible by cutting the voice in Non Linear Processor (NLP) circuit. NLP operates, as a basic rule, while the caller from an IP station terminal is talking and the voice received from the other terminal is below a specific threshold.

Audio breakup during connection to an analog circuit tends to have large remaining echo due to large wrap-around in the hybrid circuit. Audio breakup occurs when the counter party's voice from an analog circuit is below NLP's threshold, activating the NLP.



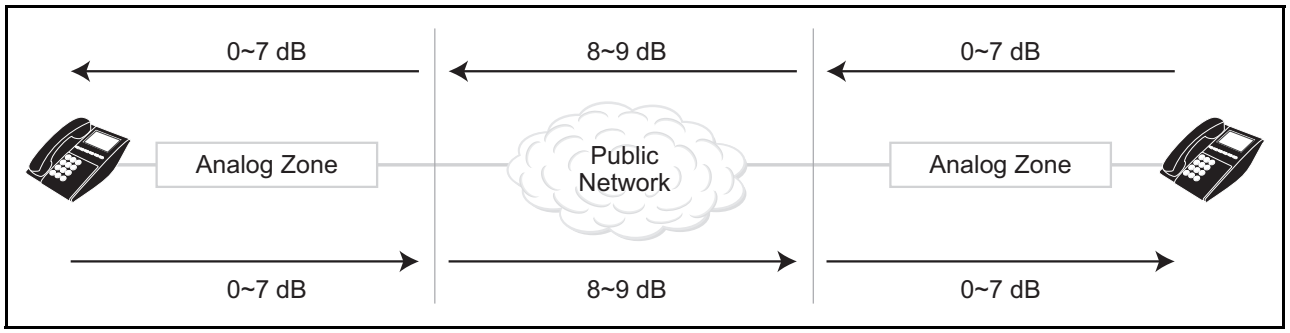
(2) Voice quality for analog network and ISDN network

Voice quality, when connected to a PSTN, is affected by the counter party's level of transmission. This value of loss greatly affects the activation of Echo Canceller NLP. The value of loss is as follows:

- Analog Network: approximately 23 dB (there are instances when it is above this level)
- ISDN Network: approximately 8 dB (there are instances when it is above this level)

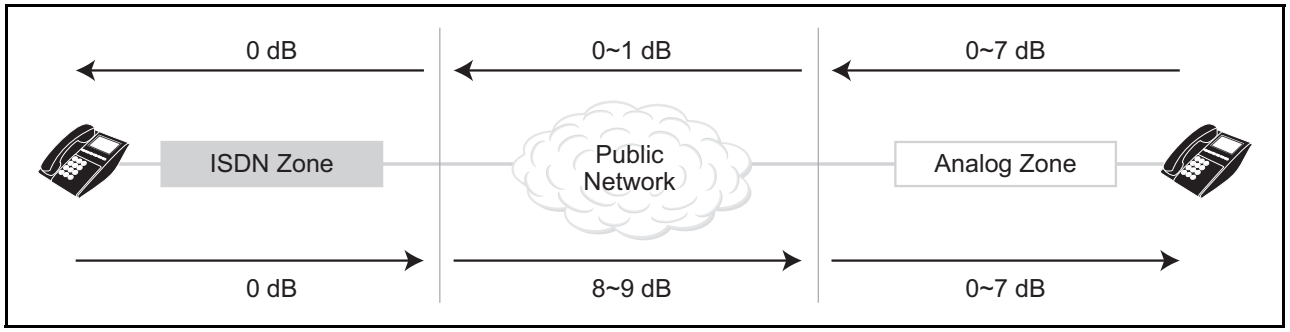
Since ISDN network does not have much loss, rate of counter party's voice recognition is high and it smoothly detects double talk. Therefore far less audio breakup is caused by NLP.

- Analog network to analog network connection



In both directions, there is a maximum loss of 23 dB (in analog zone, loss may be over 7dB).

- ISDN network to analog network connection



In ISDN network to Analog network, there is a maximum loss of 16 dB (in analog zone, loss may be over 7dB). In Analog network to ISDN network, there is a maximum loss of 8dB (in analog zone, loss may be over 7dB).

Note: Loss from this direction greatly affects the activation of Echo Canceler's NLP.

(3) Considerations for data assignment

If the above audio deterioration occurs in an IP station, make the following adjustments.

(a) Adjust the Balancing Network (BNW) of MC&MG-COT/UG50 (COT-TYPE1)/UG50-4LC2COTA (COT) COT circuit card.

How to adjust the Balancing Network (BNW)

For MC&MG-COT	See Peripheral Equipment Description (IP Devices): CODEC_COEF_MG of SET (MG side) command for each MC&MG-COT
For UG50 (COT-TYPE1)/UG50-4LC2COTA (COT)	See Peripheral Equipment Description (UG50): CODEC Filter Data Setup for each COT Card
For COT circuit card	See Circuit Card Description

(b) Use APDIL command to decrease the IP station's transmission (IP station: MC&MG-COT/COT circuit card) level to approximately 6-8 dB.

14.3 Virtual PIR Data

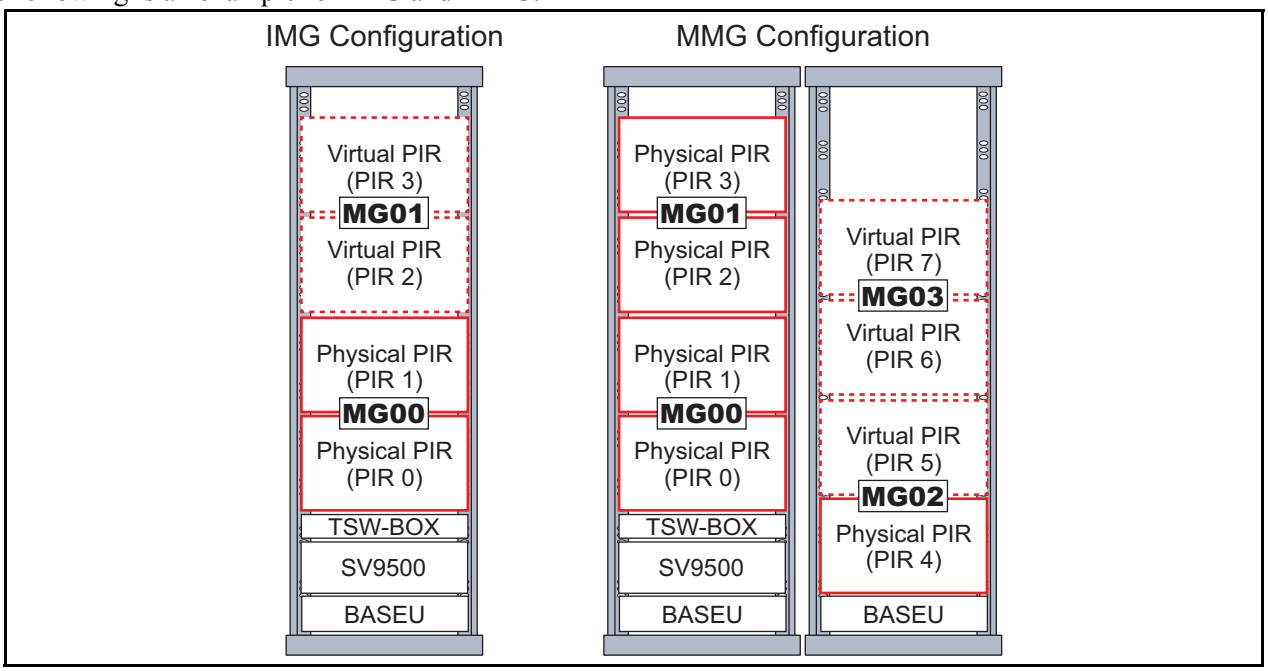
The following data can be assigned in a physically non-existent PIR (called Virtual PIR). When using Virtual PIR, the System can accommodate a maximum number of the lines with a minimum of hardware configuration.

- Virtual line (Desktop terminal Multiple Line)
- IP station data
- Internal PH
- Media Gateway (MG)
- PS station data

Note: System Initialization is required after deleted the Virtual PIR.

Note: For the IMG/MMG system: up to 4,000 lines (for IP stations) can be accommodated. For the UMGi system: up to 16,000 lines (for IP stations) can be accommodated.

The following is an example for IMG and MMG.



Service Conditions

1. The existing Virtual PIR data cannot be deleted while the System is in On-Line mode (The existing virtual PIR data must be deleted during the Off-Line mode). After deleting the Virtual PIR data, the System must be initialized.
2. IP Station data (by AISTL) cannot be assigned to the G24-G31 of Virtual PIR.
3. The Unit Data assignment (by AUNT) is required for the Virtual PIRs as well as the Physical PIRs. Also, system initialization is required if any change is made on the office data by the AUNT command.
4. When the Virtual PIR data is assigned to the physically non-existent LP via System data (ASYD, SYS1, Index 198 to Index 205), assign the Virtual PIR data in sequence (IMG 1 → IMG 15). **Note 1**
5. When the data programming procedure described in the Data Programming is not maintained, the System Message (4-D: TSW Ready Failure) may be output.

6. TSW data setting is not required for Virtual PIRs.

Note 1: ASYD, SYS1, Index 200 (IMG (LP) 4 and IMG (LP) 5) to Index 205 (IMG (LP) 14 and IMG (LP) 15) are only for UMGi system.

Programming

STEP 1: **ASYD** - Assign the number of Module Groups.

SYS1, Index 0 (01 HEX ~ 20 HEX (MG00 ~ 31)).

Note: Required to be set for both Physical PIR and Virtual PIR.

STEP 2: **ASYD** - Assign the Mounting Status for IMG (LP) 0 to IMG (LP) 15 (SYS1, Index 198 to Index 205).

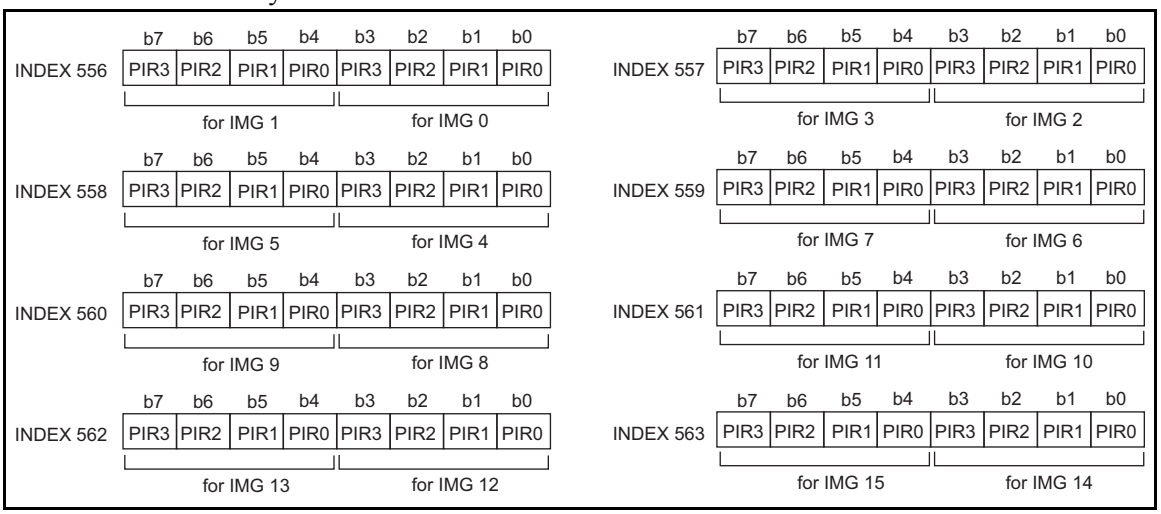
SYS1, Index 198. Mounting Status for IMG (LP) 0 and IMG (LP) 1.
SYS1, Index 199. Mounting Status for IMG (LP) 2 and IMG (LP) 3.
SYS1, Index 200. Mounting Status for IMG (LP) 4 and IMG (LP) 5.
SYS1, Index 201. Mounting Status for IMG (LP) 6 and IMG (LP) 7.
SYS1, Index 202. Mounting Status for IMG (LP) 8 and IMG (LP) 9.
SYS1, Index 203. Mounting Status for IMG (LP) 10 and IMG (LP) 11.
SYS1, Index 204. Mounting Status for IMG (LP) 12 and IMG (LP) 13.
SYS1, Index 205. Mounting Status for IMG (LP) 14 and IMG (LP) 15.

Note: Required to be set for both Physical PIR and Virtual PIR.

Note: ASYD, SYS1, Index 200 to Index 205 are only for UMGi system.

STEP 3: **ASYDL** - Assign the Virtual PIR data (SYS1, Index 556 to Index 563).

Each Bit: 0/1 = Physical PIR/Virtual PIR



Note: Only in the case where a physical PIR is installed, the related bit in Index 556 to Index 563 must be 0. In all other cases, the bit must be 1. Be sure to match the setting of ASYDL, SYS1, Index 556 to Index 563 and the actual installed configuration.

Note: ASYDL, SYS1, Index 558 to Index 563 are only for UMGi system.

STEP 4: **ASYDL** - Assign the UNIT data (SYS1, Index 1026 to Index 1033).

SYS1, Index 1026. UNIT use on MG00-MG03.
SYS1, Index 1027. UNIT use on MG04-MG07.
SYS1, Index 1028. UNIT use on MG08-MG11.
SYS1, Index 1029. UNIT use on MG12-MG15.
SYS1, Index 1030. UNIT use on MG16-MG19.
SYS1, Index 1031. UNIT use on MG20-MG23.
SYS1, Index 1032. UNIT use on MG24-MG27.
SYS1, Index 1033. UNIT use on MG28-MG31.

Note: Required to be set for both Physical PIR and Virtual PIR.

Note: ASYDL, SYS1, Index 1028 to Index 1033 are only for UMGi system.

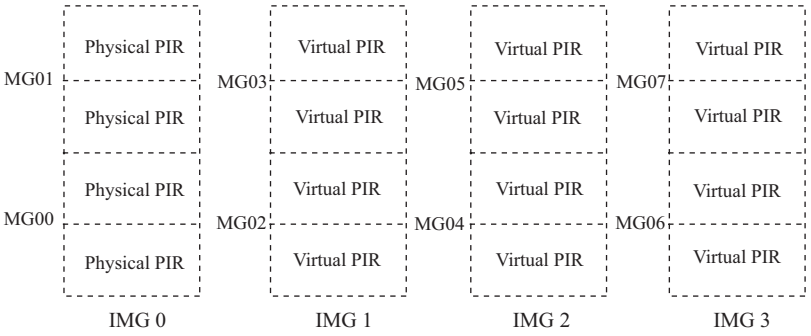
STEP 5: **AUNT** - Assign the Unit data.

MG = Module Group Number
TYPE: 1 (Select 1 for Unit data assignment)
UNIT0: 1 (Unit 0 is used)
UNIT1: 1 (Unit 1 is used)
UNIT2: 1 (Unit 2 is used)
UNIT3: 1 (Unit 3 is used)

Note: Required to be set for both Physical PIR and Virtual PIR.

Note: System initialization is required for the change to take effect.

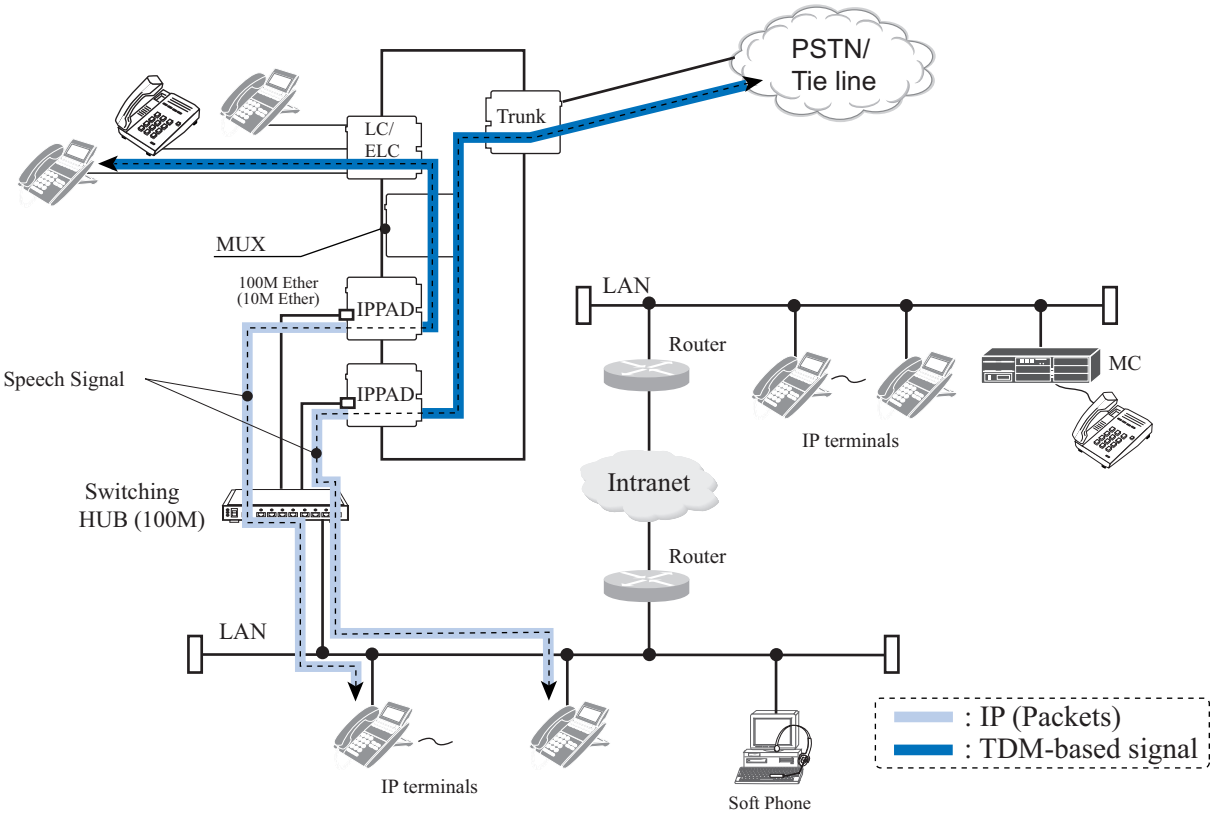
The following figure shows an example data assignment for the System including the Virtual PIR. At this time, the Unit data must be assigned for both the Physical PIR and the Virtual PIR.



1. ASYDL, System Data 1, INDEX 556 = F0 Hex
   INDEX 557 = FF Hex
   INDEX 1026 = FF Hex
   INDEX 1027 = FF Hex
2. ASYD, System Data 1, INDEX 0 = 08 Hex
   INDEX 198 = 11 Hex
   INDEX 199 = 11 Hex
3. AUNT
   MG = 00-07
   TYPE = 1 (Line/Trunk)
   UNIT 0 = 1 (UNIT 0 is used)
   UNIT 1 = 1 (UNIT 1 is used)
   UNIT 2 = 1 (UNIT 2 is used)
   UNIT 3 = 1 (UNIT 3 is used)

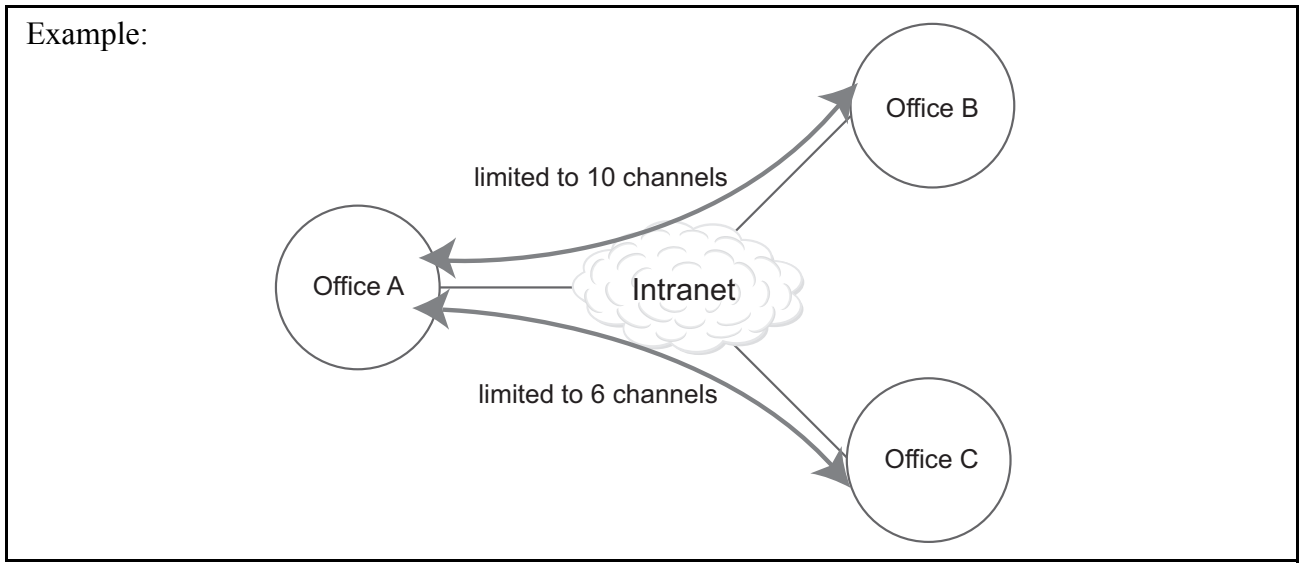
14.4 IPPAD Data

This section explains how to program the connection trunk data for IPPAD (IP Packet assembly/disassembly) circuit card. This circuit card is necessary when a call is to be established between the TDM-based Conventional Telephones/Network and IP terminals. At this time, IPPAD provides a gateway function: Voice Signal (IP Packets) sent from the LAN is disassembled as the TDM-based Signal, and the opposite directions assembled into the IP Packets. A maximum of 32 calls can be dealt with per IPPAD Card. A connection block diagram is shown below:



IPPAD: IP Packet Assembly/Disassembly (CH-IPDA for 7U-PIR, PA-32IPDA/PA-32IPDB for 8U-PIR)

It is available to limit the number of channels to be used for IPPAD per network.



The number of channels that are processed depends on codec and payload cycle used by this card. Therefore, adjustments will be needed, such as placing unnecessary channels into the Make Busy status.
See "VoIP Encryption [V-27]" for the number of available channels on the encryption status of this card.

Service Conditions

<Important Notices>

In case of performing voice signal compression, all the IPPAD circuit cards accommodated in the System should correspond to voice signal compression. In case of using voice signal compression with FCCS over IP or CCIS over IP, all the IPPAD circuit cards accommodated in the nodes inside the network should correspond to voice signal compression.

Programming

The following are the step-by-step procedures for IPPAD data programming. As shown, each port for an IPPAD Card is assigned as the Connection Trunk using the ACRD/ACTK commands.

STEP 1: **ASYDL** - SYS1, Index 810, Bit 4: Giving Priority of IPPAD in the Same Network feature **Note 1**, **Note 2**
0: Required (Default)
1: Not Required
When ASYDL, SYS1, Index 810, Bit 4=0 (Giving Priority of IPPAD in the Same Network feature is required):
- IPPAD in the same network has top priority.
- If IPPAD in the same network is all busy or cannot be used, IPPAD in another network is used.

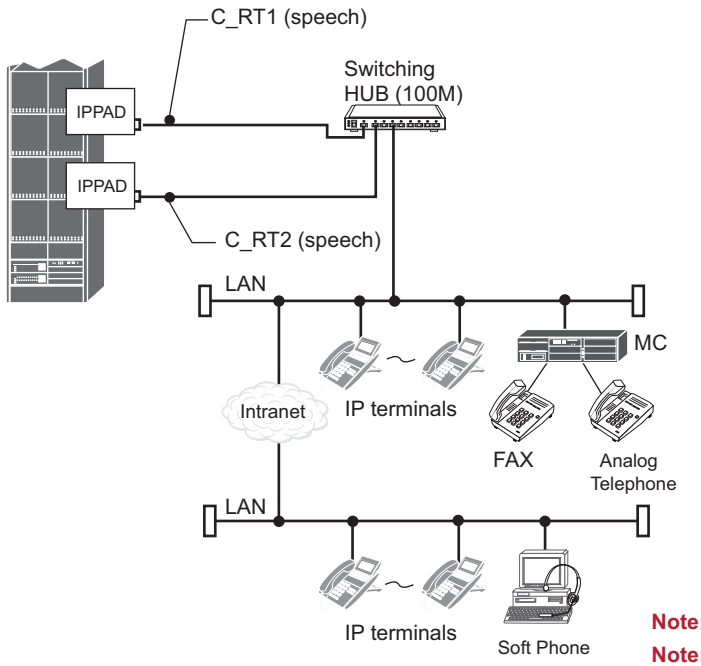
When ASYDL, SYS1, Index 810, Bit 4=1 (Giving Priority of IPPAD in the Same Network Feature is not required):
- According to the order of the assignment of Connection Route and Connection Trunk
- Using any one IPPAD in the Node, without the restriction between the networks

Note 1: This feature is also available for VLAN.

Note 2: This feature cannot be activated via the PHC.

STEP 2: **ACRD** -Assign Route Class data of Speech Routes. A sample data assignment is shown below.

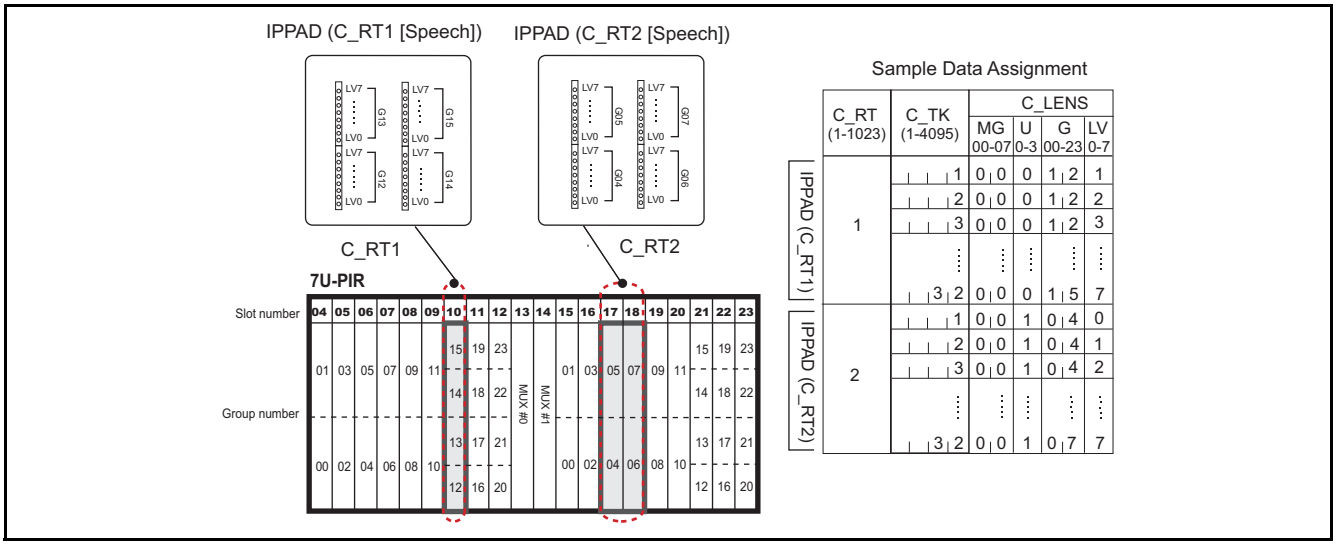
ACRD: Speech Route (LSG = 12, IPPAD = 1)



Type		Speech	Speech
CDN	C_RT	1	2
1	TF	3	3
2	TCL	4	4
3	RLP	2	2
4	SMDR	0	0
5	LSG	12	12
6	PAD		
7	TRKS	0(1)	0(1)
8	TC/EC		
9	FINT	0	0
10	FPEG	0	0
11	TC	0	0
12	MTC	0	0
13	STSEQ	0	0
14	FGH	0	0
15	MMN	0	0
16	LKIND	0	0
17	IPLYR	0	0
18	IPTRK	0	0
19	IPPAD	1	1
20	LMG	0	0

Note: This data setting is an example.
Note: "CDN20 LMG" is not used in the system.

STEP 3: **ACTK** -Assign Connection Trunk data (Connection Route, Connection Trunk and Line Equipment Numbers) of the IPPAD circuit card. A sample data assignment is shown below.

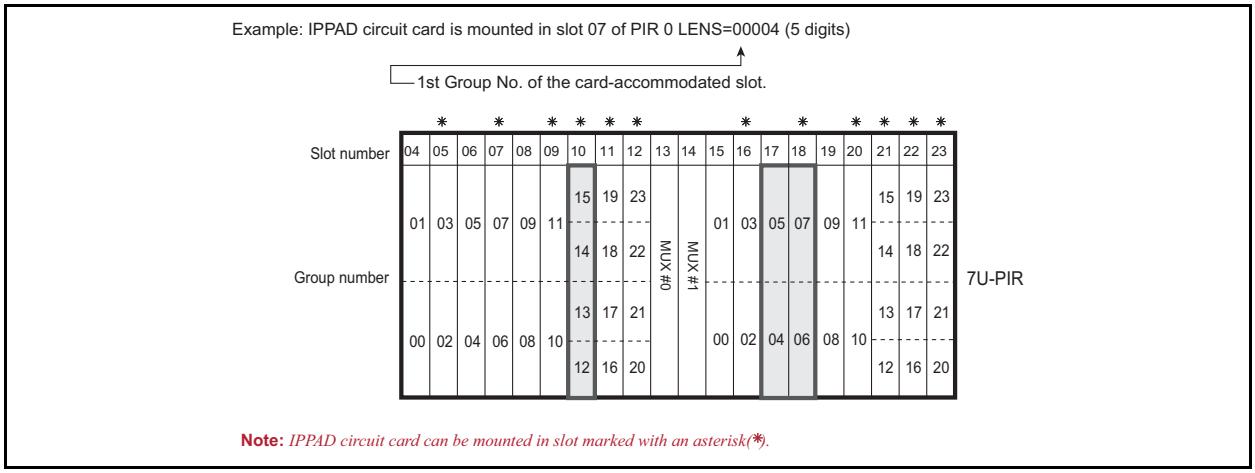


Sample Data Assignment

	C_RT (1-1023)	C_TK (1-4095)	C_LENS MG 00-07	U 0-3	G 00-23	LV 0-7
IPPAD (C_RT1)	1	1	0	0	1	2	1
		2	0	0	1	2	2
		3	0	0	1	2	3
		⋮	⋮	⋮	⋮	⋮
		32	0	0	1	5	7
IPPAD (C_RT2)	2	1	0	0	1	4	0
		2	0	0	1	4	1
		3	0	0	1	4	2
		⋮	⋮	⋮	⋮	⋮
		32	0	0	1	7	7

STEP 4: **AFPC** - For each IPPAD Connection Route assigned by the ACRD command, assign 0 (for IPPAD circuit card) to the FPC parameter.
FPC = 0 (for IPPAD circuit card)
C_RT: IPPAD Connection Route assigned by the ACRD.
FCCH: Assign "0".
P_ROUTE: Assign "0".
FCHN/FPCN: Assign "0".

STEP 5: **ALIDL** - Assign the detailed data of the IPPAD card such as Line Equipment Numbers, IP address, jitter butter, payload type/size.
LENS: Assign two-digit MG No. + one-digit Unit No. + 1st Group Number (two digits).



PH TYPE: Select IPPAD.
PH IP/IPPAD IP: Enter IP address of IPPAD card.

MASK BIT: Enter Mask Bit of IP address.
PORT: Enter the port number for control signal. (Recommendation: 61010)
REGMAX: Number of ports registered (Not used) **Note 3**
NTP_IP: IP Address of NTP Server (Not used) **Note 3**
FAX_DET: Fax Detection Timer
- •Conform to Switch Setting
- •Voice/Fax Automatic Detection
- •Fixed to Voice Mode
- •Fax Detectable Time 1 minute (Recommendation)
- •Fax Detectable Time 2 minute
- •Fax Detectable Time 3 minute
- •Fax Detectable Time 4 minute
- •Fax Detectable Time 5 minute

FAX_ECM: Specify the Fax ECM mode. (Not used) **Note 3**
FAX_TYP: Fax Mode Control Precedence (Not used) **Note 3**, **Note 4**
FAX_RATE: Fax Transmission Rate (bps)
- •14400 bps (Recommendation)
- •2400 bps
- •4800 bps
- •7200 bps
- •9600 bps
- •12000 bps

RTCP_PTN: RTCP (SR) Pattern (Not used) **Note 3**
RTCP_CYC: RTCP (SR) Cycle (fixed as 0) (5 seconds cycle)
DEF_GW IP: Enter the IP address of the default gateway. A maximum of eight IP addresses can be assigned. **Note 5**

Note 3: Since this parameter is not used, the currently displayed setting should not be changed.

Note 4: The data related to FAX should be assigned using the ARTI command with “FXPT” parameter.

Note 5: The maximum number of Default Gateway IP addresses varies depending on the firmware equipped on the IPPAD.

STEP 6: **MBCT** - Unbusy the Connection Trunk Data assigned in the previous step, by using this command.

To limit the number of channels to be used for IPPAD, use ALOCL command.

STEP 7: **ALOCL** - Assign Location ID to IP Address, PC or FPC, and then specify the connection route to be used for IPPAD.

Note: When not assigning connection route in ALOCL command, the connection routes assigned in AFPC command is used for IPPAD in the order of assignment.

- When Location ID is assigned to Network Address:

  IP ADDRESS: Enter the IP address.
  MASK BIT: Assign the mask bit in the range 1-31.
  LOC-ID: Assign Location ID to the IP address in the range of 1-4095.
  IPPAD CONNECTION ROUTE ASSIGNMENT: Place a checkmark in the checkbox.
  C_RT: Specify the connection route used for IPPAD.

- When Location ID is assigned to PC (Point Code):

Note: The IPPAD connection route assigned in the C_RT parameter is used for all CCIS over IP calls originated to the appropriate Point Code.

  PC: Enter the Point Code.
  LOC-ID: Assign Location ID to the Point Code in the range of 1-4095.
  IPPAD CONNECTION ROUTE ASSIGNMENT: Place a checkmark in the checkbox.
  C_RT: Specify the connection route used for IPPAD.

- When Location ID is assigned to FPC (Fusion Point Code):

Note: The IPPAD connection route assigned in the C_RT parameter is used for all FCCS over IP calls originated to the appropriated Fusion Point Code.

  FPC: Enter the Fusion Point Code.
  LOC-ID: Assign Location ID to the Fusion Point Code in the range of 1-4095.
  IPPAD CONNECTION ROUTE ASSIGNMENT: Place a checkmark in the checkbox.
  C_RT: Specify the connection route used for IPPAD.

To delete IPPAD data

Be sure to place the MB switch to ON position (UP) on the corresponding IPPAD Card, then follow the programming steps above in reverse order.

14.5 IP Service PAD/Echo Canceler Data

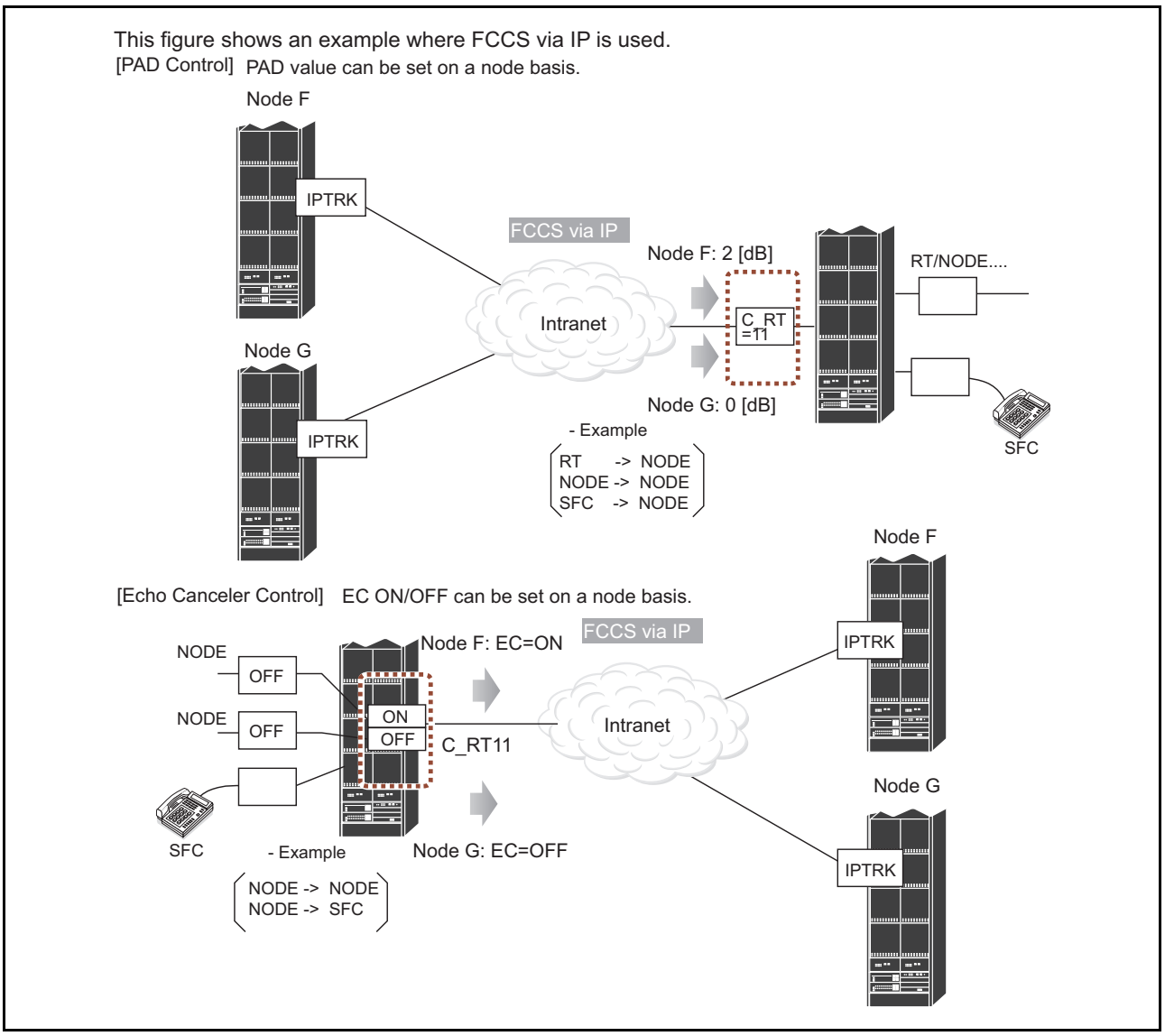
This section explains how to set IP Service PAD/EC data. The data can be set on a Node basis (when using IPTRK: refer to 11.1 PAD/ EC Control on a Node basis) or a Location ID basis (when using PH-L/G, IPPAD: refer to 11.2 PAD/EC Control on a Location ID).

Note: The PAD/EC data can also be set on a route basis as usual.

14.5.1 PAD/EC Control on a Node Basis

< PAD Control and Echo Canceler Control>

PAD/EC Control on a Node Basis



Note: FCCS Networking via IP (by IPTRK) is available only for a migrated system from the previous system.

The following is a list of the available PAD patterns:

IC Terminal Type	OG Terminal Type	IC PAD	OG PAD	DATA
RT (Trunk)	IPTRK	×	×	RT-NODE
SFC (Station/Dterm)		PAD OFF	×	NODE-SFC
ISDN Terminal		PAD OFF	×	NODE-SFC
C_RT (C_TK)		×	×	C_RT-NODE
IPTRK	IPTRK	×	×	NODE-NODE
	RT (Trunk)	×	×	NODE-RT
	SFC (Station/Dterm)	×	PAD OFF	NODE-SFC
	ISDN Terminal	×	PAD OFF	NODE-SFC
	C_RT (C_TK)	×	×	NODE-C_RT

Conditions

<PAD Control>

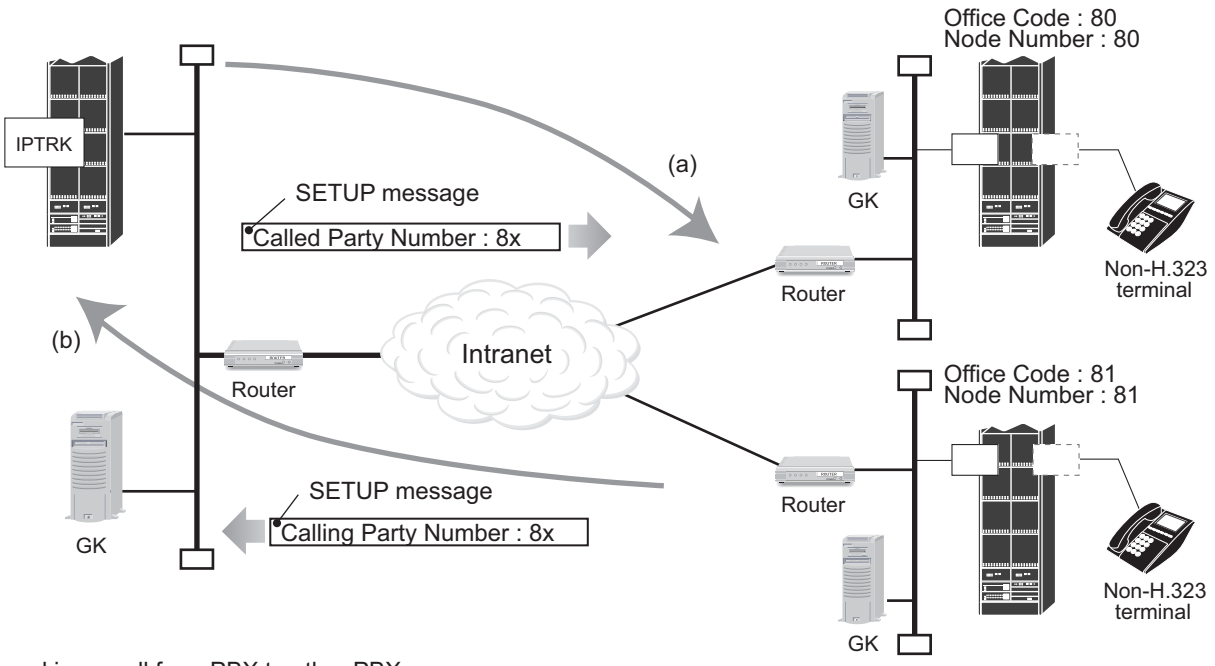
1. PAD data assigned by the AIPDL/AIPDN command is actually specified by the circuit card. Refer to the related explanation of the circuit card in this manual.
2. When the APADN data for FCCS Network has already been assigned, this feature is invalid.
3. When AFPD (used for "FCCS via IP") command data has already been assigned, this feature has precedence over the data assigned by the AFPD command.
4. This feature is available for External Trunk, C_TK, IPTRK only as APAD/APADN/AFPD command. (is not available for Stations.)
5. For this feature, use the Node Number which has been assigned in advance by using the ANTI.
6. When using an IPTRK, the PAD value can be set at the receive side. (Coding and decoding are performed after the PAD insertion. Therefore, the speech quality may not be guaranteed.)
7. When using the AIPDL/AIPDN by specifying SFC-NODE, use IC Terminal = NODE (IPTRK) and OG Terminal = SFC (Station). If IC Terminal = "SFC (Station)" and OG Terminal = "NODE (IPTRK)" are assigned, the data setting is invalid.
8. This feature is not available for a data call.

<Echo Canceler Control>

1. For this feature, use a Node Number that has been assigned in advance by using the ANTI.
2. When this data is not set, echo canceler data set by the ARTD/ACRD is used.
3. This feature is available for IPTRK-Station, or IPTRK-External Trunk. IPTRK-Internal Trunk such as ATI is not included. In this case, the echo canceler data of the Route Class data is used.
4. EC control is performed to an IPTRK or a Digital Trunk, it is not performed to an Analog Trunk even if any data has been set.

<When Connecting to the Telephony Server <Gateway>>

Considerations for Assigning Node Data (When Connecting to the Telephony Server <Gateway>)



(a) For making a call from PBX to other PBX:
Office code of the called party (8x) must be included to Called Party Number of the SETUP message.

(b) For making a call from other PBX terminal to PBX:
Office code of the calling party (8x) must be included to Calling Party Number of the SETUP message.

Note: FCCS Networking via IP (by IPTRK) is available only for a migrated system from the previous system.

Data Assignment in the Telephony Server:

- Assign the Access Code (8x) for other Telephony Server using the ASPAL/ASPAN and AFRSL/AFRSN.
- Do not delete the Access Code for Called Party Number using the AOPRL/AOPRN.
  (Or add the Office Code of the other Telephony Server using the AADC, after deleting the Access Code by the AOPRL/AOPRN.)

Data Assignment in "other" Telephony Server:

Perform one of the following for including the Self Office Code to the Calling Party Number:

- Assign the Self Office Code using the ARNP.

  or

- Assign Calling Number addition/deletion data using the ACNP and ACND.

By assigning the above data, PAD/EC Control data set by the ANTI becomes effective.

Programming

<Echo Canceler Control>

STEP 1: FCCS/CCIS via IPTRK Data Assignment

Assign Office Data for FCCS via IPTRK/CCIS via IPTRK. Refer to Programming of each service (FCCS NETWORKING VIA IP [F-36]/CCIS NETWORKING VIA IP [C-163]) in Chapter 3.

Note: FCCS Networking via IP (by IPTRK) is available only for a migrated system from the previous system.

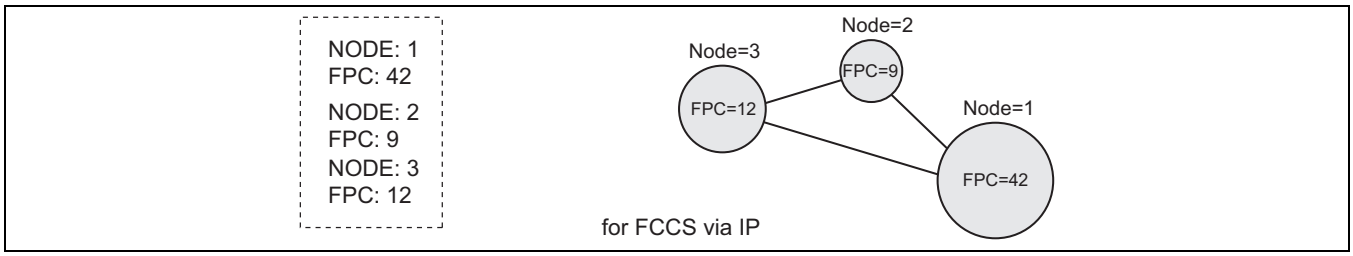
STEP 2: **ANTI** (Assignment of Node Number)

Assign a unique Node Number to each Node, using the following data depending on the system type.

FCCS over IP: FPC Number

CCIS over IP: PC

Assignment of Node Number (example - FCCS via IP)

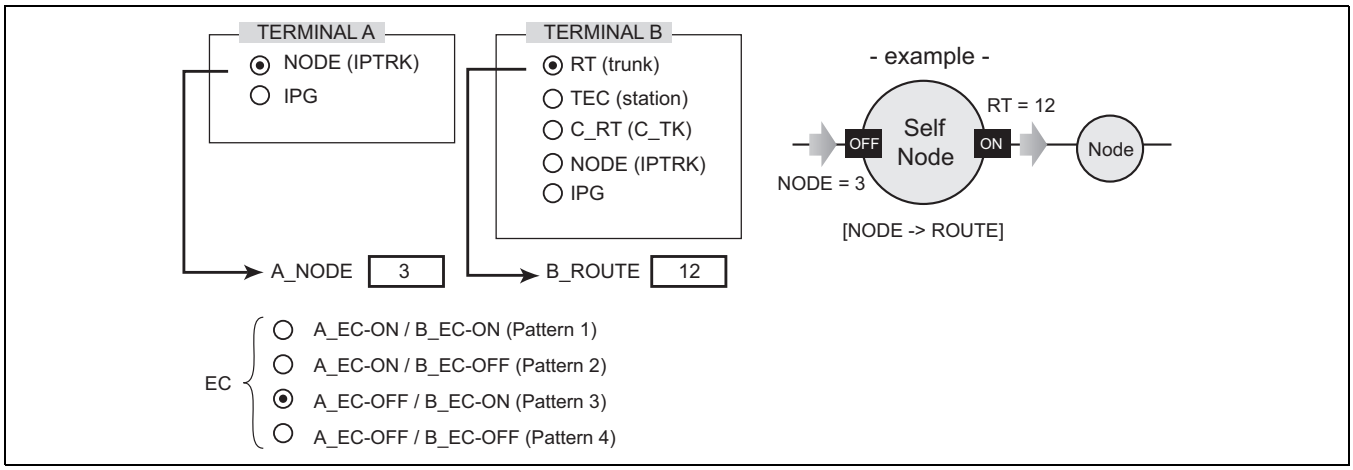


STEP 3: **AIECL/AIECN** (Assignment of IP Service Echo Canceler Data for LDM/NDM)

[AIECL]

Assign EC value data on a specified value basis. The available combinations are shown below.

Assignment of EC value by AIECL (example)



Note 1: According to the combination of Terminal A and Terminal B, required parameters vary as shown below.

TERMINAL B \ TERMINAL A	NODE	IPG
RT	I	Not used
TEC	II	Not used
C_RT	III	Not used
NODE	III	Not used
IPG	Not used	Not used

I: In case of External Trunk the conditions are the same as III. For Internal Trunk, either Pattern 2 or 4 can be selected.

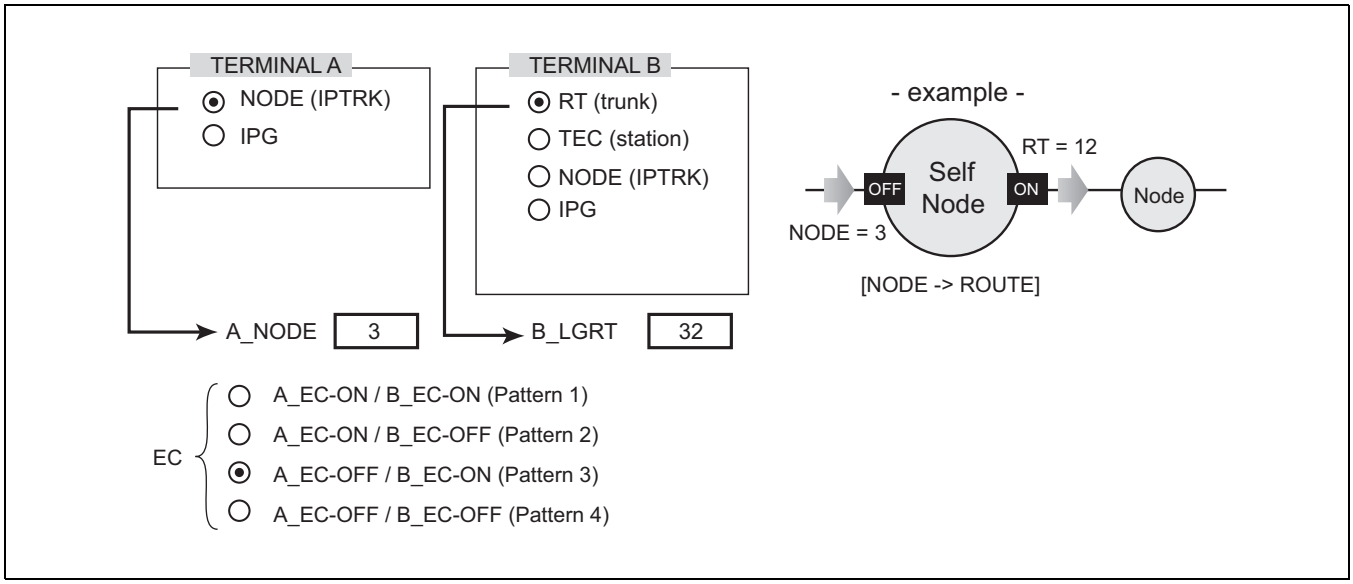
II: Pattern 2 or 4 can be selected for terminals.

III: Any pattern can be assigned.

[AIECN]

Assign EC value data on a specified value basis. The available combinations are shown below.

Assignment of EC value by AIECN (example)



Note 1: According to the combination of Terminal A and Terminal B, required parameters vary as shown below.

TERMINAL B \ TERMINAL A	NODE	IPG
LGRT	I	Not used
TEC	II	Not used
NODE	I	Not used
IPG	Not used	Not used

I: Any pattern can be selected.

II: Pattern 2 or 4 can be selected for terminals.

<PAD Control>

STEP 1: FCCS/CCIS via IPTRK Data Assignment

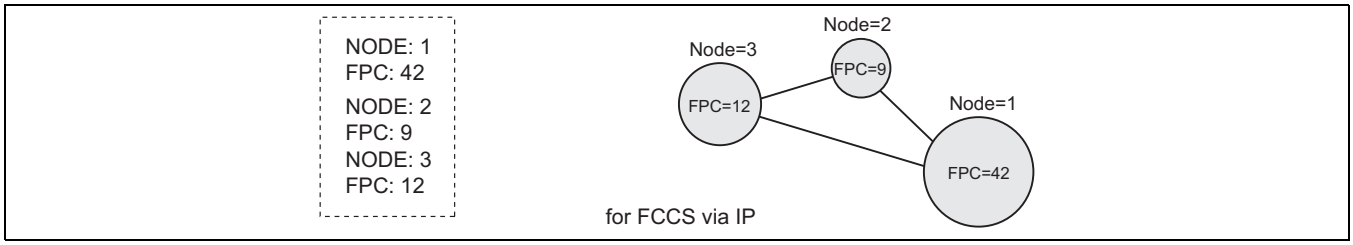
Assign Office Data for FCCS via IPTRK/CCIS via IPTRK. Refer to the Programming of each service (FCCS NETWORKING VIA IP [F-36]/CCIS NETWORKING VIA IP [C-163]) in Chapter 3.

Note: FCCS Networking via IP (by IPTRK) is available only for a migrated system from the previous system.

STEP 2: **ANTI** (Assignment of Node Number)

Assign a unique "Node Number" to each Node, using the following data depending on the system type.
FCCS over IP: FPC (Fusion Point Code) [1-253]
CCIS over IP: PC (Point Code) [1-16383)

Assignment of Node Number (example - FCCS via IP)

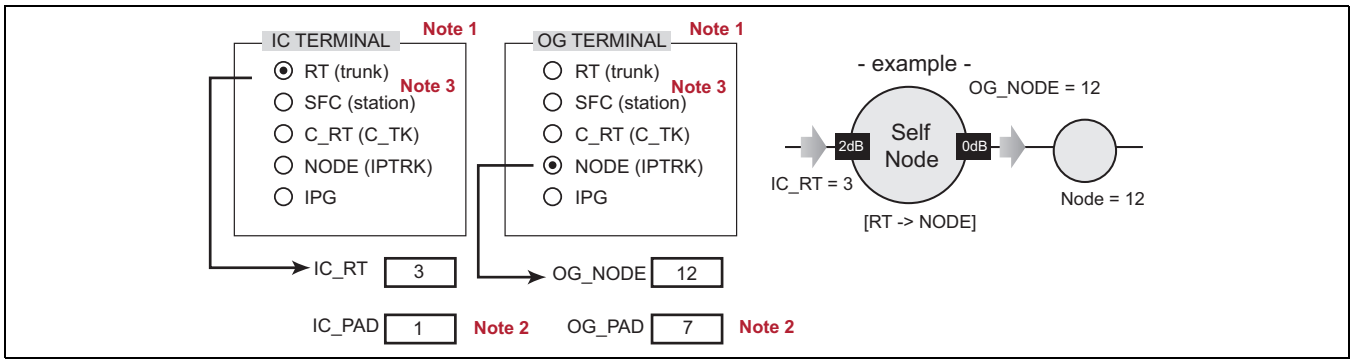


STEP 3: **AIPDL/AIPDN** (Assignment of IP Service PAD Data for LDM/NDM)

[AIPDL]

Assign PAD value data on a specified value basis. The available combinations are shown below.

Assignment of PAD value by AIPDL (example)



Note 1: Available combinations are as follows.

		OG TERMINAL				
		RT	SFC	C_RT	NODE	IPG
IC TERMINAL	RT	-	-	-	×	Not used
	SFC	-	-	-	-	Not used
	C_RT	-	-	-	×	Not used
	NODE	×	×	×	×	Not used
	IPG	Not used	Not used	Not used	Not used	Not used

Note 2: Actual PAD value is determined by the combination of this PAD data and the circuit card switch setting.

Note 3: PAD value for SFC (Station), assign 1 (fixed).

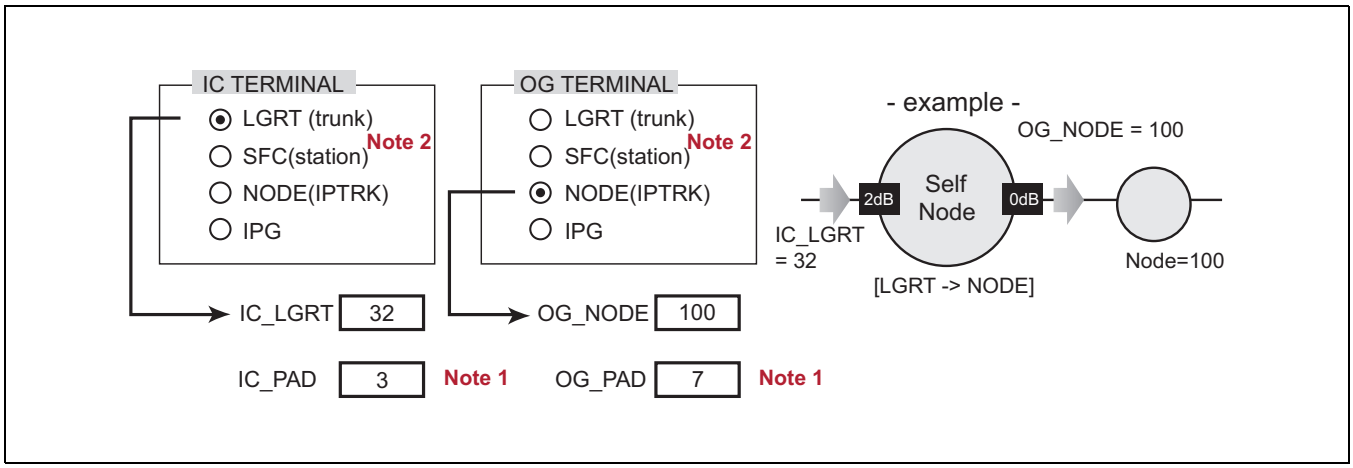
[AIPDN]

Assign PAD value data on a specified value basis. The available combinations are shown below.

AIPDN

IC TERMINAL \ OG TERMINAL	LGRT	SFC	NODE	IPG
LGRT	-	-	×	Not used
SFC	-	-	-	Not used
NODE	×	×	×	Not used
IPG	Not used	Not used	Not used	Not used

Assignment of PAD value by AIPDN (example)



Note 1: Actual PAD value is determined by the combination of this PAD data and the circuit card switch setting.

Note 2: PAD value for SFC (Station), assign 1 (fixed).

14.5.2 PAD/EC Control on a Location ID Basis

<PAD Control>

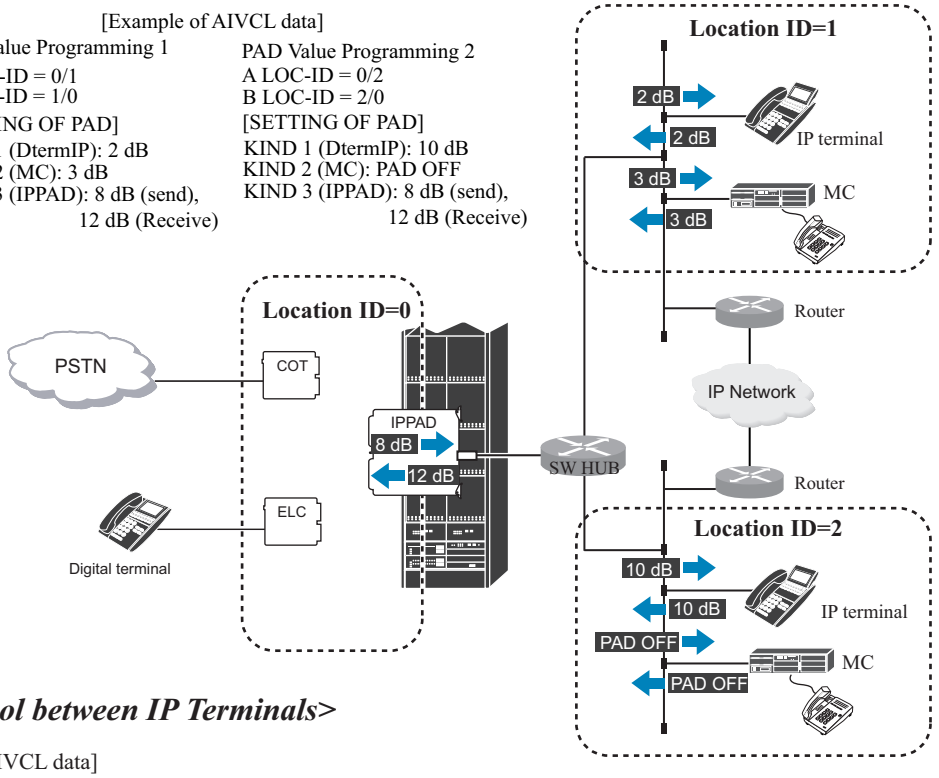
PAD Control on a Location ID Basis

<PAD Control between IPPAD and IP Terminal>

[Example of AIVCL data]

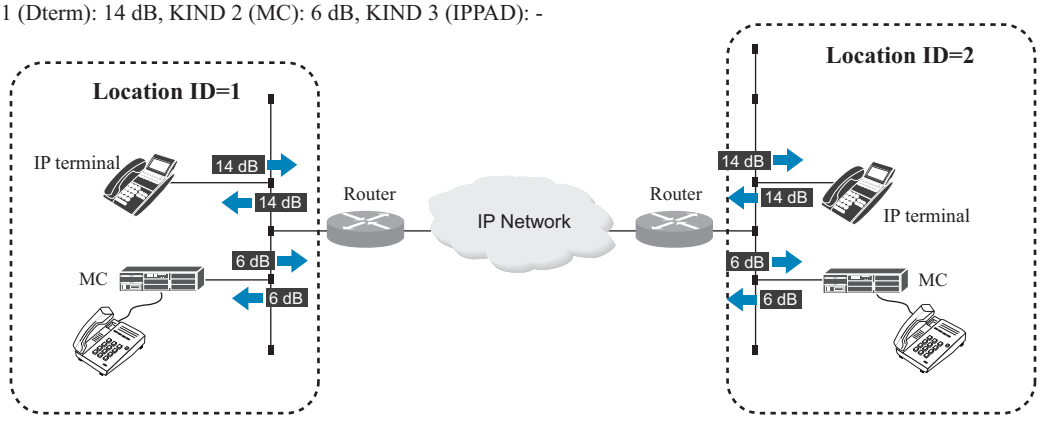
PAD Value Programming 1
A LOC-ID = 0/1
B LOC-ID = 1/0
[SETTING OF PAD]
KIND 1 (DtermIP): 2 dB
KIND 2 (MC): 3 dB
KIND 3 (IPPAD): 8 dB (send), 12 dB (Receive)

PAD Value Programming 2
A LOC-ID = 0/2
B LOC-ID = 2/0
[SETTING OF PAD]
KIND 1 (DtermIP): 10 dB
KIND 2 (MC): PAD OFF
KIND 3 (IPPAD): 8 dB (send), 12 dB (Receive)



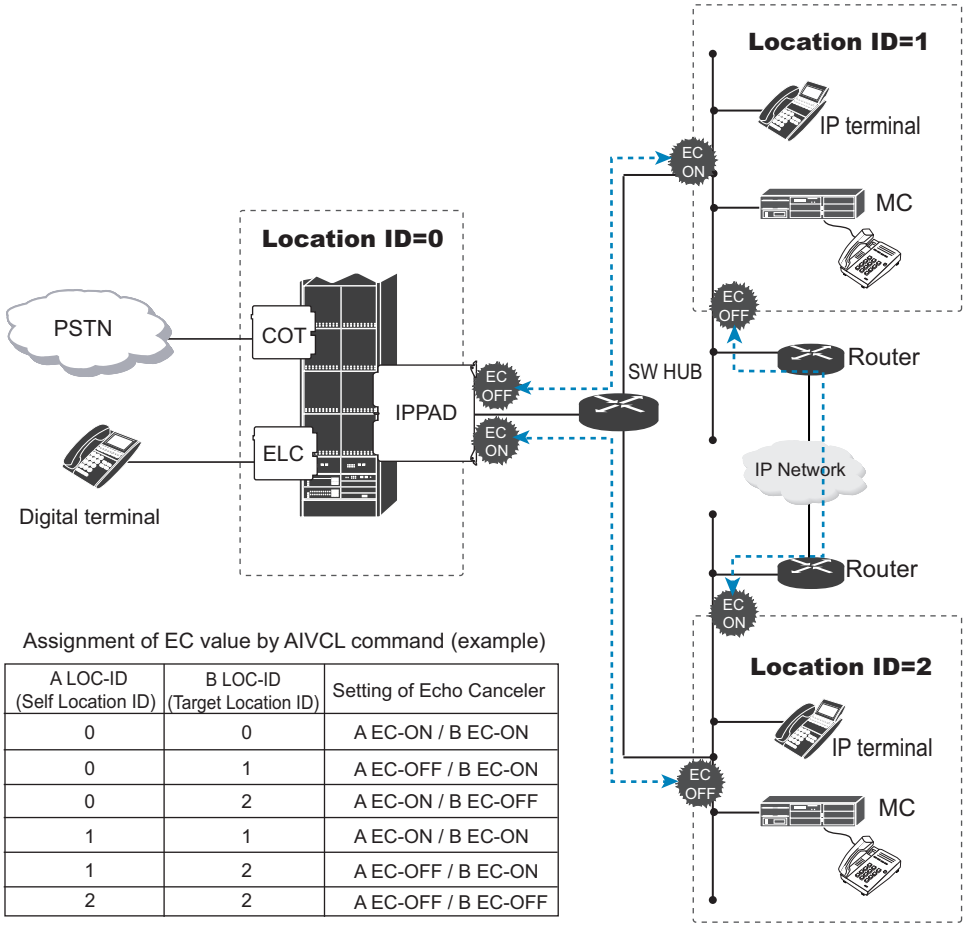
<PAD Control between IP Terminals>

[Example of AIVCL data]
PAD Value Setting 3:
A LOC-ID = 1/2, B LOC-ID = 2/1
SETTING OF PAD
KIND 1 (Dterm): 14 dB, KIND 2 (MC): 6 dB, KIND 3 (IPPAD): -



<Echo Canceler Control>

EC Control on a Location ID Basis



Assignment of EC value by AIVCL command (example)

A LOC-ID (Self Location ID)	B LOC-ID (Target Location ID)	Setting of Echo Canceler
0	0	A EC-ON / B EC-ON
0	1	A EC-OFF / B EC-ON
0	2	A EC-ON / B EC-OFF
1	1	A EC-ON / B EC-ON
1	2	A EC-OFF / B EC-ON
2	2	A EC-OFF / B EC-OFF

Programming

<PAD Control>

STEP 1: **ALOCL** - Assign the Location ID data.

TYPE:	Select “Network Address”.
IP ADDRESS:	Assign the IP Address. For the System (i.e. IPPAD), assign “0.0.0.0.”
MASK BIT:	Assign the Mask Bit in the range 1-31 except when IP address 0.0.0.0 is determined. When IP ADDRESS = 0.0.0.0, MASK BIT = 32 is automatically assigned.
LOC-ID:	Assign any number in the range 0-4095, associated with the network address assigned in the IP ADDRESS parameter. When 0.0.0.0 is assigned in the IP ADDRESS parameter, LOC-ID is automatically set to “0” (the value cannot be changed).

Note: LOC-ID=0 is determined by the Location ID of the System (IPPAD).

STEP 2: **AIVCL** - Assign the IP service voice control data.

A LOC-ID: Location ID of the Self Location. (0-4095)

B LOC-ID: Location ID of the Target Location. (0-4095)

Note: LOC-ID=0 is used for the Location ID of the System (IPPAD).

Note: Only the Location ID corresponding to the network address can be assigned on the AIVCL/AIVCN. The Location IDs corresponding to FPC or STN cannot be assigned.

[Setting of PAD]

Select the Setting of PAD check box to assign the following PAD data.

KIND 1: PAD value for the IP terminal (0-15)

0=PAD Control is not used. 5=10dB

1=2dB 6=12dB

2=4dB 7=14dB

3=6dB 8=16dB

4=8dB 9-15=PAD OFF (0dB)

KIND 2: PAD value for the Analog/Digital MC (0-3)

0=PAD Control is not used.

1=PAD OFF (0dB)

2=3dB

3=6dB

KIND3: PAD value for IPPAD card (0-15)

<Echo Canceler Control>

STEP 1: **ALOCL** - Assign the Location ID data.

TYPE:	Select “Network Address”.
IP ADDRESS:	Assign the IP Address. For the System (IPPAD), assign “0.0.0.0.”
MASK BIT:	Assign the MASK BIT in the range 1-31 except when IP ADDRESS 0.0.0.0 is determined. When IP ADDRESS = 0.0.0.0, MASK BIT = 32 is automatically assigned.
LOC-ID:	Assign any number in the range 0-4095, associated with the network address assigned in the IP ADDRESS parameter. When 0.0.0.0 is assigned in the IP ADDRESS parameter, LOC-ID is automatically set to “0” (the value cannot be changed).

Note: LOC-ID=0 is set to the location ID of the System (IPPAD).

STEP 2: **AIVCL** - Assign the IP service voice control data.

A LOC-ID: Location ID of the Self Location. (0-4095)

B LOC-ID: Location ID of the Target Location. (0-4095)

Note: LOC-ID=0 is used for the Location ID of the System (IPPAD).

Note: Only the Location ID corresponding to the network address can be assigned on the AIVCL/AIVCN. The Location IDs corresponding to FPC or STN cannot be assigned.

[Setting of Echo Canceller]

Select the “Setting of Echo Canceller” check box to assign the following Echo Canceler data.

Select the Echo Canceler value by clicking each radio button. (**Note 1**)

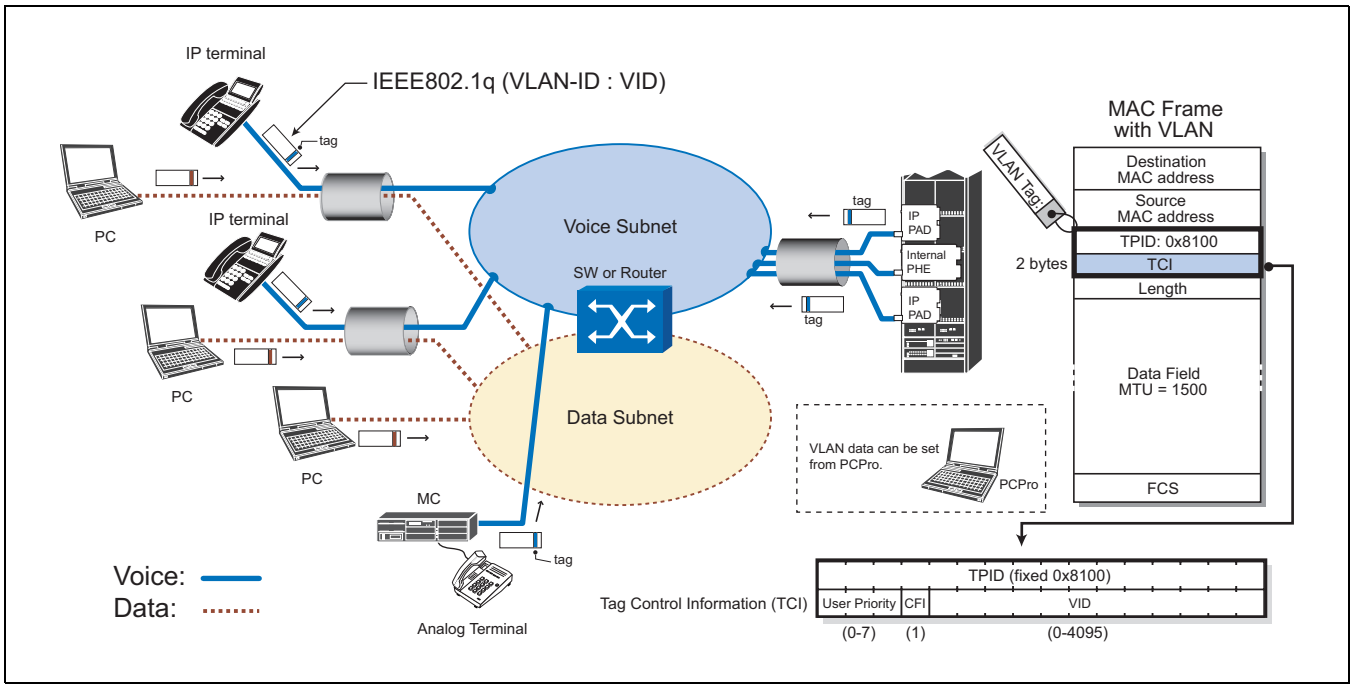
- A EC-ON/B EC-ON
- A EC-ON/B EC-OFF
- A EC-OFF/B EC-ON
- A EC-OFF/B EC-OFF

Note 1: When the same value is assigned to both A LOC-ID and B LOC-ID parameters, select either A EC-ON/ B EC-ON or A EC-OFF/B EC-OFF.

14.6 Tag VLAN Data

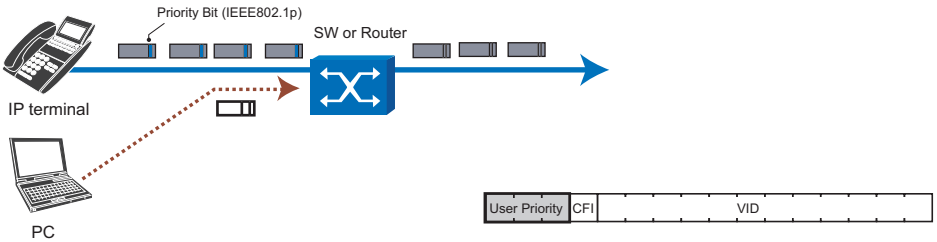
The implementation of Virtual LANs (VLANs) divides voice and data network into different segments at Layer 2. The system uses Tag VLAN, which conforms to IEEE 802.1q. To identify VLAN frame, a unique ID (VLAN-ID) must be tagged for each packet sent from the IP terminals and the internal PH/IPPAD cards.
To tag a VID to packets sent from the internal PH/IPPAD cards, the ALIDL is used. Packets from the IP terminal can be tagged by key operations on the terminal. User priority can also be used, by which you can give higher priority to voice packets to provide high speech quality. User priority bits conform to IEEE 802.1p. The following figure shows a MAC frame tagged with a VLAN tag.

Tag VLAN



Layer 2 Packet Priority Control

By using "User Priority" bits, Layer 2 packets can be processed with priority at the switch device.



[Precautions]

1. The following data can be set from the PCPro (ALIDL command). One set of data can be set on a circuit card basis.

User Priority:	User Priority to voice packet [0-7] (low ← 1, 2, 0, 3, 4, 5, 6, 7 → high)
Virtual LAN Identifier:	Virtual LAN Identifier for each internal PH/IPPAD Card [2-4094]
VLAN IP Address:	IP Address of internal PH/IPPAD for Virtual LAN
VLAN Mask Bit:	Mask Bit of internal PH/IPPAD IP Address for Virtual LAN

2. One IP terminal/PH · IPPAD belongs to a single VLAN. (cannot belong to multiple VLANs) The value of the VLAN-ID should be determined by taking the network environment into account.
3. The number of VLANs connected to a LAN is proportional to the number of accommodated internal PH/IPPAD cards.
4. This feature provides IEEE802.1pq-compatible Tag VLAN only. Port-base VLAN can be set at the network side.
5. The priority bit is basically used for giving high priority to voice packets. The value should be determined by taking the network environment into account.
6. Be sure to initialize the internal PH/IPPAD card after setting/changing the data of the card.
7. Different Sub-network addresses must be allocated to VLAN IP address (assigned to VLAN IP parameter of the ALIDL) and PH/IPPAD IP address (assigned to "PH IP/IPPAD IP" parameter of the ALIDL).
8. Same Sub-network address must be allocated to Default Gateway IP address (assigned to "Def_GW IP" parameter of ALID) and PH/IPPAD address (assigned to the PH IP/IPPAD IP parameter of ALIDL).

Internal PH/IPPAD card can belong to a maximum of 32 VLANs at the same time. The following firmware is required on IPPAD card.

Circuit Card	Firmware
IPPAD	SP-3716 (Issue 1) or later

1. Assign at least one Default GW (network address same as the PH IP address).
2. One of the IP addresses of the Default GW must correspond to the network address of the PH.
3. IP address of the Default GW must correspond to the network address of PH or VLAN.
4. The same IP address cannot be assigned as the Default GW.
5. Delete the Default GW of the same network address when deleting VLAN data.
6. Individual network address must be assigned to the IP address of the VLAN.
7. VLAN ID can be assigned the range from 1 to 4094.

8. A unique VLAN ID must be assigned for internal PH.

9. When VLAN data is modified, registration of the terminal logging in the VLAN must be performed again.

Programming

ALIDL - Assign the following VLAN data.

Note: Be sure to initialize the internal PH/IPPAD Card after assigning the VLAN data by using the ALIDL command. When the SP-3716 (Issue 1) or later firmware is equipped on IPPAD, the Card initialization after the VLAN data setting is not required.

PH LENS: Assign two-digit Module Group Number + one-digit Unit Number + First Group Number (two digits) of PH/IPPAD-accommodated slot.

TYPE: Select the kind of internal PH/circuit card from the combo box.

<PH Basic Settings>

PH IP/IPPAD IP: Enter IP address of Internal PH/IPPAD card.

MASK BIT: Enter Mask Bit of IP address.

PORT: Enter the port number for controlling signal (the default value is 61010).

REGMAX: Number of ports registered by the internal PH. (This parameter is not required for IPPAD.)

<VLAN Settings>

PRIORITY: User priority to voice packet [0-7] (low ← 1, 2, 0, 3, 4, 5, 6, 7 → high)

VLAN ID: Virtual LAN Identifier for each Internal PH/IPPAD card [1-4094]

VLAN IP: IP Address of PH/IPPAD for Virtual LAN. Be sure to assign IP address in different subnet to which PH IP belongs.

VLAN MASK BIT: Net Mask of PH/IPPAD IP address for Virtual LAN

<Def_GW Settings>

DEF_GW IP: Enter the IP address of the default gateway.

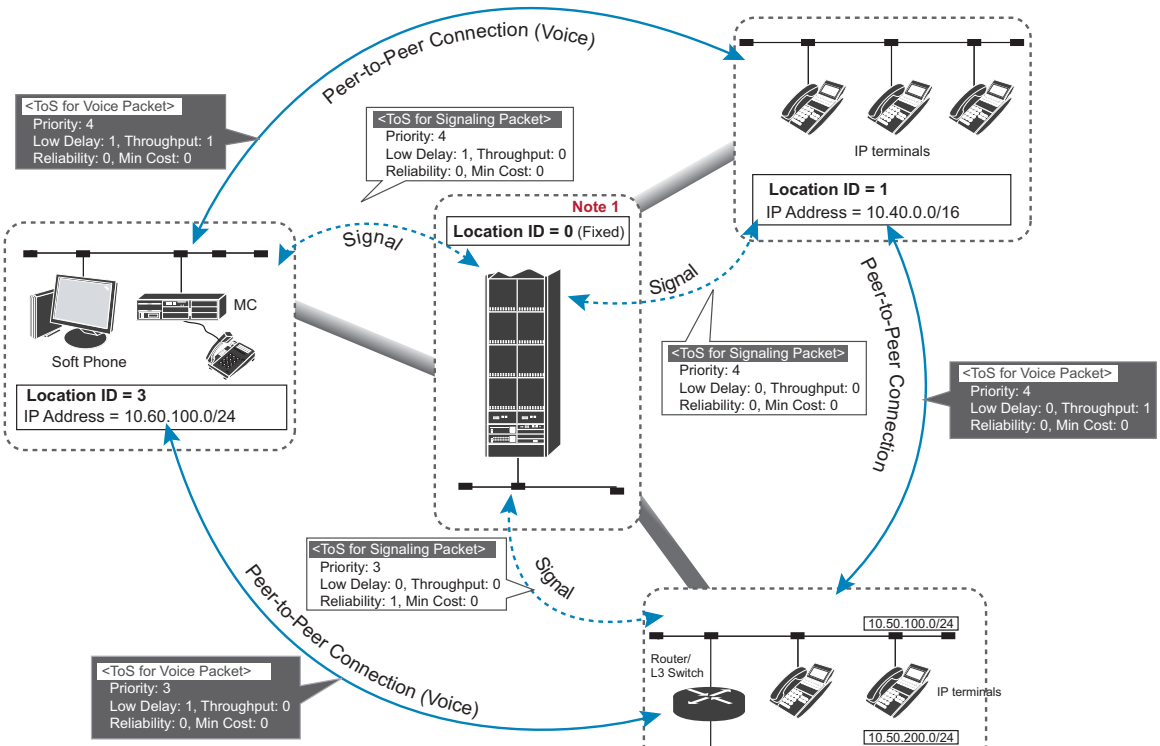
Note: When assigning two or more IP addresses, the IP addresses must be unique. Be sure to assign at least one IP address in the subnet to which PH IP belongs.

14.7 ToS Control Data

This section describes the data assignment to provide the Type of Service (ToS) control function between the IP terminals connected to the System. This function assigns priorities to the signaling packet and voice packet to ensure efficient, reliable communication within a network. Although the default ToS value is basically used, the ToS value can be changed on a Location ID basis, according to the network condition. The following figure shows an example of the network configuration using ToS Control.

Note: Each IP terminal is given a Location ID that identifies its location within the network, on a network address basis.

ToS Control



[Example]

・ Signaling Packet (Assigned by ALOCL command):

Location ID	Note 2 Priority	Low Delay	Throughput	Reliability	Min Cost
1	4	0	0	0	0
2	3	0	0	1	0
3	4	1	0	0	0

・ Voice Packet (Assigned by AIVCL command):

Location ID	The other Location ID	Note 3 Priority	Low Delay	Throughput	Reliability	Min Cost
1	1	5	0	0	0	0
1	2	4	0	1	0	0
1	3	4	1	1	0	0
2	2	5	0	0	0	0
2	3	3	1	0	0	0
3	3	5	0	0	0	0

Note 1: Location ID of the MGC location is fixed to "0".
Note 2: When no data is assigned, default value "6" is applied.
Note 3: When no data is assigned, default value "5" is applied.

Note: When ToS Control is applied for IPPAD, be sure to use all the IPPAD cards with voice compression function in the system.
- In FCCS via IP and CCIS via IP network, mount IPPAD cards corresponding to voice compression function in every node.

Note: When a SIP Handler Controlled SIP terminal is used, ToS control data can be assigned for SIP messages sent from the SIP Handler to the SIP terminal. In the Telephony Server side, ToS control data cannot be assigned for packets from the SIP terminal such as SIP messages and RTP packets (ToS control data is dependent on the SIP terminal specification).

Programming

STEP 1: **ALOCL** - Assign the Location ID data.

TYPE:	Select “Network Address”.
IP ADDRESS:	Assign the IP Address. For the System unit, assign 0.0.0.0.
MASK BIT:	Assign the Mask Bit in the range 1-31 except when IP ADDRESS 0.0.0.0 is determined. When IP ADDRESS = 0.0.0.0, MASK BIT = 32 is automatically assigned.
LOC-ID:	Assign any number in the range 0-4095, associated with the network address assigned in the IP ADDRESS parameter. When 0.0.0.0 is assigned in the IP ADDRESS parameter, LOC-ID is automatically set to “0” (the value cannot be changed).

Note: LOC-ID=0 is predetermined to the location ID of System unit/MGC (IPPAD).

- Check the Setting Type of Signaling Packet Service checkbox to assign the following ToS for Signaling Packet data.

Note: If the check box is not selected, the default settings (IP Precedence, Precedence = 6) will be provided.

Note: Initialization of IPPAD (by using MB Key or MBPM PCPro Command) and Re-registration of each IP Station are required for this data change to take effect.

Select either **IP PRECEDENCE** or **DIFFSERV**.
When IP PRECEDENCE is selected
Priority (PRECEDENCE / DSCP): Specify the precedence in the range 0-7 (low-high).
Low Delay: Place a checkmark in the checkbox to set Low Delay to ON.
Through Put: Place a checkmark in the checkbox to set High Throughput to ON.
Reliability: Place a checkmark in the checkbox to set High Reliability to ON.
Min Cost: Place a checkmark in the checkbox to set Cost Precedence to ON.
When DIFFSERV is selected
Priority (PRECEDENCE / DSCP): Specify the DSCP in the range 0-63.

STEP 2: **AIVCL** - Assign the IP service Voice Control data.
A LOC-ID: Location ID of the Self Location. (0-4095)
B LOC-ID: Location ID of the Target Location. (0-4095)

Note: LOC-ID=0 is used for the Location ID of System unit/MGC (IPPAD).

Note: Only the Location ID corresponding to the network address can be assigned on the AIVCL/AIVCN when assigning settings for locally programmed terminals. The Location IDs corresponding to FPC or STN cannot be assigned.

[Setting Type of Voice Packet Service]
Select the Setting Type of Voice Packet Service check box to assign the following ToS for Voice Packet data.

Note: If the check box is not selected, the default settings (IP PRECEDENCE, Priority (PRECEDENCE/DSCP) = 5) will be provided.

Choose either **IP PRECEDENCE** or **DIFFSERV**.
When IP PRECEDENCE is chosen
Priority (PRECEDENCE/DSCP): Specify the precedence in the range 0-7 (low-high).
Low Delay: Place a check mark in the check box to set Low Delay to ON.
Through Put: Place a check mark in the check box to set High Throughput to ON.
Reliability: Place a check mark in the check box to set High Reliability to ON.
Min Cost: Place a check mark in the check box to set Cost Precedence to ON.
When DIFFSERV is chosen
Priority (PRECEDENCE/DSCP): Specify the DSCP in the range 0-63.

14.8 PAD/EC Control IP/PHC Support

By assigning system data (ASYDL, SYS1, Index 825), default PAD/EC value that is appropriate to the network environment is automatically applied. If the value is not appropriate for a particular network or terminal, it can be adjusted to match the environment using the APDIL command in PCPro. PAD/EC value can be set on an IP device, IPPAD basis. Default PAD/value is provided on a country basis.

Note: This feature may not be available depending on the firmware version of the IP equipment. Refer to the following table.

List of Firmware Requirements for PAD/EC Control IP/PHC Support

NO.	IP TERMINAL/EQUIPMENT	FIRMWARE REQUIREMENT	REMARKS
1	IP Enabled Dterm (Dterm Series *i*)	Ver. 2.80 or higher	
2	DtermIP	Ver. 2.80 or higher	
3	DT700 Series	Available since first version	
4	DT830G/DT830	Ver. 3.00 or higher	
5	DT820 Series	Available since first version	
6	DtermIP INASET	Ver. 2.80 or higher	Not available when using ITR-LC-1 (BK)-(H).
7	DtermSP30	Ver. 4.0 or higher	
8	Soft Client SP350	Available from the first version.	
9	Analog MC	SP-3753 2AMC PROG-B Issue 10A or higher	
10	Software-based MG-SIP	Available from the first version.	
11	MG-SIP128 [MG-128SIPMGL-A]	Available from the first version.	**Note 1**
12	MG-SIP128 [MG-128SIPMGG-B/MG-128SIPMGG]	SP-4058 MGSIP PROG-J Issue 1 or higher SP-4051 MGSIP PROG-G Issue 2 or higher	**Note 1**
13	MG-SIP128 [MG-128SIPMGJ-B/MG-128SIPMGJ]	SP-4060 MGSIP PROG-L Issue 1 or higher SP-4032 MGSIP PROG-E Issue 2 or higher	**Note 1**
14	MG-SIP16 [SCA-16SIPMG(US)-B/ SCA-16SIPMG(US)]	Not available.	
15	MG-SIP96	Not available.	
16	MG-SIP16 [SCA-16SIPMGA/MG-16SIPMGA]	Not available.	
17	MG-SIP16	Not available.	

NO.	IP TERMINAL/EQUIPMENT	FIRMWARE REQUIREMENT	REMARKS
18	MC-MG(COT)	SP-3778 MCMG PROG-A Issue 8A or higher	
19	VS32	SP-3789 VS32 PROG-A Issue 1A or higher SP-3789 VS32 PROG-B Issue 1A or higher	
20	IPPAD	SW439 Ver. 3.0 or higher SW450 Ver. 1.0 or higher	
21	MG(BRI)-SIP [SCA-2BRIA/MG-2BRIA]	SP-3826 BRI MG SIP BSC PROG-A Issue 1B or higher	
22	MG(PRI) [SCA-24PRIA/MG-24PRIA]	Proprietary Protocol mode: SP-3884 MG PRI PROG-A 1A or higher SIP mode: SP-3884 5A or higher	
23	MG(PRI) [SCA-30PRIA/MG-30PRIA]	Proprietary Protocol mode: Not defined yet. SIP mode: Not defined yet.	
24	MG(COT) [SCA-6COTA]	SP-3875 6MGCOT PROG-A 1A or higher	
25	MG(COT) [SCA-6COTB]	SP-3948 6MGCOT PROG-A 1A or higher	
26	MG(COT) [SCA-6COTC]	SP-3948 6MGCOT PROG-A 1A or higher	
27	MCMG(COT) [SCA-4LC2COTA/MG-4LC2COTA]	SP-3778 MCMG PROG-A 8A or higher SP-3927 MCMG PROG-B 1A or higher	
28	8LC Card [SCA-8LCA/SCA-8LCA-EMEA]	SP-3893 8MC PROG-A 1A or higher	
29	8LC Card [SCA-8LCC/SCA-8LCC-EMEA]	SP-3983 8MC PROG-B 1A or higher	
30	MG-T1(SIP) [SCA-24DTIA]	SP-3926 MG DTI PROG-A Issue 1A or higher	
31	IPG Digital	Available since the first version	
32	IPG Analog	Available since the first version	
33	UG50	Available since the first version	

Note 1: PAD values can be changed from PCPro using the ASYD/ASYDL/ASYDN commands or from the Maintenance Console using configuration commands.

To set PAD/EC value, one of the following can be used.

14.8.1 Initial Value

Initial PAD/EC value is applied on a system basis by assigning ASYDL, SYS1, Index 825, Bit 0, Bit 1, Bit 4, and Bit 5. As to the initial value, click the Initial Value Display button.

14.8.2 Initial Value Modification

If the initial value is not adequate for a particular network, the PAD/EC value can be modified on a Connection Type (**Note 1**) basis. There are two ways to change the PAD/EC data.

(1) PAD/EC Initial Value Modification

After selecting an IP device from the list box, the default value can be modified on an IP device basis.

(2) Designated Connection Pattern

After selecting Connection Pattern (P2P Connection / Legacy Connection), PAD/EC value can be modified by using Service Feature Restriction Class (SFC) and/or Route Number (RT) in combination. This data assignment is useful when the default is not sufficient for a particular terminal such as an Off-Premise Station (OPS).

Note 1: The following Connection Type tabs in the APDIL command can be used:
- Station-to-Station tab
- Tandem tab
- OG/IC (Station) tab
- OG/IC (Trunk) tab
- Other tab160
- FCCS tab
- Advanced tab

14.8.3 Designated Station Assignment

Default PAD/EC value or PAD/EC value set by the above setting can be adjusted by applying additional value (±α) on a Station basis. To facilitate the data assignment of stations that have serial Station Numbers, 'en bloc data assignment' is also available. If a case arises where voice is too low to hear from a particular Station, the PAD value can be tuned by specifying the Station Number.

PAD Value Setting by Designated Station

	RECEIVE	SEND
DEFAULT	9 (dB)	0 (dB)
Additional Value for Designated Station	-3 (dB)	3 (dB)
APPLIED PAD VALUE	6 (dB)	3 (dB)

Note: Minus (-) indicates gain and Plus (+) indicates loss in electronics. The setting towards Minus increases volume and the setting towards Plus decreases the volume. Note that Plus symbol is not displayed in PCPro.

- Conditions

1. To use this PAD/EC control, the following data assignment is required.

   - ASYDL, SYS1, Index 825, Bit 0=1, Bit 1=0 (PAD Control IP support is in service)
   - ASYDL, SYS1, Index 825, Bit 4=1, Bit 5=0 (PAD Control PHC support is in service)
   - APDIL (PAD value assignment for IP devices, IPPAD circuit card) **Note 1**
   - APDCL (PAD value detailed assignment for IP devices, IPPAD circuit card via PHC) **Note 1**

   Note 1: When modifying the default value, this command is used.

2. This PAD/EC control has precedence over the following PAD related commands:

   - ASYDL, SYS1, Index 812, Bit 0 to Bit 4 (PAD value for IP terminal via PHC), Bit 7 (PAD control for PHC incoming/outgoing call) and Index 813, Bit 0 to Bit 3 (PAD value for Analog MC via PHC) **Note 2**
   - APAD/APADN
   - AIVCL
   - AFPD
   - ARTD/ARTDN
   - ACRD (PAD value assignment for IPPAD)
   - ASYD, SYS1, Index 4, Bit 5=0 (5 dB PAD for Station-to-Station is in service)

   Note 2: This is applied for a call originated from an IP terminal via PHC. PAD value is specified by the IP terminal kind of the calling party. The terminal kind of the called party is not considered. The following conditions exist: (1) This PAD is applied for IP terminals only. Not applied for Attendant Console. (2) This PAD control is activated when the terminal answers the call via PHC.

3. This PAD control is applied in the following precedence order:

   - When not using PHC

PRECEDENCE	DATA ASSIGNMENT
1	APDIL Command [Connection Pattern Designed]
2	APDIL Command [Initial Value Modification]
3	Initial Value

   - When using PHC

PRECEDENCE	DATA ASSIGNMENT
1	APDCN Command [Connection Pattern Designed]
2	APDCL Command [Connection Pattern Designed]
3	APDCN Command [Initial Value Modification]
4	APDCL Command [Initial Value Modification]

PRECEDENCE	DATA ASSIGNMENT
5	Initial Value

4. MG-SIP has two modes of PAD value setting: the auto mode and the local setting mode. The auto mode allows the user to set the system data from PCPro; the local setting mode allows the user to execute the configuration commands from the Maintenance Console. To use the auto mode, start the `set softdsp` command in the Maintenance Console and select “auto” for the PAD setting. For details on the `set softdsp` command, see the corresponding MG-SIP section in Peripheral Equipment Description (IP Devices).
5. When setting PAD values in the auto mode, consider the following:

   - For PAD setting on MG-SIP side, choose "BRI/PRI/T1" from the IP Devices list of the APDIL command.
   - For PAD setting of other devices connected to MG-SIP, choose "C.O.Line[D]" from the Connect to list of the APDIL command.

   In a configuration where MG-SIP / MG(BRI) / MG(PRI) are connected to the same Telephony Server, if "BRI/PRI/T1" is chosen for the IP Devices parameter, the PAD setting will be applied to all these gateways. To assign different PAD values for each gateway, use the configuration command in the Maintenance Console (local setting mode). In that case, select "local setting" for the PAD setting in the `set softdsp` command before setting PAD value (voice/FAX).
6. The following conditions must be considered in the setting of PAD values for MG-SIP:

   - When "local setting" is selected for the PAD setting in the `set softdsp` command, the PAD value specified in the configuration command takes precedence over the one specified in PCPro, even if the MG-SIP receives PAD value information from the PCPro.
   - G.711 μ-law/A-law is only available as a codec of PAD adjustment.
   - Echo Canceler Control function is not available for MG-SIP even when both EC and FAXEC parameters are set to ON with the APDIL command.
   - The Smooth PAD function is not available. Misadjustment of signal amplification caused by an unsuitable PAD value can produce clipping and unwanted noise.
7. When PAD adjustment is implemented during the activation of the multi-session function of MG-SIP, the PAD value that was set at the time of call setup is applied to RTP on the Telephony Server in the same way as a one-to-one connection between SIP network and Telephony Server.
8. PAD adjustment is available even when PCM data is processed through the μ-law/A-law conversion function of MG-SIP.
9. This PAD control cannot be applied to a legacy Station / Trunk. Note that it is applied to IPPAD circuit cards.
10. PAD value change may affect inband PB signal as well as speech level. Therefore, after the data change, make sure the change does not affect any service features that use inband PB signal.
11. When T.30 is assigned as the payload type of FAX, the PAD value for the FAX assigned by this feature becomes invalid. In this case, the PAD value for the FAX is set to 0dB (fixed).

Programming

<PAD Control>

STEP 1: **ASYDL** - Assign the following system data.
SYS1
Index 811: Tone Code by Countries

ASYDL, SYS1, Index 811			
TONE CODE	**Basic Tones**	**TONE CODE**	**EMEA Tones**
00 H	Not used	C8 H	Germany
01 H	Japan	C9 H	Italy
02 H	North America	CA H	Netherlands
03 H	Australia	CC H	Denmark
04 H	Other countries	CD H	Sweden
05 H	Hong Kong	D3 H	Spain
06 H	Malaysia	D4 H	Austria
07 H	Singapore	D5 H	Belgium
08 H	England	D6 H	England [For EMEA]
09 H	Mexico	D7 H	Greece
0A H	Taiwan	D8 H	Switzerland
0B H	New Zealand	D9 H	South Africa
0C H	Korea	Other Values	Not used
0D H	Brazil		
0E H	China		
0F H	Thailand		
10 H	Thailand (EGAT)		
11 H	Sri Lanka		

Index 825, Bit 0=1, Bit 1=0: PAD / EC IP support is in service

ASYDL, SYS1, Index 825		
Bit 1	**Bit 0**	**PAD/EC Control**
0	**0**	PAD/EC Control IP support is out of service
0	**1**	PAD/EC Control IP support is in service
1	**0**	-
1	**1**	-

When using PHC, assign also the following bits of ASYDL, SYS1, Index 825.

ASYDL, SYS1, Index 825		
Bit 5	**Bit 4**	**PAD/EC Control**
0	**0**	PAD/EC Control PHC support is out of service
0	**1**	PAD/EC Control PHC support
1	**0**	-
1	**1**	-

STEP 2: **APDIL** - Assign PAD/EC control data that supports IP. This PAD/EC control data is applied for calls not via PHC. If you want to assign PAD/EC data to PHC, go to page 130.
[Connection Type] Select an appropriate tab from the following.

Station-to-Station: Used when an IP device (Station) is connected to a Station.
Tandem: Used when an IP device (Trunk) is connected to C.O. line or TIE line.
OG/IC (Station): Used when an IP device (Station) is connected to C.O. line or TIE line.
OG/IC (Trunk): Used when an IP device (Trunk) is connected to a Station.
Other: Used when an IP device (Station/Trunk) is connected to other than above.
FCCS: Not used here.
Advanced: Used when assigning additional value to the initial value or modified value by "Initial Value Modification" or "Connection Pattern Designed" or used for IP device/IPPAD en bloc data assignment.

- **When selecting [Station-to-Station] tab**

<Initial value modification setting>

IP Devices: Select an IP device to set from the drop-down list box and then click the Setting button.

MultiLine Telephone(IP)
MC
IP-BS **Note 3**

Note 3: Not used.

Connect to: "Station" is displayed.

<Connection pattern designed setting>

Connection Pattern: Choose a connection pattern to set and then click the Setting button.

P2P connection: Peer to Peer Connection
Legacy connection: Connection between IP devices and legacy terminal

Note: Please assign the remainder of data following the parameters on the screen.

- **When selecting [Tandem] tab**

<Initial value modification setting>

IP Devices: Choose a device to set from the drop-down list box and then click the Setting button.

MC-MG(COT)
BRI/PRI/T1

Connect to: Select a destination from the drop-down list box.

C.O.Line[A]
Tie Line
C.O.Line[D]

<Connection pattern designed setting>

Connection Pattern: Choose a connection pattern to set and then click the Setting button.

P2P connection: Peer to Peer Connection
Legacy connection: Connection between IP devices and legacy terminal

Note: Please assign the remainder of data following the parameters on the screen.

- **When selecting [OG/IC (Station)] tab**

<Initial value modification setting>

IP Devices: Select an IP device to set from the drop-down list box and then click the Setting button.

MultiLine Telephone(IP)
MC
IP-BS **Note 4**

Note 4: Not used.

Connect to: Select a destination from the drop-down list box.

C.O.Line[A]
Tie Line
C.O.Line[D]

<Connection pattern designed setting>

Connection Pattern: Choose a connection pattern to set and then click the Setting button.

P2P connection: Peer to Peer Connection

Legacy connection: Connection between IP devices and legacy terminal

Note: Please assign the remainder of data following the parameters on the screen.

- **When selecting [OG/IC (Trunk)] tab**

<Initial value modification setting>

IP Devices: Choose a device to set from the drop-down list box and then click the Setting button.

MC-MG(COT)

BRI/PRI/T1

Connect to: “Station” is displayed.

<Connection pattern designed setting>

Connection Pattern: Choose a connection pattern to set and then click the Setting button.

P2P connection: Peer to Peer Connection

Legacy connection: Connection between IP devices and legacy terminal

Note: Please assign the remainder of data following the parameters on the screen.

- **When selecting [Other] tab**

<Initial value modification setting>

IP Devices: Select a device to set from the drop-down list box and then click the Setting button.

MultiLine Telephone(IP)

MC

IP-BS **Note 5**

MC-MG(COT)

BRI / PRI / T1

VPS **Note 5**

VS(MUSIC)

VS(3-party CONF)

VS(Multiple CONF)

Note 5: Not used.

Connected devices: This parameter is displayed only when "VS(3-party CONF)" or "VS(Multiple CONF)" is selected for the IP Devices parameter. Select one from the drop-down list box and then click the Setting button.

Station only

Station + C.O./Tie Line

Station + Station + UMS **Note 6**

Station + C.O./Tie Line + UMS **Note 6**

Note 6: This cannot be used when "VS(Multiple CONF)" is selected for IP Devices.

Connect to: Select a destination from the drop-down list box.

Note: The available options vary depending on the selected IP device.

Station

C.O.Line[A]

Tie Line

ATT

MUSIC

C.O.Line[D]

3-party CONF

UMS Equipment

Note: Please assign the remainder of data following the parameters on the screen.

- **When selecting [FCCS] tab**

<Connection pattern setting designed>

LAN side: Select one of the following items and then click the Setting button.

Trunk

Station

IPPAD(Other node)

Connection pattern: "Legacy connection" is displayed.

<LAN side>

Kind of connection: "Trunk," "Station" or "IPPAD (Other node)" is displayed.

RT/LGRT: Assign Route Number when "Trunk" is selected.

SFC: Assign SFC when "Station" is selected.

<TDSW side>

Kind of connection:

- When “Trunk” or “Station” is selected for the LAN side parameter, “IP-PAD(FCCS connection)” is displayed.
- When “IPPAD(Other node)” is selected for the LAN side parameter, “IP-PAD(P2P-FCCS connection)” is displayed.

C_RT: Assign Connection Route Number.

Note: Please assign the remainder of data following the parameters on the screen.

- **When selecting [Advanced] tab**

<Station designed setting>

Kind of designation: Choose either of the following.

Individual

Collective

When Individual is selected:

TN: Tenant Number

STN: Station Number

When Collective is selected:

TN: Tenant Number

START STN: 1st Station Number

END STN: Last Station Number

Note: Please assign the remainder of data following the parameters on the screen.

<Initial value modification collective setting (IP Devices)>

IP Devices: Select a device to set from the drop-down list box and then click the Setting button.

MultiLine Telephone(IP)

MC

IP-BS **Note 7**

MC-MG(COT)

BRI/PRI/T1

VPS **Note 7**

VS(MUSIC)

VS(3-party CONF)

VS(Multiple CONF)

Note 7: Not used.

Connected devices: This parameter is displayed only when "VS(3-party CONF)" or "VS(Multiple CONF)" is selected for the IP Devices parameter. Select one from the drop-down list box and then click the Setting button.

Station only

Station + C.O./Tie Line

Station + Station + UMS **Note 8**

Station + C.O./Tie Line + UMS **Note 8**

Note 8: This cannot be used when "VS(Multiple CONF)" is selected for IP Devices.

Note: Please assign the remainder of data following the parameters on the screen.

<Initial value modification collective setting (IPPAD)>

TDSW side terminal: Select one from the below and click the Setting button.

Station

C.O.Line[A]

Tie Line

ATT

C.O.Line[D]

MUSIC

3-party CONF

UMS Equipment

FCCS

Note: Please assign the remainder of data following the parameters on the screen.

STEP 3: **APDCL/APDCN** - Assign PAD control data that supports IP. This step is required to assign PAD/EC data for calls via PHC. If PHC is not used, skip this step.

- **Initial value modification tab**

Self office connection device: Select a device to set from the drop-down list box and then click the Setting button.

Legacy Station

MultiLine Telephone(IP)

MC

IP-BS **Note 9**

ATT

Legacy Trunk (Tie Line)

Legacy Trunk (C.O.Line[A])

Legacy Trunk (C.O.Line[D])

BRI/PRI/T1

MC-MG(COT)

CONF Trunk(3-party)

VS(3-party CONF)

VS(Multiple CONF)

Announcement Trunk

VS(MUSIC)

FCCS

VPS **Note 9**

UMS Equipment

Note 9: Not used.

Connected devices: This parameter is displayed only when "VS(3-party CONF)" or "VS(Multiple CONF)" is selected for the Self office connection device parameter. Select one from the drop-down list box and then click the Setting button.

Station only

Station + C.O./Tie Line

Station + Station + UMS **Note 10**

Station + C.O./Tie Line + UMS **Note 10**

Note 10: This cannot be used when "VS(Multiple CONF)" is selected for IP Devices.

Remote office PC: Enter the destination Point Code. **Note 11**

Note 11: When CCIS Link Reconnection is enabled, set a PAD value/Echo Canceller Control to be used after Link Reconnection for the Remote Office PC parameter where Self-office PC is set (i.e., Remote Office PC = Self-office PC).

Connect to: Select a connection destination type of the remote office.

Note: The available options vary depending on the selected Self office connection device.

Station

C.O.Line[A]

Tie Line

ATT

C.O.Line[D]

MUSIC

3-party CONF

UMS Equipment

No Send Terminal KIND **Note 12**

Note 12: Select this option when a counter office is NEAX2400 IPX (Feature Package 8800 (R15) or earlier)/SV9300/SV8300/NEAX2000 IPS.

Note: Please assign the remainder of data following the parameters on the screen.

- **Connection designed tab**

Connection pattern: Select a connection pattern to set and click the Setting button.

Trunk - (PHC) - Trunk

Trunk - (PHC) - Station

Station - (PHC) - Trunk

Station - (PHC) - Station

<Self Office Side>

Assign the information about the connected device at self office.

Kind of connection: This is automatically displayed.

RT/LGRT: Assign the Route Number of the connected device when you have selected “Trunk - (PHC) - Trunk” or “Trunk - (PHC) - Station” for the Connection pattern parameter.

SFC: Assign SFC of the connected device when you have selected “Station - (PHC) - Trunk” or “Station - (PHC) - Station” for the Connection pattern parameter.

<Remote Office Side>

Assign the information about the connected device at remote office.

Kind of Connection: This is automatically displayed.

PC: Assign the PC of end office.

RT/LGRT: Assign the Route Number of the connected device when you have selected “Trunk - (PHC) - Trunk” or “Station - (PHC) - Trunk” for the Connection pattern parameter.

SFC: Assign SFC of the connected device when you have selected “Trunk - (PHC) - Station” or “Station - (PHC) - Station” for the Connection pattern parameter.

Note: Please assign the remainder of data following the parameters on the screen.

Sample Data Assignment 1

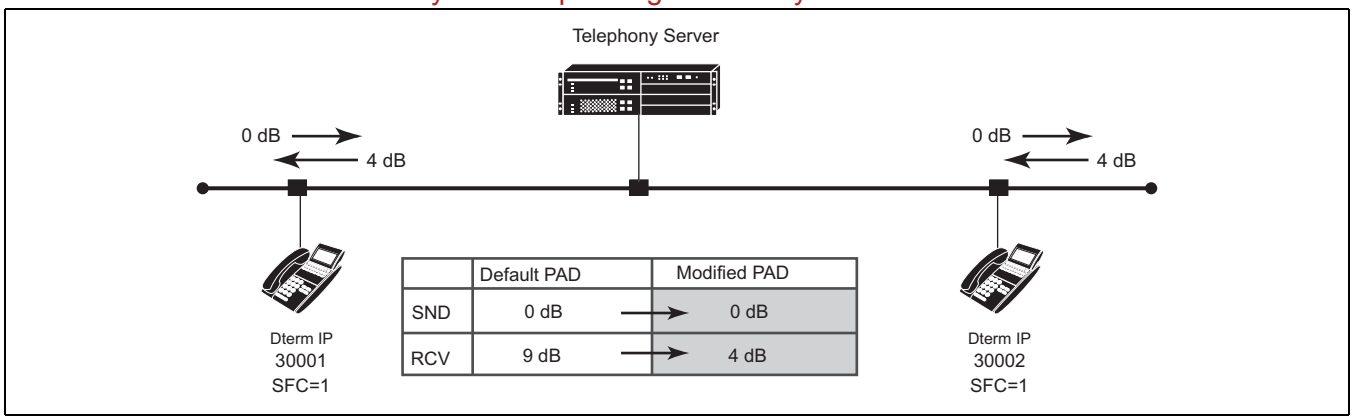
The following shows sample data for assigning PAD data. For more information about the settings of network configuration, refer to Sample Data Assignment 2.

[When not using PHC]

PAD Value Assignment for DtermIP-to-DtermIP call

The following example shows a case where default PAD value (**Note 1**) is changed for DtermIP-to-DtermIP call.

Note 1: Default PAD value may differ depending on country.



<Data Assignment>

- When using Initial Value Modification:

  APDIL: Station-to-Station tab → Initial Value Modification

  IP Devices: DtermIP
  Connection Destination: Station
  T_PAD: 0 dB
  R_PAD: 4 dB

- When using Designated Connection Pattern

  APDIL: Station-to-Station tab → Designated Connection Pattern

  Connection Pattern: P2P Connection
  IP Devices A:
  Kind: Station
  SFC: 1

  IP Devices B:
  Kind: Station
  SFC: 1

[ ] Select ALL: All items listed below are checked.
[×] Speech PAD adjustment
[×] EC is set
[ ] FAX PAD adjustment
[ ] FAX EC is set

T_PAD: 0 dB
R_PAD: 4 dB

- When using Designated Station Assignment

  APDIL: Station-to-Station tab → Designated Station Assignment

  Kind of Designation: Individual

  TN: 1
  STN: 30001 (30002)

  [ ] Select ALL: All items listed below are checked.
  [×] Speech PAD adjustment
  [×] EC is set
  [ ] FAX PAD adjustment
  [ ] FAX EC is set

  T_PAD: 0 dB
  R_PAD: 4 dB

[When using PHC]

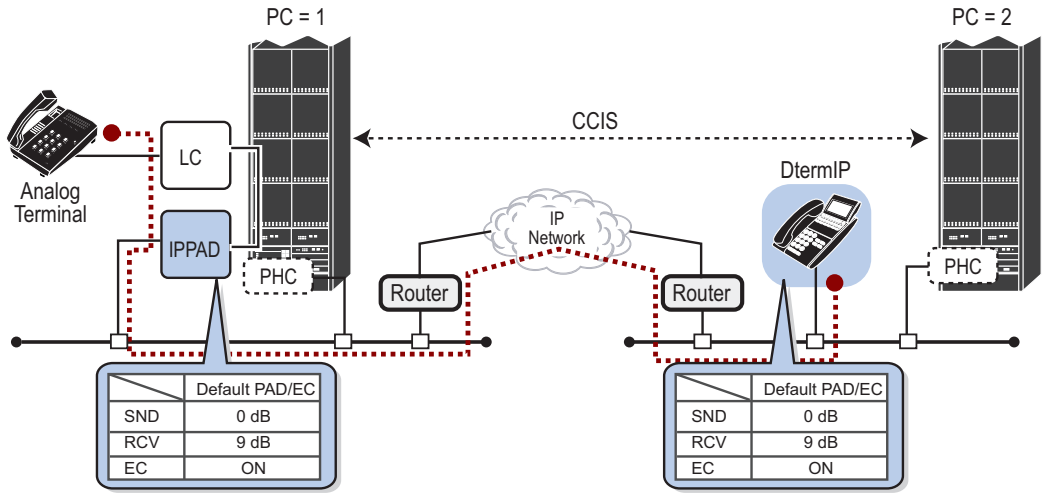
The one of the following patters can be used:

- Initial PAD/EC Value
- Initial PAD/EC Value Modification
- Designate Connection PAD/EC Assignment

Sample 1: When Initial PAD/EC Value is used.

The following example shows a case where default PAD/EC value (**Note 13**) is used for legacy terminal to DtermIP call. To apply the default value, use ASYDL, SYS1, Index 825 only is required.

Note 13: Default PAD value may differ depending on country.



Default PAD / EC is applied to IPPAD.

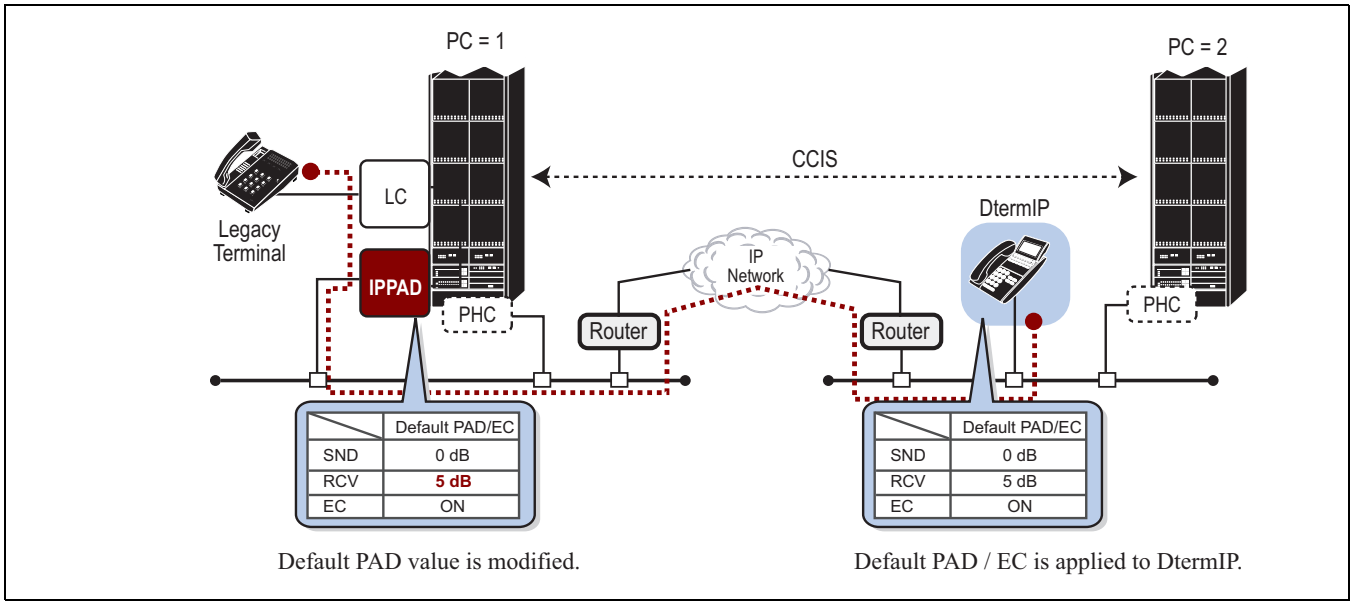
Default PAD / EC is applied to DtermIP.

Sample 2: When Initial PAD/EC Value Modification is used [1]

To change the default value, APDCL is required in addition to ASYDL, SYS1, Index 825. In this example, data assignment is required for Node (PC=1).

Select Initial Value Modification tab.
Self-node Connection Device: Legacy Terminal
Destination Point Code: 2
Connection Destination: Station
T_PAD: 0 dB
R_PAD: 5 dB
EC: ON

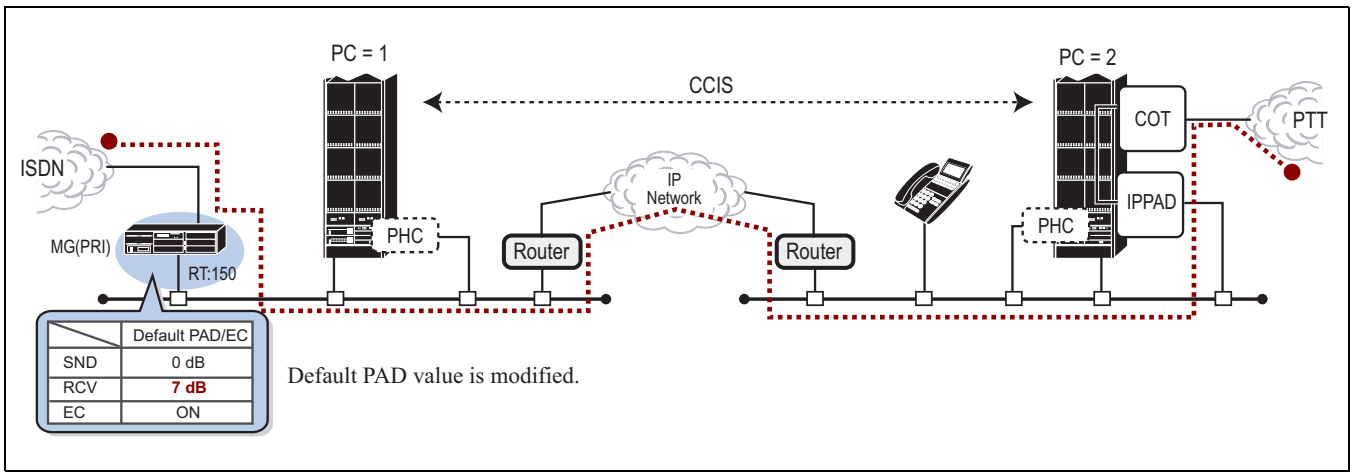
Initial PAD/EC Value Modification for MG-PRI - PHC - COT



Sample 3: When Initial PAD/EC Value Modification is used [2]

Select Initial Value Modification tab.
Self-node Connection Device: ISDN Trunk BRI/PRI or MG(BRI)/MG(PRI)
Destination Point Code: 2
Connection Destination: C.O. Line [D]
T_PAD: 0 dB
R_PAD: 7 dB
EC: ON

Initial PAD/EC Value Modification for MG-PRI - PHC - COT

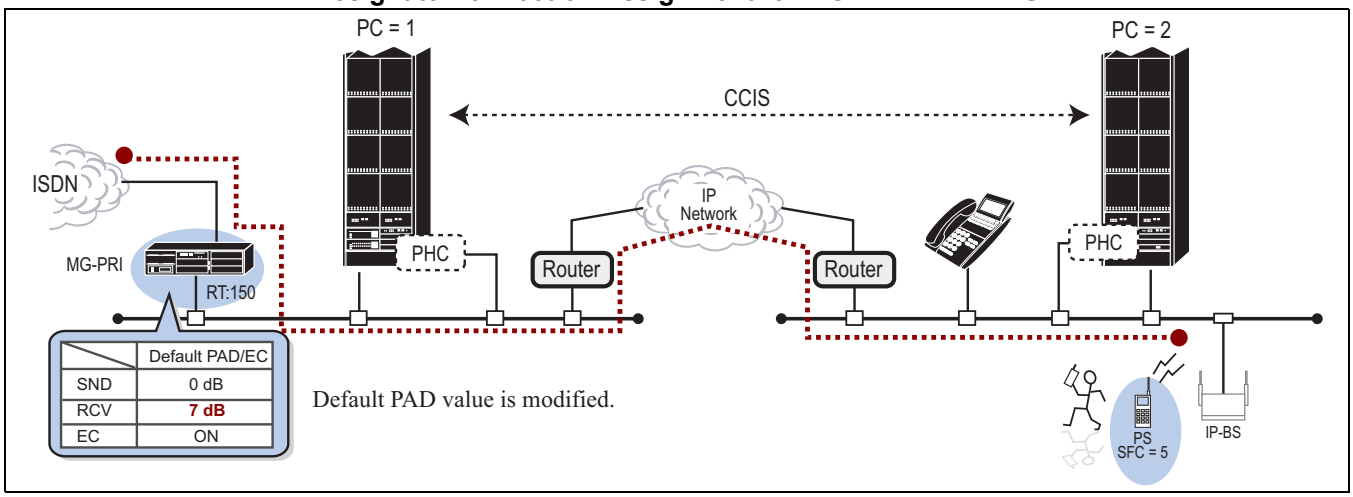


Note: When a node (PC=2) as mentioned in the above makes a tandem connection through PHC and COT, PAD/EC values specified by ARTD/APAD command in the node (PC=2) is applied to COT. Therefore, the sum (PAD value set in IPPAD + PAD value specified by ARTD/APAD command) will be used. Also besides, when you assign data by APAD command, although it usually assigns data by combinations of RT numbers for Bch in PHC and RT numbers in COT, PAD/EC values assigned to RT of PHC become invalid while PAD/EC values assigned to RT of COT become valid.

Sample 4: When Designate Connection PAD/EC Assignment is used [1]

Select [Designate Connection Assignment] tab.
[Self Node Information]
Kind: Trunk
RT/LGRT: 150
[Destination Node Information]
Kind: Station
PC: 2
SFC: 5
T_PAD: 0 dB
R_PAD: 7 dB
EC: ON

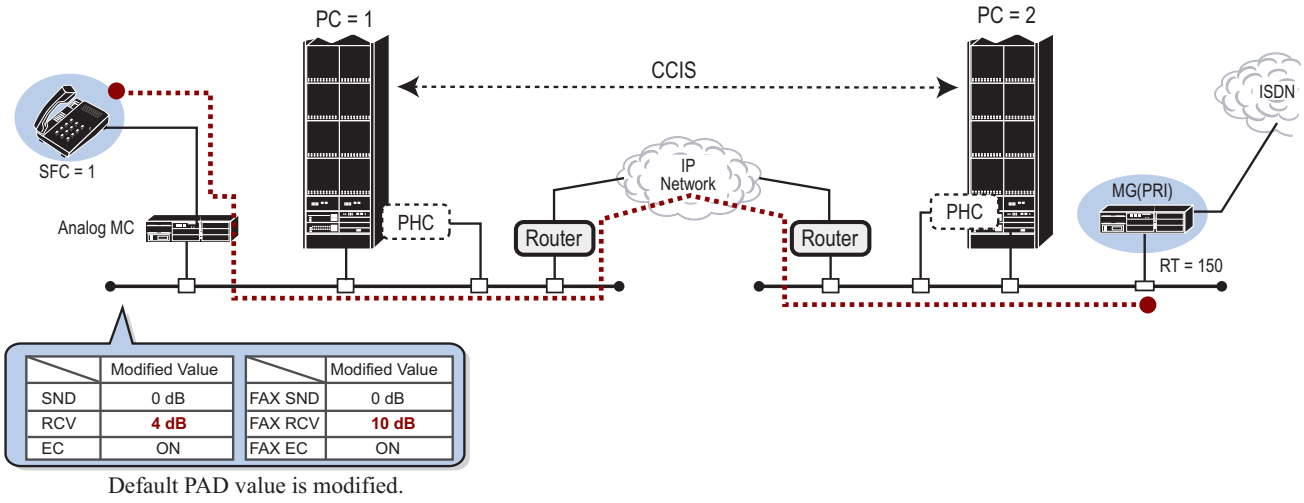
Designate Connection Assignment for MG-PRI - PHC - PS



Sample 5: When Designate Connection PAD/EC Assignment is used [2]

Select [Designate Connection Assignment] tab.
[Connection Patter]
Station-(PHC)-Trunk
[Self Node Information]
Kind: Station
SFC: 1
[Destination Node Information]
Kind: Trunk
PC: 2
RT: 150
T_PAD: 0 dB
T_FAXPAD: 0 dB
R_PAD: 4 dB
R_FAXPAD: 10 dB
EC: ON

Designate Connection Assignment for Legacy Station - PHC - MG(PRI)



	Modified Value		Modified Value
SND	0 dB	FAX SND	0 dB
RCV	**4 dB**	FAX RCV	**10 dB**
EC	ON	FAX EC	ON

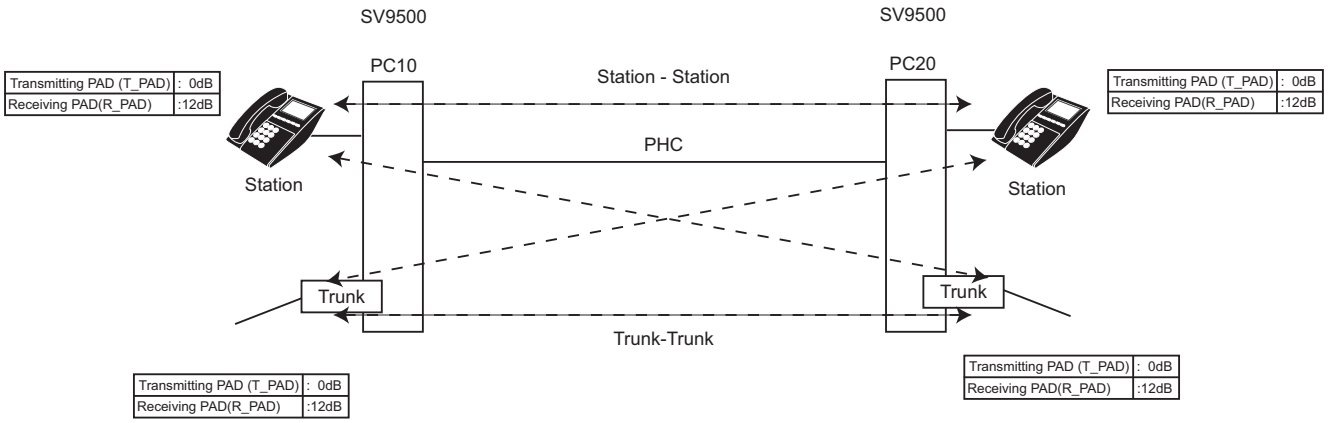
Default PAD value is modified.

Sample Data Assignment 2

This section describes the network settings for each network configuration as examples. The data settings in the following cases are described here.

- Standalone to Standalone
- Standalone to Standalone (NEAX2400 IPX (Feature Package 8800 (R15) or earlier)/SV9300/SV8300/NEAX2000 IPS)
- Standalone to FCCS
- Standalone to Peer-to-Peer FCCS
- FCCS to Standalone
- FCCS to Standalone (NEAX2400 IPX (Feature Package 8800 (R15) or earlier)/SV9300/SV8300/NEAX2000 IPS)
- FCCS to Peer-to-Peer FCCS
- Peer-to-Peer FCCS to Standalone
- Peer-to-Peer FCCS to Standalone (NEAX2400 IPX (Feature Package 8800 (R15) or earlier/SV9300/SV8300/NEAX2000 IPS)
- Peer-to-Peer FCCS to FCCS
- FCCS to FCCS
- Peer-to-Peer FCCS to Peer-to-Peer FCCS

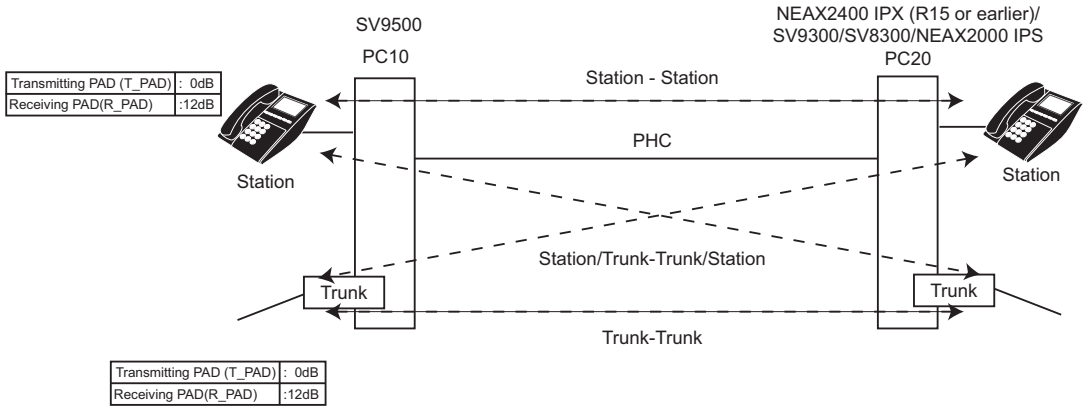
- Standalone to Standalone



		COUNTER OFFICE (PC20)	
		STATION	**TRUNK**
SELF OFFICE (PC10)	**STA-TION**	<PC10> ○ APDCL/APDCN Self Office Connection Device : Select a type of Station. Remote Office PC : PC20 Remote Office Equipment Kind : Station T_PAD : 0dB R_PAD : 12db	<PC10> ○ APDCL/APDCN Self Office Connection Device : Select a type of Station. Remote Office PC : PC20 Remote Office Equipment Kind : Select a type of Trunk. T_PAD : 0dB R_PAD : 12dB
		<PC20> ○ APDCL/APDCN Self Office Connection Device : Select a type of Station. Remote Office PC : PC10 Remote Office Equipment Kind : Station T_PAD : 0dB R_PAD : 12dB	<PC20> ○ APDCL/APDCN Self Office Connection Device : Select a type of trunk. Remote Office PC : PC10 Remote Office Equipment Kind : Station. T_PAD : 0dB R_PAD : 12dB

		COUNTER OFFICE (PC20)	
		STATION	**TRUNK**
SELF OFFICE (PC10)	**TRUNK**	<PC10> ○ APDCL/APDCN Self Office Connection Device : Select a type of trunk. Remote Office PC : PC20 Remote Office Equipment Kind : Station T_PAD : 0dB R_PAD : 12dB	<PC10> ○ APDCL/APDCN Self Office Connection Device : Select a type of trunk. Remote Office PC : PC20 Remote Office Equipment Kind : Select a type of trunk. T_PAD : 0dB R_PAD : 12dB
		<PC20> ○ APDCL/APDCN Self Office Connection Device : Select a type of Station. Remote Office PC : PC10 Remote Office Equipment Kind : Select a type of trunk. T_PAD : 0dB R_PAD : 12dB	<PC20> ○ APDCL/APDCN Self Office Connection Device : Select a type of trunk. Remote Office PC : PC10 Remote Office Equipment Kind : Select a type of trunk. T_PAD : 0dB R_PAD : 12dB

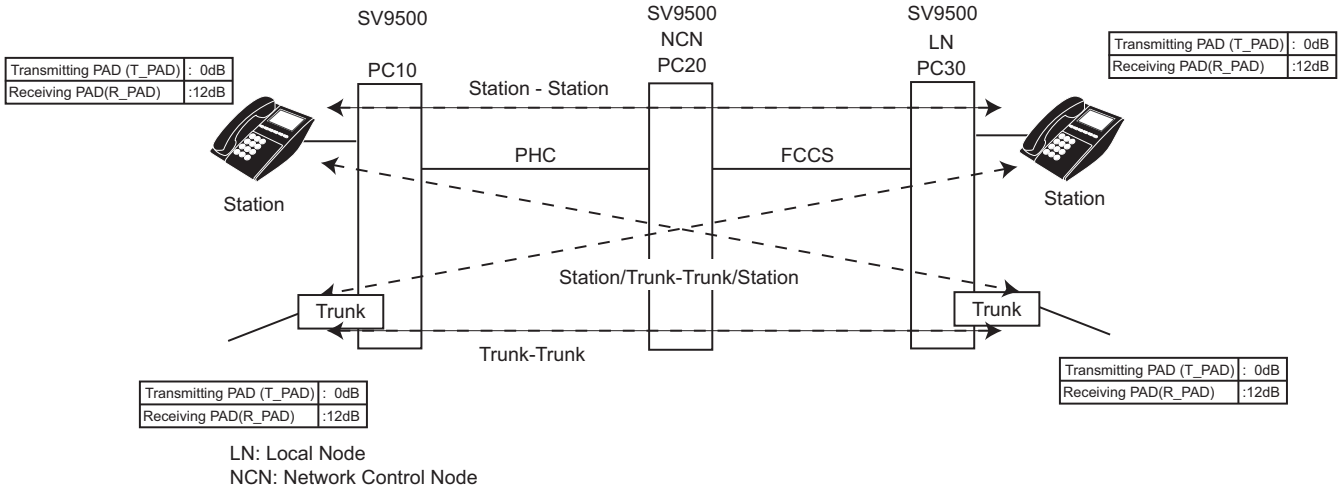
- Standalone to Standalone (NEAX2400 IPX (Feature Package 8800 (R15) or earlier)/SV9300/SV8300/ NEAX2000 IPS)



		COUNTER OFFICE (PC20)
		STATION/TRUNK
SELF OFFICE (PC10)	STA-TION	<PC10> ○ APDCL/APDCN Self Office Connection Device : Select a type of Station. Remote Office PC : PC20 Remote Office Equipment Kind : No Send Terminal T_PAD : 0dB R_PAD : 12dB
		<PC20> ○ APDCL/APDCN In the case of SV9300/SV8300/NEAX2000 IPS, refer to the appropriate manual. In the case of NEAX2400 IPX (Feature Package 8800 (R15) or earlier), settings by ASYDL, SYS1, Index 812/Index 813 or ACRD are applied.

		COUNTER OFFICE (PC20)
		STATION/TRUNK
SELF OFFICE (PC10)	TRUNK	<PC10> ○ APDCL/APDCN Self Office Connection Device : Select a type of trunk. Remote Office PC : PC20 Remote Office Equipment Kind : No Send Terminal T_PAD : 0dB R_PAD : 12dB
		<PC20> ○ APDCL/APDCN In the case of SV9300/SV8300/NEAX2000 IPS, refer to the appropriate manual. In the case of NEAX2400 IPX (Feature Package 8800 (R15) or earlier), settings by ASYDL, SYS1, Index 812/Index 813 or ACRD are applied.

- Standalone to FCCS

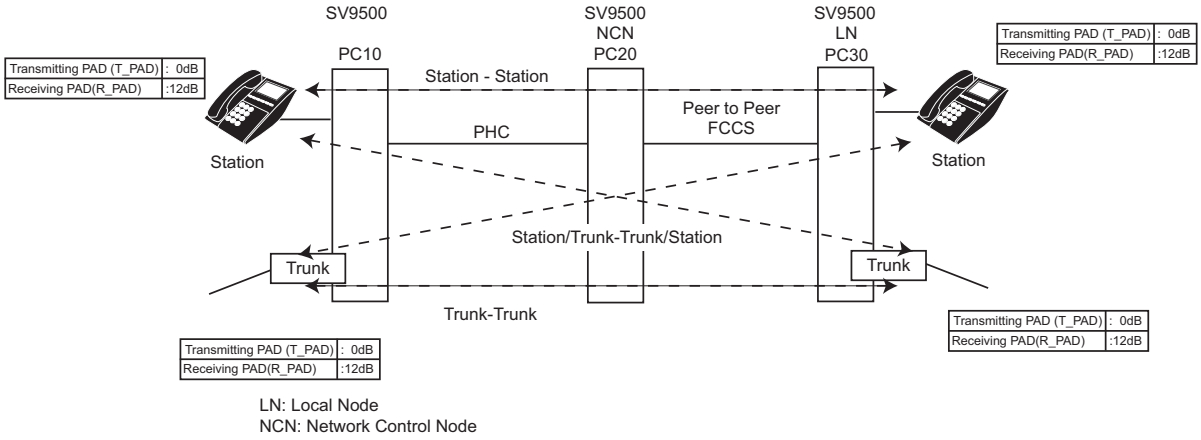


		COUNTER OFFICE (PC30)	
		STATION	**TRUNK**
SELF OFFICE (PC10)	**STA-TION**	<PC10> ○ APDCL/APDCN Self Office Connection Device : Select a type of Station. Remote Office PC : PC20 Remote Office Equipment Kind : Station T_PAD : 0dB R_PAD : 12dB	<PC10> ○ APDCL/APDCN Self Office Connection Device : Select a type of Station. Remote Office PC : PC20 Remote Office Equipment Kind : Select a type of trunk. T_PAD : 0dB R_PAD : 12dB
		<PC20(NCN)> ○ APDCL/APDCN Self Office Connection Device : Select a type of Station. Remote Office PC : PC10 Remote Office Equipment Kind : Station T_PAD : 0dB R_PAD : 12dB	<PC20(NCN)> ○ APDCL/APDCN Self Office Connection Device : FCCS Remote Office PC : PC10 Remote Office Equipment Kind : Station T_PAD : 0dB R_PAD : 12dB
		<PC30(LN)> None.	<PC30(LN)> None.

		COUNTER OFFICE (PC30)	
		STATION	**TRUNK**
SELF OFFICE (PC10)	**TRUNK**	<PC10> ○ APDCL/APDCN Self Office Connection Device : Select a type of trunk. Remote Office PC : PC20 Remote Office Equipment Kind : Station T_PAD : 0dB R_PAD : 12dB	<PC10> ○ APDCL/APDCN Self Office Connection Device : Select a type of trunk. Remote Office PC : PC20 Remote Office Equipment Kind : Select a type of trunk. T_PAD : 0dB R_PAD : 12dB
		<PC20(NCN)> ○ APDCL/APDCN Self Office Connection Device : Select a type of Station. Remote Office PC : PC10 Remote Office Equipment Kind : Select a type of trunk. T_PAD : 0dB R_PAD : 12dB	<PC20(NCN)> ○ APDCL/APDCN Self Office Connection Device : FCCS Remote Office PC : PC10 Remote Office Equipment Kind : Select a type of trunk. T_PAD : 0dB R_PAD : 12dB
		<PC30(LN)> None.	<PC30(LN)> None.

- Standalone to Peer-to-Peer FCCS

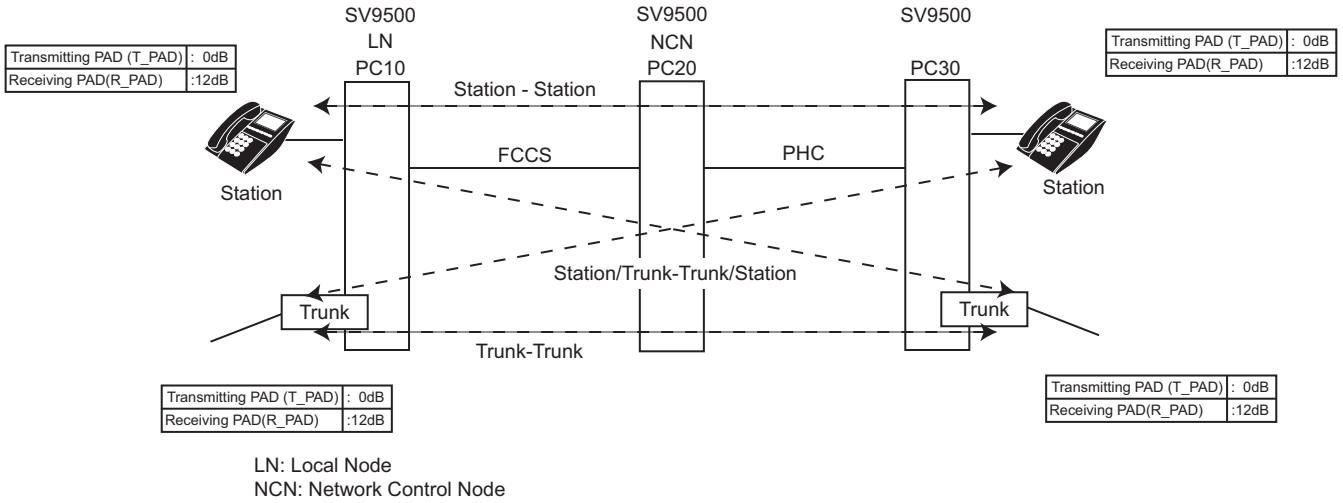
Note: In Peer-to-Peer FCCS, if you assign data by APDCL command in a node, all the nodes in the network require the setting. If you use APDCN command, data assignment is required for NCN only.



		COUNTER OFFICE (PC30)	
		STATION	**TRUNK**
SELF OFFICE (PC10)	**STATION**	<PC10> ○ APDCL/APDCN Self Office Connection Device : Select a type of Station. Remote Office PC : PC20 Remote Office Equipment Kind : Station T_PAD : 0dB R_PAD : 12dB	<PC10> ○ APDCL/APDCN Self Office Connection Device : Select a type of Station. Remote Office PC : PC20 Remote Office Equipment Kind : Select a type of trunk. T_PAD : 0dB R_PAD : 12dB
		<PC20(NCN)> ○ APDCL/APDCN Self Office Connection Device : Select a type of Station. Remote Office PC : PC10 Remote Office Equipment Kind : Select a type of trunk. T_PAD : 0dB R_PAD : 12dB	<PC20(NCN)> ○ APDCL/APDCN Self Office Connection Device : FCCS Remote Office PC : PC10 Remote Office Equipment Kind : Station T_PAD : 0dB R_PAD : 12dB
		<PC30(LN)> If PC20 is assigned by APDCL command, PC30 requires the same setting with PC20.	<PC30(LN)> If PC20 is assigned by APDCL command, PC30 requires the same setting with PC20.

		COUNTER OFFICE (PC30)	
		STATION	**TRUNK**
SELF OFFICE (PC10)	**TRUNK**	<PC10> ○ APDCL/APDCN Self Office Connection Device : Select a type of trunk. Remote Office PC : PC20 Remote Office Equipment Kind : Station T_PAD : 0dB R_PAD : 12dB	<PC10> ○ APDCL/APDCN Self Office Connection Device : Select a type of trunk. Remote Office PC : PC20 Remote Office Equipment Kind : Select a type of trunk. T_PAD : 0dB R_PAD : 12dB
		<PC20(NCN)> ○ APDCL/APDCN Self Office Connection Device : Select a type of Station. Remote Office PC : PC10 Remote Office Equipment Kind : Select a type of trunk. T_PAD : 0dB R_PAD : 12dB	<PC20(NCN)> ○ APDCL/APDCN Self Office Connection Device : Select a type of trunk. Remote Office PC : PC10 Remote Office Equipment Kind : Select a type of trunk. T_PAD : 0dB R_PAD : 12dB
		<PC30(LN)> If PC20 is assigned by APDCL command, PC30 requires the same setting with PC20.	<PC30(LN)> If PC20 is assigned by APDCL command, PC30 requires the same setting with PC20.

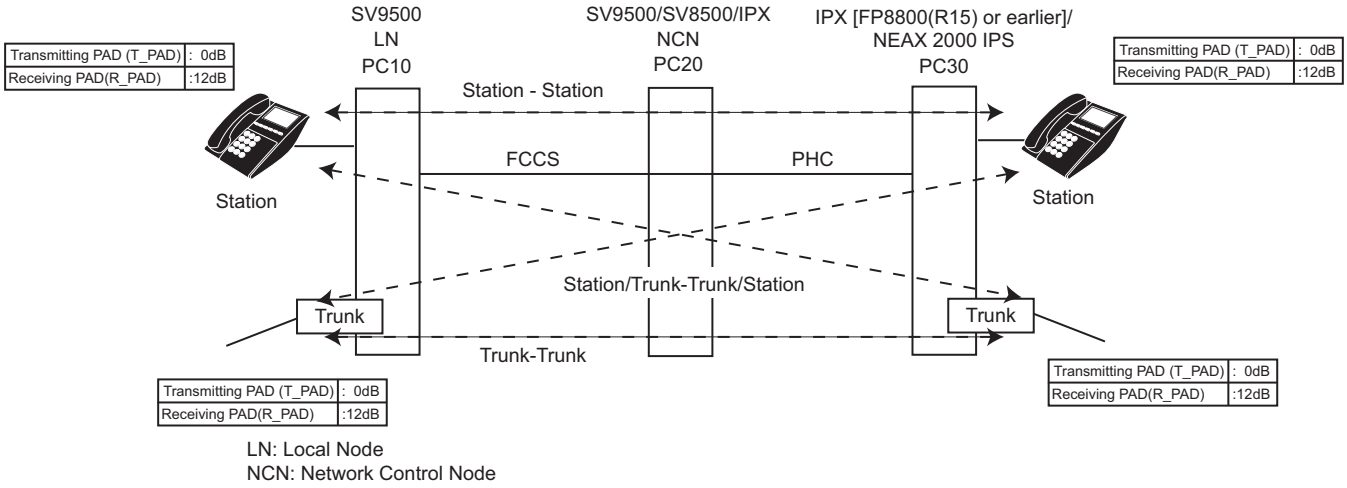
- FCCS to Standalone



		COUNTER OFFICE (PC30)	
		STATION	**TRUNK**
SELF OFFICE (PC10)	**STA-TION**	<PC10(LN)> None.	<PC10(LN)> None.
		<PC20(NCN)> ○ APDCL/APDCN Self Office Connection Device : Select a type of Station. Remote Office PC : PC30 Remote Office Equipment Kind : Station T_PAD : 0dB R_PAD : 12dB	<PC20(NCN)> ○ APDCL/APDCN Self Office Connection Device : FCCS Remote Office PC : PC30 Remote Office Equipment Kind : Select a type of trunk. T_PAD : 0dB R_PAD : 12dB
		<PC30> ○ APDCL/APDCN Self Office Connection Device : Select a type of Station. Remote Office PC : PC20 Remote Office Equipment Kind : Station T_PAD : 0dB R_PAD : 12dB	<PC30> ○ APDCL/APDCN Self Office Connection Device : Select a type of trunk. Remote Office PC : PC20 Remote Office Equipment Kind : Station T_PAD : 0dB R_PAD : 12dB

		COUNTER OFFICE (PC30)	
		STATION	**TRUNK**
SELF OFFICE (PC10)	**TRUNK**	<PC10(LN)> None.	<PC10(LN)> None.
		<PC20(NCN)> ○ APDCL/APDCN Self Office Connection Device : Select a type of Station. Remote Office PC : PC30 Remote Office Equipment Kind : Station T_PAD : 0dB R_PAD : 12dB	<PC20(NCN)> ○ APDCL/APDCN Self Office Connection Device : FCCS Remote Office PC : PC30 Remote Office Equipment Kind : Select a type of trunk. T_PAD : 0dB R_PAD : 12dB
		<PC30> ○ APDCL/APDCN Self Office Connection Device : Select a type of Station. Remote Office PC : PC20 Remote Office Equipment Kind : Select a type of trunk. T_PAD : 0dB R_PAD : 12dB	<PC30> ○ APDCL/APDCN Self Office Connection Device : Select a type of trunk. Remote Office PC : PC20 Remote Office Equipment Kind : Select a type of trunk. T_PAD : 0dB R_PAD : 12dB

- FCCS to Standalone (NEAX2400 IPX (Feature Package 8800 (R15) or earlier)/SV9300/SV8300/ NEAX2000 IPS)

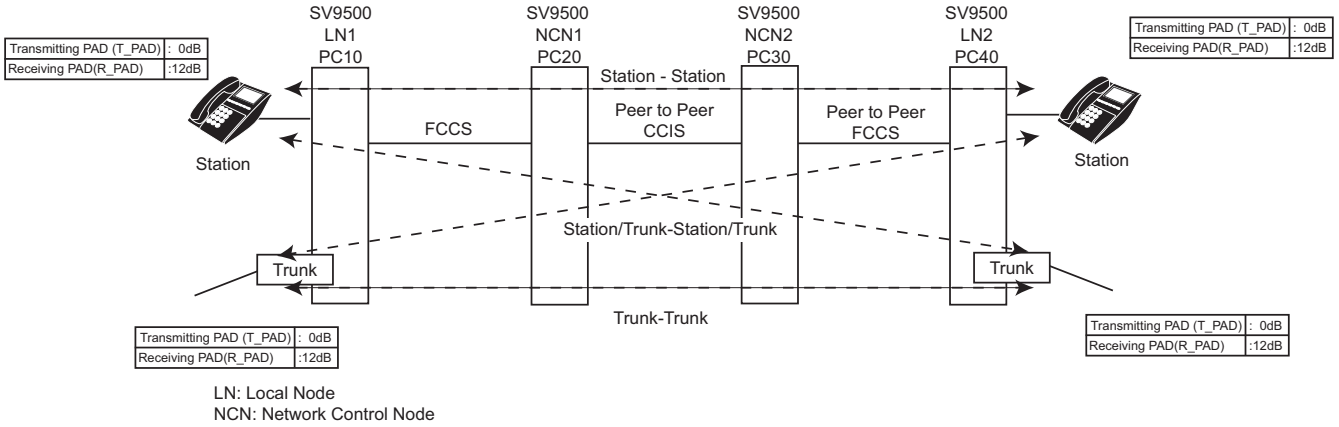


		COUNTER OFFICE (PC30)
		STATION/TRUNK
SELF OFFICE (PC10)	**STA-TION**	<PC10(LN)> None.
		<PC20(NCN)> ○ APDCL/APDCN Self Office Connection Device : FCCS Remote Office PC : PC30 Remote Office Equipment Kind : No Send Terminal T_PAD : 0dB R_PAD : 12dB
		<PC30> In the case of SV9300/SV8300/NEAX2000 IPS, refer to the appropriate manual. In the case of NEAX2400 IPX (Feature Package 8800 (R15) or earlier), settings by ASYDL, SYS1, Index 812/Index 813 or ACRD are applied.

		COUNTER OFFICE (PC30)
		STATION/TRUNK
SELF OFFICE (PC10)	TRUNK	<PC10(LN)> None.
		<PC20(NCN)> ○ APDCL/APDCN Self Office Connection Device : FCCS Remote Office PC : PC30 Remote Office Equipment Kind : No Send Terminal T_PAD : 0dB R_PAD : 12dB
		<PC30> In the case of SV9300/SV8300/NEAX2000 IPS, refer to the appropriate manual. In the case of NEAX2400 IPX (Feature Package 8800 (R15) or earlier), settings by ASYDL, SYS1, Index 812/Index 813 or ACRD are applied.

- FCCS to Peer-to-Peer FCCS

Note: In Peer-to-Peer FCCS, if you assign data by APDCL command in a node, all the nodes in the network require the setting. If you use APDCN command, data assignment is required for NCN only.



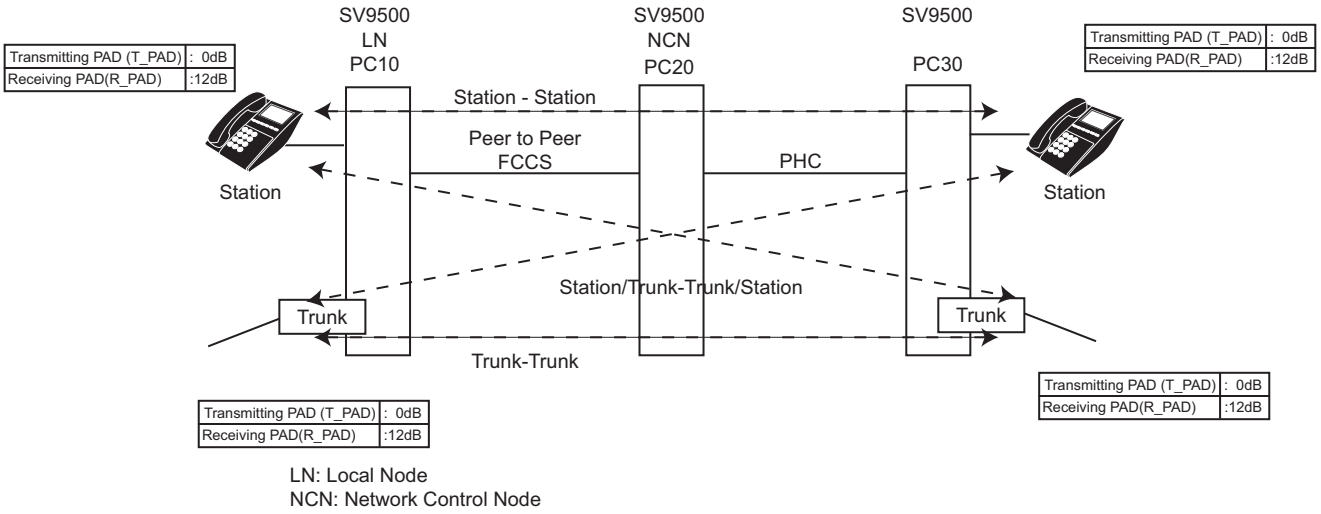
LN: Local Node
NCN: Network Control Node

		COUNTER OFFICE (PC40)	
		STATION	**TRUNK**
SELF OFFICE (PC10)	**STA-TION**	<PC10(LN1)> None.	<PC10(LN1)> None.
		<PC20(NCN1)> ○ APDCL/APDCN Self Office Connection Device : FCCS Remote Office PC : PC30 Remote Office Equipment Kind : Station T_PAD : 0dB R_PAD : 12dB	<PC20(NCN1)> ○ APDCL/APDCN Self Office Connection Device : FCCS Remote Office PC : PC30 Remote Office Equipment Kind : Select a type of trunk. T_PAD : 0dB R_PAD : 12dB
		<PC30(NCN2)> ○ APDCL/APDCN Self Office Connection Device : Select a type of Station. Remote Office PC : PC20 Remote Office Equipment Kind : Station T_PAD : 0dB R_PAD : 12dB	<PC30(NCN2)> ○ APDCL/APDCN Self Office Connection Device : Select a type of trunk. Remote Office PC : PC20 Remote Office Equipment Kind : Station T_PAD : 0dB R_PAD : 12dB
		<PC40(LN2)> If PC30 is assigned by APDCL command, PC40 requires the same setting with PC30.	<PC40(LN2)> If PC30 is assigned by APDCL command, PC40 requires the same setting with PC30.

		COUNTER OFFICE (PC40)	
		STATION	**TRUNK**
SELF OFFICE (PC10)	**TRUNK**	<PC10(LN1)> None.	<PC10(LN1)> None.
		<PC20(NCN1)> ○ APDCL/APDCN Self Office Connection Device : FCCS Remote Office PC : PC30 Remote Office Equipment Kind : Station T_PAD : 0dB R_PAD : 12dB	<PC20(NCN1)> ○ APDCL/APDCN Self Office Connection Device : FCCS Remote Office PC : PC30 Remote Office Equipment Kind : Select a type of trunk. T_PAD : 0dB R_PAD : 12dB
		<PC30(NCN2)> ○ APDCL/APDCN Self Office Connection Device : Select a type of Station. Remote Office PC : PC20 Remote Office Equipment Kind : Station T_PAD : 0dB R_PAD : 12dB	<PC30(NCN2)> ○ APDCL/APDCN Self Office Connection Device : Select a type of trunk. Remote Office PC : PC20 Remote Office Equipment Kind : Select a type of trunk. T_PAD : 0dB R_PAD : 12dB
		<PC40(LN2)> If PC30 is assigned by APDCL command, PC40 requires the same setting with PC30.	<PC40(LN2)> If PC30 is assigned by APDCL command, PC40 requires the same setting with PC30.

- Peer-to-Peer FCCS to Standalone

Note: In Peer-to-Peer FCCS, if you assign data by APDCL command in a node, all the nodes in the network require the setting. If you use APDCN command, data assignment is required for NCN only.



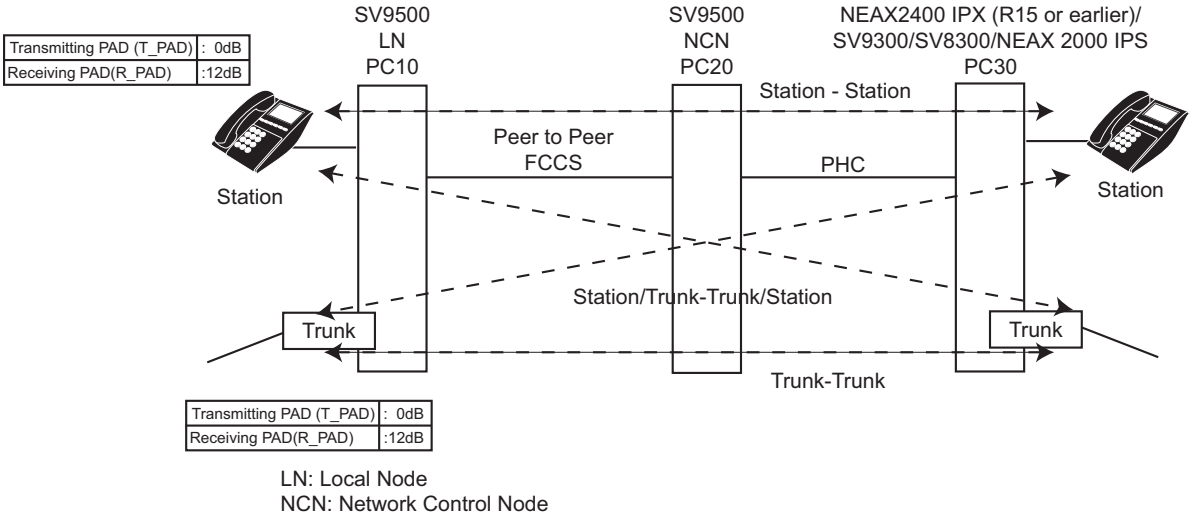
LN: Local Node
NCN: Network Control Node

		COUNTER OFFICE (PC30)	
		STATION	**TRUNK**
SELF OFFICE (PC10)	**STA-TION**	<PC10(LN)> If PC20 is assigned by APDCL command, PC10 requires the same setting with PC20.	<PC10(LN)> If PC20 is assigned by APDCL command, PC10 requires the same setting with PC20.
		<PC20(NCN)> ○ APDCL/APDCN Self Office Connection Device : Select a type of Station. Remote Office PC : PC30 Remote Office Equipment Kind : Station T_PAD : 0dB R_PAD : 12dB	<PC20(NCN)> ○ APDCL/APDCN Self Office Connection Device : Select a type of Station. Remote Office PC : PC30 Remote Office Equipment Kind : Select a type of trunk. T_PAD : 0dB R_PAD : 12dB
		<PC30> ○ APDCL/APDCN Self Office Connection Device : Select a type of Station. Remote Office PC : PC20 Remote Office Equipment Kind : Station T_PAD : 0dB R_PAD : 12dB	<PC30> ○ APDCL/APDCN Self Office Connection Device : Select a type of trunk. Remote Office PC : PC20 Remote Office Equipment Kind : Station T_PAD : 0dB R_PAD : 12dB

		COUNTER OFFICE (PC30)	
		STATION	**TRUNK**
SELF OFFICE (PC10)	**TRUNK**	<PC10(LN)> If PC20 is assigned by APDCL command, PC10 requires the same setting with PC20.	<PC10(LN)> If PC20 is assigned by APDCL command, PC10 requires the same setting with PC20.
		<PC20(NCN)> ○ APDCL/APDCN Self Office Connection Device : Select a type of trunk. Remote Office PC : PC30 Remote Office Equipment Kind : Station T_PAD : 0dB R_PAD : 12dB	<PC20(NCN)> ○ APDCL/APDCN Self Office Connection Device : Select a type of trunk. Remote Office PC : PC30 Remote Office Equipment Kind : Select a type of trunk. T_PAD : 0dB R_PAD : 12dB
		<PC30> ○ APDCL/APDCN Self Office Connection Device : Select a type of Station. Remote Office PC : PC20 Remote Office Equipment Kind : Select a type of trunk. T_PAD : 0dB R_PAD : 12dB	<PC30> ○ APDCL/APDCN Self Office Connection Device :Select a type of trunk. Remote Office PC : PC20 Remote Office Equipment Kind : Select a type of trunk. T_PAD : 0dB R_PAD : 12dB

- Peer-to-Peer FCCS to Standalone (NEAX2400 IPX (Feature Package 8800 (R15) or earlier/SV9300/SV8300/NEAX2000 IPS)

Note: In Peer-to-Peer FCCS, if you assign data by APDCL command in a node, all the nodes in the network require the setting. If you use APDCN command, data assignment is required for NCN only.

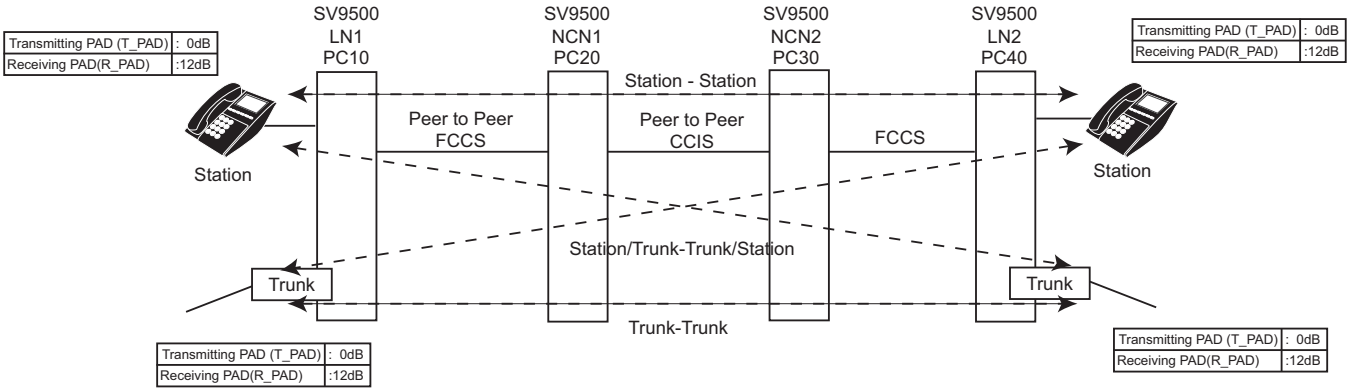


		COUNTER OFFICE (PC30)
		STATION/TRUNK
SELF OFFICE (PC10)	STATION	<PC10(LN)> If PC20 is assigned by APDCL command, PC10 requires the same setting with PC20.
		<PC20(NCN)> ○ APDCL/APDCN Self Office Connection Device : Select a type of Station. Remote Office PC : PC30 Remote Office Equipment Kind : No Send Terminal T_PAD : 0dB R_PAD : 12dB
		<PC30> In the case of SV9300/SV8300/NEAX2000 IPS, refer to the appropriate manual. In the case of NEAX2400 IPX (Feature Package 8800 (R15) or earlier), settings by ASYDL, SYS1, Index 812/Index 813 or ACRD are applied.

		COUNTER OFFICE (PC30)
		STATION/TRUNK
SELF OFFICE (PC10)	TRUNK	<PC10(LN)> If PC20 is assigned by APDCL command, PC10 requires the same setting with PC20.
		<PC20(NCN)> ○ APDCL/APDCN Self Office Connection Device : Select a type of trunk. Remote Office PC : PC30 Remote Office Equipment Kind : No Send Terminal T_PAD : 0dB R_PAD : 12dB
		<PC30> In the case of SV9300/SV8300/NEAX2000 IPS, refer to the appropriate manual. In the case of NEAX2400 IPX (Feature Package 8800 (R15) or earlier), settings by ASYDL, SYS1, Index 812/Index 813 or ACRD are applied.

- Peer-to-Peer FCCS to FCCS

Note: In Peer-to-Peer FCCS, if you assign data by APDCL command in a node, all the nodes in the network require the setting. If you use APDCN command, data assignment is required for NCN only.

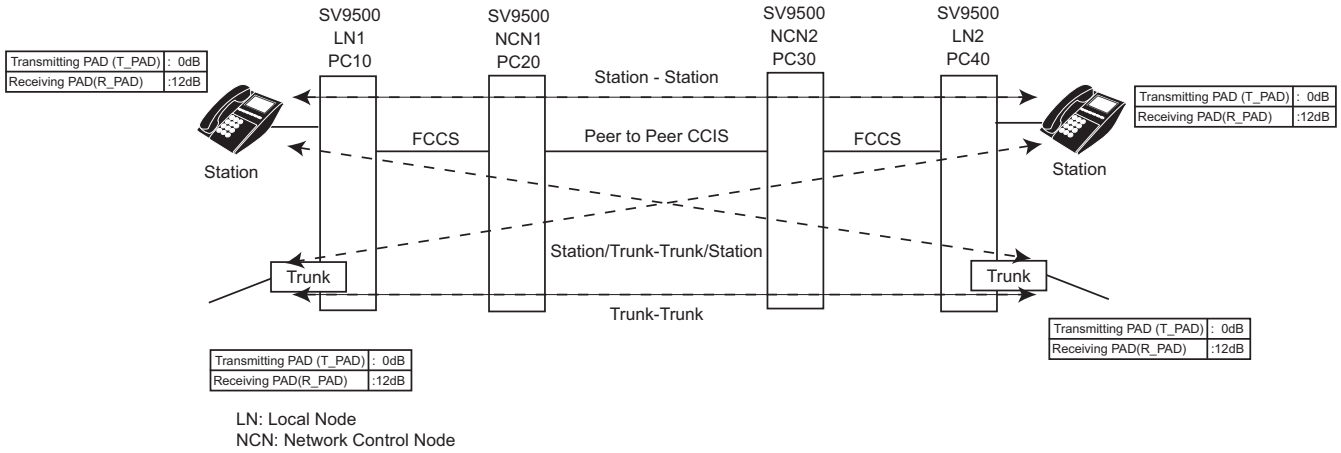


LN: Local Node
NCN: Network Control Node

		COUNTER OFFICE (PC40)	
		STATION	**TRUNK**
SELF OFFICE (PC10)	**STA-TION**	<PC10(LN1)> If PC20 is assigned by APDCL command, PC10 requires the same setting with PC30.	<PC10(LN1)> If PC20 is assigned by APDCL command, PC10 requires the same setting with PC30.
		<PC20(NCN1)> ○ APDCL/APDCN Self Office Connection Device : Select a type of Station. Remote Office PC : PC30 Remote Office Equipment Kind : Station T_PAD : 0dB R_PAD : 12dB	<PC20(NCN1)> ○ APDCL/APDCN Self Office Connection Device : Select a type of Station. Remote Office PC : PC30 Remote Office Equipment Kind : Select a type of trunk. T_PAD : 0dB R_PAD : 12dB
		<PC30(NCN2)> ○ APDCL/APDCN Self Office Connection Device : FCCS Remote Office PC : PC20 Remote Office Equipment Kind : Station T_PAD : 0dB R_PAD : 12dB	<PC30(NCN2)> ○ APDCL/APDCN Self Office Connection Device : FCCS Remote Office PC : PC20 Remote Office Equipment Kind : Station T_PAD : 0dB R_PAD : 12dB
		<PC40(LN2)> None.	<PC40(LN2)> None.

		COUNTER OFFICE (PC40)	
		STATION	**TRUNK**
SELF OFFICE (PC10)	**TRUNK**	<PC10(LN1)> If PC20 is assigned by APDCL command, PC10 requires the same setting with PC20.	<PC10(LN1)> If PC20 is assigned by APDCL command, PC10 requires the same setting with PC20.
		<PC20(NCN1)> ○ APDCL/APDCN Self Office Connection Device : Select a type of trunk. Remote Office PC : PC20 Remote Office Equipment Kind : Station T_PAD : 0dB R_PAD : 12dB	<PC20(NCN1)> ○ APDCL/APDCN Self Office Connection Device : Select a type of trunk. Remote Office PC : PC20 Remote Office Equipment Kind : Select a type of trunk. T_PAD : 0dB R_PAD : 12dB
		<PC30(NCN2)> ○ APDCL/APDCN Self Office Connection Device : FCCS Remote Office PC : PC20 Remote Office Equipment Kind : Select a type of trunk. T_PAD : 0dB R_PAD : 12dB	<PC30(NCN2)> ○ APDCL/APDCN Self Office Connection Device : FCCS Remote Office PC : PC30 Remote Office Equipment Kind : Select a type of trunk. T_PAD : 0dB R_PAD : 12dB
		<PC40(LN2)> None.	<PC40(LN2)> None.

- FCCS to FCCS

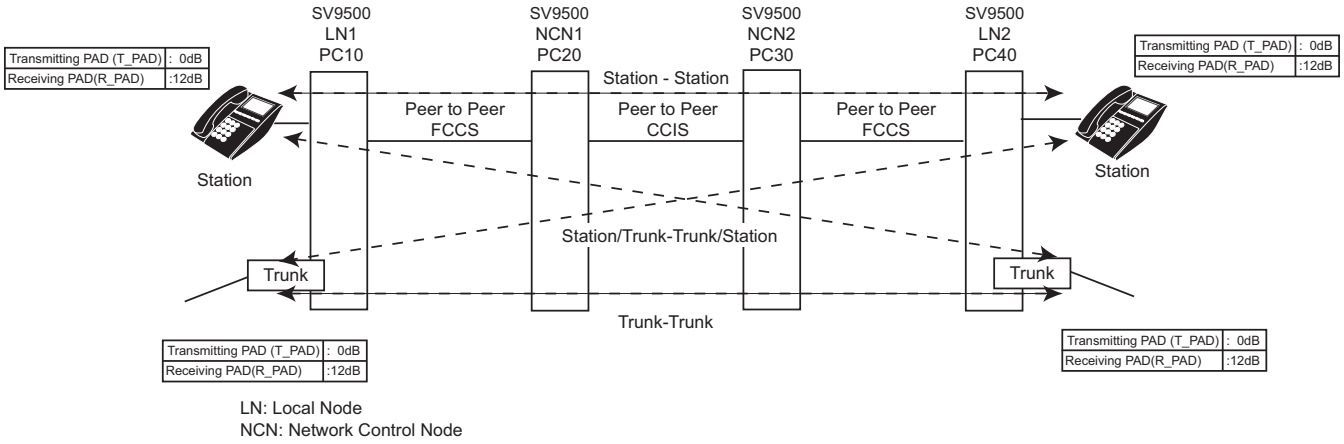


		COUNTER OFFICE (PC40)	
		STATION	**TRUNK**
SELF OFFICE (PC10)	**STA-TION**	<PC10(LN1)> None.	<PC10(LN1)> None.
		<PC20(NCN1)> ○ APDCL/APDCN Self Office Connection Device : FCCS Remote Office PC : PC30 Remote Office Equipment Kind : Station T_PAD : 0dB R_PAD : 12dB	<PC20(NCN1)> ○ APDCL/APDCN Self Office Connection Device : FCCS Remote Office PC : PC30 Remote Office Equipment Kind : Select a type of trunk. T_PAD : 0dB R_PAD : 12dB
		<PC30(NCN2)> ○ APDCL/APDCN Self Office Connection Device : FCCS Remote Office PC : PC20 Remote Office Equipment Kind : Station T_PAD : 0dB R_PAD : 12dB	<PC30(NCN2)> ○ APDCL/APDCN Self Office Connection Device : FCCS Remote Office PC : PC20 Remote Office Equipment Kind : Station T_PAD : 0dB R_PAD : 12dB
		<PC40(LN2)> None.	<PC40(LN2)> None.

		COUNTER OFFICE (PC40)	
		STATION	**TRUNK**
SELF OFFICE (PC10)	**TRUNK**	<PC10(LN1)> None.	<PC10(LN1)> None.
		<PC20(NCN1)> ○ APDCL/APDCN Self Office Connection Device : FCCS Remote Office PC : PC30 Remote Office Equipment Kind : Station T_PAD : 0dB R_PAD : 12dB	<PC20(NCN1)> ○ APDCL/APDCN Self Office Connection Device : FCCS Remote Office PC : PC30 Remote Office Equipment Kind : Select a type of trunk. T_PAD : 0dB R_PAD : 12dB
		<PC30(NCN2)> ○ APDCL/APDCN Self Office Connection Device : FCCS Remote Office PC : PC20 Remote Office Equipment Kind : Select a type of trunk. T_PAD : 0dB R_PAD : 12dB	<PC30(NCN2)> ○ APDCL/APDCN Self Office Connection Device : FCCS Remote Office PC : PC20 Remote Office Equipment Kind : Select a type of trunk. T_PAD : 0dB R_PAD : 12dB
		<PC40(LN2)> None.	<PC40(LN2)> None.

- Peer-to-Peer FCCS to Peer-to-Peer FCCS

Note: In Peer-to-Peer FCCS, if you assign data by APDCL command in a node, all the nodes in the network require the setting. If you use APDCN command, data assignment is required for NCN only.



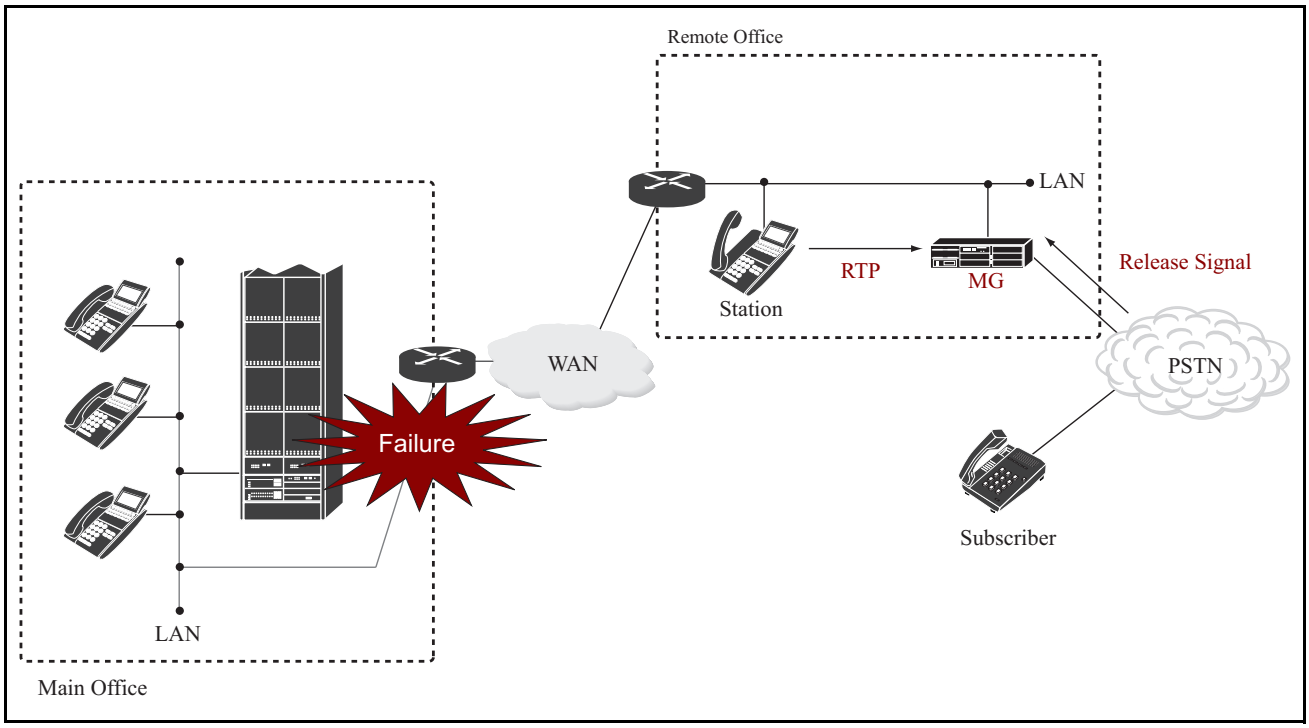
LN: Local Node
NCN: Network Control Node

		COUNTER OFFICE (PC40)	
		STATION	**TRUNK**
SELF OFFICE (PC10)	**STA-TION**	<PC10(LN1)> If PC20 is assigned by APDCL command, PC10 requires the same setting with PC20.	<PC10(LN1)> If PC20 is assigned by APDCL command, PC10 requires the same setting with PC20.
		<PC20(NCN1)> ○ APDCL/APDCN Self Office Connection Device : Select a type of Station. Remote Office PC : PC30 Remote Office Equipment Kind : Station T_PAD : 0dB R_PAD : 12dB	<PC20(NCN1)> ○ APDCL/APDCN Self Office Connection Device : Select a type of Station. Remote Office PC : PC30 Remote Office Equipment Kind : Select a type of trunk. T_PAD : 0dB R_PAD : 12dB
		<PC30(NCN2)> ○ APDCL/APDCN Self Office Connection Device : Select a type of Station. Remote Office PC : PC20 Remote Office Equipment Kind : Station T_PAD : 0dB R_PAD : 12dB	<PC30(NCN2)> ○ APDCL/APDCN Self Office Connection Device : Select a type of trunk. Remote Office PC : PC20 Remote Office Equipment Kind : Station T_PAD : 0dB R_PAD : 12dB
		<PC40(LN2)> If PC30 is assigned by APDCL command, PC40 requires the same setting with PC30.	<PC40(LN2)> If PC30 is assigned by APDCL command, PC40 requires the same setting with PC30.

		COUNTER OFFICE (PC40)	
		STATION	**TRUNK**
SELF OFFICE (PC10)	**TRUNK**	<PC10(LN1)> If PC20 is assigned by APDCL command, PC10 requires the same setting with PC20.	<PC10(LN1)> If PC20 is assigned by APDCL command, PC10 requires the same setting with PC20.
		<PC20(NCN1)> ○ APDCL/APDCN Self Office Connection Device : Select a type of trunk. Remote Office PC : PC30 Remote Office Equipment Kind : Station T_PAD : 0dB R_PAD : 12dB	<PC20(NCN1)> ○ APDCL/APDCN Self Office Connection Device : Select a type of trunk. Remote Office PC : PC30 Remote Office Equipment Kind : Select a type of trunk. T_PAD : 0dB R_PAD : 12dB
		<PC30(NCN2)> ○ APDCL/APDCN Self Office Connection Device : Select a type of Station. Remote Office PC : PC20 Remote Office Equipment Kind : Select a type of trunk. T_PAD : 0dB R_PAD : 12dB	<PC30(NCN2)> ○ APDCL/APDCN Self Office Connection Device : Select a type of trunk. Remote Office PC : PC20 Remote Office Equipment Kind : Select a type of trunk. T_PAD : 0dB R_PAD : 12dB
		<PC40(LN2)> If PC30 is assigned by APDCL command, PC40 requires the same setting with PC30.	<PC40(LN2)> If PC30 is assigned by APDCL command, PC40 requires the same setting with PC30.

14.9 MG Based Call Retention

When the station is talking to the subscriber through SIP/analog/ISDN network via MG, if a restart message is generated by the Telephony Server or health check session has timed out, MG can retain the ongoing call without the Telephony Server. The ongoing call can be released by placing the handset on hook by the station or subscriber.



Service Conditions

1. Common conditions for each MG

a.) MG equipment and firmware version available for this feature is as follows.

IP Terminal/Equipment		Firmware Version
MC&MG-COT	MCMG Box [MG-4LC2COTA]	SP-3927 MCMG PROG-B Issue 1 or later
	MCMG Card [SCA-4LC2COTA]	
	UG50 (COT-TYPE1)	No conditions
	UG50-4LC2COTA (COT)	
	UG50 (PGT-TYPE1)	Software Version 2.00 or later
	UG50-4LC2COTA (PGT)	
MG(BRI)	MG (BRI) Box [MG-2BRIA]	SP-3826 BRI MG SIP BSC PROG-A Issue 4 or later
	MG (BRI) Card [SCA-2BRIA]	
MG(PRI) (Proprietary Protocol/SIP)	MG (PRI) Box [MG-24PRIA]	SP-3884 MG PRI PROG-A Issue 8 or later
	MG (PRI) Card [SCA-24PRIA]	
	MG (PRI) Box [MG-30PRIA]	
	MG (PRI) Card [SCA-30PRIA]	
	UG50 (PRT 1.5M)	No conditions
	UG50 (PRT 2M)	
	UG50-24PRIA	
	UG50-30PRIA	

IP Terminal/Equipment		Firmware Version
MG(SIP)	MG-SIP16 Box [MG-16SIPMGA]	SP-3905 MG SIP (16) PROG-A Issue 5 or later
	MG-SIP16 Card [SCA-16SIPMGA]	SP-3905 MG SIP (16) PROG-A Issue 5 or later
	MG-SIP16 Card [SCA-16SIPMG(US)]	SP-3988 MG SIP(16) PROG-G Issue 3 or later
	MG-SIP96	SP-3987 MGSIP PROG-B Issue 3 or later
	MG-SIP128 [MG-128SIPMGL-A]	SP-4089 MGSIP(MM) PROG-A Issue 1 or later
	MG-SIP128 [MG-128SIPMGG-B/MG-128SIPMGG]	SP-4051 MGSIP PROG-G Issue 1 or later SP-4058 MGSIP PROG-J Issue 1 or later
	MG-SIP128 [MG-128SIPMGJ-B/MG-128SIPMGJ]	SP-4032 MGSIP PROG-E Issue 1 or later SP-4060 MGSIP PROG-L Issue 1 or later
	Software-based MG-SIP	SP-4078 MGSIPVM PROG-B Issue 1 or later SP-4080 MGSIPVM PROG-D Issue 1 or later SP-4085 MGSIPVM PROG-E Issue 1 or later

b.) This feature will be used if communication failure to the SV9500 or SR-MGC occurs, and Soft Reset is received from the SV9500 or SR-MGC, both because of a network failure. If a reboot is required by a failure inside the MG, the system will reboot without executing this feature.
c.) A call cannot be retained while ringing or on hold. However, if this feature is activated while hold tone is being sent out, hold tone will continue to be sent out until network failure and call termination are detected.
d.) As soon as the Health Check Error of system message [33-S] (IP PKG Failure) against MG is detected on the SV9500 or SR-MGC because of a network failure, the billing for the call via MG will be stopped. Therefore, the call retained at MG by this feature is not subject to billing on the SV9500 or SR-MGC.
e.) If a failure occurs in the network between the connected party and MG, the call cannot be retained.
f.) The VS32 Conference Server is not supported for this feature.

2. Conditions for each MG

In the case of MC&MG-COT

Note: For UG50 (PGT-TYPE1) and UG50-4LC2COTA (PGT) only the RTP related contents of condition a.) applies.

a.) If any of the following conditions are applied during call retention after a time-out occurs in Health Check, the call will be determined to be released, and the MG will be rebooted.

- When RTP reception stopped while talking:

  The time by which RTP reception is determined stopped can be set by configuration command from 3 seconds (default) to 32 seconds.

- When release signal is detected from C.O.:

  - C.O. Outgoing Call: Polarity Reversal OFF
  - C.O. Incoming Call: Disconnect Signal (Loop Off)

b.) Other equipment such as GW or TA must not be connected between C.O. and MC&MG-COT because the signal from C.O. is monitored except in the case when other equipment have a function such as the disconnect signal transmission and polarity reversal.

c.) If even one channel is being used for a call, a new call origination / termination cannot be made to the rest of channel because MC&MG-COT will not be rebooted. However, PFT line can be used for an emergency telephone until changing to on-line status from [During Call Retention] → [Call Retention released] → [Re-registration after MG reboots].

d.) When this feature is used, Calling Party Release becomes invalid. **Note 1**

- MCMG Box/Card [MG/SCA-4LC2COTA] operates as when Configuration command: SET RELEASE_SIGNAL is set to “OFF”.
- UG50 (COT-TYPE1)/UG50-4LC2COTA (COT) operates as when Web Programing: 4-1-02: COT Basic Setup - RELEASE_SIGNAL is set to “Not checked (Invalid)”.

Note 1: When this release method is used for a C.O. outgoing call, the C.O. is kept in busy status until the station is released. The release signal (polarity reversal) sent from the network side when the opposite party is released will be ignored.

In the case of MG(PRI)/MG(BRI)

a.) If any of the following conditions are applied during call retention after a time-out occurs in Health Check, the call will be determined to be released, and the MG will be rebooted.

- When RTP reception stopped while talking.

  The time by which RTP reception is determined stopped can be set by configuration command of MG from 3 seconds (default) to 32 seconds.

- When DISCONNECT, RELEASE, or RELEASE COMPLETE message of layer 3 is received from opposite node or the SV9500. **Note 2**

  Note 2: The length of call reference value (in octets) for the layer 3 is fixed to 1 byte in the case of MG(BRI), and 2 bytes in the case of MG(PRI).

- When the link between Layer 2 and the opposite node is disconnected.

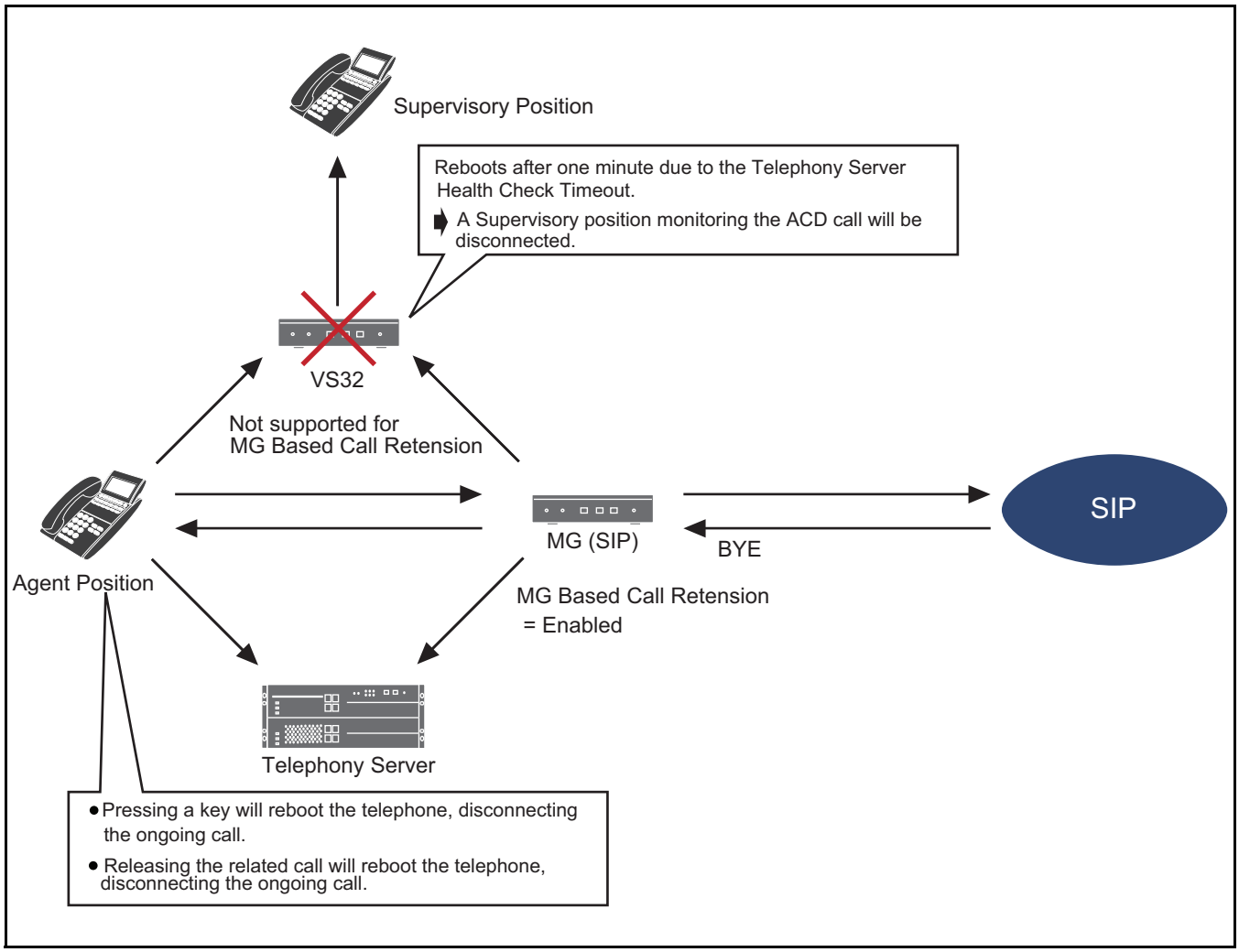
b.) If even one channel is being used for a call when a communication failure occurs between the SV9500 and MG, the MG will not be rebooted because the call will be retained until it is released. Therefore, the MG will not re-register to SR-MGC even if provided, and a new call origination / termination cannot be made to the rest of channels.
c.) In the case of MG(BRI), an emergency telephone cannot be used during call retention.
d.) If a call is assumed to be released during startup of the configuration command, it takes about five minutes to start rebooting MG(BRI)/MG(PRI) after you exit from the configuration command or enter the last command.
e.) When CALL PROTECTED RE-REGISTRATION [C-203] is enabled on MG(PRI) in Proprietary Protocol mode, it overrides this feature. See "Call Protected Re-Registration [C-203]" for further information.
f.) If STATUS ENQUIRY message of layer 3 is received from the opposite node, the call will be disconnected from the network because the status of the call cannot be recognized. For this reason, MG cannot retain the call in a network that uses STATUS ENQUIRY messages.

In the case of MG(SIP)

a.) When this feature is enabled, the warning tone generation on the MG(SIP) (audible alarm for Health Check Timeout configured with the "set hc_alarm") is automatically deactivated. In this case, the setting of the alarm generation feature is not changed.
b.) When a station call on the Telephony Server side is being put on hold or being transferred, the ongoing call is not retained by the activation of this feature.
c.) When RTP packets cannot be received from the Telephony Server while this feature is activated (e.g. When communication between telephones and the Telephony Server is not available, if call hold or call transfer is performed by a telephone user, the telephone detects a communication failure and reboots itself, stopping RTP packets.), the retained call is assumed to be ended.
d.) Billing information about calls retained by this feature will not be provided.

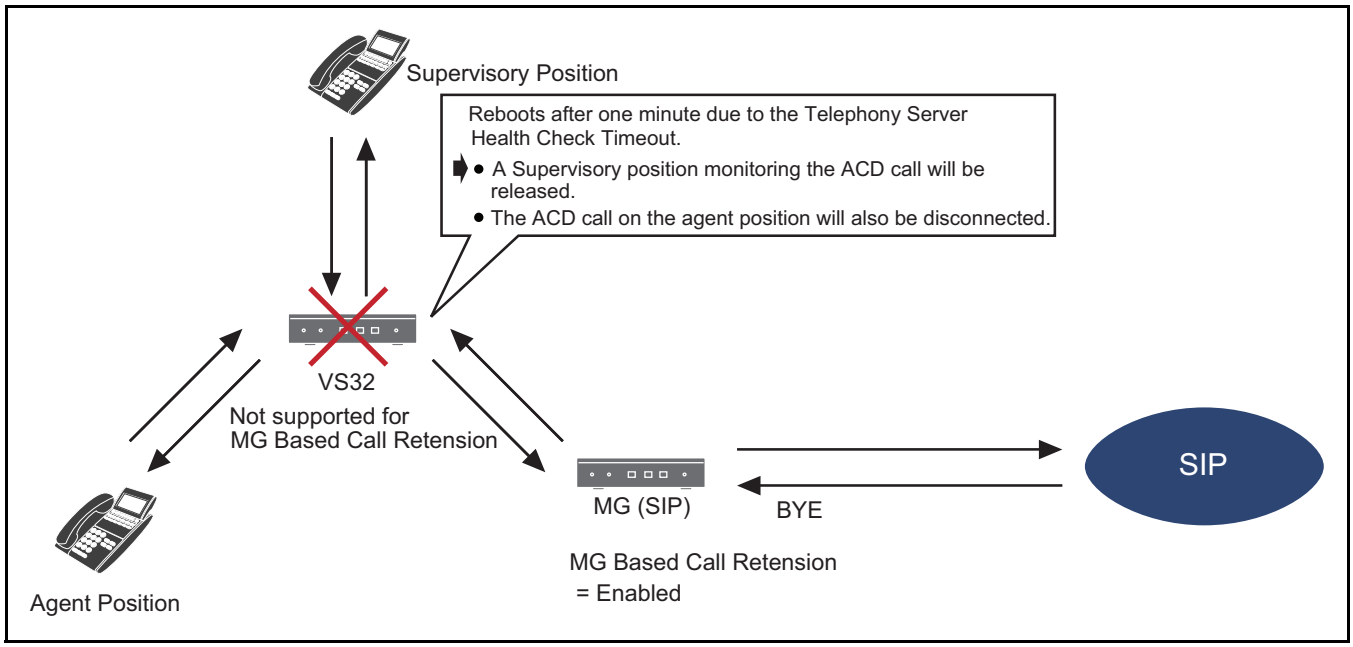
e.) When a call is being monitored by a supervisory position in an ACD system, if a restart message is generated by the Telephony Server or health check session has timed out:

- The ACD call on the agent position is retained by this feature. However, it will be disconnected by a key operation on the agent's telephone or the release of the related call.
- A call on a supervisory position is not retained since the VS32 is not supported for this feature.



f.) When a call is being barged by a supervisory position in an ACD system, if a restart message is generated by the Telephony Server or health check session has timed out;

- the ACD call on the agent position is not retained by this feature.
- a call on a supervisory position is not retained since the VS32 is not supported for this feature.



g.) In the Music On Hold [M-7] feature, the ongoing call is retained by this feature.

h.) When receiving Initial INVITE from the SIP network during call retention, the MG(SIP) returns an error response (default value: 488 Not Acceptable Here). The value of error response can be changed with the MG(SIP) configuration command “set call_hold”.

i.) When the call retention is activated during reception of a new call from a SIP network, the MG(SIP) returns an error response (default value: 488 Not Acceptable Here) to the incoming call, and at the same time, issues a disconnect order to the Telephony Server.The value of error response can be changed with the MG(SIP) configuration command “set call_hold”.

j.) This feature is activated when:

- The MG(SIP) receives Soft Reset from the Telephony Server.
- Health Check Timeout has occurred between the MG(SIP) and the Telephony Server.
- The Telephony Server performs system initialization.

k.) This feature is deactivated when:

- The reboot command is executed on the Configuration Command Setting screen.
- The “RESET” button is pressed on the MG(SIP) main unit.
- The power switch on the MG(SIP) main unit is turned off.

- The MG(SIP) is rebooted by using the MG option for the Reset Type parameter (**Note 3**) of the AMGIL command.

  Note 3: The Reset Type parameter is available since FP95-113 V3.

- The MG(SIP) is rebooted due to causes other than the above.

l.) The ongoing call retained by this feature will be disconnected when:

- The MG(SIP) does not receive RTP packets from the Telephony Server.
- The MG(SIP) has received a disconnect order from the Telephony Server.
- Ether link down is detected on the Telephony Server or in the SIP network. **Note 4**

  Note 4: A link down trap is generated after five consecutive failures of LAN cable connection for a period of one second.

- BYE request is received from the SIP network.
- Session timer is expired in the SIP network side.

m.) When all the retained calls are ended, the MG(SIP) reboots and performs reregistration.

Programming

This feature is invalid for default setting. Use the configuration command/Web Programming of MG/UG50 to make it valid. For the following data assignment, refer to the configuration command of each MG in Peripheral Equipment Description (IP Devices). For UG50, refer to Peripheral Equipment Description (UG50).

- Setting of Call Retention

  MC&MG-COT/MG(PRI) /MG(BRI)/MG(SIP) - [SET CALL_HOLD]

- Setting of the time-out value for RTP reception

  MC&MG-COT/MG(PRI)/MG(BRI) - [SET RTP_STOP_DETECT]

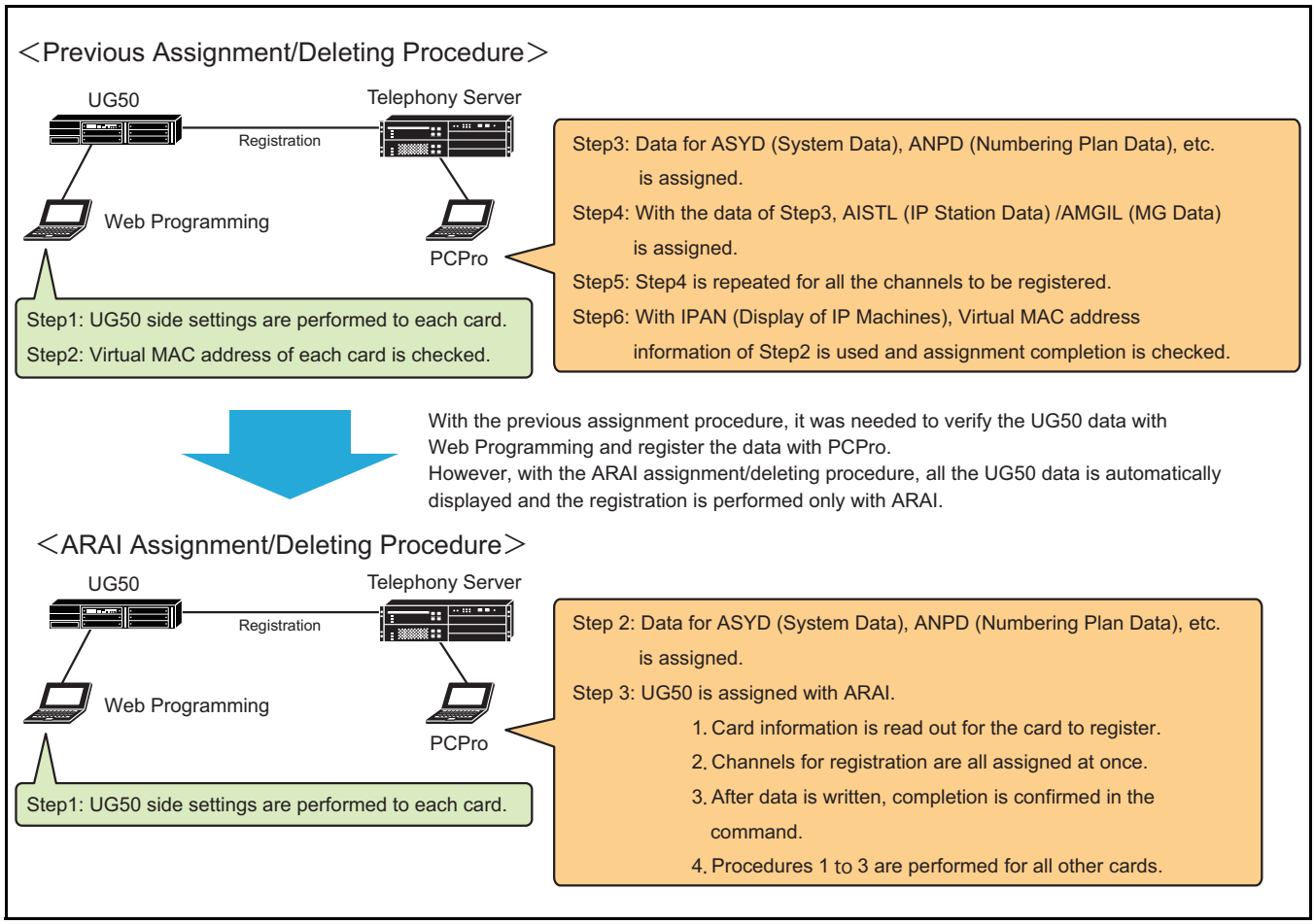
  MG(SIP) - [SET CALL_HOLD]

15. UG50 Related Data Setting

15.1 How to Use the ARAI Command to Assign or Delete Office Data for a UG50

General Description

The ARAI command collects and displays information on channels of the specified UG50, including Device Name, MAC address, slot number, etc. to facilitate office data input by using only PCPro and thus eliminating the need to look at the Web Programming screen of the UG50.



Service Conditions

<Important Notices>

1. The ARAI command is available from FP95-114 V4 Issue 2.0 and from UG50 software version 2.00.

<Interactions with Other Features>

1. If the FCCS CLUSTER feature of LOCATION DIVERSITY [L-55] is in use, the ARAI command cannot be used, use other commands as the AISTL command for displayed the needed data.

<Hardware/Software>

1. For supported UG50 cards, see "Registration Unit List" section in Startup chapter of Peripheral Equipment Description (UG50). For the settings of the cards, see "Office Data Settings on Telephony Server" section in the same chapter of Peripheral Equipment Description (UG50).

<Conditions Specific to This Feature>

1. If you modify the Device Name parameter in the UG50 that already has a channel registered with the Telephony Server, the old name prior to the modification may be displayed on the ARAI command screen. In this case, restart the UG50, and with the re-registration to the Telephony Server the new name will be displayed. However, even if you do not restart the UG50 and the old name is displayed, this does not affect the assignment of office data for the UG50
2. The following table shows the types of office data that can be entered on the ARAI command and provides a comparison with the AISTL command.

Office Data to Configure

R: Read Only, D: Data Entry, X: Available, N: Not Available

ARAI Command			AISTL Command Note 2
Parameter	Description	Read Only / Data Entry	
STS	Status of office data	R	N
SLOT	Slot Number	R	N
MAC ADDRESS	MAC address	R	X Note 3
CH	Channel Number	R	X Note 4
TN	Tenant Number	D	X
STN	Station Number	D	X Note 5
LENS	Line Equipment Number	D	X
TEC	Telephone Equipment Class	D	X
KIND	Type of IP device	R	X
RSC	Route Restriction Class	D	X
SFC	Service Feature Restriction Class	D	X
ADM/GST Note 1	Administration Station / Guest Station	D	X
ROOM CLASS Note 1	Room Class	D	X

ARAI Command			AISTL Command Note 2
Parameter	Description	Read Only / Data Entry	
ANNEX Note 1	Building in the complex (main building, Annex 1, Annex 2,...)	D	X
GROUND Note 1	Ground/Underground	D	X
FLOOR Note 1	Floor	D	X
RESULT	Result of the processing	R	N

Note 1: Displayed only when the Hotel feature is enabled (ASYD, SYS1, Index 160, Bit 0 = 1).

Note 2: This column indicates whether or not each parameter of the ARAI command is available in the AISTL command.

Note 3: The MAC ADDRESS parameter in the ARAI command corresponds to the MC-ID parameter in the AISTL command.

Note 4: The CH parameter in the ARAI command corresponds to the MC-CH parameter in the AISTL command.

Note 5: The STN parameter in the ARAI command corresponds to the IP STN parameter in the AISTL command.

Programming

15.1.1 Office Data Assignment

The following steps show how to assign the office data of each UG50 channel in the Telephony Server.

STEP 1: Start the ARAI command.

The initial screen is displayed.

If either of the following conditions is true, go to the next step. Otherwise, go to Step 3.

(a) [List of Remote Equipment Data] does not contain the IP address of the UG50 you want to update, and there are less than 256 UG50s shown.

(b) [List of Remote Equipment Data] contains the IP address of the UG50 you want to update, but the card information is empty.

STEP 2: Click the **Refresh** button.

The latest information (**Note 1**) about UG50s is loaded into the list.

Note: Information about UG50 is not displayed in [List of Remote Equipment Data] when the UG50 is:
- operating in IPG mode or Retrofit mode,
- unable to communicate with the Telephony Server, or
- running unsupported software.

Note 1: If you click the **Refresh** button when [List of Remote Equipment Data] is already filled with 256 IP addresses for the UG50s, only the slot information will be updated.

STEP 3: In [List of Remote Equipment Data], locate and select the IP address of a UG50 of your choice.

The selected item is highlighted.

Note: Consider the following:
- Up to 256 IP addresses for the UG50s are displayed in the list.
- Initializing the Telephony Server removes all records of the UG50s currently displayed in the list.
- UG50s and their associated information are listed from top to bottom in the order that they are detected.
- Slots with no information collected from the UG50 are left blank and cannot be used to assign or delete any data using the ARAI command. To refresh the screen with the latest UG50 information, click the **Refresh** button.

STEP 4: From the **Card** drop down list, select the slot containing the card with the office data that you want to assign in the Telephony Server.

If you select **"All <card type>"** (**Note 2**), you can enter office data for more than one of the same type of card at one time.

Note: The **Card** drop down list displays slots containing a card, regardless of whether or not the card is supported for the ARAI command. If a slot with an unsupported card is selected, the system disables the **GET** button, preventing access to the channel data of the card.

Note 2: The **All <card type>** option is displayed when multiple cards of the same type are present in the UG50.

STEP 5: Click the **GET** button. **Note 3**

[Autofill Specify] and [List of Remote Equipment Office Data] are displayed.

Note 3: Consider the following:
- When the **GET** button is clicked, the Telephony Server communicates with the UG50 to collect and display information about all the channels for the specified UG50 card.
- The collection process, however, can be stopped at any time by clicking the **ABORT** button. In that case, [List of Remote Equipment Office Data] only displays the office data collected before the interruption.
- If the ARAI command is collecting data and the **GET** button is pressed from another PCPro, this second PCPro is denied service and receive a pop-up error message.

If an error is detected in the information collection, an error message will appear in the **RESULT** field for the corresponding record.

STEP 6: In [List of Remote Equipment Office Data], select the check box next to each record with office data that you want to assign in the Telephony Server and then enter the office data values. To select all the records at one time, place a check mark in the **All Select** check box.

For details of office data parameters, see the table "Office Data to Configure" (**Note 4**).

Note: If you want to enter Telephone Numbers, use the ALGSL/ALGSN command in addition to the ARAI command.

Note: You can use autofill to enter office data. For details of autofill, see "[Autofill]."

Note 4: Items indicated with "R" in the "Read Only / Data Entry" column of Office Data to Configure are read only and automatically set by the command.

[Autofill]

The following autofill functions are available for each record selected:

Note: Autofill operates with two criteria:
- The lead value among the selected records.
- The value of already registered data among the selected records.

Value of already registered data criteria is the prevalent criteria. Even if autofill is operating with the lead value criteria, if there is already registered data, the value of already registered data criteria becomes prevalent and therefore autofill starts to use the data from the already registered value. If there are several already registered values, already registered data criteria is used each time.

(a) Identical

The "Identical" autofill copies the current value of the specified fields into the corresponding fields in subsequent selected records (applies to only records selected below the current record).

This function is available in the fields below:

- TN
- TEC
- RSC
- SFC
- ADM/GST **Note 5**
- ROOM CLASS **Note 5**
- ANNEX **Note 5**
- GROUND **Note 5**
- FLOOR **Note 5**

Note 5: Displayed only when the Hotel feature is enabled (ASYD, SYS1, Index 160, Bit 0 = 1).

(b) Count Up

The "Count Up" autofill populates the specified fields in the selected records with sequential numbers (with an increment of 1) starting from the number entered in the field in the top-most selected row. The sequential number generator counts all records from the top-most to the bottom-most selected rows, even including the records not selected in between.

This function is available in the fields below (**Note 6**):

- STN
- LENS

Note 6: The ARAI command does no validation. It is the users responsibility to verify the Station Number and LENS values are valid.

<Counting Station Numbers>

The "Count Up" autofill uses the following 12 symbols to count station numbers (**Note 7**):

0, 1, 2, 3, 4, 5, 6, 7, 8, 9, *, #

Asterisk (*) and a pound (#) symbol are used to count beyond 9. After "#" is reached, the digit goes to zero and "1" is added to the digit in the next place to the left.

Note 7: In this autofill function, "*" and "#" are not counted as the last digit in a station number string. These symbols, however, are counted as the digits other than the last digit.

<Counting LENS>

The "Count Up" autofill uses the following 10 symbols to count LENS:

0, 1, 2, 3, 4, 5, 6, 7, 8, 9

The range of values for each area is as follows:

MG: 0 to 31 (The maximum value depends on the System Data.)

UNIT: 0 to 3

GROUP: 0 to 31 **Note 8**

Note 8: LENS from Virtual Groups xxx240 to xxx317 cannot be used for assigning UG50 stations or trunks.

LEVEL: 0 to 7

[Autofill Data Entry]

[1] In [Identical] and/or [Count Up], select the check box for the field for which you want to use autofill.

[2] In [List of Remote Equipment Office Data], select the check box next to each record you want to enter office data for using the autofill option that you selected in [1].

Make sure that only the check box for the target records are selected.

[3] Click the **Apply** button.

STEP 7: In [List of Remote Equipment Office Data], select the check box next to each record containing office data that you want to assign in the Telephony Server, and then click the **SET** button.

Note: The writing process cannot be stopped until all the data of the selected records is written to the Telephony Server.

The result of the writing process is displayed in the **RESULT** field for each record. If an error is detected, corresponding error message will appear.

Now the data input is completed.

15.1.2 Office Data Deletion

If you want to remove unnecessary office data from [List of Remote Equipment Office Data], proceed as follows.

STEP 1: In [List of Remote Equipment Office Data], select the check box for a UG50 with office data that you want to delete, and then click the **DEL** button.

Note: The deleting process cannot be stopped until all the data of the selected records is deleted from the Telephony Server.

The result of the deleting process is displayed in the **RESULT** field for each record. If an error is detected, corresponding error message will appear.

Note: The **DEL** button deletes information that can be removed with the AISTL command.

CHAPTER 3
BUSINESS FEATURES

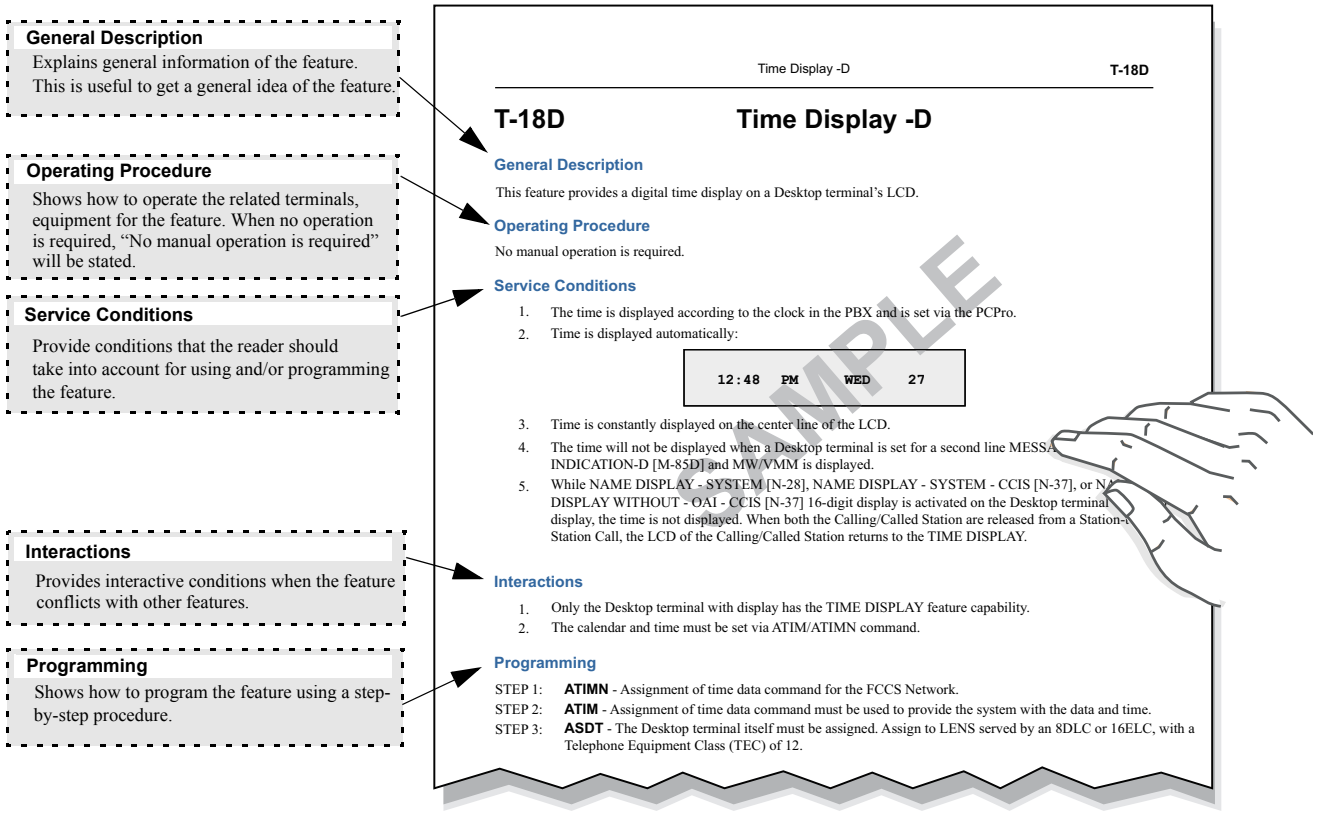
This chapter explains business features in an alphabetical order under the following headings:

- General Description
- Operating Procedure
- Service Conditions
- Interactions **Note 1**
- Programming
- FCCS Programming **Note 1**

Note 1: This item may not appear depending on service features.

The following shows an example of business features that is covered in this chapter.

Example of Business Feature Description



A-1 Attendant Camp-On with Tone Indication

General Description

This feature allows the Attendant to transfer an incoming trunk call to a busy station, placing the call in a special waiting mode (The call is "camped-on" to the busy station). The busy station receives a distinctive camp-on tone and when it becomes available, it is connected to the waiting call, either with or without the assistance of the Attendant as specified in the following options:

- Automatic Attendant Camp-On

  When the busy station becomes idle, it automatically rings and is connected to the waiting call upon answering.

- Semiautomatic Attendant Camp-On

  When the busy station becomes idle, the Attendant is informed of the availability, and then connects the waiting call to the station.

Operating Procedure

<Automatic Attendant Camp-On>

To activate from the ATTENDANT CONSOLE [A-3]

1. ATTENDANT CONSOLE [A-3] indicates an incoming call.
2. Press the ANSWER or INCOMING CALL IDENTIFICATION [I-2] key.
3. Dial the appropriate telephone number. Receive a busy tone.
4. Press the RELEASE key. The camp-on set tone is sent to the busy station and Camp-On is set.

To cancel from the ATTENDANT CONSOLE [A-3]

1. Press the LOOP key corresponding to the held call. The camp-on between the trunk call and the busy station is cancelled by this operation.
2. Press the DEST key. Hear a busy tone.
3. Press the CANCEL key, automatically returns to the trunk party.

<Semiautomatic Attendant Camp-On>

To activate from the ATTENDANT CONSOLE [A-3]

1. ATTENDANT CONSOLE [A-3] indicates an incoming call.
2. Press the ANSWER or INCOMING CALL IDENTIFICATION [I-2] key.
3. Dial the appropriate telephone number. Receive a busy tone.
4. Press the RELEASE key. The camp-on set tone is sent to the busy station and Camp-On is set.
5. Press the LOOP key to answer a recall when the station becomes idle.
6. Press the RELEASE key to connect the station and the waiting trunk party.

To cancel from the ATTENDANT CONSOLE [A-3]

1. Press the LOOP key corresponding to the held call. The camp-on between the trunk call and the busy station is cancelled by this operation.
2. Press the DEST key. Hear a busy tone.
3. Press the CANCEL key, automatically returns to the trunk party.

Service Conditions

<Common Conditions>

1. The Attendant Console must first be provided.
2. ATTENDANT CAMP-ON WITH TONE INDICATION (hereinafter also called as "Attendant Camp-On") can be used for incoming C.O. line or TIE line calls to a busy station. Attendant-assisted station-to-station calls cannot be camped-on.
3. Attendant Camp-On is controlled entirely from the ATTENDANT CONSOLE [A-3].
4. Only one incoming call may be camped-on to a particular station at any given time. Multiple Camp-On is not allowed.
5. When Attendant Camp-On is denied, the Attendant may place the Incoming Call into hold mode by pressing the HOLD key.
6. The maximum number of simultaneous Camp-Ons per ATTENDANT CONSOLE [A-3] is six. If ATTENDANT LOOP RELEASE [A-6] is enabled (ASYD, SYS2, Index 2, Bit 5=1), the maximum number is 12.
7. Attendant Camp-On may be denied if the busy station is:
   a.) dialing or in the LINE LOCKOUT [L-3] mode.
   b.) in the Make Busy state.
   c.) being rung.
   d.) already camped-on by a previous call.
   e.) protected from Camp-On by CLASS OF SERVICE - INDIVIDUAL [C-15].
   f.) protected with Data Line Privacy [ASFC SFI 11 = 0 (Data Line Privacy)].
   g.) being connected to an Attendant Loop.
   h.) activating any of the following conflicting features:

ATTENDANT OVERRIDE	[A-7]
ANNOUNCEMENT SERVICE	[A-15]
ANNOUNCEMENT SERVICE - ATTENDANT	[A-59]
BOSS - SECRETARY OVERRIDE -D	[B-5D]
BUSY VERIFICATION	[B-3]
CALL BACK	[C-1]
CALL HOLD	[C-6]
CALL TRANSFER	[C-10, C-11]
CALL WAITING - ORIGINATING	[C-31]
CALL WAITING - TERMINATING	[C-12]
CONSULTATION HOLD - ALL CALLS	[C-17]
DIAL ACCESS TO ATTENDANT	[D-2]
EXCLUSIVE HOLD -D	[E-4D]

EXECUTIVE RIGHT-OF-WAY	[E-1]
NON-EXCLUSIVE HOLD -D	[N-7D]
OFF-HOOK QUEUING	[O-7]
OUTGOING TRUNK QUEUING	[O-2]
PRIVACY -D	[P-14D]
PRIVACY RELEASE	[P-18]
THREE-WAY CALLING	[T-2]
VOICE CALL -D	[V-2]

Note: In each of the above instances, the Attendant will receive a reorder tone.

8. Attendant Camp-On may be denied on DICTATION ACCESS [D-3] and PAGING ACCESS [P-1] Connections by assigned Office Data.

9. Attendant Camp-On is available only when the busy station is in a one-on-one conversation with another extension or an outside party (through C.O. line or Tie line). There is no limit to the number of lines to which this feature can be assigned.

10. Any restriction that prevents a call from terminating directly at a station also prevents a call from waiting at that station.

11. CALL FORWARDING [C-2, 3, 5]: When call forwarding is in effect, the forwarding operation occurs before any Call Waiting is allowed. Once a call has been forwarded, the call may wait at the new station if Call Waiting is allowed.

12. HOT LINE [H-1]: Attendant Camp-On is allowed toward the Hot Line.

13. Route Restrictions not allowing a call to terminate at a station will also not allow Attendant Camp-On.

14. STATION HUNTING [S-7, 8, 9] and UNIFORM CALL DISTRIBUTION [U-1] take precedence over Attendant Camp-On. If all stations in the Station Hunting Group are busy, the call will wait at the called station.

15. TIE LINE ACCESS [T-3]: Attendant Camp-On is denied to a station over a TIE Line (distant end).

16. Attendant Camp-On and SERIAL CALL [S-15] are mutually exclusive.

17. Attendant Camp-On is usually denied to a station connected to Value-Added Modules (VAM).

18. The Automatic Attendant Camp-On and Semiautomatic Attendant Camp-On options cannot be used simultaneously. The Attendant Camp-On setting allows for either option.

19. Attendant Camp-On is not available for PSs, ISDN terminals and SP Controlled SIP terminals.

20. When a trunk call is camped on to a busy station, only the called party will hear a call waiting notification tone (camp-on set tone). In addition to the audible alert, if the called station is a Desktop terminal, the “CAMP” message appears on the LCD and the ANSWER key blinks.

21. By setting the System Data (ASYD, SYS1, Index 69, Bit 5), you can specify which station (originally called station or station forwarded to) an incoming trunk call is camped-on to when the call is forwarded with CALL FORWARDING- BUSY LINE [C-2] and encounters a busy condition.

    Note: This setting is also available when the call forwarding destination is a station in an FCCS network.

Note: When the call forwarding destination is outside of the self office, Attendant Camp-On is restricted and does not work.

22. Calls that remain camped-on for longer than the predetermined timing will initiate AUTOMATIC RECALL [A-8]. See VARIABLE TIMING PARAMETERS [V-1].

23. When an incoming trunk call is camped-on to a busy station with Attendant Camp-On, if the Attendant presses the LOOP key on the Attendant Console, the camp-on is canceled.

<Conditions Specific to Automatic Attendant Camp-On>

1. When receiving a camp-on set tone, a user of an Analog Station or Desktop terminal can place the current call on hold by pressing the switch hook and dialing a preprogrammed CALL HOLD Code (for details, see CALL HOLD [C-6]) to answer a camped-on call.

   Note: In addition, Desktop terminal users can also use the HOLD key* or ANSWER key to answer a camped-on call without releasing the original call.
   *: The HOLD key is a Line/Feature key to which the CALL HOLD [C-6] feature is assigned.

2. Automatic Attendant Camp-On may be denied if the Semiautomatic Attendant Camp-On is activated on the busy station.

<Conditions Specific to Semiautomatic Attendant Camp-On>

1. Unlike the Automatic Attendant Camp-On option, you cannot answer a camped-on call without disconnecting the first call. If you are a user of an IP/Digital terminal, the LCD screen displays “CAMP” and the ANSWER key blinks when a trunk call is camped on to your station. You cannot, however, answer the camped-on call by using the CALL HOLD Code, HOLD key or ANSWER key without releasing the original call.

2. Semiautomatic Attendant Camp-On may be denied if the Automatic Attendant Camp-On is activated on the busy station.

3. The Semiautomatic Attendant Camp-On option is also available for incoming calls on ISDN trunks.

4. When the semiautomatic camped-on station becomes idle, the Attendant Console receives a recall. If the station makes a call before the Attendant answers the recall, the system sends:

   - Ringback tone to the station when it is an analog telephone
   - No tone to the station when it is a Multiple Line terminal

Programming

STEP 1: **ASYD** - SYS1, Index 64, Bit 6. Type of Attendant Camp-On
0/1 = Automatic Attendant Camp-On/Semiautomatic Attendant Camp-On

SYS1, Index 145. This index provides timer for Camp-On call to recall to the Attendant Console. [RAM Data is 3F (hexadecimal) = 30 seconds.]

SYS2, Index 0, Bit 4 must be assigned as data "1," BUSY VERIFICATION.

SYS1, Index 69, Bit 5. Attendant Camp-On 0/1 = Set to Forwarded-from Station/Set to Forwarded-to Station

Note: All Timers settings (ASYD, SYS1, Index 128 to Index 159), operate normally on values set via the System RAM. Typically assign 00H. The RAM default values will be noted beside each System Timer.

STEP 2: **ASFC** - Assign to the telephone a Service Feature Restriction Class that allows SFI 11 (Attendant Camp-On). Data must be "1." Also allow SFI 10 (CALL HOLD [C-6]).

A-2 Attendant-Controlled Conference

General Description

This feature permits the Attendant to establish a Conference among 8-parties, 16-parties, or 32-parties. The Conferees may consist of any combination of stations and/or trunks, whether inside or outside parties. The number of conferee varies depending on the conference trunk (CFT) type as shown below:

CFT	NUMBER OF CONFEREES (including conference leader)
PA-CFTW	8-party × 2 groups
PA-CFTV	8-party × 4 groups
	16-party × 2 groups
	32-party × 1 group
CH-CFTA	8-party × 4 groups
	16-party × 2 groups
	32-party × 1 group

Operating Procedure

1. Station calls Attendant to request Conference.
2. Attendant dials the station on Circuit #0 of the CFT then presses RELEASE.
3. Attendant sequentially calls all desired parties, presses START, dials the station on Circuit #1 of the CFT, then presses RELEASE. Once all parties are connected, the Conference Leader with a Pushbutton Telephone may dial "0" to block additional participants, thereby assuring privacy.
4. If the Conference Leader dials "#", all Conferees will be disconnected. Otherwise, the Conference is terminated only after all parties have hung up.
5. To release from a Conference Circuit individually, a user must hang up. The remaining Conferees may continue without interruption.

Service Conditions

1. The Attendant Console must first be provided.
2. Conference Units are designed to provide adequate amplification, but the condition of the Central Office Connections will affect transmission.
3. Conference Extension Numbers should be placed in a STATION HUNTING - TERMINAL [S-9] Group with only the Pilot Number being published.
4. If the Conference Leader connected to the Conference Unit dials "#" during the course of the Conference, all Conferees will be disconnected.

- For privacy function
- When conference speech for less than defined members (8/16/32 parties) is made, the Conference Leader can keep another member from entering the conference by pushing "0".
- Conference Leader can set/cancel privacy by pushing "0."

- Privacy function makes available ports busy. (No calls cannot be arrived in.)
- It is unavailable to confirm whether or not Privacy function is set. The Conference Leader must be careful of using this function.
- When the Conference Leader sets privacy function and leaves the conference, the conference port is unavailable, however, the port for Conference Leader is available.
- Privacy function is canceled when all of the participants (Conference Leader and all the Conferees) have left the conference.

5. The Conference Leader with a Pushbutton Telephone must be connected to Level "0" of the Conference Circuit.
6. Override features such as BUSY VERIFICATION [B-3], ATTENDANT OVERRIDE [A-7] or EXECUTIVE RIGHT-OF-WAY [E-1] are denied on Conference Call.
7. This feature works with PA-CFTW, PA-CFTV or CH-CFTA. For setting the switches see the Circuit Card Description manual (CHAPTER 4 CARDS IN 8U-PIR for PA-CFTW and PA-CFTV and CHAPTER 3 CARDS IN 7U-PIR for CH-CFTA).
8. To disconnect a line from the conference, system data (ASYD, SYS1, Index 20, Bit 6=0) is required.
9. The Conference Leader cannot disconnect each Conferee (port) from the conference they are participated in.
10. If the Conference Leader is disconnected from the conference or the rest of members are only two parties or less, the conference trunk is still used.

Programming

STEP 1: **ASYD** - Assign System Data, SYS1.
Index 20, Bit 6=0 (Line Circuit Reverse Relay Control when Reorder Tone is connected to a station/trunk.)
Index 64, Bit 0: Assign the compression method (0/1 = μ-law/A-law)

STEP 2: **ASDT** - Assign tenant number, station number, LENS, and the Telephone Equipment Class, (TEC), to the stations of the PA-CFTW/PA-CFTV/CH-CFTA. Also the Service Feature Restriction Class and Route Restriction Class (RSC) are assigned to the telephone. The RSC should allow connections to the Incoming and Outgoing Routes that will be connected to the Conference.

TN: Tenant Number
STN: Station Number
LENS: Line Equipment Numbers (six-digits)
TEC: Telephone Class (1-31)
1: DP (10 PPS)
2: PB
→ 3: DP/PB
12: Desktop Terminal
13: Data Terminal via Desktop Terminal
14: Hotline
15: CAS Line
16: Data Terminal via Data Module
18: Virtual Circuit

23: ISDN Terminal
26: VPS
27: Eight Conference Equipment

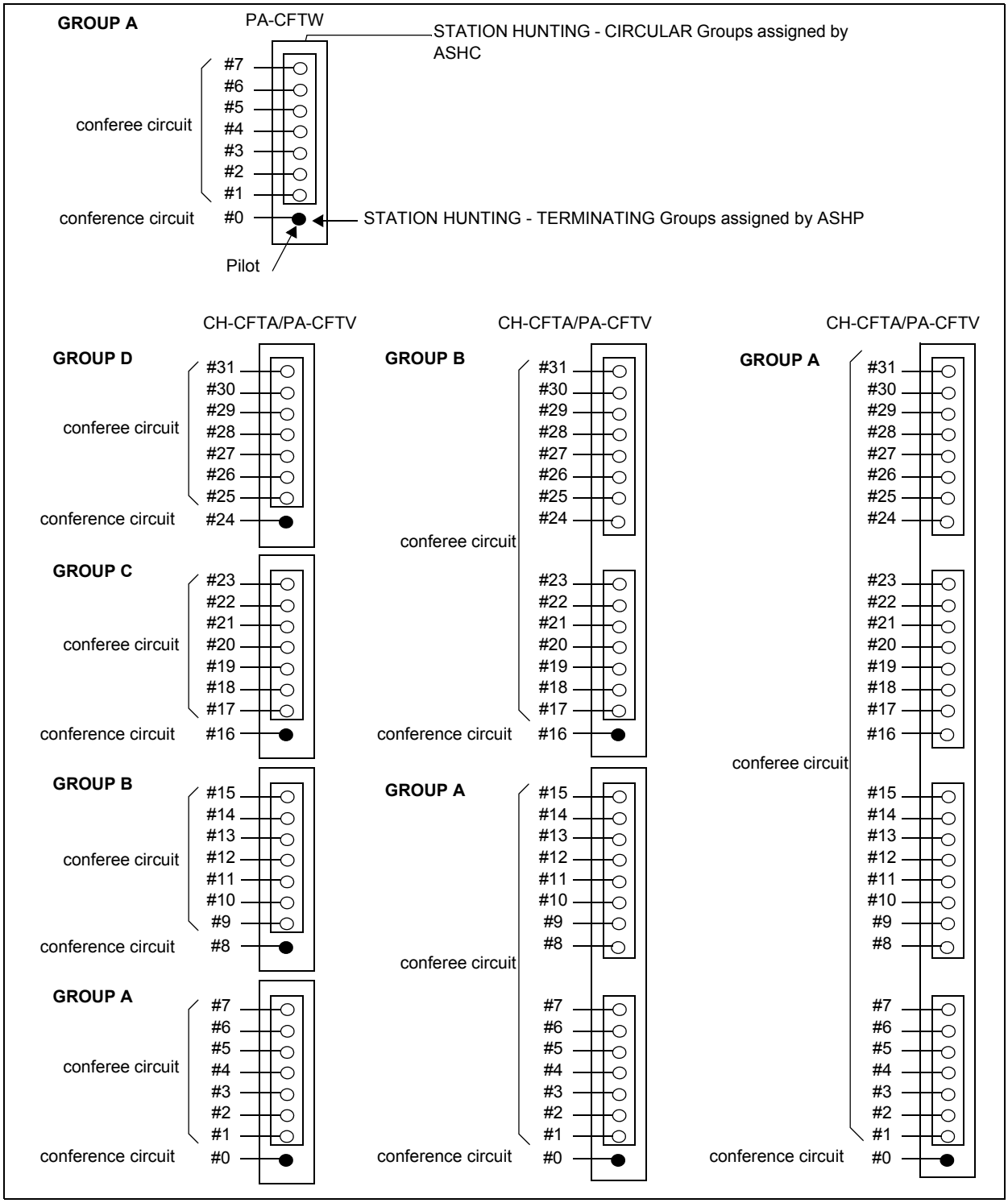
RSC: Route Restriction Class (0-15)

For assignment of RSC, ARSC.

SFC: Service Feature Restriction Class (0-15)

For assignment of SFC, use the ASFC command.

STEP 3: **ASHP** - Assign a Pilot Hunting Group using the conferees' circuits (other than conference circuit) of CFT card (PA-CFTW/PA-CFTV/CH-CFTA). Circuit #0/8/16/24 of all CFTs to be used for conference should be put into a Circular Hunt (ASHC).



STEP 4: **ASFC** - Assign 0 to Restriction Data (RES) for Service Feature Index (SFI) 11 (Data Line Privacy).

FCCS Programming

This data setting describes only the additional or replacement commands used to activate FCCS Services. When used as an FCCS feature, these commands replace the non-network commands (for example, the AAED command is replaced by the AAEDN command). Note that the data setting for FCCS Service varies depending on the node.

STEP 1: **ASHPN** - Assign a Pilot Hunting Group using the conferees' circuits (other than conference circuit) of CFT card (CH-CFTA/PA-CFTW/PA-CFTV). Circuit #0/8/16/24 of all CFTs to be used for conference should be put into a Circular Hunt (ASHCN).

STEP 2: **ALGSN** - Assign a Telephone Number to the entry of the physical station data created with the ASDT command.

STEP 3: **ASFCN** - Assign 0 to RES for SFI 11 (Data Line Privacy).

A-3 Attendant Console

General Description

The ATTENDANT CONSOLE operates on a Switched Loop basis. Six Attendant Loops terminate at each Console via the associated position circuit. The Attendant can answer, originate, hold, extend and re-enter calls through each Loop. The number of calls may be effectively increased to 12 through use of the ATTENDANT LOOP RELEASE [A-6] feature.

Operating Procedure

Refer to the description of each feature related to Attendant Console.

Service Conditions

1. Each ATTENDANT CONSOLE is equipped with dual handset jacks, one on either side. For training purposes, both jacks may be used simultaneously. For normal operation, the Attendant may choose the most convenient side.

   Note: A drop in volume may be noticed while a second handset/headset is connected.

2. A Console Switched Loop may receive a second call immediately upon release of the first, regardless of whether or not the call has been answered by a Station when using the ATTENDANT LOOP RELEASE [A-6] feature.
3. A maximum of four Consoles per PIR can be provided.
4. The maximum number of Consoles allowed per system is:

   IMG System: 8
   MMG System: 32

   Note: When the system is the SV9500 migrated from IPX, the maximum number of Consoles allowed per system is:

   IMG System: 16
   MMG System: 60

5. The ATTENDANT CONSOLE is equipped with a digital time display.
6. The ATTENDANT CONSOLE is equipped with a numerical Call Waiting display.
7. When answering a CALL FORWARDING [C-2, 3, 5, 25] Call, the Attendant can identify the Called Telephone Number on display area instead of the Trunk Number by assigned System Data.
8. The Desk Console can display a maximum of 24-digit Calling/Called Station Number (Telephone Number) on the LCD. In addition, the Calling Number of Caller ID Service can be displayed.
9. The Desk Console can display 8-Bit ASCII characters.
10. Name Display is available when Name Display Service is available for the Desktop terminal. The contents of the Name Display is the same as that of the Desktop terminal.
11. The maximum number of digits for the Name Display are as shown below:
    - 16 for the Name assigned by PCPro
    - 15 for the Calling Name sent from the Central Office

12. When the Calling Number is not received, the Route/Trunk Number or the Central Office Code (COC) is displayed.
13. The Desk Console and the current Attendant Console can be used together in a system. However, the Attendant Console can display a maximum of 24-digit Telephone Numbers and Caller ID information is not displayed.
14. The data setting of the Station Number Display Pattern for the Attendant Console (SYS1, Index 82) is unavailable for the Desk Console.
15. An ATI circuit card is required. The switch setting for CH-CS00/PA-CS33-C/PA-CS33 (ATI) is shown below:

CH-CS00

Switch Name		Setting
SW 12	1	OFF (fixed)
	4	OFF (fixed)

PA-CS33-C/PA-CS33

Switch Name		Setting
SW 16	1	OFF (fixed)
	2	ON (fixed)

16. A Terminating Node (FPC) must be assigned in the FCCS Network.
17. Identical System Data related to Attendant Console should be programmed at all Nodes.
18. A different destination for Central Office Incoming Call (Ringdown) cannot be set at each Tenant Group.
19. TOLL DENIAL/TOLL DIVERSION [T-6] is not available for the Attendant Console.
20. Because the destination station is busy state, the Attendant Console presses the SRC key to go back to the two-way connection with the Calling Party. In this case, even if the destination station becomes the idle state and the Attendant Console presses the DEST key, Busy Tone is sent to the Attendant Console.
21. PAD Control (APAD) does not apply to Attendant Consoles. When the Attendant Console is connected to analog trunk, a receiver volume (PAD Control) can be changed by adjusting PAD value of ARTD somewhere between 6db and -5db and also by adjusting the volume setting of Attendant Console.

    Note: When the volume level of analog trunk is excessively low, the PAD value will not be in the range. A gap in receiver volume may be developed between calls with analog trunk and station.

22. When major alarms or minor alarms are generated in the system, alarm lamp on Attendant Console lights red. This lamp is enabled by setting the basic data for Attendant Console.

<Conditions Specific to UMGi/RNIP System>

23. The maximum number of Consoles allowed per system is:
    UMGi/RNIP System: 32

Note: In a RNIP/UMGi system is necessary to collectively set all the Attendant Consoles in one single node that can be the CCN or any of the RNs. Therefore, the maximum number of accommodated consoles can be different depending on the system configuration (IMG/MMG).

Interactions

There are no known interactions.

Programming

Called Numbers display when Call Forwarding to ATTCON.

STEP 1: **ASYD** - SYS1, Index 70, Bit 0 (0/1 = Out of Service/In Service)

Call Waiting Lamp Display.

SYS2, Index 8 and Index 9, Bit 0 to Bit 7.

FCCS Programming

This data setting describes only the additional or replacement commands used to activate FCCS Services. When used as an FCCS feature, these commands replace the non-network commands (for example, AAED is replaced by AAEDN). Note that the data setting for FCCS Service varies depending on the Node.

Note: When using ASYDL or ASYDN to program an FCCS feature, the basic programming of ASYD is required in addition to ASYDL or ASYDN.

STEP 1: **ASYDL** - This data must be set to all the Nodes.

SYS1, Index 640: Assign the Fusion Point Code (FPC) of the Node providing the Attendant Console.

Note: Assign the Self FPC at the Terminating Node.

STEP 2: **ASPAL** - This data must be set to all the Nodes.

Assign SRV = “SSC: Service Code”, SID = 2 (Operator Call), FPC of the Node that Operator Calls are to be terminated to.

A-4 Attendant Keypad

General Description

This feature permits the Attendant to dial all calls from the ATTENDANT CONSOLE [A-3] via a Pushbutton PAD.

Operating Procedure

Refer to the description of each feature related to Attendant Console.

Service Conditions

1. Whether the System is equipped with DP or DTMF Trunk Lines, signals from the ATTENDANT KEYPAD will be converted.
2. Via the keypad, the Attendant may originate calls, process Incoming Calls, and activate various features. They are as follows:
   a.) PUSHBUTTON CALLING [P-4]
   b.) PUSHBUTTON CALLING - ATTENDANT ONLY [P-5]
   c.) ROTARY DIAL CALLING [R-5]
3. The ATTENDANT KEYPAD generates DC Signals, which in turn are converted to either DP or DTMF Signals by the Attendant Interface Circuit.
4. The keypad consists of 12 keys used to generate digits 0-9, * and #.

Programming

ASYD - SYS2, Index 2, Bit 1. This assignment will determine type of Sender used by the Attendant Console, either dial pulse or pushbutton, 0/1: DP/PB. The Attendant Console must first be provided.

SYS1, Index 134, RAM Data is 32H, four seconds. This is the Interdigit Timer for the Register/Sender Circuit on the ATI Card. For normal operation, no assignment is required via PCPro.

A-5 Attendant Lockout

General Description

This feature denies an Attendant the ability to re-enter an established Trunk/Station Connection without being recalled by the Station.

Operating Procedure

No manual operation is required once this feature has been programmed into System Data.

Service Conditions

1. This feature is mutually exclusive with:
   ATTENDANT OVERRIDE [A-7]
   BUSY VERIFICATION [B-3]
2. Lockout is inoperative on TRUNK-TO-TRUNK CONNECTION [T-10] when the Release Signal is not provided from the distant end.
3. Lockout is not used unless the feature is provided which gives Two-party Hold capability. Without this capability, the completed call is removed from the ATTENDANT CONSOLE [A-3] Loop, so the Attendant cannot reenter for assistance or any other purpose.

Interactions

1. ATTENDANT OVERRIDE [A-7]: In those cases where Answer Supervision is returned on TRUNK-TO-TRUNK CONNECTION [T-10], the Attendant can override the busy condition by using the ATTENDANT OVERRIDE [A-7] feature.
2. TRUNK-TO-TRUNK CONNECTION [T-10]: Privacy is automatically disabled for TRUNK-TO-TRUNK Calls, if the Release Signal is not provided from the distant end.
3. SPLITTING [S-4]: The Privacy feature works with the SPLITTING [S-4] feature causes automatic splitting.
4. Switched Loop Operation: Call processing is modified so that pressing a Loop key does not cause the Attendant to be bridged on an established call unless the Attendant is recalled. An exception is a TRUNK-TO-TRUNK CONNECTION [T-10] when the Release Signal is not provided from the distant end.
5. ATTENDANT-CONTROLLED CONFERENCE [A-2]: When the Attendant reenters a Conference connection to answer a recall, only the Attendant and the recalling Station are connected together.

Programming

Per-Route Basis:

ARTD - Assign CDN 44:PRV as a “1”, Line Privacy.

Per-Station Basis:

ASFC - Assign “0” to SFI 4 [Executive Right-of-Way (Called)] and SFI 11 (Data Line Privacy) to the SFC assigned to the Stations.

Per-Tenant Basis:

ASYD - SYS2, Index 0, Bit 3, Bit 4 and Bit 5. Assign data “0”.

A-6 Attendant Loop Release

General Description

This feature allows an ATTENDANT CONSOLE Loop to become available for a second call as soon as the Attendant has directed the first call to a Station, even if that Station does not answer.

Operating Procedure

1. ATTENDANT CONSOLE indicates an Incoming Call.
2. Press the ANSWER or INCOMING CALL INDICATION [I-2] key.
3. Dial the desired Telephone Number. Receive Ringback Tone.
4. Before the Station answers, press the RELEASE key.
5. The Loop is now available for another call.

Service Conditions

1. The Attendant Console must first be provided.
2. Unanswered calls will be routed to the Attendant within the predetermined time via AUTOMATIC RECALL [A-8]. See VARIABLE TIMING PARAMETERS [V-1].
3. If all Attendant Loop Circuits are busy when AUTOMATIC RECALL [A-8] is activated, the system will automatically scan for an Idle Loop every eight seconds.
4. A maximum of six calls (one per Loop) may be released simultaneously from any single ATTENDANT CONSOLE.
5. This feature provides the Attendant with the equivalent of 12 Switching Loops.
6. In a MULTIPLE CONSOLE OPERATION [M-4], the Attendant who initiated the Loop Release will be recalled.
7. ATTENDANT LOOP RELEASE is only applicable to Incoming Trunk Calls extended to an unanswered Station or busy Station (Camped-On).
8. Calls that are held by the Attendant, using the HOLD key, cannot be released from the ATTENDANT CONSOLE [A-3]. These calls remain on the Switched Loop until they are either extended by the Attendant or abandoned by the Calling Party.
9. After the ATTENDANT LOOP RELEASE is activated, the Attendant cannot interact with the call until altered by the AUTOMATIC RECALL [A-8] feature.
10. ATTENDANT CAMP-ON: This feature, when active, permits the Attendant to CAMP-ON [A-1] to a busy Called Station. Upon ATTENDANT CAMP-ON WITH TONE INDICATION [A-1], the Attendant may release the call from the ATTENDANT CONSOLE [A-3].

Interactions

1. ATTENDANT LOOP RELEASE and CALL FORWARDING - DON'T ANSWER [C-3] on calls extended to a Station via the Attendant Console (SYS2, Index 6, Bit 7) are mutually exclusive.
2. ATTENDANT LOOP RELEASE is only applicable to Incoming Station or Trunk Calls extended to an unanswered Station or busy Station. See ATTENDANT CAMP-ON WITH TONE INDICATION [A-1].
3. Calls placed on CALL HOLD [C-6] by the Attendant Console are not released from the Console. These calls remain on the Loop until they are either extended by the Attendant or abandoned by the Calling Party.
4. A maximum of six calls per Console can be released and stored in the System Memory. A seventh call extended (unanswered/camped-on) will remain in the Loop.
5. Calls processed via SERIAL CALL [S-15] are not released from the Attendant Console. These calls remain in the Loop until completion.

Programming

ASYD - SYS2, Index 2, Bit 5. Assign as data "1." Assign on a per Tenant basis.

A-7 Attendant Override

General Description

This feature permits an Attendant to enter a Busy Trunk Connection within the System, via the Attendant Console [A-3]. Upon activation, a Warning Tone will be heard by the connected parties, after which they will be connected with the Attendant in a three-way bridge.

Operating Procedure

1. Press the TKSL key.
2. Dial the desired Trunk Route (two or three digits) and the Identification Numbers (three digits).
3. Upon receiving Busy Tone, press the Busy Verification (BV) key.
4. A Two-burst Tone is sent to the connected parties.
5. The Attendant can now monitor or join the connection.
6. Press the RELEASE key to disengage.

Service Conditions

1. This feature may be used during Trunk-to-Trunk Calls (COT, FX, WATS, CCSA, TIE Lines).
2. This feature cannot be activated when the connected Station/Trunk is:
   a.) dialing.
   b.) connected to a Three-way Connection.
   c.) when a PRIVACY [P-14D, P-15D] feature is activated.
   d.) connected to a Route programmed for Line/Trunk, Privacy, CDN 44:PRV.
   e.) listening to a Tone (DT, ROT, SST, BT, RBT).

   Note: When assigning ARTD, CDN 28 (ANS) = 0 for the specified trunk, this feature is available while the connected Station/Trunk receives RBT.
   When assigning ARTD, CDN 28 (ANS) = 1 for the specified trunk, this feature is not available while the connected Station/Trunk receives RBT.

   f.) protected by DATA LINE SECURITY [D-12].
   g.) being connected to an Attendant Loop.
   h.) activating any conflicting feature, such as:

BUSY VERIFICATION	[B-3]
BOSS - SECRETARY OVERRIDE	[B-5]
CALL HOLD	[C-6]
CALL TRANSFER	[C-10, C-11]
CALL WAITING - TERMINATING	[C-12]
CONSULTATION HOLD - ALL CALLS	[C-17]
DATA LINE SECURITY	[D-12]
DATA PRIVACY ON DEMAND	[D-17]
DIAL ACCESS TO ATTENDANT	[D-2, D-71]
EXECUTIVE RIGHT-OF-WAY	[E-1]
EXCLUSIVE HOLD -D	[E-4D]
NON-EXCLUSIVE HOLD -D	[N-7D]
OFF-HOOK QUEUING	[O-7]

OUTGOING TRUNK QUEUING	[O-2]
PRIVACY -D	[P-14D]
PRIVACY RELEASE	[P-18]
THREE-WAY CALLING	[T-2]
VOICE CALL	[V-2]

In each of the above instances, the Attendant will receive Reorder Tone.

3. ATTENDANT OVERRIDE and ATTENDANT LOCKOUT [A-5] are mutually exclusive features.
4. Each Tone Burst is 0.08 seconds in duration and is used to alert both parties that the Attendant is overriding the conversation.
5. This feature can be denied on DICTATION [D-3] and PAGING [P-1] connections by assigned ARTD data.
6. It is possible, via System Data, to completely disable the sending of Warning Tone during an ATTENDANT OVERRIDE Connection. Normally the Two-burst Warning Tone is sent when the ATTENDANT OVERRIDE Connection is initiated and then is repeated approximately every 15 seconds. The initial Tone Burst is also disabled. If the Trunk is connected to a Desktop terminal, the display is as follows:



After five seconds, the display returns to the normal Conference display, CONF.

7. ATTENDANT OVERRIDE is temporarily denied and Reorder Tone is heard if any of the following features are active:
   a.) EXECUTIVE RIGHT-OF-WAY [E-1]: If the Called Trunk is connected to a line on which EXECUTIVE RIGHT-OF-WAY is being used.
   b.) Trunk Restriction: If the Called TIE Trunk was routed to Reorder Tone as a result of Trunk Restriction.
   c.) SPECIAL DIAL TONE [S-2]: If the Called Trunk is on hold.
   d.) ATTENDANT CAMP-ON WITH TONE INDICATION [A-1]: If the Called Trunk is camped-on to a busy Station.
   e.) SERIAL CALL [S-15]: ATTENDANT OVERRIDE is denied to the parties held on the Console.
   f.) ATTENDANT INTERACTION: During the normal operation of the SV9500 features, situations may arise whereby a Trunk may be connected to an ATTENDANT CONSOLE [A-3]. While the Trunk is connected to the ATTENDANT CONSOLE [A-3] or is in queue at an ATTENDANT CONSOLE [A-3], the Attendant at the Console or an Attendant at any other Console (when multiple Consoles are provided) is not allowed to bridge on the connection. The Trunk must be connected to a Station in a talking mode in order for the ATTENDANT OVERRIDE feature to function.
8. When this feature is used, the ATTENDANT CONSOLE cannot specify a Trunk accommodated in a different Node in the FCCS Network.

Interactions

1. With ATTENDANT OVERRIDE, the following Trunk types may be verified (busy or idle status):
   a.) One-way Incoming Trunks
   b.) One-way Outgoing Trunks
   c.) Bothway Trunk
2. Trunks in a Station-to-Trunk Connection or TRUNK-TO-TRUNK CONNECTION [T-10] can be verified when busy.

3. ATTENDANT LOCKOUT [A-5]: The Attendant is prevented from overriding a busy Trunk which is connected to a Loop held at the Attendant Console. Since the Lockout feature applies only to calls held on an Attendant Console Loop, ATTENDANT OVERRIDE of those Trunks will be denied and the Attendant will hear Reorder Tone.
4. CALL FORWARDING - ALL CALLS [C-5], BUSY [C-2] or DON'T ANSWER [C-3]: If an Attendant attempts to busy verify an idle or busy Station Line that has activated the Call Forwarding feature, busy verification of the Destination Station will occur. If an Attendant attempts to busy verify an idle or busy Station which is Call Forwarded to the Attendant Console, ATTENDANT OVERRIDE will not be allowed.
5. DICTATION ACCESS [D-3]: ATTENDANT OVERRIDE will be denied towards the Dictation Access Trunks.
6. PAGING ACCESS [P-1]: ATTENDANT OVERRIDE is denied to the Paging Access Trunks.
7. DATA PRIVACY - CCIS [D-67] [D-68]: To prevent interruption of Data Transmission, the Attendant cannot override a Trunk connected to a busy Station Line when the user activates this feature.

Programming

STEP 1: **ASYD** - SYS1, Index 78, Bit 0 and Bit 1.
SYS1, Index 60, Bit 5. Is the Two-burst Warning Tone to be disabled? No/Yes.
SYS1, Index 68, Bit 4. Trunk Select (TKSL) key must be enabled. Assign data "1."
SYS2, Index 0, Bit 3, assign as data "1".
SYS1, Index 158, timer between Tones.

STEP 2: **ATRK** - Assign the LENS as Route 909, Digital Conference Trunk. LENS # consists of the following:

XX	X	XX	X
Local Processor	Unit	Group	Level

Group and Level Numbers for the Digital Conference Trunk are 210, 213, 216, 221, 224, 227, 232 and 235.

STEP 3: **ASFC** - For Stations susceptible to ATTENDANT OVERRIDE, assign a Class of Service that allows SFI 4 [Executive Right of Way (Called)], and SFI 11 (Attendant Camp On).

STEP 4: **ARTD** - When a Station is connected to a Trunk, CDN 44:PRV of the route must be assigned as data "0" for ATTENDANT OVERRIDE to be allowed.

Note: To change the data for existing route (RT), initialize or reboot the trunk device such as a circuit card or MG.

FCCS Programming

This data setting describes only the additional or replacement commands used to activate FCCS Services. When used as an FCCS feature, these commands replace the non-network commands (for example, AAED is replaced by AAEDN). Note that the data setting for FCCS Service varies depending on the Node.

STEP 1: **ALRTN** - Assign this command to connect the Logical Route to the Physical Route.

STEP 2: **ARTDN** - When a Station is connected to a Trunk, CDN 44 PRV of the Route must be assigned as data "0" for ATTENDANT OVERRIDE to be allowed.

A-8 Automatic Recall

General Description

This feature works as a timed reminder. When an Incoming Central Office Call (including ISDN) handled by the Attendant remains on hold, camped-on, or ringing unanswered for a fixed interval, the Attendant is automatically alerted.

Operating Procedure

No manual operation is required.

Service Conditions

1. This feature is not available for calls originated from a station since the Attendant cannot blindly transfer them to another station.
2. AUTOMATIC RECALL [A-8] timing is preset in System Data at 30 seconds. See VARIABLE TIMING PARAMETERS [V-1] for other selectable timing choices.
3. If the Attendant reenters a held or Camped-On Trunk, timing is reset when the Trunk is returned to the previous condition.
4. The Attendant Console buzzer will sound to Signal Recall, provided no calls are being processed, in addition to the standard visual indication for Incoming Calls.
5. This feature is denied on INCOMING CENTRAL OFFICE CALL TO TIE LINE CONNECTION [I-3].
6. When an Attendant answers AUTOMATIC RECALL, either the Dialed Number or the Calling Telephone/Trunk Number can be displayed (on System Data basis).
7. The indication on the ATTENDANT CONSOLE [A-3] follows DIGITAL DISPLAY - STATION [D-5]. When the Dialed Number (destination) is displayed, the display is as follows:

Tenant Number:	Blank Space
Trunk Route Restriction Class:	Blank Space
Telephone Number:	Dialed Number (last digits)

8. When the Attendant responds to Automatic Recall, a Three-party conference will be established between the Attendant, the original party and the call forwarding destination if original party and the call forwarding destination are CCIS, ISDN or other trunks. In this case, the Attendant and the original party hear RBT until the call forwarding destination answers.
9. When the Attendant responds to Automatic Recall, the call will be reestablished between the Attendant and the original party. Afterwards, when the Attendant forwards a call and the call forwarding destination answers, the following will occur.

   a) If the call forwarding destination is a station, it will become on hold. The call between the Attendant and the original party remains in progress.

   b) If the call forwarding destination is a trunk, a Three-party conference will be established between the Attendant, the original party and the call forwarding destination.

10. This feature is available on calls extended to a branch System via a CENTRALIZED ATTENDANT SERVICE [C-20] (CAS) position.

11. SYS2, Index 6, Bit 7. On a per Tenant basis, allow this Bit for CALL FORWARD - DON'T ANSWER [C-2] via the Attendant Console. During this operation, the call will ring at the called station for the time interval set in SYS1, Index 141 (ASYD, Index 145, Bit 0 to Bit 3 must be longer than the Index 141). The call is then forwarded and rings at the forwarded station for the time interval set in Index 145. If the call is still not answered, it will be recalled to the Attendant Console via the AUTOMATIC RECALL feature.
12. SYS1, Index 70, Bit 0 (Called Number Display). This Bit does not affect the display on the Attendant Console for recalled calls. It is used for DIDs that have call forwarded to the Attendant or if the Attendant reenters the Loop on an extended call before AUTOMATIC RECALL has taken place.
13. When a Trunk Call recalls back to the Attendant Console, the Destination Telephone Number will display, and the SOURCE (SRC) LED will be lit. By pressing the SRC key, the display will change to the Originating Route and Trunk Number. By pressing the DEST key, the display will change to the destination station.
14. The station will continue to ring even as the call recalls to the Attendant Console. If the station should answer at the same time the Attendant answers the recalling call, a three-way conversation will take place (if Conferences is allowed and a Conference Card is programmed and installed). The Attendant can press the SRC, DEST, or TALK keys to achieve the desired conversation or press the RELEASE key to get out of the conversation.

Programming

ASYD - SYS1, Index 145. Automatic Recall Timer.

Note: When "00 HEX" is assigned to this data, default data is automatically set to 32 seconds. For details, refer to Command Manual.

A-8D Automatic Recall -D

General Description

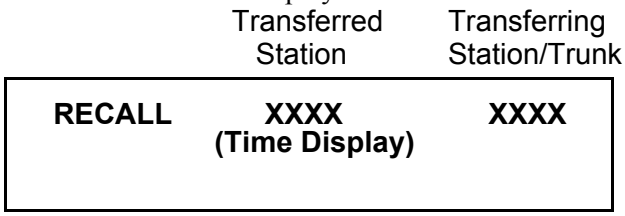
This feature works as a timed reminder. When a Desktop terminal handled call remains on hold (EXCLUSIVE HOLD [E-4D] or NON-EXCLUSIVE HOLD [N-7D]) or during an unattended transfer for a variable, programmable period of time, the Desktop terminal is automatically alerted.

Operating Procedure

No manual operation is required.

Service Conditions

1. AUTOMATIC RECALL -D timing is default in System Data at 30 seconds. See VARIABLE TIMING PARAMETERS [V-1] for other selectable timing choices.
2. If the Desktop terminal reenters a Held or Transferred Call, timing is reset when the call is returned to a RECALL position.
3. When a Held Call returns to the Desktop terminal, a continuous ring of 0.5 seconds ON, 0.5 seconds OFF, and an LED flashing at 240 IPM occurs until the call is retrieved. The upper LCD goes blank during recall. The signal occurs whether the Desktop terminal is on-hook or off-hook.
4. When an unattended transfer returns to the originator of the transfer, the Desktop terminal will ring normally for 30 seconds (VARIABLE TIMING PARAMETERS [V-1]), if the Release Signal is not provided from the distant end. The LCD displays:



Note: When the Telephone Number of the Transferred/Transferring Station exceeds eight digits, the last eight digits are displayed. However, a maximum of 16 digits Transferring Telephone Number can be displayed on the middle line of LCD depending on the System Data.

5. When a Desktop terminal holds a call by NON-EXCLUSIVE HOLD [N-7D] during a THREE-WAY CALL [T-2], the AUTOMATIC RECALL -D feature cannot be activated.

Programming

ASYD - SYS1, Index 144. Recall timing of calls placed on EXCLUSIVE HOLD [E-4D] and NON-EXCLUSIVE HOLD [N-7D]. For normal operation of 30 seconds, leave as data 00H. (RAM Data is 3FH)

SYS1, Index 140. Recall timing of Station transfers.

SYS2, Index 12, Bit 0 - Ringer for NON-EXCLUSIVE, Bit 1 - Ringer for EXCLUSIVE

0: Required

1: Not Required

A-15 Announcement Service

General Description

This feature allows a user to hear a prearranged announcement when the user dials a predetermined Access Code.

Operating Procedure

1. Lift the handset. Receive Dial Tone.
2. Dial the Announcement Trunk Access Code.
3. Listen to the message from the Announcement Machine.

Service Conditions

1. This feature requires one of the following hardware alternatives:
   a.) Central Office Trunk (COT) or 2W E&M Trunk (TLT) and an Announcement Machine.
   b.) Digital Announcement Trunk (DAT).
2. The following connections are available:
   a.) Multi Connection: many users or TIE line trunks can connect to one Announcement Trunk at the same time.
   b.) Single Connection: at any given time, only one Station/Trunk can connect to one Announcement Trunk.
3. This feature is also available for TIE line trunk access to Announcement Trunk.
4. This connection can be continuous until a user hangs up or it can be disconnected after 30 seconds. This function can be selected on a per Announcement Trunk basis.
5. Before connection to the Announcement Machine is made, the system can be programmed to send Ringback Tone.
6. When this feature is activated from a TIE line, an Answer Supervision Signal will be sent to the far end System.
7. ANNOUNCEMENT SERVICE can be provided for a maximum of 127 Trunks, which include the Announcement Trunks for CALL FORWARDING - INTERCEPT / ANNOUNCEMENT [C-25].
8. This feature is also available from the Attendant Console. The Attendant dials the Announcement Trunk Access Code and listens to a message from the Announcement Machine.
9. ANNOUNCEMENT SERVICE is provided to FCCS Nodes that have no Announcement Trunk in the Remote Node.
10. When a call is terminated from the ISDN Trunk or MG(SIP), RBT is sent to the caller if Announcement Trunk Answer Signal is not sent.
11. In AAEDN command, if the checkbox for “Do not send RBT” is OFF (sending RBT), when hearing the announcement from other Node’s VS32 via DTI-FCCS, the announcement starts to play as soon as RBT is sent. Therefore, the announcement heard after sending RBT will have its beginning cut off.
12. This feature is unavailable when a call is terminated via CCIS trunk.
13. If there is a possibility that an Announcement Trunk may be connected to another Announcement Trunk via C.O. line or TIE line, set the parameter C (Duration of connection) of the AAED command to Disconnect in 30 seconds.

Interactions

1. Each Route connected to an Announcement Machine is also an EQP. The same Route can be assigned as different EQPs. However, it is best to separate EQP functions by Route.
2. TLT, COT, or DAT cards may be used.

Programming

STEP 1: **ANPD** - Reserve a Number Level for Trunk Access. Select Normal Service (N) or Hooking Service (H) from the combo box for Connection Indexes. Assign for NND values in accordance with a predetermined Numbering Plan.

STEP 2: **ASPA** - Assign an Access Code to the Announcement Trunk. Assign SRV = "ANNC: Announcement Service - Single Announcement". The EQP Number Assignment should be as assigned in AAED. TYPE = 0/1.

STEP 3: **ARTD** - Assign the Announcement Route as shown below.

RT: 1
1-OSGS :2 2-ONSG :3 5-TF :1 6-TCL :1 7-L/T :1
15-LSG :4

Note: To change the data for existing route (RT), initialize or reboot the trunk device such as a circuit card or VS32.

STEP 4: **ARSC** - Assign a Route Restriction Class (RSC) to the Station that allows access to the Announcement Route. Assign RSC 0 to the Attendant, and allow that RSC Access to the Announcement Route.

STEP 5: **ATRK** - Assign the LENS, Announcement Trunk Route Number. Assign one or more Trunks to a specific Route used to connect to an Announcement Machine.

STEP 6: **MBTK** - Assign the Make Idle status to the Announcement Trunk.

STEP 7: **ARRC** - If TIE line, DID, or REMOTE ACCESS TO SYSTEM [R-2] Connection to the Announcement Service Trunk is required, allow Trunk-to-Trunk Connection using ARI-D-RES = 1 (Direct Connection is allowed.)

STEP 8: **AAED** - This command is used to assign Announcement Equipment.

TN: Tenant Number

EQP: Announcement Equipment Number

RT, TK:Route and Trunk Number of the Trunk connected to the Announcement Equipment.

C: Duration of connection, 0/1:Disconnect in 30 seconds/Connection is held until the Station releases. See the requirements of the application for the Announcement Equipment. For Dead Level applications, data 0 is assigned.

R: Sending RBT 0/1: sending RBT/not sending RBT.

A: Answer Signal Sending, 0/1: no answer from Incoming Trunk/answer from Incoming Trunk. Normally assign data "1".

M: Multiple Connection 0/1: Single Connection/Multiple Connection. Usually assign for Single Connection.

FCCS Programming

This data setting describes only the additional or replacement commands used to activate FCCS Services. When used as an FCCS feature, these commands replace the non-network commands (for example, AAED is replaced by AAEDN). Note that the data setting for FCCS Service varies depending on the Node.

STEP 1: **ALRTN** - Assign this command to connect the Logical Route to the Physical Route.

STEP 2: **ARRCN** - This data must be assigned to the NCN.

Release the restriction between the Terminating Trunk and the Announcement Trunk.

STEP 3: **ARTDN** - Assign the Announcement Route as shown below. Be sure to provide privacy by programming CDN 44:PRV as data "1".

LGRT: 1

1-OSGS	:2	2-ONSG	:3	5-TF	:1	6-TCL	:1	7-L/T	:1
8-RLP	:2	15-LSG	:4	44-PRV	:1				

Note: To change the data for existing route (RT), initialize or reboot the trunk device such as a circuit card or VS32.

STEP 4: **AAEDN** - This data must be assigned to the NCN.

Assign the Announcement Equipment Data.

EQUIPMENT KIND:

- DAT/Announcement Machine
- VS32

TN: Enter Tenant Number

EQP: Announcement Equipment Number

0 = Dead Level

1 = Unused Number

2-15 = Announcement Service

Duration of Connection:

- Disconnect in 30 seconds
- Connection is held until the Station releases. See the requirements of the application for the Announcement Equipment. For Dead Level applications, data 0 is assigned.

Do Not Send RBT [On/Off]

Send Answer Signal [On/Off]

Multiple Connection: Select either Single or Multiple.

LGRT: Logical Route

TK: Trunk Number **Note 1**

STEP 5: **ARSCN** - This data must be assigned to the NCN.

Note 1: Not necessary for Multiple Connection = Single.

Note: The current command data shown is set to all Nodes using the service: ASTD, ARTD, ATRK, MBTK.

A-16 Alternate Routing

General Description

This feature automatically routes Outgoing On-net or Off-net Calls over alternate facilities when the first-choice Trunk Group is busy. The user selects the first-choice Route by dialing the corresponding Access Code, and the equipment then routes the call through Alternate Trunk Groups only if the first is busy. The System will also add or delete digits, when necessary, to complete the call to the desired Station.

Operating Procedure

No manual operation is required.

Service Conditions

1. No indication is provided to a user if the call is routed over first-choice or subsequent choice facilities.
2. STATION MESSAGE DETAIL RECORDING (SMDR) [S-10] may be used with this feature.
3. The same Route can be included in two or more Alternative Groups.
4. The total number of Routes that can be included in a single Route Advance Group is 15.
5. The total number of Routes that can be included in the system is 255.
6. The same Route can be included in two or more Alternative Groups.
7. The maximum number of Trunks to be included in a Route is 255.
8. ALTERNATE ROUTING [A-16] occurs only when the dialed Code accesses the first-choice Trunk Group in the ALTERNATE ROUTING List.
9. The Outgoing Call to the Trunk belonging to the different Node in the FCCS Network is available. For more detailed information, see the Data Programming Manual - FCCS.

Interactions

1. ALTERNATE ROUTING [A-16] may be affected by TOLL RESTRICTION -3/6 DIGIT [T-7]. For example, two Routes of WATS provide the same service but for two separate Route Restriction Classes (RSC). Stations of one RSC will be allowed to advance to the second WATS Route while the other RSC will not.
2. If LEAST COST ROUTING [L-5] is used, ALTERNATE ROUTING is not used. However, they may be made available in the same system.
3. ALTERNATE ROUTING provides for no digit translation, therefore, only Routes of the same type may be used. Keep this in mind when coordinating Route Advance Groups consisting of DDD and FX Routes.

Programming

STEP 1: **ANPD** - Reserve a Number Level for Trunk Access. Select Normal Service (N) or Hooking Service (H) from the combo box for Connection Indexes. Assign NND values in accordance with a predetermined Numbering Plan.

STEP 2: **ASPA** - SRV = “OGCA: Outgoing Call with route advance”, assign Access Code, count number of Routes in a Route Advance Group, and the Route Numbers in the group.

STEP 3: **ARSC** - A Station must be assigned an RSC that will allow access to the Routes in the ALTERNATE ROUTING Route Advance Group.

STEP 4: **AMND** - must be assigned for the number of digits dialed.

FCCS Programming

Refer to the Data Programming Manual - FCCS.

A-17 Audible Indication Control

General Description

This feature allows the Attendant to adjust the audible indications provided to the ATTENDANT CONSOLE [A-3].

Operating Procedure

The Attendant Console is equipped with a Volume Control that can raise or lower the internal speaker level. The Volume Control is located on the left side of the Attendant Console. The audible indication may also be turned OFF when required.

Service Conditions

Each Attendant Console can adjust the audible indications to the ambient noise level, or completely disable audible indication where necessary.

Programming

No programming is required.

A-18 Account Code

General Description

This feature is adjunct to STATION MESSAGE DETAIL RECORDING (SMDR) [S-10], which provides a Station User with the capability to enter a Cost-accounting or Client-billing Code (up to ten digits) into the system before dialing a Long Distance Number.

Operating Procedure

To enter an ACCOUNT CODE:

1. Lift the handset. Receive Dial Tone.
2. Dial the Feature Access Code. Receive Service Set Tone.
3. Dial the ACCOUNT CODE (up to 10 digits).
4. Receive Dial Tone and dial the Called Number.

Note: Feature Access Code - Access Code for ACCOUNT CODE entry will be a maximum of three digits.

To enter an ACCOUNT CODE after an AUTHORIZATION CODE [A-20]:

1. Lift the handset. Receive Dial Tone.
2. Dial the Feature Access Code for AUTHORIZATION CODE and receive Service Set Tone.
3. Dial the AUTHORIZATION CODE (up to 10 digits) and receive Second Service Set Tone.
4. Dial the ACCOUNT CODE and receive Dial Tone.
5. Dial the Called Number.

Service Conditions

1. The maximum number of digits for an ACCOUNT CODE is 10. There is no limitation to the number of Account Codes used per system.

   Note: Service Set Tone after the Feature Access Code can be eliminated by System Data. System Data is common for ACCOUNT CODE, AUTHORIZATION CODE [A-20] and FORCED ACCOUNT CODE [F-7].

2. A Station User can enter an ACCOUNT CODE of fewer digits than the maximum length defined and indicate the end of the entry by pressing the “#” key.
3. The user may decide if ACCOUNT CODE follows the AUTHORIZATION CODE (on a per tenant basis).
4. ACCOUNT CODE dialing is also effective while in CONSULTATION HOLD [C-17]. A Special Dial Tone is received after entering the ACCOUNT CODE.
5. Stations are assigned this feature via Class of Service.
6. ACCOUNT CODEs can be output in the Station Message Detail Recording (SMDR) record for calls handled with OUTGOING TRUNK QUEUING [O-2] and OFF-HOOK QUEUING [O-7].
7. When the Called Number (including Trunk Access Code) is 23 digits or more, the 23rd and subsequent digits will be output to the Account Code Field of Station Message Detail Recording (SMDR).
8. If the Called Number (including Trunk Access Code) with 23 or more digits is dialed after the AUTHORIZATION CODE [A-20], the Calling Party hears Reorder Tone.

Interactions

1. If FORCED ACCOUNT CODEs [F-7] are used, ACCOUNT CODEs are not used.

2. ACCOUNT CODE is also effective while in CONSULTATION HOLD [C-17]. A Special Dial Tone is received after the ACCOUNT CODE.
3. When using this feature on a Single-line Telephone or Desktop terminal, the user can use the SPEED CALLING - SYSTEM, GROUP, INDIVIDUAL or SPEED CALLING - ONE TOUCH (Desktop terminal) to complete the call.
4. ACCOUNT CODES cannot be entered by a Station upon answering an Incoming Trunk Call. The ACCOUNT CODE can be entered by the Attendant Console before extending the call to a Station.
5. The Access Code for ACCOUNT CODEs is restricted to a maximum of three digits.
6. STATION MESSAGE DETAIL RECORDING (SMDR) [S-10] Split Billing (System Data Option): On Transferred Calls, or Incoming Calls via the Attendant Console, the ACCOUNT CODE will be recorded in all the call records.
7. When OUTGOING TRUNK QUEUING [O-2] is initiated, the ACCOUNT CODE will be recorded when the call is completed.
8. When a Desktop terminal user initiates the LAST NUMBER CALLED [L-6D] feature, the ACCOUNT CODE will be recorded, if previously dialed.
9. When a SIP Handler Controlled SIP terminal is used, dial Feature Access Code, Account Code and Called Number by preset dialing. To hear a Service Set Tone (SST) and a Dial Tone (DT), enable the IP-DTG Tone Send Control Bit through the AUACL/AUACN command.

Programming

STEP 1: **ASYD** - SYS1, Index 43, Bit 2. Is Service Set Tone sent after the ASPA Access Code for ACCOUNT CODE? 0/1: No/Yes. Assign data "1".

SYS2, Index 3, Bit 4. Is ACCOUNT CODE used with Authorization Code? 0/1: No/Yes.

SYS1, Index 93, Bit 6=0.

STEP 2: **ANPD** - Reserve a Number Level for Service Feature Access. Select Normal Service (N), Hooking Service (H), or Busy Service (B) from the combo box for Connection Indexes. (applicable to any feature assigned to this level). Assign NND in accordance with a predetermined Numbering Plan.

STEP 3: **ASPA** - For SRV = "SSC: Service Code", SID = 41, assign the Access Code. Assign the number of digits in the ASPA Service Feature Access Code plus the number of digits that will appear in the ACCOUNT CODE for the NND parameter. When ACCOUNT CODE operates with AUTHORIZATION CODE [A-20], the above assignment is omitted. In ASPA, assign for SRV = "SSC: Service Code", SID = 42. The NND value will include the number of digits in the ASPA Service Feature Access Code plus the number of digits that appear in the ACCOUNT CODE.

For example:

45 - 511234 - 617- 555 - 2323

2 + 6 = 8

ACC + ACCT.CODE = NND

For example:

44 - 623423 - 511234 - 617 - 555 - 2323

2 + 0 + 6 = 8

ACC + AUTH CODE + ACCT.CODE = NND

STEP 4: **ASFC** - For Stations to use ACCOUNT CODE, assign a Class of Service that allows SFI 27 (Acct. Codes/Auth. Codes/Forced Acct.Codes).

STEP 5: **AMND** - If Authorization Codes are used with ACCOUNT CODES, then for Tenant 0, assign the Destination Code (DC) as the first digit of all Authorization Codes and the Maximum Necessary Digits (MND) of the Authorization Codes. For the above example, a DC of six is assigned with an MND of six.

STEP 6: **AATC** - Assign when Authorization Code is used.

FCCS Programming

This data setting describes only the additional or replacement commands used to activate FCCS Services. When used as an FCCS feature, these commands replace the non-network commands (for example, AAED is replaced by AAEDN). Note that the data setting for FCCS Service varies depending on the Node.

Note: When using ASYDL or ASYDN to program an FCCS feature, the basic programming of ASYD is required in addition to ASYDL or ASYDN.

STEP 1: **ANPDL** - Reserve a Number Level for Service Feature Access. Select Normal Service (N), Hooking Service (H), or Busy Service (B) from the combo box for Connection Indexes. (applicable to any feature assigned to this level). Assign NND in accordance with a predetermined Numbering Plan.

STEP 2: **ASPAL** - For SRV = "SSC: Service Code", SID = 41, assign the Access Code. Assign the number of digits in the ASPAL Service Feature Access Code plus the number of digits that will appear in the ACCOUNT CODE for the NND parameter. When ACCOUNT CODE operates with AUTHORIZATION CODE [A-20], the above assignment is omitted. In ASPAL, assign for SRV = "SSC: Service Code", SID = 42. The NND value will include the number of digits in the ASPAL Service Feature Access Code plus the number of digits that appear in the ACCOUNT CODE.

A-19 Attendant Night Transfer

General Description

When the ATTENDANT CONSOLE [A-3] is in Night Mode, any operator calls (dial 0 calls) are automatically routed to a predetermined Night Station.

Operating Procedure

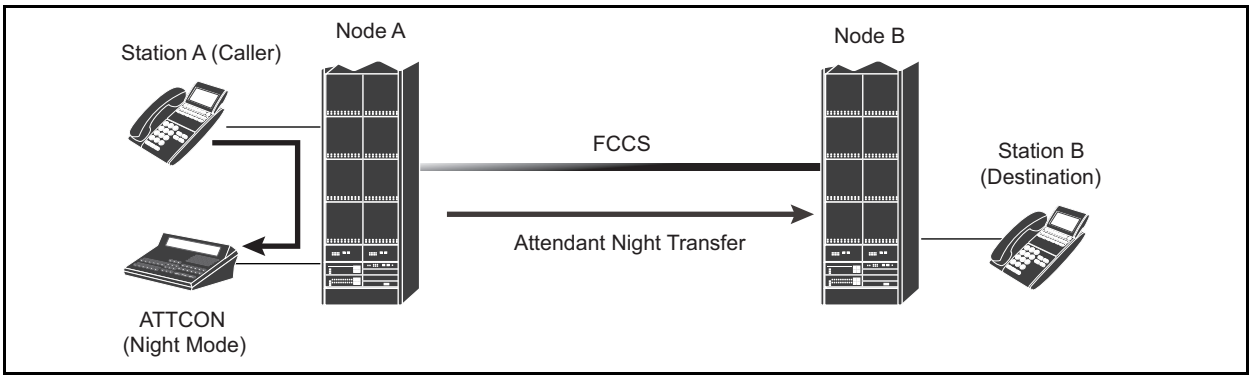
To answer a Night Call

1. The station user dials the operator call code. Receives Ringback Tone.
2. An Operator Call is automatically transferred to the Night Answer Station.
3. The Calling Party hears Ringback Tone.
4. Ringing Signal is sent back to the Night Station.
5. The Night Station goes off-hook to answer the call. If necessary, this station can transfer the call to another station within the system or an outside party, using the CALL TRANSFER - ALL CALLS [C-11] feature, or transfer to an outside party, using TRUNK-TO-TRUNK CONNECTION [T-10] for Incoming Tie Trunk Calls to the operator.

Service Conditions

1. If the Night Station that an Operator Call has been directed to is busy, the Calling Party will receive a Ringback Tone until the Night Station becomes idle.
2. This feature may be provided with NIGHT CONNECTION - FLEXIBLE [N-2] or NIGHT CONNECTION - FIXED [N-1] to a Night Station. The Night Connection Station can also be assigned as a Night Station for ATTENDANT NIGHT TRANSFER.
3. The Night Station can be assigned as a station in a STATION HUNTING [S-7, 8, 9] Group.
4. One Night Station per tenant is available in multi-tenant arrangements.
5. The Night Station can set CALL FORWARDING - ALL CALLS [C-5], CALL FORWARDING - BUSY LINE [C-2], CALL FORWARDING - DON'T ANSWER [C-3] to another station. When the Night Station has set CALL FORWARDING - BUSY LINE [C-2] to the Attendant, the call is placed in a queue until the Night Station becomes idle. When the Night Station has set CALL FORWARDING - DON'T ANSWER [C-3] to the Attendant, the call is not forwarded. When the Night Station has set CALL FORWARDING - ALL CALLS [C-5] to the Attendant, Reorder Tone is sent.
6. PRIORITY CALL [P-9] and OFF-HOOK ALARM [O-6] are not routed by this function.
7. The Night Station can be assigned as a station in a Uniform Call Distribution (UCD) [U-1] Group.
8. If an Incoming Call terminates at the Night Station assigned by NIGHT CONNECTION - FIXED/ FLEXIBLE [N-1, N-2], and the station has set CALL FORWARDING - ALL CALLS [C-5] to the ATTENDANT CONSOLE [A-3], the Incoming Call will be forwarded to a Night Station assigned by ATTENDANT NIGHT TRANSFER [A-19].
9. This operation is not applicable to LDN Calls. Refer to LDN NIGHT CONNECTION [L-28] for more information.
10. Operator Calls from a Station or TIE Line Trunk in a Satellite System (Centralized Attendant Service (CAS) [C-20]) can be transferred to the Night Station.
11. Programmed Night Stations can use CALL HOLD [C-6], CALL TRANSFER [C-10, 11] and THREE-WAY CALLING [T-2], provided these features are programmed in CLASS OF SERVICE - INDIVIDUAL [C-15].

12. Operator calls after Consultation Hold is not transferred to the Night Station.
13. Individual Attendant Calls are not transferred to the Night Station assigned by ATTENDANT NIGHT TRANSFER.
14. The Night Station can set CALL FORWARDING - ALL CALLS - OUTSIDE [C-28], CALL FORWARDING - BUSY LINE - OUTSIDE [C-60], and CALL FORWARDING - DON'T ANSWER - OUTSIDE [C-74].
15. When a DIT call terminates to a Station that has set CALL FORWARDING - ALL CALLS [C-5] to Attendant, the call will be forwarded to a Night Station assigned by this service.
16. In case a Station places an Incoming Call on hold and calls an Attendant, this service is not activated and the Station receives Reorder Tone (including CALL FORWARDING - ALL CALLS [C-5] and CALL FORWARDING - BUSY LINE [C-2]).
17. An Incoming Call is transferred to a Station in the same Node that accommodates the Calling Station/ Trunk in the FCCS Network.
18. A Telephone Number cannot be used, the Physical Station Number must be used.
19. When the Operator Call from PS is not answered within a fixed period of time, some PSs release the call automatically.
20. When originating a call to the Attendant Console where the feature is set (from a Desktop terminal), NXF is displayed on the LCD.
21. For an ISDN Terminal, Phantom Number/Phantom Telephone Number are not available.
22. This feature is available via FCCS. A call to the ATTENDANT CONSOLE [A-3] in Night Mode can be routed to a predetermined Destination Station in the Remote Node.



23. When feature data is mixed on the DM, LDM, and NDM, data registered on the NDM has priority. Data registered on LDM takes precedence over data registered on DM (the priority is NDM⟶LDM⟶DM).

Interactions

1. There can be only one ATTENDANT NIGHT TRANSFER Station per tenant.
2. The ATTENDANT NIGHT TRANSFER Telephone Number must be programmed using the PCPro. This Telephone Number may be the same number as NIGHT CONNECTION - FIXED/FLEXIBLE [N-1, N-2], but each must be individually assigned. PRIORITY CALL [P-9] and OFF-HOOK ALARM [O-6] Calls cannot be answered by the ATTENDANT NIGHT TRANSFER Station.
3. Stations CALL FORWARDING [C-2, C-3, C-5] calls to the Operator (dial 0) will be directed to the ATTENDANT NIGHT TRANSFER Station.
4. CALL WAITING [C-12]: This feature cannot be activated at the ATTENDANT NIGHT TRANSFER Station for Operator Directed (dial 0) Calls.
5. STATION HUNTING [S-7,8,9]: If the ATTENDANT NIGHT TRANSFER Station is in a Station Hunting Group and all Stations in the group are busy, the call will wait in queue at the ATTENDANT NIGHT TRANSFER Station and will be answered by the first available Station in the Hunt Group.

Programming

ASID - Assign the Tenant, Group Number (GN) 1-7, and Function (F) = 3, ATTENDANT NIGHT TRANSFER. Assign the Point Of Destination as a Station. Assign the Station Number.

Note: ASGD assignment is not required. All Stations in the Tenant Group will be affected.

FCCS Programming

This feature is available via FCCS. Data assignment shown below is required.

When Data is registered to Local Data Memory (LDM)
ASIDL -Specify the Destination Telephone Number for Attendant Night Transfer.
TN: Tenant Number
F: Function = 3 (Attendant Night Transfer)
GN: Group Number = 1 (Fixed for the feature)
S/A: Assign “S (Telephone Number of the Station)”
STELN: Special Telephone Number (Destination Telephone Number) **Note 1**

Note 1: This parameter appears when “S” is assigned at S/A parameter.

When Data is registered to Network Data Memory (NDM)
ASIDN -Specify the Destination Telephone Number for Attendant Night Transfer.
TN: Tenant Number
F: Function = 3 (Attendant Night Transfer)
GN: Group Number = 1 (Fixed for the feature)
S/A: Assign “S (Telephone Number of the Station)”
STELN: Special Telephone Number (Destination Telephone Number) **Note 2**

Note 2: This parameter appears when “S” is assigned at S/A parameter.

A-20 Authorization Code

General Description

An AUTHORIZATION CODE is a numerical code dialed by users (up to 10 digits), which will override the Station CLASS OF SERVICE - INDIVIDUAL [C-15] for facility access restriction.

The AUTHORIZATION CODE can be masked on the Desktop terminal display if necessary using the AUTHORIZATION CODE DISPLAY ELIMINATION [A-99].

When a wrong code is received from a Station/Trunk, detailed information on the unauthorized user is output as System Message [26-M]. This message is Toll Fraud Report.

Operating Procedure

There are two procedures for entering an AUTHORIZATION CODE in the System Data base. One of the following choices will be employed:

Procedure 1

- Without an ACCOUNT CODE [A-18]:

1. Lift the handset and receive Dial Tone.
2. Enter the Feature Access Code and receive Service Set Tone.
3. Enter the AUTHORIZATION CODE (up to 10 digits).
4. Receive Dial Tone.
5. Enter the number to be called.

Note: The Feature Access Code for AUTHORIZATION CODE will be a maximum of three digits.

- With an ACCOUNT CODE [A-18]:

1. Lift the handset and receive Dial Tone.
2. Enter the Feature Access Code and receive Service Set Tone.
3. Enter the AUTHORIZATION CODE and receive second Service Set Tone.
4. Enter the ACCOUNT CODE [A-18] and receive Dial Tone.
5. Enter the number to be called.

Procedure 2 (only for LEAST COST ROUTING [L-5] calls)

- Without an ACCOUNT CODE [A-18]:

1. Lift the handset and receive Dial Tone.
2. Enter the number to be called.
3. If an AUTHORIZATION CODE is required, the caller will receive Special Dial Tone.
4. If Special Dial Tone is received, enter the AUTHORIZATION CODE (up to 10 digits) or the call is denied.

- With an ACCOUNT CODE [A-18]:

1. Lift the handset and receive Dial Tone.
2. Enter the Feature Access Code for the ACCOUNT CODE [A-18] and receive Service Set Tone.
3. Enter the ACCOUNT CODE [A-18] and receive Dial Tone.
4. Enter the number to be called.
5. If AUTHORIZATION CODE is required, the caller will receive Special Dial Tone.
6. If Special Dial Tone is received, enter AUTHORIZATION CODE (up to 10 digits) or the call is denied.

Note: The Feature Access Code for ACCOUNT CODE will be a maximum of three digits.

Service Conditions

1. A maximum of 10 digits for AUTHORIZATION CODE can be dialed, and 8 digits will be registered on STATION MESSAGE DETAIL RECORDING [S-10].
2. The maximum number of AUTHORIZATION CODEs is 4,000. This number is dependent upon system configuration.

   These specifications are based on the digits being assigned as follows:

1st digit:	A (when A equals a unique digit from 0 - 9)
2nd digit:	B - C (when b, c equal any two unique digits from 0 - 9)
3rd digit:	D-E (when d, e equal any two unique digits from 0 - 9)
4th digit:	F-G-H (when f, g, h equal any three unique digits from 0 - 9)
5th - 8th digits:	Any combination 0 - 9
9th and 10th digits:	Assigned using the checksum. These are for validation purposes only. See ETI: 091 for more details.

   Note: See ETI: 082 for details on AUTHORIZATION CODE/FORCED ACCOUNT CODE [F-7] assignment.

3. Service Set Tone is common to AUTHORIZATION CODE, ACCOUNT CODE [A-18] and FORCED ACCOUNT CODE [F-7], but can be eliminated by System Data as required.
4. AUTHORIZATION CODE and FORCED ACCOUNT CODE [F-7] cannot be provided simultaneously for the same Tenant Group. System Data determines, on a per Tenant basis, whether AUTHORIZATION CODE or FORCED ACCOUNT CODE [F-7] is used.
5. AUTHORIZATION CODE dialing (Procedure 1) is also effective with CONSULTATION HOLD [C-17]. Special Dial Tone is received instead of Dial Tone after AUTHORIZATION CODE.
6. Procedure 2 is effective only for LEAST-COST ROUTING - 3/6-DIGIT [L-5] calls.
7. Stations must be assigned for Procedure 1 and 2 via CLASS OF SERVICE - INDIVIDUAL [C-5] programming in System Data.
8. In Procedure 1, System Data can determine, on a per Tenant basis, whether or not an ACCOUNT CODE [A-18] follows the AUTHORIZATION CODE.
9. Assigned AUTHORIZATION CODEs can be used for the REMOTE ACCESS TO SYSTEM [R-2] function.
10. OG TRUNK QUEUING [O-2] and OFF-HOOK QUEUING [O-7] can be set according to the Service Feature Restriction Class of the user's AUTHORIZATION CODE.
11. AUTHORIZATION CODE can be recorded in the STATION MESSAGE DETAIL RECORDING [S-10] for Procedure 2.
12. The registered/non-registered condition of the AUTHORIZATION CODE is not included in the information block of Condition B on the SMDR output format.
13. Attendant cannot activate this feature by Procedure 1.
14. When the user goes on-hook or hooking, changed class goes back to normal class.
15. For service conditions for the AUTHORIZATION CODE DISPLAY ELIMINATION [A-99].
16. When originating a call using AUTHORIZATION CODE from a Station whose SFC is allowed to use PAD LOCK [P-57], the temporarily used RSC is retained even if the user goes on-hook.

Interactions

1. If FORCED ACCOUNT CODEs [F-7] are used, AUTHORIZATION CODEs are not used within the same tenant.
2. AUTHORIZATION CODEs may be used in conjunction with ACCOUNT CODEs [A-18].
3. When assigning an AUTHORIZATION CODE Restriction of 1 in AATC, the RSC must be constructed in ARSC.

4. AUTHORIZATION CODEs can be unrestricted or assigned one of 15 Route Restriction Classes (RSC) - 15.
5. There are no restrictions regarding the first digit of AUTHORIZATION CODEs, except what is programmed in AMND. Tenant 0 does not conflict with the System Numbering Plan. This is true when separate tenant tables are provided. If common tenant tables are used, a conflict between LCR(S) and OGC(A), used with SMDR, will occur.

   Note: The Access Code to the feature AUTHORIZATION CODE must be assigned within the System Numbering Plan.

6. If ASPA, SRV = "LCR: Least Cost Routing" / "LCRS: Register Sender LCR" AH parameter is assigned as data "1", the request for an AUTHORIZATION CODE is determined by the ARSC assignment of the Station in relationship to the chosen route.
7. When a Desktop terminal user initiates the LAST NUMBER DIALED [L-6D] feature, the AUTHORIZATION CODE will be recorded if previously dialed.
8. The AMND command determines the length of AUTHORIZATION CODEs, based on the first digit of the code.
9. AUTHORIZATION CODEs are recorded by STATION MESSAGE DETAIL RECORDING [S-10], Procedure 2.
10. AUTHORIZATION CODEs cannot be entered by the Attendant Console.
11. PEG COUNTS [P-7] of invalid AUTHORIZATION CODEs are not available.
12. AUTHORIZATION CODEs can be dialed by a user when establishing a CONSULTATION HOLD [C-17] connection.
13. For Procedure 2, LCR(S) must be provided.

Programming

Note: There are two methods of Authorization Code assignment: Full Implementation and the Checksum method. In the Full Implementation method, the entire AUTHORIZATION CODE is assigned in the AATC command and ASYD, SYS1. Index 12, Index 13, and Index 14 are assigned 00H. In the Checksum method, partial assignment for each AUTHORIZATION CODE is assigned in the AATC command. The last one or two digits (programmable option) are determined by the system according to the data assignments in ASYD, SYS1, Index 12, Index 13, and Index 14.

STEP 1: **ASYD** - SYS1, Index 4, Bit 4. Temporary Class Conversion. Assign as data "1". SYS1, Index 12. If Index 12=0, no Checksum method is used, and Index 13 and Index 14 are ignored, regardless of their assignments. Each Bit in Index 12 directly correlates to a digit in the AUTHORIZATION CODE assigned in AATC command, and will determine the first digit of the Checksum method. For this example, 26428 has been assigned as the AUTHORIZATION CODE in AATC. Arbitrarily pick any Bit 0 to Bit 4, and flag those Bits as data 1.

Note: Bit 0 to Bit 7 may be used when eight-digit AUTHORIZATION CODEs are assigned in the AATC command.

	Bit 4	Bit 3	Bit 2	Bit 1	Bit 0
AATC	8	2	4	6	2

	Bit 4	Bit 3	Bit 2	Bit 1	Bit 0
SYS1, Index 12	0	1	0	0	1
Designation	d4	d3	d2	d1	d0

SYS1, Index 13. Each Bit in Index 13 directly correlates to a digit in the AUTHORIZATION CODE assigned in AATC command, and will determine the second digit of the Checksum method. Pick any Bit 0 to Bit 4, but not the same combination of Bits as picked in Index 12.

	Bit 4	Bit 3	Bit 2	Bit 1	Bit 0
AATC	8	2	4	6	2
SYS1, Index 13	1	0	0	1	0
Designation	d4	d3	d2	d1	d0

SYS1, Index 14 determines the algorithm used to calculate the last two digits of every AUTHORIZATION CODE. (Only decimal numbers may be used in this index, therefore, hexadecimal values A to F may not be assigned). SYS1, Index 14 = X and Y, where X and Y represent any arbitrarily chosen decimal digit 1- 9, and where Y is used to determine the first digit and X is used to determine the second digit in the Checksum method. The following data, 37H, has been used for the example calculation:

	X				Y			
SYS1, Index 14	**Bit 7**	**6**	**5**	**4**	**3**	**2**	**1**	**0**
DATA	0	0	1	1	0	1	1	1

Note: If only one digit is to be added via the Checksum method, then Index 13 data will be 00h and Index 14, digit X (Bit 4 to Bit 7) = 0.

Example Calculation:

In AATC command, 26428 has been assigned. In SYS1, Index 12, Bit 0 and Bit 3 were chosen; 26428

2 + 2 = 4.

The decimal value of Bit 0 to Bit 3 of Index 14 = 7.

4 + N =7. N = 3.

7 is obtained as any whole number greater than 4 that ends with a 7, the decimal value as determined in SYS1, Index 14, Bit 0 to Bit 3. In AATC, 26428 has been assigned. In SYS1, Index 13, Bit 1 and Bit 4 were chosen; 26428.

8 + 6 = 14.

The decimal value of Index 14, Bit 4 to Bit 7 is 3.

14 + N = 23. N = 9.

23 is obtained as any whole number greater than 14 that ends with a 3, the decimal value determined in SYS1, Index 14, Bit 4 to Bit 7. The AUTHORIZATION CODE will be 26428-39.

Note: Digits 39 determined by the Checksum method will be unique to the AUTHORIZATION CODE 26428. 26428 will not be followed by any digits other than 39.

SYS1, Index 43, Bit 2, is Service Set Tone sent after AUTHORIZATION CODE. ASPA Access Code? 0/1: No/Yes. Assign data "1".

SYS1, Index 43, Bit 0, is AUTHORIZATION CODE required for DISA? 0/1: required/not required. Normally assign data "0."

SYS1, Index 93, Bit 6. Are the AMND tables Separate or Common for tenants in the system? 0/1: Separate/Common. Assign data "0" for separate tables.

SYS2, Index 3, Bit 4: Is Account Code used with AUTHORIZATION CODE? 0/1: No/Yes.

SYS2, Index 3, Bit 5: Type of code? 0/1: AUTHORIZATION CODE/ Forced Account Code; assign data "0".

STEP 2: **ANPD** - Reserve a Number Level for feature access. Select Normal Service (N), Hooking Service (H), or Busy Service (B) from the combo box for Connection Indexes. (if applicable to any feature in this level). Assign NND in accordance with a predetermined Numbering Plan.

STEP 3: **ASPA** - For SRV = "SSC: Service Code", SID = 42, assign the Access Code. For the NND parameter, no assignment is required when AUTHORIZATION CODE operates by itself. When Account Code operates with AUTHORIZATION CODE, assign for SRV = "SSC: Service Code", SID = 42. The NND value will include the number of digits in the AUTHORIZATION CODE Access Code, and Account Code. For example:

45	–	26428	–	623423	–	9-1-617-555-2323
(ACC)		(Auth Code)		(Account Code)		(Destination Code)
ASPA		ACC=45				NND=8 (ACC + Account Code)

For SRV = "LCR: Least Cost Routing" / "LCRS: Register Sender LCR", if operating Procedure 2 is to be used, (dialing the AUTHORIZATION CODE after the number dialed), then assign data "1" to the AH parameter when programming the LCR(S) Access Code. Therefore, operating Procedure 2 is only available with LEAST-COST ROUTING - 3/6 DIGIT [L-5].

STEP 4: **ASFC** - For Stations allowed AUTHORIZATION CODE, assign a Class of Service that allows SFI 27.

STEP 5: **AMND** - For Tenant 0, assign the Destination Code (DC) as the first digit of all AUTHORIZATION CODEs and the Maximum Necessary Digits (MND) of the AUTHORIZATION CODE. For the above example, a DC of 2 is assigned with an MND of 7. The MND value will include the digits added by the Checksum method.

STEP 6: **AATC** - Assign the TN, Digit Code (DC) (the actual code itself), the AUTHORIZATION CODE RESTRICTION (ACR) as data "1" and Service Feature Restriction Class (SFC) and Route Restriction Class (RSC). If ACR is assigned as data 2. Allowed: RSC and SFC parameters are not programmed. (SFC parameters are not operational.)

STEP 7: **LATC** - List up all the AUTHORIZATION CODEs. This is the simplest way to determine what digits have been added to the system with the Checksum method. This is not required if the Checksum method is not used.

FCCS Programming

This data setting describes only the additional or replacement commands used to activate FCCS Services. When used as an FCCS feature, these commands replace the non-network commands (for example, AAED is replaced by AAEDN). Note that the data setting for FCCS Service varies depending on the Node.

Note: When using ASYDL or ASYDN to program an FCCS feature, the basic programming of ASYD is required in addition to ASYDL or ASYDN.

STEP 1: **ANPDL** - Reserve a Number Level for feature access. Select Normal Service (N), Hooking Service (H), or Busy Service (B) from the combo box for Connection Indexes. (if applicable to any feature in this level). Assign NND in accordance with a predetermined Numbering Plan.

STEP 2: **ASPAL** - For SRV = "SSC: Service Code", SID = 42, assign the Access Code. For the NND parameter, no assignment is required when AUTHORIZATION CODE operates by itself. When Account Code operates with AUTHORIZATION CODE, assign for SRV = "SSC: Service Code", SID = 42. The NND value will include the number of digits in the AUTHORIZATION CODE Access Code, and Account Code.

For SRV = "LCR: Least Cost Routing" / "LCRS: Register Sender LCR", if operating Procedure 2 is to be used, (dialing the AUTHORIZATION CODE after the number dialed), then assign data "1" to the AH parameter when programming the LCR(S) Access Code. Therefore, operating Procedure 2 is only available with LEAST-COST ROUTING - 3/6 DIGIT [L-5].

STEP 3: **LATCN** - List up all the AUTHORIZATION CODEs. This is the simplest way to determine what digits have been added to the system with the Checksum method. This is not required if the Checksum method is not used.

A-29 Authorization Code - Tie Line Incoming Routes

General Description

This feature offers an Outside Party the ability to enter an AUTHORIZATION CODE [A- 20] through an incoming TIE Line Trunk.

Note: This feature is available in North America only.

Operating Procedure

When the incoming route is restricted or when an AUTHORIZATION CODE [A-20] is always required:

1. The Called Number, office code, and terminal number are received by the TIE Line Trunk.
2. The outside Calling Party hears a Special Dial Tone.
3. Dial the AUTHORIZATION CODE [A-20].
   a.) The connection is established when the AUTHORIZATION CODE [A-20] is valid.
   b.) Busy Tone is sent to the Calling Party when the AUTHORIZATION CODE [A-20] is invalid.

Service Conditions

1. The outside Calling Party must be able to generate DTMF Signals to the System when entering AUTHORIZATION CODE [A-20].
2. An AUTHORIZATION CODE [A-20] can be used for Incoming Calls to a uniform Station or for LEAST-COST ROUTING [L-5] tandem connections.
3. CALL WAITING - TERMINATING [C-12] is available when the Called Station is busy, after the AUTHORIZATION CODE [A-20] is entered and validated.
4. INTER-OFFICE OFF-HOOK QUEUING [I-11] is available if the called route is busy, after the AUTHORIZATION CODE is entered and validated.
5. Through programming, the system defines the AUTHORIZATION CODE [A-20] to be required for all TIE Line Incoming Calls or when a restriction is encountered. This is assigned on a per Tenant basis.

Note: There are two methods of AUTHORIZATION CODE assignment - Full Implementation and the Checksum. In the Full Implementation method, the entire AUTHORIZATION CODE is assigned in the AATC command, and SYS1, Index 12, Index 13 and Index 14 are assigned 00H. In the Checksum method, partial assignment for each AUTHORIZATION CODE is assigned in the AATC command. The last one or two digits (programmable option) are determined by the system according to the data assignments in ASYD, SYS1, Index 12, Index 13 and Index 14.

Interactions

1. If FORCED ACCOUNT CODEs [F-7] are used, AUTHORIZATION CODEs-TIE LINE INCOMING ROUTES are not allowed within the same tenant.
2. AUTHORIZATION CODEs-TIE LINE INCOMING ROUTES may be used in conjunction with ACCOUNT CODEs [A-18].
3. When assigning an AUTHORIZATION CODE Restriction of 1 in AATC, the RSC must be constructed in ARSC.
4. AUTHORIZATION CODEs can be unrestricted or assigned one of 15 Route Restriction Classes (RSC 1 to 15).

5. There are no restrictions regarding the first digit of AUTHORIZATION CODEs, except what is programmed in AMND. Tenant 0 does not conflict with the System Numbering Plan.

   Note: The Access Code to the feature AUTHORIZATION CODE - TIE LINE INCOMING ROUTES must be assigned within the System Numbering Plan.

6. If ASPA, SRV = "LCR: Least Cost Routing" / "LCRS: Register Sender LCR" AH parameter is assigned as data "1", the request for an AUTHORIZATION CODE is determined by the ARSC assignment of the Station in relationship to the route chosen. For AUTHORIZATION CODE - TIE LINE INCOMING ROUTES, no RSC is assigned against the incoming TIE Line, therefore, AUTHORIZATION CODEs are always required. Trunk connection will be determined against the RSC of the dialed code.
7. When a Desktop terminal user initiates the LAST NUMBER CALLED -D [L-6D] feature, the AUTHORIZATION CODE will be recorded if previously dialed.
8. The AMND command determines the length of AUTHORIZATION CODEs based on the first digit of the code.
9. AUTHORIZATION CODEs are recorded by STATION MESSAGE DETAIL RECORDING [S-10].
10. AUTHORIZATION CODEs cannot be entered by the Attendant Console.
11. PEG COUNTS [P-7] of invalid AUTHORIZATION CODEs are not available.
12. AUTHORIZATION CODEs can be dialed by a user, when establishing a CONSULTATION HOLD [C-17] connection.
13. For Procedure 2, LCR(S) must be provided.
14. For senderized TIE Lines, see UNIVERSAL SENDER [U-3]. Also, the ANND command must be programmed for senderized TIE Lines. If other services, such as, STATION MESSAGE DETAIL RECORDING [S-10] and LEAST-COST ROUTING [L-5], the AMND command will be programmed. If digit translation is required, go to the ASTP and AOSP command.
15. For MISCELLANEOUS TRUNK RESTRICTION [M-3], restrictions to TIE trunks are determined by the Route Service Class (RSC) assigned to the AUTHORIZATION CODE.
16. When the incoming route passes through MG-SIPs, this feature cannot be used.

Programming

STEP 1: **ASYD** - SYS1, Index 4, Bit 4=1: Temporary Class Conversion is required.

SYS1, Index 12. If Index 12=0, no Checksum method is used, and Index 13 and Index 14 are ignored, regardless of their assignments. Each Bit in Index 12 directly correlates to a digit in the AUTHORIZATION CODE assigned in AATC, and will determine the first digit of the Checksum method. For this example, 26428 has been assigned as the AUTHORIZATION CODE in AATC. Pick any Bit 0 to Bit 4, and flag those Bits as data 1.

Note: Bit 0 to Bit 7 may be used when eight-digit AUTHORIZATION CODEs are assigned in the AATC command

	Bit 4	Bit 3	Bit 2	Bit 1	Bit 0
AATC	8	2	4	6	2
SYS1, Index 12	0	1	0	0	1
Designation	d4	d3	d2	d1	d0

SYS1, Index 13. Each Bit in Index 13 directly correlates to a digit in the AUTHORIZATION CODE

assigned in AATC command, and will determine the second digit of the Checksum method. Pick any Bit 0 to Bit 4, but not the same combination of Bits as picked in Index 12.

	Bit 4	Bit 3	Bit 2	Bit 1	Bit 0
AATC	8	2	4	6	2
SYS1, Index13	1	0	0	1	0
Designation	d4	d3	d2	d1	d0

SYS1, Index 14 determines the algorithm used to calculate the last two digits of every AUTHORIZATION CODE. (Only decimal numbers may be used in this index, therefore, hexadecimal values A to F may not be assigned).

SYS1, Index 14=X and Y, where X and Y represent any arbitrarily chosen decimal digit 1- 9, and where Y is used to determine the first digit, and X is used to determine the second digit in the Checksum method. The following data, 37H, has been used for the example calculation:

SYS1, Index 14	X				Y			
	Bit 7	6	5	4	3	2	1	0
DATA	0	0	1	1	0	1	1	1

Note: If only one digit is to be added via the Checksum method, then SYS1, Index 13 data will be 00h and Index 14, digit X (Bit 4 to Bit 7) = 0.

Example Calculation:

In AATC command, 26428 has been assigned. In SYS1, Index 12, Bit 0 and Bit 3 were chosen; 26428

2 + 2 = 4.

The decimal value of Bit 0 to Bit 3 of Index 14 = 7.

4 + N =7. N = 3.

7 is obtained as any whole number greater than 4 that ends with a 7, the decimal value as determined in SYS1, Index 14, Bit 0 to Bit 3. In AATC command, 26428 has been assigned. In SYS1, Index 13, Bit 1 and Bit 4 were chosen; 26428

8 + 6 = 14.

The decimal value of Index 14, Bit 4 to Bit 7, is 3.

14 + N = 23. N = 9.

23 is obtained as any whole number greater than 14 that ends with a 3, the decimal value determined in SYS1, Index 14, Bit 4 to Bit 7. The AUTHORIZATION CODE will be 26428-39.

Note: Digits 39, determined by the Checksum method, will be unique to the AUTHORIZATION CODE 26428. 26428 will not be followed by any digits other than 39.

SYS1, Index 43, Bit 2, is Service Set Tone sent after AUTHORIZATION CODE

ASPA Access Code? 0/1: No/Yes. Assign data “1”. SYS1, Index 93, Bit 6. Are the AMND tables Separate or Common for tenants in the system? 0/1: Separate/Common. Assign data “0” for separate tables.

SYS2, Index 3, Bit 4: Is Account Code used with AUTHORIZATION CODE? 0/1: No/Yes.

SYS2, Index 3, Bit 5: Type of code: 0/1 AUTHORIZATION CODE/ Forced Account Code; assign data “0”.

STEP 2: **ANPD** - Reserve a Number Level for feature access. Select Normal Service (N), Hooking Service (H), or Busy Service (B) from the combo box for Connection Indexes. (if applicable to any feature in this level). Assign NND in accordance with a predetermined Numbering Plan.

STEP 3: **ASPA** - For SRV = "SSC: Service Code" SID = 42, assign the Access Code. For the NND parameter, no assignment is required when AUTHORIZATION CODE [A-20] operates by itself. When Account Code [A-18] operates with AUTHORIZATION CODE - TIE LINE INCOMING ROUTES, assign for SRV = "SSC: Service Code", SID = 42. The NND value will include the number of digits in the AUTHORIZATION CODE Access Code, and Account Code.

Note: Procedure 1 is not available with AUTHORIZATION CODE - TIE LINE INCOMING ROUTES. For example:

45 -	*26428 -*	*623423 -*	*9-1-617-555-2323*
(ACC)	*(AUTH CODE)*	*(ACCOUNT CODE)*	*(DESTINATION CODE)*
ASPA	*ACC = 45*		*NND = 8 (ACC+ACCOUNT CODE)*

For SRV = "LCR: Least Cost Routing" / "LCRS: Register Sender LCR", if operating Procedure 2 is used, (dialing the AUTHORIZATION CODE after the number dialed), assign data "1" to the AH parameter when programming the LCR(S) Access Code. Therefore, operating Procedure 2 is only available with LEAST-COST ROUTING - 3/6 DIGIT [L-5].

STEP 4: **ASFC** - For Stations allowed AUTHORIZATION CODE - TIE LINE INCOMING ROUTES, assign a Class of Service that allows SFI 27.

STEP 5: **AMND** - For Tenant 0, assign the Destination Code (DC) as the first digit of all AUTHORIZATION CODEs and the Maximum Necessary Digits (MND) of the AUTHORIZATION CODE. For the above example, a DC of 2 is assigned with an MND of 7. The MND value will include the digits added by the Checksum method.

STEP 6: **AATC** - Assign the TN, Digit Code (DC) (the actual code itself), the AUTHORIZATION CODE Restriction (ACR) as data "1", and Service Feature Restriction Class (SFC) and Route Restriction Class (RSC). If ACR is assigned as data 2, Allowed: RSC and SFC parameters are not programmed. (SFC parameters are not operational.)

STEP 7: **LATC** - List all the AUTHORIZATION CODEs. This is the simplest way to determine what digits have been added to the system with the Checksum method. This is not required if the Checksum method is not used.

TIE Line Programming

STEP 8: **ARTD** - Assign the TIE Line routes as shown below:

RT:1

1-OSGS	:2	2-ONSG	:3	5-TF	:1	6-TCL	:1	7 L/T	:1
8-RLP	:2	15-LSG	:1						

Note: To change the data for existing route (RT), initialize the circuit card.

STEP 9: **ATRK** - Assign the LENS, TIE Line Route Number, Trunk Number, and Tenant Number.

STEP 10: **MBTK** - Assign the Make Idle status to the TIE Line Trunks.

STEP 11: **ANPD**- Assign a Number Level for Trunk Access. Select Normal Service (N) or Hooking Service (H) from the combo box for Connection Indexes. Number of Necessary Digits is usually 1 or 2. Busy Lamp Field is not activated.

STEP 12: **ASPA** - Assign the Access Code as assigned in ANPD. Type of Service, SRV = "OGC: Outgoing Call" for Outgoing Trunk. Assign the Route Number associated with this Access Code.

STEP 13: **ARSC** - The Route Restriction Class (RSC) must be constructed to allow the Station access to the involved routes.

FCCS Programming

This data setting describes only the additional or replacement commands used to activate FCCS Services. When used as an FCCS feature, these commands replace the non-network commands (for example, AAED is replaced by AAEDN). Note that the data setting for FCCS Service varies depending on the Node.

STEP 1: **ANPDL** - Reserve a Number Level for feature access. Select Normal Service (N), Hooking Service (H), or Busy Service (B) from the combo box for Connection Indexes. (if applicable to any feature in this level). Assign NND in accordance with a predetermined Numbering Plan.

STEP 2: **ASPAL** - For SRV = "SSC: Service Code", SID = 42, assign the Access Code. For the NND parameter, no assignment is required when AUTHORIZATION CODE [A-20] operates by itself. When Account Code [A-18] operates with AUTHORIZATION CODE - TIE LINE INCOMING ROUTES, assign for SRV = "SSC: Service Code", SID = 42. The NND value will include the number of digits in the AUTHORIZATION CODE Access Code, and Account Code.

Note: Procedure 1 is not available with AUTHORIZATION CODE - TIE LINE INCOMING ROUTES.

For SRV = LCR (Least Cost Route Selection) / LCRS (Register Sender LCR), if operating Procedure 2 is used, (dialing the AUTHORIZATION CODE after the number dialed), assign data "1" to the AH parameter when programming the LCR(S) Access Code. Therefore, operating Procedure 2 is only available with LEAST-COST ROUTING - 3/6 DIGIT [L-5].

STEP 3: **LATCN** - List all the AUTHORIZATION CODEs. This is the simplest way to determine what digits have been added to the system with the Checksum method. This is not required if the Checksum method is not used.

TIE Line Programming

STEP 4: **ARTDN** - Assign the TIE Line routes as shown below:

RT:1

1-OSGS	:2	2-ONSG	:3	5-TF	:1	6-TCL	:1	7 L/T	:1
8-RLP	:2	15-LSG	:1						

Note: To change the data for existing route (RT), initialize the circuit card.

STEP 5: **ANPDL** - Assign a Number Level for Trunk Access. Select Normal Service (N) or Hooking Service (H) from the combo box for Connection Indexes. Number of Necessary Digits is usually 1 or 2. Busy Lamp Field is not activated.

STEP 6: **ARSCN** - The Route Restriction Class (RSC) must be constructed to allow the Station access to the involved routes.

Note: When using ASYDL or ASYDN to program an FCCS feature, the basic programming of ASYD is required in addition to ASYDL or ASYDN.

A-30 Automatic/Manual Intercom

General Description

This feature permits multiple Desktop terminal users to simultaneously call all the Stations in a pre-assigned group, regardless of their idle/busy status. An INTERCOM member can optionally override another INTERCOM Station in a Two-party INTERCOM Connection. This service is referred to as INTERCOM BRIDGE. INTERCOM Service is separated into two types, AUTOMATIC INTERCOM (AICM) and MANUAL INTERCOM (MICM).

Automatic Intercom

All Stations in the INTERCOM Group are alerted simultaneously with a lamp indication. Ringing is directed to one predefined Station, as in a button and buzzer INTERCOM arrangement.

Manual Intercom

All Stations in the INTERCOM group are called simultaneously with a lamp indication. The Calling INTERCOM Station manually sends a one-second tone to the Called INTERCOM Signal Station, as in a button and buzzer INTERCOM arrangement.

Operating Procedure

- AUTOMATIC INTERCOM

To initiate an Automatic Intercom Call

1. The INTERCOM Station presses the AICM key and goes off-hook (within five to ten seconds). **Note 1**
2. The INTERCOM Station will hear Ringback Tone. A specified Station rings.
3. One of the Called INTERCOM Group Stations answers the call and is connected to the Calling Station.

To answer an Automatic Intercom Call

1. The AICM key will flash, indicating an Incoming INTERCOM Call. The Called Station will also ring according to the pre-assigned system parameters.
2. If the Called Station is already engaged in a conversation with a Non-Intercom Call, the Station may press the AICM key after pressing the HOLD key. Ringing will stop.

To bridge into an Automatic Intercom Call

1. When two Stations are engaged in an INTERCOM Call, a third Station in the same INTERCOM Group presses the AICM key and goes off-hook. **Note 1**
2. A Three-party Conference is established. The Called Stations may hear a Warning Tone when the Station bridges into the call. This Warning Tone is pre-assigned in System Data.

- MANUAL INTERCOM

To originate a Manual Intercom Call

1. A Station presses the MICM key and goes off-hook (within five seconds). **Note 1**
2. The Station will hear Ringback Tone.
3. The Calling Station sends a chime to the INTERCOM Station by pressing the SIG key.
4. When one of the Called INTERCOM Stations answers the call, the Calling Station will be connected. Also, all Stations in the group share the common talk path and can pick up an unanswered INTERCOM Call.

To answer a Manual Intercom Call

1. The MICM key will flash, indicating an Incoming INTERCOM Call. The Called Station may also hear

one second of ringing from the Calling INTERCOM Station.

2. If the Called Station in the INTERCOM Group is already engaged in a conversation with a Non-Intercom Call, the Station may press the MICM key after pressing the HOLD key.

To bridge into a Manual Intercom Call

1. When two Stations are engaged in an INTERCOM Call, a third Station, belonging to the same INTERCOM Group, will press the MICM key and go off-hook.**Note 1**
2. A Three-party Conference is established. The Called Stations may hear a Warning Tone when the Station bridges into the call. This Warning Tone is pre-assigned in System Data.

Note 1: When ASYD, SYS1, Index 384, Bit 0=0 is set, you can skip off-hook operation to initiate Automatic Intercom Call or originate Manual Intercom Call. However, off-hook operation cannot be skipped to bridge into an Automatic Intercom Call or a Manual Intercom Call regardless of the ASYD, SYS1, Index 384, Bit 0 data setting.

Use of the SIG key

1. The Manual INTERCOM Signal (SIG) key is operational at all times. An INTERCOM Call is not required. If the Signaled Station is ringing from another call, the Manual INTERCOM Signal will interrupt that ringing.

- PRIME-LINE PICK UP -D [P-13D]

  The Desktop terminal may be programmed to seize the Incoming Call on the AUTOMATIC, MANUAL, or DIAL INTERCOM [D-87] Lines before the Prime Line. If two Incoming Calls terminate simultaneously to the Prime Line and the MICM Line, when the user goes off-hook, PRIME LINE PICK UP -D [P-13D] will seize the MICM line before the Prime Line. If only the Prime Line is ringing when the user goes off-hook, PRIME LINE PICK UP -D [P-13D] will seize the Prime Line.

Service Conditions

1. Only Desktop terminals can be assigned as a member of an INTERCOM Group.
2. All members of the same INTERCOM Group must be assigned in the same Module Group (MG).
3. Up to 16 Stations may be assigned to one INTERCOM Group. Each Station in the INTERCOM Group can be assigned as both a Manual and Automatic INTERCOM member of two different groups.
4. No more than one connection can be established at the same time in one INTERCOM Group.
5. A Three-party Conference can be established in an INTERCOM Group if INTERCOM Bridge Service is available. If a fourth party tries to activate INTERCOM Bridge Service, the connection will be denied.
6. A maximum of 50 (per MG) Automatic or Manual INTERCOM Groups are allowed.
7. It is possible to assign one group of Stations as an AICM Group, Dial INTERCOM Group, and an MICM Group at the same time.
8. The DISTINCTIVE RINGING [D-10] sent from an Incoming AICM Station takes priority over the ringing of Multi Line Appearances.
9. Features normally available to Multi Line Appearances are not available for INTERCOM Lines.
10. INTERCOM Lines cannot be held with the activation of the HOLD key.
11. When MANUAL SIGNALING [M-47] is performed by a user, this ringing (one second) takes priority over any other Incoming Call indications from the Multi Line Appearance.
12. When a Station is moved from one INTERCOM Group to another INTERCOM Group, through system commands, the AICM or MICM key on the associated Desktop terminal must be canceled before the change is made.

13. Port traffic and Station PEG COUNT [P-7] reports will not record the use of the INTERCOM feature.
14. Flex keys on the Desktop terminal cannot be used while on an INTERCOM Call. Only the SPEAKER key can be used for hands-free operation.
15. FCCS is not available for this feature.
16. On SIP Multiple Line station, the pattern of Ringback Tone is the same as Ringing Tone.
17. The AICD command must be assigned before the AKYD command.
18. No programming is required to enable the SIG key to remain active at all times.

Interactions

1. The RECALL key will not operate on an INTERCOM Call.
2. THREE-WAY CALLING [T-6] must first be provided.
3. When assigning the PRI parameter in AKYD, the following data selections are available.

Priority PRI	1	2	3
0	Prime Line	-	-
1	Dial Intercom	Prime Line	-
3	Automatic Intercom	Prime Line	-
3	Manual Intercom	Prime Line	-
4	Automatic Intercom	Manual Intercom	Prime Line

4. The SIG key, when pressed by a Calling Station, will interrupt other Ringing Signals at the Called Station.
5. Hands-free operation is available with AUTOMATIC/MANUAL INTERCOM. When an Incoming Call terminates to a Station, by pressing the SPEAKER key, the line seized is based on the PRI assignment of AKYD.
6. MANUAL SIGNALING [M-47] will not operate if the DATA key has been pressed.

Programming

STEP 1: **ASYD** - SYS1, Index 59, Bit 7. Is AICM/MICM in service? 0/1: No/Yes.

SYS1, Index 69, Bit 6. Is a Warning Tone required when creating a three-way AICM/MICM Call? 0/1: No/Yes.

SYS3, Index 3. Assign the data 20H for Multiple Ringings per Ringing Period (for Automatic Intercom) if necessary.

SYS3, Index 7. Assign the Ringing Pattern (for Automatic Intercom) if necessary.

STEP 2: **AICD** - Assign the following data:

MG: Module Group

G-ID: Group ID Number per MG

A/M: Automatic/Manual Signal (0/1)

- A: Automatic
- M: Manual

BCR: Bridge Call Restriction (0/1)

- 0: Restricted
- 1: Allowed

For example, Station 1000 is to initiate AUTOMATIC/MANUAL INTERCOM to Station 1001. In AICD you must make two line entries:

ICM STN: 1000 SIG STN: 1001
ICM STN: 1001 SIG STN: 1000

TN: Tenant Number
ICM STN: Intercom Station Number
SIG STN: Signaling Station Number

STEP 3: **AKYD** - Assign a Station INTERCOM Line according to the following parameters:

TN: 1
STN: Station Number
TP: Type of Dterm (0-3)
PRI: Form of Line Preference (0-3)
KYN: Key Number (1-40)
KYI: Service Index
0: Key not used
1: Feature Key
2: Multi Line Key
KD: Kind of Line
0: Multi Line
1: INTERCOM Line
2: DSS Console Key
3: -
4: MultipleLine Appearance
ICM: Kind of INTERCOM
0: AUTOMATIC INTERCOM
1: MANUAL INTERCOM
2: DIAL INTERCOM

If Manual Signaling is used, assign a Line/Feature key as Type:Function: Manual Signaling key [60]

Note: AKYD A/M Data must be deleted before deleting AICD.

A-43 AC Clearing

General Description

This feature is provided to prevent the occurrence of double-line seizure by transmitting a Continuous Ringing Signal that supervises the balance condition of each Exchange Line while in an idle state.

Note: This feature is available only in Australia.

Operating Procedure

No manual operation is required.

Service Conditions

1. This feature is available in Australia only.
2. This feature is activated on Exchange Lines that are in the unbalanced idle condition.

Programming

STEP 1: **ARTD** - Assign the Route Class Data of the Trunk for AC CLEARING. Assign CDN: 39. BC (Central Office Line Balance Condition): 0/1 = Not balanced/Balanced, assign “0”.

Note: To change the data for existing route (RT), initialize or reboot the trunk device such as a circuit card or MG.

A-52 Account Code - Attendant

General Description

This feature provides the ATTENDANT CONSOLE [A-3] with the capability to enter an Account Code into the system after talking with a Central Office Outgoing/Incoming Call.

Operating Procedure

The Attendant talks with the Central Office Trunk Call.

1. Press the START key.
2. Dial the Account Code Feature Access Code. Service Set Tone is heard.
3. Dial the Account Code (up to ten digits).
4. Dial the desired Telephone Number.

Service Conditions

1. The Account Code may be a maximum of ten digits.
2. When the Account Code is entered again for the same call and Station, the last code entered will be recorded on STATION MESSAGE DETAIL RECORDING (SMDR) [S-10, 47].
3. Account Code dialing is available for an ATTENDANT CONSOLE [A-3] connected to a Trunk.
4. The Feature Access Code for Account Code will be a maximum of three digits.
5. An Attendant can enter an Account Code of fewer digits than the maximum length defined, indicating the end of entry by "#" key.

Programming

STEP 1: **ASYD** - SYS1, Index 43, Bit 2. Is Service Set Tone sent after the ASPA Access Code for ACCOUNT CODE? 0/1: No/Yes. Assign data "1".

SYS2, Index 3, Bit 4. Is ACCOUNT CODE used with Authorization Code? 0/1: No/Yes.

SYS1, Index 93, Bit 6=0.

STEP 2: **ANPD** - Reserve a Number Level for Service Feature Access. Select Normal Service (N), Hooking Service (H), or Busy Service (B) from the combo box for Connection Indexes. (applicable to any feature assigned to this level). Assign NND in accordance with a predetermined Numbering Plan.

STEP 3: **ASPA** - For SRV = "SSC: Service Code", SID = 41, assign the Access Code. Assign the number of digits in the ASPA Service Feature Access Code plus the number of digits that will appear in the ACCOUNT CODE for the NND parameter. When ACCOUNT CODE operates with AUTHORIZATION CODE [A-20], the above assignment is omitted. In ASPA, assign for SRV = "SSC: Service Code", SID = 42.

The NND value will include the number of digits in the ASPA Service Feature Access Code plus the number of digits that appear in the ACCOUNT CODE.

For example:

45 - 511234 - 617- 555 - 2323

2 + 6 = 8

ACC + ACCT.CODE = NND

For example:

44 - 623423 - 511234 - 617 - 555 - 2323

2 + 0 + 6 = 8

ACC + AUTH CODE + ACCT.CODE = NND

STEP 4: **ASFC** - For Stations to use ACCOUNT CODE, assign a Class of Service that allows SFI 27 (Acct. Codes/Auth. Codes/Forced Acct. Codes).

STEP 5: **AMND** - If Authorization Codes are used with ACCOUNT CODES, then for Tenant 0, assign the Destination Code (DC) as the first digit of all Authorization Codes and the Maximum Necessary Digits (MND) of the Authorization Codes. For the above example, a DC of six is assigned with an MND of six.

STEP 6: **AATC** - Assign when Authorization Code is used.

FCCS Programming

This data setting describes only the additional or replacement commands used to activate FCCS Services. When used as an FCCS feature, these commands replace the non-network commands (for example, AAED is replaced by AAEDN). Note that the data setting for FCCS Service varies depending on the Node.

Note: When using ASYDL or ASYDN to program an FCCS feature, the basic programming of ASYD is required in addition to ASYDL or ASYDN.

STEP 1: **ANPDL** - Reserve a Number Level for Service Feature Access. Select Normal Service (N), Hooking Service (H), or Busy Service (B) from the combo box for Connection Indexes. (applicable to any feature assigned to this level). Assign NND in accordance with a predetermined Numbering Plan.

STEP 2: **ASPAL** - For SRV = "SSC: Service Code", SID = 41, assign the Access Code. Assign the number of digits in the ASPAL Service Feature Access Code plus the number of digits that will appear in the ACCOUNT CODE for the NND parameter. When ACCOUNT CODE operates with AUTHORIZATION CODE [A-20], the above assignment is omitted. In ASPAL, assign for SRV = "SSC: Service Code", SID = 42.

The NND value will include the number of digits in the ASPAL Service Feature Access Code plus the number of digits that appear in the ACCOUNT CODE.

A-53 All Zone Paging

General Description

This feature provides both the Attendant and a user with dial access to Multiple Zone Paging Equipment.

Operating Procedure

To operate from a Station

1. The Station goes off-hook. Receive Dial Tone.
2. The Station dials the ALL ZONE PAGING Access Code.
3. Connections to Multiple Paging Equipment are established.

Service Conditions

1. This feature allows only Paging Access. PAGING TRANSFER [P-10] and MEET-ME PAGING [M-11] are not provided.
2. A maximum of seven Paging Zones can be accessed simultaneously.
3. When a user or an ATTENDANT CONSOLE [A-3] has a held Station or Trunk, this feature is not activated.
4. If the Paging Trunk is busy, the Paging Connection is not established to the busy zone. PRIORITY PAGING [P-30] Access Code + ALL ZONE PAGING [A-53] Access Code dialing is possible. Even if this dialing operation is performed, the PRIORITY PAGING [P-30] function is not activated when the line is busy (it has the same result as ALL ZONE PAGING Access Code dialing).
5. One Trunk circuit is required for each PAGING ACCESS [P-1].
6. Amplifiers and speakers must be customer provided.
7. Where TENANT SERVICE [T-12] is provided, separate Access Codes and hardware should be provided for each Tenant.
8. The ALL ZONE PAGING Access Code is a maximum of three digits [Outgoing Call Alternative (OGCA)].
9. The following can be used as a paging trunk interface:

    - COT Circuit Cards
    - TLT Circuit Cards
    - MC&MG-COT (4LC2COT)
    - MG-COT (6COT)
    - UG50 (GCD-PGTA) **Note 1 Note 2**

    Note 1: UG50 (GCD-4COTA/B/C) cannot be used as a paging trunk interface.

    Note 2: Available from UG50 software version 2.00.

10. Loop Signaling is the recommended signaling type.
11. This feature can be activated via FCCS.

Interactions

1. MEET-ME PAGING [M-1, 11] and ALL ZONE PAGING [A-53] can be provided simultaneously in the same system.
2. To directly answer ALL ZONE PAGING by the Attendant Console, the Paged Party should use the INDIVIDUAL TRUNK ACCESS [I-4] feature.
3. Paging is limited to Stations or Attendant Consoles. Other parties (Central Office Trunk Calls) must have the Attendant Console or user perform the Paging operation.
4. BUSY VERIFICATION [B-3], EXECUTIVE RIGHT-OF-WAY [E-1]: These features will be denied if the Paging Route has been assigned for privacy. Verify the data in the assignment of Route Data command, ARTD, CDN 44:PRV. If TCL is assigned as "8", CDN 44:PRV is not required.
5. For simple ALL ZONE PAGING, assignments in ASFC are not required. Refer directly to the other Paging features, such as PAGING TRANSFER [P-10].
6. The Attendant Console can dial Priority Page Access Code and release a station.
7. When disconnecting from a Paging Trunk, the Trunk becomes idle immediately.
8. ALL ZONE PAGING is not possible from CONSULTATION HOLD [C-17].
9. CALL HOLD [C-6]: A Paging Trunk cannot be placed on Call Hold.
10. The Desktop terminal LCD indicates BUSY when the Paging Trunk is Busy. Reorder Tone is returned.
11. If Priority Page is to be used with ALL ZONE PAGING, ARTD, CDN 6:TCL must be assigned as "8", Page with Ringback Tone. SYS1, Index 73 affects the operation of Page Routes assigned Page with Ringback Tone. Therefore, if the Priority Page is used with ALL ZONE PAGING, then MEET-ME PAGING [M-1, 11] and PAGING TRANSFER [P-10] are not available to the system.
12. Priority Page access will not disconnect an Attendant Console engaged in a Paging Trunk Connection.

Programming

STEP 1: **ASYD** - SYS1, Index 73, Bit 0 and Bit 1. For the Page function only, assign Data 00.

STEP 2: **ARTD** - Assign the Paging Route as shown below:

RT:1

1-OSGS	:2	2-ONSG	:3	5-TF	:1	6-TCL	:8	7 L/T	:1
8-RLP	:2	15-LSG	:1						

Note: To change the data for existing route (RT), initialize the circuit card.

STEP 3: **ATRK** - Assign the LENS, Paging Trunk, Route Number, Trunk Number, and Tenant Number.

STEP 4: **MBTK** - Assign the Make Idle status to the Paging Trunks.

STEP 5: **ANPD** - Assign a level for Trunk Access. Select Normal Service (N) or Hooking Service (H) from the combo box for Connection Indexes. Assign the Number of Necessary Digits, according to a predetermined Numbering Plan. Busy Lamp Field is not activated.

STEP 6: **ASPA** - Assign Paging Trunk Route Data for two or more Routes, with a maximum of seven. (CI = "N: Normal", SRV = "OGCA: Outgoing Call with route advance", COUNT = 2 - 7, RT = Paging Trunk Route).

STEP 7: **ARSC** - Assign to the Station, which is allowed access to Paging Trunks, a Route Restriction Class (RSC) that is allowed these Routes in Route Restriction Index (RRI) 2 and 3.

FCCS Programming

This data setting describes only the additional or replacement commands used to activate FCCS Services. When used as an FCCS feature, these commands replace the non-network commands (for example, AAED is replaced by AAEDN). Note that the data setting for FCCS Service varies depending on the Node.

STEP 1: **ALRTN** - Assign this command to connect the Logical Route to the Physical Route.

STEP 2: **ARTDN** - Assign this command in addition to assigning data for ARTD.

Note: To change the data for existing route (RT), initialize the circuit card.

STEP 3: **ANPDN** - Assign a level for Trunk Access. Select Normal Service (N) or Hooking Service (H) from the combo box for Connection Indexes.
Assign the Number of Necessary Digits, according to a predetermined Numbering Plan. Busy Lamp Field is not activated.

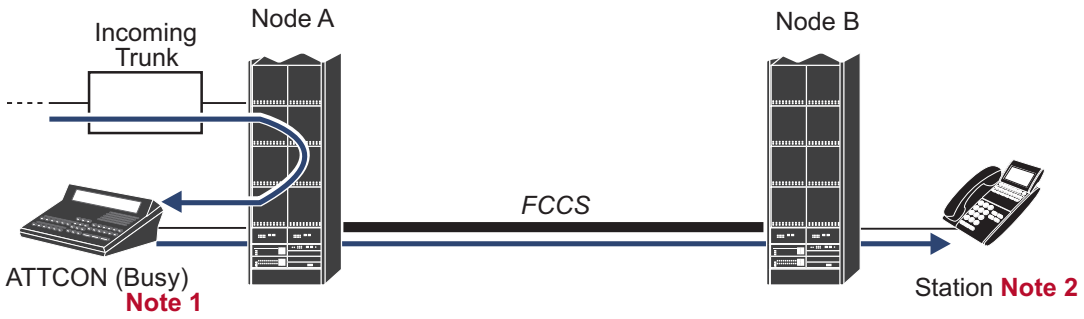
STEP 4: **ASPAN** - Assign Paging Trunk Route Data for two or more Routes, with a maximum of seven, (CI = N, SRV = "OGCA: Outgoing Call with route advance", COUNT = 2 - 7, RT = Paging Trunk Route).

STEP 5: **ARSCN** - Assign to the Station, which allowed access to Paging Trunks, a Route Restriction Class (RSC) that is allowed these Routes in Route Restriction Index (RRI) 2 and 3.

A-55 Automatic Change of Night Service

General Description

This feature allows calls directed to an Attendant from a Trunk Line to be transferred according to the Night Connection Service when the Attendant does not answer the call for a predetermined time.
The feature enables Incoming Trunk Calls, directed to the Attendant Console, to overflow to a predetermined Station on the FCCS Network via FCCS, also.



Note 1: When Attendant/Operator does not answer a call, the call overflows.
Note 2: This is the Station transfered by Night Connection Service.

Note: This feature is not available in North America.

Operating Procedure

1. An Incoming Trunk Call arrives at the ATTENDANT CONSOLE [A-3].
2. The Attendant does not answer for a predetermined time.
3. Night Connection Service is activated for the Incoming Call.

Service Conditions

1. This feature is only available while the system is in Day Mode (not in Night Mode).
2. This feature is provided for Incoming Calls to the ATTENDANT CONSOLE [A-3] from a Central Office Trunk Line (TCL 1, 2, 3) or Ringdown Trunk Line.
3. This feature is not available for INDIVIDUAL ATTENDANT ACCESS [I-6], which allows a call to be directed to a particular ATTENDANT CONSOLE [A-3].
4. This feature is not available concurrently with ATTENDANT NIGHT TRANSFER [A-19].
5. The Overflowing Timer Value is determined by SYS1, Index 140 (the default is 40 seconds).
   When the destination does not answer the Overflowed Call, the time value is reset to 40 seconds.
6. This feature is not available for calls to be transferred outside by following the features:

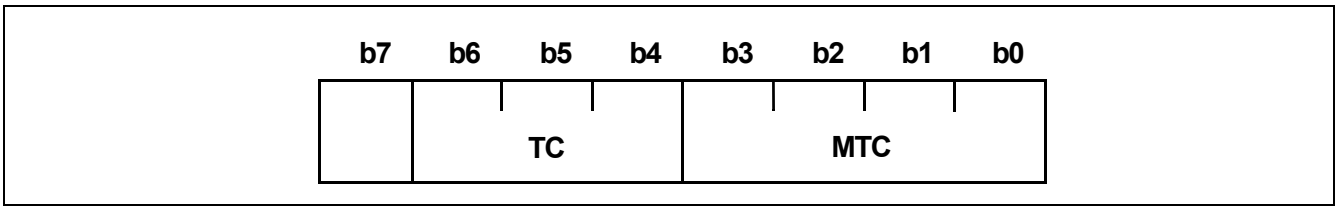
CALL FORWARDING - ALL CALLS - OUTSIDE	[C-28]
CALL FORWARDING - BUSY LINE - OUTSIDE	[C-60]
CALL FORWARDING - DON'T ANSWER - OUTSIDE	[C-74]
LDN NIGHT CONNECTION	[L-28]
NIGHT CONNECTION OUTSIDE - SYSTEM	[N-17]

7. When a Station activated by this service has set CALL FORWARDING - DON'T ANSWER [C-3] to an Attendant (or outside), the station is connected to TRUNK ANSWER FROM ANY STATION (TAS) [T-8] by No Answer Time-out, but it will remain calling that station in the case of TAS Restriction.

8. In the case of DIRECT INWARD DIALING (DID) [D-8] Termination, this feature is available for an Operator Call, LDN, Intercept, and Forwarded Calls to an Attendant in CALL FORWARDING - ALL CALLS/BUSY LINE/DON'T ANSWER [C-5, 2, 3].
9. DID includes CCIS terminating call via MFC and Central Office Trunks.
10. In the case of DIRECT-IN TERMINATION (DIT) [D-7], this feature is available when a station has set CALL FORWARDING - ALL CALLS/BUSY LINE [C-5, 2] to the Attendant.
11. AUTOMATIC CHANGE OF NIGHT SERVICE and DELAY ANNOUNCEMENT - ATTENDANT [D-114] are mutually exclusive features. DELAY ANNOUNCEMENT ATTENDANT [D-114] takes priority over AUTOMATIC CHANGE OF NIGHT SERVICE.
12. A PS cannot be assigned as Overflowing Destination.
13. When Call Forwarding - All Calls [C-5] or Call Forwarding - Busy Line [C-2] is set to the Overflowing Station, an Overflowed Call to the station is transferred to the destination.
14. When Call Forwarding - All Calls [C-5] is set to the Overflowing Station in Hunting Group (Station Hunting - Circular [S-7]/Station Hunting - Terminal [S-9]/Uniform Call Distribution (UCD) [U-1]), an Overflowed Call to the station is transferred by Call Forwarding - All Calls. (Hunting function is not activated.)
15. When Call Forwarding - Busy Line [C-2] is set to the Overflowing Station in Hunting Group (Station Hunting - Circular [S-7]/Station Hunting - Terminal [S-9]/Uniform Call Distribution (UCD) [U-1]), an Overflowed Call to the station is hunted (only when all stations in the Hunting Group are busy, Call Forwarding - Busy Line is activated.)
16. When Call Forwarding - All Calls [C-5]/Call Forwarding - Busy Line [C-2] to outside is set to the Overflowing Station, the call is not overflowed (the Attendant Console is called).
17. Overflowing Destination/Timer is determined by the System Data of the node that accommodates the Trunk which a call terminates.
18. The Incoming Call continues to ring at the Attendant Console when designated FCCS trunks are all busy, or locked out, etc.
19. FPC of a node which transmits Day/Night changeover information is determined by SYS1, Index 640, and the information is transmitted to a node of FPC designated by SYS1, Index 704.

Programming

STEP 1: **ASYD** - SYS1, Index 140, Bit 0 to Bit 6: No Answer Timer for Automatic Change of Night Service.



MTC = 1 Hexadecimal to F Hexadecimal (1 -15)
TC = 3 Hexadecimal (two seconds) or 7 Hexadecimal (eight seconds)
(No Answer Timer) = (MTC) × (TC) (seconds)

Note: Default data is set to 40 seconds.

SYS2, Index 15, Bit 1 (Automatic Change of Night Service): 0/1 = Out/In Service, Assign "1".

STEP 2: **ACSI** - Assign the Connection Service Index (CSI) for AUTOMATIC CHANGE OF NIGHT SERVICE.

Assign CSI = 4 (Direct-In Termination (Night Mode))

STEP 3: **ACFR** - Assign the Call Forwarding Restriction Data.

TSFI = 2 [Direct in Termination (Night only)]

CCI: Assign the appropriate Call Category Index Number [1 = LDN (Central Office Line), 4 = TIE Line].

RES = 1 (Transfer Service is allowed)

FCCS Programming

The following command assignments are required when your system is configured as an FCCS Network.

STEP 1: **ASYDL** - SYS1, Index 640, Bit 0 to Bit 7. Assign FPC of the node which transfers the Day/Night changeover information. **Note 1**

SYS1, Index 704 to Index 735. Assign FPC of the node to which Attendant Console transfers the Day/Night changeover information.**Note 1**

STEP 2: **ASYD** - SYS1, Index 140. This determines the Timer Value at which the Incoming Call will overflow to the Destination Station. **Note 1**

SYS2, Index 15, Bit 1. Allowance of ATTENDANT OVERFLOW. Assign 0/1 = Out/In Service. **Note 1**

STEP 3: **ANPDL/ANPDN** - Reserve a number level for feature access. Assign following data in accordance with a predetermined plan.

TN: Tenant Number
1stDC: 1st digit Code (0-9,*,#)
CI: Select Normal Service (N), Hooking Service (H), or Busy Service (B) from the combo box for Connection Indexes.
NND: Number of Necessary Digits (1-6)

BLF: Busy Lamp Field Display Activated on ATT

STEP 4: **ASPAL/ASPAN** - Determine the kind of service and the Service Access Code to be executed.

TN: Tenant Number (1-255)
ACC: Access Code (maximum 6-digit)
CI: Connection Status Index (N: Normal, H: Hooking, B: Busy)
SRV: Kind of Service (TELN: Telephone Number)
NND: Necessary Digit of Telephone Number

STEP 5: **ASDT** - Assign the Station Data for Destination Overflow Station.

TN: Tenant Number (1-255)
STN: Station Number
LENS: Line Equipment Numbers
TEC: Telephone Class
SFC: Service Feature Restriction Class

STEP 6: **ALGSL/ALGSN** - Assign Telephone Number to the Destination Station Number programed by ASDT command above.

TYPE=1 (Line Equipment Numbers/Phantom Index is used to specify the Telephone Number)

UGN: User Group Number (1-255)
TELN: Telephone Number (maximum 16-digit)
FPC: Fusion Point Code (Only ALGSN)
LENS/PHIN:Line Equipment Numbers/Phantom Index of the destination Station Number

TYPE=2 (Station Number is used to specify the Telephone Number)

UGN: User Group Number (1-255)
TELN: Telephone Number (maximum 16-digit)
FPC: Fusion Point Code (Only ALGSN)
TN: Tenant Number
STN: Station Number of destination

STEP 7: **ACSIL** - Assign Overflowing Destination of Attendant Console. **Note 1**

RT/LGRT:External Route Number/Logical Route Number (1-899)
TK: Trunk Number (1-255)
CSI: Connection Service Index, assign 4 [Direct-In Termination (Night Mode)]
UGN: User Group Number (1-255) **Note 2**
TELN: Telephone Number (a maximum of 16 digits) **Note 2**

STEP 8: **ACFR** - Assign the Call Forwarding Restriction Data.

TSFI: Transfer Service Feature Index 2 = Direct Inward Dialing (DID)
CCI: Call Category Index [1 = LDN (Central Office Line), 4 = TIE Line)]
RES: Restriction Data (0 = Transfer service is not allowed, 1 = Transfer service is allowed)

Note 1: Assign the data to the node that a call terminates to.

Note 2: This parameter is displayed when CSI = 2, 3 or 4.

A-59 Announcement Service - Attendant

General Description

This feature allows the Attendant, via the ATTENDANT CONSOLE [A-3], to hear a prearranged announcement when the Attendant dials a predetermined Access Code.

Operating Procedure

1. Press an Idle Loop key.
2. Dial the Announcement Trunk Access Code.
3. Listen to the message from Announcement Machine.

Service Conditions

1. This feature requires one of the following hardware alternatives:
   a.) Central Office Trunk (COT) or 2W E&M Trunk and Announcement Machine.
   b.) Digital Announcement Trunk (DAT).
2. The following connections are available:
   a.) Multi Connection: several Attendants can connect to one Announcement Trunk at the same time.
   b.) Single Connection: one Attendant can connect to one Announcement Trunk at a time.
3. This connection can be continuous until the Attendant presses the CANCEL key or can be discontinued after 30 seconds. This function can be selected on a per Announcement Trunk, and per Tenant basis.
4. Before connection to the Announcement Machine is made, the system can be programmed to return Ringback Tone. This is selected on a per Tenant and per announcement basis.
5. This service can be provided for a maximum of 127 announcements per Tenant, which includes the Announcement Trunks for:

ANNOUNCEMENT SERVICE	[A-15]
CALL FORWARDING - ALL CALLS - ANNOUNCEMENT	[C-69]
CALL FORWARDING - INTERCEPT / ANNOUNCEMENT	[C-25]

6. ANNOUNCEMENT SERVICE is provided to FCCS Nodes that have no Announcement Trunk using the Trunk in the Remote Node.
7. When a call is terminated from the ISDN Trunk or MG(SIP), RBT is sent to the caller if the Announcement Trunk Answer Signal is not sent.

Programming

STEP 1: **ANPD** - Reserve a Number Level for Trunk Access. Select Normal Service (N) or Hooking Service (H) from the combo box for Connection Indexes. Assign for NND values in accordance with a predetermined Numbering Plan.

STEP 2: **ASPA** - Assign an Access Code to the Announcement Trunk. Assign SRV = "ANNC: Announcement Service - Single Announcement". The EQP number assignment should be as assigned in AAED. TYPE = 0/1.

STEP 3: **ARTD** - Assign the Announcement Route as shown below. Be sure to provide privacy by programming CDN 44:PRV as data "1".

RT: 1

1-OSGS	:2	2-ONSG	:3	5-TF	:1	6-TCL	:1	7-L/T	:1		
8-RLP	:2	15-LSG	:4								

STEP 4: **ARSC** - Assign a Route Restriction Class (RSC) to the Station that allows access to the Announcement Route. Assign RSC 0 to the Attendant, and allow that RSC access to the Announcement Route.

STEP 5: **ATRK** - Assign the LENS, Announcement Trunk Route Number. Assign one or more Trunks to a specific Route used to connect to an Announcement Machine.

STEP 6: **MBTK** - Assign the Make Idle status to the Announcement Trunk.

STEP 7: **ARRC** - If TIE Line, DID, or REMOTE ACCESS TO System [R-2] Connection to the Announcement Service Trunk is required, allow Trunk-to-Trunk Connection using ARI-D-RES = 1 (Direct Connection is allowed.)

STEP 8: **AAED** - This command is used to assign Announcement Equipment.

Dead Level:

Assign:

TN: Tenant Number

EQP: (Announcement Equipment Number) = 0 (Dead Level). For example, Level 8 is not assigned. An announcement is returned rather than a Reorder Tone.

RT, TK:Route and Trunk Number of the Trunk connected to the Announcement Equipment.

Unused Number:

Assign:

TN: Tenant Number

EQP: 1 (Unused Number). For example: Extension 7836 is not in the system. An Announcement is returned rather than a Reorder Tone.

RT, TK:Route and Trunk Number of the Trunk connected to the Announcement Equipment.

Announcement Service:

Assign:

TN: Tenant Number

EQP: = 2 - 15

RT, TK:Route and Trunk Number of the Trunk connected to the Announcement Equipment.

C: Duration of connection, 0/1:Disconnect in 30 seconds/connection is held until the Station releases. See the requirements of the application for the Announcement Equipment. For Dead Level applications, data 0 is assigned.

R: Sending RBT 0/1: sending RBT/not sending RBT. **Note 1**

Note 1: Assign “1” (not sending Ringback Tone) when the “M” parameter = “0” (Single Connection).

A: Answer Signal Sending, 0/1: no answer from Incoming Trunk/answer from Incoming Trunk. Normally assign data “1”.

M: Multiple Connection 0/1: Single Connection/Multiple Connection. Assign usually for Single Connection.

FCCS Programming

This data setting describes only the additional or replacement commands used to activate FCCS Services. When used as an FCCS feature, these commands replace the non-network commands (for example, AAED is replaced by AAEDN). Note that the data setting for FCCS Service varies depending on the Node.

STEP 1: **ALRTN** - Assign this command to connect the Logical Route to the Physical Route.

STEP 2: **ARTDN** - Assign the Announcement Route as shown below. Be sure to provide privacy by programming CDN 44:PRV as data "1".

LGRT: 1

1-OSGS	:2	2-ONSG	:3	5-TF	:1	6-TCL	:1	7-L/T	:1
8-RLP	:2	15-LSG	:4	44-PRV	:1

STEP 3: **AAEDN** - This data must be assigned to the NCN.

Assign the Announcement Equipment Data.

EQUIPMENT KIND:

- DAT/Announcement Machine
- VS32

TN: Enter Tenant Number

UGN: Enter User Group Number

EQP: Announcement Equipment Number

0 = Dead Level

1 = Unused Number

2-15 = Announcement Service

Duration of Connection:

0 = Disconnect in 30 seconds

1 = Connection is held until the Station releases. See the requirements of the application for the Announcement Equipment. For Dead Level applications, data 0 is assigned.

Do Not Send RBT [On/Off]

Note: Assign "Do not Send RBT" when Multiple Connection = Single.

Send Answer Signal [On/Off]

Note: Normally assign "Send."

Multiple Connection:

0 = Single Connection

1 = Multiple Connection

LGRT: Logical Route

TK: Trunk Number **Note 2**

STEP 4: **ARRCN** - This data must be assigned to the NCN.
Release the restriction between the Terminating Trunk and the Announcement Trunk.

STEP 5: **ARSCN** - This data must be assigned to the NCN.
Release the restriction between the Announcement Trunk and the Station.

Note 2: Not necessary for Multiple Connection = Single.

Note: The current command data shown is set to all Nodes using the service: ASTD, ARTD, ATRK, MBTK

A-60 Attendant-to-Attendant Calling

General Description

This feature permits an Attendant, via an Attendant Console, to access a non-specific free Attendant by dialing the Operator Access Code, or to call a particular Attendant Console by dialing an Individual Attendant Identification Number.

Operating Procedure

1. Press an Idle Loop key.
2. Dial the Individual Attendant Identification Number or Operator Call Code.

Service Conditions

1. An Attendant Identification Number must be published for each Attendant.
2. ATTENDANT NIGHT TRANSFER [A-19] cannot be activated for calls via this feature.
3. The Operator Call Code should be a one-digit number. Normally, 0 is assigned as the Operator Call Code.
4. No Operation keys on the Called Attendant Console can be used while performing Attendant-to-Attendant Calling.
5. Only the CANCEL key on the Calling Attendant Console can be used to disconnect Attendant-to-Attendant Calling.
6. An Attendant can receive only one Attendant-to-Attendant Call at a time. An INCOMING CALL IDENTIFICATION (ICI) key can be designated for receipt of TF: INTER-POSITION TRANSFER CALLs and Attendant-to-Attendant Calls.
7. An Attendant cannot extend an Attendant-to-Attendant Call to any other destination.
8. With an Individual Attendant Identification Number, an Attendant Console/Desk Console within an FCCS Network can be specified.
9. The Operator Call Code is dialed. The call is routed to the predetermined Attendant Console by ASPAL.
10. When assigning Function Code to KYN 1 of an Attendant Console by AAKP, this feature cannot be used.

Programming

STEP 1: **ASYD** - SYS1, Index 9. Assign the number of Attendant Consoles/Desk Consoles.
STEP 2: **ASPA** - Assign SRV = “SSC: Service Code”, SID = 2 (Operator Call).
STEP 3: **ASAT** - Assign Identification Numbers to the specific Attendant Consoles.

FCCS Programming

This data setting describes only the additional or replacement commands used to activate FCCS Services. When used as an FCCS feature, these commands replace the non-network commands (for example, AAED is replaced by AAEDN). Note that the data setting for FCCS Service varies depending on the Node.

Note: When using ASYDL or ASYDN to program an FCCS feature, the basic programming of ASYD is required in addition to ASYDL or ASYDN.

STEP 1: **ASYDN** - SYS1, Index 9. Assign the number of Attendant Consoles / Desk Consoles.

STEP 2: **ASPAL** - Set to all the Nodes (accommodates the Attendant Console). Assign SRV = "SSC: Service Code", SID = 2 (Operator Call), FPC of the Node that Operator Calls are to be terminated to.

STEP 3: **ALGNN** - Set to the NCN.

STEP 4: **ASATN** - Set to the NCN. Assign Identification Numbers to the specific Attendant Consoles.

A-68 Automatic Hold

General Description

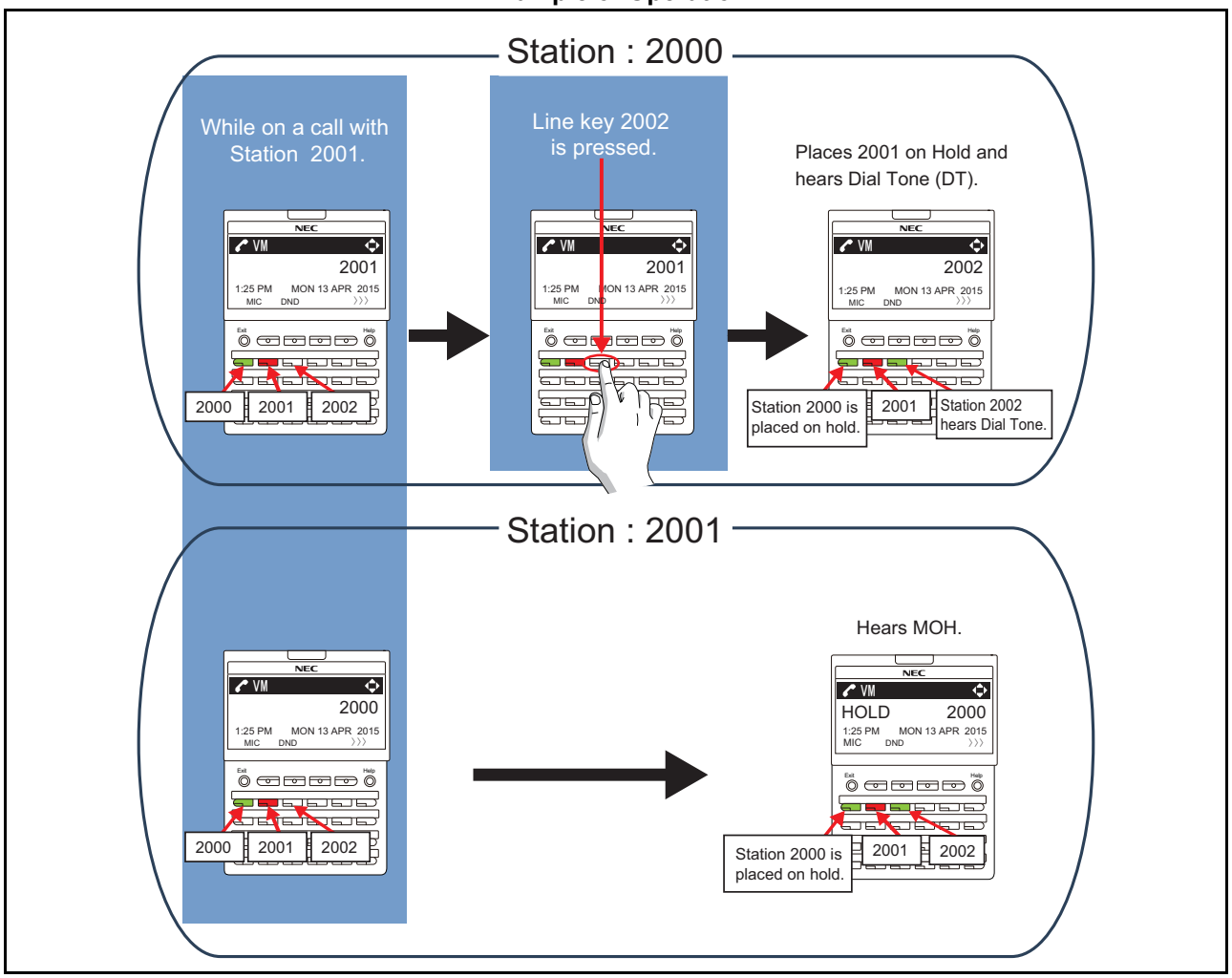
This feature allows users to automatically place the first party on hold and answer the second party by just pressing the intended Line key. When this feature is used, an opposite party is placed on hold under the same conditions when HOLD key is pressed during a conversation. Hold status can be selected from the following by office data setting.

- Non-Exclusive Hold -D [N-7D]
- Hold Fixation Mode
- Call Park Group -D [C-176D]

Operating Procedure

Press a Line key while on a call with a station/trunk.

Example of Operation



Note: This feature is only available for Line keys (not currently being used) to which KD=0 (Multi-Line) or KD=4 (Same Number Sub Line) of AKYD command is assigned.

Service Conditions

<Important Notices>

1. The following system data setting is required to use this feature.

   ASYDL, SYS1, Index 1248, Bit 1=1

   Note: Initialize the terminal (for IP Multiple Line terminal/SIP Multiple Line terminal) or ELC card (for Digital Multiple Line terminal) for the change of this system data to take effect. For details, see Programming section.

2. This feature is only available for Line keys to which KD=0 (Multi-Line) or KD=4 (Same Number Sub Line) of AKYD command is assigned (when pressed during a call in progress).

   Note: This feature is not available during an Intercom call or a call with an Attendant Console.

3. While a line used currently (original line) is in other state than busy state such as in calling state and in held state, this feature does not function in the line. When you press another Line key, the existing line is released then the line you pressed will be connected.
4. When a Line key is pressed during a call in progress, status of each terminal is as follows:

Status of Each Party (Basic Pattern)

Case	Status of Line Key to be Pressed	Automatic Hold	Kind of Opposite Party	Status of Opposite Party	Status of Operating Party
1	Idle * Lamp goes off.	Disabled	Station/Trunk	Released	Seizing Second Party's line (hearing DT).
			ATT/Intercom		
		Enabled	Station/Trunk	Hearing MOH	
			ATT/Intercom	Released	
2	Receiving an incoming call or having a call held by another terminal * Lamp flashes red.	Disabled	Station/Trunk	Released	Answering the incoming call and starting the conversation.
			ATT/Intercom		
		Enabled	Station/Trunk	Hearing MOH	
			ATT/Intercom	Released	
3	Talking with/calling to another line * Lamp lights red.	Disabled	Station/Trunk	Status does not change (continues to talk).	
			ATT/Intercom		
		Enabled	Station/Trunk	Hearing MOH	Hold **Note 1**
			ATT/Intercom	Status does not change (continues to talk).	
4	Talking with the operating Party. * Lamp lights green.	Disabled	Station/Trunk		
			ATT/Intercom		
		Enabled	Station/Trunk		
			ATT/Intercom		

Note 1: After a line is held by Non-Exclusive Hold -D [N-7D], re-pressing the HOLD key changes its state to EXCLUSIVE HOLD -D [E-4D].

5. When Line keys not supporting this feature are pressed during a call in progress, status of each terminal is as follows:

Status of Each Party (When Line Keys Not Supporting This Feature are Pressed)

KD: Kind of Line (AKYD)	Status of Opposite Party	Status of Operating Party
KYI=2 (Multi-Line), KD=1 (Intercom)	Released	Intercom Calling
KYI=2 (Multi-Line), KD=2 (DSS key)	Hearing MOH	Forwarding to the appropriate Station

<Interactions with Other Features>

1. Interaction conditions between this feature and other Hold features are as follows:
   a.) Although you can use this feature even in Hold Fixation mode, you cannot seize an idle line or answer an incoming call.
   b.) After this feature is performed to place a call on hold, you cannot use the Hold button to perform Exclusive Hold -D [E-4D] against the held line. Exclusive Hold-D [E-4D] is able to function only when the line is held by the Hold button. [See **Note 1** in the Status of Each Party (Basic Pattern) list.]
   c.) Place an opposite party on hold once and then press the HOLD key to place the party on Exclusive Hold -D [E-4D].
   d.) When this feature is used while Call Park Group -D [C-176D] is enabled, a call is placed on hold under the conditions of Call Park Group -D [C-176D].
   e.) When this feature is used while Call Park Group -D [C-176D] is disabled, a call is placed on hold under the conditions of Non-Exclusive Hold -D [N-7D].
   f.) When this feature is used while Call Park Group -D [C-176D] with the CALL PARK GROUP -D HOLD key is enabled, a call is placed on hold under the conditions of Non-Exclusive Hold -D [N-7D].
2. After a Secretary answers a call terminated to Boss' line while this feature is enabled, the Secretary can transfer the call to the Boss by pressing the Boss's Line key.

Note: The status of Secretary when connecting to an Incoming call is the same as Case 4 of Status of Each Party (Basic Pattern). After the Boss' Line key is pressed, the existing feature (Boss - Secretary Transfer [B-11]) is activated.

3. When the BROKERAGE HOT LINE key is pressed while this feature is enabled, a call in progress is placed on hold and Hotline destination is called.
4. This feature cannot be activated from an Emergency Telephone and a station currently talking with an Emergency Telephone. When these station users press an idle Line key, the first call is released and the pressed Line key (second party) is seized. When these stations press a busy Line key, "RST" appears on the LCD and the conversation with the first party continues.
5. This feature is not available during an Intercom call. When an idle Line key is pressed during an Intercom call, the first call (Intercom call) is released and the pressed Line key (second party) is seized (for how each terminal operates, see Status of Each Party (Basic Pattern)). Also, when an Intercom Line key is

pressed during a call in progress, this feature cannot be activated (for how each terminal operates, see Status of Each Party (When Line Keys Not Supporting This Feature are Pressed)).

6. This feature is not available during a call with an Attendant Console. When an idle Line key is pressed during a call with an Attendant Console, the first call (call with the Attendant Console) is released and the pressed Line key (second party) is seized (for how each terminal operates, see Status of Each Party (Basic Pattern)).
7. This feature is not available during a conference call with Add-On Conference - 8 Party [A-121]. When an idle Line key is pressed during a conference call, the conference call is released and the pressed Line key (second party) is seized.

<Hardware/Software>

1. Requirement for circuit card to use this feature is shown in the table below.

Equipment	Firmware
CH-16ELCA	SP-4035 16ELCA PROG-A Issue 1 or later
PA-16ELCN-A **Note 2**	SP-4013 16ELCN PROG-A Issue 1 or later

Note 2: This feature only works when the Terminal Mode Setting of Switch 11 (SW11 5-6) on ELC circuit card is set to ON/ON (Standard Setting). For details on the switch settings of the circuit cards, see sections related to ELC cards in the Circuit Card Description.

2. When the held party is an IP terminal and the connection with the party is disconnected due to the accidents such as network failure or unplugging of cables, the lamp of the Line key on the Desktop terminal that held the call keeps flashing. If a user of this Desktop terminal tries to retrieve the call in this condition, dial tone is heard and the lamp of the Line key will stop flashing and turn on.

<Interactions with Networks>

None.

<Conditions Specific to This Feature>

None.

<Other Conditions>

None.

Programming

STEP 1: **ASYDL** - SYS1, Index 1248, Bit 1. Assign 1 to enable Automatic Hold [A-68].

Note: Initialize the terminal (for IP Multiple Line terminal/SIP Multiple Line terminal) or ELC card (for Digital Multiple Line terminal) for the change of this system data to take effect.

STEP 2: **AKYD** - Assign Multi-line data as follows:
KYI = 2 (Multi-line)
0/1 = OFF/ON
KD (Kind of Line): 0 (Multi-line) or 4 (Same Number Sub Line)

- When KD = 0 (Multi-line):

  Line TN: Enter the tenant number of Multi-line.
  STN: Enter the station number of Multi-line.
  RG: Assign Ring Information for each Multi-line (0-7).

- When KD = 4 (Same Number Sub Line):

  LENS: Enter the LENS of the Same Number Sub Line.

  Note: When KD = 4 (Same Number Sub Line), the following data setting is required.
  ASFC SFI 212 = 1 (Same Number Sub Line)

A-69 Automatic Idle Return

General Description

This feature enables a Desktop terminal to become idle automatically after a predetermined time when the line is released by the opposite Station. Therefore, manual operation for the line release (such as on-hook, speaker, monitoring, or call originating) is not required.

Operating Procedure

No manual operation is required.

Service Conditions

1. This feature is available for the following status:
   - While a conversation with hands-free operation is in progress
   - While a conversation using handset is in progress **Note 1**
   - While a conversation using headset is in progress **Note 1**

   Note 1: Available since FP95-114 V4 Issue 2.0.
2. This feature may be assigned based on AUTOMATIC IDLE RETURN (ASFC SFI 136=1) of each terminal.

   Note: The conditions of this feature for SIP Handler Controlled SIP terminal are as follows:
   - this feature does not based on Service Feature Class. For details, refer to SIP HANDLER - SIP TERMINAL [S-167].
   - when this feature is activated on off-hook state, Dial Tone (DT) connection may not be established. After the automatic line release, activation of the terminal depends on the type of terminal.

   Note: This feature is not available for SP Controlled SIP terminal, when this feature is activated, the terminal hears Reorder Tone only.

   Note: When using Sub Line, the setting depends on the SFC of My Line which accommodates the Sub Line. Not depended on the SFC of Sub Line.
3. This feature is not available when the Station has a call placed on hold.
4. This feature is activated when Reorder Tone (ROT) is sent to a Desktop terminal individually.
5. Release Timer for this feature can be set by the system data (ASYD, SYS1, Index 401, Bit 0-3).
6. When using this feature with handset/headset, assign the system data (ASYDL/ASYDN, SYS1, Index 686, Bit 4=1).

   Note: Available since FP95-114 V4 Issue 2.0.

7. The following conditions apply to use handset/headset on this feature. **Note 2**

   - After the line is released automatically, Dial Tone (DT) connection on the Prime Line will be established. However, when call restriction of Prime Line is set, the tone of line will be silence (being idle status). **Note 3**
   - Line Lockout [L-3] is not supported.

   Note 2: Available since FP95-114 V4 Issue 2.0.

   Note 3: When using Sub Line, this condition is also applicable.

Interactions

This feature is not available when the following features are enabled:

- ATT CAMP-ON WITH TONE INDICATION [A-1]
- AUTOMATIC/MANUAL INTERCOM [A-30]
- ATTENDANT CAMP-ON WITH TONE INDICATION - CCIS [A-45]
- CALL HOLD [C-6]
- CALL TRANSFER - ALL CALLS [C-11] **Note 4**
- DIAL INTERCOM [D-87]
- OFF-HOOK ALARM [O-6] **Note 5**
- TRUNK LINE APPEARANCE -D [T-23D]

Note 4: For Camp-On by Station only.

Note 5: When the Transferred Station assigned by OFF-HOOK ALARM is in the LINE LOCKOUT [L-3] status or make busy status, this feature is available.

Programming

STEP 1: **ASYD** - SYS1, Index 401, Bit 0 through Bit 3. Assign the timer value for AUTOMATIC IDLE RETURN.

Note: According to the timer value, the time to hear Reorder Tone (ROT) is determined. When the data is set to 0, the timer value will be 0-2. Then, no tone will be connected in nearly every case.

STEP 2: **ASYDL/ASYDN** - SYS1, Index 686, Bit 4: Availability of AUTOMATIC IDLE RETURN for handset and headset (off-hook). 0/1 = Available only when hands-free operation is activated (on-hook)/Available when activating hands-free operation (on-hook) and using handset or headset (off-hook).

Note: This system data is available since FP95-114 V4 Issue 2.0.

STEP 3: **ASFC** - For Stations allowed AUTOMATIC IDLE RETURN, assign a Class of Service that allows SFI 136 (Automatic Idle Return).

A-78 Automatic Number Identification (ANI)

General Description

This feature automatically sends the Calling Subscriber's Number to the Called Party after a response is received from the Distant Office. This feature is used for Outgoing Central Office Calls [Enhanced 911 (E911) Calls] or TIE Line Calls using MF Signaling.

Note: This feature is available only in North America.

Operating Procedure

No manual operation is required. Automatic Number Identification (ANI) Calls are placed using normal call origination procedures.

Service Conditions

1. The ANI feature for Enhanced 911 (E911) Outgoing Connections is available to the following connections:
   a.) Call from a Station by LCR, MF Signaling.
   b.) Call from an Attendant Console by LCR, MF Signaling.
2. ANI Sending Number by Enhanced 911 (E911) - ANI UNIFIED NUMBER OF DIGITS [E-18] is NXX-XXXX, where:
   NXX: Determined for each system
   XXXX: Telephone Number
3. Sending-digit patterns in Enhanced 911 (E911) - ANI UNIFIED NUMBER OF DIGITS [E-18]:
   a.) Outgoing Call from a Station:
   - If ANI Data is assigned:

   KP-0-NXX-XXXX-ST
   Telephone Number
   Information digit

   - If ANI Data is not assigned:

   KP-2-ST
   Information digit

   b.) Outgoing Call from ATTENDANT CONSOLE [A-3]:
   - Individual ATTENDANT CONSOLE [A-3] access available and ANI Data is assigned:

   KP-0-NXX-XXXX-ST
   Individual Attendant Console
   Information digit

   - Individual Attendant Console access not available and ANI Data is assigned:

   KP-0-NXX-XXXX-ST
   ANI sending number
   Information digit

   - ANND Data is not assigned or is invalid:

   KP-1-ST
   Information digit
4. An independent Route is required for an Enhanced 911 (E911) - ANI Interface.

5. It takes some time for a speech path to be established after a Calling Subscriber Number has been sent out to the Remote Office through the Enhanced 911 (E911) - ANI Interface.
6. A Direct Distance Dialed (DDD) Seizure Tone is not sent after the digits have been sent out.
7. Other interactions are the same as for MF Signal dialing.
8. FCCS is not available for this feature.

Programming

STEP 1: **ARTD** - Assign Route Class Data to the Route Number of the Outgoing Trunk.

RT:1
1-OSGS :7 2-ONSG :4 3-ISGS :7 4-INSG :4 15-LSG :5
19-EANI :1

Note: To change the data for existing route (RT), initialize the circuit card.

STEP 2: **AAND** - Assign ANI sending digit as per the Route Number and the number of digits of the Caller's Number.

When AT = 0:
RT: Route Number
AT: 0/1 = STA Outgoing / ATT Outgoing
CC 0
KT: Expansion of Station Number Length **Note 1**
0/1 = OFF/ON
STNL: Station Number Length (When KT = 1)
SKIP: Call Originator Digit to be Skipped
ADD: ANI Information Additional Number of Digits
DC: ANI Sending Number

Note 1: When 0 has been assigned to parameter KT and when the Station Numbers used consist of a different number of digits, Outgoing Routes must be divided on the basis of each number of digits.
When 1 has been assigned to parameter KT, Outgoing Routes need not be divided, even if the Station Numbers used consist of a different number of digits.

A-82 Automated Attendant

General Description

This feature allows a public or private network user to access the system without the Attendant or station user's assistance. After or during an announcement, the outside user may originate calls over any or all of the System facilities.

Operating Procedure

1. The Calling Party (Public or Private Network) lifts handset. Receive Dial Tone.
2. The Calling Party dials Automated Attendant Number. Receive Ringback Tone through either the Public/ Private Network. Announcement asking the caller to dial the desired Telephone Number is provided.
3. The Calling Party dials the desired Telephone Number. Receive Ringback Tone (if not busy or restricted).
4. The Called Party answers.

Service Conditions

1. The Incoming Routes from the Central Office Lines for Automated Attendant cannot be used for REMOTE ACCESS TO SYSTEM [R-2].
2. When a Ringdown Incoming Call encounters registers all busy status, call termination is retried approximately one second later. The call will wait until one of the receivers becomes available, but will not be placed in a queue.
3. System Data can select to provide or not provide the first announcement for an Incoming Call terminated to Automated Attendant. If first announcement sending is not provided, the Calling Party hears Ringback Tone from the System.
4. When a Ringdown Incoming Call encounters Announcement Trunk Busy/Blockout for the first announcement status, the Calling Party hears Ringback Tone from the Originating Office until the call is connected to the Announcement Trunk. When a TIE Line Incoming Call encounters Announcement Trunk Busy/Blockout for the first announcement status, the system returns Dial Tone.
5. Redial or Forced Release after the second announcement may be selected in System Data:
   a.) When the Calling Party has dialed a Dead Number. (When Dead Number Announcement is provided, it takes priority over the second announcement.)
   b.) When the Calling Party has abandoned dialing.
   c.) When the connection is restricted:
      - The Called Station is restricted from Incoming Trunk Connection.
      - The Calling Party has dialed an Access Code for Trunk Call Origination and encounters Digit Restriction. When a Trunk Call has encountered an Access Code Restriction, the second announcement is provided (digit reception ready), and the Calling Party can redial.
      - The Calling Party has dialed an Access Code for Trunk Call Origination and encounters an Inter-digit Pause Time-out Connection Restriction.
      - The Calling Party originates a Trunk Call and encounters Trunk Busy Status.
   d.) When the Called Station does not answer (the threshold value is 30 seconds), except when CALL FORWARDING - DON'T ANSWER [C-3] is set on the Called Station.
   e.) When the Called Station is busy, except when CALL FORWARDING - BUSY LINE [C-2] is set on the Called Station.

6. During the first or second announcement, the system can receive DTMF Signals. However, when a DTMF Signal is included in the Recorded Announcement, the announcement stops (first digit reception). Then, after providing the second announcement, either redial or forced release is selected (inter-digit time-out).
7. After the first or the second announcement, the system does not return any Tone until register time-out.
8. For the first or second Register time-out, either transfer to Attendant, or forced release can be selected by the System Data. However, forced release is executed, regardless of the above-mentioned system selection, for Central Office Lines for which release cannot be monitored. Call termination to the Night Transfer Station and TAS connection is not available.
9. To provide or not provide the second announcement can be selected in System Data. When second announcement sending is not designated, the system sends out Dial Tone. The call is single/multiple connected to the second Announcement Trunk, and Ringback Tone is not returned before sending the second announcement.
10. When a call encounters Announcement Trunk Busy/Blockout status in a Single Connection to the second announcement, the system returns a Dial Tone.
11. Tandem Switching as Automated Attendant is a service for Trunk-to-Station Connection, Tandem Connection is not available. When Tandem Connection or Authorization Code service is required, this service is activated by changing the Incoming Routes to that of REMOTE ACCESS TO SYSTEM [R-2].
12. If the Calling Party does not dial the desired Telephone Number after the first announcement, the call can be transferred to the Attendant after a predetermined period of time (prepause time-out), and the Calling Party receives Ringback Tone. However, if the Attendant is not available, Reorder Tone is returned.
13. When an Incoming Call has terminated to a Station, the following services are available:

CALL FORWARDING - ALL CALLS	[C-5]
CALL FORWARDING - BUSY LINE	[C-2]
CALL PICKUP	[C-7, C-30]
CALL WAITING - TERMINATING	[C-12]
STATION HUNTING	[S-7, S-8, S-9]

14. When an Incoming Call has terminated to a Station, CALL FORWARDING - DON'T ANSWER [C-3] is not available.
15. DAT card or VS32 is (**Note 1**) required for this service.

    Note 1: For details on the announcement feature of VS32, see description relating to announcement feature in VS32 Conference Server [V-26].

16. The length of the announcement depends on the key setting of the DAT circuit card. In case of Multiple Connection, it is fixed at 30 seconds. In case of Single Connection, it may be the length of the actual message (including pause) to a maximum of 64 seconds. When the message ends, the Announcement trunk is released.
17. The second announcement is not provided to the ISDN trunk.
18. Since the announcement is provided for a maximum of 30 seconds, the message recording should be completed within 30 seconds, unless a Single Connection is used.
19. This feature is available for Stations in the FCCS Network only.

Programming

STEP 1: **ASYD** - SYS1, Index 7, Bit 5:

Processing on unobtainable Number/Dead Level Dialed, dialing abandoned by Calling Station, Connection Restriction, and Called Station unanswered.

0/1: Redial is prompted by sending the second announcement/forcible disconnect

Bit 6: Processing on PB Receiver Time-out.

0/1: Forcible Disconnection/Transfer to ATT

Index 128: Assign the timer value of Register Prepause Time.

Index 139: Assign the value of No Answer Timer (Call Forwarding - All Calls, Call Back, etc.)

Index 246, Bit 4:

Process of Incoming Calls (Ringdown) after they encounter their first, second PB receiver time-out in a sequence of Automated Attendant.

0/1: Forced Release/Transfer to ATT

STEP 2: **ARTD** - Assign the Route Data of the Trunk for AUTOMATED ATTENDANT.

RT:1

1-OSGS :2 4-INSG :2 or 7 6-TCL :4

Note: To change the data for existing route (RT), initialize or reboot the trunk device such as a circuit card or MG.

Assign the Route Data of the Digital Announcement Trunk.

Note: To change the data for existing route (RT), initialize or reboot the trunk device such as a circuit card or VS32.

STEP 3: **ATRK** - Assign the Trunk Data of both the Trunk for AUTOMATED ATTENDANT and the Digital Announcement Trunk. Also, assign the Trunk Data of PB Receiver (RT = 906).

STEP 4: **MBTK** - Cancel the make-busy of the Trunk assigned by ATRK.

STEP 5: **ACSI** - Assign CSI = 5 (Remote Access to Node) to the Route and Trunk Number of the Trunk for AUTOMATED ATTENDANT.

STEP 6: **ACFR** - Assign the Restriction Release of TSFI = 7 (Remote Access to System)

STEP 7: **AAED** - Assign the Announcement Equipment for AUTOMATED ATTENDANT.

Announcement Equipment for sending the first announcement.

(EQP = 34, C = 0, R = 1, A = 1, M = 0)

Announcement Equipment for sending the second announcement **Note 2**

(EQP = 35, C = 0, R = 1, A = 1, M = _) **Note 3**

Note 2: When ASYD, SYS1, Index 7, Bit 5=0, assign the Announcement Equipment to send the second announcement.

Note 3: When selecting M (0 = Single/1 = Multi) for the second announcement, take the following into account:
Do not assign the second announcement in the same Route as the first announcement when Multi (M = 1) is selected. In the case of Multi (M = 1), the DAT Trunk is forcibly released when 30 seconds have elapsed from the beginning of the second announcement. Note that the connection of the DAT Trunk still remains even if the playback reaches the end of the Recorded Announcement. In the case of Single (M = 0), the DAT Trunk is automatically released when the playback reaches the end of the Recorded Announcement.

STEP 8: **ASTD** - Assign the following State Translation Data:
STM = 0, STS = 2, ST = 1 (allows a Tandem Connection inclusive of COT)
STM = 4, STS = 0, ST = 1 (Remote Access to System in service)

STEP 9: **ARRC** - Cancel the restrictions on the Trunk Route for AUTOMATED ATTENDANT and the route for the Digital Announcement Trunk.
ICRT: (Trunk Route for AUTOMATED ATTENDANT)
OGRT: (Route for Digital Announcement Trunk)
ARI-D-RES

STEP 10: **ARAC** - Assign an Idle Station Number as the Remote Access Code.

FCCS Programming

This data setting describes only the additional or replacement commands used to activate FCCS Services. When used as an FCCS feature, these commands replace the non-network commands (for example, AAED is replaced by AAEDN). Note that the data setting for FCCS Service varies depending on the Node. **Note 4**

Note 4: When using ASYDL or ASYDN to program an FCCS feature, the basic programming of ASYD is required in addition to ASYDL or ASYDN.

STEP 1: **ALRTN** - Assign this command to connect the Logical Route to the Physical Route.

STEP 2: **ARTDN** - Assign the Route Data of the Trunk for AUTOMATED ATTENDANT.
LGRT:1
1-OSGS :2 4-INSG :2 or 7 6-TCL:4

Note: To change the data for existing route (RT), initialize or reboot the trunk device such as a circuit card or MG.

Assign the Route Data of the Digital Announcement Trunk.

Note: To change the data for existing route (RT), initialize or reboot the trunk device such as a circuit card or VS32.

STEP 3: **AAEDN** - Assign the Announcement Equipment for AUTOMATED ATTENDANT.
Announcement Equipment for sending the first announcement.
EQP = 34, Duration of Connection: Disconnect in 30 seconds. Do not Send RBT. Send Answer Signal. Multiple Connection = Single.
Announcement Equipment for sending the second announcement
EQP = 35, Duration of Connection: Disconnect in 30 seconds. Do not Send RBT. Send Answer Signal. Multiple Connection = Single.

STEP 4: **ARRCN** - Cancel the restrictions on the Trunk Route for AUTOMATED ATTENDANT and the Route for the Digital Announcement Trunk.
IC LGRT:Trunk Route for AUTOMATED ATTENDANT
OG LGRT:Route for Digital Announcement Trunk
ARI-D-RES

A-98 Answer Hold - Attendant

General Description

This feature enables an Attendant to answer an Incoming Call by pressing the ANSWER key or flashing ICI key. If the Attendant is already engaged in a call, pressing one of these keys places the first call on hold and automatically connects the second one. Use of the ANSWER key or ICI key speeds call handling, while Answer Hold prevents accidental call dropping.

Operating Procedure

To answer a call while engaged in a Two-party Call

1. While the Attendant is engaged in a Station/Trunk Call, another call terminates at the Attendant Console.
2. The Attendant presses the ANSWER key or ICI key. The original call is placed on hold (MUSIC ON HOLD [M-7]). The LED of the ATND key goes out and that of the HOLD key lights.
3. The Attendant is connected to the second party. (ATND key LED lights.)

Note: When the ANSWER key is pressed, the calls connected to the Attendant will be placed on hold (if there is an Idle Loop).

To answer a call while engaged in a Three-way Call

1. While engaged in a Three-way Call, another call terminates at the Attendant Console.
2. The Attendant presses the ANSWER key or ICI key. Currently lit LEDs of the ATND key and the ANSWER key go out. The TALK key LED goes out and that of the SRC key lights. The parties formerly connected to the Attendant remain connected to each other.
3. The LEDs of ICI key and the ATND key light. The Attendant is connected to the Incoming Call.

Service Conditions

1. Activation of this service is determined in individual (Two-party Call or Three-way Call) System Data.
2. When the Attendant is engaged in a Two-party Call:
   a.) If there is no Idle Loop when the Attendant presses the ANSWER key or ICI Key, the party currently connected to the Attendant will be placed on hold (MUSIC ON HOLD [M-7]), but the Attendant cannot answer the second call.
   b.) When the number of available Attendants exceeds the number of Incoming Calls, the status of each Attendant Console will be one of the following:
      (1) The connected call will be placed on hold, and the Attendant will be connected to the second call.
      (2) The connected call will be placed on hold, and the Attendant Console will become idle.
      (3) No change.
   c.) ATTENDANT LOOP RELEASE [A-6] is not available when the call is placed on hold.
3. When the Attendant is engaged in a Three-way Call:
   a.) If there is no Idle Loop when the Attendant presses the ANSWER key or ICI key, one of the following occurs:
      (1) When both the Calling and the Called Parties are Trunk and when the Called Party has not answered yet, the call is held on the Loop and the Attendant cannot answer the next Incoming Call.
      (2) When Called Party has already answered the call, the Attendant can answer the next Incoming Call (the previous call is not held on the Loop).
   b.) ATTENDANT LOOP RELEASE [A-6] is available when the call is placed on hold.

c.) When the number of available Attendants exceeds the number of Incoming Calls, the status of Attendant Console will be one of the following:
(1) The Attendant will be released from the current call and will be connected to the next Incoming Call.
(2) The Attendant will be released from the current call and Attendant Consoles will be idle.
(3) No change.

4. AUTOMATIC RECALL [A-8] is available for a call held by the Attendant.

Programming

ASYD - SYS1, Index 246, Bit 0: Answer Hold - Attendant from a Two-party Connection. 0/1 = Out/In Service

SYS1, Index 246, Bit 1: Answer Hold - Attendant from a Three-way Connection. 0/1 = Out/In Service

Assign the data of the Attendant Console by referring to the Assignment Procedure of ATTENDANT CONSOLE [A-3].

A-99 Authorization Code Display Elimination

General Description

This feature eliminates AUTHORIZATION CODE [A-20] and FORCED ACCOUNT CODE [F-7] from being displayed on the Desktop terminal for security purposes.

Operating Procedure

- Procedure 1
  (Access Code for Authorization Code/Forced Account Code) + (SST) + (Authorization Code/Forced Account Code) + (DT) + (OG Access Code) + (OG Code)
  Dial: (DT) - 10 + (SST) - 1234 + (DT) - 9 - 753 - 7000
  Display on Desktop terminal is: 10XXXX97537000
  XXXX = Authorization Code/Forced Account Code
- Procedure 2
  (OG Access Code) + (OG Code) + (SPDT) + (Authorization Code/Forced Account Code)
  Dial: (DT) - 9 - 753 - 7000 + (SPDT) - 1234
  Display on Desktop terminal is: 97537000XXXX
  XXXX = Authorization Code/Forced Account Code

Service Conditions

1. When the Station dials the AUTHORIZATION CODE/FORCED ACCOUNT CODE, Xs appear on the display.
2. LAST NUMBER CALLED [L-6] and SAVE AND REPEAT -D [S-24D] features keep dialed information, but display Xs when these Feature keys are pressed.
3. SPEED CALLING STATION -D [S-2D] keeps dialed digits, and all digits are displayed at the same time when this Feature key is pressed.
4. ACCOUNT CODE [A-18] is displayed on Desktop terminal.
5. This feature is available only for the Calling Party's Desktop terminal.
6. This feature is available for the following dialing procedures:

Procedure	Before Dialing	After Dialing
OG Access without Held Party	Yes	Yes
OG Access with Held Party	Yes	Yes
Attendant	No	No

7. If the network is connected by a Trunk (including a CCIS Trunk), and AUTHORIZATION CODE/ FORCED ACCOUNT CODE is requested at the other Node, this function is not applied.
8. The Calling Party needs to confirm SPDT/SST if AUTHORIZATION CODE/FORCED ACCOUNT CODE is required. The first digit of AUTHORIZATION CODE/FORCED ACCOUNT CODE may or may not be recognized as AUTHORIZATION CODE/FORCED ACCOUNT CODE, if the inter-digit timing immediately before the AUTHORIZATION CODE/FORCED ACCOUNT CODE is too short.

Programming

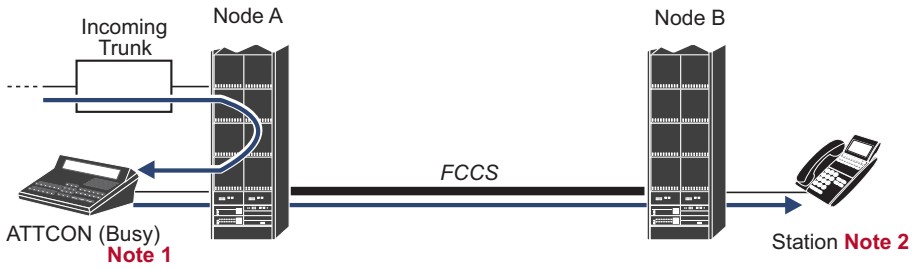
STEP 1: **ASFC** - Assign SFI 137 (Authorization Code Display Elimination) = 1 to the Originating Terminal.
STEP 2: **ASYD** - SYS1, Index 63, Bit 1. This enables Authorization Code Display Elimination.

A-105 Attendant Overflow

General Description

This feature allows Incoming Trunk Calls directed to the ATTENDANT CONSOLE [A-3] to overflow to a predetermined Night Transfer Destination Station. The Trunk Call will ring the Attendant for a pre-programmed time period before overflowing to the Destination Station.
This feature enables the Incoming Trunk Calls, directed to the ATTENDANT CONSOLE, to overflow to a predetermined Station in the FCCS Network via FCCS, also.

Note: This feature is only available in North America.



Note 1: When Attendant/Operator does not answer a call, the call overflows.
Note 2: This is the Station transfered by Night Connection Service.

Operating Procedure

1. An Incoming Trunk Call arrives at the ATTENDANT CONSOLE [A-3].
2. The Attendant is busy or on another call for longer than the assigned timer (the Incoming Caller has waited for 30 seconds or longer).
3. The Incoming Call is directed to the Destination Overflow Station.

Service Conditions

1. This feature is provided for all Incoming Ringdown Trunk Calls, DIRECT INWARD DIALING (DID) [D-8] Trunk Calls and LISTED DIRECTORY NUMBER (LDN) [L-24] Trunk Calls directed to the ATTENDANT CONSOLE [A-3].
2. This feature is available only while the system is in Day mode (not available for Night mode).
3. For DIRECT-IN TERMINATION [D-7] (DIT) Calls, this feature is available when the receiving Station has set CALL FORWARDING - ALL CALLS/BUSY LINE [C-5, 2] to the Attendant.
4. For DID Call Termination, this feature is available for LDN, Intercept, and Forwarded Calls to the Attendant, using CALL FORWARDING - ALL CALLS/BUSY LINE/DON'T ANSWER [C-5, 2, 3].
5. This feature is available for DID Calls routed to the Attendant via CCIS.
6. This feature is also not available for INDIVIDUAL ATTENDANT ACCESS [I-6], which allows a call to be directed to a particular Attendant Console.
7. This feature is not available concurrently with ATTENDANT NIGHT TRANSFER [A-19].
8. Overflowing Timer Value is determined by SYS1, Index 140 (the default is 40 seconds). When the destination does not answer the overflowed call, the Timer Value is reset to 40 seconds.

9. When the designated Station used for ATTENDANT OVERFLOW has set CALL FORWARDING - DON'T ANSWER [C-3] to the Attendant or an Outside Number, the Incoming Caller will be directed to TRUNK ANSWER FROM ANY STATION [T-8] (TAS) after the timer expires. If the Incoming Trunk is restricted from connecting to the TAS Circuit, the caller will continue to ring at the designated Station.

   Note: This feature is not available for calls to be transferred by using the following features:

CALL FORWARDING - ALL CALLS - OUTSIDE	*[C-28]*
CALL FORWARDING - BUSY LINE - OUTSIDE	*[C-60]*
CALL FORWARDING - DON'T ANSWER - OUTSIDE	*[C-74]*
LDN NIGHT CONNECTION	*[L-28]*
NIGHT CONNECTION - OUTSIDE	*[N-17]*

10. A PS cannot be assigned as Overflowing Destination.
11. The Incoming Call continues to ring at the Destination Station when designated FCCS Trunks are all busy, or locked out, etc.
12. FPC of a Node which transmits Day/Night Changeover information is determined by SYS1, Index 640, and the information is transmitted to a Node of FPC designated by SYS1, Index 704.
13. The Overflow Station and the Attendant must be in the same node.

Interactions

1. All Routes and Trunk Numbers must be assigned for Ringdown Trunks that are to be terminated to the Attendant Console and that require overflow.
2. For DID Trunks, the LDN will be terminated to the Attendant Console with overflow capabilities when CSI = 4.
3. ATTENDANT OVERFLOW [A-105] and DELAY ANNOUNCEMENT ATTENDANT [D-114] are mutually exclusive features. DELAY ANNOUNCEMENT ATTENDANT [D-114] takes priority over ATTENDANT OVERFLOW [A-105].
4. When CALL FORWARDING - ALL CALLS [C-5] or CALL FORWARDING - BUSY LINE [C-2] is set to the Overflowing Station, an overflowed call to the Station is transferred to the destination.
5. When CALL FORWARDING - ALL CALLS [C-5] is set to the Overflowing Station in the Hunting Group (STATION HUNTING - CIRCULAR [S-7]/STATION HUNTING - TERMINAL [S-9]/UNIFORM CALL DISTRIBUTION (UCD) [U-1]), an overflowed call to the Station is transferred by CALL FORWARDING - ALL CALLS. (Hunting Function is not activated).
6. When CALL FORWARDING - BUSY LINE [C-2] is set to the Overflowing Station in Hunting Group (STATION HUNTING - CIRCULAR [S-7]/STATION HUNTING - TERMINAL [S-9]/UNIFORM CALL DISTRIBUTION (UCD) [U-1]), an overflowed call to the Station is hunted (only when all Stations in the Hunting Group are busy is CALL FORWARDING - BUSY LINE [C-2] activated.)
7. When CALL FORWARDING - ALL CALLS [C-5]/CALL FORWARDING - BUSY LINE [C-2] to outside is set to the Overflowing Station, the call is not overflowed (the Attendant Console is called).
8. Overflowing Destination/Timer is determined by System Data of the Node that accommodates the Trunk which a call terminates.

Programming

STEP 1: **ASYD** - SYS1, Index 140. This determines the timer value at which the Incoming Call will overflow to the Destination Station.

SYS2, Index 15, Bit 1. Set to data "1". This enables the feature to be assigned via each Tenant.

STEP 2: **ACSI** - Assign CSI = 4: Direct-In Termination (Night Mode). This enables the overflow station to be assigned.

Note: Make sure you assign all Routes that apply in ACSI. For example, ISDN is on Route 1, and DID is on Route 5. They must be assigned in ACSI in order for the Overflow Call to be routed to the predetermined station.

STEP 3: **ACFR** - Allow TSFI 2 to the CCI type [1=LDN (Central Office Line), 4 = TIE Line] of Trunks that will overflow.

FCCS Programming

The following command assignments are required when your system is configured as an FCCS Network.

STEP 1: **ASYDL** - SYS1, Index 640, Bit 0 to Bit 7. Assign FPC of the Node which transfers the Day/Night Changeover information. **Note 1**

SYS1, Index 704 to Index 735. Assign FPC of the Node to which Attendant Console transfers the Day/Night Changeover information. **Note 1**

STEP 2: **ASYD** - SYS1, Index 140. This determines the timer value at which the Incoming Call will overflow to the Destination Station. **Note 1**

SYS2, Index 15, Bit 1. Allowance of ATTENDANT OVERFLOW. Assign 0/1 = Out/In Service. **Note 1**

STEP 3: **ANPDL/ANPDN** - Reserve a Number Level for Feature Access. Assign following data in accordance with predetermined Plan.

TN: Tenant Number
1stDC: 1st digit Code (0-9,*,#)
CI: Select Normal Service (N), Hooking Service (H), or Busy Service (B) from the combo box for Connection Indexes.
NND: Number of Necessary Digits (1-6)
BLF: Busy Lamp Field Display Activated on ATT

STEP 4: **ASPAL/ASPAN** - Determine the kind of service and the service Access Code to be executed.

TN: Tenant Number (1-255)
ACC: Access Code (maximum of six digits)
CI: Connection Status Index (N: Normal, H: Hooking, B: Busy)
SRV: Kind of Service (TELN: Telephone Number)
NND: Necessary Digit of Telephone Number

STEP 5: **ASDT** - Assign the Station Data for Destination Overflow Station.

TN: Tenant Number (1-255)
STN: Station Number
LENS: Line Equipment Numbers
TEC: Telephone Class
SFC: Service Feature Restriction Class

STEP 6: **ALGSL/ALGSN** - Assign Telephone Number to the Destination Station Number programmed by ASDT above.

TYPE = 1 (Line Equipment Numbers/Phantom Index is used to specify the Telephone Number)
UGN: User Group Number (1-255)
TELN: Telephone Number (maximum of 16 digits)
FPC: Fusion Point Code (Only ALGSN)
LENS/PHIN:Line Equipment Numbers/Phantom Index of the Destination Station Number
TYPE=2 (Station Number is used to specify the Telephone Number)
UGN: User Group Number (1-255)
TELN: Telephone Number (maximum of 16 digits)
FPC: Fusion Point Code (Only ALGSN)
TN: Tenant Number
STN: Station Number of destination

STEP 7: **ACSIL** - Assign Overflowing Destination of Attendant Console. **Note 1**

RT/LGRT:External Route Number/Logical Route Number (1-899)
TK: Trunk Number (1-255)
CSI: Connection Service Index, assign 4 [Direct-In Termination (Night Mode)].
UGN: User Group Number (1-255) **Note 2**
TELN: Telephone Number (maximum of 16 digits) **Note 2**

Note 1: Assign the data to the Node that a call terminates.

Note 2: This parameter is displayed when CSI = 2, 3 or 4.

STEP 8: **ACFR** - Assign the Call Forwarding Restriction data.

TSFI: Transfer Service Feature Index 2 = Direct Inward Dialing (DID)
CCI: Call Category Index [1 = LDN (Central Office Line), 4 = TIE Line]
RES: Restriction Data (0 = Transfer service is not allowed, 1 = Transfer service is allowed)

A-121 Add-On Conference - 8 Party

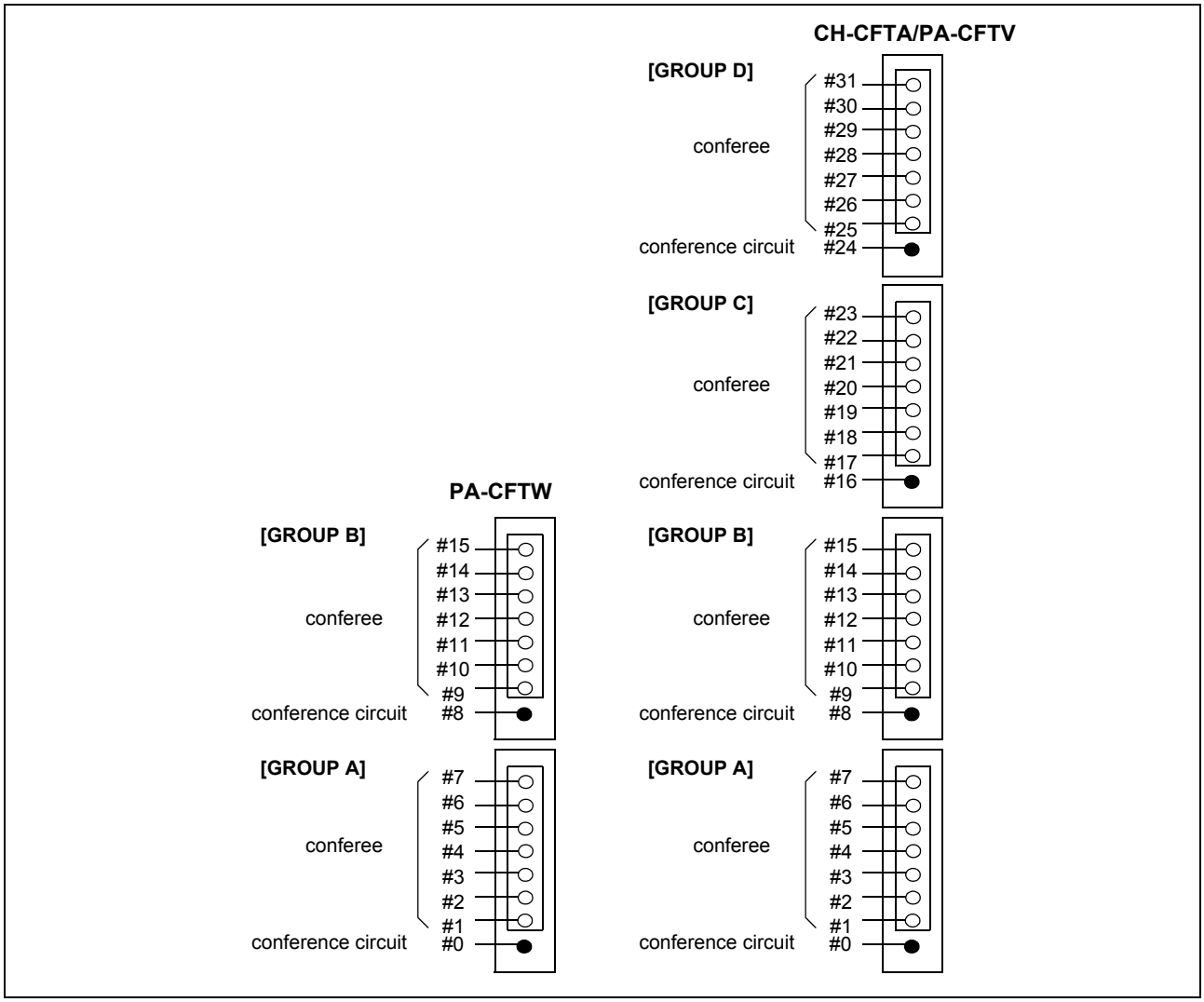
General Description

This service allows the Eight-Party Conference to be established any time by any Conferee by the use of the conference trunk.
The following words are defined in the description:

Item	Description
Conference member or Conferee	A member or station that is connecting to conference trunks or in a conference call.
Operating station	A station that calls a new conferee.
Called station	A station that is called by the operating station.
First station	The first station that is on a call with the operating station.

For using CFT cards:
The number of conferences per CFT varies depending on the CFT type as follows:

- PA-CFTW:8-party × 2 group
- PA-CFTV:8-party × 4 group
- CH-CFTA:8-party × 4 group



For using VS32:
A maximum of four conferences of Eight-Party Conference can be established by assigning and using the following combination of station numbers to the VS32 Conference Server channel.

CH#0 (Conference leader port) and CH#1 through #7 (Conferee port).

CH#8 (Conference leader port) and CH#9 through #15 (Conferee port).

CH#16 (Conference leader port) and CH#17 through #23 (Conferee port).

CH#24 (Conference leader port) and CH#25 through #31 (Conferee port).

ch	0	1	2	3	4	5	6	7	8	9	10	11	12	13	14	15	16	17	18	19	20	21	22	23	24	25	26	27	28	29	30	31
1																	Eight-Party								Eight-Party							
2	Eight-Party										Eight-Party						Eight-Party								Eight-Party							
3	Eight-Party										Eight-Party																					

Operating Procedure

There are the following operations:

- To establish an Eight-Party Conference (Basic operation):
- To operate establishing an Eight-Party Conference with a called station before the called station answers:

To establish an Eight-Party Conference (Basic operation):

1. Station A calls Station B. Station B answers.
2. Station A (or B) as an operating station performs switch hook flash (or presses the TRANSFER key) and dials Station C's number. Station C answers.
3. Station A (or B) (operating station) that had dialed Station C's number presses the CONF key. The Three-way Conference is established.

   Note: The conditions are as follows:
   - Pressing the CONF key to establish the conference is available for only Desktop terminals allowed Add-On Conference - 8 Party (ASFC/ASFCN SFI:131).
   - When the operating station is not allowed Add-On Conference - 8 Party (ASFC/ASFCN SFI:131) but Three-Party Conference trunk of RT909 is available, Three-Party Conference is established by pressing the CONF key and CONF is displayed on the LCD of Desktop terminals.

4. Operating Station A (B, or C) performs switch hook flash (or presses the TRANSFER key) and dials the number of Station D which is to be a called station. Station D answers.

5. Station A (B, or C) (operating station) that had dialed the number of Station D (called station) presses the CONF key. The Four-party Conference is established. In the same way, any user connected to the conference trunk can call other conference members asynchronously to establish the Eight-Party Conference by performing switch hook flash, dialing a station number to invite the conference and pressing the CONF key.

   - When an operating station is not allowed Add-On Conference - 8 Party (ASFC/ASFCN SFI:131), the station cannot be joined the conference and is restricted if the operating station user presses the CONF key.
   - If you try to invite the ninth conferee by performing switch hook flash and dialing the number of the ninth conferee, the operation is restricted when the CONF key is pressed.

6. When a conference member (station or trunk line) is released, a new conference member can be invited to the vacant conference port in the same way.

To operate establishing an Eight-Party Conference with a called station before the called station answers:

Note: SFI:131 (Add on Conference - 8 Party) and SFI:138 (Consultation Hold Enhancement) must be set to allowed by the ASFC/ASFCN command.

1. Station A calls Station B.
2. Station B answers. The two party calling is established.
3. Station A performs switch hook flash and dials Station C's number. The Station B user hears music on hold, and Station C is ringing.
4. < To make a conference from ringing state >
   When the CONF key of Station A is pressed before Station C answers, Station A is on a conference call with Station B using a conference trunk. Station C is called from the port of the conference trunk.

   Note: When CALL FORWARDING - DON'T ANSWER [C-3] is assigned to Station C, the call is transferred. However, if CALL FORWARDING - DON'T ANSWER - OUTSIDE [C-74] is assigned to Station C, the call is not transferred and Station C continues ringing.

5. In 4., when the Operating Station A user presses the Disconnect key assigned to Line/Feature keys or Soft keys, Station C can be disconnected. Station A and Station B are still on the conference call using the conference trunk.

   - The Disconnect key can be used by the operating station only. The target to be disconnected is the last station called by the operating station whose Disconnect key is pressed.
     When the operating station user presses the Disconnect key during the conference, the target to be disconnected is the opposite station on the call just before the conference call is established.
     The called stations that are invited by other conference members cannot be disconnected.
   - Pressing the Disconnect key by the conference member (Station B) who invites no stations to the conference is restricted.
   - When disconnecting the opposite party by using Disconnect key, RELEASE is displayed on the LCD of the terminal pressed the Disconnect key.

6. In 4., when Station C answers the call from the port of the conference trunk, the Three-Party Conference is established with Station A, B, and C.
7. For inviting Station D, any one of the conference members (Station A, B, or C) performs switch hook flash and dials Station D's number. Station D is ringing.
8. In 7., when the CONF key of the Station B is pressed before Station D answers, Station B is back to the conference call. Station D is called from the conference trunk.
9. < Transition to a conference from being in held state >
   In 7., when the Hold key of the Station D is pressed after Station D answers, Station B is placed on hold. When the CONF key of the Station B is pressed, Station B is back to the conference call. Station D will be in a state placing the conference trunk on hold.
   - Station B can invite other conference members while Station D places the conference trunk on hold.
10. When the Disconnect key assigned to Line/Feature keys or Soft keys of the Station B is pressed, held Station D is released. Station A, B and C are still on the conference call.
    - When disconnecting the opposite party by using the Disconnect key, RELEASE is displayed on the LCD of the terminal pressed the Disconnect key.
11. After canceling the hold state in 9. by Station D, the Four-party Conference is established with Station A, B, C and D by using the conference trunk.
12. In the same way, a maximum of eight parties conference can be established by operating switch hook flash, dialing station number of other conference members to invite to the conference asynchronously, and pressing the CONF key by any member connected the conference trunk.
13. When a conference member (station or trunk line) is released, a new conference member can be invited to the vacant conference port in the same way.

Service Conditions

<Important Notices>

1. To prevent a mistaken connection, the conference trunk is assigned in a different Tenant from the general Station/Trunk, and the connection between the conference trunk and the other Station/Trunk is set to be restricted (Inter-Tenant Restriction).

   (a) The conference trunk cannot be called from a Station, Outside Party, or the Attendant Console (Reorder Tone is sent out when attempted).

   (b) When a station (Station A) connected to the conference trunk transfers a call to another station (Station B) and Station A performs on-hook whether Station B answers or not, the conference trunk recalls to Station A but Station B will not be connected to the conference.
   However, if Station A performs on-hook while the call between Station A and Station B is being held, the conference trunk will not recall Station A and Station B will be connected to the conference.

   (c) The conference trunk cannot be assigned as the destination of CALL FORWARDING - ALL CALLS [C-5]/BUSY LINE [C-2]/DON'T ANSWER [C-3].

2. Assign System Data for Routine Diagnosis to check for Trunk Ineffective Hold Detect and Trunk Ineffective Hold Release for any conference trunks that may be held up.

<Interactions with Other Features>

1. This service can coexist in the same system with THREE-WAY CALLING [T-2] using the Three-way conference trunks in RT909 and RT913, and with the current Conference Service.
2. Status Number 9 of Soft Key Pattern (in ADSL) is supported during a Three-Party Conference. However, it is not supported during an Add-On Conference - 8 Party. If you customize Soft Key Pattern on an Add-On Conference - 8 Party, use Status Number 4 (converse) of Soft Key Pattern (in ADSL).
3. The following services are not available for the Stations/Trunks connecting to the Conference Trunk:

ATTENDANT CAMP-ON WITH TONE INDICATION	[A-1]
BUSY VERIFICATION	[B-3]
BROKERAGE - HOTLINE	[B-6] **Note 4**
BUSY VERIFICATION - CCIS	[B-9]
CALL TRANSFER - ALL CALLS	[C-11]
CALL WAITING - TERMINATING	[C-12]
CALL WAITING - ORIGINATING	[C-31]
EXECUTIVE RIGHT-OF-WAY	[E-1]
EXECUTIVE RIGHT-OF-WAY - CCIS	[E-8]
NAME DISPLAY	[N-28]
PRIVACY RELEASE	[P-18]

Note 4: When a conference member presses a Sub Line key assigned as a BROKERAGE - HOTLINE [B-6] during a conference, the conferee's station is released from the conference and originates a call to the BROKERAGE - HOTLINE [B-6] destination.

4. This feature does not support DIAL PLAN TO SUPPORT IP CENTREX FUNCTIONALITY [D-166].
5. Name data such as Name Display and System Phonebook should not be assigned for a conference trunk.
6. When you call a station set with CALL FORWARDING - DON'T ANSWER - OUTSIDE [C-74] while participating in a conference, the forwarding is invalid and the called station continues to ring. (This is a condition of CALL FORWARDING - DON'T ANSWER - OUTSIDE [C-74].) If the forwarding destination is a station, the forwarding is valid.
7. Traffic data of this feature is not reflected to PEG COUNT [P-7].
8. EXCLUSIVE HOLD -D [E-4D] is not supported for conferees.
9. A called station in a ringing state using VOICE CALL [V-2], INTERNAL ZONE PAGING [I-21] and VOICE CALL - CCIS [V-7] cannot participate in the conference by operating < To make a conference from ringing state >.
   However, the called station can participate in the conference by pressing the CONF key on the operating station after the called station answers the ringing call.
10. For FLEXIBLE NETWORK RE-ROUTING [F-41], if network failure occurs on a called station side when the station is called for a conference, automatic re-routing does not work.
11. Do not invite a terminal that is assigned as the Emergency Telephone on EMERGENCY CALL [E-11]. (Recall by performing switch hook flash or recall manually by pressing the TRANSFER key and automatically does not work.)

12. The conditions for the billing are as follows:

    (a) In case of Trunk Call Origination, the billing is performed on the Station which originated the Trunk Call until the Trunk is released. Therefore, there may be several billing messages sent for the same time for the same Station.

    (b) When a station joins a conference while the station is ringing or being held, station number of the conference port is recorded as “Calling Party” for the call between the station and the conference trunk in Station-to-Station Call Billing. (Setting SFI:222 = 1 (8-Party Conference Billing Enhancement) by the ASFC/ASFCN command, the conference member connecting with conference port 0 can be recorded as “Calling Party.”) On the other hand, when a station joins a conference while the station is engaged in a call or being held, the operating station is recorded as “Calling Party” in Station-to-Station Call Billing.
    When a trunk party joins a conference, the operating party is recorded as “Calling Party” in Trunk Call Billing, regardless of whether the party is ringing, engaged in a call or being held.

(c) The following table shows the billing information for Station-to-Station Call Billing by using a combination of SFI: 222 (8-Party Conference Billing Enhancement) and by system data (ASYDL, SYS1, Index 589, Bit 3=1) (Billing after transfer over different MA-IDs).

Preconditions / State		Billing after transfer over different MA-IDs: Out of Service (ASYDL, SYS1, Index 589, Bit 3=0)				Billing after transfer over different MA-IDs: In Service (ASYDL, SYS1, Index 589, Bit 3=1)			
		SFI: 222 = 0		SFI: 222 = 1		SFI: 222 = 0		SFI: 222 = 1	
		Calling Party	Called Party	Calling Party	Called Party	Calling Party	Called Party	Calling Party	Called Party
Ringing	Conference Transition (Two-way to Three-way) **Note 4**	Station number of conference port	The number of called station	The number of operating station	The number of called station	The number of conferees connected with conference port 0	The number of called station	The number of conferees connected with conference port 0	The number of called station
	A New Member Invitation after Conference Transition			The number of conferees connected with conference port 0					
Calling	Conference Transition (Two-way to Three-way) **Note 4**	No output billing information	The number of called station	No output billing information	The number of called station	The number of operating station	The number of called station	The number of operating station	The number of called station
	A New Member Invitation after Conference Transition								
Holding	Conference Transition (Two-way to Three-way) **Note 4**	Station number of conference port	The number of called station	The number of conferees connected with conference port 0	The number of called station	The number of operating station	The number of called station	The number of operating station	The number of called station
	A New Member Invitation after Conference Transition								

Note 4: In addition to the table information above, the information in Station-to-Station Call Billing for the call between the first station and the operating station is also output.

<Hardware/Software>

1. The following firmwares are required for the conference trunks:

Hardware	Conditions
CH-CFTA **PA-CFTW** **PA-CFTV**	No conditions.
VS32 **Note 5**	The firmware version SP-3891 CA-VS32 PROG-A Issue 10 or later.

Note 5: Available since FP95-115 V5.

2. The following devices cannot be a conference member:

   - Attendant Consoles
   - SP controlled SIP Terminals **Note 6**
   - SIP Handler controlled SIP Terminals **Note 7**
   - ISDN terminals **Note 6**
   - PS **Note 6**, **Note 8**
   - WLAN Handset (MG Series) **Note 8**
   - Softphone **Note 8**

   Note 6: When THREE-WAY CALLING [T-2] for Three-Party Conference trunks in RT909 is available for the terminals, the call is transferred to the current Conference Service by pressing the CONF key regardless of the setting for SFI:131.
   To be called from any conference member whose CONF key is pressed is restricted.

   Note 7: Assignment of RES=0 (Restricted) to SFI:131 (Add-On Conference - 8 Party) in Service Restriction Class for SIP Handler controlled SIP Terminals is required because this feature is not supported.
   The terminals can be a conference member as Called station or First station.

   Note 8: The device cannot be a conference member when using SOFT PHONE WIRELESS HANDSET [S-145] or SOFT PHONE WLAN HANDSET [S-159].

3. This feature cannot be activated by a Single Line Telephone.
4. On the LCD of the Desktop terminals, the number of the Stations and Trunks attending the Conference is shown as STA:3, TRK:2 in the upper line (displays the number of conference members). Name Display Service is not available.
5. During the Conference, the CONF lamp remains lit.
6. When using CFT cards with tone indication enabled, no tone is provided when a conference leader port (Port 0) leaves the conference.
7. In the same system, a conference trunk using CFT cards and a conference trunk using VS32 can coexist.

<Interactions with Networks>

1. In a FCCS network, a Conference Trunk is required in the node to which the first station to perform a conference operation belongs. For that reason, accommodate Add-On 8-Party Conference trunks to each node in the FCCS network, and configure Station Hunting - Terminal on each node. Assignment of pilot station number of Station Hunting - Terminal on the self node to the pilot station number on system data (ASYD, SYS1, Index 432 to 434) is also required.
2. For using VS32, only FCCS over IP (Peer-to-Peer) is supported in a FCCS network.

<Conditions Specific to This Feature>

1. The Conference Trunk will be treated as a Station (TEC=27).
2. Any one of the following lines can be a conference member:
   a.) Station
   b.) C.O. (COT and ISDN)
   c.) TIE **Note 9**
   d.) CCIS
   e.) SIP Network (Network Carrier and TIE)

   Note 9: When the called station is External Trunk of ACIS line, a conference cannot be established using < To make a conference from ringing state >. However, if the called station in ACIS answers the call from the operating station and the CONF key on the operating station is pressed, the conference is established.

3. In the following cases, this feature by pressing the CONF key is not available:
   a.) When service is not set in System Data (ASYD, SYS1, Index 432 to 434)
   b.) When service is restricted (SFI:131 Add-On Conference - 8 Party by the ASFC/ASFCN command) **Note 10**
   c.) When the Conference Trunk is not assigned, not accommodated, or busy.

   Note 10: Pressing the CONF key to establish the conference is available for only Desktop terminals allowed Add-On Conference - 8 Party (ASFC/ASFCN SFI: 131)

   In the cases above, if THREE-WAY CALLING [T-2] using Three-Party Conference trunks in RT909 is available for the terminals, a Three-Party Conference by pressing the CONF key is established and CONF is displayed on the terminal LCD.
4. The Stations connected to the Conference trunks can call the Conferees separately at any time.
5. When all of the ports of the Conference trunks are used by Conferees, no other Conferee can be added to the Conference. Pressing the CONF key, RST is displayed on the terminal LCD.
6. The Station/Trunk which is connected to the Conference trunks can be released and exit from the Conference at any time.

   Note: When the conferee exits the conference call, the other members may hear Reorder Tone (ROT) for a short time.

7. An Incoming Call from a Ringdown Trunk (Route Class Data: LSG=0 or LSG=3, except MC-MG(COT) **Note 11**), where a release cannot be detected, cannot be connected to the Conference trunks.

Note 11: The following equipment are included:

- MG (6COT)
- UG50 (COT-TYPE1)
- UG50-MC-MG (COT)
- UG50 (6COT)
- UG50 (COT-TYPE2)

8. When all the Trunks and Stations connected to the Conference trunks are released, the Conference Trunk becomes free. Until then, the same Trunk cannot be used by another group for a Conference.
9. When the Conferee presses the RETAIN CONF key, the LCD shows the follows for five seconds, depending on the connection:
   a.) One Station and multiple Outside Parties RETAIN CONF
   b.) One Station and one Outside Party REST
   c.) Multiple Stations REST

   Note: Only in case "a" above will the Conference status be retained.

10. When a Desktop terminal user who has Retain Conference presses the RETAIN CONF key a second time, Retain Conference is canceled. At that time, the LCD of the Desktop terminal indicates DISCONNECT CONF and the Conference Trunk is released after the Stations perform an on-hook.
11. When only two parties (Station/Station, Station/Outside Party, Outside Party/Outside Party) are connected to the Conference trunks, if either one of the parties is released, the Conference Trunk is released.
12. The Station setting Retain Conference can only cancel Retain Conference or perform on-hook (no other operation is available). When the Station performs on-hook, the RETAIN CONF lamp turns OFF.
13. When one Desktop terminal (station) and multiple outside parties (trunks) are connected to the conference trunk, the conference by the outside parties can continue by pressing the RETAIN CONF key and performing on-hook after setting retain conference from the Desktop terminal. The Desktop terminal that assigned the settings of the retain conference becomes idle and cannot return to the retained conference.

    - When one Desktop terminal and multiple outside parties are connected to the conference trunk, the conference trunk is released forcibly when the Desktop terminal goes on-hook without pressing the RETAIN CONF key.

    - In other cases, when a station that is not a Desktop terminal is connected with one outside party (not multiple parties), the conference trunk is forcibly released when the station goes on-hook.

    Note: Pressing the CONF key on the operating station before the called station answers or while it is ringing or placed on hold, the conference call can be established. (For details, see To operate establishing an Eight-Party Conference with a called station before the called station answers:.)
    For this operation, SFI:131 (Add on Conference - 8 Party) and SFI:138 (Consultation Hold Enhancement) is required for the operating station.

14. The operating station user can press the CONF key to connect to the conference before the called station answers or while the operating station is being held by the called station under the following conditions.
    a.) For using this feature, assign the System Data (ASYDL/ASYDN, SYS1, Index 678, Bit 3=1).
    b.) This feature is available only for Desktop terminals where SFI:131 (Add-On Conference - 8 Party) and SFI:138 (Consultation Hold Enhancement) are enabled by the ASFC/ASFCN command.

c.) Even if one conferee is left in the conference while a called station is ringing or being held, the conference call remains without release. The remaining station user hears no tone until the called station that is ringing or being held joins the conference or the remaining station invites a new conference member to the conference.

d.) When all conferees release the conference call while one or more called stations are ringing or being held, the called stations are automatically released and the conference trunk is also released.

e.) If the called station connects to the conference while hearing ringback tone or music on hold sent from the counter office, the ringback tone or the music is sent to all the other conferees. In this case, the operating station can release the called station by pressing the Disconnect key.

f.) If any of the items listed below exists, when an operating station user presses the CONF key when calling a transfer destination through CONSULTATION HOLD [C-17], then RST/REST will be displayed on the LCD of operating station. Likewise, when an operating station user presses the CONF key while being held by the transfer destination, the operating station will connect to the conference using Three-Party Conference trunk (Consultation Hold Enhancement).

- No trunk data for Eight-Party Conference is assigned
- No Eight-Party Conference trunk is accommodated in the system
- Eight-Party Conference trunks are all busy
- Eight-Party Conference trunks are in link block status

g.) If the operating station user presses the CONF key and connects to the conference while calling a target party via P2P-CCIS, the information (Station Number) about the connected Eight-Party Conference trunk appears on the LCD of the called station.

h.) Assign FKY=13 (Three-Way Calling) to Status Number: 15 (Others) for soft key pattern data using the ADSL/ADSLN command in order to operate a conference using soft keys.

15. The key of the Eight-Party Conference trunk must be set to not detect PB signals. For details on the switch settings, refer to Circuit Card Description.

16. When a conferee places a conference trunk on hold, the other conferees need to be set to not to hear the music on hold.

< For CH-CFTA/PA-CFTW/PA-CFTV >

For details on the switch settings, refer to Circuit Card Description.

< For VS32 >

Even if a conferee places a conference trunk on hold, the other conferees do not hear the music on hold.

< For all>

The other conferees hear music on hold in the followings:

- The Conference member is a trunk and hears the music on hold from the counter office.

17. For Eight-Party Conference trunk, the pilot number for station hunting set in system data (ASYD, SYS1, Index 432 to 434) can be assigned up to six digits. The telephone number of station number assigned to the Eight-Party Conference trunks can also be up to six digits.

18. A station number assigned to Level-0 of an Eight-Party Add-On Conference, will be the pilot number for the conference. Pilot numbers for all Eight-Party Add-On Conferences can be assigned in the ASHP/ ASHPN command. When dialed, this allows the system to hunt for a conference that is not in use.

19. When an operating station user dials any vacant number or vacant level during a conference call, the operating station does not connect to the announcement service but hears Reorder Tone (ROT). When the operating station is restricted from outgoing calls on a C.O. line or TIE line and originates a call with trunks or a TIE line as a Conferee, the operating station does not connect to the outgoing restriction announcement service but hears Reorder Tone (ROT).

20. The conditions of displaying the number of conference members on the LCD of Desktop terminals are as follows:
    a.) In the following cases, the number of Conferees are not displayed but the station number of conference trunk is displayed.
    - A call is transferred to other Desktop terminal by performing Call Forwarding - All Calls while participating in a conference.
    - A held call is returned to the holding party by pressing the ANSWER key while talking with another call after the call is held using CALL HOLD.

    b.) In the following cases, the number of Conferees are not displayed but a service corresponding to the various features (SERVICE DISPLAY -D [S-25D]) is displayed.
    - When a called station belonging to a Call Pickup Group is in a ringing state from a conference port < To make a conference from ringing state > or To operate establishing an Eight-Party Conference with a called station before the called station answers:, the called station is picked up by another member of the Call Pickup Group.
      (The service of CALL PICKUP - GROUP is displayed.)

    In addition, when establishing an Add-On conference by combining other features, the number of Conferees is not displayed but station number of the conference port may be displayed or SERVICE DISPLAY -D [S-25D] may be displayed depending on the combined services.
21. Terminals operate per the table below when a terminal failure or a communication failure (such as power-off of IP equipment, network failure or by a disconnected LAN cable) during an Eight-Party Conference.

Note: The failure includes the following:
- If a patrol function is activated by using RLST command while talking.
- If disconnected by the trunk ineffective hold detect during routine diagnosis.
- Re-registration of the terminals for conference members.
- Re-registration of the VS32.

Location of Failures	Terminal Behavior
The conference leader port 0	The calls connected with Eight-Party Conference are all released.
Conference member connected with the conference leader port 0	The calls connected with Eight-Party Conference are all released. **Note 12**
Any of the conference port 1 to 7	System releases only the conferees that are connected with the conference port occurred the failures. **Note 13**
Any of the conference member connected with the conference port 1 to 7	System releases only the conferees occurred the failures. **Note 13**

Note 12: When a modular cable is pulled out from Digital terminal accommodated in ELC card, all the calls connected with Eight-Party Conference are not released but only the conferee using the Digital terminals whose cables are disconnected is released.

Note 13: In this case of failure, the number of released stations or trunks is deducted from the number of Conferees displayed on the terminal LCD of other Conferees.

Programming

- < Basic system data settings >
- < Conference Transition Settings before Called Station Answers >

< Basic system data settings >

Note: When using a VS32, the system data of Internal PHE and SP-PHI are required to be assigned beforehand. For details, refer to the PH Data Setting in CHAPTER 2.

STEP 1: **ASYD** - SYS1, Index 8, Bit 0 to Bit 7. Assign the number of Tenants.
SYS1, Index 20, Bit 6=0 (Line Circuit Reverse Relay Control when Reorder Tone is connected to a station/trunk.)
SYS1, Index 92, Bit 4=0/1. This enables Tenant-to-Tenant Route Restriction Service.
SYS1, Index 94, Bit 0=0/1. This enables Tenant-to-Tenant Restriction Service.
SYS1, Index 432, Bit 0 to Bit 3. Assign the first digit of the Pilot Number used for this service. **Note 14**
SYS1, Index 432, Bit 4 to Bit 7. Assign the second digit of the Pilot Number. **Note 14**
SYS1, Index 433, Bit 0 to Bit 3. Assign the third digit of the Pilot Number. **Note 14**
SYS1, Index 433, Bit 4 to Bit 7. Assign the fourth digit of the Pilot Number. **Note 14**
SYS1, Index 434, Bit 0 to Bit 3. Assign the fifth digit of the Pilot Number. **Note 14**
SYS1, Index 434, Bit 4 to Bit 7. Assign the sixth digit of the Pilot Number. **Note 14**

Note 14: The conditions for setting ASYD, SYS1, Index 432 to 434 are as follows:
- One to six digits may be used.
- For the digit "0" use the HEX value format - "A."
- For the digit "*" use the HEX value format - "B."
- For the digit "#" use the HEX value format - "C."
- Assign the bits accordingly. If a digit is not used, assign those bits as "0". If a bit is used for 0, see the following example for Station 7001:
  SYS1, Index 432 is A7,
  SYS1, Index 433 is 1A,
  SYS1, Index 434 is 00.

STEP 2: **ASYDL** - This data must be assigned to all the nodes that have Add-On Conference - 8 Party implemented across FCCS.

Note: This step is required when using CFT cards or performing Add-On Conference - 8 Party on DTI-FCCS.

SYS1, Index 527, Bit 4=1 (Dial Pulse (DP) Relay Broadcasting to FCCS Link: Valid)

STEP 3: **ASDT** - Assign a station number to a conference trunk.

Note: This step is required when using CFT cards.

TN: Tenant Number **Note 15**
STN: Station Number **Note 16**, **Note 17**
LENS: Line Equipment Numbers
TEC: 27 (8 Conference Equipment) **Note 18**
RSC: Route Restriction Class **Note 18**
SFC: Service Feature Restriction Class **Note 18**

Note 15: To prevent dialing the station number of the conference trunk directly, assign the station number to a tenant that is different from general stations or trunks, and then restrict the connection with the general stations or trunks (Tenant Restriction).

Note 16: To prevent dialing the station number of the conference trunk directly, assign the conference station numbers to include a "#" or "*."

Note 17: When using only one 8-Party Add-On Conference trunk, the station number assigned to Level 0 of the conference trunk must be the same as the station number assigned in system data (ASYD, SYS1, Index 432 to 434).

Note 18: Assign a class that is different from general stations for the RSC and SFC.

Note: When using Telephone Numbers, the following setting is required. Also, assignment of the NDM commands (ASFCN, ASHCN, ASHPN, ARSCN) is also required after this step.
- ALGSN (Assignment of Telephone Station Data for NDM)
  Assign the Telephone Number for the station number assigned in the ASDT command.

STEP 4: **AIVSL** - Assign a station number to the following channels of VS32.

Note: This step is required when using VS32.

- CH#0 (Conference leader port) and CH#1 through #7 (Conferee port)
- CH#8 (Conference leader port) and CH#9 through #15 (Conferee port)
- CH#16 (Conference leader port) and CH#17 through #23 (Conferee port)
- CH#24 (Conference leader port) and CH#25 through #31 (Conferee port)

VS-ID: Assign the MAC Address of the VS32.

[BASE DATA] tab
CH: Assign the channel number of the VS32.
KIND: 8-CONF
CONF KIND: ADD-ON
TN: Tenant number **Note 19**

STN: Station number **Note 20**, **Note 21**
LENS: Line Equipment Numbers
TEC: 27 (8 Conference Equipment) (Display only) **Note 22**
RSC: Route Restriction Class **Note 23**
SFC: Service Feature Restriction Class **Note 23**

Note 19: To prevent dialing the station number of the conference trunk directly, assign the station number to a tenant that is different from general stations or trunks, and then restrict the connection with the general stations or trunks (Tenant Restriction).

Note 20: To prevent dialing the station number of the conference trunks directly, assign the conference station numbers to include a "#" or "*."

Note 21: When using only one 8-Party Add-on Conference trunk, the station number assigned to Level 0 of the conference trunk must be the same as the station number assigned in system data (ASYD, SYS1, Index 432 to 434).

Note 22: Selecting ADD-ON in CONF KIND, TEC=27 is selected automatically.

Note 23: Assign a class that is different from general stations for the RSC and SFC.

Note: When using Telephone Numbers, the following is required. Also, assignment of the NDM commands (ASFCN, ASHCN, ASHPN, ARSCN) is also required after this step.
- ALGSN (Assignment of Telephone Station Data for NDM)
  Assign the Telephone Number for the station number set by ASDT command.

STEP 5: **ASFC/ASFCN** - Specify the information related to Service Feature Restriction for the stations.

Enter a Service Feature Index (SFI) and then specify Restriction Data (RES) for the Service Feature Restriction Class (SFC) of the stations.

- For stations connected to the CFT cards
  SFI 32: RES = 1 (Line Circuit Reverse Relay Control (for 16LC) is available.)
- For stations to use this feature
  SFI 131: RES = 1 (Add on Conference - 8 Party is available.)

Assign the following data for the SFC of the conference trunks.

- For stations to allow Conference trunk.
  SFI 221: RES = 1 (Conference trunk is allowed.)

STEP 6: **ASHC/ASHCN** - Assign a STATION HUNTING - CIRCULAR Group for Stations programmed to LENS Levels 1 through Level 7 of the CFT cards.

STN: Station number of the first member station in the Hunt Group (Station number on port 1 of the conference trunks).

[Station List in Group]

STN: Station number of the member station in the Hunt Group (Station numbers on port 2 to 7 of the conference trunks).

<< When using ASHCN command >>

UGN: User group number

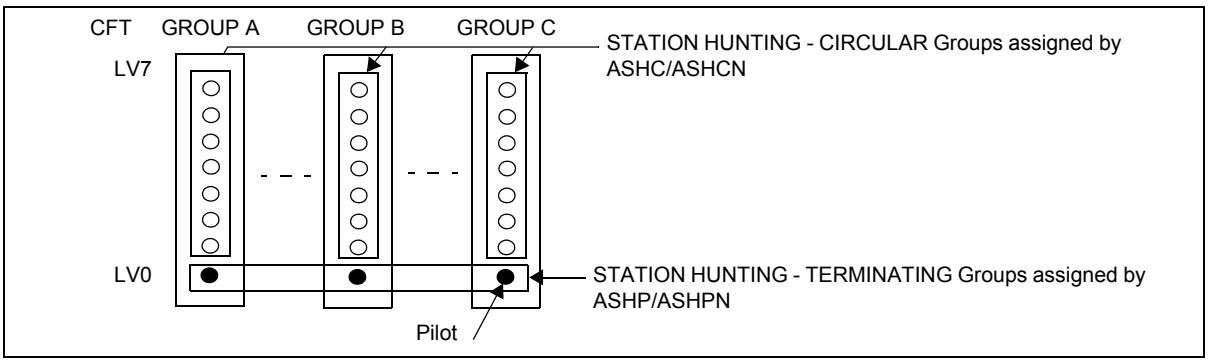
TELN: Telephone Number of the first member station in the Hunt Group (Station number on port 1 of the conference trunks).

[Station List in Group]

TELN: Telephone Number of the member stations in the Hunt Group (Station number on port 2 to 7 on the conference trunks).

FPC: Fusion Point Code (Display only)

STEP 7: **ASHP/ASHPN** - When the system accommodates multiple CFT cards, assign STATION HUNTING - TERMINATING Group for the Stations programmed to the LENS on Level 0 of each CFT card. See the figure below:



STN: Station number of the Pilot Station in the Hunt Group (Station number on port 0 of the GROUP A conference trunk). **Note 24**

[Station List in Group]

STN: Station number of the Pilot Station in the Hunt Group (Station number on port 0 of the conference trunks of GROUP B conference trunk. Continue to add the Pilot Station numbers for any other Eight-Party Add-On Conference Groups).

<< When using ASHPN command >>

UGN: User group number

TELN: Telephone Number of the Pilot Station in the Hunt Group (Station number on port 0 of the GROUP A conference trunk). **Note 24**

[Station List in Group]

TELN: Telephone Number of the Pilot Station in the Hunt Group (Station number on port 0 for the GROUP B conference trunk. Continue to add the Pilot Station numbers for any other Eight-Party Add-On Conference Groups).

FPC: Fusion Point Code (Display only)

Note 24: The station number of the Pilot Station in the Hunt Group must be the same as the station number assigned in system data (ASYD, SYS1, Index 432 to 434).

STEP 8: **AKYD/AKYD_T** - Assign the Line/Feature key Retain Conference [83] to any Desktop terminals that require the use of the Retain Conference feature.

Note: This step is required when assigning the Retain Conference feature to a Line/Feature key.

KYN: Key number
KYI: 1 (Function Key)
FKY: 83 (Retain Conference)

STEP 9: **ADSL/ADSLN** - Assign the features Three-Way Calling [13] and Retain Conference [83] to any Desktop terminal which require the use of RETAIN CONFERENCE.

Note: This step is required when assigning the Retain Conference feature to a Soft key.

SKP: Soft key pattern
SN: 15 (Others)
SKN: Soft key number
FKY: 13 (Three-Way Calling)
FKY: 83 (Retain Conference)

STEP 10: **ARSC/ARSCN** - Assign Restriction Data between Routes and the RSC.
STEP 11: **ATNR** - Assign Restriction Data between Tenants.

Note: Assign RES=0 (Restricted) to all the TRIs of the Tenants where the conference trunk is registered.

< Conference Transition Settings before Called Station Answers >

Note: Assign system data for CONSULTATION HOLD RELEASE [C-169] and < Basic system data settings > beforehand. For CONSULTATION HOLD RELEASE, refer to CONSULTATION HOLD RELEASE [C-169] in this manual.

STEP 1: **ASYDL** - Enable the operating station to connect to the conference before the called station answers or while the operating station is placed on hold by the called station.
SYS1, Index 678, Bit 3=1 (In service)

STEP 2: **ASFC/ASFCN** - Specify the information related to Service Feature Restriction for the stations.
Enter a Service Feature Index (SFI) and then specify Restriction Data (RES) for the Service Feature Restriction Class (SFC) of the stations.
- For stations to use Consultation Hold Enhancement
  SFI 138: RES = 1 (Consultation Hold Enhancement is available.)
- For stations to use Send Warning Tone to interrupted parties when Three-Way-Calling is established using Consultation Hold Enhancement
  SFI 140: RES = 1 (Send Warning Tone to interrupted parties when Three-Way-Calling is established using Consultation Hold Enhancement is available.)

When the conference members connecting with the conference port 0 will be recorded as “Calling Party” in Station-to-Station Call Billing to the conference call while ringing or placing on hold, assign the following to the SFC of the conference trunk.
- For stations to use 8-Party Conference Billing Enhancement
  SFI 222: RES = 1 (8-Party Conference Billing Enhancement is available.)

A-125 Analog Caller ID (Class SM)

General Description

This feature allows the Desktop terminal to display Calling Party's Number and/or Name on the LCD when a call from an analog Central Office Line is terminated. In addition, the Calling Number can be output as a part of SMDR information. Caller ID information can be displayed on an Analog Telephone with a display device or the display of a Desktop terminal.

Operating Procedure

No manual operation is required.

Service Conditions

1. This service is available when an Incoming Call from Central Office Line is terminated directly to a Station.
2. First ringing pattern for this service is two-seconds ON and four-seconds OFF. After that, ringing pattern assigned by System Data is applied.
3. The Calling Numbers received by using this feature or current MF/ISDN feature may be output to the SMDR expanded area.
4. A Calling Number cannot be received during conversation.
5. The Calling Numbers which may be added to SMDR output are the dial code only. The Calling Name is not added.
6. This service is not provided for the Attendant Console.
7. This service is not applied to SAVE AND REPEAT [S-24D] Service. The Calling Number received may not be saved.
8. A maximum of 256 calls per LP are available at the same time.
9. When an Incoming Call is sent as a test, the LCD does not display both the Number and the Name.
10. Display conditions are shown below. Also, the ID display of ISDN Service has the same conditions.
    a.) It is assigned by System Data in which display of the Calling Number or Name has priority on termination. When the Calling Name has priority, the LCD displays:

Calling Number XXXXXXXX **(Time Display)** **Calling Name**	* 1 * 2

When the Calling Number has priority, the LCD displays:

Calling Name XXXXXXXX **(Time Display)** **Calling Number**	* 1 * 2

*** 1** - In ISDN Service, the sub-address is displayed. When the Dterm does not receive the data pertaining to the sub-address, nothing is displayed.
*** 2** - If the prior display is not assigned by System Data, the other is displayed.

b.) The Calling Number and Name may be exchanged by pressing the Line/Feature key, according to the following conditions:
- When My Line is Prime Line, displays may be exchanged during termination or conversation.

- When My Line is not the Prime Line, displays may be exchanged only during conversation.

c.) When the Received Numbers of a Calling Number and Calling Name exceed the Display Field, the decision to put the digits to the right or left may be chosen in System Data. An "*" is indicated on the exceeded digit position.

d.) The display is flashing during termination. It is lit during conversation.

e.) The following are examples of displays when the received digits of a Calling Number and Name exceed the Display Field:

1234567*	Calling Number
(Time Display)	
ABCDEFGHIJKLMNOPQRSTUVW*	Calling Name

OR

***4567890**	Calling Number
(Time Display)	
***DEFGHIJKLMNOPQRSTUVWXYZ**	Calling Name

OR

ABCDEFG*	Calling Name
(Time Display)	
1234567890123456789012345*	Calling Number

OR

***TUVWXYZ**	Calling Name
(Time Display)	
8901234567890123456789012345	Calling Number

11. This feature is not supported via Analog MC.

Programming

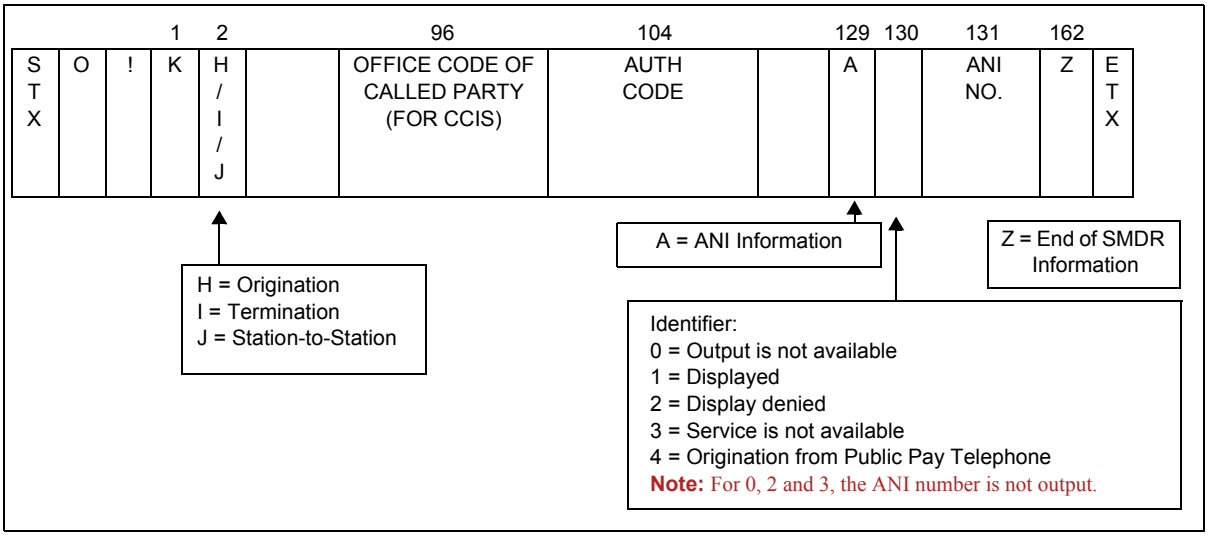
See TRUNK LINE APPEARANCE [T-23] for information on how to accommodate Central Office Line on a Desktop terminal Line/Feature key.

STEP 1: **ASYD** - SYS1, Index 241, Bit 6. 0/1 = When the length of Caller ID exceeds the limit, only the former/latter part of data appears on the LCD.

SYS1, Index 390, Bit 0. Caller ID display pattern on the LCD. 0/1 = Calling Number/Calling Name has priority.

SYS1, Index 390, Bit 2. When using Dterm two-line LCD, the lower LCD can be used for displaying Caller ID. 0/1 = Out/In Service.

When Calling Number is output to SMDR, the following expanded format is used:



SYS1, Index 288, Bit 5. Expanded SMDR Output for SMDR A Apparatus. 0/1 = Out/In Service.
SYS1, Index 290, Bit 5. Expanded SMDR Output for SMDR B Apparatus. 0/1 = Out/In Service.
SYS1, Index 292, Bit 5. Expanded SMDR Output for SMDR C Apparatus. 0/1 = Out/In Service.
SYS1, Index 294, Bit 5. Expanded SMDR Output for SMDR D Apparatus. 0/1 = Out/In Service.
SYS1, Index 296, Bit 0. Output of Calling Number (ANI information). 0/1 = Out/In Service. Assign data "1".
SYS1, Index 300, Bit 0. 0/1 = -/Billing information is sent to the Main Node in Expanded SMDR format in the case of CENTRALIZED BILLING - CCIS [C-55].

STEP 2: **ASFC** - Assign a Service Feature Restriction Class that allows SFI 14 (Station Message Detail Recording - Trunk Basis) to Stations allowed STATION MESSAGE DETAIL RECORDING (SMDR) [S-10].
Assign a Service Feature Restriction Class that allows SFI 146 (Analog Caller ID - Station) to Analog Stations allowed this service.

STEP 3: **ARTD** - Assign Route Class Data for the Central Office Trunk.
15-LSG :3 50-DPLY :1 **Note 1**
When Calling Number is output to SMDR:
10-SMDR :1

Note: To change the data for existing route (RT), initialize or reboot the trunk device such as a circuit card or MG.

Note 1: Up to 16 digits of a Caller ID can be shown on the display.

STEP 4: **ATRK** - Assign Trunk Data for the Central Office Trunk.

STEP 5: **MBTK** - Cancel the make busy of the Central Office Trunk.

STEP 6: **AKYD** - Assign Feature key Call ID Change Display [113].

STEP 7: **ADSL** - Assign ID CHG key, SN = 4 (Communication in Progress), Feature key Call ID Change Display [113].
This command is used for assigning a function to each Soft key on the Desktop terminal.

<When MG-COT is used for C.O. trunk>

STEP 1: **ASYD** - SYS1, Index 390, Bit 0: Caller ID display pattern on the LCD
0/1 = Calling Name/Calling Number has priority.

STEP 2: **ASYDL** - SYS1, Index 672, Bit 6=1: MG-COT Caller ID is in service

STEP 3: **ARTD** - Assign Route Class Data for the Central Office Trunk.

15-LSG : 3 (Caller ID [Loop])
-or-
15-LSG : 14 (Caller ID [Ground])
-and-
50-DPLY : 1 (Caller ID display is enabled) **Note 2**
114-DCH : 1 (Channel Integration is ON)
125-MG COT : 1 (Available)

Note: To change the data for existing route (RT), initialize the circuit card.

Note 2: Up to 16 digits of a Caller ID can be shown on the display.

STEP 4: **ARTI** - Assign Trunk Application Data.

68-VIR : 2 (MG Connection)

STEP 5: **AMGIL** - Assignment of Media Gateway Information.
Refer to the Peripheral Equipment Description for details.

FCCS Programming

This data setting describes only the additional or replacement commands used to activate FCCS Services. When used as an FCCS feature, these commands replace the non-network commands (for example, AAED is replaced by AAEDN). Note that the data setting for FCCS Service varies depending on the Node. **(Note 3)**

STEP 1: **ALRTN** - Assign this command to connect the Logical Route to the Physical Route.

STEP 2: **ARTDN** - Assign Route Class Data for the Central Office Trunk.

15-LSG :3 50-DPLY :1 **Note 4**
When Calling Number is output to SMDR:
10-SMDR :1

Note: To change the data for existing route (RT), initialize or reboot the trunk device such as a circuit card or MG.

Note 3: When using ASYDL or ASYDN to program an FCCS feature, the basic programming of ASYD is required in addition to ASYDL or ASYDN.

Note 4: Up to 16 digits of a Caller ID can be shown on the display.

A-127 Analog Caller ID - Station

General Description

This feature allows a Called Station to receive caller ID information (time, Calling Number and Name) from Central Office when a call terminates via the analog COT. The Calling Number received from the ISDN network, MF-ANI information received from Central Office with the MF feature or the calling Telephone Number within the System can change into caller ID information and be sent to the Called Station.
This feature is also provided for the Analog Station equipped with the function of receiving the Calling Number. This function is called Analog Caller ID - Station by Modem Sender.

Operating Procedure

No manual operation is required.

Service Conditions

<Caller ID with CH-12COTB/PA-8COTBJ>

1. This service is provided for Analog Stations.
2. This feature adheres to the following specifications and recommendation:
   Bellcore Caller ID (for North America) --- Bellcore (Telcordia) GR-30-CORE
3. Message types of caller ID information sent to Called Station are as follows:
   04H: Calling Number Delivery Information (for Calling Number)
   80H: Call Setup (for Calling Number and Name, or reason not to inform.)
4. When a call is answered by one party (intermediate station) then transferred to another (target station), the caller ID displayed on the target station is the intermediate station's number and not the original caller's number. The caller ID is not updated even when the intermediate party drops out of the call.
5. If a Calling Number exceeds ten digits, the first or last exceeded digits will not be displayed according to System Data.
6. This service is not provided if a call does not terminate directly on a line: in the case of termination using such services as Call Pickup, Executive Right of Way.
7. This service is not available when a Calling Party is the Attendant Console.
8. This service is not available for the 1A2 key telephone.
9. This service is not available in Boss - Secretary service.
10. The first ringing upon termination is two-second ring and four-second silence. A ringer pattern is provided from the second ringing.
11. The following firmware is required to be equipped on COT card:

CIRCUIT CARD	FIRMWARE
PA-8COTBJ	SW-317 8COTBJ PROG-A
CH-12COTB	SP-4065 12COTB PROG-A

<Analog Caller ID - Station by Modem Sender compatible with Bellcore Caller ID>

1. This service is also provided for Analog Stations equipped with the function of receiving the Calling Number.

2. This feature adheres to the following specifications and recommendation:
   Bellcore Caller ID (for North America) --- Bellcore (Telcordia) GR-30-CORE
3. Message types of caller ID information sent to a called Analog Station are as follows:
   - Message type - 04H: Calling Number Delivery Information (for Calling Number)
   - Message type - 80H: Call Setup (for Calling Number, Name, or reason not to inform)
   - Parameter - 02H: Calling line DN parameter
     - 04H: Reason for absence of calling DN Parameter
     - 07H: Name
     - 08H: Reason for absence of name
4. When the reason not to inform the caller ID is received through Central Office Line, the reason is notified.
5. The function equipped with the Station for restricting the caller ID notification can not be used.
6. The caller ID notification can be restricted only when originating the Trunk Call.
   When restricting the caller ID notification on each conversation, dial the number following the Numbering Plan preassigned at the self-Office.
7. Station allowed Analog Caller ID - Station by Modem Sender (ASFC, SFI 168) can receive a maximum of 10 digits per Calling Number.
   Station allowed Expanded Analog Caller ID - Station by Modem Sender (ASFC, SFI 194) can receive a maximum of 20 digits per Calling Number. If a Calling Number exceeds 10 or 20 digits, the first or last exceeded digits are not displayed according to the System Data (ASYD, SYS1, Index 241, Bit 1).
8. This feature is available for Station allowed Analog Caller ID - Station by Modem Sender (ASFC, SFI 168) or Expanded Analog Caller ID - Station by Modem Sender (ASFC, SFI 194). When a Station is allowed both SFI 168 and SFI 194, SFI 194 has higher priority than SFI 168.
9. When a call is answered by one party (intermediate station) then transferred to another (target station), the caller ID displayed on the target station is the intermediate station's number and not the original caller's number. The caller ID is not updated even when the intermediate party drops out of the call.
10. The caller ID notification is not provided for the following services because the call is not terminated to line using the services.
    CALL PICKUP - GROUP [C-7]
    CALL WAITING-TERMINATING [C-12]
    CALL WAITING-ORIGINATING [C-31]
    EXECUTIVE RIGHT-OF-WAY [E-1]
    Also, the caller ID notification is not provided to the Boss Station for Boss - Secretary service.
11. The system starts to count for No Answer timer when the caller ID notification process is complete.
12. The Incoming Call using this service is also counted as the maximum number of calls terminated at the same time in a system.
13. Since some telephones can not display the Telephone Number using "*" or "#" (other than the numerical digits), Telephone Number of numerical digits is recommended to use.
14. The maximum number of Register/Sender Trunk in an LP is 64. And, when the maximum number of Register/Sender Trunks is assigned (64 × LP no.), the ISDN/CCIS call can not be originated or terminated (the same as the usual system condition).
15. The following messages are notified when the reason for restricting caller ID notification is received: P (Privacy Security) and O (Outside Area where Service is not provided
16. Calling Number conversion (add or delete preassigned numbers) is not available when the caller ID information is received via Analog Caller ID Trunk.
17. When the Calling Party or the intermediate party is Attendant Console, or the Calling Party number is not received when the call is terminated via Central Office Line or TIE Line, the Calling Number can not be notified.

18. When the Name Display data is assigned to the Calling Party in Station-to-Station Calling, the Calling Party's name is also sent to the Called Party. Note that only 1 byte character data can be effective in this case. Since the name data is sent without confirming whether the proper character is used or not, characters that are not proper may cause abnormal status to the system.
19. When all the Sender Trunks are busy, the Calling Number cannot be notified.
20. To activate Analog Caller ID - Station by Modem Sender service in the FCCS Network, the Sender Trunk for sending Modem signal (RT=937, ATRK) should be accommodated in the all Nodes where Stations providing the service is installed.
21. Assigning SFI=168 and SFI=194 in ASFC/ASFCN command for non-Analog Caller ID station will cause a failure of displaying caller ID afterward. Therefore, be sure to assign SFI=168 and SFI=194 in ASFC/ASFCN command toward Analog Caller ID station.
22. The following circuit cards require the firmware condition as follows:

CIRCUIT CARD	FIRMWARE
PA-8RSTAE	SP-3899 8RSTAD PROG-B
PA-16LCBY	SP-3396 16LCBX PROG-A
PA-16LCCD	SP-3876 16LCCD PROG-A

Note: PA-16LCBY/PA-16LCCD and PA-8RSTAE must be used together for caller ID.

<Analog Caller ID - Station by Modem Sender compatible with ETSI Caller ID>

1. This feature adheres to the following specification and recommendation:
   ETSI Caller ID (for EMEA) --- ETSI EN 300-659-3 V1.3.1
2. Analog Stations compatible with ETSI Caller ID are available in this feature. (Be sure to execute performance test for each terminal before using this feature.)
3. When a call is answered by one party (intermediate station) then transferred to another (target station), the caller ID displayed on the target station is the intermediate station's number and not the original caller's number. The caller ID is not updated even when the intermediate party drops out of the call.
4. When the number of digits exceeds the limit to display, the caller ID is displayed on the called station deleting the last or first digits depending on the system data setting (ASYD, SYS1, Index 241, Bit 6).
   Up to 20 digits can be displayed on the called station.
5. The caller ID notification is not provided for the following services because the call is not terminated to line using the services.
   CALL PICKUP - GROUP [C-7]
   CALL WAITING-TERMINATING [C-12]
   CALL WAITING-ORIGINATING [C-31]
   EXECUTIVE RIGHT-OF-WAY [E-1]
   Also, the caller ID notification is not provided to the Boss Station for Boss - Secretary service.
6. One second ON - four seconds OFF for the first ringing is used for the incoming call termination. Then specified ringer pattern is applied from the second ringing.
7. If the Calling Party or the intermediate party is Attendant Console, or the Calling Party number is not received when the call is terminated via Central Office Line or TIE Line, “the cause of No ID” will be notified.
8. For the destination station of BLIND TRANSFER TO STATION [B-18], caller ID of original Calling Party is notified when the intermediate station disconnects the line before Register/Sender Trunk receives the information. Caller ID of the intermediate station is notified when the intermediate station disconnects the line after or during Register/Sender Trunk receives the information.
9. When a Sender cannot be seized, or information sending error occurs, caller ID is not notified. The called party receives the call without caller ID.
10. Caller ID is informed in Call Set-up message (CODE 80 Hex) from ISDN line. The following parameters are used in Call Set-up message for each specification.

PARAMETER TYPE	CODE (Hex)	ETSI CALLER ID SPECIFICATION
Date and Time	01	-
Calling Line Identity or Reason for absence of Calling Line Identity	02 04	X X
Calling Party Name or Reason for absence of Calling Party Name	07 08	- -
Visual Indicator	0B	-
Message Identification	0D	-
Last Message CLI	0E	-
Complementary Date and Time	0F	-
Complementary Calling Line Identity	10	-

PARAMETER TYPE	CODE (Hex)	ETSI CALLER ID SPECIFICATION
Number of Message	13	-
Type of Calling User	16	-
Network Provider Identity	30	-
Selection of Terminal Function	40	-
Display Information	50	-
Extension for network operator use	E0	-
Network operator parameter	E1 to FF	-

X: Available -: Not available

Note: Assign “0” to the related SFIs other than SFI 239 ETSI Caller ID (that is, SFIs 146, 153, 168 and 194).

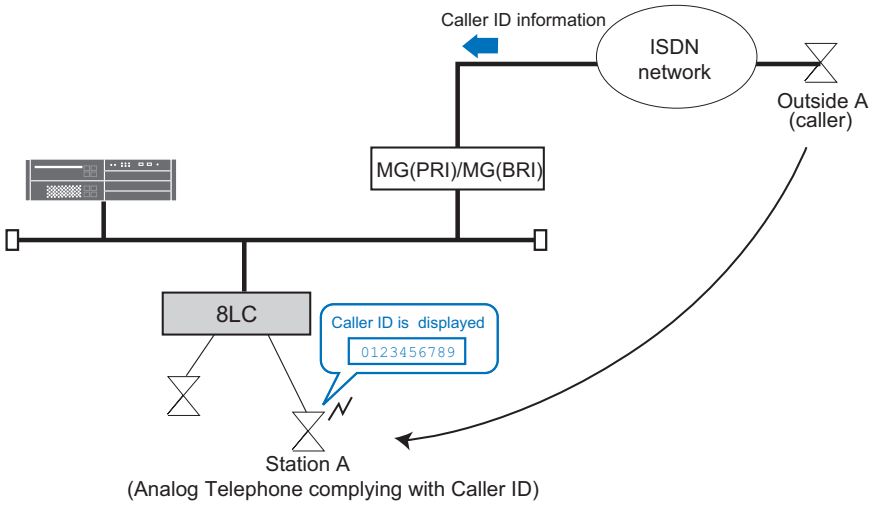
11. To activate Analog Caller ID - Station by Modem Sender service in the FCCS Network, the Sender Trunk for sending Modem signal (RT=937, ATRK) should be accommodated in the all Nodes where Stations providing the service is installed.
12. This feature is not supported via Analog MC.
13. The following circuit cards are required for this service:

CIRCUIT CARD	FIRMWARE
PA-8RSTAE	SP-3899 8RSTAD PROG-B Issue 5 or later version
PA-16LCCD	SP-3876 16LCCD PROG-A Issue 2 or later version

Note: PA-16LCCD and PA-8RSTAE must be used together for caller ID.

<Service Conditions for Analog Caller ID with 8LC (8LC card/UG50 (LC) [SIP]/UG50-8LC)>

Caller ID information (caller's number) can be displayed on called Analog Telephone stations connected to 8LC when incoming calls terminate from station, CCIS network, DP line, and ISDN line with caller ID. Caller ID for ETSI (EMEA) and Bellcore (North America) are supported depending on the Analog Telephone terminal types and line specification.



1. This feature adheres to the following specifications and recommendation:
   - ETSI Caller ID (for EMEA) --- ETSI EN 300-659-3 V1.3.1
   - Bellcore Caller ID (for North America) --- Bellcore (Telcordia) GR-30-CORE

   Analog TEL terminals compatible with ETSI/Bellcore Caller ID are available in this feature. (Be sure to execute performance test for each terminal before using this feature.)
2. The following firmware is required to be equipped on 8LC card.
   - For SCA-8LCA/SCA-8LCA-EMEA: SP-3893 8MC PROG-A Issue 4.0.0.0 or later version
   - For SCA-8LCC/SCA-8LCC-EMEA: SP-3983 8MC PROG-B Issue 1.0.0.0 or later version

   8LC not complying with caller ID cannot be used.
3. To activate this feature, assign the Service Feature Restriction Class (SFC) data depending on each specification.
   a.) For ETSI Caller ID
   SFI 239=1 (ETSI Caller ID is allowed) and SFI 146=0 (Analog Caller ID-Station is restricted)
   b.) For Bellcore Caller ID
   SFI 239=0 (ETSI Caller ID is restricted) and SFI 146=1 (Analog Caller ID-Station is allowed)
   c.) For Bellcore Caller ID complying with Single Message
   SFI 239=0 (ETSI Caller ID is restricted), SFI 146=1 (Analog Caller ID-Station is allowed), and SFI 153=1 (Analog Caller ID-Station (Single Mode) is allowed)

Note: When both SFIs 146 and 239 are assigned to be allowed, Bellcore Caller ID (SFI 146) takes precedence over ETSI Caller ID (SFI 239).

Note: When ETSI Caller ID is in service (SFI 239=1), Analog Caller ID-Station by Modem Sender feature is not activated.

4. When a call is answered by one party (intermediate station) then transferred to another (target station), the caller ID displayed on the target station is the intermediate station's number and not the original caller's number. The caller ID is not updated even when the intermediate party drops out of the call.
5. When the number of digits exceeds the limit to display, the caller ID is displayed on the called station deleting the last or first digits depending on the system data setting (ASYD, SYS1, Index 241, Bit 6). The maximum number of digits to be displayed on the called station is:
    - ETSI Caller ID --- 20 digits
    - Bellcore Caller ID --- 10 digits/20 digits
6. Caller ID information is not sent out for the features without line termination (e.g. Call Pickup, Executive Right of Way).
7. Caller ID information is not sent out when the call originator is an Attendant Console.
8. Caller ID information is not sent out for Secretary Call features.
9. The following ringing is used for the incoming call termination:
    - ETSI Caller ID --- 1 second ON - 4 seconds OFF for the first ringing
    - Bellcore Caller ID --- 2 seconds ON - 4 seconds OFF for the first ringing

    Specified ringer pattern is applied from the second ringing.
10. Branch station does not send out caller ID information.
11. Caller ID is informed in Call Set-up message (CODE 80 Hex) from ISDN line. The following parameters are used in Call Set-up message for each specification.

PARAMETER TYPE	CODE (Hex)	CALLER ID SPECIFICATION	
		ETSI	Bellcore
Date and Time	01	-	X
Calling Line Identity or Reason for absence of Calling Line Identity	02 04	X X	X X
Calling Party Name or Reason for absence of Calling Party Name	07 08	- -	X X
Visual Indicator	0B	-	-
Message Identification	0D	-	-
Last Message CLI	0E	-	-
Complementary Date and Time	0F	-	-
Complementary Calling Line Identity	10	-	-
Number of Message	13	-	-
Type of Calling User	16	-	-
Network Provider Identity	30	-	-
Selection of Terminal Function	40	-	-
Display Information	50	-	-
Extension for network operator use	E0	-	-
Network operator parameter	E1 to FF	-	-

X: Available -: Not available

12. Assigning SFI=146 and SFI=153 in ASFC/ASFCN command for non-Analog Caller ID station will cause a failure of displaying caller ID afterward. Therefore, be sure to assign SFI=146 and SFI=153 in ASFC/ASFCN command toward Analog Caller ID station.

13. To activate Analog Caller ID - Station by Modem Sender service in the FCCS Network, the Sender Trunk for sending Modem signal (RT=937, ATRK) should be accommodated in the all Nodes where Stations providing the service is installed.

Interactions

1. Only one Route Number can be assigned for each Sender and Register in a PA-8RSTAE circuit card.
2. In Hotel system, this service is provided only for administration stations.
3. This service is not provided for the call using OAI system.

Programming

<**Analog Caller ID - Station by Modem Sender feature for Bellcore Caller ID**>

STEP 1: **ASFC** - Assign the Service Feature Restriction Class to each Analog Station.

DAY/NIGHT: Day/Night Mode
TN: Tenant Number
SFI: 168=1 (Analog Caller ID Station by Modem Sender) **Note 1**

or

SFI: 194=1 (Expanded Analog Caller ID Station) **Note 1**
SFI: 239=0 (ETSI Caller ID is out of service) **Note 2**
When Calling Number Delivery (Message Type - 04H) is notified
SFI: 153=1 **Note 1**

Note 1: Assigning SFI=153, SFI=168 and SFI=194 in ASFC/ASFCN command for non-Analog Caller ID station will cause a failure of displaying caller ID afterward. Therefore, be sure to assign SFI=153, SFI=168 and SFI=194 in ASFC/ASFCN command toward Analog Caller ID station.

Note 2: Setting SFI 239=0 is required to activate “Analog Caller ID Station by Modem Sender” or “Expanded Caller ID Station by Modem Sender” for Bellcore Caller ID.

STEP 2: **ATRK** - Assign the Sender Trunk data.

RT: 937

Note: RT 937 must be programmed in level 4, 5, 6, or 7 for FSK Sender.
Level 0, 1, 2, or 3 may be used for RT 902 or PB/MF Sender.

Note: Optional ASYD, SYS1, Index 241, Bit 6. When a Calling Number exceeds ten digits:
0/1= The last/first exceeded digits are deleted.

<Analog Caller ID - Station by Modem Sender feature for ETSI Caller ID>

STEP 1: **ATRK** - Assign the Sender Trunk data.
RT: 937

Note: RT 937 must be programmed in level 4, 5, 6, or 7 for FSK Sender. Level 0, 1, 2, or 3 may be used for RT 902 or PB/MF Sender.

STEP 2: **MBTK** - Cancel the make-busy of Sender Trunk.

STEP 3: **ASFC/ASFCN** - Assign a Service Feature Restriction Class
D/N: Day/Night Mode
TN: Tenant Number
SFI: 168=0 (Analog Caller ID Station by Modem Sender is out of service)
SFI: 194=0 (Expanded Analog Caller ID Station is out of service)
SFI: 239=1 (ETSI Caller ID)
SFC: Service Feature Restriction Class
RES: 1 (Allowed)

STEP 4: **ASYD** - SYS1, Index 241, Bit 6: 0/1=Display the first/last 20 digits of Caller ID

Note: Assign ACNP and ACND commands for adding the access code of incoming route.

STEP 5: **ACNP** - Assign Calling Number Pattern.
IC/OG: IC: Incoming Call (Calling Party Recognition)
RT: External Route Number of incoming trunk.
CNP: Calling Number Pattern [1-255]

STEP 6: **ACND** - Assign Calling Number Data.
CNP: Calling Number Pattern assigned in ACNP command [1-255]
SKIP: Number of Skip Digits
ADD: Number of Digits of the Additional Number [0-24]
DC: Additional Digit Information [Maximum 24 digits]

<When MG-COT is used for C.O. trunk>

Note: Firmware SP3875 6MGCOT PROG-A Issue 4.0.0.0 or later version is required when using SCA-6COTB.

STEP 1: **ASYD** - SYS1, Index 390, Bit 0: Caller ID display pattern on the LCD
0/1 = Calling Name/Calling Number has priority.
STEP 2: **ASYDL** - SYS1, Index 672, Bit 6=1: MG-COT Caller ID is in service
STEP 3: **ARTD** - Assign Route Class Data for the Central Office Trunk.
15-LSG : 3 (Caller ID [Loop])
-or-

15-LSG : 14 (Caller ID [Ground])
-and-
50-DPLY : 1 (Caller ID display is enabled **Note 3**)
114-DCH : 1 (Channel Integration is ON)
125-MG COT : 1 (Available)

Note: To change the data for existing route (RT), initialize the circuit card.

Note 3: Up to 16 digits of a Caller ID can be shown on the display.

STEP 4: **ARTI** - Assign Trunk Application Data.
68-VIR : 2 (MG Connection)

Note: To change the data for existing route (RT), reboot the MG.

STEP 5: **AMGIL** - Assignment of Media Gateway Information.
Refer to the Peripheral Equipment Description for details.

<When 8LC Card>
STEP 1: **ASFC/ASFCN** - Assign a Service Feature Restriction Class
SFI: 146=0 (To restrict Analog Caller ID-Station for North America specification) **Note 4**
239=1 (To allow ETSI Caller ID)
STEP 2: **ASYD** - SYS1, Index 241, Bit 6: 0/1=Display the first/last 20 digits of caller ID
*Assign ACNP and ACND commands for adding the access code of incoming route.
STEP 3: **ACNP** - Assign Calling Number Pattern.
IC/OG: IC: Incoming Call (Calling Party Recognition)
RT: External Route Number of incoming trunk.
CNP: Calling Number Pattern [1-255]
STEP 4: **ACND** - Assign Calling Number Data.
CNP: Calling Number Pattern assigned in ACNP command [1-255]
SKIP: Number of Skip Digits
ADD: Number of Digits of the Additional Number [0-24]
DC: Additional Digit Information [Maximum 24 digits]

<When 8LC card/UG50 (LC) [SIP]/UG50-8LC>

STEP 1: **ASFC/ASFCN** - Assign a Service Feature Restriction Class

SFI: 146=1 (To allow Analog Caller ID-Station for North America specification) **Note 4**
153: 0/1=To restrict/To allow Caller ID-Station (Single Mode) **Note 4**, **Note 5**
239=0 (To restrict ETSI Caller ID)
194=1 (Expanded Analog Caller ID Station)

Note 4: Assigning SFI=146 and SFI=153 in ASFC/ASFCN command for non-Analog Caller ID station will cause a failure of displaying caller ID afterward. Therefore, be sure to assign SFI=146 and SFI=153 in ASFC/ASFCN command toward Analog Caller ID station.

Note 5: Assign SFI 153=0 when informing date/time, calling party name, and the reason of caller ID blocking. Assign SFI 153=1 when informing the caller number only.

STEP 2: **ASYD** - SYS1, Index 241, Bit 6: 0/1=Display the first/last 10 digits of caller ID

Note: Assign ACNP and ACND commands for adding the access code of incoming route.

STEP 3: **ACNP** - Assign Calling Number Pattern.

IC/OG: IC: Incoming Call (Calling Party Recognition)
RT: External Route Number of incoming trunk.
CNP: Calling Number Pattern [1-255]

STEP 4: **ACND** - Assign Calling Number Data.

CNP: Calling Number Pattern assigned in ACNP command [1-255]
SKIP: Number of Skip Digits
ADD: Number of Digits of the Additional Number [0-24]
DC: Additional Digit Information [Maximum 24 digits]

A-139 Account Code Override

General Description

This feature allows an Account Code to be entered during a conversation, overriding the original Account Code entered for that conversation.

Note: This feature is available in North America only.

Operating Procedure

1. While in conversation with an Incoming or Outgoing Call, the Station user presses the Flash or the TRANSFER key, enters the Account Code Override Access Code followed by the desired Account Code.
2. The Station user will hear a short tone and be returned to the original conversation, when the user hangs up, the last entered Account Code will be output to SMDR.

Service Conditions

1. This feature is not available for Station-to-Station Calls.
2. Desktop terminal users may store, on a Station Speed Dial Button, the TRANSFER key Code and the Special Access Code for Account Code Override.
3. An Account Code with fewer digits than the maximum length defined may be used by pressing # after the Account Code is entered, to identify the end of the code.
4. All Interactions and Limitations of Account Code [A-18] apply to this feature.
5. Desktop terminal users may store, on a Station Speed Dial Button, the TRANSFER key Code and the Special Access Code for Account Code Override.

Programming

ASPA - For SRV = “SSCA: Service Code Appendix”, SIDA = 107, assign the Access Code. Assign the number of digits in the ASPA Service Feature Access Code plus the number of digits that will appear in the Account Code for the NND Parameter.

A-142 Anonymous Call Block

(Series 8800 software feature)

General Description

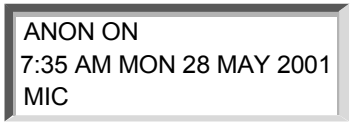
This feature allows a Station User to block calls that do not present a Calling Party Number. Anonymous Call Block also allows a Trunk Caller to be routed to a common announcement instead of receiving a Busy Tone.

Note: This feature is only available in North America.

Operating Procedure

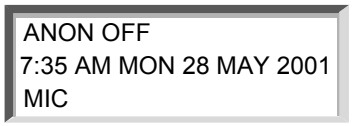
To turn on Anonymous Call Block

1. Lift the handset and dial the Anonymous Call Block ON Access Code (SSCA SIDA 131).
2. Wait for Service Set Tone.
3. Replace the handset.
4. The Desktop terminal LCD displays:



To turn off Anonymous Call Block

1. Lift the handset and dial the Anonymous Call Block OFF Access Code (SSCA SIDA 132).
2. Wait for Service Set Tone.
3. Replace handset.
4. The Desktop terminal LCD displays:



Service Conditions

1. Anonymous Call Block is normally off. To activate, use the Anonymous Call Block activation code.
2. When Anonymous Call Block is activated, any incoming trunk call that originally has Caller ID (Calling Party Number/ANI) but does not present a valid Calling Party Number or ANI will be restricted. (Example, a COT trunk that never had Caller ID would not present a valid Calling Party Number but would not be restricted.)
3. Call Forward-All [C-5] and Call Forward-Busy [C-2] take precedence over Anonymous Call Block.
4. To use this feature on a Desktop terminal Sub Line, Anonymous Call Block must also be set from each Sub Line.
5. For MF DID to receive reorder tone, when announcements are not used, set ASYD, SYS1, Index 59, Bit 2=1.
6. DIT Trunks will receive ring-no-answer if blocked using Anonymous Call Block.
7. Incoming call to stations that are Forward-All or Forward-Busy to restricted stations, will hear busy on Anonymous Call Block.
8. Restricted incoming calls will follow Call Forward Busy.

Programming

STEP 1: **ANPD/ANPDL/ANPDN** - Assign a number level for Anonymous Call Block.

STEP 2: **ASPA/ASPAL/ASPAN** - Assign the following access codes for CI (N: Normal):

Anonymous Call Block - ON: SRV = "SSCA:Service Code Appendix", SIDA = 131

Anonymous Call Block - OFF:SRV = "SSCA:Service Code Appendix", SIDA = 132

STEP 3: **DICBD** - Command shows if Anonymous Call Block is inactive/active.

AN BLK ACT = 0: Inactive

AN BLK ACT = 1: Active

STEP 4: **ASFC** - Assign SFI 189 RES = 1

Note: If Anonymous Call Block has been set, prior to SFI-189 being changed from a '1' to a '0', Anonymous Call Block will remain. Likewise the user will be restricted from turning it off.

STEP 5: **AAED/N** - Assign Announcement Equipment Number 63 to route Call Block calls to an announcement.

"R" must be set to "0" to send ring back. If "R" is set to "1", ISDN will not operate with the announcement. When using a DAT card for the announcement equipment, set the card to Multiple Playback.

STEP 6: **ARRC/N** - Allow Incoming Routes to Announcement Routes.

To delete

The deletion procedure is the reverse order of assigning the feature.

A-143 Assisted Conference 8 Party

General Description

This feature will allow a station user to move an existing 2 party call to the Add-On Conference - 8 Party [A-121] bridges without interrupting the current call.
It will also allow a station user to join an already in process Add-On Conference - 8 Party [A-121], call with out action by a current attendee.

This is accomplished by dialing the Assisted Conference Access code and then My-line of the terminal to conference with. The system will determine if the terminal is on a sub-line and use the sub-line automatically if needed. The calling party does not need to know the line number the called party is using.feature allows a Station User to block calls that do not present a Calling Party Number. Anonymous Call Block also allows a Trunk Caller to be routed to a common announcement instead of receiving a Busy Tone.

Note: This feature is only available in North America.

Operating Procedure

Example Environment

Station 1000 is an executive.
Station 2000 is 1000's assistant.
Station 2000 has a line appearance of 1000 appear in its key data along with a virtual line 2200.
*1 is assigned as the Assisted Conference Access Code. (ASPA)

Example Operation

1. Station 1000 (Executive) is on a call to an outside party.
2. Station 2000 goes off hook and dials the Assisted Conference Access code and the my line number of the Executive 1000 (*1-1000).
    a.) If Executive 1000 is using a sub-line the system will determine which sub-line is in use on 1000 and set the barge to that sub line.
3. All three parties are moved on to the conference bridge in a normal Add-on Conference 8 party [A-121] conference state.
4. Station 2000 (or 1000) can then use Add-on Conference 8 party [A-121] or Call Hold Conference [C-146] to add other parties.

(The same operation can be preformed if the original call is already an Add-on conference 8 party [A-121] call)

Example Operation Consult call with voice announce

1. Station 1000 (Executive) is on a call to another party on RT 1 TRK 1.
2. Station 2000 calls makes outgoing call or receives an incoming call on RT 2 TRK 1.
3. Station 2000 press transfer and dials Assisted Conference Access code and the my line number of the Executive 1000 (*1-1000).
    a.) RT 2 TRK 1 is placed on consultation hold.
    b.) If Executive 1000 is using a sub-line the system will determine which sub-line is in use on 1000 and set the barge to that sub line.
4. 1000, 2000 and RT 1 TRK 1 are moved on to the conference bridge in a normal 8 party add-on conference state. RT 2 TRK 1 remains on consultation hold.

5. Station 2000 can then press CONF to add RT 2 TRK 1 to the conference.
6. This becomes a normal Add-On Conference 8 party [A-121] state.

(The same operation can be preformed if the original call is already an 8 party add-on conference call)

Service Conditions

1. All the other A-121 service conditions apply.
2. Firmware must be SP-3654 (04.00 or above) for PA-CFTV and PA-CFTW.
3. The assistant must go off hook on an idle line to dial the Assisted Conference access code.
   a.) Or press transfer and get dial tone.
4. There must be an 8 party Add-on Conference port available for this feature.
5. OAI is not supported for 8 party Assisted Conference.
6. Wireless terminals are not supported for 8 party Assisted Conference.
   a.) If an 8 party Conference bridge is not available the assistant is given ROT.
7. Desktop terminal, Virtual, Outside Trunks, and Single line stations can be moved to the conference bridge using this feature.
8. Outside trunks must have answer and disconnect supervision to be used with this feature.
9. This feature is allowed on an SFC basis.
10. Emergency call [E-11] and 8 Party Assisted Conference are mutually exclusive features.
11. This feature will hunt for the line in use on the dialed executive station. (My line).
    a.) If the executive is on a sub-line and the My-line of the executive station is dialed, the system will automatically conference in the sub-line being used.
    b.) If the executive set does not have an active 2 party call, or 8 party conference and the executives My-line is idle, the system will return ROT.
    c.) If the Executive does not have an active line, but the My-line of the executive is in use by another station as a sub-line the conference will be established to the station using the executive's My-line.
    d.) The target station will not hunt to the add-on module of the dialed number.

        (a) All lines that need to be hunted must be on the physical Desktop terminal.

    e.) The system will only hunt the first 32 lines of the Executive station.
12. Once a party is in an 8 party conference state the assisted conference feature may not be used.
    a.) Only parties outside of the conference can use the assisted conference feature.
13. Each IP station or IP Trunk in the conference will use a separate IPPAD channel.
14. If the Executive or the party the executive is talking to is an IP station or Trunk they may here the cut through of voice because of IP packet delay between the terminal and the IPPAD.
15. See all other interactions for Add-On Conference - 8 Party [A-121].
16. For a quick connection PA-CFTV must set SW14-6 = On, SW14-7 = On and CH-CFTA SW13-5 = Off, SW13-6 = Off.
    This will allow immediate cut through of the voice path. If this is not done the Executive will hear momentarily a silence when the feature is used.

Programming

All normal Programming for A-121 apply plus the below steps.

STEP 1: **ASYD** SYS1, Index 17, Bit 3= 0/1 No Tone/Tone on barge in

STEP 2: **ASYDL** SYS1, Index 803, Bit 1=1 for the DISCONNECT Feature key adds.
SYS1, Index 803, Bit 2=1 for 8 party DISCONNECT key.

STEP 3: **ASPA/ASPAL/ASPAN**

TN: X
CI: N: Normal
H: Hooking, SRV = "SSCA: Service Code Appendix", SIDA = 163
Assign for the Assisted Conference access code.
CI is for N: Normal and H: Hooking.

STEP 4: **ASFC** - Assign a Service Feature Restriction Class that allows SFI 250, to perform Assisted Conference.

Assign a Service Feature Restriction Class that allows SFI 251.
0 = allow assist conference without station line appearance.
1 = Allow assisted conference only with station line appearance.

B-2 Busy Lamp Field - Flexible

General Description

This feature provides the Attendant with visual indications of line status - busy, idle, or LINE LOCKOUT [L-3] for a particular group of Stations via a designated lamp in the dial pad on the Desk Console.

Operating Procedure

1. Press an Idle Loop key.
2. Key in first digit for three-digit station numbers or first two digits for four-digit station numbers. For example:
   a.) To monitor Stations 200 to 209, dial "20".
   b.) To monitor Stations 2220 to 2229, dial "222".
3. A BLF lamp indicates the status of the station line, such as busy (steady on), idle (off), or Line LOCKOUT [L-3] (flashing).
   a.) ANPD - 1stDC = 3, NND = 3
   b.) ASPA - ACC = 300, SRV = "STN: Station Number"
   To monitor STN 3000 to STN 3009, key in 3000.

Note: For the Desk Console, dial all digits except the last digit of the desired number. The Desk Console indicates only ten Stations' status at once with the last digit.

Service Conditions

<Important Notices>

This feature does not work for a station number containing an asterisk (*) or number (#) sign.

<Interactions with Other Features>

1. For three-digit Telephone Numbers, the digital display will show the first two digits of the group being monitored (110s, 120s, etc.). Lamps 0-9 will provide visual indication of the status of each Station in the selected group.
2. For four-digit Telephone Numbers, the digital display will show the first three digits of the group being monitored (1110s, 1120s, etc.). Lamps 0-9 will provide visual indication of the status of each Station in the selected group.
3. For a Desk Console, the BLF display is available for three through five-digit Station Numbering Plans.

   Note: Not available for Station Numbering Plans with two digits or less.

4. The BLF display is enabled/disabled through the configuration setting of Desk Console.
5. The BLF display feature overrides the CALL WAITING LAMP [C-13] feature.
6. The lamp indication for LINE LOCKOUT [L-3] is an LED flashing at 60 pulses per minute.
7. LINE LOCKOUT [L-3] lamp indication can be provided by System Data programming.
8. The BLF does not indicate busy/idle status for Stations located in CCIS Satellite Offices.
9. For the FCCS Network, when assigning a Telephone Number to a Station, the Telephone Number is displayed for the Station.

10. For the FCCS Network, the BLF display is available for three-digit to five-digit physical station numbers/ Telephone Numbers.

    Note: For a Desk Console, BLF display is applicable for 3 through five-digit Station Numbering Plans.

11. For FCCS Network, when using BLF display among multiple nodes, all Attendant Consoles must be accommodated in one node.
12. To use this feature, the user must initialize the system after SYS1, Index 257, Bit 0=1 is assigned.

<Conditions Specific to MMG/UMGi/RNIP System>

For FCCS Network, set all the Attendant Consoles in one node. It is advisable to set Attendant Consoles in a node that is not part of a UMGi/RNIP system and has a MMG system.

Interactions

1. CENTRALIZED ATTENDANT SERVICE (CAS) [C-20]: At the CAS Main Location, when the ATTENDANT CONSOLE [A-3] is handling a call from a CAS branch over a Release Line Trunk (RLT), the BLF will give no indication of busy or idle status of Branch Location Stations.
2. The BLF indicates the station status after the hundreds digit is keyed in by the Attendant.
3. A two-digit Digital Display Indicator is used to display the thousands and hundreds digit of the displayed stations.
4. The BLF display is also available for Hotel system.
5. The BLF display is available for all stations including ISDN terminals, PS, and SIP terminals.
6. LINE LOCKOUT [L-3] of PS cannot be identified by the BLF lamp.
7. This feature is available even after a PS performed handover.
8. This feature is not available for a PS roaming to a visitor node.

Programming

STEP 1: **ASYD** - SYS1, Index 16, Bit 0 to Bit 5. Place one to six-digit station numbers in service. Assign 1FH, Busy Lamp Field - Flexible (BLF display) is available for three and four-digit Station Numbering Plans only.
SYS1, Index 70, Bit 1. Assign as data “0”. For Stations in Line Lockout, flash on BLF display.
SYS2, Index 7, Bit 0. Assign as data “1”. Enables BLF display on a tenant basis.

STEP 2: **ANPD** - For the levels reserved for station numbering, assign BLF display in service. Select Normal Service (N) or Hooking Service (H) from the combo box for Connection Indexes. NND value is assigned in accordance with the Numbering Plan.

Note: Consideration should be given when the system has stations of mixed-digit numbering (two- digit STN, three-digit STN, etc.) with the same first digit. In this case, the BLF display on Desk Consoles can only show the status of stations where the same value is provided for the Number of Necessary Digits (NND) parameter of both the ASPA and ANPD commands.

FCCS Programming

STEP 1: **ASYD** - SYS1, Index 16, Bit 0 to Bit 5. Place one to six-digit station numbers in service. Assign 1FH, Busy Lamp Field - Flexible (BLF display) is available for three and four-digit Station Numbering Plans only.
SYS1, Index 70, Bit 1. Assign "0". For Stations in Line Lockout, the lamp on BLF display.
SYS2, Index 7, Bit 0. Assign "1". Enables BLF display on a tenant basis.

STEP 2: **ANPDL** - For the levels reserved for station numbering, assign BLF display in service. Select Normal Service (N) or Hooking Service (H) from the combo box for Connection Indexes. NND value is assigned in accordance with the Numbering Plan.

STEP 3: **ASYDL** - SYS1, Index 640. Assign FPC (Fusion Point Code) of the Node installed Desk Console to all the Nodes in the FCCS network. For the Node installed Desk Console, assign the Self Node FPC.
SYS1, Index 650, Bit 6. Assign as data: 0/1 = When BLF information clear has started, the system message is sent/not sent.
SYS1, Index 650, Bit 7. Assign as data: 0/1 = BLF information is cleared/not cleared.

STEP 4: **ASFC -** Permits this feature with the Service Feature Restriction Class for each stations.
SFI 218 = 1 (Send BLF information over FCCS)

Note: Consideration should be given when the system has stations of mixed-digit numbering (two- digit TELN, three-digit TELN, etc.) with the same first digit. In this case, the BLF display on Desk Consoles can only show the status of stations where the same value is provided for the Number of Necessary Digits (NND) parameter of both the ASPAL and ANPDL commands.

B-3 Busy Verification

General Description

When an Attendant places a call to a busy Station, this feature allows the Attendant to break into the connection. When this feature is initiated from the Console, the system sends a Warning Tone to the Stations before establishing a Three-party Conference. Additionally, the Attendant Monitor Function can be activated depending on System Data.

Operating Procedure

To activate

1. Press an Idle Loop key and dial the desired Telephone Number. Receive the Busy Tone.
2. Press the BV key. Two-burst Tones are sent to the connected parties. Attendant may now monitor or join the connection.
3. Press the RELEASE key to disengage.

To monitor the call in progress before entering the Busy Connection from a Station

1. Press an Idle Loop key and dial the desired Telephone Number. Receive the Busy Tone.
2. Press the BV key. The Attendant can monitor the connection. (BV Lamp lights.)
3. Press the TALK key. A Warning Tone is sent, and a Three-way Call is established (TALK Lamp lights).

   Note: If the Called Station is a Desktop terminal, the LCD displays OPE BV for five seconds, then CNF/CONF.
   If Counter Party of the Called Station is a Desktop terminal, the LCD displays OPE BV for five seconds, then CNF/CONF.

4. Press the CANCEL key to disengage. Attendant returns to the Incoming Call. The originally connected parties remain connected.

Service Conditions

1. This feature may be used during Station-to-Station and Station-to-Trunk Calls (COT, FX, WATS, CCSA, or TIE Lines).
2. BUSY VERIFICATION cannot be established to a Station when the Station is:
   a.) Dialing
   b.) Listening to a Tone
   c.) Being rung
   d.) Being connected to an Attendant Loop
   e.) Activating a conflicting feature such as:

ATTENDANT CAMP-ON WITH TONE INDICATION	[A-1][A-45]
ATTENDANT OVERRIDE	[A-7]
BOSS - SECRETARY OVERRIDE -D	[B-5D]
CALL HOLD	[C-6]
CALL TRANSFER	[C-10] [C-11] [C-50] [C-54]
CALL WAITING - ORIGINATING	[C-31]
CALL WAITING - TERMINATING	[C-12]
CONSULTATION HOLD - ALL CALLS	[C-17, -17D] [C-58]
DATA LINE SECURITY	[D-12] [D-66]

DATA PRIVACY ON DEMAND	[D-17] [D-68]
DIAL ACCESS TO ATTENDANT	[D-2] [D-71]
EXCLUSIVE HOLD -D	[E-4D]
EXECUTIVE RIGHT-OF-WAY	[E-1]
NON-EXCLUSIVE HOLD -D	[N-7D]
PRIVACY -D	[P-14D]
PRIVACY RELEASE	[P-18]
THREE-WAY CALLING	[T-2]
VOICE CALL	[V-2]

f.) Modem Pooling Connection
g.) Data Call through Data Adapter
h.) Data Call through Data Module
i.) The Calling Station is connected to an Attendant Loop

Note: In each of the above instances, the Attendant will receive Reorder Tone.

3. Before the Attendant is added to an existing connection, a Two-burst Tone is applied to alert the parties of the Attendant's presence. Sending this Tone Burst can be suppressed by System Data.
4. BUSY VERIFICATION and ATTENDANT LOCKOUT [A-5] are mutually exclusive features.
5. This feature can be denied on DICTATION ACCESS [D-3] and PAGING ACCESS [P-1] Connections by assigned System Data assignment.
6. PAGING ACCESS [P-1]: BUSY VERIFICATION will be denied on a Station Line that is connected to Paging Equipment. If the initiating Paging Station goes on-hook and is waiting for a Call Back Call from the Paged Party, BUSY VERIFICATION will be allowed on that Station line.
7. DICTATION ACCESS [D-3]: BUSY VERIFICATION will be denied on a Station that is connected to a Dictation Trunk. Attempts to busy-verify a Station on this connection will result in a Reorder Tone being sent to the Attendant.
8. STATION HUNTING (CIRCULAR [S-7], SECRETARIAL [S-8], or TERMINAL [S-9]): If an Attendant attempts to busy-verify a busy Station Line in a Hunt Group, the call will hunt. That is, STATION HUNTING will take precedence over BUSY VERIFICATION. If all Stations in a Hunt Group are busy, BUSY VERIFICATION takes place on the dialed Station.
9. The BUSY VERIFICATION feature is assigned to Attendant Consoles only.
10. ATTENDANT LOCKOUT [A-5]: The Attendant is prevented from busy-verifying a Station Line that is connected to a Loop held on the ATTENDANT CONSOLE [A-3]. Because the LOCKOUT feature applies only to calls held on the Attendant Console Loop, BUSY VERIFICATION of those Station Lines will be denied, and the Attendant will hear the Reorder Tone.
11. CALL BACK [C-1]: When Station A is busy and Station B has activated this feature toward Station A, and then goes on-hook to wait for Station A to become idle, both Stations can be busy-verified. When Station A goes on-hook and the CALL BACK [C-1] process has begun, neither Station can be busy-verified until the Talking Connection between Stations A and B has been established. The Attendant will hear the Reorder Tone when BUSY VERIFICATION attempts are made during the denied period.
12. CALL FORWARDING (ALL CALLS [C-5], BUSY LINE [C-2]): If an Attendant attempts to busy-verify a (busy or idle) Station Line that has the CALL FORWARDING (ALL CALLS [C-5], BUSY LINE [C-2]) feature activated to another Station Line, the call will be forwarded to the other line, and BUSY VERIFICATION will proceed as if the Attendant had attempted to busy-verify the other line. If an

Attendant attempts to busy-verify a (busy or idle) Station Line that has the CALL FORWARDING (ALL CALLS [C-5], BUSY LINE [C-2]) feature activated to the Attendant, BUSY VERIFICATION will be denied.

13. CALL HOLD [C-6]: When two Stations have established a Talking Connection, and one of the Stations has a third party on hold, the two talking parties can be verified, but the party on hold cannot be busy-verified.
14. CALL WAITING [C-12, 31]: If an Attendant attempts to busy-verify a Station Line that already has a Waiting Call, BUSY VERIFICATION will be denied. Also, if the Attendant attempts to busy-verify a Station Line that is waiting for another line, the BUSY VERIFICATION attempt will be temporarily denied, because such Waiting Calls always involve either a connection to the Attendant position or the Calling Party is listening to the Ringback Tone.
15. UNIFORM CALL DISTRIBUTION (UCD) [U-1]: The Attendant can busy-verify a Station Line that is part of a UCD Group. That is, the BUSY VERIFICATION Call will not hunt. If BUSY VERIFICATION is attempted on the Group Number, UCD Hunting will take precedence over BUSY VERIFICATION.
16. It is determined in System Data to allow monitoring of the Busy Connection before establishing a Three-way Call.
17. The Attendant cannot monitor the call when activating the INDIVIDUAL TRUNK ACCESS [I-4].
18. When the Attendant presses the TALK key and enters a Three-party Call, the Attendant cannot return to the monitor mode.
19. While monitoring, only the operations via the TALK key or CANCEL key are valid.
20. If one of the originally connected parties disconnects while the Attendant is monitoring the call, the Reorder Tone is sent to the Attendant and the other party. The third party will remain on hold.
21. While being monitored with this feature, you cannot control the call in progress (for example, perform switch hook flash, place it on hold, or transfer it). Depending on Hooking Timer specification or data assignment of analog terminal, the ongoing station call may be disconnected by pressing Recall/Transfer key.
22. The billing during the monitor mode is continued from the status before monitoring.
23. The Node accommodating the Attendant Console requires CFT function for FCCS Service.
24. When Called Station activates CALL FORWARDING- BUSY LINE [C-2], and the call forwarding destination is busy:

    - When call forwarding destination is a station (including stations in FCCS network), ASYD, SYS1, Index 69, Bit 5 determines whether to break into the Forwarded-from Station, or into the Forwarded-to Station.

    - When the call forwarding destination is in outside of the self office, BUSY VERIFICATION is restricted and does not work.

Interactions

1. This feature is allowed whenever the Destination Telephone Set is engaged in any state other than:
   a.) a Three-way Connection,
   b.) a PRIVACY [P-14D, 15D] feature is activated,
   c.) connected to a Route programmed to have Line/Trunk, Privacy, CDN 44:PRV,
   d.) listening to a Tone (Dial Tone, Reorder Tone, Service Set Tone, Busy Tone, Ringback Tone).
2. With the ATTENDANT OVERRIDE [A-7] feature, the following Trunk types may be verified busy or idle status:
   a.) One-way Incoming Trunks,

b.) One-way Outgoing Trunks,
c.) Bothway Trunk.

3. Trunks in a Station-Trunk Connection, TRUNK-TO-TRUNK CONNECTION [T-10], and ATTENDANT CONTROLLED CONFERENCE [A-2] can be verified when busy.
4. ATTENDANT LOCKOUT [A-5]: The Attendant is prevented from verifying a busy Station Line which is connected to a Loop held at the Attendant Console. Since the LOCKOUT feature applies only to calls held on an Attendant Console Loop, BUSY VERIFICATION of those Station Lines will be denied and the Attendant will hear the Reorder Tone.
5. CALL FORWARDING - ALL CALLS [C-5], BUSY [C-2]: If an Attendant attempts to busy-verify a (busy or idle) Station Line that has the CALL FORWARDING (ALL CALLS [C-5], BUSY [C-2]) feature activated to another Station Line, the call will be forwarded to the other line, and BUSY VERIFICATION will proceed as if the Attendant had attempted to busy-verify the other line. If an Attendant attempts to busy-verify a (busy or idle) Station Line that has the CALL FORWARDING (ALL CALLS [C-5], BUSY [C-2]) feature activated to the Attendant, BUSY VERIFICATION will not be denied.
6. CALL HOLD [C-6]: When two Stations have established a connection and one of the Stations has a third party on hold, the two talking parties can be busy verified, but the party on hold cannot be busy verified.
7. DATA LINE SECURITY [D-12]: An Attendant cannot busy-verify a busy Station Line that has a DATA LINE SECURITY [D-12] Class of Service. If BUSY VERIFICATION is attempted, the Attendant will hear the Reorder Tone.
8. CALL WAITING [C-12, 31]: If an Attendant attempts to busy-verify a Station Line that already has a Waiting Call, BUSY VERIFICATION will be denied. Also, if the Attendant attempts to busy-verify a Station Line that is waiting for another line, the BUSY VERIFICATION attempt will be temporarily denied because such Waiting Calls always involve either a connection to the Attendant position or the Calling Party is listening to a Ringback Tone.
9. UNIFORM CALL DISTRIBUTING (UCD) [U-1]: The Attendant can busy-verify a Station Line that is part of a UCD Group. That is, the BUSY VERIFICATION Call will not hunt. If BUSY VERIFICATION is attempted on the Group Number, UCD Hunting will take precedence over BUSY VERIFICATION.
10. DATA PRIVACY - CCIS [D-67, 68]: To prevent interruption of Data Transmission, the Attendant cannot busy-verify a Station Line when this feature is used.
11. DICTATION ACCESS [D-3]: Dictation Access Trunks cannot be busy-verified by the Attendant. The Attendant will hear the Reorder Tone.
12. PAGING ACCESS [P-1]: Paging Access Trunks cannot be busy-verified by the Attendant.

Programming

STEP 1: **ASYD** - SYS1, Index 60, Bit 5. Is the Two-burst Warning Tone to be disabled? 0/1: No/Yes.
SYS1, Index 68, Bit 4. Trunk Select (TKSL) key must be enabled. Assign data “1”.
SYS1, Index 158. Timer between Tones.
SYS1, Index 179, Bit 4. Is ATT Monitor Service - in service? 0/1: No/Yes.
SYS2, Index 0, Bit 3. Attendant Override Service. Assign as data “1”.
SYS2, Index 0, Bit 4. Attendant Busy Verification Service. Assign as data “1”.
SYS1, Index 69, Bit 5. Busy Verification. 0/1 = Break into Forwarded-from Station/Break into Forwarded-to Station

STEP 2: **ATRK** - Assign these LENS as Route 909, Digital Conference Trunk. LENS consists of:
Group and Level Numbers for the Digital Conference Trunk are 210, 213, 216, 221, 224, 227, 232, and 235.

STEP 3: **ASFC** - for Stations subject to ATTENDANT OVERRIDE [A-7], assign a Class of Service that

allows SFI 4, [Executive Right-of-Way (Called)], and SFI 11 (Attendant Camp-On).

STEP 4: **ARTD** - When a Station is connected to a Trunk, CDN 44:PRV of the Route must be assigned as data "0" for ATTENDANT OVERRIDE [A-7] to be allowed.

Note: To change the data for existing route (RT), initialize or reboot the trunk device such as a circuit card or MG.

FCCS Programming

This data setting describes only the additional or replacement commands used to activate FCCS Services. When used as an FCCS feature, these commands replace the non-network commands (for example, AAED is replaced by AAEDN). Note that the data setting for FCCS Service varies depending on the Node.

STEP 1: **ALRTN** - Assign this command to connect the Logical Route to the Physical Route.

STEP 2: **ARTDN** - When a Station is connected to a Trunk, CDN 44:PRV of the Route must be assigned as data "0" for ATTENDANT OVERRIDE [A-7] to be allowed.

B-5D Boss - Secretary Override -D

General Description

This feature enables a Secretary to Voice Announce a call to a Boss when he is currently on his My Line.

Operating Procedure

- Station 200 - Boss
- Station 201 - Secretary

Boss is currently connected to Trunk A. Call at Trunk B, intended for 200, reroutes to Station 201 by way of Call Forwarding - Busy Line.

1. Secretary answers Station 201.
2. Secretary asks the caller to hold and presses the CALL HOLD [C-6] Feature key or TRANSFER key and CALL HOLD Code (CALL WAITING - ORIGINATING [C-31] key), Secretary hears Dial Tone.
3. Secretary presses the SPEED CALLING - ONE TOUCH [S-26D] key on which the BOSS - SECRETARY OVERRIDE Tone Code pause (RECALL key) and the Boss' Telephone Number (200) has been programmed.
4. Secretary hears Ringback Tone. Boss receives three bursts of Camp-On Tone (to distinguish it from the single burst for an Attendant Camp-On).

Boss' Response Options

Case 1:

1. Boss presses the ANSWER key and talks with Secretary (Trunk A is placed on hold).
2. Secretary hangs up, Boss speaks with Trunk B.

Case 2:

1. Boss presses the ANSWER key and talks with Secretary (Trunk A is placed on hold).
2. Boss presses 201 key and talks with Trunk B. Secretary hears Reorder Tone.
3. Boss presses ANSWER to reconnect with Trunk A. Trunk B is placed on hold.

Case 3:

1. If Boss does not respond to three bursts of waiting Tone, Secretary presses the RECALL key to resume talking to Trunk B.

Case 4:

1. Boss presses the ANSWER key and talks with Secretary (Trunk A is placed on hold). Boss denies the call.
2. Boss presses TRANSFER to return to Trunk A. Secretary can also press the TRANSFER key, so that both Boss and Secretary return to original callers.
3. Secretary is returned to Trunk B.

Service Conditions

1. The Secretary must have a Dterm for this operation. The Boss may have a Single Line telephone, but since no Line key or the CALL HOLD Feature key is available, the Boss's switch hook flashes to alternate between Trunk A and Trunk B. To disconnect from one Trunk, the Boss hangs up. The remaining Trunk recalls to the Single Line Set. The Held Party hears Ringback Tone.
2. If Boss and Secretary have Multi Line operations, they must have each other's My Line appearing on their Dterm.
3. This feature is available only when the Boss and the Secretary are on their own My Lines.

4. This feature must be allowed in CLASS OF SERVICE - INDIVIDUAL [C-15] via System Data programming.
5. BOSS - SECRETARY OVERRIDE is denied if the Boss is connected to the ATTENDANT CONSOLE.
6. If provided in the system, the Held Caller will hear MUSIC ON HOLD [M-7].
7. The call is recorded in SMDR for the Secretary's Station (201).
8. This feature is denied to a UCD [U-1] Group.
9. BOSS - SECRETARY OVERRIDE [B-5] overrides DO NOT DISTURB -D [D-11D] if set for the Boss.
10. BOSS - SECRETARY OVERRIDE [B-5] is denied under the following conditions:

ATTENDANT CAMP-ON WITH TONE INDICATION	[A-1]
ATTENDANT OVERRIDE	[A-7]
BUSY VERIFICATION	[B-3]
CALL BACK	[C-1]
CALL HOLD	[C-6]
CALL TRANSFER	[C-10] [C-11] [C-50] [C-54]
CALL WAITING - ORIGINATING	[C-31]
CALL WAITING - TERMINATING	[C-12]
CONSULTATION HOLD - ALL CALLS	[C-17] [C-58]
DATA LINE SECURITY	[D-12] [D-66]
DATA PRIVACY ON DEMAND	[D-17] [D-68]
DIAL ACCESS TO ATTENDANT	[D-2] [D-71]
EXCLUSIVE HOLD -D	[E-4D]
EXECUTIVE RIGHT-OF-WAY	[E-1]
NON-EXCLUSIVE HOLD -D	[N-7D]
OFF-HOOK QUEUING	[O-7]
OUTGOING TRUNK QUEUING	[O-2]
PRIVACY -D	[P-14D]
PRIVACY RELEASE	[P-18]
THREE-WAY CALLING	[T-2]
VOICE CALL	[V-2]

11. FCCS is not available for this feature.

Programming

STEP 1: **ASYD** - SYS2, Index 1, Bit 0 and Bit 1. Consultation Hold allowed or denied. 00: Denied; 01: Originating and Terminating Calls Allowed, Tandem Call Denied; 11: All Calls Allowed.

SYS2, Index 10. Assign data 10H to enable CALL WAITING ORIGINATING [C-31] Service.

STEP 2: **ANPD** - Reserve a Number Level for Service Feature Access. Select Normal Service (N), Hooking Service (H), or Busy Service (B) from the combo box for Connection Indexes. Assign NND in accordance with a predetermined Numbering Plan.

STEP 3: **ASPA** - Assign an Access Code to CALL HOLD set, SID = 1 for Connection Status Index (CI) of "H: Hooking". Assign an Access Code to CALL WAITING - ORIGINATING [C-31] service. SRV = "SSC: Service Code", SID = 5. Assign for a Connection Index of "N: Normal" and "B: Busy".

Assign an Access Code to BOSS - SECRETARY OVERRIDE Tone, SRV = "SSCA: Service Code Appendix", SIDA = 53. Assign for Connection Status Index of "N: Normal" and "H: Hooking".

STEP 4: **ASFC** - Assign Station to receive BOSS - SECRETARY OVERRIDE [B-5] a Feature Class that allows the following SFIs: 5 [Call Waiting -Originating (Calling)], 6 [Call Waiting - Originating (Called)], 10 (Call Hold), 11 (Attendant Camp On), and 51 (Boss - Secretary Features). Assign these SFIs as data "1".

STEP 5: **AKYD** - Optional. Assign Feature key CALL HOLD [10]. (The operation of TRANSFER key and Access Code is also available.) Optional. Assign to a Line/Feature key a CALL WAITING - ORIGINATING [C-31] key. Assign Feature key Call Waiting [4]. (The operation of the TRANSFER key and Access Code is also available).

STEP 6: **ADSL** - Assign SN = 5 (Consultation Hold) and Feature key Call Hold [10] and Feature key Call Waiting [4].

This command is required only for assigning the function to each Soft key.

B-6 Brokerage - Hotline

General Description

This feature provides a HOTLINE [H-1] function for Desktop terminal.

Operating Procedure

To access from Dterm

1. Lift handset or press the SPEAKER key.
2. Press a Sub Line key associated with the predetermined desired Station. The Destination Station is automatically dialed. Ringback Tone is heard, the Destination Station answers.
3. After completion of conversation, hang up or press the SPEAKER key.

   Note: To make another HOTLINE [H-1] Call instantaneously, press another Sub Line key without going on-hook and off-hook.

4. If the desired Station is busy - While hearing Busy Tone, press the EXECUTIVE RIGHT-OF-WAY (OVER) key. Receive Warning Tone. The caller can override the call in progress.

Service Conditions

1. One of the Line/Feature keys must be assigned as a My Line.
2. A Station to be set Hotline (Calling Party) cannot belong to Hunt [S-7, 8, 9] or UCD [U-1] Group.
3. Single Line Station or Virtual Lines (SOFTWARE LINE APPEARANCE [S-33]) can be assigned as a BROKERAGE - HOTLINE Destination. (ISDN Terminal cannot be assigned as the destination.)
4. When a Hotline Call is made from a Dterm, the Dterm's display shows the Destination Station Number but doesn't show the SERVICE DISPLAY [S-25] of Hotline.
5. This Hotline feature is applicable for Intra-Office Connection within the system.
6. My Line of a Dterm cannot be assigned as the HOTLINE Feature key.
7. Destinations must be Stations or an ATTENDANT CONSOLE [A-3] of the system.
8. When a destination is an ATTENDANT CONSOLE [A-3], the Hotline Call terminates as an ATND Call.
9. ATTENDANT CAMP-ON [A-1], EXECUTIVE RIGHT-OF-WAY [E-1], CALL WAITING [C-31] and CALL BACK [C-1] can be prohibited to the Hotline Connection through CLASS OF SERVICE - INDIVIDUAL [C-15].
10. When a Destination Station is busy, STEP CALL [S-13], CALL BACK [C-1], CALL WAITING [C-31], or MESSAGE REMINDER [M-19] cannot be activated.
11. Station Hunting works if the Destination Station is in a Hunt Group.
12. UCD function works if the Destination Station is a UCD Station, but Busy Tone is given if all the UCD Stations within the group are busy (no UCD queuing).
13. Calling party hears Reorder Tone (ROT) in the following cases.
    a.) When the Called Station has set CALL FORWARDING - ALL CALLS - OUTSIDE [C-28]
    b.) When the Called Station has set CALL FORWARDING - BUSY LINE - OUTSIDE [C-60] and line is busy
    c.) When the Called Station has set LOGGED OUT IP STATION-CALL DESTINATION [L-53] (for Trunk) and in logout status
14. Overriding is available only via the OVER key on Desktop terminal. If switch hook flash is attempted, the caller receives Reorder Tone.

15. In the following cases, overriding by the OVER key is not available, and the caller receives Reorder Tone.
    a.) the Conference-function Trunks are all busy.
    b.) the Destination Station is not engaged in a Two-party Call, as when
        - Call is terminating to the Destination Station.
        - The Destination Station has a call on hold.
        - The Destination Station is off-hook.
        - The Destination Station is already engaged in a Three-party Call.
    c.) the destination is the Attendant or is connected to an Outside Party.
16. When EXECUTIVE RIGHT-OF-WAY [E-1] is used with this feature, it is not activated under any of the following conditions:
    - when utilizing EXECUTIVE RIGHT-OF-WAY [E-1] using switch hook flash and Access Code.
    - when all Three-party Conference Trunks are busy.
    - when the destination is not in the Two-party Connection (an Incoming Call is in the middle of terminating to the destination or the destination places a call on hold or the destination is in off-hook status or the Three-party Connection including the destination has already been established).
    - when the destination is connected with the Attendant Console.
17. This feature can be activated via FCCS.
18. Only the Station assigned a Telephone Number is available for the Station assignment when activating this feature via FCCS.
19. The Station, the Attendant Console/Desk Console (Operator Call), and the Trunk can be a destination for Hotline.
20. When Call Forwarding - All Calls/Busy Line/Don't Answer Service has been set to the Called Station, these features are available for the call to the Station.
21. When a Calling Party encounters a busy Station, the call may bridge into the connection, depending on the data assigned by ASYD.

Programming

STEP 1: **ASYD** - SYS2, Index 0, Bit 1. Is Executive Right-of-Way in service? 0/1:No/Yes. **Note 1**

STEP 2: **ASDT** - Assign to a PA-16LC Circuit the following data:

TN: Tenant Number

STN: Station Number

LENS: Line Equipment Numbers (six digits) - Actual LENS for TEC 14, Virtual LENS for TEC 18

TEC: Telephone Class (1-31)

1: DP (10 PPS)
2: PB
3: DP/PB
12: Dterm
13: Data Terminal via Dterm
→ 14: HOTLINE
15: CAS Line
→ 18: Virtual Circuit

RSC: Route Restriction Class (0-15)

SFC: Service Feature Restriction Class (0-15)

STEP 3: **AHLS** - Assign the Station Numbers programmed above for the following parameters:
HOT TN:Hotline Tenant Number
HOT STN:Hotline Station Number
TYPE: 1 = Station
CON TN: Destination Tenant Number
CON STN: Destination Station Number

STEP 4: **ASFC** - When the station that Hotline is performed is Telephone Number(TEC=18), assign SFC of Telephone number as follows.
SFC: Service Feature Restriction Class[0-15]
SFI 36=1(Brokerage-Hotline)

STEP 5: **AKYD** - Assign to a Dterm a Line Appearance of the Hotline Station Number.
TN: 1
STN: Station Number
TP: Type of Dterm (0-3)
KYN: Key Number (1-40)
KYI: Service Index
0: Key not used
1: Feature Key
→ 2: Multi Line Key
KD: 0
TN, STN:Tenant Number and Station Number of Multi Line (Virtual)
RG: Ringing assignment per Multi Line Appearance
0: No ringing
1: Ringing in Day mode only
2: Ringing in Night mode only
3: Not Used
4: Ringing in Day and Night Mode - 30 second delay.
5: Ringing in Day and Night Mode - 20 second delay.
6: Ringing in Day and Night Mode - 10 second delay.
7: Ringing in Day and Night Mode.

STEP 6: **AKYD** - When Executive Right-of Way is enabled using ASYD, assign Executive Right-of-Way to a Line/Feature key. **Note 2**
KYN: Key Number(1-40)
KYI=1(Feature Key)
FKY=3(Executive Right-of-Way)

STEP 7: **ASFC** - When Executive Right-of Way is enabled using ASYD, lift restriction on Executive Right-of-Way. **Note 1**
SFI 3=1(Executive Right-of-Way)
SFI 4=1(Tone Block)
SFI 11=1(Data Line Security)

This data setting describes only the additional or replacement commands used to activate FCCS Services. When used as an FCCS feature, these commands replace the non-network commands (for

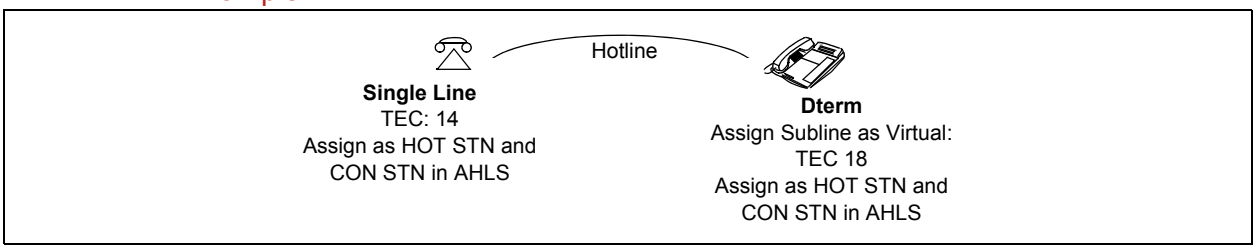
example, AAED is replaced by AAEDN). Note that the data setting for FCCS Service varies depending on the Node.

AHLSN - This data must be assigned to the NCN.

HOT UGN: Hotline User Group Number
HOT TELN: Hotline Telephone Number of the Calling Party
TYPE: 1 (the Called Party is a Station)
CON UGN: User Group Number of the Called Party
CON TELN: Telephone Number of the Called Party

Note 1: Set STEP 1,6,7 when Executive Right-of-Way is assigned.

Note 2: The following data should also be set in addition to the commands above.
The commands to be set to the Node accommodating the Calling Party: ASDT/AKYD
The following ASFC command data must be assigned at all Nodes to be set BROKERAGE - HOTLINE Service. SFI 36 (Brokerage - Hotline), RES 1 (allowed)
Example:



B-8 Boss - Secretary - Message Waiting Lamp Control

General Description

This feature allows a Secretary, via the Station, to control the Call Indicator Lamp or Message Waiting (MW) display on the Boss' Station.

Note: The Call Indicator Lamp is used as the Message Waiting Lamp.

Operating Procedure

1. CALL FORWARDING [C-3, 5] is not set on Boss' Station.
2. An Incoming Call for Boss' Station is answered on Secretary's Desktop terminal.
3. Secretary sets the Call Indicator Lamp or Message Waiting display on Boss' Station. See Procedure 1 below.
4. Boss's Station Call Indicator Lamp or the Message Waiting display is ON.
5. Secretary cancels the Call Indicator Lamp or Message Waiting display on Boss' Station.

To set Message Waiting at a Boss' Station

Procedure 1 (Secretary's Station requires a Desktop terminal with a LED)

1. While remaining connected to the Calling Party, press the MESSAGE WAITING Set key.
2. The LCD will indicate Message Waiting Set (for five seconds, in place of time display).
3. Message Waiting is set on Boss' Station. Call Indicator Lamp or the Message Waiting display is ON.
4. Secretary will not hear Service Set Tone.

Procedure 2

1. Secretary releases the call after taking the message and presses the RECALL key. Dial Tone is heard.
2. Secretary presses the MESSAGE WAITING Set key. LCD will indicate Message Waiting Set.
3. Special Dial Tone (SPDT) is heard and Secretary's LCD will display Message Waiting.
4. Secretary dials Boss' station number.
5. Secretary will hear Service Set Tone. Message Waiting is set to Boss' Station. Call Indicator Lamp or the Message Waiting display is ON.

To cancel Message Waiting indication

Procedure 1

1. While remaining connected, Secretary presses the MESSAGE WAITING Cancel key.
2. The LCD will indicate Message Waiting Cancel (for five seconds, in place of time display).
3. Boss' Message Waiting indication is then canceled. Call Indicator Lamp or the Message Waiting display is OFF.

Procedure 2

1. Secretary presses Boss' Line Appearance. Dial Tone is heard.
2. Press the MESSAGE WAITING Cancel key. Special Dial Tone (SPDT) is heard and Secretary's LCD will display Message Waiting.
3. Secretary dials Boss' station number.
4. Service Set Tone is heard. Secretary's LCD will display Message Waiting Cancel.
5. Boss' Message Waiting indication is canceled.

Service Conditions

1. Any number of Desktop terminal stations within the system can be assigned to activate this feature.
2. Secretary must have a Desktop terminal.
3. Any attempt to set BOSS - SECRETARY MESSAGE WAITING LAMP CONTROL to a Station that already has an Message Waiting indication will receive Message Waiting Busy following Procedure 1 or a Busy Tone following Procedure 2.
4. Any Station provided with the Sub Line can set or cancel BOSS - SECRETARY MESSAGE WAITING LAMP CONTROL indication on that Sub Line.
5. This feature is not restricted by CLASS OF SERVICE - INDIVIDUAL [C-15] on either the Boss or Secretary Stations.
6. BOSS - SECRETARY MESSAGE WAITING LAMP CONTROL indication is cleared by System Initialization or a Local Processor Initialization.
7. Message Waiting control is provided to Stations using the Desktop terminal Sub Line when:
   a.) one Station is talking with another Station.
   b.) a Special Purpose Button is pushed after a call is established.
   c.) the Station is calling another Station in the Network.
   d.) CALL WAITING [C-12, 13] is being set up.
   e.) the Station is using VOICE CALL [V-2].
   f.) the ATTENDANT CONSOLE [A-3] indication shows an Incoming Call.
8. Message Waiting for one Station of the network can be Controlled from the ATTENDANT CONSOLE [A-3] and any other Station at the same time. The controlled Station cannot identify the Controlling Party.
9. The Message Waiting indication on the Boss' Station appears as follows:

Type of Station Set	Message Waiting Indication
Desktop terminal	**MW** **(Time Display)**

10. The LCD on the Secretary's Station displays the following messages:

Status of Boss' Station		LCD Desktop Terminal	Type of Tone
MESSAGE WAITING (already set)	SETTING	**MW BUSY** **(Time Display)**	
	CANCELLING	**MW CANCEL** **(Time Display)**	
MESSAGE WAITING (not set)	SETTING	**MW SET** **(Time Display)**	Call Indicator is ON

11. Message Waiting cannot be activated on a Virtual Line. The LCD displays Message Waiting Rest, and Reorder Tone is heard when activation from a Virtual Line is attempted.

    Note: Reorder Tone is not heard while talking to the Calling Party.

12. This feature can control the Call Indicator Lamp or Message Waiting display from the Sub Line during:
    a.) conversation with the Calling Party.
    b.) special code dialing after replacing the handset or pressing the SPEAKER key.
    c.) Ringback Tone for Station Calling.
    d.) MUSIC ON HOLD [M-7] when the other party has pressed the HOLD key.
    e.) dialing of Telephone Number.
    f.) Trunk Origination or Called Number dialing.
    g.) calls to or talking to Attendant.
    h.) THREE-WAY CALLING [T-2].
    i.) CONSULTATION HOLD - ALL CALLS [C-17].
    j.) calls by VOICE CALL [V-2].
    k.) Call Waiting Ringback Tone with Call Waiting set.
    l.) Secretary calls to Boss by BOSS - SECRETARY TRANSFER [B-11].
13. Message Waiting can be canceled for a Station in the same Module Group using the My Line during:
    a.) conversation with the Calling Party.
    b.) Ringback Tone for Station Calling.
    c.) MUSIC ON HOLD [M-7] when the other party has pressed the HOLD key.
    d.) THREE-WAY CALLING [T-2].
    e.) CONSULTATION HOLD - ALL CALLS [C-17].
    f.) calls by VOICE CALL [V-2].
    g.) Call Waiting Ringback Tone with Call Waiting set.
14. CALL WAITING [C-12, 13] set from any other Station causes the Message Waiting Indication for the party setting the Call Waiting to go OFF.
15. THREE-WAY CALLING [T-2] established by EXECUTIVE RIGHT-OF-WAY [E-1] causes the Message Waiting Indication connected to the Interrupted Party to go OFF when the interrupting party presses the MESSAGE WAITING Cancel key. When using a Soft key, press the Message Waiting OFF to cancel.
16. FCCS is not available for this feature.
17. If Boss' telephone to which the secretary transfers a call is busy, the Call Indicator Lamp flashes without giving any ringing tone.

    Note: Ringing tone will be provided if the Boss' terminal is idle or is in off-hook status including pressing the SPEAKER key.

Interactions

1. Use of the following features has no effect on this feature. BOSS - SECRETARY MESSAGE WAITING LAMP CONTROL can be set/canceled to a Station that has previously activated any of the following features:
   a.) DO NOT DISTURB -D [D-21D]
   b.) CALL BACK [C-1] set to Boss' Station
   c.) CALL BACK [C-1] set from Boss' Station
   d.) CALL FORWARDING - ALL CALLS [C-5] will not follow Forwarding data
   e.) CALL FORWARDING - DON'T ANSWER [C-3] will not follow Forwarding data
   f.) CALL FORWARDING - BUSY [C-2] will not follow Forwarding data.
2. When the Boss' Line is answered by the Secretary, the following services may be activated either by the Secretary or by another Station towards the Secretary. These services will not interfere with the Secretary's ability to set/cancel Message Waiting:
   a.) PRIVACY -D [P-14D]
   b.) CONSULTATION HOLD - ALL CALLS [C-17]
   c.) CALL WAITING - ORIGINATING [C-31] set to Boss' Station
   d.) CALL WAITING - ORIGINATING [C-31] set from Boss' Station
   e.) ATTENDANT CAMP ON WITH TONE INDICATION [A-1] set to Boss' Station
   f.) THREE-WAY CALLING [T-2]
   g.) NON-EXCLUSIVE HOLD -D [N-7D], EXCLUSIVE HOLD -D [E-4D] or DUAL HOLD [D-32]

Programming

STEP 1: **ASYDL** - SYS1, Index 1088, Bit 2=1 (is Ringing tone provided if Boss' telephone is in busy status when secretary transfers a call?): 0/1 = Provided/Not provided

STEP 2: **AKYD** - Assign to a Line/Feature key a MESSAGE WAITING LAMP Set key. Feature key BOSS-SECRETARY MSG WAIT Set [9] and a MESSAGE WAITING LAMP Cancel key. Feature key BOSS-SECRETARY MSG WAIT- Cancel [11].

STEP 3: **ADSL** - Assign SN = 4 (Communication in Progress) when operating Procedure 1, SN = 2 (Originating) when operating Procedure 2. Assign Feature key BOSS-SECRETARY MSG WAIT - Set [9] and Feature key BOSS-SECRETARY MSG WAIT - Cancel [11].
This command is required only for assigning the function to each Soft key.

STEP 4: **ASFC** - Assign the Stations a Feature Class that allows SFI 74 [Msg Waiting Lamp from the ATT or Station (Called)] and SFI 82 [Msg Waiting Lamp from the Station (Calling)].

B-11 Boss - Secretary Transfer

General Description

This feature allows a Secretary to Voice Announce a call to a Boss when the Secretary answers a call to the Boss' Line.

Operating Procedure

To transfer from Dterm to Dterm

1. Call terminates to Boss' Line.
2. Secretary answers the call on the Sub Line of Secretary's Dterm.
3. Secretary presses the Line key of Answered Call. Boss and Secretary talk by Voice Call. The LCD displays:

Boss/Secretary

ICM	**XXXX**
(Time Display)	

4. Boss answers the call. The LCD displays:

Secretary

TRANSFER	**XXXX**
(Time Display)	

5. Secretary hangs up. Calling Party talks to Boss. The LCD displays:

Calling Party

	XXXX
(Time Display)	

To transfer from Dterm to a Single Line Telephone

1. Call terminates to Boss' Line.
2. Secretary answers the call on the Sub Line of Secretary's Desktop terminal.
3. Secretary presses the Line key of the Answered Call. Boss' Station rings. The LCD displays (Secretary):

TRANSFER	**XXXX**
(Time Display)	

4. Boss answers the call.
5. Secretary hangs up. Calling Party talks to Boss.

To cancel

1. Secretary presses the Line key.
2. Secretary talks to the Calling Party.

Service Conditions

1. Secretary must have Boss' My Line appearing on the Desktop terminal in a Multi Line operation.
2. Secretary must have a Desktop terminal for this operation. Boss may have a Single Line Telephone.
3. This feature is only available when the Secretary is talking on the Boss' My Line.
4. This feature must be allowed in CLASS OF SERVICE - INDIVIDUAL [C-15] via System Data.
5. BOSS - SECRETARY TRANSFER [B-11] is denied if the Secretary is connected to the ATTENDANT CONSOLE [A-3].
6. If provided in the system, the Held Caller will hear MUSIC ON HOLD [M-7].
7. The call is recorded in STATION MESSAGE DETAIL RECORDING (SMDR) [S-10] for the Boss' Station.
8. This feature will override CALL FORWARDING - BUSY LINE [C-2] set for the Boss, unless the Boss' Line is forwarded to the Secretary.
9. THREE-WAY CALLING [T-2] (Boss, Secretary and Calling Party) is not available.
10. A BOSS - SECRETARY TRANSFER Call cannot be picked up by CALL PICKUP [C-7, 30].
11. Name Display Service is not available when the Secretary is calling the Boss or when the Secretary is talking to the Boss.
12. When the Boss is using a Desktop terminal, the Boss must always press the Line key in order to answer the call transferred from the Secretary.
13. FCCS is not available for this feature.
14. Assigning ASYDL, SYS1, Index 1088, Bit 2=1 stops Boss's telephone from providing ringing tone during busy state of Boss's telephone: Call indicator lamp (Line key) on the Boss's telephone flashes with no ringing tone.
    This setting is valid only when the Secretary answers a call to the Boss' Line and transfers it to the Boss's telephone. However, this setting becomes invalid in some cases. In the following conditions, Boss's telephone is ringed even though the person is in busy state.
    - Line is idle.
    - The Speaker button is pressed.
    - Off the hook

Programming

STEP 1: **ASYDL** - Optional. SYS1, Index 1088, Bit 2
Assign "1" to this bit when disabling ringing tone on the Boss's telephone during its busy status. If not, assign "0".

STEP 2: **ANPD** - Reserve a Number Level for Feature Access. Select Normal Service (N), Hooking Service (H), or Busy Service (B) from the combo box for Connection Indexes. Assign NND in accordance with a predetermined Numbering Plan.

STEP 3: **ASFC** - Assign Station to receive Boss - Secretary Override a Feature Class that allows the following SFIs 5 [Call Waiting - Originating (Calling)], 6 [Call Waiting - Originating (Called)], 10 (Call Hold), 11 (Attendant Camp-On) and 51 (Boss - Secretary Features). Assign these SFIs as "1".

STEP 4: **AKYD** - Boss' My Line must be assigned as a Sub Line on the Secretary's telephone.

B-12 Brokerage Hotline - Outside

General Description

This feature provides a HOTLINE - OUTSIDE [H-9] function for a Desktop terminal.

Operating Procedure

To access a BROKERAGE HOTLINE - OUTSIDE from a Desktop terminal

1. Lift the handset or press the SPEAKER key.
2. Press a Sub Line key associated with a predetermined desired Outside Party. The Destination Station is automatically dialed; Ringback Tone is heard. The Destination Station answers.
3. After completion of the conversation, hang up or press the SPEAKER key.

To immediately make another BROKERAGE HOTLINE - OUTSIDE Call

1. Press another Sub Line Button associated with another Station without going on/off-hook. The RELEASE key on the Add-On Module can be used as the RELEASE key.
2. A maximum of 39 BROKERAGE HOTLINEs - OUTSIDE can be used to access distant destinations.

Service Conditions

1. One of the Line keys of the Desktop terminal must be assigned as the My Line.
2. When a BROKERAGE HOTLINE - OUTSIDE Call is made, the Desktop terminal display shows the connected Trunk Number but does not show a SERVICE DISPLAY [S-25] for the BROKERAGE - HOTLINE.
3. The My Line of a Desktop terminal cannot be assigned as a BROKERAGE HOTLINE - OUTSIDE.
4. The Virtual Line of a Single Line Station can be assigned as the Hotline Station.
5. BROKERAGE HOTLINE - OUTSIDE uses SPEED CALLING - SYSTEM [S-3] memory. The total number of BROKERAGE HOTLINE - OUTSIDE Destinations is limited by the capacity of the SPEED CALLING - SYSTEM [S-3].
6. ATTENDANT CAMP-ON [A-1], CALL BACK [C-1], CALL WAITING [C-31], and EXECUTIVE RIGHT-OF-WAY [E-1] can be prohibited to the BROKERAGE HOTLINE - OUTSIDE Connection through CLASS OF SERVICE - INDIVIDUAL [C-15].
7. A Hotline Station user confronted with all Trunks busy or all Senders busy at the time of calling hears the Reorder Tone. No indication is shown on the user's Desktop terminal LCD.
8. Outside Connection when a user calls in BROKERAGE HOTLINE - OUTSIDE Service is used when Trunk Classification (DDD, TIE, CCSA) and the Trunk Number is indicated on the LCD indication.
9. Outside Connection when a user answers in BROKERAGE HOTLINE - OUTSIDE Service is used when Time Duration, Trunk Classification (DDD, TIE, CCSA) and the Trunk Number is indicated on the LCD indication.
10. Station Message Detail Recording (SMDR) is available for Outgoing Calls in BROKERAGE HOTLINE - OUTSIDE Service.
11. A Desktop terminal with an Add-On Module attached is used for applications requiring at least 15, but fewer than 39 destinations. In this case, one of the Line keys of the Desktop terminal must be assigned as the My Line.
12. An AC Adapter is required for the Add-On Module for the Desktop terminal phones.
13. This feature can be activated via FCCS.

14. Only the Station assigned a Telephone Number is available for the Station assignment when activating this feature via FCCS.
15. The Station, the Attendant Console / Desk Console (Operator Call), and the Trunk can be a destination for Hotline.
16. When Call Forwarding - All Calls/Busy Line/Don't Answer Service has been set to the Called Station, these features are available for the call to the Station.
17. When a Calling Party encounters a busy Station, the call may bridge into the connection, depending on the data assigned by ASYD.
18. When Telephone Number to be set by ASPD command is 8 digits or more, Desktop terminal LCD can display only 8 digits from the top digit.
19. A SIP Handler Controlled SIP terminal cannot be used as a Hotline Station but it can be used as a destination of a Hotline Station.
20. This feature is not supported for MC&MG-COT or MG-COT (including UG50).

Programming

STEP 1: **ASDT** - Assign to a 16LC Circuit the following data:

TN: Tenant Number
STN: Station Number
LENS: Line Equipment Numbers (six digits)
TEC: Telephone Class (1 - 31)
1: DP (10 PPS)
2: PB
3: DP/PB
12: Dterm
13: Data Terminal via Dterm
→ 14: Hotline
15: CAS Line
18: Virtual Circuit
RSC: Route Restriction Class (0 - 15)
SFC: Service Feature Restriction Class (0 - 15)

STEP 2: **ASPD** - Assign ADC, maximum of three digits. Assign CD = Called Number, maximum 24 -digits, corresponding to the ADC.

STEP 3: **AHLS** - Assign the Station Number programmed above for the following parameters:

HOT TN: Hotline Tenant Number
HOT STN: Hotline Station Number
TYPE: 2=Calling Party is an Outside Party using Abbreviated Digit Code (ADC)
3=Called Party is a Trunk
CON ADC: Speed Calling Code of the Destination Station
CON TN: Destination Tenant Number
CON STN: Destination Station Number
CON RT: Destination Route Number
CON TK: Destination Trunk Number

STEP 4: **AKYD** - Assign to a Dterm a Line Appearance of the Hotline Station Number.

TN: Tenant Number

STN: Station Number
TP: Type of Dterm (0-3)
KYN: Key Number (1 - 40)
KYI: Service Index
0: Key not used
1: Feature Key
→ 2: Multi Line Key
KD: 0
RG: Ringing Assignment per Multi Line Appearance
0: No ringing
1: Ringing in Day mode only
2: Ringing in Night mode only
3: Not Used
4: Ringing in Day and Night Mode - 30 second delay.
5: Ringing in Day and Night Mode - 20 second delay.
6: Ringing in Day and Night Mode - 10 second delay.
7: Ringing in Day and Night Mode.

FCCS Programming

This data setting describes only the additional or replacement commands used to activate FCCS Services. When used as an FCCS feature, these commands replace the non-network commands (for example, AAED is replaced by AAEDN). Note that the data setting for FCCS Service varies depending on the Node.

STEP 1: **AHLSN** - This data must be assigned to the NCN. Assign the Station Number programmed above for the following parameters:

HOT UGN: Hotline User Group Number of the Calling Party
HOT TELN: Hotline Telephone Number of the Calling Party
TYPE: 2 (Calling Party is an Outside Party using Abbreviated Digit Code (ADC))
3 (Called Party is a Trunk)
CON UGN: User Group Number of the Called Party (for TYPE = 2)
CON ADC: Speed Calling Code assigned by the ASPD command (for TYPE = 2)
CON LRN: Logical Route Number of the Called Party (for TYPE = 3)
CON TK: Trunk Number of the Called Party (for TYPE = 3)

Note: The following data should also be set in addition to the commands above.
The commands to be set to the Node accommodating the Calling Party: ASDT/AKYD/ASPD
If Hotline TEC 14, the ASFC command data must be assigned at all the Nodes to be set Brokerage - Hotline Service: SFI 36 (Brokerage - Hotline), RES 1 (allowed).
When assigning Virtual Hotline, set the ASFC command data to TEC 18.
If Hotline is Virtual TEC 18, assign SFI 36 (Brokerage - Hotline). If Hotline is a TEC 14, do not assign SFI 36 (Brokerage - Hotline).

B-13 Blocking of Trunk Usage

General Description

This feature is activated to set make busy status on a Bothway Trunk, so that the Trunk cannot be used for either Incoming or Outgoing Calls.

Note: This feature is only available in Australia.

Operating Procedure

Bothway Trunk Make Busy is both set and canceled by MBTK.

Service Conditions

1. At office which sets Bothway Trunk Make Busy
   a.) The system does not detect any Incoming Calls received at the Bothway Trunk placed in the Make Busy state by this service.
   b.) As soon as Bothway Trunk Make Busy is set, the BL lamp (red) starts to flash at 120 IPM and the SUP lamp (yellow) lights up on the corresponding Trunk Card.
   c.) The Trunk is completely released from the Make Busy state only after an Outgoing Call is placed to it, following Make Busy release by the appropriate command.
   d.) As soon as the Make Busy state is completely released, both the BL and SUP lamps on the Trunk Card go off.
   e.) This service can be used to set Make Busy on Bothway Trunk only.
   f.) It is not possible to set Make-Busy on a line currently engaged in a call.
   g.) Make Busy state is not released by system, LP or circuit card initialization.
2. At office at which Bothway Trunk Make Busy is assigned
   a.) When the distant office sets up Incoming and Outgoing Make Busy, the corresponding Trunks are placed to Make Busy state and both the BL and SUP lamps on the Card start to flash at 120 IPM.
   b.) Make Busy is released following the receipt of an Incoming Call from the distant office to the Trunk which has been in the Make Busy state.
   c.) Outgoing Trunk Make Busy established by the Trunk Make Busy setting at the distant office may be released by LP initialization, though not by circuit card initialization.
   d.) Both the BL and SUP lamps go off as soon as the Outgoing Trunk Make Busy status is released.

Programming

STEP 1: **ASYD** - SYS1, Index 7, Bit 2 (Trunk Blocking-Wink Signal Tie Lines): 0/1 = Out/In Service.
Assign "1".

STEP 2: **ARTD** - Assign the Route Class Data for the Trunk Route.
< CDN: 48 >
BL (Detection of Blocking Signal): 0/1 = No detection/Detection.
Assign "1".

Note: To change the data for existing route (RT), initialize or reboot the trunk device such as a circuit card or MG.

STEP 3: **MBTK4** - Assign make busy data for the Trunk.
MB = 2 (Make Busy (Bothway)).

B-17 Blind Transfer To Attendant

General Description

This feature allows a Station to transfer a Held Call (Station or Trunk) to the Attendant Console and hang up without having to wait for the Attendant to answer.

Operating Procedure

<For terminals other than Standard SIP Terminal>

1. While connected to another party (Station or Trunk), the user presses switch hook or the TRANSFER key. Receives Special Dial Tone.
2. User dials the Access Code for operator. Receives Ringback Tone while the call terminates to the Attendant at the RECALL key. The Transferred Party hears MUSIC ON HOLD [M-7].
3. User can either:
   a.) Hang up before the Attendant answers, and call will be connected to the Attendant when the Attendant answers.

   Note: The call will terminate at the Attendant's INCOMING CALL IDENTIFICATION (ICI) key depending on the Call Class (LDN, FX, WATS, CCSA, TIE, ATND) of the party being transferred. Peg count will be incremented by one, according to the Call Class. The Transferred Party hears Ringback Tone.

   b.) Wait for the Attendant to answer and then hang up after announcing that a call is being transferred.

   Note: When this method is used, the call is identified at the Attendant's INCOMING CALL IDENTIFICATION (ICI) key as the RECALL key. Peg count for RECALL will be incremented by one.

<For Standard SIP Terminals>

Refer to the manual of each Standard SIP Terminal for operating procedures.

Note: This feature is available for Standard SIP Terminal.

Service Conditions

1. Reorder Tone will be sent to the Station if an attempt is made to transfer to an Attendant in Night mode.
2. This feature is not available on a Tenant basis.
3. If the Attendant does not answer the Transferred Call within the predetermined time set for Transfer Call Recall, the Transferring Station will not be recalled.
4. This feature is allowed only when a Station is engaged in a Two-party Connection (Station or Trunk).
5. Reorder Tone will be sent to the transfer Originating Station if that Station is restricted from accessing the Attendant.
6. Attendant peg count is incremented by one for the following:
   a.) RECALL when the Attendant answers before the Transferring Station releases.

b.) LDN, FX, WATS, CCSA, TIE, or ATND when the Attendant answers after the Transferring Station releases.

7. The System will not set a Priority Queue for the call being transferred when the Station originating the transfer hangs up before the Attendant answers.
8. CALL TRANSFER - ATTENDANT [C-10] is restricted when this feature is activated.
9. If a call is blind transferred to a Station that has set CALL FORWARDING - DON'T ANSWER [C-3] to the Attendant, the following conditions will occur:
   a.) if No Recall on Call Transfer is in service, the call will forward in a Don't Answer scenario. BLIND TRANSFER TO STATION [B-18] must be in service.
   b.) if Recall on Call Transfer is in service, the Transferred Call will recall to the Transferring Party.
10. This feature is available for Stations in the FCCS Network only.

<Conditions particular to Standard SIP Terminals>

SP Controlled SIP terminal:

1. This feature is available for SP Controlled SIP terminals which are confirmed by the connection test of NEC as well as in compliance with IETF RFC3261 and the Internet-Drafts listed below.
   - "draft-ietf-sipping-service-examples-12 (2.4. Transfer - Unattended)"
   - "draft-ietf-sipping-cc-transfer-07"
2. This feature is not available when the control of the terminal is changed over to SR-MGC.
3. SP Controlled SIP terminals are essentially disconnected right after the Blind Transfer is performed.
4. Step Call cannot be activated for SP Controlled SIP terminals when the transferred destination is busy.
5. A SP Controlled SIP terminal which operated Blind Transfer is called back by the Telephony Server when a transferring ends in a failure caused by cases such as when the transferred destination is busy or when misdialed.

SIP Handler Controlled SIP terminal:

1. When a SIP Handler Controlled SIP terminal is in Multi Line Mode, only BLIND TRANSFER through the REFER method is available. BLIND TRANSFER through the secondary session is unavailable.

Programming

STEP 1: **ASYD** - SYS1, Index 17, Bit 4=1, Blind Transfer to Attendant In Service.
SYS2, Index 1, Bit 0 and Bit 1=1.

STEP 2: **ASFC** - Cancel the restriction of Blind Transfer to Attendant from SFC of the Station.
SFI 103 (Assignment of No Answer Timer), RES = 1.

STEP 3: **ANPD** - 0 = Hooking Service (H) NND = 1.

STEP 4: **ASPA** - CI = "H: Hooking" SRV = "SSC: Service Code", SID = 2.

FCCS Programming

This data setting describes only the additional or replacement commands used to activate FCCS Services. When used as an FCCS feature, these commands replace the non-network commands (for example, AAED is replaced by AAEDN). Note that the data setting for FCCS Service varies depending on the Node. **Note 1**

STEP 1: **ANPDL** - 0 = Hooking Service (H), NND = 1.

STEP 2: **ASPAL** - CI = “H: Hooking” SRV = “SSC: Service Code”, SID = 2, FPC = “x”.

Note 1: When using ASYDL or ASYDN to program an FCCS feature, the basic programming of ASYD is required in addition to ASYDL or ASYDN.

B-18 Blind Transfer To Station

General Description

This feature allows two Service Feature Restriction Classes to be assigned to Stations, enabling services such as Voice Mail, Announcement Machine, etc. depending on the Service Feature Restriction Class.

1. A Service Feature Restriction Class which allows No Answer Timer to be extended when the Blind Transferred-to Station does not answer within the predetermined period of time interval.
2. A Service Feature Restriction Class which activates CALL FORWARDING - DON'T ANSWER [C-3] set to the Blind Transferred-to Station when CALL FORWARDING - DON'T ANSWER [C-3] is available.

Operating Procedure

<For terminals other than Standard SIP Terminal>

To transfer a call in progress

1. While connected to another party (Station or Trunk), the user presses switch hook or the TRANSFER key on Dterm. Receive Special Dial Tone.
2. User dials the third Station. Receive Ringback Tone. If the user remains on the line, the Transferred Party hears MUSIC ON HOLD [M-7].
3. At this point, the user can hang up before the third Station answers. The Transferred Party hears Ringback Tone.

<For Standard SIP Terminals>

Refer to the manual of each Standard SIP Terminal for operating procedures.

Service Conditions

1. This feature is allowed only when a Station is engaged in a Two-party Connection (Station or Trunk).
2. Samples of Connection Patterns available under BLIND TRANSFER Service are as follows:
   a.) Station A blind-transfers Incoming Trunk/Station to Station B (if Index 69=0).
   b.) Station A blind-transfers Incoming Trunk/Station Call to the Attendant (if Index 69=1).
   c.) Station A blind-transfers Incoming Trunk/Station Call to Station B, which has set CALL FORWARDING - DON'T ANSWER to Station C [the call is forwarded to Station C (if Index 69=1)].
   d.) Station A blind-transfers Incoming Trunk/Station Call to Station B, which has set CALL FORWARDING - DON'T ANSWER Trunk (the call is forwarded to outside Station).
3. The Service Feature Restriction Class that extends No Answer Timer must be the same as the one used for BLIND TRANSFER TO ATTENDANT [B-17]. This Service Feature Restriction Class is assigned to the Transferring Station.
4. Depending on System Data assignment, the following conditions occur when CALL FORWARDING - DON'T ANSWER feature is set on the Blind Transferred-to Station:
   a.) the Transferring Station is recalled.
   b.) the Transferred-to Station rings continuously.
   c.) the call is forwarded when the CALL FORWARDING - DON'T ANSWER Timer expires.
5. If the Destination Station for Blind Transfer has set CALL FORWARDING - DON'T ANSWER [C-3] to a Station/Trunk, Call Forwarding is activated when a predetermined No Answer Timer expires.

6. If you use this feature in the FCCS network, the feature is not available for stations outside the FCCS network.
7. When PS becomes Out of Zone while holding a call (switch hook flash), it will be recalled.
   However, if it remains in Out of Zone state, the following will operate.
   If CALL FORWARDING - PS INCOMING INCOMPLETE [C-155] is set, the aborted call will be transferred to the preprogrammed destination. If AUTOMATIC ANNOUNCEMENT - CALLED PS OUT OF ZONE [A-129] is set, the system automatically provides an announcement.
8. When you blind transfer a call to a PS station in the Local Node, and if the PS station goes out of zone before answering the call, you will be recalled.
9. When ASYD - SYS1, Index 69, Bit 2=1 (Call Forward Recall Tone. A burst of Ringback Tone to alert the person receiving a recall when a BLIND TRANSFER - TO STATION is not answered) is assigned: when a SIP station/PHS receives a recall when a BLIND TRANSFER - TO STATION call is not answered, there will be no burst of Ringback Tone.

<Conditions particular to Standard SIP Terminals>

SP Controlled SIP Terminal:

1. This feature is available for SP Controlled SIP terminals which are confirmed by the connection test of NEC as well as in compliance with IETF RFC3261 and the Internet-Drafts listed below.
   - “draft-ietf-sipping-service-examples-12 (2.4. Transfer - Unattended)”
   - “draft-ietf-sipping-cc-transfer-07”
2. This feature is not available when the control of the terminal is changed over to SR-MGC.
3. SP Controlled SIP terminals are essentially disconnected right after the Blind Transfer is performed.
4. Step Call cannot be activated for SP Controlled SIP terminals when the transferred destination is busy.
5. A SP Controlled SIP terminal which operated Blind Transfer is called back by the Telephony Server when a transferring ends in a failure caused by cases such as when the transferred destination is busy or when misdialed.

SIP Handler Controlled SIP terminal:

1. When a SIP Handler Controlled SIP terminal is in Multi Line Mode, only BLIND TRANSFER through the REFER method is available. BLIND TRANSFER through the secondary session is unavailable.

Programming

STEP 1: **ASYD** - SYS1, Index 69, Bit 0:

Processing for a case where the Transfer Destination Station does not answer the call within a predetermined period of time. 0/1 = the Transferring Station is recalled/the Transfer Destination Station is continuously rung, or if the Call Forwarding - Don't Answer is set at the Destination Station, Call Forwarding - Don't Answer is executed.

SYS1, Index 247: Assign the timer value of No Answer Timer for Blind Transfer to Station.

SYS1, Index 448, Bit 2: When Blind Transfer is activated via CCIS:

0 = If Incoming Call is terminated to the Trunk that cannot detect the release signal, the call is forcedly released when the destination party does not answer within a period of time (60 seconds).

1 = Incoming Call is forcedly released when the destination party does not answer within the predetermined timer assigned by ASYD, SYS1, Index 448, Bit 4 to Bit 7.

SYS1, Index 448, Bit 4 to Bit 7: Assign the forced release timer value when ASYD, SYS1, Index 448, Bit 2=1. (Default value is 64 seconds.)

STEP 2: **ASFC** - Assign the following data to the SFC of the Station:

SFI 103: Assign the timer value of No Answer Timer for BLIND TRANSFER TO STATION.

RES: 0/1 = Index 140/Timer value assigned by ASYD, SYS1, Index 247
0 = Index 140
1 = Index 247

Also, when ASYD, SYS1, Index 69, Bit 0=1 has been assigned, assign the following data to the SFC of the Station:

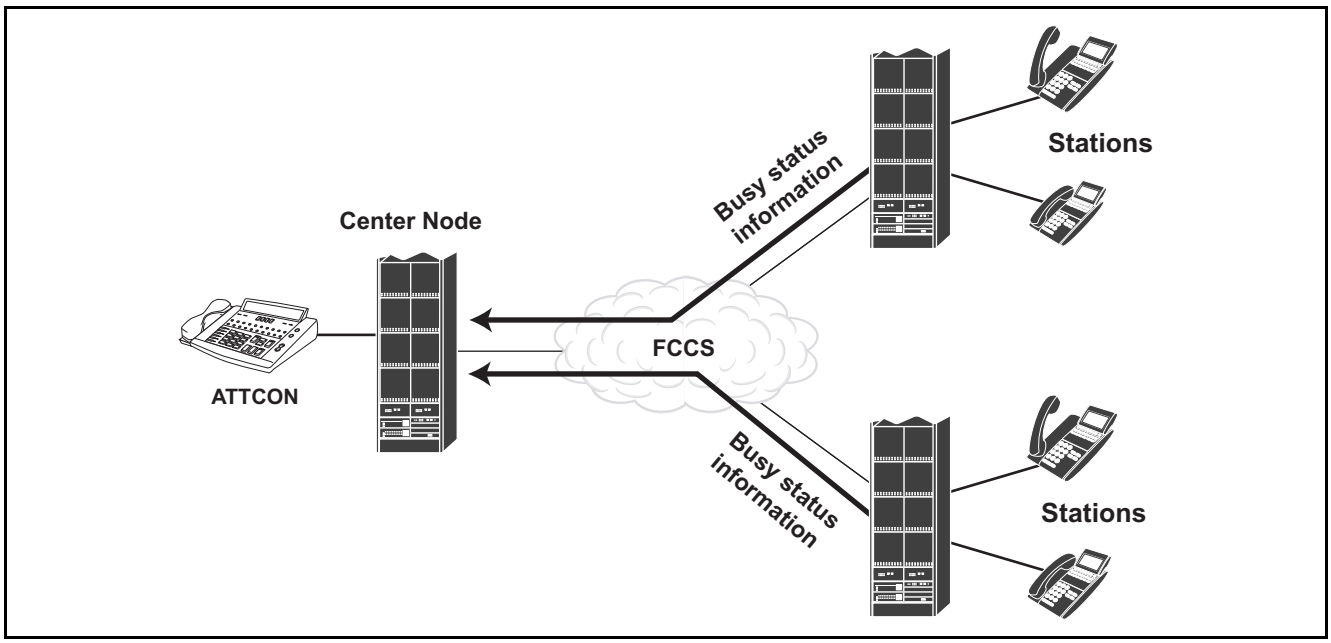
SFI 104: Processing in a case when the Transfer Destination Station does not answer within a predetermined period of time.

RES: 0/1 The Transfer Destination Station is continuously rung/Call Forwarding - Don't Answer is executed if it is set at the Transfer Destination Station.

B-29 BLF over FCCS

General Description

The system can exchange Busy Lamp Field (BLF) data. As shown below, the busy status information of the other nodes within FCCS network can be displayed on the ATTCON of the Center Node designated by the System data.



Operating Procedure

Refer to the Data Programming Manual - FCCS.

Service Conditions

Refer to the Data Programming Manual - FCCS.

Programming

Refer to the Data Programming Manual - FCCS.

B-30 Blind Transfer with Recall

General Description

This feature combines the BLIND TRANSFER TO STATION [B-18] with AUTOMATIC RECALL -D [A-8D] to allow blind transfer with recall capability. If the target of a blind transfer has CFD set, the call will follow the forward. If CFD is not set, the call will recall back to the transferring Station.

Note: This feature is only available in North America.

Operating Procedure

No Station user operation is required for Blind Transfer and Recall work from same Station.

Service Conditions

1. All of the service conditions for BLIND TRANSFER TO STATION [B-18] and AUTOMATIC RECALL -D [A-8D] apply.

Interactions

1. BLIND TRANSFER TO STATION [B-18] and AUTOMATIC RECALL -D [A-8D] must be provided.

Programming

See the programming sections of BLIND TRANSFER TO STATION [B-18] and AUTOMATIC RECALL -D [A-8D]. Then perform the following.

STEP 1: **ASYDL** - SYS1, Index 805, Bit 6=0/1 B-30 out of service / B-30 in service

C-1 Call Back

General Description

This feature provides the ability for a Calling Station that has dialed a busy Station to dial a CALL BACK code. This feature also provides the ability for a Desktop terminal, which has dialed a busy Station, to press the CALL BACK key. When this has been done, the Calling Station will be rung as soon as the busy Station becomes available, provided the Calling Station is also idle.

Operating Procedure

To set using Call Back Code

1. Dial the desired Telephone Number. Receive Busy Tone.
2. Press switch hook for Special Dial Tone.

   Note: The availability of special dial tone is changed depending on the combination of the Step Call - CCIS Restriction (ASYD, SYS2, Index 6, Bit 2) and SHF method (ASYD, SYS1, Index 68, Bit 0). Refer to Step Call - CCIS [S-59] in Data Programming Manual - CCIS for the detail.

3. Dial CALL BACK Code. Receive Service Set Tone.
4. Hang up.
5. When the Station becomes idle, Calling Station will ring. Upon answering, the Called Station will be rung.

To cancel using Call Back Code

1. Lift the handset. Receive Dial Tone.
2. Dial CALL BACK Cancel Code. Receive Service Set Tone.

<The following procedures are applied for Desktop terminals only.>

To set using the CALL BACK key

1. Dial desired Telephone Number. Receive Busy Tone.
2. Press the CALL BACK key. Receive Service Set Tone. The LCD displays:

Called Station

CB SET	**XXXX**
(Time Display)	

3. Hang up.
4. When the desired Station becomes idle, the Calling Station will ring. The LCD displays:

Called Station (Flashing)

CB	**XXXX**
(Time Display)	

5. Upon answering, the Called Station will be rung.

6. Called Desktop terminal's LCD displays:

Calling Station

CB	XXXX
(Time Display)	

To cancel using the CALL BACK key

1. Lift the handset or press the SPEAKER Key. Receive Dial Tone.
2. Press the CALL BACK Key. Receive Service Set Tone. CALL BACK is canceled.
3. The LCD displays:

CB CANCEL
(Time Display)

Service Conditions

1. If CALL BACK is not answered within 30 seconds, ringing will stop and CALL BACK will be automatically canceled.

   Note: No Answer Time is variable (ASYD, SYS1, Index 139).

2. As with certain other CLASS OF SERVICE - INDIVIDUAL [C-15] features, CALL BACK may be assigned via Feature Class.
3. CALL BACK to a busy Station cannot be established if that Station is in the LINE LOCKOUT [L-3] mode.
4. There is no time limit for CALL BACK. Once activated, it will remain in status memory until completed or manually canceled.
5. A Station cannot activate CALL BACK to more than one Station at a time.
6. CALL BACK cannot be set by more than one Station at a time (Called Party).
7. A Station cannot use CALL BACK and OUTGOING TRUNK QUEUING [O-2] at the same time.
8. When the Calling Station is called back, STATION HUNTING [S-7, 8, 9] and CALL PICKUP [C-7] programming will be ignored.
9. There is no limit to the number of Stations that may access this feature simultaneously.
10. A Delay Timer can be set in system programming, on a system-wide basis. This allows a Station, to which CALL BACK has been set, to initiate another call within a predetermined time period after becoming idle, before the CALL BACK feature is activated. The Delay Timer can be set in multiples of two seconds for up to a maximum of 14 seconds. (Delay Timer is unavailable for PHS/PCS/WLAN Handset.)
11. CALL BACK Service cannot be set when FCCS Links are all busy.
12. CALL BACK Service that has already been set is automatically canceled when all FCCS Links are busy at the moment the Target Station becomes idle.
13. When Called Station activates CALL FORWARDING- BUSY LINE [C-2], and the call forwarding destination is busy:

- When call forwarding destination is a station (including stations in FCCS network), ASYD, SYS1, Index 69, Bit 5 determines whether to set CALL BACK to the Forwarded-from Station, or to the Forwarded-to Station.
- When the call forwarding destination is in outside of the self office, CALL BACK is restricted and does not work.

<The following conditions are applied for Desktop terminals only.>

1. CALL BACK may be assigned via CLASS OF SERVICE - INDIVIDUAL [C-15].
2. CALL BACK can only be activated from My Line of a Desktop terminal.
3. CALL BACK cannot be established to a busy Station when the Station is engaged in any of the following:

ATTENDANT CAMP-ON WITH TONE INDICATION	[A-1]
CALL HOLD	[C-6]
OFF-HOOK QUEUING	[O-7]
OUTGOING TRUNK QUEUING	[O-2]

Interactions

1. OUTGOING TRUNK QUEUING [O-2] and CALL BACK cannot be accessed simultaneously.
2. ASYD, SYS1, Index 139, No Answer Timer is also used for Station-to-Station Call, DID, or TIE Line Call for CALL FORWARDING - DON'T ANSWER [C-3], and CALL HOLD [C-6].
3. CALL WAITING - ORIGINATE [C-31] automatic setting and CALL BACK are mutually exclusive.
4. CALL BACK cannot be activated toward the Attendant Console.
5. CALL HOLD [C-6]: Activation of CALL BACK is not allowed if the Calling or Called Station has a party on Call Hold. CALL BACK can be activated on a Desktop terminal that is holding a call or is being held by EXCLUSIVE HOLD [E-4D] or NON-EXCLUSIVE HOLD [N-7D].
6. When CALL FORWARDING - ALL CALLS [C-5] has been set from Station A to Station B, and Station B is busy, the CALL BACK feature will be set to Station A.
7. CALL BACK cannot be activated when using CONSULTATION HOLD [C-17].
8. CALL BACK cannot be set when all FCCS Links are busy.
9. When the FCCS Links are all busy at the moment the Target Station becomes idle, CALL BACK that has already been set is automatically cancelled.
10. To operate this feature from SIP Handler Controlled SIP terminal, perform the one digit dial hooking operation while hearing Busy Tone (BT) and dial the access code for this feature. To hear Busy Tone (BT), enable IP-DTG with the AUACL/AUACN command.

Programming

STEP 1: **ASYD** - SYS1, Index 68, Bit 0. 0/1: SHF and Access Code/last digit of Telephone Number + Access Code.
Index 71, Bit 0 through Bit 3. Call Back Automatic Cancel Timer (TC=2-F)
3.5 × (TC-1) < value < 3.5 × TC (minutes)
Index 71, Bit 7. Enable Call Back Automatic Cancel Timer 0/1: No/Yes
Index 139. No Answer Timer for CALL BACK. Assign 00H. (RAM Data is 3FH = 30 seconds.)
SYS2, Index 0, Bit 0. Is CALL BACK enabled on a per Tenant basis? 0/1: No/Yes.

SYS2, Index 4, Bit 0. CALL BACK and OUTGOING TRUNK QUEUING [O-2] Access Codes are same or separate? 0/1: Separate/Same.

SYS3, Index 3, Bit 5. Enable Ringer Pattern 5 for CALL BACK. See DISTINCTIVE RINGING [D-10]. (Unavailable for PS/WLAN Handset.)

SYS3, Index 7. Assign data 24H for four seconds ON, two seconds OFF.

SYS1, Index 5, Bit 7. CALL BACK - Delay Timer: 0/1 = Out/In Service. (Unavailable for PHS/PCS/WLAN Handset.)

SYS1, Index 5, Bit 4 through Bit 6. Delay Timer. Value equals two seconds x the Data set. (Unavailable for PHS/PCS/WLAN Handset.)

SYS1, Index 69, Bit 5. CALL BACK: 0/1 = Set to Forwarded-from Station/Set to Forwarded-to Station

STEP 2: **ANPD** - Reserve a Number Level for Service Feature Access and Cancel. Select Normal Service (N) or Busy Service (B) from the combo box for Connection Indexes. Assign NND in accordance with a predetermined Numbering Plan.

STEP 3: **ASPA** - Assign an Access Code to CALL BACK - Entry, SRV = "SSC: Service Code", SID = 3 and a Connection Status Index (CI) for "B: Busy" service. For CALL BACK - Cancel, SRV = "SSC: Service Code", SID = 6. Assign a Connection Status Index (CI) for "N: Normal" service.

STEP 4: **ASFC** - Assign a Service Feature Restriction Class that allows SFI 2 (CALL BACK) to Stations to be allowed CALL BACK.

STEP 5: **AKYD** - For Desktop terminals, CALL BACK may be assigned to a Line/Feature key. Assign Feature key CALL BACK [5].

STEP 6: **ATNR** - Allow Tenant Restriction Index TRI 0, Station-to-Station Calling for Inter and Intra-Tenant Connections.

FCCS Programming

This data setting describes only the additional or replacement commands used to activate FCCS Services. When used as an FCCS feature, these commands replace the non-network commands (for example, AAED is replaced by AAEDN). Note that the data setting for FCCS Service varies depending on the Node.

Note: When using ASYDL or ASYDN to program an FCCS feature, the basic programming of ASYD is required in addition to ASYDL or ASYDN.

STEP 1: **ANPDL** - Reserve a Number Level for Service Feature Access and Cancel. Select Normal Service (N) or Busy Service (B) from the combo box for Connection Indexes. Assign NND in accordance with a predetermined Numbering Plan.

STEP 2: **ASPAL** - Assign an Access Code to CALL BACK - Entry, SRV = "SSC: Service Code", SID = 3 and a Connection Status Index (CI) for "B: Busy" service. For CALL BACK - Cancel, SRV = "SSC: Service Code", SID = 6. Assign a Connection Status Index (CI) for "N: Normal" service.

Note: System Timers may be set on a network basis in ASYDN.

C-2 Call Forwarding - Busy Line

General Description

This feature permits a call to a busy Station to be immediately forwarded to a predesignated Station or to the ATTENDANT CONSOLE [A-3].

If a Called Station is in a STATION HUNTING [S-7, 8, 9] Group and forwards calls to another Hunt Group, it can be determined through System Data that the Calling Party has been directed either to the Called Parties Hunt Group or to the Terminating Parties Hunt Group when all of the Forwarded Stations are busy.

For Desktop terminals:
CALL FORWARDING - BUSY LINE may be set or canceled by the user for all Multi Line Appearances on the Desktop terminals. Additionally, a Single Line user may set CALL FORWARDING - BUSY LINE for all Sub Lines on the Desktop terminals.

Operating Procedure

To set from an Individual Station using Access Code

1. Lift the handset. Receive Dial Tone.
2. Dial the specific CALL FORWARDING - BUSY LINE Access Code. Receive Special Dial Tone.
3. Dial the desired target Telephone Number. Receive Service Set Tone.

Note: A tone will not be provided when an access code is used to set this feature on SP Controlled Standard SIP terminal.

To cancel from an Individual Station using Cancel Code

1. Lift the handset. Receive Dial Tone.
2. Dial the specific CALL FORWARDING - BUSY LINE Cancel Code. Receive Service Set Tone.

To set from the ATTENDANT CONSOLE [A-3]

1. Press an Idle Loop key.
2. Dial the specific CALL FORWARDING - BUSY LINE Access Code. Receive Special Dial Tone.
3. Dial the TENANT SERVICE [T-12] Number (two or three digits).
4. Dial the Originating Telephone Number.
5. Dial the desired Target Telephone Number. Receive Service Set Tone.

To cancel from the ATTENDANT CONSOLE [A-3]

1. Press an Idle Loop key.
2. Dial the specific CALL FORWARDING - BUSY LINE Cancel Code.
3. Dial the TENANT SERVICE [T-12] Number (two or three digits).
4. Dial the Originating Telephone Number. Receive Service Set Tone.

<The following procedures are applied for Desktop terminals only.>

To set from an Individual Station using a Line/Feature key

1. Lift the handset or press the SPEAKER key. Receive Dial Tone.
2. Press the CALL FORWARDING - BUSY LINE FORWARD - BY key. Receive Special Dial Tone.
3. Dial the desired Target Telephone Number. Receive Service Set Tone. The LED of the associated Feature key lights and the LCD displays:

Target Station

FORWARD SET XXXX (Time Display)

4. Replace the handset or press the SPEAKER key.
5. If the Target Station is the Attendant Console, the LCD displays:

FWD SET (Time Display)

To monitor

1. Press the CALL FORWARDING - BUSY LINE FORWARD - BY Feature key. The LCD displays:

Target Station

FORWARD XXXX (Time Display)

2. If the Target Station is the Attendant Console:

FWD SET (Time Display)

To cancel

1. Lift the handset or press the SPEAKER key. Receive Dial Tone.
2. Press the FORWARD-BY key. Receive Service Set Tone. The LCD displays:

FORWARD CANCEL (Time Display)

3. The associated LED goes out.
4. Replace the handset or press the SPEAKER key.

To set CALL FORWARDING - BUSY LINE to a Desktop terminal Multi Line:

1. Press the Multi Line key for which CALL FORWARDING - BUSY LINE feature is to be set.
2. Lift the handset or press the SPEAKER key. Receive Special Dial Tone.
3. Press the CALL FORWARDING - BUSY LINE "FORWARD-BY" Feature key. Receive Special Dial Tone.
4. Dial the desired Target Telephone Number. Receive Service Set Tone. The LCD displays:

Target Station

FORWARD SET XXXX (Time Display)

5. The LED of the associated Feature key lights on the Desktop terminal for which this Multi Line is a My Line.

6. If the Target Station is the Attendant, the LCD displays:

FWD SET OPR (Time Display)

7. Replace the handset or press the SPEAKER key.

To monitor CALL FORWARDING - BUSY LINE to a Desktop terminal Multi Line

1. Press the Multi Line key for which CALL FORWARDING - BUSY LINE is to be monitored.
2. Press the FORWARD - BY key. The LCD displays:

Target Station

FORWARD XXXX (Time Display)

If the Target Station is the Attendant Console:

FWD SET OPR (Time Display)

To cancel CALL FORWARDING - BUSY LINE to a Desktop terminal Multi Line

1. Press the Multi Line key for which CALL FORWARDING - BUSY LINE feature is to be canceled.
2. Lift the handset or press the SPEAKER key. Receive Dial Tone.
3. Press the FORWARD - BY key. Receive Service Set Tone. The LED of the associated Feature key will go out on the Desktop terminal for which this Multi Line is a My Line. The LCD displays:

FORWARD CANCEL (Time Display)

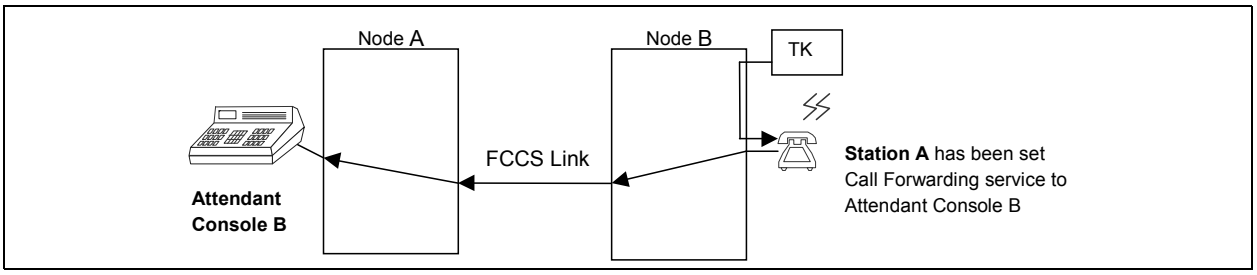
4. Replace the handset or press the SPEAKER key.

Service Conditions

1. This feature may be set and canceled by either the Station or the Attendant.
2. CALL FORWARDING - BUSY LINE Data remains in the Data Memory, even if the system is re-initialized or in the event of a power failure, but it must be saved to HD.
3. This feature is assigned to Stations through CLASS OF SERVICE [C-15] programming in System Data.
4. Registration memories for CALL FORWARDING - BUSY LINE and CALL FORWARDING - DON'T ANSWER [C-3] are common. Changing System Data will enable separation of the two, permitting separate Access Codes to be assigned.
5. CALL FORWARDING - BUSY LINE can be provided on a Tenant and/or an individual Station basis.
6. Call Forwarding on a Tenant basis allows any Incoming Calls, which encounter a busy condition, to be forwarded to a predetermined Station or the ATTENDANT CONSOLE [A-3].
7. Individual CALL FORWARDING - BUSY LINE supersedes system basis Call Forwarding.
8. If all the Stations in the Terminating Hunt Group are busy, the Calling Party will hear Busy Tone and will not return to the Called Party's Hunt Group.

9. When MULTIPLE CALL FORWARDING - BUSY LINE [M-24] is activated, hunting will be performed at the last destination of the forwarded call, even though the multiple jumped stations are in a Hunt Group.
10. When the Pilot Station of a STATION HUNTING [S-7, 8, 9] Group activates CALL FORWARDING - BUSY LINE to a station (not a member of a Hunt Group) and system parameters specify that call hunting will be activated at the forwarded station, the system will send Busy Tone if the forwarded station is busy. If the system parameters specify hunting at the forwarding station, and the forwarded station is busy, the call will hunt in the forwarding stations Hunt Group.
11. When a station sets CALL FORWARDING - ALL CALLS [C-5] to a pilot in a STATION HUNTING [S-7, 8, 9] Group and the pilot activates CALL FORWARDING - BUSY LINE to another station, if system parameters define the call to hunt to the forwarded Hunt Group and all stations are busy, the Calling Party will hunt at the called stations Hunt Group. If system parameters define the call to hunt to the forwarding Hunt Group and all stations are busy, the Calling Party will then hunt to the forwarding stations Hunt Group.
12. If the Called Station is the pilot of a Uniform Call Distribution (UCD) [U-1] Group and activates CALL FORWARDING - BUSY LINE, the Calling Station will queue to the Uniform Call Distribution (UCD) [U-1] Group and will not call forward.
13. If a Calling Station, ATTENDANT CONSOLE [A-3], or Trunk receives a Busy Tone after CALL FORWARDING - BUSY LINE is set, and then activates EXECUTIVE RIGHT - OF - WAY [E-1], CALL WAITING - ORIGINATING [C-31], CALL WAITING - TERMINATING [C-12] or BUSY VERIFICATION [B-3], these services will be activated to the Calling Station.
14. There is no limit to the number of stations that can set CALL FORWARDING - BUSY LINE simultaneously.
15. The operator answering a call forwarded call can identify the Called Telephone Number on the ATTENDANT CONSOLE [A-3] display.
16. When a Station User places a call to a Busy Station which has been set CALL FORWARDING - BUSY LINE [C-2], the Calling Party can override the Call Forwarding feature, activating features associated with busy status such as CALL BACK [C-1], EXECUTIVE RIGHT-OF-WAY [E-1], STEP CALL [S-13], provided that the Service Feature Restriction Class (SFC) of the Calling Party permits this feature. Otherwise, the call is forwarded to a predetermined destination as usual. See the following command assignment to program this feature:
    ASFC - DAY / NIGHT
    TN: Tenant Number
    SFI: Service Feature Index 128 (Call Forwarding - Busy Line Override)
    SFC: Service Feature Restriction Class
    RES: Allowed
17. The OAI Monitor Number and the ACD Pilot Number cannot be assigned as the Call Forwarding Target Telephone Number except when called from a Station or a TIE Line Trunk.
18. When a call terminates to a Busy Station setting CALL FORWARDING - BUSY LINE, depending on the Calling Station Service Feature Restriction Class, it can be selected to make the station busy (not to activate CALL FORWARDING - BUSY LINE) and to activate such services as ATTENDANT CAMP-ON [A-1], BUSY VERIFICATION [B-3], CALL WAITING - TERMINATING [C-12], STEP CALL [S-13], etc.
19. This service is not available when the Called Party is the Attendant Console or Monitoring Station. However, the service is available when these Stations are designated as the Transfer Destination of CALL FORWARDING - BUSY LINE.
20. This service is available only when the Called Party is busy and has set CALL FORWARDING - BUSY LINE. In case CALL FORWARDING - BUSY LINE is set as the second hop and after in the Multiple Call Forwarding Link, this service is not available.

21. This service is available in the following occasions:
    a.) the destination of the CALL FORWARDING - BUSY LINE is the Calling Station.
    b.) the Calling Party is the Attendant Console or the Night Station, and the destination of the CALL FORWARDING - BUSY LINE is also the Attendant Console or the Night Station.
22. SPLIT CALL FORWARDING [S-99] is available only when the Calling Party is the Station or the CCIS Line.
23. A Digital Announcement Trunk cannot be set as the Forwarding Destination.
24. When using FCCS, a Telephone Number must be set to the Call Forwarding Destination.
25. A call can be forwarded to an Attendant Console in the Remote Node via FCCS.



26. When a call is forwarded through FCCS, the Numbering Plan of the originally Called Station must be identical with that of the forwarded Attendant Console.
27. The forwarding destination for this feature can be assigned with the ACFO/ACFOL/ACFON command on a tenant basis. At this time, the following conditions must be considered:
    a.) To forward an incoming trunk call to a station via FCCS, specify the destination station with TELN through the ACFOL/ACFON command.
    b.) Setting call forwarding destination from an individual station or Attendant Console overrides the setting specified with the ACFO/ACFOL/ACFON command.
    c.) The feature data via DM, LDM and NDM can coexist in a system. However, data registered on NDM has top priority. Data registered on LDM has precedence over data registered on DM. The priority is NDM → LDM → DM.
28. The forwarding destination for this feature can be assigned with ACFO/ACFOL/ACFON command on a route basis. For details, see CALL FORWARDING DESTINATION PER TRUNK ROUTE - INCOMING CALL [C-209].

    Note: Data assignment on a route basis is available since FP95-114 V4 Issue 2.0 (earlier than FP95-114 V4 Issue 2.0, the destination can be assigned with the ACFO/ACFOL/ACFON command on a tenant basis only).

29. If a Called Station is in a UNIFORM CALL DISTRIBUTION (UCD) Group or a Station Hunting Group, UNIFORM CALL DISTRIBUTION [U-11] and STATION HUNTING [S-7, 8, 9] are activated first rather than Call Forwarding - Busy Line (CF-1).
30. Specific Attendant Number cannot be set as Forwarding Destination Number.
31. Assign the System Data (ASYDL, SYS1, Index 677, Bit 3=1), and press the CALL FORWARDING - BUSY LINE FORWARD - BY key, a maximum of 16 digits Call Forwarding Information can be displayed.

    Note: This feature is available only for the terminal that can display 24 digits on LCD.

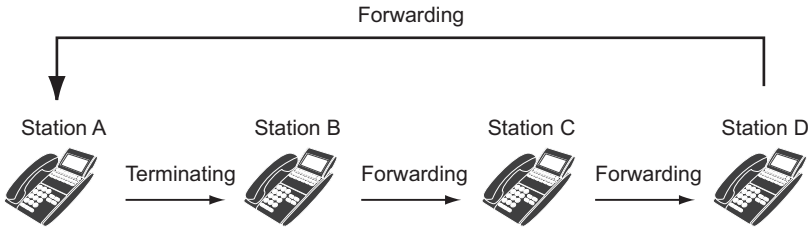
32. When SYS1, Index 69, Bit 1 [(Bit 2 - Call Forward Recall Tone). A burst of Ringback Tone to alert the person receiving a call that this is a CALL FORWARDING - ALL CALLS Call. In service? 0/1:No/Yes. Normally assigned as data "0".] is assigned, there will be no burst of Ringback Tone if the call is received by a SIP station.
33. Multiple Call Forwarding - Busy Line/All Calls/Logout (Logged Out IP Station - Call Destination [L-53]) can be selected to use or not by ASYD command [SYS1, Index 69, Bit 7 (0/1 = Out of service/In service)].

- When using Multiple Call Forwarding - Busy Line/All Calls/Logout.

  A maximum number of Multiple Call Forwarding - All Calls/Busy Line/Logout occurrences can be selected by ASYD command [SYS1, Index 5, Bit 1 to Bit 3].

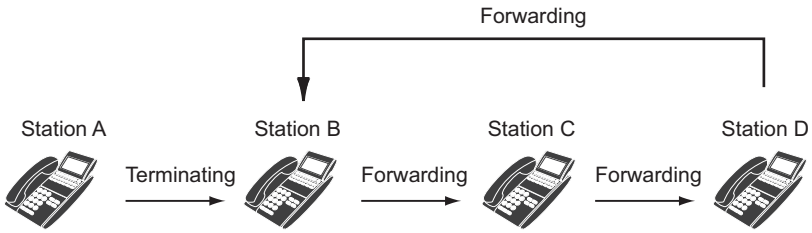
  The following shows the call forwarding operation that is looped within the assigned number of Multiple Call Forwarding - Busy Line/All Calls/Logout occurrences.

< If target station is Station A. >



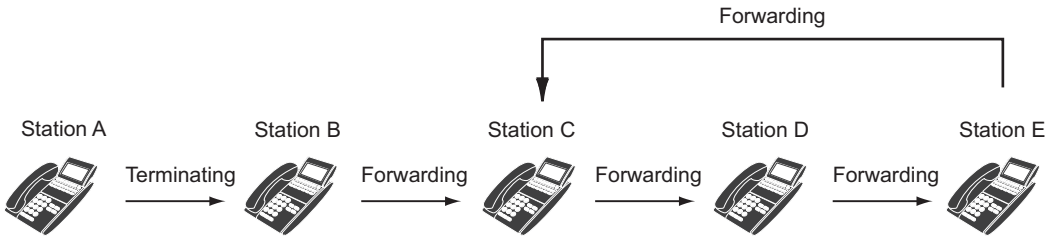
The call of Station A calls Station D. If Station D is busy, Station A hears a busy tone.

< If call forwarding is looped among Station B-D >



The call of Station A calls Station C. If Station C is busy, Station A hears a busy tone.

< If call forwarding is looped among Station C-E >



The call of Station A calls Station C. If Station C is busy, Station A hears a busy tone.

- When not using Multiple Call Forwarding - Busy Line/All Calls/Logout.

Depending on the call forwarding setting for destination terminal, the operation will change as follows.

Example: Call Forwarding Operation among Station A-D When Station A calls Station B

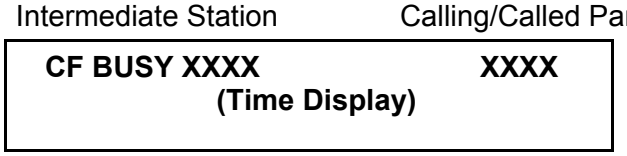
Station B sets	Station C sets	Operation
"Call Forwarding - All Calls [C-2]/Logged Out IP Station - Call Destination [L-53]" for Station C	"Call Forwarding - Busy Line [C-2]" for Station D	Station C is called.
"Call Forwarding - All Calls [C-2]/Logged Out IP Station - Call Destination [L-53]" for Station C	"Call Forwarding - All Calls [C-2]/Logged Out IP Station - Call Destination [L-53]" for Station D	Station D is called.
"Call Forwarding - Busy Line [C-2]" for Station C	"Call Forwarding - All Calls [C-2]/Logged Out IP Station - Call Destination [L-53]" for Station C	Station C is called.
"Call Forwarding - Busy Line [C-2]" for Station C	"Call Forwarding - Busy Line [C-2]" for Station D	Station A hears a busy tone.

<The following conditions are applied for Desktop terminals only.>

1. CALL FORWARDING - BUSY LINE may also be accessed via the dial pad or SPEED CALLING - ONE TOUCH [S-22].

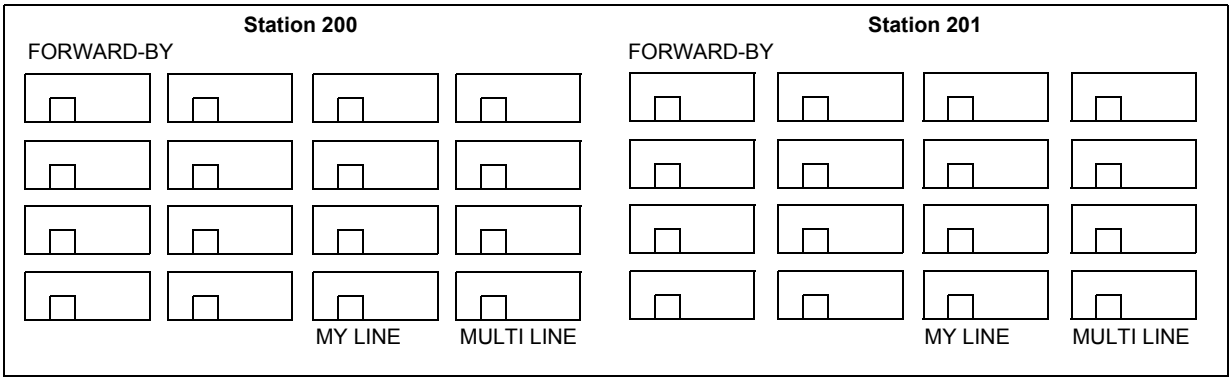
   Note: If this feature is accessed via the dial pad, it must be canceled via the dial pad.

2. If separate Access Codes are programmed into System Data for CALL FORWARDING - DON'T ANSWER [C-3] and CALL FORWARDING - BUSY LINE [C-2], separate Feature keys must be programmed on each Desktop terminal to equip it with both features.
3. Individual CALL FORWARDING - BUSY LINE supersedes Tenant basis CALL FORWARDING.
4. The display of an Intermediate Station on a Desktop terminal's LCD may be allowed or denied according to System Data. This assignment is done on a system basis. The LCD displays:



Note: When the Telephone Number of the Intermediate or Calling/Called Station exceeds eight digits, the last eight digits are displayed. However, a maximum of 16 digits Calling/Called Telephone Number can be displayed on the middle line of LCD depending on the System Data.

5. CALL FORWARDING - BUSY LINE supersedes STATION HUNTING [S-7, 8, 9].
6. This feature can be restricted according to the Service Feature Restriction Class of the Desktop terminal My Line. Even after the CALL FORWARDING - BUSY LINE [C-2] has been set to a Desktop terminal Multi Line, incoming calls to the Multi Line cannot be transferred if the Service Feature Restriction Class of the My Line restricts this feature.
7. A user may set CALL FORWARDING - BUSY LINE for a Multi Line Appearance based on the CLASS OF SERVICE - INDIVIDUAL [C-15], as assigned to the Station setting the service. The figure below is a simplified Key Sheet Assignment for Stations 200 and 201. The CLASS OF SERVICE - INDIVIDUAL [C-15] for Station 200 is allowed CALL FORWARDING - BUSY LINE, but the CLASS OF SERVICE - INDIVIDUAL [C-15] for Station 201 is not. Station 200 is able to set CALL FORWARDING - BUSY LINE for Station 201. The associated Feature key LED on Station 201 will illuminate.
   Even when a Sub Line of Desktop terminal is assigned as the Destination Station, the call is not forwarded if the My Line of that Desktop terminal is not allowed CALL FORWARDING - DON'T ANSWER.



Note: The FORWARD - BY key is same as the CALL FORWARDING - BUSY LINE Feature key.

8. This feature is applicable to CALL FORWARDING - OUTSIDE [C-28] and CALL FORWARDING/ MULTIPLE CALL FORWARDING - BUSY LINE - CCIS [C-46] [M-41].

Interactions

1. If Access Codes for CALL FORWARDING - BUSY LINE and CALL FORWARDING - DON'T ANSWER [C-3] are the same, only one key needs to be assigned in AKYD.
2. An ATTENDANT CAMP-ON [A-1] Call is terminated at Station A, and Station A has CALL FORWARDING - BUSY LINE set to Station B. If Station B is busy and part of a STATION HUNTING [S-7, 8, 9] Group, the call will Camp-On at Station A and will not hunt. Set via ASYD, SYS1, Index 69, Bit 5=0.
3. When CALL WAITING [C-12, 31], ATTENDANT CAMP-ON [A-1], and CALL BACK [C-1] is directed toward Station A, which has Call Forwarding set to Station B, and Station B is busy, Station A will receive Call Waiting Tone. Set via ASYD, SYS1, Index 69, Bit 5=0.
4. A non-controlling UNIFORM CALL DISTRIBUTION (UCD) Station is allowed to activate CALL FORWARDING - BUSY LINE [C-2]. When this station sets CALL FORWARDING - BUSY LINE, only calls directly dialed to this station are call forwarded.

5. The Attendant attempts BUSY VERIFICATION [B-3] to Station A. Station A has CALL FORWARDING - BUSY LINE set to Station B. Station B is busy. BUSY VERIFICATION is allowed at Station A. If Station A has set CALL FORWARDING - BUSY LINE to the Attendant Console, BUSY VERIFICATION [B-3] is allowed at Station A.
6. When CALL FORWARDING - BUSY LINE is in effect, the Activating Station may be used to originate calls.
7. When a call is set to call forward, the Destination Station is recorded for STATION MESSAGE DETAIL RECORDING (SMDR) [S-10].
8. STATION-TO-STATION CALLING [S-11] must be provided.
9. A Calling Station, Attendant Console, or Trunk can successfully activate EXECUTIVE RIGHT-OF-WAY [E-1], CALL WAITING - ORIGINATING [C-31], CALL WAITING - TERMINATING [C-12], or BUSY VERIFICATION [B-3] after having been call forwarded and having received Busy Tone. A trunk cannot activate EXECUTIVE RIGHT-OF-WAY [E-1] or BUSY VERIFICATION [B-3]. Set via ASYD, SYS1, Index 69, Bit 5=1.
10. A tone will not be provided when an access code is used to set this feature on SP Controlled Standard SIP terminal.
11. When SIP Handler Controlled SIP terminal is in Multi Line Mode, the Call Forwarding setting applies to the seized line while setting. In normal case, where My Line is not used by another station, the Call Forwarding setting applies to My Line. To set the Call Forwarding applying to Sub Line, use the ACFS/ ACFS_T commands.

Programming

<Call Forwarding - Busy Line with Access Code>

STEP 1: **ASYD** - SYS1, Index 5, Bit 0. Are Access Codes for CALL FORWARDING - BUSY LINE and CALL FORWARDING - DON'T ANSWER the same or separate? 0/1: Same/Separate.
SYS1, Index 5, Bit 1 to Bit 3. Maximum number of Multiple Call Forwarding - Busy Line/All Calls/ Logout (Logged Out IP Station - Call Destination [L-53]). **Note 1**

Note 1: This data is valid only when SYS1, Index 69, Bit 7 is assigned as"1".

STEP 2: **ASYDL** - SYS1, Index 677, Bit 3.
Display a maximum of 16 digits Call Forwarding Information on LCD? 0/1 = No (default)/Yes.

STEP 3: **ANPD** - Reserve a number level for Feature Access and Cancel.
Select Normal Service (N) from the combo box for Connection Indexes (applicable to any feature assigned to this level).
For Desktop terminal, select Normal Service (N), Hooking Service (H), or Busy Service (B) from the combo box for Connection Indexes.
Assign NND in accordance with a predetermined Numbering Plan.

STEP 4: **ASPA** - Assign an Access Code to CALL FORWARDING - BUSY LINE Entry, SID = 10 and to CALL FORWARDING - BUSY LINE Cancel, SID = 11. Assign Connection Status Index (CI) for "N: Normal" service.

STEP 5: **ASFC** - Assign a Service Feature Restriction Class that allows SFI 9 (Call Forwarding - Busy Line) to the Stations that receive CALL FORWARDING - BUSY LINE.

STEP 6: **AKYD** - For Desktop terminal sets, CALL FORWARDING - BUSY LINE may be assigned to a Line/ Feature key. Assign Feature key Call Forwarding - Busy Line [1].

STEP 7: **ADSL** - Assign SN = 3 (Called Station is Busy) and Feature key Call Forwarding - Busy Line [1]
This command is required only for assigning the function to each Soft key.

STEP 8: **ATNR** - Allow Tenant-to-Tenant Connection for Inter and Intra-tenant Connections. Assign Tenant Restriction Index (TRI) = 1. For this feature to be set, assign Inter and Intra-Tenant Connection TRI = 3 from the Attendant Console. This allows a Station in one Tenant to be call forwarded to a Station in the same or different Tenant. Also assign TRI = 0, Station-to-Station Calling.

<Call Forwarding - Busy Line with PCPro command (per Station)>

STEP 1: **ASYD** - SYS1, Index 79, Bit 2=1. Split Call Forwarding Service: 0/1 = Out/In service.

Note: This data is required only when Split Call Forwarding service is used.

STEP 2: **ACFS** - Specify a forwarding destination per station.
* When you set the data with Telephone Number, use ACFS_T command.
TN: Tenant Number
STN: Station Number
TYPE: Call Forwarding Type
1: Call Forwarding
2: Split Call Forwarding
SRV: Call Forwarding Kind
B: Busy Line
RCV: Kind of Received Call
This parameter (RCV) is available when "TYPE = Split Call Forwarding" is chosen.
T: Trunk
S: Station
CFD: Call Forwarding Destination Number (Maximum 24 digits)

<Call Forwarding - Busy Line with PCPro command (per Tenant)>

- Setting with Physical Station Number

ACFO - Specify a forwarding destination with Physical Station Number.
TN: Tenant Number
CF = 1: Call Forwarding - Busy Line
CFI = STN
STN: Physical Station Number of forwarding destination

Note: When you specify an Attendant Console as a forwarding destination, choose CFI = Attendant Console (ATT).

- Setting with Telephone Number

ACFOL/ACFON - Specify a forwarding destination with Telephone Number.
TN: Tenant Number
CF = 1: Call Forwarding - Busy Line
CFI = TELN
UGN: User Group Number of call forwarding destination
TELN: Telephone Number of forwarding destination (Maximum 16 digits)

Note: When you specify an Attendant Console as a forwarding destination, choose CFI = Attendant Console (ATT).

FCCS Programming

This data setting describes only the additional or replacement commands used to activate FCCS Services. When used as an FCCS feature, these commands replace the non-network commands (for example, AAED is replaced by AAEDN). Note that the data setting for FCCS Service varies depending on the Node.

Note: When using ASYDL or ASYDN to program an FCCS feature, the basic programming of ASYD is required in addition to ASYDL or ASYDN.

ASYDL - This Data must be assigned to all the nodes setting Call Forwarding service.
SYS1, Index 640: Assign the Fusion Point Code (FPC) of the node providing the Attendant Console.

Note: The data of ASFC is assigned to the node setting Call Forwarding service.

C-3 Call Forwarding - Don’t Answer

General Description

This feature permits a call to an unanswered Station to be forwarded to a predesignated Station, or to the Attendant, when the Called Station doesn’t answer after a predetermined time interval. If the individual Call Forwarding - Don’t Answer Timer is not assigned, the Default Timer assigned by System Data programming is used.
For Desktop terminals, CALL FORWARDING - DON’T ANSWER may be set or canceled by the user for all Multi Line Appearances on the Desktop terminal. Additionally, a single Station user may set CALL FORWARDING - DON’T ANSWER to all Sub Lines on the Desktop terminal.

Operating Procedure

To set from an individual Station using Access Code

1. Lift the handset. Receive Dial Tone.
2. Dial the specific CALL FORWARDING - DON’T ANSWER Access Code. Receive Special Dial Tone.
3. Dial the desired Target Telephone Number. Receive Service Set Tone.

Note: A tone will not be provided when an access code is used to set this feature on SP Controlled Standard SIP terminal.

To cancel from an individual Station using Cancel Code

1. Lift the handset. Receive Dial Tone.
2. Dial the specific CALL FORWARDING - DON’T ANSWER Cancel Code. Receive Service Set Tone.

To set from the ATTENDANT CONSOLE [A-3]

1. Press an Idle Loop key.
2. Dial the specific CALL FORWARDING - DON’T ANSWER Access Code. Receive Special Dial Tone.
3. Dial the TENANT SERVICE [T-12] Number (two or three digits).
4. Dial the Originating Telephone Number.
5. Dial the desired Target Telephone Number. Receive Service Set Tone.

To cancel from the ATTENDANT CONSOLE [A-3]

1. Press an Idle Loop key.
2. Dial the specific CALL FORWARDING - DON’T ANSWER Cancel Code. Receive Special Dial Tone.
3. Dial the TENANT SERVICE [T-12] Number (two or three digits).
4. Dial the Originating Telephone Number. Receive Service Set Tone.

To assign individual Call Forwarding - Don’t Answer Timer to each Station

1. Use the ADAI command.
   a.) Parameter NAT 139 overrides ASYD, SYS1, Index 139
   b.) Parameter NAT 140 overrides ASYD, SYS1, Index 140
   c.) Parameter NAT 141 overrides ASYD, SYS1, Index 141
   d.) Parameter NAT 247 overrides ASYD, SYS1, Index 247
2. Timer assigned to Called Party is valid.

Note: ADAI command is available only for Australia and North America.

<The following procedures are applied for Desktop terminals only.>

To set from My Line on an individual Station using a Line/Feature key

1. Lift the handset or press the SPEAKER key. Receive Dial Tone.
2. Press the CALL FORWARDING - DON'T ANSWER FORWARD - NA key. Receive Special Dial Tone.
3. Dial the desired Target Telephone Number. Receive Service Set Tone. The LCD displays:

Target Station

FORWARD SET	**XXXX**
(Time Display)	

To monitor from a Desktop terminals from My Line

1. Press the CALL FORWARDING - DON'T ANSWER FORWARD - NA key. The LCD displays:

FORWARD	**XXXX**
(Time Display)	

OR

FWD SET
(Time Display)

To set for a Multi Line Appearance other than the My Line

1. Lift the handset or press the SPEAKER key.
2. Press the Multi Line Appearance. Receive Dial Tone. Press the CALL FORWARDING - DON'T ANSWER FORWARD - NA key. Receive Special Dial Tone.
3. Dial the desired Target Telephone Number. Receive Service Set Tone. The LED of the associated Feature key does not light at the Station setting the service. If the Multi Line Appearance is another Desktop terminal's My Line, that Station's Feature key LED will light. The setting Station's LCD displays:

Target Station:

FORWARD SET	**XXXX**
(Time Display)	

4. Replace the handset or press the SPEAKER key.
5. If the Target Station is the Attendant, the LCD displays:

FWD SET OPR
(Time Display)

To monitor for a Multi Line Appearance other than the My Line

1. Press the CALL FORWARDING - DON'T ANSWER FORWARD - NA key. The LCD displays:

Target Station

FORWARD	**XXXX**
(Time Display)	

To cancel from My Line on an individual Station using a Line/Feature key

1. Lift the handset or press the SPEAKER key. Receive Dial Tone.
2. Press the CALL FORWARDING - DON'T ANSWER FORWARD - NA key. The associated LED goes out. Receive Service Set Tone. The LCD displays:

FORWARD CANCEL **(Time Display)**

To cancel for a Multi Line Appearance other than the My Line

1. Lift the handset or press the SPEAKER key.
2. Press the Multi Line Appearance. Receive Dial Tone. Press the CALL FORWARDING - DON'T ANSWER FORWARD - NA key. Receive Service Set Tone. The LCD displays:

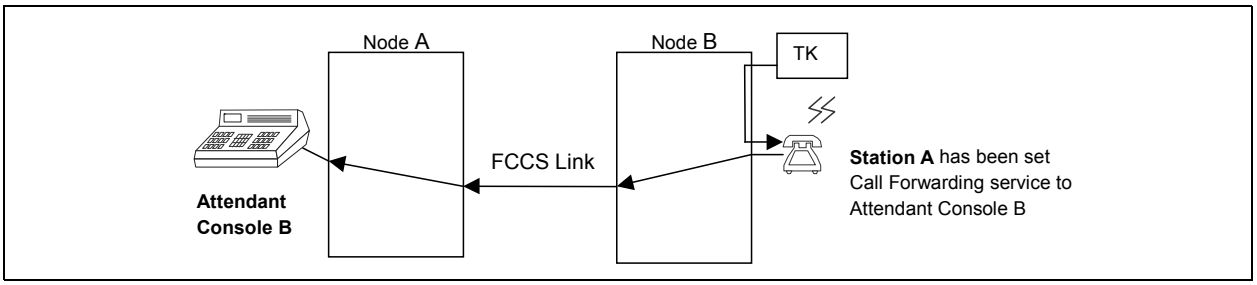
FORWARD CANCEL **(Time Display)**

   The associated LED that is lit at another Station goes out.
3. Replace the handset or press the SPEAKER key.

Service Conditions

1. This feature may be set and canceled by either the user or the Attendant.
2. When the Attendant extends a call to a Station that has set CALL FORWARDING - DON'T ANSWER, the Called Station will ring for ten seconds. If the call is not answered, AUTOMATIC RECALL [A-8] is activated. This timing can be changed through the VARIABLE TIMING PARAMETERS [V-1] feature.
3. This feature is assigned by CLASS OF SERVICE - INDIVIDUAL [C-15] programming in System Data.
4. A Station programmed for access to CALL FORWARDING - DON'T ANSWER is automatically allowed CALL FORWARDING - BUSY LINE [C-2].
5. There is no limitation to the number of Stations that can set the CALL FORWARDING - DON'T ANSWER simultaneously.
6. CALL FORWARDING - DON'T ANSWER can be provided on either a Tenant or an individual basis.
7. Individual CALL FORWARDING - DON'T ANSWER takes precedence over Tenant basis Call Forwarding.
8. Call Forwarding on a Tenant basis allows direct Incoming Trunk Calls that encounter a Don't Answer condition to be forwarded to a predetermined Station or the ATTENDANT CONSOLE [A-3].
9. The operator answering a Call Forwarding Call can identify the Called Telephone Number on the ATTENDANT CONSOLE [A-3] display.
10. CALL FORWARDING - DON'T ANSWER from the Attendant is only activated after pressing the RELEASE key.
11. If the Station's own Telephone Number is set for CALL FORWARDING - DON'T ANSWER, then this feature is changed to CALLING PARTY RECOGNITION SERVICE (CALL FORWARDING - DON'T ANSWER [C-103]).

12. When the Called Station does not answer for a predetermined time interval, Stations are assigned this feature according to the TENANT SERVICE [T-12] Class of the Calling and Destination Stations. In the following cases, when the Calling Station and Destination Station are restricted by TENANT SERVICE [T-12] Class, this feature is not activated, and the Called Station continues to ring:
    a.) when the Station Called is setting CALL FORWARDING - DON'T ANSWER,
    b.) when a Station holding an original call (Station or Trunk) on CONSULTATION HOLD [C-17] places a call to a Station setting CALL FORWARDING - DON'T ANSWER.
13. Call Transfer - Attendant (Trunk): Trunk Calls transferred by the Attendant are assigned this feature when the Called Station does not answer for a predetermined time interval according to Route Restriction Class of the Trunk and the Destination Stations. When restricted by Route Restriction Class, this feature is not activated, and the Called Station continues to ring.
14. Call Transfer - Attendant (Station): Station Calls transferred by the Attendant are assigned this feature when the Called Station does not answer for a predetermined time interval according to TENANT SERVICE [T-12] Class of the Originating and Destination Stations. When restricted by Tenant Service Class, this feature is not activated, and the Called Station continues to ring.
15. Speed Calling List Numbers may be programmed as the Destination Telephone Number. This permits the sent digits to exceed the 12-digit maximum.
16. CALL FORWARDING - DON'T ANSWER Data remains in the Data Memory, even if the system or LP is re-initialized.
17. SPLIT CALL FORWARDING [S-99]: When a call is forwarded to the parties predesignated by SPLIT CALL FORWARDING [S-99], Service Conditions are the same as described in Items 12, 13, and 14.
18. When a call is transferred from the Attendant to a Station that has set CALL FORWARDING - DON'T ANSWER, this feature may be restricted or activated according to SYS2, Index 6, Bit 7.
19. Digital Announcement Trunk cannot be set as Forwarding Destination.
20. If individual Call Forwarding - Don't Answer Timer is not given, the Default Timer is used by ASYD.
21. A call can be forwarded to an Attendant Console in the Remote Node via FCCS.



22. When a call is forwarded through FCCS, the Numbering Plan of the originally Called Station must be identical with the one of the Forwarded Attendant Console.
23. The forwarding destination for this feature can be assigned with the ACFO/ACFOL/ACFON command on a tenant basis. At this time, the following conditions must be considered:
    a.) To forward an incoming trunk call to a station via FCCS, specify the destination station with TELN through the ACFOL/ACFON command.
    b.) Setting call forwarding destination from an individual station or Attendant Console overrides the setting specified with the ACFO/ACFOL/ACFON command.

c.) The feature data via DM, LDM and NDM can coexist in a system. However, data registered on NDM has top priority. Data registered on LDM has precedence over data registered on DM. The priority is NDM → LDM → DM.

24. The forwarding destination for this feature can be assigned with ACFO/ACFOL/ACFON command on a route basis. For details, see CALL FORWARDING DESTINATION PER TRUNK ROUTE - INCOMING CALL [C-209].

Note: Data assignment on a route basis is available since FP95-114 V4 Issue 2.0 (earlier than FP95-114 V4 Issue 2.0, the destination can be assigned with the ACFO/ACFOL/ACFON command on a tenant basis only).

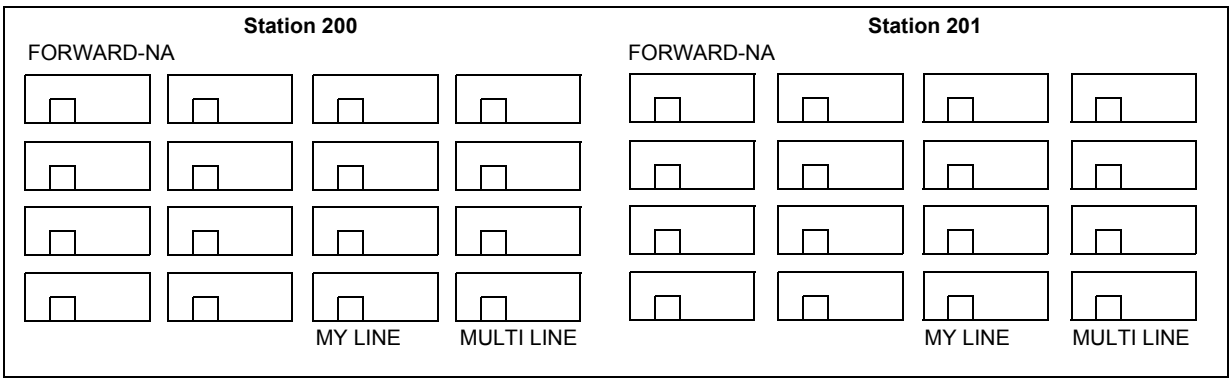
25. Specific Attendant Number cannot be set as Forwarding Destination Number.
26. Assign the System Data (ASYDL, SYS1, Index 677, Bit 3=1), and press the CALL FORWARDING - DON'T ANSWER FORWARD - NA key, a maximum of 16 digits Call Forwarding Information can be displayed.

Note: This feature is available only for the terminal that can display 24 digits on LCD.

27. A station number including self office code cannot be assigned as the destination number of PS. (It can be assigned but not be transferred)

The following conditions are applied for Desktop terminals only.

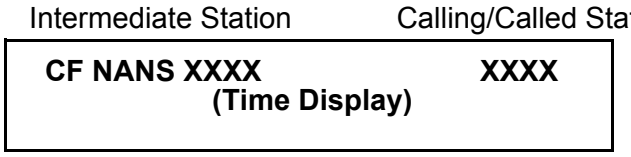
1. If separate Access Codes are programmed into System Data for CALL FORWARDING - DON'T ANSWER and CALL FORWARDING - BUSY LINE [C-2], each feature must be programmed on a separate Feature key at the Desktop terminal.
2. CALL FORWARDING - DON'T ANSWER may also be accessed via the dial pad or SPEED CALLING - ONE TOUCH -D [S-26D]. If this feature is accessed via the dial pad, it must be canceled via the dial pad.
3. CALL FORWARDING - DON'T ANSWER is not applicable to CONSULTATION HOLD - ALL CALLS [C-17], and CALL TRANSFER - ALL CALLS [C-11].
4. A user may set CALL FORWARDING - DON'T ANSWER for a Multi Line Appearance based on the CLASS OF SERVICE - INDIVIDUAL [C-15] as assigned to the Station (My Line) setting the service. The figure below is a simplified Key Sheet Assignment for Stations 200 and 201. The CLASS OF SERVICE - INDIVIDUAL [C-15] for Station 200 is allowed CALL FORWARDING - DON'T ANSWER, but the CLASS OF SERVICE - INDIVIDUAL [C-15] for Station 201 is not. Station 200 is able to set CALL FORWARDING - DON'T ANSWER for Station 201. The associated Feature key LED on Station 201 will illuminate.



Note: The FORWARD - NA key is same as the CALL FORWARDING - DON'T ANSWER Feature key.

Even when a Sub Line of Desktop terminal is assigned as the Destination Station, the call is not forwarded if the My Line of that Desktop terminal is not allowed CALL FORWARDING - DON'T ANSWER.

5. This feature is applicable to CALL FORWARDING - ALL CALLS - OUTSIDE [C-28], CALL FORWARDING - DON'T ANSWER - CCIS [C-47], and MULTIPLE CALL FORWARDING -DON'T ANSWER - CCIS [M-42].
6. The display of an Intermediate Station on a Desktop terminal's LCD may be allowed or denied according to System Data. This arrangement is done on a system basis. The LCD displays:



Note: When the Telephone Number of the Intermediate or Calling/Called Station exceeds eight digits, the last eight digits are displayed. However, a maximum of 16 digits Calling/Called Telephone Number can be displayed on the middle line of LCD depending on the System Data.

Interactions

1. If Access Codes for CALL FORWARDING - BUSY LINE and CALL FORWARDING - DON'T ANSWER are the same, only one key needs to be assigned in AKYD.
2. ACFO is applicable to DID, DIT, and TIE Line Calls only.
3. A Non-Controlling UCD Station is allowed to activate CALL FORWARDING - DON'T ANSWER. When this Station sets Call Forwarding, calls directed to this Station will be Call Forwarded. The member Station is not removed from the UCD Group under the conditions of the Call Forwarding feature used. When it is the Station's turn to ring, that call is affected by CALL FORWARDING - DON'T ANSWER.
4. CALL FORWARDING - DON'T ANSWER to an Attendant Console will not operate when the system is in Night mode. If ATTENDANT NIGHT TRANSFER [A-19] is provided, the call is forwarded to the ATTENDANT NIGHT TRANSFER [A-19] Station.
5. If the CALL FORWARDING-DON'T ANSWER Target Station is a Desktop terminal that has activated DO NOT DISTURB [D-21D], Call Forwarding is not activated. The Station will ring, and the timer setting in SYS1, Index 145 is ignored.
6. The CALL FORWARDING - DON'T ANSWER feature is set from Station A to Station B if Station A is idle and Station B is busy. The timer setting in SYS1, Index 145 is ignored.
7. When a call is Call Forwarded, the Destination Station is recorded for STATION MESSAGE DETAIL RECORDING (SMDR) [S-10].
8. STATION-TO-STATION CALLING [S-11] must first be provided.
9. A tone will not be provided when an access code is used to set this feature on SP Controlled Standard SIP terminal.
10. When SIP Handler Controlled SIP terminal is in Multi Line Mode, the Call Forwarding setting applies to the seized line while setting. In normal case, where My Line is not used by another station, the Call Forwarding setting applies to My Line. To set the Call Forwarding applying to Sub Line, use the ACFS/ ACFS_T commands.

Programming

<Call Forwarding - Don't Answer with Access Code>

STEP 1: **ASYD** - SYS1, Index 5, Bit 0. Are the Access Codes for CALL FORWARDING - BUSY LINE [C-2] and CALL FORWARDING - DON'T ANSWER the same or separate? 0/1: Same/Separate.

SYS1, Index 69, Bit 1 and Bit 2 is a Short Tone to alert person receiving the call that this is a CALL FORWARDING - DON'T ANSWER Call? 0/1: No/Yes.

Note: There will be no Short Tone if the call is received by a SIP station.

SYS1, Index 139. Assign the No Answer Timer for Station-to-Station, DID and TIE Line Calls. For 30 seconds, assign 00H. (RAM Data is 3FH.)

Note: By using ADAI, No Answer Timer can be assigned on a Terminal basis. ADAI command is available only for Australia and North America.

SYS1, Index 141. Assign the No Answer Timer for Incoming Calls via the Attendant Console. It will then be forwarded to the next Station according to SYS1, Index 145 timer. For ten seconds, assign 00H. (RAM Data is 35H.)

SYS1, Index 145. Assign the time an Incoming Call via the Attendant Console will ring at the CALL FORWARDING - DON'T ANSWER Station before recalling to the Attendant Console. For 30 seconds, assign 00H. [RAM Data is 9FH.]

SYS2, Index 6, Bit 7. Enable CALL FORWARDING - DON'T ANSWER via the Attendant Console.

STEP 2: **ASYDL** - SYS1, Index 677, Bit 3.
Display a maximum of 16 digits Call Forwarding Information on LCD? 0/1 = No (default)/Yes.

STEP 3: **ANPD** - Reserve a Number Level for Feature Access and Cancel. Select Normal Service (N), Hooking Service (H), or Busy Service (B) from the combo box for Connection Indexes. (in order to be applicable to any feature assigned to this level). Assign NND in accordance with a predetermined Numbering Plan.

STEP 4: **ASPA** - Assign an Access Code to CALL FORWARDING - DON'T ANSWER entry, SID = 12, and to CALL FORWARDING-DON'T ANSWER cancel, SID = 13. Assign CI for "N: Normal" service.

STEP 5: **ASFC** - Assign the Stations to receive CALL FORWARDING - DON'T ANSWER a Service Feature Restriction Class that allows SFI 8 (Call Forwarding - Don't Answer).

STEP 6: **ADSL** - Assign SN=1 (Originating; DT Connection) and Feature key Call Forwarding Don't Answer; FDN [22]. This command is used for assigning the function to each Soft key.

STEP 7: **AKYD** - For Dterm sets, CALL FORWARDING - DON'T ANSWER may be assigned to a Line/ Feature key. Assign Feature key Call Forward - Don't Answer [22].

STEP 8: **ATNR** - Allow Tenant-to-Tenant Connection for Inter and Intra-Tenant Connections. Assign Tenant Restriction Index (TRI) = 1. For this feature to be set by the Attendant Console, assign Inter and Intra-Tenant Connection via the Attendant Console, TRI = 3. Also assign TRI = 0, Station-to-Station Calling.

CFI of either the Attendant Console or a Station.

STEP 9: **ADAI** - Assign NAT139, NAT140, NAT141 and NAT247 for Individual Call Forwarding - Don't Answer Timer.

Note: ADAI command is available only for Australia and North America.

<Call Forwarding - Don't Answer with PCPro command (per Station)>

STEP 1: **ASYD** - SYS1, Index 79, Bit 2=1. Split Call Forwarding Service: 0/1 = Out/In service.

Note: This data is required only when Split Call Forwarding service is used.

STEP 2: **ACFS** - Specify a forwarding destination per station.
* When you set the data with Telephone Number, use ACFS_T command.
TN: Tenant Number
STN: Station Number
TYPE: Call Forwarding Type
1: Call Forwarding
2: Split Call Forwarding
SRV: Call Forwarding Kind
D: Don't Answer
RCV: Kind of Received Call
This parameter (RCV) is available when "TYPE = Split Call Forwarding" is chosen.
T: Trunk
S: Station
CFD: Call Forwarding Destination Number (Maximum 24 digits)

<Call Forwarding - Don't Answer with PCPro command (per Tenant)>

- Setting with Physical Station Number

ACFO - Specify a forwarding destination with Physical Station Number.
TN: Tenant Number
CF = 2: Call Forwarding - Don't Answer
CFI = STN **Note 1**
STN: Physical Station Number of forwarding destination

- Setting with Telephone Number

ACFOL/ACFON - Specify a forwarding destination with Telephone Number.
TN: Tenant Number
CF = 2: Call Forwarding - Don't Answer
CFI = TELN **Note 2**
UGN: User Group Number of call forwarding destination
TELN: Telephone Number of forwarding destination (Maximum 16 digits)

Note 1: When you specify an Attendant Console as a forwarding destination, choose CFI = Attendant Console (ATT).

Note 2: When you specify an Attendant Console as a forwarding destination, choose CFI = Attendant Console (ATT).

FCCS Programming

This data setting describes only the additional or replacement commands used to activate FCCS Services. When used as an FCCS feature, these commands replace the non-network commands (for example, AAED is replaced by AAEDN). Note that the data setting for FCCS Service varies depending on the Node.

Note: When using ASYDL or ASYDN to program an FCCS feature, the basic programming of ASYD is required in addition to ASYDL or ASYDN.

ASYDL - This data must be assigned to the Node setting Call Forwarding Service.
SYS1, Index 640: Assign the FPC of the Node providing the Attendant Console.

Note: The data of ASFC is assigned to the Node setting Call Forwarding Service.

C-5 Call Forwarding - All Calls

General Description

This feature permits all calls destined for a particular Station to be routed to another Station, or to the Attendant, regardless of the busy or idle status of the Called Station. Activation and cancellation may be accomplished by the user or the Attendant. This feature may be set or canceled by the user for all Multi Line Appearances on the Dterm. Additionally, a single user may set this feature to all Sub Lines on the Dterm.

Operating Procedure

To set and cancel from an individual Station using the same Access Code Note 1 Note 2

- For setting:

1. Lift the handset. Receive Dial Tone.
2. Dial the CALL FORWARDING - ALL CALLS entry Access Code. Receive Special Dial Tone.
3. Dial the desired Forwarding Destination Number. Receive Service Set Tone.

- For cancellation:

1. Lift the handset. Receive Dial Tone.
2. Dial the CALL FORWARDING - ALL CALLS entry Access Code. Receive Special Dial Tone.

Note 1: Available from FP95-114 V4 Issue 2.0 with system data ASYDL, SYS1, Index 686, Bit 0= 1.

To set and cancel from an individual Station using different Access Codes Note 2

- For setting:

1. Lift the handset. Receive Dial Tone.
2. Dial the CALL FORWARDING - ALL CALLS entry Access Code. Receive Special Dial Tone.
3. Dial the desired Forwarding Destination Number. Receive Service Set Tone.

- For canceling:

1. Lift the handset. Receive Dial Tone.
2. Dial the CALL FORWARDING - ALL CALLS cancel Access Code. Receive Service Set Tone.

To set from the ATTENDANT CONSOLE [A-3]

1. Press an Idle Loop key.
2. Dial the CALL FORWARDING - ALL CALLS entry Access Code. Receive Special Dial Tone.
3. Dial the TENANT SERVICE [T-12] Number (two digits).
4. Dial the Originating Telephone Number.
5. Dial the desired Forwarding Destination Number. Receive Service Set Tone.

To cancel from the ATTENDANT CONSOLE [A-3]

1. Press an Idle Loop key.
2. Dial the specific CALL FORWARDING - ALL CALLS cancel Access Code. Receive Special Dial Tone.
3. Dial the TENANT SERVICE[T-12] Number (two digits).
4. Dial the Originating Telephone Number. Receive Service Set Tone.

Note 2: A tone will not be provided when an access code is used to set this feature on SP Controlled Standard SIP terminal.

<The following procedures are applied for Desktop terminal only.>

To set from My Line using a Line/Feature key

1. Lift the handset or press the SPEAKER key. Receive Dial Tone.
2. Press the CALL FORWARDING - ALL CALLS Feature key. Receive Special Dial Tone.
3. Dial the desired Forwarding Destination Number. Receive Service Set Tone. The LED will light. The LCD displays:

Target Station

FORWARD SET XXXX **(Time Display)**

4. Replace the handset or press the SPEAKER key.

To monitor from My Line

1. Press the CALL FORWARDING - ALL CALLS Feature key. The LCD displays:

Target Station

FORWARD XXXX **(Time Display)**

OR

Operator

FWD SET **(Time Display)**

To cancel from My Line using a Line/Feature key

1. Lift the handset or press the SPEAKER key. Receive Dial Tone.
2. Press the CALL FORWARDING - ALL CALLS Feature key. Receive Service Set Tone. The LED of the associated Feature key will go out. The LCD displays:

FORWARD CANCEL **(Time Display)**

3. Replace the handset or press the SPEAKER key.

To set for a Multi Line Appearance other than the My Line

1. Lift the handset or press the SPEAKER key.
2. Press the Multi Line Appearance. Receive Dial Tone. Then press the CALL FORWARDING - ALL CALLS Feature key. Receive Special Dial Tone.
3. Dial the desired Forwarding Destination Number. Receive Service Set Tone. The LED of the associated Feature key does not light at the Station setting the service. If the Multi Line Appearance is another Dterm's My Line, that Station's Feature key LED will light. The setting Station's LCD displays:

Target Station

FORWARD SET XXXX (Time Display)

4. Replace the handset or press the SPEAKER key.
5. If the Target Station is the operator, the LCD displays:

FWD SET OPR (Time Display)

To monitor for a Multi Line Appearance other than the My Line

1. Press the Multi Line Appearance. Then press the CALL FORWARDING - ALL CALLS Feature key. The LCD displays:

Target Station

FORWARD XXXX (Time Display)

To cancel for a Multi Line Appearance other than the My Line

1. Lift the handset or press the SPEAKER key. Receive Dial Tone.
2. Press the Multi Line Appearance. Receive Dial Tone. Press the CALL FORWARDING - ALL CALLS Feature key. Receive Service Set Tone. The LCD displays:

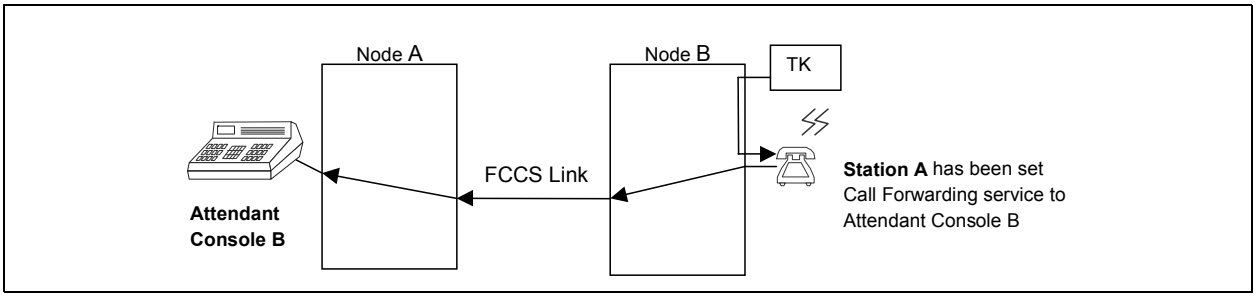
FORWARD CANCEL (Time Display)

   The associated LED that is lit at another Station goes out.
3. Replace the handset or press the SPEAKER key.

Service Conditions

1. Stations with this feature set may originate any calls and access the CALL FORWARDING - ALL CALLS cancellation Code.
2. There is no limitation to the number of Stations that can set CALL FORWARDING - ALL CALLS simultaneously.
3. Stations may be assigned this feature via CLASS OF SERVICE - INDIVIDUAL [C-15] programming in System Data.
4. When a call is forwarded to another Station, one burst of Ringing Signal will be sent to the Station to indicate that it is Call Forwarded.
5. Stations activating CALL FORWARDING - ALL CALLS may be restricted from placing any calls, except to the Attendant. (This is an optionally programmable feature).
6. When the Operator answers a Forwarded Call, System Data programming can provide a Called Telephone Number display on the ATTENDANT CONSOLE [A-3].
7. CALL FORWARDING - ALL CALLS Data remains in the Data Memory, even if the system is re-initialized or in the event of a power failure.

8. If the Station's own Telephone Number is set for CALL FORWARDING - ALL CALLS, this feature is changed to CALLING PARTY RECOGNITION SERVICE (CALL FORWARDING - ALL CALLS) [C-103].
9. When the Station user setting CALL FORWARDING - ALL CALLS goes off-hook, Special Dial Tone can be provided instead of Dial Tone (activation of this service is determined by System Data).
10. The OAI Monitor Number and the ACD Pilot Number cannot be assigned as the Call Forwarding Target Telephone Number except when called from a Station or TIE Line Trunk.
11. When a destination of CALL FORWARDING - ALL CALLS has already been assigned, even if the feature is restricted by the new Office Data or changing of Day/Night mode, the assigned data can still be used or canceled.
12. A Digital Announcement Trunk can be set as Forwarding Destination.
13. A Telephone Number must be set to the call Forwarding Destination.
14. A call can be forwarded to an Attendant Console in the Remote Node via FCCS.



15. When a call is forwarded through FCCS, the Numbering Plans of the originally Called Station and the Forwarded Attendant Console must be the same.
16. Specific Attendant Number cannot be set as Forwarding Destination Number.
17. Assign the System Data (ASYDL, SYS1, Index 677, Bit 3=1), and press the CALL FORWARDING - ALL CALLS Feature key, a maximum of 16 digits Call Forwarding Information can be displayed.

    Note: This feature is available only for the terminal that can display 24 digits on LCD.

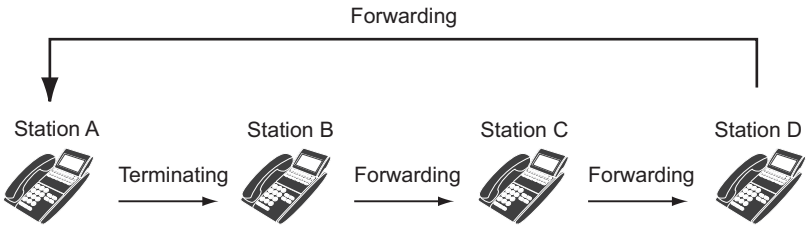
18. Multiple Call Forwarding - Busy Line/All Calls/Logout (Logged Out IP Station - Call Destination [L-53]) can be selected to use or not by ASYD command [SYS1, Index 69, Bit 7 (0/1 = Out of service/In service)].

    - When using Multiple Call Forwarding - Busy Line/All Calls/Logout.

      A maximum number of Multiple Call Forwarding - All Calls/Busy Line/Logout occurrences can be selected by ASYD command [SYS1, Index 5, Bit 1 to Bit 3].

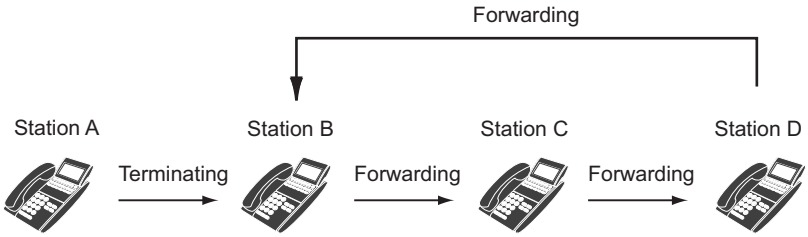
The following shows the call forwarding operation that is looped within the assigned number of Multiple Call Forwarding - Busy Line/All Calls/Logout occurrences.

< If target station is Station A. >



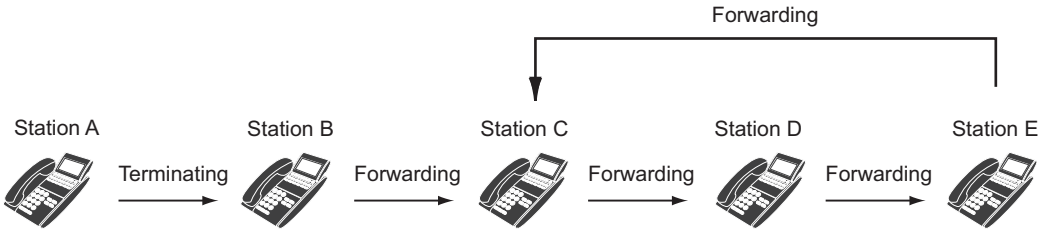
The call of Station A calls Station D. If Station D is busy, Station A hears a busy tone.

< If call forwarding is looped among Station B-D >



The call of Station A calls Station C. If Station C is busy, Station A hears a busy tone.

< If call forwarding is looped among Station C-E >



The call of Station A calls Station C. If Station C is busy, Station A hears a busy tone.

- When not using Multiple Call Forwarding - Busy Line/All Calls/Logout.

Depending on the call forwarding setting for destination terminal, the operation will change as follows.

Example: Call Forwarding Operation among Station A-D When Station A calls Station B

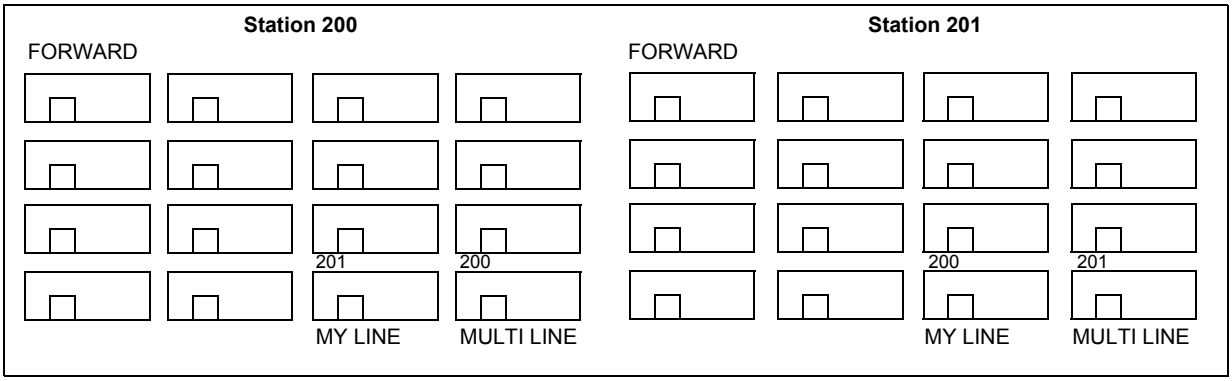
Station B sets	Station C sets	Operation
"Call Forwarding - All Calls [C-2]/Logged Out IP Station - Call Destination [L-53]" for Station C	"Call Forwarding - Busy Line [C-2]" for Station D	Station C is called.
"Call Forwarding - All Calls [C-2]/Logged Out IP Station - Call Destination [L-53]" for Station C	"Call Forwarding - All Calls [C-2]/Logged Out IP Station - Call Destination [L-53]" for Station D	Station D is called.
"Call Forwarding - Busy Line [C-2]" for Station C	"Call Forwarding - All Calls [C-2]/Logged Out IP Station - Call Destination [L-53]" for Station C	Station C is called.
"Call Forwarding - Busy Line [C-2]" for Station C	"Call Forwarding - Busy Line [C-2]" for Station D	Station A hears a busy tone.

19. By registering the Call Forwarding - All Calls Access Code + the desired Forwarding Destination Number to the CF Setting button of a Line/Feature key on a Desktop terminal, it is possible to enable and disable CALL FORWARDING - ALL CALLS [C-5] just by pressing the CF Setting button. For a detailed explanation, see CALL FORWARDING (CF) BUTTON - ONE TOUCH [C-208].

Note: CALL FORWARDING (CF) BUTTON - ONE TOUCH [C-208] is available from FP95-114 V4 Issue 2.0.

<The following conditions are applied for Desktop terminal only.>

1. CALL FORWARDING - ALL CALLS can only be set from the My Line or Sub Line.
2. CALL FORWARDING - ALL CALLS may also be accessed via the dial pad or SPEED CALLING - ONE TOUCH [S-26D]. If this feature is accessed via the dial pad, it must also be canceled via the dial pad.
3. A Station user may set CALL FORWARDING - ALL CALLS for a Multi Line Appearance based on the CLASS OF SERVICE - INDIVIDUAL [C-15] assigned to the Station setting the service. The figure below is a simplified Key Sheet Assignment for Stations 200 and 201. The CLASS OF SERVICE - INDIVIDUAL [C-15] for Station 200 is allowed CALL FORWARDING - ALL CALLS, but the CLASS OF SERVICE - INDIVIDUAL [C-15] for Station 201 is not. Station 200 is able to set CALL FORWARDING - ALL CALLS for Station 201. The associated Feature key LED on Station 201 will illuminate.



Note: The FORWARD key is same as the CALL FORWARDING - ALL CALLS Feature key.

4. This feature is applicable to CALL FORWARDING - ALL CALLS - OUTSIDE [C-28] and CALL FORWARDING and MULTIPLE CALL FORWARDING - ALL CALLS - CCIS [C-45] [M-40].
5. The display of an Intermediate Station on a Dterm's LCD may be allowed or denied according to System Data. This assignment is done on a system basis. The LCD displays:

Intermediate Station Calling/Called Party

CF ALL XXXX	**XXXX**
(Time Display)	

Note: When the Telephone Number of the Intermediate or Calling/Called Station exceeds eight digits, the last eight digits are displayed. However, a maximum of 16 digits Calling/Called Telephone Number can be displayed on the middle line of LCD, depending on the System Data.

Interactions

1. Users are unaccustomed to hearing Ringback Tone when responding to their own Station's ring. Inform users when providing Ringback Tone with CALL FORWARDING - ALL CALLS in SYS1, Index 69, Bit 1.
2. Call Forwarding can only be activated from an idle Station.
3. When CALL WAITING [C-12, 31], ATTENDANT CAMP-ON [A-1], and CALL BACK [C-1] are directed toward Station A, with CALL FORWARDING set to Station B, Station B will receive Call Waiting Tone.
4. When a Station in a STATION HUNTING [S-7, 8, 9] Group initiates CALL FORWARDING - ALL CALLS, calls normally destined for this Station will not be received. The Hunt Group operates as if the Station were busy.
5. A Non-Controlling UCD Station is allowed to activate CALL FORWARDING [C-5]. With this Station Call Forwarded, only calls that are directly dialed to this Station are Call Forwarded. The member Station is removed from the UCD Group under the conditions of this Call Forwarding feature.
6. If an Attendant attempts BUSY VERIFICATION [B-3] to Station A, which has CALL FORWARDING - ALL CALLS set to Station B, BUSY VERIFICATION [B-3] proceeds as if the Attendant had attempted to Verify Station B. If Station A has set CALL FORWARDING - ALL CALLS to the Attendant Console, BUSY VERIFICATION [B-3] is allowed at Station A if Station A is busy.
7. When CALL FORWARDING - ALL CALLS is in effect, the Call Forwarded Station may be used to originate calls. See SYS2, Index 6, Bit 4.
8. When a Station is Call Forwarded, the Destination Station is recorded for STATION MESSAGE DETAIL RECORDING (SMDR) [S-10].
9. CALL FORWARDING - ALL CALLS can be overridden, allowing Station B to Initiate or Transfer Calls back to Station A.
10. STATION-TO-STATION CALLING [S-11] must first be provided.
11. A tone will not be provided when an access code is used to set this feature on SP Controlled Standard SIP terminal.
12. When SIP Handler Controlled SIP terminal is in Multi Line Mode, the Call Forwarding setting applies to the seized line while setting. In normal case, where My Line is not used by another station, the Call Forwarding setting applies to My Line. To set the Call Forwarding applying to Sub Line, use the ACFS/ ACFS_T commands.

Programming

STEP 1: **ASYD** - SYS1, Index 4, Bit 6. One burst of ringing at the Forwarding Station when Call Forwarding - All Calls is in service? 01: No/Yes.
SYS1, Index 5, Bit 1 to Bit 3. Maximum number of Multiple Call Forwarding - Busy Line/All Calls/ Logout (Logged Out IP Station - Call Destination [L-53]). **Note 3**
SYS1, Index 69, Bit 1 (Bit 2 - Burst of Tone for Recall). A burst of Ringback Tone to alert the person receiving a Call Forwarding - All Calls Call. In service? 0/1: No/Yes. Normally assigned as data "0". **Note 4**
SYS1, Index 248, Bit 7. Dial Tone to be sent out when a Station on which CALL FORWARDING - ALL CALLS is set, goes off-hook to place an Outgoing Call 0/1: Dial Tone/Special Dial Tone.
SYS 2, Index 6, Bit 4. Enables Station set for Call Forwarding - All Calls to be allowed to use phones normally. If data "0" is assigned, the Station will only be able to call the Attendant Console. Assign on a per Tenant basis.

STEP 2: **ASYDL** - SYS1, Index 677, Bit 3.
Display a maximum of 16 digits Call Forwarding Information on LCD? 0/1 = No (default)/Yes.
SYS1, Index 686, Bit 0=1. **Note 5**
Disabling Call Forwarding - All Calls with entry Access Code is in service.

STEP 3: **ASYDN** - SYS1, Index 686, Bit 0=1. **Note 5**
Disabling Call Forwarding - All Calls with entry Access Code is in service.

STEP 4: **ANPD** - Reserve a Number Level for Feature Access and Cancel (**Note 6**). Select Normal Service (N), Hooking Service (H), or Busy Service (B) from the combo box for Connection Indexes. Assign NND in accordance with a predetermined Numbering Plan.

STEP 5: **ASPA** - Assign an Access Code to Call Forwarding - All Calls entry SID = 8, and to Call Forwarding - All Calls cancel SID = 9 (**Note 6**). Assign Connection Status Index (CI) for "N: Normal" service.

STEP 6: **ASFC** - Assign a Service Feature Restriction Class that allows SFI 7 (Call Forwarding - All Calls) to Stations that will activate Call Forwarding - All Calls.

STEP 7: **AKYD** - For Dterm sets, Call Forwarding - All Calls may be assigned to a Line/Feature key. Assign Feature key Call Forwarding - All Calls [2].

STEP 8: **ADSL** - Assign SN = 1 (Originating) and Feature key Call Forwarding - A - All Calls [2].
This command is required only for assigning the function to each Soft key.

STEP 9: **ATNR** - Allow Tenant-to-Tenant Connection for Inter and Intra-Tenant Connections. Assign Tenant Restriction Index (TRI) = 1. For this feature to be set by the Attendant Console, assign Inter and Intra-Tenant Connection TRI = 3, via the Attendant Console. Also allow TRI = 0, Station-to-Station Calling.

Note 3: This data is valid only when SYS1, Index 69, Bit 7 is assigned as"1".

Note 4: There will be no burst of Ringback Tone if the call is received by a SIP station.

Note 5: Available from FP95-114 V4 Issue 2.0.

Note 6: When SYS1, Index 686, Bit 0=1 (Disabling Call Forwarding - All Calls with entry Access Code is in service), Call Forwarding - All calls cancel is not required because the cancellation can be performed with the entry Access Code.

FCCS Programming

This data setting describes only the additional or replacement commands used to activate FCCS Services. When used as an FCCS feature, these commands replace the non-network commands (for example, AAED is replaced by AAEDN). Note that the data setting for FCCS Service varies depending on the Node.

Note: When using ASYDL or ASYDN to program an FCCS feature, the basic programming of ASYD is required in addition to ASYDL or ASYDN.

ASYDL - This data must be assigned to the Node setting Call Forwarding Service.
SYS1, Index 640: Assign the Fusion Point Code (FPC) of the Node providing the Attendant Console.

Note: The data of ASFC is assigned to the Node setting Call Forwarding Service.

C-6 Call Hold

General Description

This feature permits any user to hold a call in progress by performing a switch hook flash and dialing a pre-programmed CALL HOLD Code or pressing the HOLD Feature key, thus allowing that Line to be used for originating another call or returning to a previously Held Call.

Operating Procedure

To hold a call in progress using Access Code

1. Press the switch hook. Receive Special Dial Tone.
2. Dial CALL HOLD Code. Receive normal Dial Tone.
3. The call in progress is held. The user may initiate a new call.

To release a new call and return to the held call

1. Go on-hook, releasing the new call.
2. The original call rings back to Station.
3. Go off-hook and continue with the original call.

To hold a new call and return to original call

1. Press the switch hook. Receive Special Dial Tone.
2. Dial the CALL HOLD Code. The new call is now held.
3. The original call is automatically returned.

Note: By repeating the above steps, users can talk alternately with two parties.

<The following procedures are applied for Desktop terminal only.>

To hold a call in progress without the HOLD Feature key

1. Press the TRANSFER key. Receive Special Dial Tone. The LCD displays:

TRANSFER **(Time Display)**

2. Dial the CALL HOLD [C-6] Code. Receive normal Dial Tone.
3. The user may initiate a new call.

To release a new call and return to the original call

1. Go on-hook, releasing the new call, or press the RECALL key.
2. The original call will ringback to the Dterm.

To hold a new call and return to the original call

1. Press the TRANSFER key. Receive Special Dial Tone.
2. Dial CALL HOLD [C-6] Code. The new call in progress is held, the original call is automatically returned.

Note: By repeating the above steps, user can talk alternately with two parties.

To hold a call in progress with the HOLD key programmed to a Line/Feature key

1. While a call is in progress, press the HOLD key.
2. The call is placed on HOLD, Dial Tone is restored.
3. A new call may be dialed.

To release a new call and return to the original call

1. Go on-hook, releasing the new call, or press the RECALL key.

To hold a new call and return to the original call

1. Press the HOLD key. The new call in progress is held, the original call is automatically returned.

Note: By repeating the above steps, user can talk alternately with two parties.

On a Dterm, a SPEED CALLING - ONE TOUCH [S-26] may be used to place a call on HOLD

1. Press FEATURE key, plus a SPEED CALLING - ONE TOUCH [S-26] key. The LCD will display any previously stored data.
2. Press the RECALL key. The LCD displays:



   This acts as a switch hook flash. See FLASH ENTRY -D [F-4D].
3. Dial CALL HOLD [C-6] Access Code.
4. Press the SPEED CALLING - ONE TOUCH [S-26] key. The LCD displays:



5. This key may now be used to access the CALL HOLD feature.
6. When engaged in a conversation, press the SPEED CALLING - ONE TOUCH [S-26] key with the stored CALL HOLD Access Code.
7. Dial Tone is restored to the telephone, and user is free to make another call.

To operate a Dterm via the FLASH BUTTON [F-3D]

1. When engaged in a conversation, press the programmable Feature/Line key programmed as a FLASH BUTTON [F-3D]. The LCD will display:



2. Dial the CALL HOLD [C-6] Access Code. The LCD goes blank and Dial Tone is restored to the user.

Service Conditions

1. Lines freed through use of this feature may also be used for answering Incoming Calls via the CALL PICKUP - GROUP [C-7] or TRUNK ANSWER FROM ANY STATION [T-8] features.
2. If the Controlling user does not dial any further digits after the CALL HOLD Code, removal of Dial Tone will occur after 12 seconds; Reorder Tone is then applied for 30 seconds followed by muting of the Station set.
3. Calls will remain on hold until the Controlling user either replaces the handset, causing the Held Call to ringback, or will press the TRANSFER key or Flash Button and redial the Hold Code to return to the original call.
4. Only one call at a time may be held per Station Line, and the Held Call cannot be added to another call as in THREE-WAY CALLING [T-2].
5. Stations may be assigned this feature via CLASS OF SERVICE - INDIVIDUAL [C-15] programming in System Data.
6. There is no limitation to the number of Stations that may simultaneously access this feature.
7. CALL WAITING [C-12, 31] cannot be activated if the Target Station has a call on HOLD.
8. CALL BACK [C-1]:
   a.) A user cannot activate CALL HOLD when CALL BACK [C-1] has already been activated.
   b.) A user cannot use the CALL HOLD feature while CALL BACK [C-1] is in effect toward it.
   c.) A user with a call on HOLD cannot activate the CALL BACK [C-1] feature.
9. CALL PICKUP [C-7, 30]: A user is denied use of the CALL PICKUP [C-7, 30] feature when holding one call and in a talking state with another call.
10. THREE-WAY CALLING [T-2]: This feature used in conjunction with the CALL HOLD feature, allows the Calling Station user to transfer a second call while holding the first call. After transferring the second call, the Calling Station user is recalled by the party on hold.
11. EXECUTIVE RIGHT-OF-WAY [E-1]: Executive Override of a Station that has a call on HOLD and is connected to another call is allowed. However, Executive Override of a Station on HOLD is denied (Reorder Tone is heard).
12. CALL FORWARDING [C-2, 3, 5]: This feature interacts with CALL HOLD as follows: CALL HOLD ringbacks are not forwarded. Instead, the Calling Station is recalled with the Held Call after going on-hook.
13. BUSY VERIFICATION [B-2]: Busy Verification of a Station on HOLD is denied. Busy Verification of a Station that has a call on HOLD is allowed if the Station is in a talking state with another party.
14. OUTGOING TRUNK QUEUING [O-2, 2D]: A Station user cannot be placed in a Ringback Queue while holding another call in hold.
15. Attendant Calls cannot be held.
16. ATTENDANT CAMP-ON [A-1]: Call hold cannot be activated when ATTENDANT CAMP-ON has already been set.
17. This feature is not available for ISDN trunk, ISDN terminal, and PS.
18. This feature cannot be used when the party on the other end is PS.

19. The Operating Station is recalled by the party on CALL HOLD after going on-hook in the following situations:
    - operational error (dialing a wrong number or an incomplete number, etc.)
    - the second station is busy
    - the second station is ringing
    - when engaged in a two-party connection with the second station
    - etc.
20. During a new call after activating the CALL HOLD feature on the Desktop terminal, press the ANSWER key to return to the original call. This operation is same as dialing CALL HOLD [C-6] Code after pressing the TRANSFER key, or pressing the HOLD key.
21. A SIP Multiple Line terminal, a Soft Phone (SIP) and an MG with a setting that hold tone is sent out from the terminal itself plays the hold tone back from the beginning when the counter station dials a Call Hold access code.
22. If External IP Music-on-Hold of VS32 CONFERENCE SERVER [V-26] is used, the hold tone after dialing the Call Hold access code is replayed from the beginning.

Interactions

1. CONSULTATION HOLD [C-17] must be allowed for all features that will operate by using the switch hook to make a connection. This includes CALL TRANSFER [C-11], CALL PICKUP [C-7], THREE-WAY CALLING [T-2], etc.
2. A Station can CALL HOLD only one call at a time.
3. CALL WAITING [C-12, 31]: When a Station has a call on CALL HOLD, CALL WAITING [C-12, 31] features are denied to both the holding and the held Station.
4. CALL BACK [C-1]:
    a.) A user cannot activate CALL HOLD when CALL BACK [C-1] has already been activated.
    b.) A user cannot use the CALL HOLD feature while CALL BACK [C-1] is in effect toward it.
    c.) A user with a call on HOLD cannot activate the CALL BACK [C-1] feature.
5. NON-EXCLUSIVE and EXCLUSIVE HOLD [N-7D, E-4D] are not affected by CALL BACK [C-1].
6. CALL PICKUP [C-7, 30]: A Station user is allowed use of the CALL PICKUP [C-7, 30] feature when holding Station B to pickup Station C.
7. EXECUTIVE RIGHT-OF-WAY [E-1]: Executive Override of a Station that has a call on HOLD and is connected to another call is allowed. However, Executive Override of a Station on HOLD is denied (Reorder Tone is heard).
8. AUTOMATIC RECALL [A-8] of a Held Call will not be affected by CALL FORWARDING [C-2, 3, 5].
9. BUSY VERIFICATION [B-2]: Busy Verification of a Station on HOLD is denied. Busy Verification of a Station that has a call on HOLD is allowed if the Station is in a talking state with another party

10. OUTGOING TRUNK QUEUING [O-2]: A Station user cannot queue a Trunk while using CALL HOLD.
11. Calls that include the Attendant Console cannot have access to CALL HOLD.
12. ATTENDANT CAMP-ON WITH TONE INDICATION [A-1]: This feature is denied if a Station has a call on CALL HOLD.
13. CALL HOLD cannot be activated if a call has been previously camped-on. A switch hook flash will seize the Camped-On Call.
14. Calls remain on CALL HOLD until the Controlling Station replaces the handset, causing the Held Call to ringback. If the Controlling Station does not answer the recall within the preset time-out interval (SYS1, Index 139), the connection is released.
15. This feature cannot be used with SP Controlled/SIP Handler Controlled Standard SIP terminals.

Programming

STEP 1: **ASYD** - SYS1, Index 139. No Answer Timer for CALL HOLD. (When Station A places Station B on HOLD, and hangs up, Station A will ring after a preset period of time has elapsed. If Station A does not answer, then Station B goes from Ringback Tone to Reorder Tone.)

SYS2, Index 1, Bit 0 and Bit 1. CONSULTATION HOLD [C-17] allowed or denied. 00: denied; Originating and Terminating Calls allowed, Tandem Calls denied; 11: All calls allowed.

STEP 2: **ANPD** - Reserve a Number Level for Feature Access and Cancel. Select Normal Service (N), Hooking Service (H), or Busy Service (B) from the combo box for Connection Indexes. Assign NND in accordance with a predetermined Numbering Plan.

STEP 3: **ASPA** - Assign an Access Code for CALL HOLD, SID = 1 for Connection Status Index (CI) of "H: Hooking".

STEP 4: **ASFC** - Assign a Service Feature Restriction Class that allows SFI 10 (Call Hold) to Stations that are allowed CALL HOLD.

STEP 5: **AKYD** - Optional. Assign CALL HOLD to a Line/Feature key. Assign Feature key Call Hold [10]. (The operation of TRANSFER key and Access Code is also available). Assign Feature key Flash Button [21].

STEP 6: **ADSL** - Assign SN = 5 (Consultation Hold) and Feature key Call Hold [10].

This command is required only for assigning the function to each Soft key.

To operate a Dterm via the FLASH BUTTON [F-3D], assign SN = 4 (Communication is in Progress) and Feature key Flash Button [21].

FCCS Programming

This data setting describes only the additional or replacement commands used to activate FCCS Services. When used as an FCCS feature, these commands replace the non-network commands (for example, AAED is replaced by AAEDN). Note that the data setting for FCCS Service varies depending on the Node.

Note: When using ASYDL or ASYDN to program an FCCS feature, the basic programming of ASYD is required in addition to ASYDL or ASYDN.

STEP 1: **ANPDL** - Reserve a Number Level for Feature Access and Cancel. Select Normal Service (N), Hooking Service (H), or Busy Service (B) from the combo box for Connection Indexes. Assign NND in accordance with a predetermined Numbering Plan.

STEP 2: **ASPAL** - Assign an Access Code for CALL HOLD, SID = 1 for Connection Status Index (CI) of "H: Hooking".

Note: System Timers may be set on a network basis in ASYDN.

C-7 Call Pickup - Group

General Description

This feature permits a user to answer any calls directed to other Lines in the user's preset CALL PICKUP - GROUP by using a Line/Feature key or dialing a Pickup Code.

This feature may be used by seizing Dial Tone from any Multi Line Appearance on the Desktop terminal.
A user can be assigned to an additional CALL PICKUP - GROUP, referred to as EXPANDED CALL PICKUP - GROUP.

Operating Procedure

To pick up an Incoming Call using Access Code

1. A Station within the CALL PICKUP - GROUP rings.
2. Lift the handset. Receive Dial Tone.
3. Dial the CALL PICKUP Code. The call is connected to the Station.

To pick up an Incoming Call (CALL HOLD [C-6]) using Access Code

1. A Station within the CALL PICKUP - GROUP rings.
2. The Station flashes the switch hook. Receives Special Dial Tone.
3. Dial the CALL PICKUP Code. The call is connected to the Station.
4. The first party is put on CALL HOLD [C-6] status.

To pick up an Incoming Call within the EXPANDED CALL PICKUP - GROUP using Access Code

1. A Station within the EXPANDED CALL PICKUP - GROUP rings.
2. Lift the handset. Receive Dial Tone.
3. Dial the Access Code for EXPANDED CALL PICKUP. The call is connected to the Station.

<The following procedures are applied for Desktop terminal only.>

To pick up an Incoming Call using a Line/Feature key

1. A Station within the CALL PICKUP - GROUP rings.
2. Lift the handset. Receive Dial Tone.
3. Press the CALL PICKUP - GROUP Feature key. The call is connected to the Station. The LCD displays:

Called Station	Calling Station Trunk
PICKUP XXXX	**XXXX**
(Time Display)	

Note: When the Telephone Number of the Called/Calling Station exceeds eight digits, the last eight digits are displayed. However, a maximum of 16-digit Calling Telephone Number can be displayed on the middle line of the LCD, depending on the System Data.

To pick up an Incoming Call CALL HOLD [C-6] using a Line/Feature key

1. A Station within the CALL PICKUP - GROUP rings.
2. Press the TRANSFER/HOLD Feature key. Receive Special Dial Tone.
3. Press the CALL PICKUP - GROUP Feature key. The call is connected to the Station.

4. The first party is put on CALL HOLD [C-6] status.

To pick up an Incoming Call within the EXPANDED CALL PICKUP - GROUP using a Line/Feature key

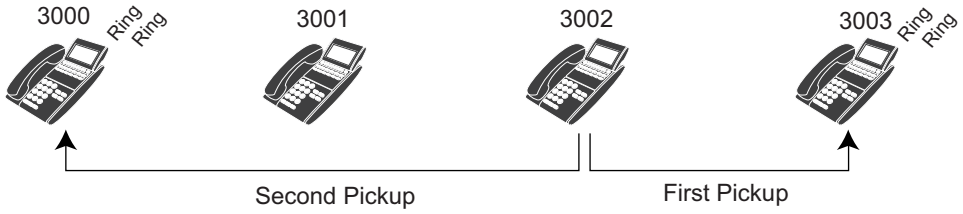
1. A Station within the EXPANDED CALL PICKUP - GROUP rings.
2. Lift the handset. Receive Dial Tone.
3. Press the Feature key for EXPANDED CALL PICKUP. The call is connected to the Station.

Service Conditions

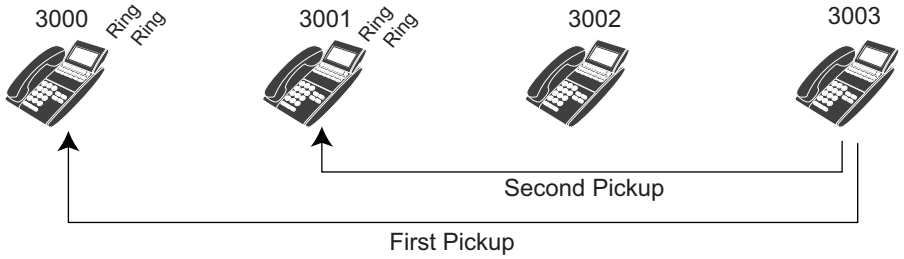
1. This feature may be provided on a per line basis via System Data.
2. Any Station assigned a fully restricted CLASS OF SERVICE - INDIVIDUAL [C-15] will be unable to answer Incoming Central Office Calls using this feature.
3. A CALL PICKUP - GROUP is defined as those Stations having the ability to pick up each other's calls using the CALL PICKUP feature.
4. A maximum of 100 Stations can be assigned to each group. There is no limit to the number of groups per system.
5. PS can pick up a call except for the time for roaming from self office to another office. Multi Line of Desktop terminal which accommodates PS, or L2 line of PS Dual Line capable can pick up the call during the PS roaming.
6. If the CALL PICKUP - GROUP Code is dialed while two or more Stations within the group are ringing, the Switching System will connect the calls in the order of priority assigned to the Call Pickup Chain. This feature allows the user to pickup from the next Station in assigned order. See the following figure:

Assume that a total of 4 stations (3000, 3001, 3002 and 3003) are configured as a Call Pickup Group.
These Stations are assigned as members of a Call Pickup Group in the following order :
3000 to 3001 to 3002 to 3003

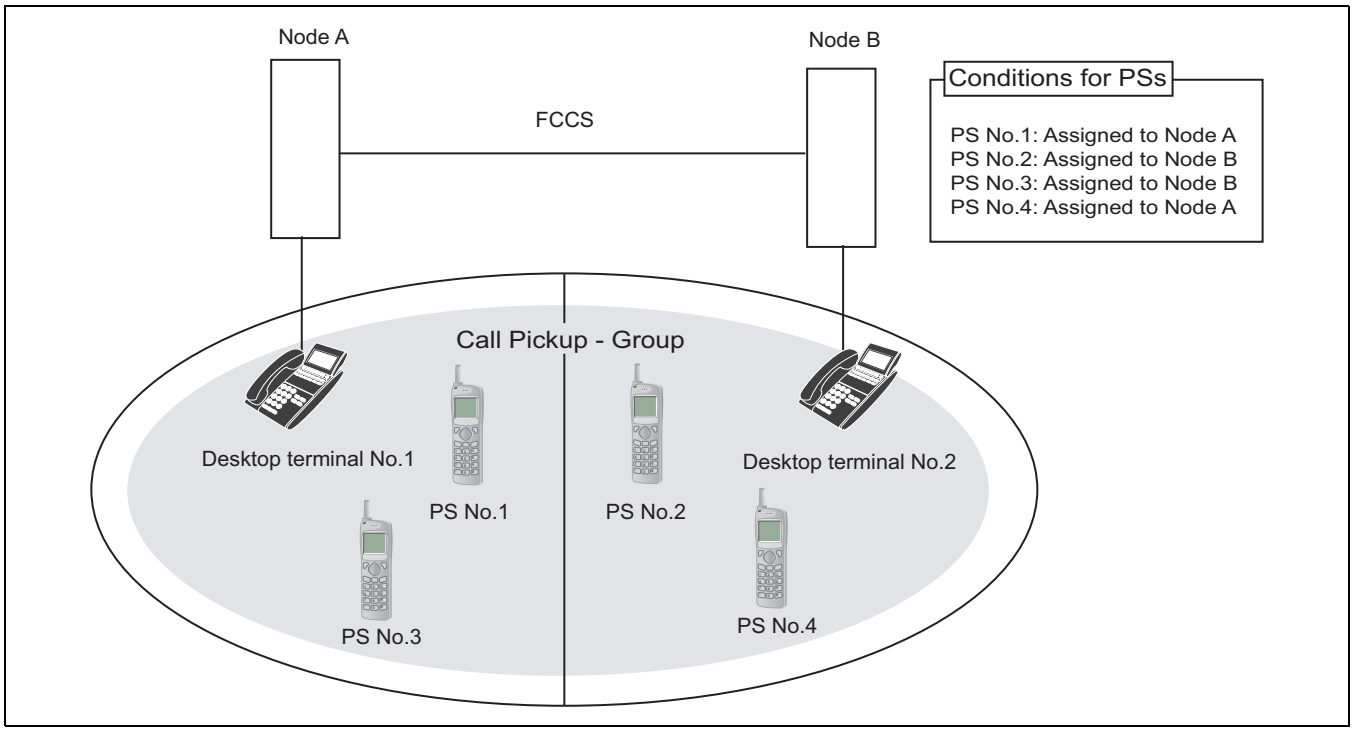
[Example 1] When Stations 3000 and 3003 are ringing:
If the user answers via Station 3002, the user will pick up a call terminated to Station 3003 first.
The user will pick up Station 3000, if Station 3000 is still ringing.



[Example 2] When Stations 3000 and 3001 are ringing:
If the user answers via Station 3003, the user will pick up a call terminated to Station 3000 first.
The user will pick up Station 3001, if Station 3001 is still ringing.



7. The conditions for Call Pickup-Group including PSs are as follows:



Call Pickup Decision Table (X: Available/N: Not Available)

Ringing Terminals	Terminal picking up a call					
	Desktop terminal No.1	Desktop terminal No.2	PS No.1	PS No.2	PS No.3	PS No.4
Desktop terminal No.1	-	-	-	-	-	-
Desktop terminal No.2	-	-	-	-	-	-
PS No.1	-	-	-	-	-	-
PS No.2	-	-	-	-	-	-
PS No.3	N Note 1	N Note 1	N Note 1	N Note 1	-	N Note 1
PS No.4	N Note 1	N Note 1	N Note 1	N Note 1	N Note 1	-

Note 1: If L2 (SubLine) feature is used in the ringing terminal, Call Pickup is available.

8. To register a group including PS, use the ACPGN command. If the ACPG command is used in this case, Picking up any incoming call to PS will not be available.
9. A CALL PICKUP - GROUP cannot be used to answer OUTGOING TRUNK QUEUING [O-2] or CALL BACK [C-1] from another Station.

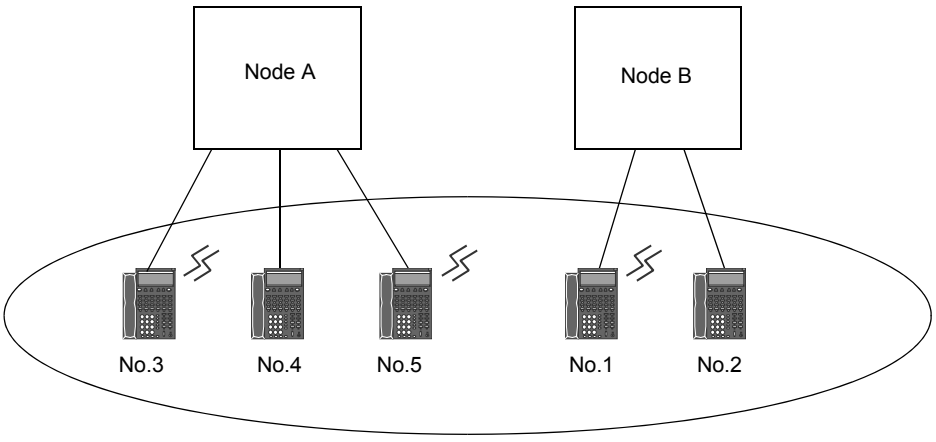
10. A busy Station can accomplish Call Pickup by flashing the switch hook and then dialing the CALL PICKUP - GROUP Code. The call in progress will automatically be placed on hold.
11. The CALL PICKUP - GROUP feature cannot be used to answer the following types of calls:
    - Automatic callback from the station number requested through CALL BACK [C-1]
    - Automatic callback from the outgoing trunk queue requested through OUTGOING TRUNK QUEUING [O-2]
    - Recall indicating unsuccessful connection to another party in CALL HOLD [C-6]
    - Call from the paged individual through Paging Answer Access Code in the Delay System of PAGING TRANSFER [P-10]
12. A maximum of 100 Stations can be assigned to each EXPANDED CALL PICKUP - GROUP. There is no limit to the number of groups per system.
13. A user cannot pickup a call in the EXPANDED CALL PICKUP - GROUP when engaged in another call.
14. This feature is available via FCCS.
15. The allowable Station Number type for a member of CALL PICKUP - GROUP or CALL PICKUP EXPAND - GROUP is the Telephone Number only.
16. When using the FCH Card with:
    a.) Stations in up to a maximum of four different Nodes can be assigned to a CALL PICKUP - GROUP or an EXPANDED CALL PICKUP - GROUP.
    b.) When a CALL PICKUP - GROUP/EXPANDED CALL PICKUP - GROUP has Stations that belong to different Nodes, and the Multiple Call Pickup/Call Pickup Expand Stations are ringing simultaneously, if ringing Stations belong to the same Node, the user will pick up the call in assigned order. However, if ringing Stations belong to different Nodes, the user cannot pick up the call in assigned order or terminated order. The user will first pick up the call terminated to the Station belonging to the same Node of the user's Station. See the figures below.

CALL PICKUP - GROUP

The number below the Station is the assigned order.

Example: All the Stations No.1 through No. 5 belong to the same CALL PICKUP - GROUP. When Stations No. 1, No. 3 and No. 5 are ringing:

- If the user answers via Station No.4, the user will pick up a call terminated to Station No.5 first (this feature allows the user to pickup from the next Station in assigned order). The user will then pick up Station No. 3, if Station No. 3 is still ringing.
- If the user answers via Station No. 2, the user will pick up a call terminated to Station No. 1 first.

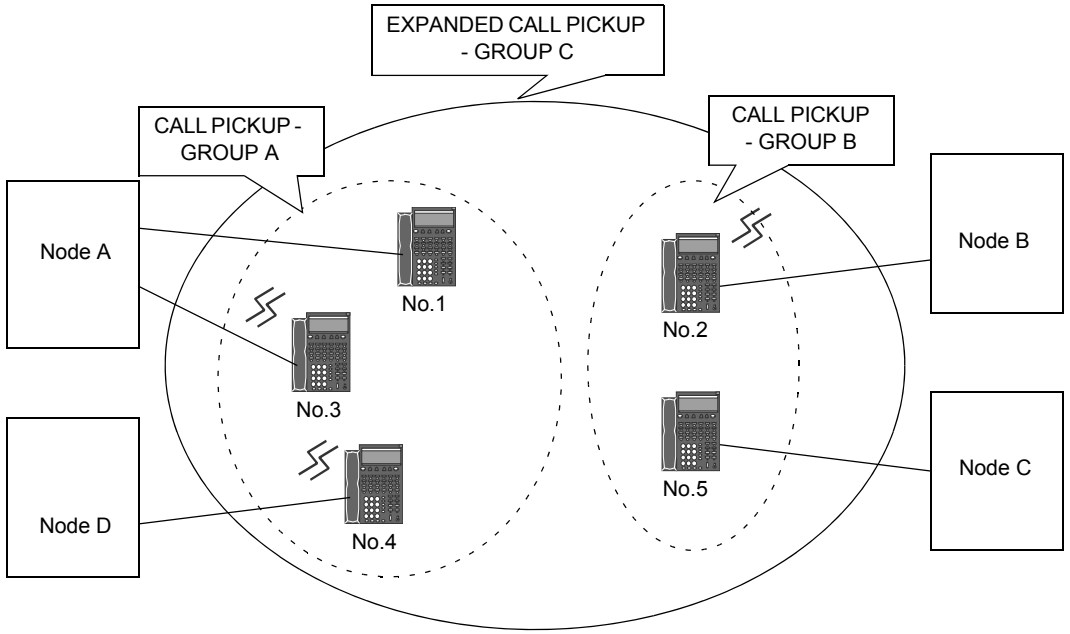


EXPANDED CALL PICKUP - GROUP

The number below the Station is the assigned order.

Example: All the Stations No.1 through No.5 belong to the same EXPANDED CALL PICKUP - GROUP C.
When Stations No.2 through No.4 are ringing simultaneously:

- If the user answers via Station No. 1, the user will pick up a call terminated to Station No. 3 first.
- If the user answers via Station No. 5, the user will pick up a call terminated to Station No. 2 first.



c.) The number of CALL PICKUP - GROUPs or EXPANDED CALL PICKUP - GROUPs is not limited.

17. A Station user cannot answer an Incoming Call to the Station in another CALL PICKUP - GROUP or EXPANDED CALL PICKUP - GROUP.
18. Even if a Station has been assigned to the existing CALL PICKUP - GROUP or EXPANDED CALL PICKUP - GROUP by Physical Station Number, the Station can also be assigned to the same CALL PICKUP - GROUP or an EXPANDED CALL PICKUP - GROUP via FCCS Service.
19. When the Station holds a call (switch hook flash or Call Hold) which terminated through FCCS, the Station cannot pick up a call via CALL PICKUP/CALL PICKUP EXPAND.
20. When both Stations, a Calling Station and a Station that will picks up a call, place another call on hold, this service is activated on the condition that all the Stations (the Calling Station, the Station that picks up the call, the ringing Station, and the held Stations) belong to the same Node. If any of the Stations belong to a different Node, this service is not activated.
21. The CALL PICKUP - GROUP/EXPANDED CALL PICKUP - GROUP can be set on a Node basis, even in the use of Telephone Numbers. In other words, the features are available via the data written on the Local Data Memory (LDM), not only on the Data Memory (DM) or the Network Data Memory (NDM). The following are conditions in this case:
    a.) The features are to be programmed by using the Telephone Number (TELN) of each Station.
    b.) A maximum of 100 Stations can be assigned per CALL PICKUP - GROUP.
    c.) A maximum of 100 Stations can be assigned per EXPANDED CALL PICKUP - GROUP.
    d.) A Station can belong to one CALL PICKUP - GROUP only.

e.) A Station can belong to one EXPANDED CALL PICKUP - GROUP only.
f.) No limitation is attached to the number of CALL PICKUP - GROUPs/EXPANDED CALL PICKUP - GROUPs per Node.
g.) When multiple Stations are ringing in a CALL PICKUP - GROUP/EXPANDED CALL PICKUP - GROUP, the calls won't be picked up in the terminating order or in the assigned order of Stations.
h .) A Station cannot pick up a call to a member of other CALL PICKUP - GROUPs/EXPANDED CALL PICKUP - GROUPs.
i .) A call to be picked up can be that from a Station, a Trunk or an Attendant Console.
j.) When there is a restriction on the connection between an Incoming Call and the Answering Station, the restriction is applied. As a result of this, the Station cannot pick up the call.
k.) The features, though using the Telephone Numbers, cannot be used together with the same features via FCCS.
l.) Stations denied access to a Route via Route Restriction Class may not be able to pick up an Incoming Trunk Call.
m.) A specific Attendant Number cannot be assigned for these features.
n.) CALL PICKUP - GROUPs via DM and groups via LDM can coexist in a system.

22. When using the FCCS Network, the specific Attendant Number cannot be assigned to a CALL PICKUP - GROUP or an EXPANDED CALL PICKUP - GROUP, which is activated with NDM Data.
23. When programmed on LDM, a call by Automatic Wake-up [A-27] or Timed Reminder [T-62] cannot be picked up.
24. Assign CALL PICKUP - GROUP data using NDM commands in FCCS network to register the member stations in a group across multiple nodes.
25. Stations that are not belonging to any CALL PICKUP - GROUP can be a member of an EXPANDED CALL PICKUP - GROUP. These Stations can pick up a call directed to an appropriate group.
26. You can answer a call ringing on a virtual line of another station. Also, you can answer a call by picking it up from a virtual line.
27. Virtual lines can be assigned to a Call Pickup Group.

Interactions

1. CALL HOLD [C-6]: To pick up a call while engaged in a Two-party Connection, the Station can switch hook flash to place the other party on CALL HOLD [C-6] and dial the Call Pickup Code. The first party is put on CALL HOLD [C-6].
2. CALL BACK [C-1]: The Call Back Station Originator being called back cannot be picked up by a Station with the CALL PICKUP - GROUP feature. If the call back is directed to the Terminating Station, then Call Pickup may be used to answer the call. Any other Station attempting to pick up the call will result in the Station user receiving Reorder Tone.
3. OUTGOING TRUNK QUEUING [O-2]: The OUTGOING TRUNK QUEUING Station being recalled cannot be picked up by a Station with the CALL PICKUP - GROUP feature. An attempt to pick up the call will result in the Station receiving Reorder Tone.
4. CALL PICKUP - ONE TOUCH KEY [C-207] is available in combination with this feature.
5. RESTRICTION: A Station assigned as fully restricted or with Inward Restrictions cannot use the CALL PICKUP feature if assigned to a CALL PICKUP - GROUP.
6. THREE-WAY CALLING [T-2]: CALL PICKUP can be used only after a THREE-WAY CONFERENCE has been reduced to a Two-party Connection or the Station has returned to an idle state.

7. STATION MESSAGE DETAIL RECORDING (SMDR) [S-10]: If a billing after transfer over different MA-IDs is out of service (ASYDL, SYS1, Index 589, Bit 3=0), when a call is answered using the CALL PICKUP - GROUP feature, the SMDR record is not created.
   If a billing after transfer over different MA-IDs is in service (ASYDL, SYS1, Index 589, Bit 3=1), the SMDR record is created. The number of a station using this feature is output to the SMDR record as a called party number.
8. CONSULTATION HOLD - ALL CALLS [C-17] must be provided.
9. Station A will not be able to pick up Station B, if Station A is engaged in any state other than idle or a simple Two-party Connection.
10. Stations denied access to a Route via Route Restriction Class may not use CALL PICKUP - GROUP to answer restricted Incoming Trunk Calls.
11. CALL PICKUP - GROUP is not available for AUTOMATIC/MANUAL INTERCOM [A-30] or DIAL INTERCOM [D-87] Calls.
12. If more than one Station Line in a CALL PICKUP - GROUP is ringing, the individual call answered is selected by the system. The system priority is based on a circular chain arrangement with first priority given to the Station following the Station dialing the pick up Code.
13. A call ring in a recall state cannot be picked up with this feature.

Programming

STEP 1: **ANPD** - Reserve a Number Level for Feature Access and Cancel. Select Normal Service (N) from the combo box for Connection Indexes. Assign NND in accordance with a predetermined Numbering Plan.

STEP 2: **ASPA** - Assign an Access Code for the CALL PICKUP - GROUP, SID = 7 and the CALL PICKUP EXPAND, SIDA = 63 (if necessary) for Connection Status Index (CI) of "N: Normal" and "H: Hooking".

STEP 3: **ACPG** - Assign members of CALL PICKUP - GROUP. If two Stations are ringing simultaneously, the Station that was programmed first will be picked up first.

STEP 4: **ACPE** - Assign the EXPANDED CALL PICKUP - GROUP (if necessary).

STEP 5: **AKYD** - Optional. For Desktop terminal, assign a Line/Feature key as the CALL PICKUP Feature key. Assign Feature key Call Pickup - Group [12]. If necessary, assign Feature key Expanded Call Pickup - Group [115].

STEP 6: **ADSL** - Assign SN = 1 (Originating) and Feature key Call Pickup Group [12]. When answering an Incoming Call while the user is on the line, assign SN = 4 (Communication is in Progress) and Feature key Call Pickup Group [12]. This command is required for assigning the function of each Soft key.

Note: When using a Soft key on a Desktop terminal, use ADSL and ADKS for the necessary data assignments. The Status Number (SN) of the ADSL command may be "1 DT" or "4 Communication in Progress".

FCCS Programming

This data setting describes only the additional or replacement commands used to activate FCCS Services. When used as an FCCS feature, these commands replace the non-network commands (for example, AAED is replaced by AAEDN). Note that the data setting for FCCS Service varies depending on the Node.

STEP 1: **ANPDN** - This command must be assigned to the NCN.

Reserve a Number Level for Feature Access and Cancel. Select Normal Service (N) from the combo box for Connection Indexes.

Assign NND in accordance with a predetermined Plan.

STEP 2: **ASPAN** - This command must be assigned to the NCN. Assign Access Code for CALL PICKUP, CI = N, SRV = "SSC: Service Code", SID = 7 (CALL PICKUP - GROUP). If necessary, assign SRV = "SSCA: Service Code Appendix", SIDA = 63 (EXPANDED CALL PICKUP - GROUP).

CI = "N: Normal"

SRV = "SSC: Service Code"

SID = 7 (CALL PICKUP - GROUP)

STEP 3: **ACPGN** - This command must be assigned to the NCN.

Assign members of CALL PICKUP - GROUP.

UGN: User Group Number

TELN: Telephone Number in CALL PICKUP - GROUP

STEP 4: **ACPEN** - This command must be assigned to the NCN.

Assign the EXPANDED CALL PICKUP - GROUP (if necessary).

UGN: User Group Number

TELN: Telephone Number in EXPANDED CALL PICKUP - GROUP

When entering data via LDM

STEP 1: **ANPDL** - Assign 1st Digit Code (1stDC) and Number of Necessary Digits (NND) for Feature Access and Cancel. Select Normal Service (N) from the combo box for Connection Indexes.

STEP 2: **ASPAL** - Assign Access Code for CALL PICKUP, CI = "N: Normal", SRV = "SSC: Service Code", SID = 7 (CALL PICKUP - GROUP). If necessary, assign SRV = "SSCA: Service Code Appendix", SIDA = 63 (EXPANDED CALL PICKUP - GROUP).

STEP 3: **ACPGL** - Assign members of CALL PICKUP - GROUP.

UGN: User Group Number

TELN: Telephone Number of member Stations

STEP 4: **ACPEL** - Assign members of EXPANDED CALL PICKUP - GROUP (if necessary).

UGN: User Group Number

TELN: Telephone Number of member Stations

STEP 5: **AKYD** - Optional. Assign a Line/Feature key as the CALL PICKUP Feature key. KYI = 1 (Feature key), Feature key Call Pickup Group [12]. If necessary, assign Feature key Expanded Call Pickup - Group [115].

Note: When using a Soft key on a Desktop terminal, use ADSL and ADKS for necessary data assignment. At this time, Status Number (SN) of ADSL may be 1 DT or 4 Communication in Progress.

For Call Pickup - Group enhancement, see **Call Pickup Preview** (next page).

Call Pickup Preview

General Description

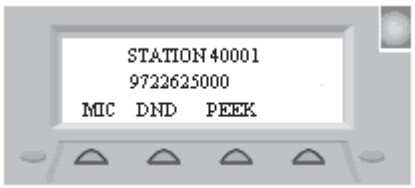
This is an enhancement to [C-7] Call Pickup - Group for Desktop terminals. This enhancement will allow calling party number/ANI or station number, up to a maximum of 15 digits, to be previewed on a Desktop terminal LCD display before answering. If name display is in service, a maximum of 16 ASCII characters can be displayed. Note that this feature is only for North America.

Operating Procedure

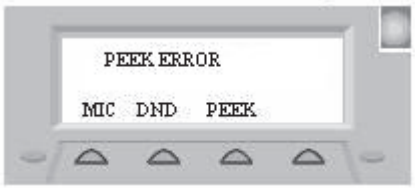
To Preview an Incoming Call

1. A Station within the CALL PICKUP - GROUP rings.
2. Press the CALL PICK UP PREVIEW Feature key.
3. LCD displays the following based on the incoming call type:
   a.) NAME
   b.) Station number/CPN/ANI

Note: A maximum of 15 digits are displayed.



Display of invalid CPN/ANI



Service Conditions

1. Call Pickup Preview is only available for Desktop terminal stations assigned in the same call pickup group. Consult the SV9500 feature and programming manual for complete details on feature [C-7] Call Pick Up Group.
2. If Call Pickup Preview is activated while two or more stations within the group are ringing, the system will view the call in the order of priority assigned to the Call Pickup Group Chain. See [C-7] Call Pickup for more detail.
3. Call Pickup Preview for Expanded call pickup - group is not available.
4. Call Pickup Preview will override the top and middle line display for approximately 5 seconds. After the 5 seconds the previous display will return. If no name is received the top display will remain as is.
5. If no stations are ringing within the Peek station call pick up group, “Peek Error” is displayed.
6. Call Pick Preview is not available for ISDN when ASFC SFI 94 is valid on the ringing station.
7. Route name assigned in ADTRL (Assignment of Dterm Display Route Data) will not be displayed in place of name.
8. On an internal station to station call, if the calling station is the same as the “Peek” station, “Peek Error” is displayed.
9. Call Pickup Preview name is not available for Guest Name Display Through PMS [G-5].
10. Call Pickup Preview is not available for PHS.
11. Call Pickup Preview will not show service type displays such as transfer, conference etc.
12. Call Pickup Preview will not display station numbers beginning with “# or *”, the display will be “PEEK ERROR”.

Interactions

1. Consult [C-7] Call Pickup -D for a complete list of interactions.
2. Call Pick Up Preview can used in an idle or consultation hold condition.

Programming

STEP 1: **ACPG** - Assign members of CALL PICKUP - GROUP. (See note)

STEP 2: **AKYD** - Assign a Line/Feature key as the CALL PICKUP Preview Feature key. KYI = 1 (Feature Key), Feature key Call Pickup Preview [166].

STEP 3: **ADSL** - Assign SKP - Soft key pattern to set, SN-status, SKN - Soft key number, FKY = 166

FCCS Programming

Additional FCCS Programming

This data setting describes only the additional commands used to activate FCCS Services. Note that the data setting for FCCS Service varies depending on the Node.

ACPGN - This command must be assigned to the NCN. Assign members of CALL PICKUP - GROUP.

UGN: User Group Number

TELN: Telephone Number in CALL PICKUP - GROUP

Note: ACPG setting is performed regardless of the LDM/NDM programming.

Take reverse procedures of above if you want to delete the settings.

C-8 Call Processing Indication

General Description

This feature provides visual indications at the ATTENDANT CONSOLE [A-3] of all calls being handled by the Attendant.

Operating Procedure

None.

Service Conditions

Each ATTENDANT CONSOLE [A-3] is provided with six dedicated Switched Loop keys. Each Loop key is associated with five lamp indicators (BUSY, RING, ATND, ANSWER, or HOLD) to display the status of the call on that Loop. The indicators may be on, OFF or flashing (400 milliseconds on and 400 milliseconds OFF).

Interactions

1. When the ATTENDANT LOOP RELEASE [A-6] feature is activated, the status of the call is removed from the ATTENDANT CONSOLE [A-3] until alerted by the AUTOMATIC RECALL [A-8] feature.
2. See DIGITAL DISPLAY - STATION [D-5], DIGITAL DISPLAY - TRUNK [D-6], and ATTENDANT LOOP RELEASE [A-6].

Programming

No programming is required.

Note: These indications are generic to the system and are enabled when the Attendant Console is programmed.

C-9 Call Queueing

General Description

This feature enables an Attendant to handle a series of Exchange Network Calls in the order of their arrival, eliminating unnecessary delays.

Operating Procedure

Press ATTENDANT CONSOLE [A-3] ANSWER key to receive calls in order of queue.

Service Conditions

1. Incoming calls arriving at the Attendant Console will light the CALL WAITING LAMP [C-13].
2. When an Incoming Call lights an INCOMING CALL IDENTIFICATION [I-2] (LDN, RECALL, etc.) Lamp, the Attendant may answer it out of the queueing sequence by pressing the indicated key.
3. Automatic Call Distribution is not used in Multi-Console operation. All Incoming Call Indications appear at each Attendant Console, so that the call can be answered by any Console. See MULTIPLE CONSOLE OPERATION [M-4] and INCOMING CALL IDENTIFICATION [I-2].
4. If an Incoming Call has been answered simultaneously by multiple Attendants, only the Attendant that operated the ANSWER key first will be connected to the call. The other Attendant's ANSWER key is non-effective when pressed.

Interactions

1. ASYD - SYS1, Index 64, Bit 1 and Bit 2. Display of Call Waiting Indication; 00 - PA Lamp is on 10 - Always.
   SYS2, Index 8 and Index 9. What kinds of calls are counted as Calls Waiting?
   SYS2, Index 14, Bit 0 to Bit 3. When CALL WAITING value exceeds a threshold value, the Call Waiting Lamp will flash.
   SYS2, Index 14, Bit 4. Will CALL WAITING - INDICATION flash for all Calls? 0/1: All Calls/Pegged Calls.
2. NIGHT CONNECTION FIXED/FLEXIBLE [N-1, 2], TRUNK ANSWER ANY STATION [T-8]: If the system is changed from Day to Night mode, calls waiting in queue will be removed and disconnected.
3. Incoming Call Indications are common to all the Attendant Consoles.
4. If the NIGHT CONNECTION - FIXED [N-1] Station is a Pilot Number in a STATION HUNTING - TERMINAL [S-9] Hunt Group, Incoming Calls will hunt to an Idle Station in the group. If all Stations in the group are busy, the call queues on the Pilot Number of the Hunt Group. When any Station becomes idle, the call is removed from queue and completed.
5. If the system is changed from Night to Day mode, Incoming Calls will be routed to the Attendant Console. A call physically ringing at the Night Station is not re-routed and is handled by the NIGHT CONNECTION - FIXED [N-1] Station.
6. If a power failure occurs, only the calls in queue that have POWER FAILURE TRANSFER [P-3] associated with their Trunk will be connected to Power Failure Stations.

Programming

No programming is required. This is generic to the system after programming the Attendant Console.

C-10 Call Transfer - Attendant

General Description

This feature permits a user, while connected to an Exchange Network Call, to signal the Attendant and have the Attendant transfer the call to another Station within the system.

Operating Procedure

Calling the Attendant

1. While engaged in a Central Office Trunk Call, press the switch hook. Receive Special Dial Tone.
2. Dial the operator Access Code.
   a.) ATTENDANT CONSOLE [A-3] RECALL Lamp flashes and buzzer sounds.
   b.) The Station receives Ringback Tone.

Answering by the Attendant

1. Refer to the Command Manual

If the Station wishes to return to the Central Office Trunk Call while the Attendant is being called

1. Press the switch hook. The RECALL lamp goes out; the buzzer stops.
2. Ringback Tone ceases. The Station returns to Central Office Trunk Call.

Note: For all Desktop terminals, use the TRANSFER key instead of the switch hook.

Service Conditions

1. This feature is available for Incoming and Outgoing Central Office Trunk Calls and Station-to-Station Calls.
2. When the ATTENDANT CONSOLE [A-3] is in the NIGHT CONNECTION [N-1, 2] mode, the Transferring Station will receive Reorder Tone if ATTENDANT NIGHT TRANSFER [A-19] is not provided.
3. The CALL TRANSFER - ATTENDANT feature allows a user, while participating in a Two-party Connection (a Two-party Connection can be comprised of two Stations or a Station and a Trunk), to call the Attendant so that the Attendant may transfer the call or provide other assistance as required.
4. If all Registers are busy and the user switch hook flashes, the flash is ignored by the system and the parties remain connected.

Interactions

1. When the HOTLINE [H-1] set is equipped with a dial, the user switch hook flashes to get Special Dial Tone and dials “0”. Ringback Tone is provided and the Attendant is notified of the call.
2. CALL TRANSFER - ATTENDANT: This feature can be used by a fully restricted Station. The Attendant can transfer a fully restricted Station to another Station only.
3. This feature is not operable when NIGHT CONNECTION - FIXED [N-1] or FLEXIBLE [N-2] is activated. To process a transfer, use CALL TRANSFER - ALL CALLS [C-11] feature. Reorder Tone is returned when CALL TRANSFER - ATTENDANT is attempted.
4. If the SPLITTING [S-4] and ATTENDANT LOCKOUT [A-5] features are activated, this feature can be activated.
5. Refer to BLIND TRANSFER TO ATTENDANT [B-17] for blind transfer from Station to Attendant Console.

6. A HOTLINE [H-1] Station may transfer a call to the Attendant Console.
7. A CALL TRANSFER to a Station with CALL FORWARDING [C-2, 3, 5] set will forward according to the conditions of the feature set, unless the Station is set to CALL FORWARD - ALL CALLS [C-5] to the Attendant Console. CALL FORWARDING - OVERRIDE [C-26] supersedes CALL FORWARDING - ALL CALLS [C-5].
8. A CALL TRANSFER to a Station in a STATION HUNTING [S-7, 8, 9] Group will hunt according to the conditions of the STATION HUNTING [S-7, 8, 9] Group.
9. A CALL TRANSFER to a Dterm that has DO NOT DISTURB [D-21D] set will receive Reorder Tone.

Programming

STEP 1: **ASYD** - SYS2, Index 1, Bit 0 and Bit 1. Assign data "11" to allow CONSULTATION HOLD - ALL CALLS [C-17] for all types of connections.

STEP 2: **ATNR** - Allow Tenant-to-Tenant Connection for Inter and Intra-Tenant Connections. Assign Tenant Restriction Index (TRI) 0 and 4 on an Inter and Intra-Tenant basis.

C-11 Call Transfer - All Calls

General Description

This feature permits a user to transfer Incoming or Outgoing Central Office and Intra-office Calls to another Station within the system without Attendant assistance.

The user can set Camp-On function to the busy destination while transferring an Incoming Call (Camp-On by Station). With this feature, the Trunk Party is called back directly, as the destination becomes idle.

Operating Procedure

To transfer a call in progress using switch hook flash

1. While connected with the first party, press the switch hook. Receive Special Dial Tone.
2. Dial the third party. Receive Ringback Tone.
3. At this point, the user can either:
   a.) Hang up before the third party answers. The first and third parties will be connected when third party answers.
   b.) Wait for third party to answer and announce the transfer while keeping the first party in CONSULTATION HOLD - ALL CALLS [C-17] condition. When the user hangs up, the first and third parties will be automatically connected.

To set the Camp-On by Station function to the Busy Station

- Manual Operation

1. While connected with the first party, press the switch hook. Receive Special Dial Tone. At this time, the Calling Trunk Party hears music on hold.
2. Dial the third party, and the third party is busy. Receive Busy Tone.
3. Press the switch hook. Receive Special Dial Tone, and then dial the Access Code of CALL BACK. Receive Service Set Tone.
4. Replace the handset. The Station to be set Camp-On by Station (destination) hears Warning Tone. At this time, the Calling Trunk Party hears Ringback Tone.
5. The Target Station starts to ring when the target Station lifts the handset.
6. The Target Station (Called Party) can call back to the Trunk Party directly by answering.

- Automatic Operation

1. While connected with the first party, press the switch hook. Receive Special Dial Tone. At this time, the Calling Trunk Party hears music on hold.
2. Dial the third party, and the third party is busy. Receive Busy Tone.
3. Replace the handset. The Station to be set Camp-On by Station hears Warning Tone. At this time, the Trunk Party hears Ringback Tone.
4. When the Target Station becomes available, the Target Station starts to ring.
5. The Target Station can call back to the Trunk Party directly as the Station answers.

To answer the Camp-On by Station Call from the Busy Station

1. After receiving Warning Tone, the third party in the condition that the communication is in progress can answer the call by pressing the ANSWER key. At this moment, the call connected by the third Station is placed on hold.

<The following procedures are applied for Desktop terminal only.>

To transfer a call in progress using the TRANSFER key

1. While connected with the first party, press the TRANSFER key. Receive Special Dial Tone.
2. Dial the third party. Receive Ringback Tone.
3. At this point, the user can either wait and announce the call or hang up before the transfer is completed. The LCD display for an announced transfer will be:

Transferred Party



The LCD for an unannounced transfer will be:

Transferring Party Transferred Party



Note: When the Telephone Number of the Transferring/Transferred Station exceeds eight digits, the last eight digits are displayed. However, a maximum of 16 digit Transferred Telephone Numbers can be displayed on the middle line of the LCD, depending on System Data.

To set the Camp-On by Station function to the Busy Station

- Manual Operation

1. While connected with the first party, press the TRANSFER key. Receive Special Dial Tone. The Calling Trunk Party hears music on hold. The Desktop terminal LCD of the Mediate Party (Operating Station) displays:



2. Dial the third party, and the third party is busy. Receive Busy Tone. The LCD of the Operating Station displays:

The Extension Number of the third party



3. Method of operation:
   a.) When the Access Code is used, press the FLASH key. Receive Special Dial Tone and no indication on the LCD, then dial the Access Code of CALL BACK. Receive Service Set Tone.



   b.) When the Feature key/Soft key is used, press the CALL BACK key. Receive Service Set Tone. The LCD of the operating Station displays as mentioned above in the way of using the Access Code.
4. Replace the handset. The Station to be set Camp-On by Station (destination) hears Warning Tone. The LCD of the operating Station displays:



The Calling Trunk Party hears Ringback Tone. The LCD of the target party displays:



(TRK): Trunk kind or the Calling Number

5. The Target Station starts to ring when the Target Station lifts the handset. The LCD of the Target Party displays:



(TRK): Trunk kind or the Calling Number

6. The Target Station (Called Party) can call back to the Trunk Party directly by answering the ringing. The LCD of the Target Party displays:

ELAPSED TIME for the communication



(TRK): Trunk kind or the Calling Number

- Automatic Operation

1. While connected with the first party, press the switch hook. Receive Special Dial Tone. The Calling Trunk Party hears music on hold. The LCD of the operating Station displays:



2. Dial the third party, and the third party is busy. Receive Busy Tone. The LCD of the Operating Station displays:

The Extension Number of the third party



3. Replace the handset. The Station to be set Camp-On by Station hears Warning Tone. The LCD of the Operating Party displays:



The LCD Display is changed after the Trunk receives Ringback Tone:



The LCD Display of the Target Party:



(TRK): Trunk kind or the Calling Number

The Trunk Party hears Ringback Tone.

4. When the Target Station becomes available, the Target Station starts to ring. The LCD of the Target Party displays:



(TRK): Trunk kind or the Calling Number

5. The Target Station can call back to the Trunk Party directly as the Station answers the ringing. The LCD of the Target Party displays:

ELAPSED TIME for the communication



(TRK): Trunk kind or the Calling Number

To answer the Camp-On by Station Call from the Busy Station

1. After receiving Warning Tone, the third party can answer the call by pressing the ANSWER key. The original call connected by the third Station is placed on hold.

Service Conditions

1. The Station performing the CALL TRANSFER - ALL CALLS can drop off the connection before the Called Party answers. If the Called Station does not answer within the predetermined time, Ringing Signal is returned to the Transferring Station. See VARIABLE TIMING PARAMETERS [V-1] for other selectable timing choices.
2. In the case of Outgoing Calls (No Answer Supervision), this feature may be activated only after 18 seconds have elapsed since the last digit dialed.
3. Outgoing Calls with Answer Supervision may be transferred immediately after call is answered (no time delay).
4. If all Registers are busy and the user switch hook flashes, the flash is ignored by the system, and the parties remain connected.
5. When TRUNK-TO-TRUNK CONNECTION [T-10] is provided with the system, an Incoming or Outgoing Network or TIE Line Call may be transferred to an Outgoing Network Trunk, if Release Signal is provided from distant end. The user who transfers the call gives line control to the two remaining parties upon hanging up. If the Release Signal is not provided, Call Transfer from Trunk-to-Trunk will not function.
6. The THREE-WAY CALLING [T-2] feature can comprise not more than three parties on one connection. The three parties connected can be as follows:
   a.) three Stations
   b.) two Stations and one Trunk
   c.) one Station and two Trunks.
7. The maximum number of simultaneous THREE-WAY CALLING [T-2] Connections per system is dependent on the quantity of CFT-function Circuits equipped (traffic engineered).
8. Reorder Tone is returned to the user when the following actions occur:
   a.) The user's line is restricted by Outward or Miscellaneous Trunk Restrictions by the Class of Service,
   b.) The user dials the number of an unequipped line or termination is restricted (Route Restrictions),
   c.) The user dials an invalid first digit.
9. Transferring a call in progress by switch hook flash, if you hang up the call while receiving Special Dial Tone within four minutes, Busy Tone will be given to the call being held. And no call is transferred to any Station.
10. The Camp-On by Station cannot be set to the Station via CCIS.
11. The Camp-On by Station cannot be set in the following cases:
    a.) When the Calling Station:
        sets or is set CALL BACK [C-1].
        holds a call.
        Service Feature Restriction Class is not allowed.
        uses Sub Line.
    b.) When the Target Station:
        is in LINE LOCKOUT [L-3] state.
        is in male-busy state.
        sets or is set CALL BACK [C-1].
        holds a call.
        has already been set another Camp-On.
        answers another Camp-On with the ANSWER key.

is in Consultation Hold.
is in Exclusive Hold.
is set Fixed Connection - Nailed Down Connection.
sets Do Not Disturb -D [D-11D]
Incoming Route Restriction Class is not allowed.

12. The Camp-On by Station can be set to one Station only once.
13. The Camp-On by Station is restricted when the Target Station is in the Three-party Connection.
14. The Camp-On by Station is not available for the Target Station in LINE LOCKOUT [L-3] condition.
15. The Camp-On by Station is not available in the following cases:
    a.) When the Transferred Station is dialing the number.
    b.) When a call is terminating to the Transferred Station (the Station is ringing).
    c.) When the Transferred Station hears Busy Tone or Reorder Tone.
    d.) When the Transferred Station hears Special Dial Tone.
16. The Station allowed to set the Camp-On by Station (operating Station) is only the Dterm and the Analog Station.
17. When using Camp-On by Station, 1 Dial Hooking cannot be used.
18. The Station to set the Camp-On by Station (Transferred Station) is the Analog Station, the Desktop terminal and the ISDN terminal.
19. When the Camp-On by Station is set to the Desktop terminal:
    a.) The ANSWER key of the Desktop terminal lights when Warning Tone is received.
    b.) At the Station to be set the Camp-On by Station, the LCD displays the Service Set indication (CAMP) and the Trunk Kind or the Calling Number if the Caller ID notification is received.
20. The following conditions should be considered when this service is set by manual operation (see Operating Procedure) because the service becomes effective when the operating Station is released from the connection:
    a.) When the operating Station is not released within the predetermined time period after receiving Service Set Tone, the service which has been set will be ineffective. And the operating user hears Busy Tone.
    b.) When the Transferred Destination Station becomes available before the Transferring Station is released from the connection after the service was set (after the operating user hears Service Set Tone), the Trunk Incoming Call is connected with the Destination Station after the operating Station (Mediate Party) replaces the handset.
    c.) When the Transferred Destination Station becomes LINE LOCKOUT condition or the destination becomes make busy before the Transferring Station is released from the connection after the service was set (after the operating user hears Service Set Tone), the Called Station (Mediate Party) is recalled even though the Station is disconnected.
21. The following are the conditions when the Destination Station does not answer within the time period of the No Answer Timer:
    a.) When the timer time-out occurs due to that the Destination Station not being answered within the predetermined time period after the operating Station is disconnected, the operating Station is called back.
    b.) When the No Answer Timer expires and the Destination Station has not answered within the predetermined time period, the call management is decided by the System Data (recall to the operating Station or CALL FORWARDING - DON'T ANSWER activation).
22. The Service Conditions for Warning Tone are as follows:
    a.) The Destination Station receives Call Waiting Tone when Camp-On by Station has been set.

b.) The amount of time between tones is decided on a system basis.

23. A call can be transferred to a Station in the Remote Node via FCCS link.
24. Camp-On by Station can be set to the Station in the Self Office or the Self Node in the FCCS Network.
25. When PS becomes Out of Zone while holding a call (switch hook flash), it will be recalled. However, if it remains in Out of Zone state, the following will operate.
    If CALL FORWARDING - PS INCOMING INCOMPLETE [C-155] is set, the aborted call will be transferred to the pre programmed destination.
    If AUTOMATIC ANNOUNCEMENT - CALLED PS OUT OF ZONE [A-129] is set, the system automatically provides an announcement.
26. Camp-On by Station is unavailable for PS.
27. When a Control Station (second party) transfers a Station or Trunk call to another Station (third party) and the Control Station goes on-hook during ringing, if the third party's line is busy, the Control Station will be recalled right away. If the third party does not answer the call in time (30 seconds), the Control Station will be recalled after the time expires.

    Note: When the Control Station does not answer the call recalled, operations are different depending on the third party's state.
    If the third party's line is busy and the Control Station is recalled, but it does not answer:
    - Ringing the Control Station is stopped, and then the line held by the Control Station is disconnected regardless of whether the held line is Trunk or Station line. If the Control Station is a PS Station, the PS Station is kept ringing.

    If the third party does not answer the call and the Control Station is recalled but it does not answer:
    - The Control Station is kept ringing regardless of whether the held line is Trunk or Station line.

28. When an attendant console places a station or a trunk call on hold and transfers it to another station then the transferring destination releases the connection after answering the transferred call, the attendant console hears a silence.
    When an attendant console places a station or a trunk call on hold and transfers it to C.O. line party then the transferring destination releases the connection after answering the transferred call, the attendant console hears a Reorder Tone (ROT).
29. When a SIP Handler Controlled SIP terminal is talking with the transferring destination and the transferring destination is released, the SIP Handler Controlled SIP terminal returns to the state putting the previous call on Call Hold (the Call Hold state is not canceled automatically). An operation to cancel the Call Hold state is needed to talk with the previous call.

Interactions

1. This feature is denied whenever the Transferring Station is engaged in any state other than a simple Two-party Connection.
2. See TRUNK-TO-TRUNK CONNECTION [T-10] for additional programming if the call is to be transferred off-network.
3. If a Station is in an established THREE-WAY CALLING [T-2] Connection, that Station cannot activate any features. If a Station switch hook flashes to activate a feature, the flash is ignored and the Three-way Connection remains established.
4. Features that cannot be activated if a party is involved in a CONSULTATION HOLD [C-17] or in an established THREE-WAY CONFERENCE [T-2] Connection are:

ATTENDANT CAMP-ON WITH TONE INDICATION	[A-1]
ATTENDANT OVERRIDE	[A-7]
BOSS - SECRETARY OVERRIDE -D	[B-5D]
BUSY VERIFICATION	[B-3]
CALL BACK	[C-1]
CALL FORWARDING (Activate or Cancel)	[C-2, 3, 5]
CALL PICKUP - DIRECT	[C-30]
CALL WAITING - TERMINATING	[C-12]
EXECUTIVE RIGHT-OF-WAY	[E-1]
MESSAGE REMINDER	[M-19]
OUTGOING TRUNK QUEUING Cancel	[O-2]
SPEED CALLING - STATION Entry	[S-21]
TRUNK ANSWER FROM ANY STATION (TAS)	[T-8]

5. Upon establishing a CONSULTATION HOLD [C-17] Connection and dialing a third party, if the Station user receives busy or Reorder Tone instead of Ringback Tone, the Station user flashes once to disconnect the Tone and reconnect the second party.
6. Upon establishing a CONSULTATION HOLD [C-17] Connection and dialing a MISCELLANEOUS TRUNK ACCESS [M-2] Code, if no Dial Tone is heard (after the Trunk is seized), if a mistake is made in dialing, or if an Intercept Announcement or a Permanent Tone is heard, the Station user must go on-hook and wait for Ringback Tone.
7. The Station user, who establishes CONSULTATION HOLD [C-17], will be reconnected to the party in soft hold when the third party disconnects.
8. A Call Transfer to a Station with CALL FORWARDING [C-2, 3, 5] set will forward according to the conditions of the feature set.
9. A Call Transfer to a Station in a STATION HUNTING [S-7, 8, 9] Group will hunt according to the conditions of the STATION HUNTING [S-7, 8, 9] Group.
10. A Call Transfer to a Desktop terminal that has DO NOT DISTURB [D-21D] set will receive Reorder Tone.
11. A Call Transfer Call may be answered by CALL PICKUP - GROUP/DIRECT [C-7, 30].
12. The Camp-On by Station is not available when DO NOT DISTURB -D [D-11D] has been set to the Target Station. **Note 1**
13. The Camp-On by Station is not available when the Target Station uses PRIVACY RELEASE [P-18]. **Note 1**
14. When the Camp-On by Station is set by automatic operation and the Destination Station is the UCD Group member, the Camp-On by Station feature takes priority over BLIND TRANSFER TO BUSY UCD if it is available.

15. The TRANSFER key on the Desktop terminal is used in place of the switch hook.

Note 1: The operating Station hears Busy Tone when the destination is in the above condition.

Programming

STEP 1: **ASYD** - SYS1, Index 69, Bit 0. Return Transferred Call to origin of Call Transfer after the Recall Timer elapses (only applicable to Blind Transfers). SYS1, Index 69, Bit 2. Short Tone on recall of CALL TRANSFER - ALL CALLS. Ringback Tone is briefly heard by the person answering the recall (only applicable to Blind Transfers).

SYS1, Index 140 if ASFC SFI 103 = 0. Index 247 if ASFC SFI 103 = 1. Assign the Recall Timer for a Station-to-Station CALL TRANSFER - ALL CALLS. For 30 seconds, assign data 00H. (TC 3 = 2 seconds, TC 7 = 8 seconds). RAM Data 3FH for 30 seconds.

SYS2, Index 1, Bit 0 and Bit 1. Assign data "11" to allow CONSULTATION HOLD.

SYS1, Index 156 and Index 157. How long before SHF is possible.

SYS2, Index 0, Bit 0=1 (Allow the restriction for Call Back Service)

SYS2, Index 5, Bit 2. Camp-On by Station is set automatically after the operating Station is released from the connection after encountering the busy Target Station.

0: No (Recall to the mediate Station)

1: Yes (Camp-On Service is set)

SYS2, Index 10, Bit 7. Number of times that the Warning Tone (Call Waiting Tone) is sent to the Target Station.

0/1:Once/At Intervals

Note: When "1" is assigned, the interval of this Tone is assigned in SYS1, Index 152.

SYS1, Index 248, Bit 6. When the Destination Station has been set Call Forwarding - Don't Answer.

0/1: Recall to the Station/Call Forwarding - Don't Answer is activated

STEP 2: **ASTD** - Assignment of State Translation Data

STM: 2
STS: 2 (Outgoing call to a C.O. Line or a Tie Line after holding the COT)
ST: 0/1 (Restricted/Allowed)

STM: 2
STS: 3 (Outgoing call to a C.O. Line after holding a station or a Tie Line)
ST: 0/1 (Restricted/Allowed)

STEP 3: **ANPD** - Assign the first Number Level for the Feature Access Code.

STEP 4: **ASPA** - Assign the Feature Access Code.

TN: Tenant Number
ACC: Access Code for CALL BACK
CI: "H: Hooking" (Switch hook Flash)
SRV: "SSC: Service Code"
SID: 3 (Call Back; Entry)

STEP 5: **AKYD** - Assign CALL BACK Feature key.

KYI: 1 (Feature Key)
FKY: 5 (Call Back)

ADSL - Assign CALL BACK Soft Key.

FKY: 5 (Call Back)
SN: 8 (Encounters the busy Station while holding an original call)

STEP 6: **ASFC** - Assign the Service Feature Restriction Class on Camp-On by Station.
SFI 174 (Camp-On by Station) RES 1 (Allowed)

STEP 7: **ATNR** - Allow Tenant-to-Tenant Connection for Inter and Intra-tenant Connections. Assign Tenant Restriction Index (TRI) 0 and 4 on an Inter and Intra-tenant basis.

STEP 8: **ARSC** - The station receiving the call must be assigned a Route Restriction Class (RSC) that will allow the station to be connected to the trunk, if a trunk is involved.

FCCS Programming

This data setting describes only the additional or replacement commands used to activate FCCS Services. When used as an FCCS feature, these commands replace the non-network commands (for example, AAED is replaced by AAEDN). Note that the data setting for FCCS Service varies depending on the Node.

Note: When using ASYDL or ASYDN to program an FCCS feature, the basic programming of ASYD is required in addition to ASYDL or ASYDN.

STEP 1: **ALRTN** - Assign this command to connect the Logical Route to the Physical Route.

STEP 2: **ARSCN** - The Station receiving the call must be assigned a Route Restriction Class (RSC) that will allow the Station to be connected to the Trunk, if a Trunk is involved.

C-12 Call Waiting - Terminating

General Description

This feature enables a busy Station to receive a second Incoming Call. A Camp-On Indication Tone is sent to the busy Station, the user can use a switch hook flash to answer the second call. A switch hook flash may be used to alternate between the two calls.

Operating Procedure

To activate from a Station

1. Dial the desired Telephone Number.
2. If the Called Station is busy, CALL WAITING - TERMINATING is automatically set. The Calling Station receives Call Waiting Ringback Tone. Call Waiting Tone (one beep) is sent to the busy Station.

To activate on a Direct-In Trunk Call

1. The System receives the Called Telephone Number.
2. If the Called Station is busy, CALL WAITING - TERMINATING is automatically set. The Calling Party receives Call Waiting Ringback Tone. Call Waiting Tone (two beeps) is sent to the busy Station.

To answer

1. Call Waiting Tone is heard during the call in progress.
2. A switch hook flash or pressing the ANSWER key on the Desktop terminal will hold the existing call.
3. CALL WAITING - TERMINATING Call is automatically connected.
4. Another switch hook flash or pressing the ANSWER key on the Desktop terminal will return to the original call and hold the second call.

OR

1. Call Waiting Tone is heard during the call in progress.
2. The Called Station hangs up. Priority ringing is sent.
   Priority ringing = 0.4 seconds ON 0.2 seconds OFF0.8 seconds ON
   0.2 seconds OFF0.4 seconds ON0.4 seconds OFF
   Lift the handset to answer.

Service Conditions

1. Call Waiting Tone is one beep for a Station Call and two beeps for a Trunk Call. A beep is 200 milliseconds ON/OFF.
2. Periodic Call Waiting Tone can be eliminated by System Data programming on a per Tenant basis.
3. CALL WAITING - TERMINATING must be allowed in the call originator's CLASS OF SERVICE - INDIVIDUAL [C-15] programming to be activated.
4. There is no limit to the number of Stations that can receive this feature simultaneously.
5. CALL WAITING - TERMINATING is denied when the Called Station is connected to the ATTENDANT CONSOLE [A-3].
6. CALL WAITING - TERMINATING can be set only when the Called Station is busy on a Two-party Call with another Station or Trunk.
7. Call Waiting Ringback Tone changes to normal Ringback Tone for the Calling Party when the Called Station goes on-hook.
8. CALL WAITING - TERMINATING cannot be activated while in CONSULTATION HOLD [C-17] mode.

9. CALL WAITING - TERMINATING cannot be used in conjunction with DIAL ACCESS TO ATTENDANT [D-2].
10. CALL WAITING - TERMINATING cannot be activated if the Target Station has accessed, entered, or is engaged in any of the following features or modes:

ATTENDANT CAMP-ON WITH TONE INDICATION	[A-1]
ATTENDANT OVERRIDE	[A-7]
BOSS - SECRETARY OVERRIDE -D	[B-5D]
BUSY VERIFICATION	[B-3]
CALL BACK	[C-1]
CALL HOLD	[C-6]
CALL TRANSFER	[C-10, C-11]
CALL WAITING - ORIGINATING	[C-31]
CONSULTATION HOLD - ALL CALLS	[C-17]
DATA LINE SECURITY	[D-12]
DATA PRIVACY ON DEMAND	[D-17]
DIAL ACCESS TO ATTENDANT	[D-2]
EXCLUSIVE HOLD -D	[E-4D]
EXECUTIVE RIGHT-OF-WAY	[E-1]
LINE LOCKOUT	[L-3]
NON-EXCLUSIVE HOLD -D	[N-7D]
OUTGOING TRUNK QUEUING	[O-2]
PRIVACY -D	[P-14D]
PRIVACY RELEASE	[P-18]
THREE-WAY CALLING	[T-2]
VOICE CALL	[V-2]

11. The following features take precedence over CALL WAITING - TERMINATING:

CALL FORWARDING - ALL CALLS	[C-5]
CALL FORWARDING - BUSY LINE	[C-3]
STATION HUNTING - CIRCULAR	[S-7]
STATION HUNTING - SECRETARIAL	[S-8]
STATION HUNTING - TERMINAL	[S-9]

12. When Called Station activates CALL FORWARDING- BUSY LINE [C-2], and the call forwarding destination is busy:

- When call forwarding destination is a station (including stations in FCCS network), ASYD, SYS1, Index 69, Bit 5 determines whether to set CALL WAITING - TERMINATING to the Forwarded-from Station, or to the Forwarded-to Station.
- When the call forwarding destination is in outside of the self office, CALL WAITING - TERMINATING is restricted and does not work.

Interactions

1. This feature is allowed whenever the Destination Telephone Set is in any state other than:
   a.) Engaged in Three-way Connection,
   b.) A PRIVACY feature is activated,
   c.) Connected to a Route programmed to have Line/Trunk, Privacy, CDN 44:PRV,
   d.) Listening to a Tone (Dial Tone, Reorder Tone, Service Set Tone, Busy Tone, Ringback Tone).
   e.) Connected to the Attendant Console.

2. To answer a CALL WAITING - TERMINATING call, SYS2, Index 0, Bit 0 and Bit 1, CONSULTATION HOLD [C-17]. Assign data "11" to allow CONSULTATION HOLD.
3. To answer a CALL WAITING - TERMINATING Call from a Dterm, press the ANSWER key. The Answering Party can alternate between the calls by pressing the ANSWER key.
4. Enabling ATTENDANT LOOP RELEASE [A-6] (SYS2, Index 6, Bit 7), a call Camped-On to a Station via the Attendant Console will wait at that Station for the time set in Index 145. When the preset time has elapsed, the call will return to the Attendant Console.
5. If ATTENDANT LOOP RELEASE [A-6] is not provided and CALL FORWARDING - DON'T ANSWER [C-3] via the Attendant Console is provided, a call extended to a Station via the Attendant Console will wait at the Station for the time set in Index 141 and will then be forwarded to the Destination Station for the time set in Index 145. Index 145 is the total time before recall. When that time expires, the call will return to the Attendant Console.
6. When CALL FORWARDING [C-2, 3, 5] is in effect, the forwarding operation occurs before CALL WAITING - TERMINATING is allowed, unless the CALL FORWARDING - OVERRIDE [C-26] feature is applicable. If CALL FORWARDING - OVERRIDE [C-26] is used, the call will wait. After a call has been forwarded, the call may wait at the new Station if CALL WAITING - TERMINATING is allowed.
7. BUSY VERIFICATION [B-3]: This feature is denied to a Station that is in a CALL WAITING - TERMINATING state.
8. EXECUTIVE RIGHT-OF-WAY [E-1]: This feature is denied to a Two-party Connection that is in a CALL WAITING-TERMINATING state.
9. STATION HUNTING [S-7, 8, 9]: This feature takes precedence over CALL WAITING - TERMINATING.
10. LINE LOCKOUT [L-3]: CALL WAITING - TERMINATING is not allowed to a Station in LINE LOCKOUT [L-3]
11. OUTGOING TRUNK QUEUING [O-2]: CALL WAITING - TERMINATING is not allowed on any Station involved in THREE-WAY CALLING [T-2].
12. When using this feature against a Station that is engaged in Tandem connection other than CCIS-CCIS, SFI 4 and SFI 11 require to be set to 1 for SFC 0 by using ASFC/ASFCN.
13. This feature cannot be set from SIP Handler Controlled SIP terminals. Also this feature cannot be set from different terminals to SIP Handler Controlled SIP terminals.

Programming

STEP 1: **ASYD** - SYS1, Index 64, Bit 6. Leave as data "0".

SYS1, Index 145. Assign the Recall Timer for a call extended by the Attendant Console that is not answered. Normally assign data 00H for 30 seconds. (RAM Data 9F for 30 seconds).

SYS2, Index 10, Bit 6. Assign 1H for automatic. Bit 7 is used for CALL WAITING TONE INDICATION. 0/1: Once/Repeat-per setting of SYS1, Index 152.

SYS1, Index 69, Bit 5. CALL WAITING - TERMINATING 0/1 = Set to Forwarded-from Station/Set to Forwarded-to Station

STEP 2: **ASFC** - Assign a Service Feature Index (SFI) of 11 (Data Line Privacy = 0/Attendant Camp-On = 1) to the Station. SFI 5 [Call Waiting - Originating (Calling)] and 6 [Call Waiting - Originating (Called)], RES = 1.

STEP 3: **ARTD** - For CALL WAITING - TERMINATING to operate automatically for TIE Lines and DID Routes, assign CDN 46:CW, Call Waiting, as data "1".

Note: To change the data for existing route (RT), initialize the circuit card.

FCCS Programming

This data setting describes only the additional or replacement commands used to activate FCCS Services. When used as an FCCS feature, these commands replace the non-network commands (for example, AAED is replaced by AAEDN). Note that the data setting for FCCS Service varies depending on the Node.

STEP 1: **ALRTN** - Assign this command to connect the Logical Route to the Physical Route.

STEP 2: **ARTDN** - For CALL WAITING - TERMINATING to operate automatically for TIE Lines and DID Routes, assign CDN 46:CW, Call Waiting, as data "1".

Note: To change the data for existing route (RT), initialize the circuit card.

C-13 Call Waiting Lamp

General Description

This feature provides a visual indication to the Attendant when one or more calls are waiting to be answered.

Operating Procedure

Refer to the Command Manual.

Service Conditions

1. All Incoming Calls to the ATTENDANT CONSOLE [A-3] that have to be answered will signal the Attendant via the CALL WAITING LAMP.
2. The CALL WAITING LAMP can be set to flash at 60 IPM only after "x" number of calls are waiting. "x" can be anywhere from 1 to 15 calls waiting per TENANT SERVICE [T-12].
3. In the case of MULTIPLE CONSOLE OPERATION [M-4], the CALL WAITING LAMP will light on all Consoles until one of the positions has processed the calls.
4. Where TENANT SERVICE [T-12] is provided, the CALL WAITING LAMP will light only on the Consoles of the Called Tenant.
5. In addition to the CALL WAITING LAMP indication, each Console is provided with a two-digit display that shows the number of calls waiting.

Interactions

1. The CALL WAITING LAMP lights if any of the following Incoming Calls are waiting to be answered at the ATTENDANT CONSOLE [A-3]:

LDN	Listed Directory Number
ATND	Attendant Dial "0" Calls, CALL FORWARDING - ALL CALLS [C-5]
RECALL	Attendant Recall Calls
FX	Foreign Exchange
WATS	Wide Area Telephone Service
TIE	TIE Line Calls
BUSY	Call Forwarding - Busy Line to Attendant
NANS	Call Forwarding - Don't Answer to Attendant
TF	Inter-Position Transfer
ICPT	Call Forwarding - Intercept
CCSA	Common Channel Switching Arrangement

2. In addition to the CALL WAITING LAMP, an audible buzzer is sounded whenever a call is waiting and the Attendant is available to answer calls. The buzzer volume can be adjusted and silenced by operation of a Volume Control key. If the Attendant is on a Loop, a buzzer can be activated.
3. An indication of the type of calls waiting is given by the INCOMING CALL IDENTIFICATION [I-2] feature.
4. When TENANT SERVICE [T-12] is used, the CALL WAITING LAMP will flash only on the Consoles assigned to the Called Tenant. The Tenant Console is assigned in ATRK. The Route of the Tenants of the individual Trunks are assigned in ATRK as well.

Programming

STEP 1: **ASYD** - SYS1, Index 64, Bit 1 and Bit 2. When will the number of calls waiting appear on the Attendant Console? 00: When Position Available Lamp is ON (Attendant Console is not actively processing a call); 01: When Busy Lamp Field is OFF; 10: Continuously.

SYS2, Index 8 and Index 9. These bits flag types of Incoming Calls to the Attendant Console to be counted as calls waiting. This is assigned on a per Tenant basis.

Index 8:

Bit 0 - Is an LDN Call counted as a Call Waiting? 0/1: No/Yes.

Bit 1 - Is an Incoming Station Call counted as a Call Waiting? 0/1: No/Yes.

Bit 2 - Is an Attendant Console Recall counted as a Call Waiting? 0/1: No/Yes.

Bit 3 - Is an FX Call counted as a Call Waiting? 0/1: No/Yes.

Bit 4 - Is a WATS Call counted as a Call Waiting? 0/1: No/Yes.

Bit 5 - Is a CCSA (DID) Call counted as a Call Waiting? 0/1: No/Yes.

Bit 6 - Is a TIE Line Call counted as a Call Waiting? 0/1: No/Yes.

Bit 7 - Is a CALL FORWARDING - BUSY LINE [C-2] Call counted as a Call Waiting? 0/1: No/Yes.

Index 9:

Bit 0 - Is a CALL FORWARDING - DON'T ANSWER [C-3] Call counted as a Call Waiting? 0/1: No/Yes.

Bit 1 - Is a CALL FORWARDING - INTERCEPT [C-25, 25D] Call counted as a Call Waiting? 0/1: No/Yes.

Bit 2 - Will this key (type of call is assigned in AAKP) be counted as a Call Waiting? 0/1: No/Yes.

Bit 3 - Is an INTERPOSITION TRANSFER [I-5] Call counted as a Call Waiting? 0/1: No/Yes.

Note: The types of calls listed above are the default types of Incoming Calls. SYS2, Index 8, Bit 6 to Bit 7, and Index 9, Bit 0 to Bit 3 only turn ON or OFF the ability to count an Incoming Call to a specific key as a Call Waiting. A different type of Incoming Call may be assigned via AAKP.

SYS2, Index 14, Bit 0 to Bit 3. When CALL WAITING value exceeds a threshold value, the Call Waiting Lamp will flash.

SYS2, Index 14, Bit 4. Will CALL WAITING - INDICATION flash for All Calls? 0/1: All Calls/ Pegged Calls.

Note: This data is required only when assigning "x" number of calls waiting.

STEP 2: **ARTD** - Flag the type of Route in CDN 6:TCL, Trunk Class.

Note: To change the data for existing route (RT), initialize the circuit card.

STEP 3: **AAKP** - This command may be used to change the meaning of the top six Call Identification keys of the Attendant Console. (See SYS2, Index 8 and Index 9). The following functions may be assigned to L1 - L6:

Function:

(1) CAS
(2) Off-Hook Alarm
(3) Priority Call 1
(4) Priority Call 2
(5) Priority Call 3

C-14 CCSA Access

General Description

This feature enables connection to a Common Channel Switching Arrangement (CCSA) Network. CCSA Networks provide customers with the use of a completely private inter-facility dial system. They use individual facilities for dedicated Access Lines and Trunks that terminate in Common Control Switching Equipment and in various types of Telephone Equipment at Customer Locations. **Note 1**

Note 1: This feature is available in North America only.

Operating Procedure

To place an Outgoing CCSA Call

1. Lift the handset. Receive Dial Tone.
2. Dial the CCSA ACCESS Code. Receive Second Dial Tone.
3. Dial the desired number.
4. Wait for the call to be processed.

To answer an Incoming CCSA Call

1. The Attendant presses either the CCSA or ANSWER key.
2. The call is extended to the desired Station in normal fashion.

Service Conditions

1. CCSA Trunks are physically the same as those used for TIE Trunks. The CCSA Trunks may be pushbutton or rotary in either direction.
2. Careful consideration should be given to System Numbering Plans before assigning CCSA ACCESS Codes to avoid numbering conflicts.
3. CCSA ACCESS Codes can be either one or two digits.
4. Pad Control can be used on a CCSA Call. Refer to TIE LINE CONNECTION WITH PAD CONTROL [T-5].
5. LEAST COST ROUTING - 3/6-DIGIT [L-5] may access a CCSA Trunk Group for Off-net Calling only.
6. Tandemming through on a CCSA Network requires an Attendant and TRUNK-TO-TRUNK CONNECTION [T-10].

Programming

ARTD - For CCSA TIE Line Routes, assign CDN 6:TCL, TRUNK CLASS as a CCSA Route:

RT:1

1-OSGS	:2/7	2-ONSG	:3	3-ISGS	:2/7	4-INSG	:3	5-TF	:3
6-TCL	:5	7-L/T	:1	8-RLP	:2	15-LSG	:5/8 **Note 2**		

Note 2: Assign “5” for TIE Lines and “8” for DID Trunks. For all other programming, see TIE LINE ACCESS [T-3].

Note: To change the data for existing route (RT), initialize the circuit card.

C-15 Class of Service - Individual

General Description

This feature permits all Stations to be assigned a Class of Service in accordance with the degree of system use desired. Each Station is assigned one Class from each of three groups.

Operating Procedure

CLASS OF SERVICE - INDIVIDUAL is assigned via System Data entry. Thereafter, each Station is automatically provided with the desired degree of service.

Service Conditions

1. The CLASS OF SERVICE - INDIVIDUAL feature in the System contains three categories: Telephone Equipment Class (TEC), Service Feature Restriction Class (SFC), and Route Service Class (RSC).
2. TEC determines the type of Terminal and the Signalling Arrangement necessary.
3. Trunk Route Restriction Class provides 15 Groups. Separate Restriction Patterns for each group are programmed in System Data. Group RSC0 is reserved for ATTENDANT CONSOLE use only.
4. Fifteen types of Service Feature Restriction Classes are available for access to the following special features. These classes are assigned to each Station in order to determine which features a particular Station will be permitted to use.

ATTENDANT CAMP-ON WITH TONE INDICATION	[A-1]
CALL BACK	[C-1]
CALL FORWARDING	[C-2, C-3, C-5]
CALL HOLD	[C-6]
DATA LINE SECURITY	[D-12]
EXECUTIVE RIGHT-OF-WAY	[E-1]
OUTGOING TRUNK QUEUING	[O-2]
STATION MESSAGED DETAIL RECORDING (SMDR)	[S-10]
SPEED CALLING - SYSTEM	[S-3]
TRUNK ANSWER FROM ANY STATION (TAS)	[T-8]

   For details, refer to the Command Manual.
5. Stations may be programmed to change SFC and RSC when the Attendant has set NIGHT CONNECTION [N-1, 2]. This is accomplished via System Data entry. For further information, refer to DAY/NIGHT CLASS OF SERVICE [D-15].
6. TEC, RSC, and SFC are independent, with no interaction between them.
7. RSC and SFC can be assigned independently on a TENANT SERVICE [T-12] basis.
8. The system software is capable of eliminating 5db loss on Station-to-Station Calls on a per Station basis.
9. The system software is capable of eliminating DISTINCTIVE RINGING [D-10] on a per Station basis (for OPX Stations).
10. RSC can be overridden by a new RSC, using the AUTHORIZATION CODE [A-20] feature.

Interactions

The following features interact with the fully restricted Station

1. CALL TRANSFER - ALL CALLS [C-11]: A fully restricted Station can be transferred to a Station or the Attendant by another party using CALL TRANSFER - ALL CALLS [C-11]. The fully restricted Station

may transfer a Station-to-Station Call to a Station or the Attendant.

2. CALL FORWARDING [C-2, 3, 5]: Station-to-Station Calls can be forwarded to a fully restricted Station. An attempt to forward an Incoming Trunk Call to a fully restricted Station will result in Reorder Tone.
3. STATION HUNTING [S-7, 8, 9]: Any call directed to a busy, fully restricted Station will hunt to an alternate Station. Careful consideration is required when assigning a fully restricted Station to a Hunt Group, since Incoming Trunk Calls directed to a fully restricted Station will result in Reorder Tone.
4. TRUNK ANSWER FROM ANY STATION (TAS) [T-8]: An attempt by a fully restricted Station to answer a call via the TAS feature will result in Reorder Tone.

The following features interact with Inward Route Restricted Stations

1. THREE-WAY CALLING [T-2]: When a user is restricted from receiving Direct Calls from a Trunk Group, the user cannot be added to an established connection that includes the restricted Trunk Group.
2. CALL FORWARDING [C-2, 3, 5]: Incoming Trunk Calls cannot be forwarded to an Inward Route Restricted Station.
3. CALL PICKUP [C-7, 30]: Incoming Trunk Calls cannot be answered by an Inward Route Restricted Station.
4. ATTENDANT CAMP-ON [A-1]: This feature is denied on calls extended to an Inward Route Restricted Station. Attempts will be routed to Reorder Tone.
5. STATION HUNTING [S-7, 8, 9]: Hunting of an Incoming Trunk Call to an Inward Route Restricted Station is not allowed.
6. TRUNK ANSWER FROM ANY STATION [T-8]: A user restricted from receiving direct Incoming Calls via a given Trunk Group cannot use the TAS feature to answer calls received via that Trunk Group.
7. NIGHT SERVICE [N-1, 2]: Restrictions can be changed for a given Station when the system is switched to the night mode of operation. Therefore, the assignment of an Inward Route Restricted Station as a Night Station can be allowed (see DAY/NIGHT CLASS OF SERVICE [D-15].)
8. CODE CALLING ACCESS [C-21]/MEET ME PAGING [M-1]: Connection of an Inward Route Restricted Line to an answer-back channel is not denied when the call is an Incoming Exchange Call.

Programming

Day/Night Class

STEP 1: **ASYD** - SYS1, Index 75. Should the following commands be developed for Day/Night. 0/1: No/Yes

Note: DAY/NIGHT CLASS OF SERVICE [D-15] - If Day/Night operation is to remain the same, the following Bits should be assigned as data “0”. Only those restrictions that will change between Day and Night should be assigned as data “1”. Further assignments MUST be made in the corresponding command.

Bit 0 - Develop ARSC for Day/Night?
Bit 1 - Develop ASFC for Day/Night?
Bit 2 - Develop ATNR for Day/Night?
Bit 3 - Develop ATDP for Day/Night?

Telephone Class

STEP 1: **ASDT** - Assign Tenant Number, Station Number, LENS, and the TEC to the Station. This is used in regular Business Systems. Assign the SFC and RSC.

TN: Tenant Number
STN: Station Number
LENS: Line Equipment Numbers (six digits)
TEC: Telephone Class (1-31)
1: DP (10 PPS)
2: PB
3: DP/PB
12: Dterm
13: Data Terminal via Dterm
14: Hotline
15: CAS Line
16: Data Terminal via Data Module
18: Virtual Circuit
23: ISDN Terminal
26: Voice Mail (Digital Voice Mail Ports)
27: Eight Conference Equipment
RSC: Route Restriction Class (0-15). For assignment of RSC, ARSC.
SFC: Service Feature Restriction Class (0-15). For assignment of SFC, ASFC.

STEP 2: **ASCL** - This command may be used to change the CLASS OF SERVICE - INDIVIDUAL for existing Stations. The RSC, SFC, and TEC may all be changed.

Service Feature Restriction Class

STEP 1: **ASFC** - A possible 16 (0-15) different SFCs may be constructed allowing or denying access to the various features.

Route Restriction Class

STEP 1: **ASYD** - SYS2, Index 1, Bit 4 and Bit 5. Should a restricted Station be routed to the Attendant Console or receive Reorder Tone? Reorder Tone: Bit 4=0, Bit 5=0. ATTENDANT CONSOLE: Bit 4=1, Bit 5=0.

STEP 2: **ARSC** - A possible 16 (0 - 15) different Route Restriction Classes may be constructed, either allowing a call or subjecting the call to TOLL DENIAL/DIVERSION [T-7] or TOLL RESTRICTION - 3/6-DIGIT [T-7].

Note: RSC 0 is reserved for Attendant Console use only.

STEP 3: **ATDP** - Assignment of Toll Distinctive Pattern command is assigned when Toll Restriction is required. When Toll Restriction changes from Day to Night, then ATDP must be assigned for both Day mode and Night mode.

Note: If a restricted Station is routed to the ATT, all RRI 3 must be RES = 2.

C-17 Consultation Hold - All Calls

General Description

This feature permits a user to hold any Incoming or Outgoing Public Network or TIE Line Call, or any Intra-Office Call, while originating a call to another Station within the system.

Operating Procedure

To hold the original call and place the second call using switch hook flash

1. Press the switch hook. Receive Special Dial Tone.
2. The original call is held.
3. Dial the second Telephone Number. Receive Ringback Tone.
4. The second Station answers. The CONSULTATION HOLD state has been entered.

To return to the original call using switch hook flash

1. In any of the following cases, the Calling Station can return to the original call by pressing the switch hook:
   a.) The second Station called for CONSULTATION HOLD is busy.
   b.) The Calling Station cannot gain access to second Station because of restriction or some other reason.
   c.) The second Station does not answer.
2. If the second Station hangs up, the Calling Station will be returned to the original call.
3. If the Originating Station flashes the switch hook, a THREE-WAY CALL [T-2] will be initiated.

<The following procedures are applied for Desktop terminal only.>

To hold the original call and place the second call using the TRANSFER key

1. Press the TRANSFER key. Receive Special Dial Tone.
2. The original call is held.
3. Dial the second Telephone Number. Receive Ringback Tone.
4. The second Station answers. The CONSULTATION HOLD state has been entered.

To return to the original call using the TRANSFER key

1. In any of the following cases, the Calling Station can return to the original call by pressing the TRANSFER key:
   a.) Second Station Called for consultation is busy.
   b.) Calling Station cannot gain access to the second Station because of a restriction or some other reason.
2. If the second Station hangs up, the Calling Station will be returned to the original call.
3. If the second Station stays on, pressing the TRANSFER key returns the original call to the Dterm while the second call is being held.

By pressing the CONF key, a THREE-WAY CALL [T-2] will be initiated.

Note: After pressing the CONF key, you cannot reestablish a call with the 2nd party while having the 3rd party on hold state.

Service Conditions

1. When TRUNK-TO-TRUNK CONNECTION [T-10] is provided with the system, an Outgoing Public Network or TIE Line Call can be made by the user.
2. For Outgoing Calls, this feature may be activated only after 18 seconds have elapsed since the last digit dialed.
3. CALL PICKUP - GROUP [C-7] is available on CONSULTATION HOLD.
4. A call can be forwarded to a Station in the Remote Node via FCCS.

Interactions

1. This feature is denied whenever the Destination Telephone Set is engaged in any state other than a simple Two-party Connection.
2. CONSULTATION HOLD - ALL CALLS must always be made available when features require switch hook flashing.
3. Index 156 and Index 157 also are used for SMDR, establishing an Arbitrary Connection Timing.
4. If a party is involved in a CONSULTATION HOLD - ALL CALLS state or in an established THREE-WAY CALLING [T-2] Connection, attempts to override the connection are denied. These features are:

EXECUTIVE RIGHT-OF-WAY	[E-1]
CALL WAITING - TERMINATING	[C-12]
BUSY VERIFICATION	[B-3]
ATTENDANT CAMP-ON WITH TONE INDICATION	[A-1]
ATTENDANT OVERRIDE	[A-7]

5. Features that cannot be activated while a user has a party in CONSULTATION HOLD - ALL CALLS are:

EXECUTIVE RIGHT-OF-WAY	[E-1]
TRUNK ANSWER FROM ANY STATION	[T-8]
CALL BACK, Activate or Cancel	[C-1]
CALL PICKUP - DIRECT	[C-30]
CALL FORWARDING, Activate or Cancel	[C-2, 3, 5]
OUTGOING TRUNK QUEUING, Cancel	[O-2]
SPEED CALLING - INDIVIDUAL, Entry	[S-21]
MESSAGE REMINDER	[M-19]

6. All other features may be activated as long as the user's line CLASS OF SERVICE - INDIVIDUAL [C-15] allows that feature.
7. Upon establishing a CONSULTATION HOLD - ALL CALLS Connection and dialing a third party, if the user receives Busy or Reorder Tone instead of Ringback Tone, the user flashes once to disconnect the Tone and reconnect the second party.
8. Upon establishing a CONSULTATION HOLD - ALL CALLS Connection and dialing a MISCELLANEOUS TRUNK ACCESS [M-2] Code, if no Dial Tone is given (after the Trunk is seized), or if a mistake is made in dialing, or if Intercept Announcement or a Permanent Tone is given, the user must go on-hook and wait for Ringback Tone.
9. A user, who established CONSULTATION HOLD - ALL CALLS, is reconnected to the party in soft hold when the third party is disconnected.
10. When using CALL TRANSFER - ALL CALLS [C-11] from a Station (second party) to another Station or Trunk (third party), the Control Station can transfer the call during ringing (blind transfer). When the Control Station goes on-hook, the Station in soft hold is connected to the Third Party's Connection and hears Ringback Tone. A 30-second timer is set when the Controlling Station goes on-hook. If the timer times out before the third party answers, the Control Station will receive Ringback Tone.

11. Reorder Tone is returned to the Station using CONSULTATION HOLD - ALL CALLS when the following actions occur:
    - The user attempts to place an Off-net Call using MISCELLANEOUS TRUNK ACCESS [M-2] on a Restricted Route.
    - The user dials the number of an unassigned Station or Station residing in an inaccessible Tenant via Tenant Service.
    - The user dials vacant or invalid first digit (no assignment in ANPD either for Level or Connection Status).
    - The TRANSFER key or FLASH-BUTTON -D [F-3D] is used instead of momentarily pressing the switch hook. A FLASH BUTTON -D[F-3D] is required for a Connection Index of B, Busy.
12. When a SIP Handler Controlled SIP terminal is talking with the transferring destination and the transferring destination is released, the SIP Handler Controlled SIP terminal returns to the held call but the call hold state is not canceled automatically. An operation to cancel the call hold state is needed.

Programming

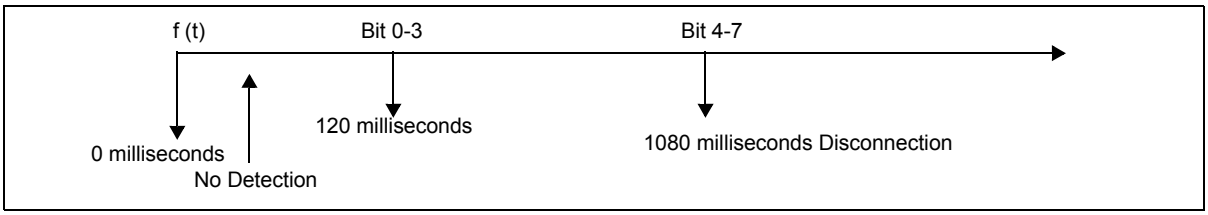
ASYD - SYS1, Index 156. Trunk Soft Hold Timer A, used with Second Dial Tone Trunks. Assign data 00H for 30 seconds. (Data 31H may be used for a value of two seconds if CALL TRANSFER - ALL CALLS [C-11] is to be used immediately after connecting to the Trunk.) This is the time between Trunk Connection and when switch hook flash is allowed on a Non-senderized Trunk Connection.

SYS1, Index 157. Trunk Soft Hold Timer A, used with Senderized Trunks. Assign data 00H for a value of 18 seconds. (Data 31H may be used for a value of two seconds, if CALL TRANSFER - ALL CALLS [C-11] is to be used immediately after connecting to the Trunk.). This is the time between Trunk Connection and when switch hook flash is allowed on a Senderized Trunk Connection. It is also used to determine when SMDR billing will begin.

SYS2, Index 1, Bit 0 and Bit 1. CONSULTATION HOLD allowed or denied? 00: denied; 01: Originating and Terminating Calls allowed, Tandem Call denied; 11: All calls allowed. Normally assign data “11”.

SYS3, Index 2, Bit 0 to Bit 3. Switch hook flash starts timer. To calculate this value: (1 ~ FH) × 120 milliseconds = timer.

SYS3, Index 2, Bit 4 to Bit 7. Switchhook flash ends timer. To calculate this value: (1 ~ FH) × 120 milliseconds = timer. For both beginning and ending timers, assign 91H for 120 - 1080 milliseconds.



C-20 Centralized Attendant Service (CAS)

General Description

For applications requiring multiple sites, but having common interests or operation, the System can operate in a Main (attended) or Satellite (unattended) configuration. Typically, Incoming Calls to the Satellite are routed to the Main Location for processing by the Central Attendants within the CAS Network. This feature can be used to improve overall communication efficiency.

Operating Procedure

Listed Directory Number (LDN) Call

1. Upon arrival of an LDN Call at a Satellite System, the LDN Call is automatically directed to the Main System via a Release Link Trunk.
2. Upon receiving the Incoming CAS Call, the ATTENDANT CONSOLEs [A-3] will display the Incoming CAS Call by flashing the CAS Lamp.
3. The Attendant answers the Incoming CAS Call by pressing the CAS or ANSWER key. Hear the Incoming Call Tone (480 Hz - 440 Hz - 480 Hz) for 300 milliseconds.
4. The Attendant presses SHF, causing a switch hook flash. Hear Special Dial Tone from the Satellite System.
5. The Attendant dials the required Telephone Number in the Satellite System.
6. Upon hearing Ringback Tone, the Attendant presses the RELEASE key to make the Release Link Trunk available to serve another call.

Recall on Don't Answer Call

1. A Satellite System is configured to re-establish a connection from a Calling Party via a Release Link Trunk to an Attendant in the Main System, if the Called Station does not answer within a prefixed time.
2. Upon receiving the recall on a No Answer Call, the ATTENDANT CONSOLE [A-3] will display the Incoming CAS Call by flashing the CAS Lamp.
3. The Attendant answers the Incoming CAS Call by pressing the CAS or ANSWER key. Hear continuous Ringback Tone (440 Hz - 480 Hz) for 300 milliseconds.
4. After hearing the burst of continuous Ringback Tone, the Attendant and the Calling Party talk over Ringback Tone because of ringing the Called Station continuously during Recall Connection.
5. If the Calling Party desires to hold the connection, the Attendant presses the RELEASE key to make the Release Link Trunk available to serve another call.

Camp-On

1. If the Attendant directs the Satellite System to connect a call to a Busy Station, the Satellite System transmits a Camp-On Confirmation Tone (100 millisecond burst of 440 Hz) to the Attendant, also a camp-on indication Tone (200 millisecond burst of 480 Hz) to the Called Station, and reconnects the Calling Party to the Attendant.
2. If the Calling Party desires to wait, the Attendant presses the RELEASE key to make the Release Link Trunk available to serve another call.
3. If the Camped-On Station remains busy for longer than the prefixed time, the Satellite System reconnects the Calling Party via a Release Link Trunk.
4. Upon receiving the recall on a Camped-On Call, the ATTENDANT CONSOLE [A-3] will display the Incoming CAS Call by flashing the CAS Lamp.

5. The Attendant answers the Incoming CAS Call by pressing the CAS or ANSWER key. Hear the Camp-On Identification Tone (100 millisecond burst of 440 Hz).
6. When the Camped-On Station becomes idle, the Satellite System rings the camped-on Station.

Remote Hold

1. The Attendant talks with a Central Office party.
2. The Attendant presses the SHF key. Hear Special Dial Tone.
3. The Attendant dials the Remote Hold Access Code (one or two digits). The Central Office party is placed in the hold state at the Satellite System, and the Attendant hears the Hold Confirmation Tone (five, 50 millisecond bursts of 440 Hz).
4. The Attendant presses the RELEASE key to make the Release Link Trunk available to serve another call.
5. If the Central Office party remains in the hold state for longer than the prefixed time, the Satellite System reconnects the Central Office party via the Release Link Trunk.
6. Upon receiving the recall on the Hold Call, the ATTENDANT CONSOLE [A-3] will display the Incoming CAS Call by flashing the CAS Lamp.
7. The Attendant answers the Incoming CAS Call by pressing the CAS or ANSWER key. Hears the Hold Recall Identification Tone (five, bursts of 440 Hz).
8. The Attendant is now able to complete the call.

Operator Call

1. When an Operator Call Code (one or two digits) is received from a Station or TIE Line Trunk in the Satellite System, an Operator Call is directed to the Main System via Release Link Trunks.
2. Upon receiving the Incoming CAS Call, the ATTENDANT CONSOLE [A-3] will display the Incoming CAS Call by flashing the CAS Lamp.
3. The Attendant answers the Incoming CAS Call by pressing the CAS or ANSWER key. Hear the Operator Call Identification Tone (two, 100 millisecond bursts of 440 Hz).
4. At the expiration of this Tone, the Attendant talks with the Calling Party.
5. The Attendant's procedure for serving this call is similar to that for an LDN Call.

Call Transfer - Attendant

1. After any call is completed to a Satellite Station, the Called Party Station can call the Attendant in the Main System to transfer the Calling Party to another Station in the Satellite System. The operation of this feature is similar to that of an Operator Call.
2. The Attendant answers the Incoming CAS Call by pressing the CAS or ANSWER key. Hear the Operator Call Identification Tone (two, 100 millisecond bursts of 440 Hz).
3. After answering the call, the Attendant presses the SHF key, causing the Satellite System to disconnect the Station that dialed the Operator Call Code and connects to the Calling Party.

Night Service for the Incoming CAS Call in the Main System

1. In Night mode, the Incoming CAS Call is directed to a Night Station to be answered.
2. When the Night Station answers that call, the Night Station hears the Identification Tone depending on the call.
3. The Night Station momentarily switch hook flashes. Hear the Special Dial Tone from the Main System.
4. Dials the Flash Signal Sending Access Code (one or two digits). Hear Special Dial Tone.
5. After this, the operation of the Night Station is similar to that of the Attendant.
6. The Night Station must be provided with DTMF Signaling.

Service Conditions

The following applies to a System used at the Main CAS Location (Attendant Location)

1. The number of Release Link Trunks is determined by overall Satellite to Main Location traffic requirements.
2. RLTs can be seized in one direction only, from the Satellite System to the Main System.
3. The Attendant cannot originate or INTER-POSITION TRANSFER [I-5] a call on an RLT.
4. The Attendant cannot extend an Incoming CAS Call to a Station, Trunk, or another Attendant in the Main System.
5. CALL PROCESSING INDICATION [C-8] is not provided to the Attendant serving the Incoming CAS Call.
6. The Attendant at the Main System cannot connect RLT Calls to Main System users.
7. The Main System Attendant cannot busy verify a Station in the Satellite System.

The following applies to a System used as a Satellite Location

1. The number of Station ports used will equal the number of RLTs required.
2. A call extended by an Attendant in the Main System can be recorded on STATION MESSAGE DETAIL RECORDING (SMDR) [S-10].
3. An optional ATTENDANT CONSOLE [A-3] may be provided at a Satellite System to allow a local Attendant to control access to WATS or FX Trunks.
4. CAMP-ON [A-1] and Remote Hold are not provided for an Operator Call.
5. The NIGHT CONNECTION - FIXED [N-1]/FLEXIBLE [N-2] and TAS feature can be provided at the Branch Location.
6. A System can be equipped with 100 RLTs per Hunt Group.
7. FCCS is not available for this service.

Interactions

1. Release Link Trunks are not used for Voice Calls between Systems. Release Link Trunks cannot be transferred to a Station in the Main System. TIE Lines must be installed for regular Voice Connection between systems.
2. CAS Systems have no effect on Tandem Connection, LCR (6)/LCRS (7), or any other Trunking application.
3. The following applies to a system used at the Main CAS Location (Attendant Location): The Attendant Console cannot originate or INTERPOSITION TRANSFER [I-5] a call on an RLT. The Attendant Console at the Main System cannot connect RLT Calls to Main System Station users.
4. A System can be equipped with 100 RLTs per STATION HUNTING [S-7, 8, 9] Group. This is a constraint of STATION HUNTING [S-7, 8, 9] and UNIFORM CALL DISTRIBUTION [U-1], not CAS.
5. Listed Directory Numbers (LDN) may be provided for the various Satellite Systems. Because these LDNs will ring in only at the CAS key of the Attendant Console, indications to distinguish among the multiple LDNs are not specifically available. However, the Route Number is displayed and can be used to distinguish Satellite Systems.
6. During the Day mode of operation, if all RLTs are busy, Incoming COT Calls will wait and Ringback Tone is sent to the Calling Party.
7. If the Attendant Console at the Main System should switch hook flash to get Dial Tone from the Satellite and all ports are busy, Reorder Tone is sent to the Attendant Console. The Attendant can switch hook flash COT Party.

8. The CAS feature allows Attendant Console to be located at both the Main and Satellite Locations. When the Satellite Location goes to the Night mode, translations are available to connect Incoming Central Office Calls to the RLT Circuits. This is accomplished by ACSI and the NIGHT CONNECTION FIXED/ FLEXIBLE [N-1, 2] feature.
9. When the Night Stations are located at the Satellite System and no Attendant Console is provided, a Night key must be installed at the Satellite Location to activate Night Service. When the key is operated, the CAS operations are suspended, and Incoming Calls are terminated at the NIGHT CONNECTION FIXED/ FLEXIBLE Stations [F-1, 2] or via TRUNK ANSWER FROM ANY STATION [T-8]. When answered, the Night Station can transfer the call to the Main Location via the TIE Trunks but not the RLT Circuits. A PA-D07 (Attendant Console) circuit card must be supplied to the Satellite System.
10. PAGING ACCESS [P-1]: This feature allows the CAS Attendant to access Paging Equipment located at the Branch Location upon answering an RLT Call. See PAGING ACCESS [P-1] and MEET-ME PAGING [M-1].
11. CODE CALLING ACCESS [C-21]: This feature allows the CAS Attendant Console to access CODE CALLING Equipment located at the Satellite Location upon answering an RLT Call.
12. STATION MESSAGE DETAIL RECORDING (SMDR) [S-10]: SMDR is provided for calls extended by a CAS Attendant Console.

Programming

Main System

STEP 1: **ASYD** - SYS1, Index 150. Assign data 00H. This is the Guard Timer between the Attendant Console switch hook flash and the dialing of digits.

SYS1, Index 151. Assign data 00H. This allows a 480 milliseconds SHF. Timer Class is given as 32 milliseconds. For variable Hooking values, adjust the Miscellaneous Timer Class in Bit 0 to Bit 3.

SYS2, Index 13, Bit 1. Assign a data "0" for a Main System. This Bit applies to the PFT circuit card. This application is not common.

STEP 2: **ARTD** - Assign the CAS Routes (Release Link Trunks, RLT) in ARTD as follows. Be sure to assign CDN 6:TCL as data "7", CA Line.

RT:1

1-OSGS	:2	2-ONSG	:3	5-TF	:3	6-TCL	:7*	7-L/T	:1

Note: 15-LSG:5**

Note: *This Bit will provide a Dterm Trunk display when used as a Night Station.

Note: **This Bit is used when the TLT Card is used at the Main System Location for the Release Link Trunks. At the Satellite Location, a TLT Card is used for Release Link Trunk (RLT). If a COT Card is used as the RLT at the Main Location, assign LSG as data 0 Loop.

Note: To change the data for existing route (RT), initialize the circuit card.

STEP 3: **ATRK** - Program the LENS of the RLT Trunks. These Trunks must be COT circuit cards.

STEP 4: **MBTK** - Make idle the Release Link Trunk.

STEP 5: **AAKP** - To provide the Attendant Console with a FLASH key, the assignment of Attendant Console key Pattern (AAKP) must be programmed. Assign the ATN: Attendant Number (TRK Number for RT 901); KYN 5; and FUNC 1, CAS Line. (If this data is not assigned, CAS Trunks will not ring into the

Attendant Console.)

STEP 6: **ANND** - For the Outgoing Connection Parameter, OG; assign Route Number and the Number of Necessary Digits. The NND is the number of digits in the Satellite Switches' Station Numbering Plan.

Night Station - CAS

STEP 1: **ANPD** - Reserve a Number Level for Feature Access.

STEP 2: **ASPA** - Assign an Access Code to the Hooking, SRV = "SSC: Service Code"; SID = 43. Assign for a Connection Status Index (CI) of "H: Hooking".

STEP 3: **ACFR** - Assignment of Call Forwarding Restriction. Allow Call Class Index (CCI) 14, CAS Trunk Main, a Transfer Service Feature Index (TSFI) of 5, TAS.

STEP 4: **ACSI** - Assign the Incoming CAS Route and Trunk to the Night Station and Tenant Number. Assign the CSI = 4 for Night Connection.

TAS

STEP 1: **ANPD** - Reserve a Number Level for Feature Access.

STEP 2: **ASPA** - Assign an Access Code to TRUNK ANSWER ANY STATION, SRV = "SSC: Service Code"; SID = 16. Assign for a Connection Status Index (CI) of "N: Normal".

STEP 3: **ACFR** - Assignment of Call Forwarding Restriction. Allow Call Class Index (CCI) 14, CAS Trunk Main, a Transfer Service Feature Index (TSFI) of 5, TAS.

STEP 4: **ATAS** - Assign the TAS Circuit to the system. Refer to TRUNK ANSWER FROM ANY STATION [T-8] for details.

Note: A second call will go to TRUNK ANSWER FROM ANY STATION if the first call is being answered by the Night Station.

Satellite System

STEP 1: **ASYD** - SYS1, Index 149. Assign 00H for Recall Timer to Attendant Console at the Main System Location. This value is 30 seconds.

SYS2, Index 11, Bit 7. Will DIT Park in a Busy Station queue? 0/1: Yes/No, it will be re-routed to the Attendant Console. Assign data "0" for Queuing.

SYS2, Index 13, Bit 1. Assign data "1" for Satellite System.

STEP 2: **ANPD** - Reserve a Number Level for Feature Access. Select Normal Service (N), Hooking Service (H), or Busy Service (B) from the combo box for Connection Indexes. Assign NND in accordance with a predetermined Numbering Plan. Reserve a Number Level for Station Numbers. If the Satellite System does not have an Attendant Console, 0 can be used as a Station Number. If not, consider a Numbering Plan for the Release Link Trunks that will be assigned to the Stations assigned a TEC = 15.

STEP 3: **ARSC** - Construct a Route Restriction Class that allows a Direct Connection to all Incoming Trunks that will be served by the Attendant Console at the Main System Location. Assign via Route Restriction Index, RRI 1, Direct Connection on Incoming Trunks.

STEP 4: **ASFC** - Construct a Service Feature Restriction Class that allows Station Message Detail Service SFI 14. Do not provide CALL FORWARDING [C-2, 3, 5].

STEP 5: **ASDT** - Assign ports of the 16LC that receive the Release Link Trunks as TEC 15. The RSC assignment should allow a Direct Connection with all required Incoming Trunks. Allow these Trunks via Route Restriction Index (RRI) = 0 and 1. The SFC assigned is separately constructed and can include SFI 14 [Station Message Detail Recording (SMDR) - Trunk Basis]. See STATION MESSAGE DETAIL RECORDING (SMDR) [S-10]. When assigning Station Numbers, keep in mind that one of these Stations will be the lead Station in a UCD Group and that Station Number will be used by the Satellite Stations to access the Attendant Console at the Main System Location. To keep the user from becoming confused when a different number appears on the LCD, other than the one

dialed, make the Station Numbers appear as sequential Circuit Numbers such as 100, 101, 102, etc.

Note: If a COT Card is used at the Main Location for the RLTs, 16LC Cards are used at the Satellite Location.

STEP 6: **ASHU** - Assign the Release Link Trunk Stations to a UCD Group. See UNIFORM CALL DISTRIBUTION [U-1]. One Station Number will be designated as the lead Station. This number will simulate Dial Access to Attendant. Also, Incoming Trunk Calls will be distributed among the available Release Link Trunks evenly. ASHC may also be used, STATION HUNTING - CIRCULAR [S-7].

STEP 7: **ACSI** - Assign the Incoming Satellite Trunks to be answered at the Main System CSI = 3, Direct In Termination, Day and Night Mode. Then assign the Stations programmed above. This ties Incoming Trunks to the TLTs.

STEP 8: **ACFR** - Assignment of Call Forwarding Restriction. Upon determining which types of Call Class Indexes (CCI) will be answered at the Main System, such as LDN (DDD), WATS, FX. Allow those CCI a Transfer Service Feature Index of 15, CAS Line.

STEP 9: **ACSA** - (Optional) Use this command rather than ACSI if CAS operation is desired in Day mode, and if Incoming Trunks are to stay at the Satellite and be answered by TAS or a NIGHT CONNECTION - FIXED/FLEXIBLE [F-1/2] Station in Night mode.

C-21D Code Calling Access -D

General Description

This feature provides the ATTENDANT CONSOLE [A-3] and users dial access to Code Calling Equipment.

Note: This feature is only available in North America.

Operating Procedure

1. Go off hook. Receive Dial Tone.
2. Dial the CODE CALLING ACCESS Code.
3. Dial the Code Number for the unit desired.

Service Conditions

1. Amplifiers, speakers, and Code Calling Units must be provided by the customer.
2. Loop Start Central Office Trunks are used for interfacing Code Calling Equipment. One Central Office Trunk circuit is required per Paging Port and per Answer Back Port.
3. FULLY RESTRICTED STATIONS: A fully restricted Station may make only dial “0” and Station-to-Station Calls. Therefore, the CODE CALL ACCESS is denied.
4. HOTLINE [H-1]: A Hotline Station user can be programmed to connect to a Code Call Unit when the Station goes off-hook.
5. ROUTE RESTRICTION: A Station that is Code Calling Route Restricted is denied dial access to the Code Call Unit. Restricted Calls are routed to Reorder Tone. If allowed by the user’s Class of Service, Attendant assistance can be used to complete a Code Call for a Route-Restricted Station.
6. Attendants, users and TIE Trunk users can dial an Access Code and a two or three-digit Called Party Code to activate a Coded Signal.
7. The Desktop terminal display will indicate the Elapsed Call Time and Trunk Number.

Interactions

PAGING TRANSFER [P-10] may be used along with CODE CALLING ACCESS. Refer to PAGING TRANSFER [P-10] for additional programming.

Programming

STEP 1: **ANPD** - Reserve a Number Level for Miscellaneous Trunk Access. Select Normal Service (N) or Hooking Service (H) from the combo box for Connection Indexes. Assign NND in accordance with a predetermined Numbering Plan.

STEP 2: **ASPA** - Assign the Access Code in the level as assigned in ANPD above. Type of Service, SRV = "OGC: Outgoing Call" for Outgoing Trunk. Assign the Route Number associated with this Access Code.

STEP 3: **ARTD** - Assign the CODE CALLING ACCESS Route in ARTD as follows:

RT:1
1-OSGS :2 2-ONSG :3 5-TF :1 6-TCL :9 7-L/T :1
8-RLP :2

Note: To change the data for existing route (RT), initialize the circuit card.

STEP 4: **ATRK** - Program the LENS of the Paging Trunks. These Trunks must be COT or 8TLT Cards.

STEP 5: **MBTK** - Make idle the Paging Trunk.

C-22D Called Station Status Display -D

General Description

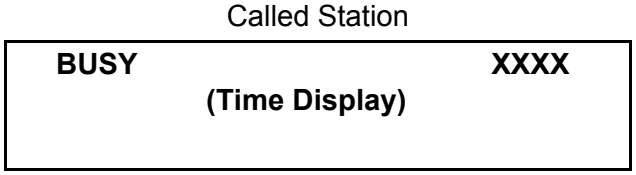
This feature provides for the status of a Called Station to appear on the LCD of a Calling Desktop terminal.

Operating Procedure

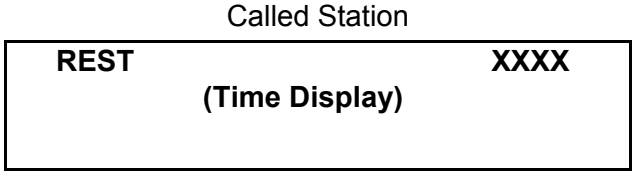
Lift handset or press the SPEAKER key. Dial desired Station. The LCD display indicates the status of the Called Station.

Service Conditions

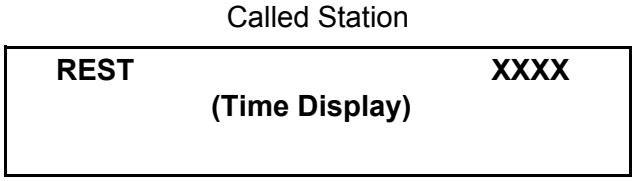
1. If the Called Station is idle, the display flashes its number until the call is answered. If the Called Station is busy, the LCD displays:



2. If the Called Station is in DO NOT DISTURB -D [D-11D], the LCD displays:



3. If the Calling Station encounters a restricted Station, the LCD displays:



4. CALLED STATION STATUS DISPLAY -D can be restricted on a system basis. The LCD displays:

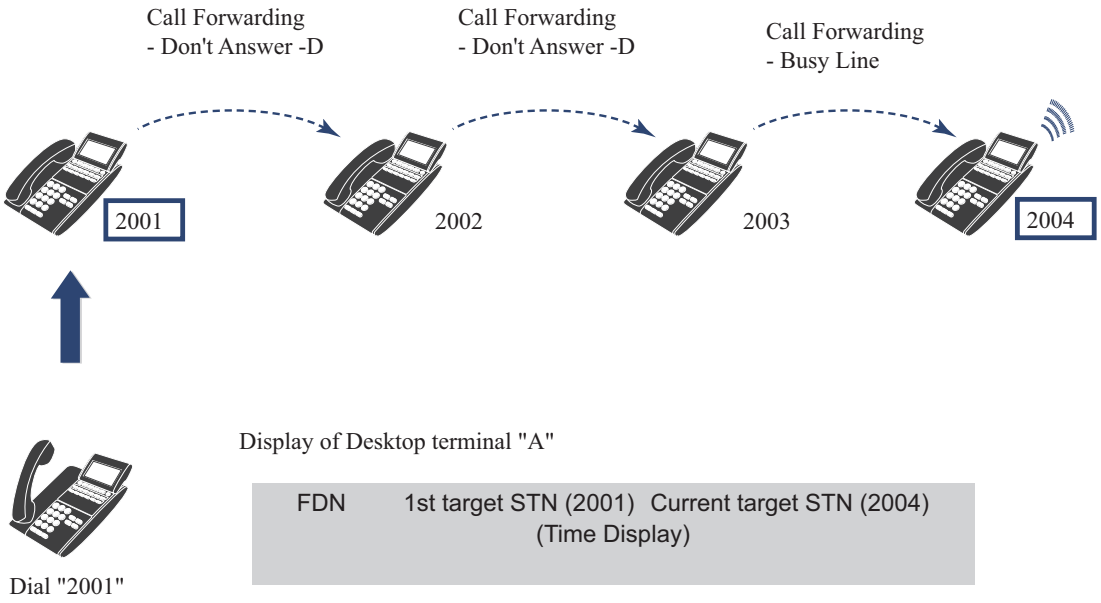


5. If calling a target station by Call Forwarding - All Calls, Call Forwarding - Busy Line, Call Forwarding - Don't Answer -D, Logged Out IP Station - Call Destination, or Call Forwarding PS Incoming Call Incomplete, the LCD of calling party's Desktop terminal displays as follows.

FWD 1st target STN Current target STN (Time Display)

Note: If Call Forwarding - All Calls, Call Forwarding - Busy Line, Call Forwarding - Don't Answer -D, Logged Out IP Station - Call Destination, or Call Forwarding PS Incoming Call Incomplete are mixed on Multi-hops forwarding, the kind of forwarding and the first station number that are displayed on the LCD of calling party's Desktop terminal and called party's Desktop terminal is the kind of forwarding and the station number of station that is forwarded first.

Note: If Call Forwarding - All Calls, Call Forwarding - Busy Line, Call Forwarding - Don't Answer -D, Logged Out IP Station - Call Destination, or Call Forwarding PS Incoming Call Incomplete are mixed on Multi-hops forwarding, the current target station number to be displayed on the LCD of calling party's Desktop terminal is the station number that is currently called.



Programming

ASYD - SYS1, Index 78, Bit 1. Is the Service Feature Display in service? 0/1: No/Yes.

C-24D Calling Number Display -D

General Description

This feature provides for the Station and Trunk Number of an Incoming Call to appear on the Desktop terminal's LCD. This display will flash while the call is ringing, then appear steady when the call is answered.

Operating Procedure

No manual operation is required.

Service Conditions

1. When a call terminates on a line other than the PRIME-LINE [P-13D], the LCD displays the Incoming Station determined by LINE PRE-SELECTION -D [L-9D].
2. LINE PRE-SELECTION -D [L-9D] is governed by a timer.

Programming

No programming is required.

C-25 Call Forwarding - Intercept / Announcement

General Description

This feature provides for interception of STATION-TO-STATION [S-11], DIRECT INWARD DIALING [D-8], Attendant-to-Station and CCSA ACCESS [C-14] Calls that cannot be completed (unassigned Station, Level, etc.). These calls are automatically routed to a Recorded Announcement informing the caller that an Inoperative Number was reached and giving the Listed Directory Number for information.

This feature permits a Station-originated Call, upon encountering a restricted Outgoing Number, to automatically be routed to a Recorded Announcement informing the caller that the Dialed Number is restricted for this Station.

This feature permits a Station-originated Call, upon encountering a Trunk Busy Condition, to automatically be routed to a Recorded Announcement informing the Main System caller that all the Outgoing Trunks are busy.

Operating Procedure

No manual operation is required, once this feature has been programmed in System Data.

Service Conditions

1. CALL FORWARDING - INTERCEPT / ANNOUNCEMENT requires one of the following hardware alternatives:
   a.) Central Office Trunk (COT) or 2W E&M Trunk and an Announcement Machine, or
   b.) Digital Announcement Trunk (DAT).
2. If the Outside Party or Attendant forgets to hang up, the system will automatically disconnect in 30 seconds (provided this is programmed in System Data). If no disconnect timing is programmed, the Outside Party will control disconnect.
3. The forwarding destination for this feature can be set on a tenant basis or on a route basis.

   Note: The following conditions apply:

   - Data assignment on a route basis is available since FP95-114 V4 Issue 2.0 (earlier than FP95-114 V4 Issue 2.0, it can be set on a tenant basis only).
   - To set the forwarding destination on a route basis, see CALL FORWARDING DESTINATION PER TRUNK ROUTE - INCOMING CALL [C-209].

4. The following call conditions, which cannot be completed, can be routed to intercept:
   a.) vacant level
   b.) unassigned Telephone Number
   When Attendant is assigned as the forwarded destination by Call Forwarding-Intercept, the LCD on the caller's station shows the same display as Call Forwarding-Don't Answer.
5. Only one common message can be provided for the different intercept conditions.
6. There is no method by which to exempt individual DID, CCSA, and TIE Trunks from intercept.
7. A Station dials the LEAST COST ROUTING [L-5] Access Code and then an Outside Number. If that Outside Number is not listed in LEAST COST ROUTING [L-5] Tables (Outgoing Routing Patterns, OPR), the call will be routed to an Intercept Announcement.
8. If a different message is required for each of the three intercept conditions, three Trunks or three Announcement Machines must be provided.

9. When a call is terminated from the ISDN Trunk or MG(SIP), RBT is sent to the caller if Announcement Trunk Answer Signal is not sent.
10. When a call with Consultation Hold is terminated from a counter office, if the Announcement Trunk Answer Signal exists, announcement trunk will operate. To return to the call, press the DISC key. For more information about it, refer to Consultation Hold Release [C-169].
    Note that pressing the hooking key disables the announcement trunk to operate although it returns to a call.

When encountering a Restricted Route

11. This feature is valid when Station to Outgoing Trunk Connection is restricted.
12. This feature is applied equally to all Trunk Routes in the system. It cannot be denied to some Routes and available to other Routes.
13. This service is valid when a call is redirected to an OG Trunk via CALL FORWARDING - OUTSIDE [C-28]. The Announcement will be heard when the Calling Station is also a restricted Station.
14. This service is not valid for TRUNK-TO-TRUNK CONNECTION [T-10] when the Called Station has set CALL FORWARDING - OUTSIDE [C-28] via DID or TIE Trunk.
15. This service is not valid for Outgoing Calls that do not go out when all Trunks or Senders are busy.

When encountering All Trunks Busy

11. This feature is valid for Outgoing Station-to-Trunk or Tandem Trunk-to-Trunk Connections.
12. Automatic OUTGOING TRUNK QUEUING [O-2], OFF-HOOK QUEUING [O-7] and ROUTE ADVANCE [R-6] are mutually exclusive with this feature. These features take priority over OUTGOING TRUNK BUSY - ANNOUNCEMENT [O-21].
13. ANNOUNCEMENT SERVICE is provided to FCCS Nodes that have no Announcement Trunk using the Trunk in the Remote Node.
14. When CALL FORWARDING - INTERCEPT / ANNOUNCEMENT [C-25] or CALL FORWARDING - INTERCEPT / ANNOUNCEMENT - ATTENDANT [C-75] is used in an FCCS Network, the additional data shown below is required.
    a.) The Logical Route Number should be applied to the Physical Route that is provided the service.
    b.) The restriction of Route-to-Route Connection between FCCS Nodes should be released.
15. When using the FCCS Network, CALL FORWARDING - INTERCEPT / ANNOUNCEMENT [C-25] or CALL FORWARDING - INTERCEPT / ANNOUNCEMENT - ATTENDANT [C-75] is not provided for an Incoming Call using AUTOMATED ATTENDANT [A-82], DIRECT-IN TERMINATION [D-7], or REMOTE ACCESS TO SYSTEM [R-2].
16. When using the FCCS Network, the Call Forwarding for an Incoming Call routed to a Dead Level or an Unused Number has priority over the Announcement Service for the Incoming Call.
17. This feature allows the Incoming Trunk Call that cannot be completed [including a CCIS Terminating Call that is abandoned by a Route Restriction Class (RSC) of the Calling Party] to be forwarded to a designated Station in the Remote Node via FCCS. The Call Forwarding Destination Number must be programmed by using the Telephone Number (TELN) of the Station.
18. To forward an Incoming Trunk Call to a predetermined Station in the Remote Node via FCCS, the data assignment of ACFOL/ACFON is required. When Call Forwarding is activated by using ACFOL/ ACFON, only one Station can be registered as the Call Forwarding Destination in a system.
19. The Call Forwarding Destination that is registered by ACFOL/ACFON is available only when an Incoming Call is forwarded to a Station.
20. The feature data via DM, LDM and NDM can coexist in a system. However, data registered on NDM has priority. Data registered on LDM has precedence over data registered on DM (the priority is NDM $\rightarrow$ LDM $\rightarrow$ DM).

Interactions

1. Each Route connected to an Announcement Machine is also an EQP. The same Route can be assigned as different EQPs. However, it is best to separate EQP functions by Route.
2. TLT, COT, or DAT Card may be used.

Programming

STEP 1: **ACFO** - Enable this feature by allowing CF = 3: Call Forwarding - Intercept. The CFI parameter will come up for assignment. An Attendant Console may be programmed so that Intercepted Calls will terminate at the Call Forwarding Intercept key. If no assignment is made, the System program goes to AAED for Announcement Trunk information because the destination of the transfer is a Trunk and not a Station or Attendant Console.

STEP 2: **ARTD** - Assign the Announcement Route as shown below:

RT:1

1-OSGS	:2	2-ONSG	:3	5-TF	:1	6-TCL	:1	7-L/T	:1
8-RLP	:2	15-LSG	:0*						

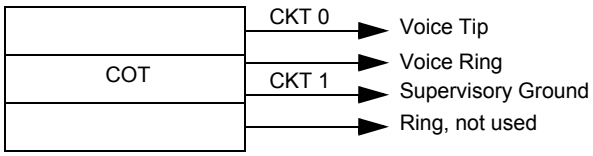
Note: *If a TLT circuit card is used, assign five, E&M.

Note: To change the data for existing route (RT), initialize or reboot the trunk device such as a circuit card or VS32.

STEP 3: **ATRK** - Assign the LENS, Announcement Trunk Route Number, Trunk Number, and Tenant Number.

STEP 4: **MBTK** - Assign the Make Idle status to the Announcement Trunk.

Note: If supervision is required, use the first two circuits of the COT circuit card. Move the switch on the COT to PGT.



STEP 5: **ARRC** - If TIE Line or REMOTE ACCESS TO SYSTEM [R-2] Connection to the ANNOUNCEMENT SERVICE Trunk is required, allow Trunk-to-Trunk Connection using ARI-D-RES, Direct Connection.

STEP 6: **AAED** - Used to assign Announcement Equipment. Assign:

TN: Tenant Number

EQP: Announcement Equipment Number

- 0: Dead Level
- 1: Unused Number (LCR OPR not programmed)
- 2: Available
- 3: Available
- 4: Outgoing Trunk Group Busy Announcement
- 5: Available

6: Available
7: Outgoing Route Restriction Announcement
8-15: Available for ANNOUNCEMENT SERVICE [A-15] applications.

RT, TK: Route and Trunk Number of the Trunk connected to the Announcement Equipment.

C: Duration of connection, 0/1: Disconnect in 30 seconds/the connection is held until the Station releases. See the requirements of the Announcement Equipment.

R: Sending RBT, 0/1: Sending/Not sending. Normally assign data "0".

Note: Assign "1" (not sending Ringback Tone) when the "M" parameter = 0 (Single Connection).

A: Answer Signal Sending, 0/1: No answer from Incoming Trunk/answer from Incoming Trunk. Normally assign data "0". No Answer Signal is sent to the Central Office, therefore the Calling Party will not be billed for the call.

M: Multiple Connection, 0/1: Single Connection/Multiple Connection. See the requirements of the Announcement Equipment.

STEP 7: **ACFR** - Assignment of Call Forwarding Restriction. Upon determining which types of Call Class Indexes (CCI) will be answered via CALL FORWARDING - INTERCEPT, for example, CCSA and DID Calls: allow those CCIs a Transfer Service Feature Index of 1, for CALL FORWARDING - ALL CALLS, NO ANSWER, BUSY.

FCCS Programming

This data setting describes only the additional or replacement commands used to activate FCCS Services. When used as an FCCS feature, these commands replace the non-network commands (for example, AAED is replaced by AAEDN). Note that the data setting for FCCS Service varies depending on the Node.

STEP 1: **ALRTN** - This data must be assigned to the NCN. Assign the Logical Route Number to the physical route.

STEP 2: **AAEDN** - This data must be assigned to the NCN. Assign the Announcement Equipment Data.

EQUIPMENT KIND:
- DAT/Announcement Machine
- VS32

TN: Enter Tenant Number

EQP: Announcement Equipment Number
0 = Dead Level
1 = Unused Number
2-15 = Announcement Service

Duration of Connection:
- Disconnection occurs in 30 seconds
- Connection is held until the Station releases. See the requirements of the application for the Announcement Equipment. For Dead Level applications, data 0 is assigned.

Do Not Send RBT [On/Off]

Note: Assign Do not Send RBT when Multiple Connection = Single.

Send Answer Signal [On/Off]
Multiple Connection: Select either Single or Multiple.
LGRT: Logical Route
TK: Trunk Number **Note 1**
ANP: Announcement Pattern (0 - 255)
REP: Repeat Pattern (0 - 255)

STEP 3: **ARRCN** - This data must be assigned to the NCN.
Release the restriction between the Terminating Trunk and the Announcement Trunk.

STEP 4: **ARSCN** - This data must be assigned to the NCN.
Release the restriction between the Announcement Trunk and the Station.

Note 1: Not necessary for Multiple Connection = Single.

Note: The current command data shown is set to all Nodes using the service: ASTD, ARTD, ATRK, MBTK

To forward the Incoming Trunk Call that cannot be completed to the predetermined Station in the Remote Node via FCCS, the following data assignment is required.

ACFOL/ACFON - TN = Tenant Number, Call Forwarding Service, CF = 3: Call Forwarding - Intercept, Forwarding Destination, CFI = TELN (Forwarded to Station). When TELN is assigned to the CFI Parameter, assign the Telephone Number (a maximum of 16 digits) of the Forwarding Destination Station to the TELN Parameter.

C-26 Call Forwarding - Override

General Description

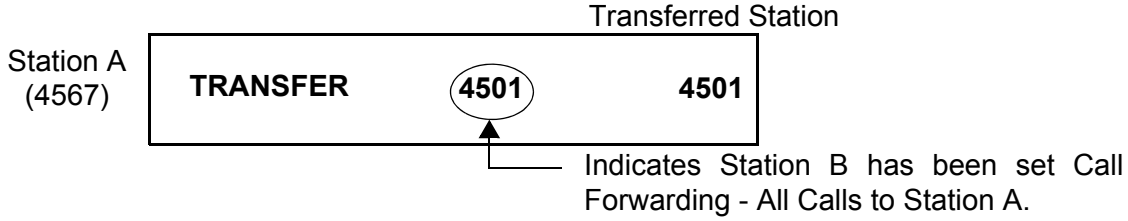
This feature allows a Target user (Station A) to call the Station (Station B) which has set CALL FORWARDING - ALL CALLS [C-5] to it. If the Called Station (Station B) has set the CALL FORWARDING - BUSY LINE [C-2] to the Calling Station (Station A), the Calling Station (Station A) hears Busy Tone and can activate CALL BACK [C-1], EXECUTIVE RIGHT-OF-WAY [E-1], or CALL WAITING - ORIGINATING [C-31].

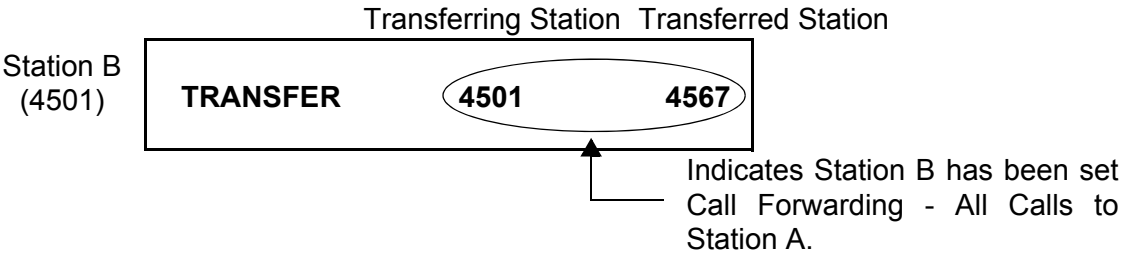
Operating Procedure

Station A presses EXECUTIVE RIGHT-OF-WAY [E-1] or CALL WAITING [C-31] after hearing Busy Tone.

Service Conditions

1. This feature is allowed to all Stations in the System, except the ATTENDANT CONSOLE [A-3].
2. Any Station that has set CALL FORWARDING - ALL CALLS [C-5] can activate any feature, if the Station is allowed the feature via CLASS OF SERVICE [C-15].
3. When Station A (Desktop terminal) calls Station B (Desktop terminal), or if Station A places an Incoming Call into hold condition and calls Station B, each Dterm LCD displays as shown below:





Programming

ASFC - Assign Service Feature Restriction Class that allows SFI 156 (Call Forwarding - Override Indication) to the Dterm Stations allowed LCD indication of this service.

C-27D Call Waiting - Answer -D

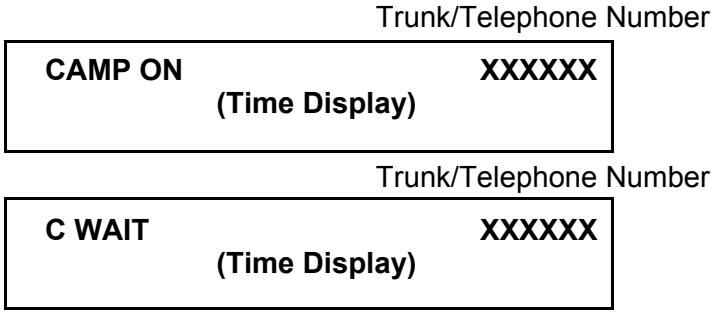
General Description

This feature allows a Desktop terminal user to answer a CAMPED-ON [A-1] or CALL WAITING [C-12, 31] Call while putting an existing call on HOLD [C-6], by pressing the ANSWER key.

Operating Procedure

CAMP-ON [A-1] or CALL WAITING [C-12, 31] to a busy Desktop terminal is activated:

1. The user hears camp-on indication Tone or Call Waiting Tone during a call in progress. The LCD of the Desktop terminal displays:



2. The user presses the ANSWER key and is connected to the Waiting Call.
3. To return to the original call, the Desktop terminal user re-presses the ANSWER key.

Service Conditions

1. Service conditions are essentially the same as those of CALL WAITING - TERMINATING [C-12] for ordinary Telephone Sets.
2. Pressing RECALL will disconnect the party presently engaged in conversation.

Interactions

1. This feature is allowed whenever the Destination Telephone is engaged in any state other than:
   a.) engaged in a Three-way Connection,
   b.) a Privacy feature is activated,
   c.) connected to a Route programmed to have Line/Trunk, Privacy, CDN 44:PRV,
   d.) listening to a Tone (Dial Tone, Reorder Tone, Service Set Tone, Busy Tone, Ringback Tone).
2. CALL WAITING - ANSWER -D is operated by pressing the ANSWER key located on the Desktop terminal instrument. This feature is used for ATTENDANT CAMP-ON [A-1] and CALL WAITING - ORIGINATING [C-31].
3. Periodic Call Waiting Tones will not operate for ATTENDANT CAMP-ON WITH TONE INDICATION [A-1].
4. This feature cannot be set from SIP Handler Controlled SIP terminals. Also this feature cannot be set from different terminals to SIP Handler Controlled SIP terminals.

Programming

STEP 1: **ASYD** - SYS1, Index 152. Timer Class 3 = Two seconds. MTC×TC = Interval for CALL WAITING.

MTC assigned in hexadecimal to Bit 0 to Bit 3 of this index. Default value is ten seconds.

SYS2, Index 10, Bit 6. Should CALL WAITING - ORIGINATING [C-31] operation be automatic upon encountering a busy Station? 0/1: No/Yes. Assign on a per Tenant basis.

SYS2, Index 10, Bit 7. Should the Warning Tone set in SYS1, Index 152 be periodically sent? 0/1: No/Yes. Assign on a per Tenant basis.

STEP 2: **ASFC** - Assign a Service Feature Restriction Class that allows SFIs 5 [Call Waiting - Originating (Calling)], 6 [Call Waiting - Originating (Called)], and 11 [Data Line Privacy = 0/Attendant Camp-On = 1). This allows the ability to originate a CALL WAITING - ORIGINATE (SFI 5), to terminate a CALL WAITING - ORIGINATING [C-31] (SFI 6), and to receive Camp-On Tone (SFI 11).

Commands related to CALL WAITING - ORIGINATING [C-31]

STEP 3: **ANPD** - Reserve a Number Level for Feature Access. Select Normal Service (N), Hooking Service (H), or Busy Service (B) from the combo box for Connection Indexes. Assign NND in accordance with a predetermined Numbering Plan.

STEP 4: **ASPA** - Assign an Access Code to CALL WAITING - ORIGINATING [C-31], SRV = "SSC: Service Code", SID = 5 for Connection Status Index (CI) of "B: Busy". Assign the NND value as equal to the digits in the Feature Access Code.

FCCS Programming

This data setting describes only the additional or replacement commands used to activate FCCS Services. When used as an FCCS feature, these commands replace the non-network commands (for example, AAED is replaced by AAEDN). Note that the data setting for FCCS Service varies depending on the Node.

Note: When using ASYDL or ASYDN to program an FCCS feature, the basic programming of ASYD is required in addition to ASYDL or ASYDN.

STEP 1: **ANPDL** - Reserve a Number Level for Feature Access. Select Normal Service (N), Hooking Service (H), or Busy Service (B) from the combo box for Connection Indexes. Assign NND in accordance with a predetermined Numbering Plan.

STEP 2: **ASPAL** - Assign an Access Code to CALL WAITING - ORIGINATING [C-31], SRV = "SSC: Service Code", SID = 5 for Connection Status Index (CI) of "B: Busy". Assign the NND value as equal to the digits in the Feature Access Code.

Note: System Timers may be set on a network basis in ASYDN.

C-28 Call Forwarding - All Calls - Outside

General Description

This feature allows a call, which originates from a Station or Trunk and is destined for a Station, to be routed to another Station outside the System via the appropriate Trunk Line. This feature can also be activated from a Desktop terminal Multi Line.

Operating Procedure

To set and cancel from an individual Station using the same Access Code Note 1

- For setting:

1. Lift the handset. Receive Dial Tone.
2. Dial the CALL FORWARDING - ALL CALLS - OUTSIDE entry Access Code (same as CALL FORWARDING - ALL CALLS [C-5]) (**Note 2**). Receive Special Dial Tone.
3. Dial the desired Forwarding Destination Number. Receive Service Set Tone.

- For cancellation:

1. Lift the handset. Receive Dial Tone.
2. Dial the CALL FORWARDING - ALL CALLS - OUTSIDE entry Access Code (same as CALL FORWARDING - ALL CALLS [C-5]). Receive Special Dial Tone.

Note 1: Available from FP95-114 V4 Issue 2.0 with system data ASYDL, SYS1, Index 686, Bit 0= 1.

Note 2: A tone will not be provided when an access code is used to set this feature on SP Controlled Standard SIP terminal.

To set and cancel from an individual Station with different Access Codes

- For setting (**Note 3**):

1. Lift the handset. Receive Dial Tone.
2. Dial the Call Forwarding - All Calls - Outside entry Access Code (same as CALL FORWARDING - ALL CALLS [C-5]). Receive Special Dial Tone.
3. Dial the desired Forwarding Destination Number (or Speed Calling Access Code and Abbreviated Code). If the ATTENDANT CONSOLE [A-3] is setting this feature, dial the entry Access Code, Tenant Number, Originating Telephone Number, and then the desired Outside Number. Receive Service Set Tone.

Note 3: A tone will not be provided when an access code is used to set this feature on SP Controlled Standard SIP terminal.

- For canceling:

1. Lift the handset. Receive Dial Tone.
2. Dial the Call Forwarding - All Calls - Outside cancel Access Code, (same as CALL FORWARDING - ALL CALLS [C-5] Cancel Code). Receive Service Set Tone.

To set from a Multi Line

1. Press a Line/Feature key to be used for Call Forwarding - All Calls - Outside.
2. Lift the handset or press the SPEAKER key. Receive Dial Tone.
3. Press the CALL FORWARDING -ALL CALLS - OUTSIDE Feature key. Receive Special Dial Tone.
4. Dial the desired Forwarding Destination Number (or Speed Calling Access Code and Abbreviated Code). Receive Service Set Tone. The Call Forwarding - All Calls FKY LED will light on the phone of the station that this feature has been set. No lamp indication will be given on the multi-line of the setting phone once this feature is set. The LCD displays:

FORWARD SET **(Time Display)**

5. Replace the handset or press the SPEAKER key.

To monitor from a Multi Line

1. Press the Multi Line Appearance, then press the CALL FOWARDING -ALL CALLS Feature key. The LCD displays:

Outside Number*

FORWARD	**XXXXXXXX**
(Time Display)	

Note: *A dialed Abbreviated Number is transferred into the Forwarding Number, and the first 16 digits are displayed on LCD, even if a maximum of 24 digits can be stored.

To cancel from a Multi Line

1. Lift the handset or press the SPEAKER key. Receive Dial Tone. Press the CALL FOWARDING -ALL CALLS Feature key. Receive Service Set Tone. The LED of the associated Feature key will go out. The LCD displays:

FORWARD CANCEL **(Time Display)**

2. Replace the handset or press the SPEAKER key.

Service Conditions

1. Call Forwarding - All Calls - Outside may be set or cancelled by either a user or the Attendant.
2. Stations may be assigned this feature via CLASS OF SERVICE - INDIVIDUAL [C-15] programming in System Data.
3. For Call Forwarding - All Calls - Outside, Speed Call List Numbers may be programmed as the Destination Telephone Number. This permits the sent digits to exceed the 12-digit maximum.
4. When using this feature, either the Calling Station or the Forwarding Station is recorded by STATION MESSAGE DETAIL RECORDING (SMDR) [S-10] as the Originating Station of calls forwarded. The selection between the above Stations is decided by System Data.
5. The Calling Desktop terminal display for Call Forwarding and Busy Line - Outside appear as regular Outgoing Central Office Calls. The Calling Desktop terminal display for Call Forwarding Don't Answer - Outside appears as a STATION-TO-STATION CALLING - CCIS [S-57] Call.

6. Call Forwarding - All Calls - Outside can be restricted by the Trunk Route Restriction Class (RSC) of the Originating Station.
7. There is no limit to the number of Stations that may activate this feature simultaneously.
8. Restrictions for Outgoing Connection by Call Forwarding - All Calls - Outside are as follows:
   a.) When originating a call from a Station: RSC is verified when this service is in effect.
   b.) When originating a call from a Trunk: Restriction for Tandem Connections between the incoming trunk and the outgoing trunk is selected for the call where Call Forwarding - All Calls service occurs.
9. When a caller places a call via OGCA and when the first Outgoing Route and the Calling Party are restricted, this service cannot be set. In this case, the service can be set via the following procedure:
   a.) assign a Dummy Route,
   b.) clear the restriction between the Dummy Route and all the Stations,
   c.) assign the Dummy Route as the first Route of OGCA.
10. When a destination of Call Forwarding - All Calls - Outside has already been assigned, even if the feature is restricted by the new Office Data or changing of Day/Night mode, the assigned data can still be used or canceled.
11. A maximum of 24 digits, including Trunk Access Code, can be stored.
12. This feature allows a call to terminate to a Trunk outside of the FCCS Network. Accordingly, the Access Code + Office Code or the Access Code including Office Code is required before dialing the Telephone Number to recognize the Terminating Node.
13. When the Incoming Call is forwarding to the Public Network, be sure to add the Outgoing Access Code + Destination Area Code or the Outgoing Access Code including Destination Area Code Number before dialing Subscriber Number.
14. A call can be forwarded to a Trunk in the Remote Node via FCCS.
15. Assign the System Data (ASYDL, SYS1, Index 677, Bit 3=1), and press the Call Forwarding - All Calls - Outside Feature key, a maximum of 16 digits Call Forwarding Information can be displayed.

    Note: This feature is available only for the terminal that can display 24 digits on LCD.

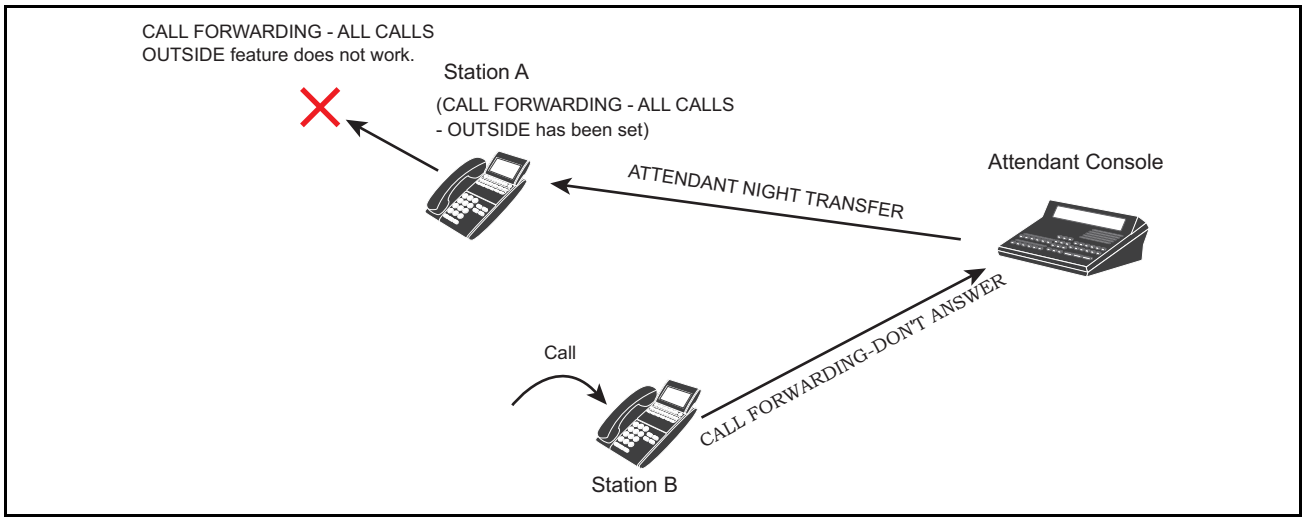
16. By registering the Call Forwarding - All Calls Access Code - Outside + the desired Forwarding Destination Number to the CF Setting button of a Line/Feature key on a Desktop terminal, it is possible to enable and disable CALL FORWARDING - ALL CALLS - OUTSIDE [C-28] just by pressing the CF Setting button. For a detailed explanation, see CALL FORWARDING (CF) BUTTON - ONE TOUCH [C-208].

    Note: CALL FORWARDING (CF) BUTTON - ONE TOUCH [C-208] is available from FP95-114 V4 Issue 2.0.

Interactions

1. A 24-digit limitation exists in memory for Call Forwarding - Outside.
2. Call Forwarding - Outside is an enhancement of the Call Forwarding - All Calls feature.
3. This feature may be set from the Attendant Console.
4. A call may be Call Forwarded - Outside to a SPEED CALLING - SYSTEM [S-3] or SPEED CALLING - STATION/GROUP [S-21, 23] Number. This will allow a Long-distance Number that exceeds 12 digits.
5. A user can verify that Call Forwarding - Outside has been correctly activated and can inform the designated Outside Party of the activation by dialing the Forwarded Station from any Station in the system.

6. If a Station is restricted from outside calling, Call Forwarding - Outside will be denied. TOLL RESTRICTION - 3/6 DIGIT [T-7] will be applied to Call Forwarding - Outside via SPEED CALLING - SYSTEM, STATION, GROUP [S-3, 21, 23], if Tenant Data assigned in System Data 2 applies Toll Restriction to Speed Calling.
7. Upon activation of this feature, STATION MESSAGE DETAIL RECORDING (SMDR) [S-10] will record the Outside Exchange Number whenever a call is forwarded.
8. A Station can use the OUTGOING TRUNK QUEUING [O-2] feature to queue an outgoing trunk and a call that is being forwarded via the Call Forwarding - Outside feature. The call will be outpulsed when a Trunk becomes available.
9. A Station assigned to a STATION HUNTING [S-7, 8, 9] Group will be temporarily removed from the Hunt Group if it initiates Call Forwarding - Outside.
10. If a call using the CALL BACK [C-1] feature is placed from a Station that has activated Call Forwarding - Outside, the Call Back will ring at the Forwarding Station, not the outside Station.
11. When OUTGOING TRUNK QUEUING [O-2] (on-hook) is initiated by the Forwarding Station, the ring back will be to the Forwarding Station and not to the outside Station.
12. A DIRECT-IN TERMINATION [D-7] call may be sent through Call Forwarding - Outside.
13. The number of digits (maximum 12) that will be accepted before forwarding is set is determined by AMND.
14. This feature allows a call to terminate to a Trunk outside of the FCCS Network. Accordingly, “the Access Code + Office Code” or “the Access Code including Office Code” is required before dialing a Telephone Number to recognize the Terminating Node.
15. When forwarding the Incoming Call to the Public Network, be sure to add the Outgoing Access Code + Destination Area Code or the Outgoing Access Code including Destination Area Code Number before dialing a Subscriber Number.
16. When CALL FORWARDING - ALL CALLS - OUTSIDE [C-28] is set to Station A (Destination Station for ATTENDANT NIGHT TRANSFER [A-19]), the following call routing is not available.
    a.) Station B has set CALL FORWARDING - DON’T ANSWER [C-3] to the Attendant Console.
    b.) A call terminates to Station B when the system is in Night mode.
    c.) Station B does not answer the call. It is transferred to the Attendant Console after a predetermined period of time.
    d.) The Attendant Console transfers the call to Station A (Destination Station for ATTENDANT NIGHT TRANSFER [A-19]).
    e.) Although Station A has set CALL FORWARDING - ALL CALLS - OUTSIDE [C-28], the call cannot be transferred to the outside party.



17. A tone will not be provided when an access code is used to set this feature on SP Controlled Standard SIP terminal.
18. When a SIP Handler Controlled SIP terminal is in Multi Line Mode, the Call Forwarding setting applies to the seized line while setting. In normal case, where My Line is not used by another station, the Call Forwarding setting applies to My Line. To set the Call Forwarding applying to Sub Line, use the ACFS/ ACFS_T commands.

Programming

STEP 1: **ASYD** - SYS1, Index 4, Bit 6. One burst of ringing all CALL FORWARDING - ALL CALLS [C-5] in service = 0/1: No/Yes. SYS1, Index 20, Bit 7. Calling or Forwarded Party shows on SMDR = 0/1: Calling/Forwarded.

SYS2, Index 6, Bit 4. Call-forwarded Station restricted from making calls = 0/1: Yes/No.

STEP 2: **ASYDL** - SYS1, Index 677, Bit 3.
Display a maximum of 16 digits Call Forwarding Information on LCD? 0/1 = No (default)/Yes.
SYS1, Index 686, Bit 0=1. **Note 4**
Disabling Call Forwarding - All Calls with entry Access Code is in service.

STEP 3: **ASYDN** - SYS1, Index 686, Bit 0=1. **Note 4**
Disabling Call Forwarding - All Calls with entry Access Code is in service.

STEP 4: **ANPD** - Reserve a Number Level for Feature Access and Cancel (**Note 5**). Select Normal Service (N), Hooking Service (H), or Busy Service (B) from the combo box for Connection Indexes. Assign NND in accordance with a predetermined Numbering Plan.

STEP 5: **ASPA** - Assign an Access Code to CALL FORWARDING -ALL CALLS [C-5] set SID = 8, and to CALL FORWARDING -ALL CALLS [C-5] cancel SID = 9 (**Note 5**). Assign Connection Status Index (CI) for "N: Normal" service.

Note 4: Available from FP95-114 V4 Issue 2.0.

Note 5: When SYS1, Index 686, Bit 0=1 (Disabling Call Forwarding - All Calls with entry Access Code is in service), Call Forwarding - All calls cancel is not required because the cancellation can be performed with the entry Access Code.

STEP 6: **ASFC** - Assign the Station a Service Feature Restriction Class that allows SFI 7 (Call Forwarding - All Calls). And also assign either SFI 25 (Call Forward Outside - Local) or SFI 26 (Call Forward Outside - Long Distance). Assign SFI 151 (Call Forwarding - Outside) when Call Forwarding - Outside is performed, RSC of the Forwarding Station is used.

Note: SFI 25 checks AMND for 11 digits or less (Local Service). SFI 26 checks AMND for 12 digits or more (Long Distance Service).

Note: SFI 151 (Call Forwarding - Outside) is not available when the System Data (ASYDL/ASYDN, SYS1, Index 1393, Bit 5) is set to 1.

STEP 7: **ARRC** - Allow trunk-to-trunk connection between an incoming route and an outgoing route that may be used for this feature. The incoming trunk includes calls extended by the Attendant Console and DID Calls. The outgoing trunk would be DDD or Trunks seized when using SPEED CALLING - GROUP [S-23] or SPEED CALLING - SYSTEM [S-3]. If LCR is used, allow trunk-to-trunk connection to the Outgoing LCR Flag Route, usually Route 31.

To forward tie line calls, you need to allow trunk-to-trunk connection between tie lines by setting the following properties:

IC RT: External route number of a tie line for incoming calls

OG RT: External route number of a tie line for outgoing calls

ARI A-RES = 1: Alternate routing is Allowed

ARI D-RES = 1: Alternate routing is Allowed

STEP 8: **AKYD** - For Dterm sets, Call Forwarding - Outside may be assigned to a Line/Feature key. Assign Feature key Call Forwarding - All Calls [2].

STEP 9: **ADSL** - Assign SN = 1 (Originating; DT Connection) and Feature key Call Forwarding - All Calls [2].

STEP 10: **ATNR** - Allow Tenant-to-Tenant Connection for Inter and Intra-Tenant Connections. TRI = 1. For this feature to be set by the Attendant Console, assign Inter and Intra-Tenant Connection via the Attendant Console, TRI = 3.

STEP 11: **AMND** - Local Outgoing Destination Codes (DC) must be programmed to allow Local Outgoing Calls such as 92, 93, 94, etc.

STEP 12: **ACFS** - Call Forwarding can also be assigned, displayed, or deleted by this command.

FCCS Programming

This data setting describes only the additional or replacement commands used to activate FCCS Services. When used as an FCCS feature, these commands replace the non-network commands (for example, AAED is replaced by AAEDN). Note that the data setting for FCCS Service varies depending on the Node.

Note: When using ASYDL or ASYDN to program an FCCS feature, the basic programming of ASYD is required in addition to ASYDL or ASYDN.

STEP 1: **ANPDL** - Reserve a Number Level for Feature Access and Cancel. Select Normal Service (N), Hooking Service (H), or Busy Service (B) from the combo box for Connection Indexes. Assign NND in accordance with a predetermined Numbering Plan.

STEP 2: **ASPAL** - Assign an Access Code to CALL FORWARDING -ALL CALLS [C-5] set SID = 8, and to CALL FORWARDING -ALL CALLS [C-5] cancel SID = 9. Assign Connection Status Index (CI) for "N: Normal" service.

STEP 3: **ARRCN** - Allow trunk-to-trunk connection between an incoming trunk and an outgoing trunk that will be involved with this application. The incoming trunk includes calls extended by the Attendant Console and DID Calls. The outgoing trunk would be DDD or Trunks seized when using SPEED CALLING - GROUP [S-23] or SPEED CALLING - SYSTEM [S-3]. If LCR is used, allow trunk-to-trunk connection to the Outgoing LCR Flag Route, usually Route 31.

Note: System Timers may be set on a network basis in ASYDN.

C-29 Call Park

General Description

This feature enables the Attendants or Station Users to "Park" calls against their own Extension Numbers. Calls can easily be retrieved from any Station within the system.

Operating Procedure

To Park a call from the ATTENDANT CONSOLE [A-3]

1. The Attendant presses the CALL PARK Feature key. The CALL PARK Number (Individual Attendant Number) is automatically selected and displayed at the ATTENDANT CONSOLE [A-3].
2. The Attendant receives Service Set Tone.
3. The Attendant presses either the RELEASE or CANCEL key.

To retrieve a Parked Call from any Station within the system

1. Dial the CALL PARK Remote Retrieval Code and the Individual Attendant Number that has Parked a call. The Parked Call is reconnected.

Note: An attendant cannot retrieve a Parked Call.

To Park a call from a Station

1. Momentarily press the switch hook. Receive Special Dial Tone.
2. Dial the CALL PARK Access Code. Receive Service Set Tone.
3. Replace the handset.

To retrieve a Parked Call from the Originating Station

Dial the CALL PARK Local Retrieval Code. The Parked Call is reconnected.

To retrieve a Parked Call from a Different Station

Dial the CALL PARK Remote Retrieval Code and the number of the Station that Parked the call. The Parked Call is reconnected.

Service Conditions

1. Calls Parked by a Station are automatically Parked on that Station's Extension Number. The My Line Number is used for a Dterm.
2. CALL PARK may be activated only during calls involving two parties.
3. The system provides one Attendant CALL PARK Number for each Attendant. The Individual Attendant Number is the CALL PARK Number. See INDIVIDUAL ATTENDANT ACCESS [I-6].
4. A Station or Attendant can only Park one call at a time. If a Station or Attendant attempts to Park a second call, Reorder Tone will be heard.
5. This feature may be assigned to Stations via CLASS OF SERVICE - INDIVIDUAL [C-15] programming in System Data.
6. The dedicated key (for Parking calls) on the Attendant Console is located between the HOLD and Busy Verification (BV) keys. An Attendant cannot use a CALL PARK Access Code.
7. A user can Park a call only by dialing a CALL PARK Access Code.
8. Stations can Originate and Receive Calls while having a call Parked.

9. The following features cannot be activated to a Station that has set CALL PARK:

ATTENDANT CAMP-ON WITH TONE INDICATION	[A-1]
ATTENDANT OVERRIDE	[A-7]
BUSY VERIFICATION	[B-3]
CALL WAITING - ORIGINATING	[C-31]
CALL WAITING - TERMINATING	[C-12]
EXECUTIVE RIGHT-OF-WAY	[E-1]

10. A Station in an ATTENDANT-CONTROLLED CONFERENCE [A-2] cannot be Parked.
11. There is no limit to the number of calls that can be Parked simultaneously within a system.
12. A Parked Call will recall to the Station that has set the CALL PARK. The Recall Timer is preprogrammed into System Data. A call Parked by the Attendant will not recall.
13. CALL PARK is denied for the following Station status:

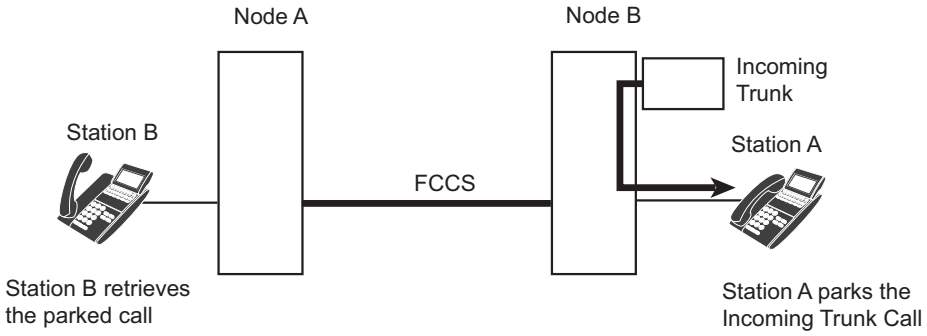
ATTENDANT OVERRIDE	[A-7]
BOSS - SECRETARY OVERRIDE -D	[B-5D]
BUSY VERIFICATION	[B-3, B-9]
CALL TRANSFER	[C-10, C-11]
CALL WAITING - ORIGINATING	[C-31]
CALL WAITING - TERMINATING	[C-12]
CONSULTATION HOLD - ALL CALLS	[C-17, C-58]
DATA LINE SECURITY	[D-12, D-66]
DIAL ACCESS TO ATTENDANT	[D-2, D-71]
EXCLUSIVE HOLD -D	[E-4D]
EXECUTIVE RIGHT-OF-WAY	[E-1, E-8]
NON-EXCLUSIVE HOLD -D	[N-7D]
PRIVACY RELEASE	[P-18]
THREE-WAY CALLING	[T-2]
VOICE CALL	[V-2, V-7]

14. CALL PARK is denied for the following Attendant Console status:

ATTENDANT OVERRIDE	[A-7]
BUSY VERIFICATION	[B-3]
CALL TRANSFER - ATTENDANT	[C-10]
Call on Hold	
THREE-WAY CALLING	[T-2]

15. An Attendant Console cannot be Parked.
16. A CALL PARK Connection remains Parked until retrieved or until the party disconnects.
17. A Parked Call can only be retrieved from an Idle Station.
18. An ISDN Terminal User cannot use this feature.
19. An Incoming Call from the ISDN Trunk can be Parked from a Station/Attendant.
20. A Parked Call by an Attendant can be retrieved from a Station by dialing the CALL PARK Retrieval Code and the Individual Attendant Number. Attendant can Park a call only when the call is on SRC side.
21. This feature is available via FCCS Network.

For example, an Incoming Trunk Call is parked from Station A, the call parked by Station A is retrieved from Station B via FCCS.



22. A Parked Call cannot be retrieved from a Station in the Remote Node when all FCCS Links are busy.
23. Feature data may be programmed on either the Local Data Memory (LDM) or the Network Data Memory (NDM).
24. The Night Attendant Console cannot Park a call.
25. A Parked Call will recall to the Station that has set the CALL PARK after a predetermined period of time assigned by the system data (ASYD, SYS1, Index 142) has elapsed (the default value is 500 seconds). If the recalling parked call is from an analog C.O line (except MC&MG-COT/MG-COT), it will be released after a predetermined period of time assigned by the system data (ASYD, SYS1, Index 139) has elapsed (the default value is 30 seconds). As for other than the analog C.O line (except MC&MG-COT/MG-COT), the recalling will be continued.
26. A Parked Call by the Desktop terminal is always retrieved by using the My Line of the Desktop terminal. Even if a Desktop terminal Parks a call by using the Sub Line, the Parked Call is retrieved by using My Line of the Desktop terminal.

Interactions

Only calls directly terminated to the Attendant Console can be Parked. This includes calls transferred to the Attendant from another Station or Attendant Console.

Programming

STEP 1: **ASYD** - SYS1, Index 139, Bit 0 to Bit 3 (MTC), Bit 4-6 (TC). Assign No Answer Timer for CALL PARK recall ringing. If a Parked Call from an analog C.O line (except MC&MG-COT/MG-COT) recalls and the set timer value (default is 30 seconds) elapses, the call will be released. SYS1, Index 142. For CALL PARK Recall, the Miscellaneous Timer Counter is assigned in Bit 0 to Bit 3. The Timer Class is assigned in Bit 4 to Bit 6.

YS2, Index 1, Bit 0 and Bit 1. CONSULTATION HOLD - ALL CALLS [C-17] allowed or denied. 00: denied; 01: Originating and Terminating Calls allowed, Tandem Call denied; 11: All calls allowed.

STEP 2: **ANPD** - Reserve a Number Level for Feature Access and Retrieval. Select Normal Service (N), Hooking Service (H), or Busy Service (B) from the combo box for Connection Indexes. Assign NND in accordance with a predetermined Numbering Plan.

STEP 3: **ASPA** - Assign an Access Code to:

CALL PARK Access Code, SRV = "SSC: Service Code", SID = 61, Connection Index of Hooking, CI = "H: Hooking".

CALL PARK Local Retrieval Code, SRV = "SSC: Service Code", SID = 62, Connection Index of "N: Normal".

CALL PARK Remote Retrieval Code, SRV = "SSC: Service Code", SID = 63, NND = number of digits in the Access Code, and assign a Connection Index of "N: Normal".

STEP 4: **ASFC** - Assign a Service Feature Restriction Class that allows SFI 67 (Call Park - Access and Answer) to the applicable Stations. This allows the Station to Park a call and to retrieve a Parked Call. Assign Stations a Service Feature Restriction Class that allows SFI 68 (Call Park Called). This allows a Station to be Parked by another Station or Attendant Console.

STEP 5: **ASAT** - This command must be assigned to designate a specific Station Number to the Attendant Console. This number is used to identify the Attendant Console when Parking a call.

FCCS Programming

This data setting describes only the additional or replacement commands used to activate FCCS Services. When used as an FCCS feature, these commands replace the non-network commands (for example, AAED is replaced by AAEDN). Note that the data setting for FCCS Service varies depending on the Node.

Note: When using ASYDL or ASYDN to program an FCCS feature, the basic programming of ASYD is required in addition to ASYDL or ASYDN.

STEP 1: **ANPDL** - Reserve a Number Level for Feature Access and Retrieval. Select Normal Service (N), Hooking Service (H), or Busy Service (B) from the combo box for Connection Indexes. Assign NND in accordance with a predetermined Numbering Plan.

STEP 2: **ASPAL** - Assign an Access Code to:

CALL PARK Access Code, SRV = "SSC: Service Code", SID = 61, FPC = "x", Connection Index of Hooking, CI = "H: Hooking".

CALL PARK Local Retrieval Code, SRV = "SSC: Service Code", SID = 62, FPC = "x", Connection Index of "N: Normal".

CALL PARK Remote Retrieval Code, SRV = "SSC: Service Code", SID = 63, FPC = "x", NND = number of digits in the Access Code, and assign a Connection Index of "N: Normal".

STEP 3: **ASATN** - This command must be assigned to designate a specific Station Number to the Attendant Console. This number is used to identify the Attendant Console when Parking a call.

Note: System timers may be set on a network basis in ASYDN.

The following command assignments are required when the Parked Call is retrieved from a Station in the Remote Node via FCCS.

STEP 1: **ASYD** - SYS1, Index 139, Bit 0 to Bit 3 (MTC), Bit 4 to Bit 6 (TC). Assign No Answer Timer for CALL PARK recall ringing. If a Parked Call from an analog C.O line (except MC&MG-COT/MG-COT) recalls and the set timer value (default is 30 seconds) elapses, the call will be released. SYS1, Index 142, Bit 0 to Bit 3. Assign the Miscellaneous Timer Counter for CALL PARK recall.

If the set Timer Value elapses, a Parked Call will recall to the Station that has set the CALL PARK.

The Timer Class is assigned in Bit 4 to Bit 6. SYS1, Index 6, Bit 4 to Bit 6. This data assignment is required to Park a call from the Attendant.

Bit 4 to Bit 6 (ATTCON Key Arrangement Pattern): Bit 4=0, Bit 5=0, Bit 6=0

STEP 2: **ANPDL/ANPDN** - Reserve a Number Level for feature access. Assign the following data in accordance with the predetermined Numbering Plan.

TN: Tenant Number
1stDC: 1st Digit Code (0-9,*,#)
CI: Connection Status Index (N = Normal, H = Hooking)
NND: Number of Necessary Digits (1-6)
BLF: Busy Lamp Field Display Activated On ATT (Fixed to "0")

STEP 3: **ASPAL/ASPAN** - Assign an Access Code:

CALL PARK Access Code, SRV = "SSC: Service Code", SID = 61, Connection Index of Hooking, CI = "H: Hooking"

CALL PARK Local Retrieval Code, SRV = "SSC: Service Code", SID = 62, Connection Index of "N: Normal.

CALL PARK Remote Retrieval Code, SRV = "SSC: Service Code", SID = 63, NND = Number of Digits in the Access Code, and assign a Connection Index of "N: Normal".

STEP 4: **ALGSL/ALGSN** - Assign Telephone Number to the Station Number programmed by ASDT.

TYPE = 1 (Line Equipment Numbers/Phantom Index is used to specify the Telephone Number)
UGN: User Group Number (1-255)
TELN: Telephone Number (maximum of 16 digits)
FPC: Fusion Point Code (only ALGSN)
LENS/PHIN: Line Equipment Numbers/Phantom Index of the Station Number
TYPE = 2 (Station Number is used to specify the Telephone Number)
UGN: User Group Number (1-255)
TELN: Telephone Number (maximum of 16 digits)
FPC: Fusion Point Code (only ALGSN)
TN: Tenant Number
STN: Station Number of destination

STEP 5: **ASFC** - The same data must be assigned to all the nodes.

Assign a Service Feature Restriction Class that allows SFI 67 (Call Park - Access and Answer) to the applicable Stations. This allows the Station to Park a call and to retrieve a Parked Call. Assign a Service Feature Restriction Class that allows SFI 68 (Call Park - Called) to the applicable Stations. This allows a Station to be Parked by another Station or Attendant Console.

C-30 Call Pickup - Direct

General Description

This feature allows a user to pick up a call to any other Station in the system by dialing a specific CALL PICKUP - DIRECT code.

Operating Procedure

1. Lift the handset. Receive Dial Tone.
2. Dial the CALL PICKUP - DIRECT Code. Receive Dial Tone.
3. Dial the specific Telephone Number to be picked up. Incoming Call is connected to user Station.

Service Conditions

1. This feature is provided on a per line basis, based on the Station CLASS OF SERVICE - INDIVIDUAL [C-15].
2. A Station in a fully restricted class cannot answer a Central Office Incoming Call using CALL PICKUP - DIRECT.
3. If the ringing Call is a result of CALL BACK [C-1], CALL HOLD [C-6] Recall, or OUTGOING TRUNK QUEUING [O-2] Recall, it cannot be picked up by any Station.
4. This feature cannot be accessed from the ATTENDANT CONSOLE [A-3].
5. The Stations in this procedure can belong to the different TENANT SERVICE [T-12] Groups. A TN 2 can pick up a TN 1 call.
6. A busy Station can accomplish CALL PICKUP - DIRECT by placing an existing call on CALL HOLD [C-6].
7. CALL PICKUP - DIRECT cannot be accessed while in the CONSULTATION HOLD - ALL CALLS [C-17] mode.
8. CALL PICKUP - ONE TOUCH KEY [C-207] is available in combination with this feature.
9. When a Station holds a call (CALL HOLD) terminated through FCCS, the Station cannot pick up a call.
10. When both the Calling Station and the Station that will pick up a Call Hold the call, if one of the Held Calls is terminated through FCCS, the Station cannot pick up the call.
11. The call terminated through FCCS cannot be picked up from the other Station in the system by dialing the Physical Station Number.
12. PS can pick up a call arrived in a specific Station.
13. A call which has terminated to a PS/ISDN terminal cannot be answered using this feature (the caller hears busy tone). However, if PS line is assigned as a Multi Line of Desktop terminal (APLAML/APLAMN), calls terminating to the corresponding PS can be answered using this feature.
14. You can answer a call ringing on a virtual line of another station. Also, you can answer a call by picking it up from a virtual line.

Interactions

1. After answering via CALL PICKUP - DIRECT, the CALL TRANSFER - ALL CALLS [C-11] feature can be used.

2. A STATION MESSAGE DETAIL RECORDING (SMDR) [S-10]: If a billing after transfer over different MA-IDs is out of service (ASYDL, SYS1, Index 589, Bit 3=0), when a call is answered using the CALL PICKUP - DIRECT feature, the SMDR record is not created.
   If a billing after transfer over different MA-IDs is in service (ASYDL, SYS1, Index 589, Bit 3=1), the SMDR record is created. The number of a station using this feature is output to the SMDR record as a called party number.
3. The CALL PICKUP - DIRECT feature cannot be used to answer a call from the paged individual through Paging Answer Access Code in the Delay System of PAGING TRANSFER [P-10].
4. You can pick up an incoming call which arrives to Call Pickup Group of CALL PICKUP - ONE TOUCH KEY [C-207].

Note: Available since FP95-115 V5.

Programming

STEP 1: **ASYD** - SYS2, Index 1, Bit 0 and Bit 1. CONSULTATION HOLD - ALL CALLS [C-17] allowed or denied. 00: denied; 01: Originating and Terminating Calls allowed, Tandem Call denied; 11: All calls allowed.

STEP 2: **ASYDL/ASYDN** - SYS1, Index 685, Bit 5=0/1 (Picking up an incoming call which arrives to Call Pickup Group of Call Pickup - Direct [C-30] = Disabled/Enabled)

Note: Available since FP95-115 V5.

STEP 3: **ANPD** - Reserve a Number Level for Feature Access. Select Normal Service (N), Hooking Service (H), or Busy Service (B) from the combo box for Connection Indexes.

STEP 4: **ASPA** - Assign an Access Code to CALL PICKUP - DIRECT, SRV = “SSC: Service Code”, SID = 35 for Connection Status Index (CI) of “N: Normal”.

STEP 5: **ASFC** - Assign a Service Feature Restriction Class that allows SFI 29 (Direct Call Pickup) to the Stations.

FCCS Programming

This data setting describes only the additional or replacement commands used to activate FCCS Services. When used as an FCCS feature, these commands replace the non-network commands (for example, AAED is replaced by AAEDN). Note that the data setting for FCCS Service varies depending on the Node.

Note: When using ASYDL or ASYDN to program an FCCS feature, the basic programming of ASYD is required in addition to ASYDL or ASYDN.

STEP 1: **ANPDL** - Reserve a Number Level for Feature Access. Select Normal Service (N), Hooking Service (H), or Busy Service (B) from the combo box for Connection Indexes.

STEP 2: **ASPAL** - Assign an Access Code to CALL PICKUP - DIRECT, SRV = “SSC: Service Code”, SID = 35 for Connection Status Index (CI) of “N: Normal”.

C-31 Call Waiting - Originating

General Description

This feature provides selected Stations with Camp-On capability to a busy Internal Station.

Operating Procedure

To activate

1. Dial the desired Telephone Number. Receive Busy Tone.
2. Switch hook flash. Receive Special Dial Tone, dial the CALL WAITING - ORIGINATING code, or press the CW key on the Desktop terminal.
3. Receive Call Waiting Ringback Tone. Call Waiting Tone (three beeps) is sent to the busy Station.

OR

1. Dial the CALL WAITING - ORIGINATING Access Code. Receive Dial Tone.
2. Dial the desired Telephone Number.
3. If the Called Station is busy, CALL WAITING - ORIGINATING is automatically set. Receive Call Waiting Ringback Tone. Call Waiting Tone (three beeps) will be sent to the busy Station.

Note: Soft key can be used for the Desktop terminal.

To answer

1. Call Waiting Tone is heard during a call in progress.
2. Switch hook flash, or press the ANSWER key on the Desktop terminal to hold the existing call.
3. The CALL WAITING - ORIGINATING Call is automatically connected.
4. Another switch hook flash or pressing the ANSWER key can be used to return to the original call, and the second call will be held.

OR

1. Call Waiting Tone is heard during the call in progress.
2. The Called Station hangs up. Priority ringing is sent.
3. Lift the handset to answer.

Service Conditions

1. Once CALL WAITING - ORIGINATING is activated to a given Station, other Stations attempting CALL WAITING - ORIGINATING to this Station will receive Reorder Tone.
2. Call Waiting Tone is three bursts, 200 milliseconds ON and 200 milliseconds OFF. The repetition rate can be programmed in System Data.
3. Periodic Call Waiting Tone can be eliminated by System Data programming on a per Tenant basis.
4. The ability to activate this feature must be assigned in the call originators CLASS OF SERVICE - INDIVIDUAL [C-15] programming.
5. CLASS OF SERVICE - INDIVIDUAL [C-15] for the Called Station must be programmed for CALL WAITING - TERMINATING [C-12].
6. There is no limit to the number of Stations that may access this feature simultaneously.
7. CALL WAITING - ORIGINATING is denied when the Called Station is connected to the ATTENDANT CONSOLE [A-3].

8. CALL WAITING - ORIGINATING can be set only if the Called Station is busy on a Two-party Call with another Station or Trunk.
9. When the Called Station goes on-hook, the Call Waiting Ringback Tone changes to Normal Ringback Tone to the Calling Station.
10. CALL WAITING - ORIGINATING cannot be activated while in CONSULTATION HOLD [C-17].
11. CALL WAITING - ORIGINATING cannot be used in conjunction with DIAL ACCESS TO ATTENDANT [D-2].
12. CALL BACK [C-1] activation is denied if a call is waiting at the Calling or Called Station Line.
13. If a Station has set CALL FORWARDING [C-2, 3, 5], a call will wait at the Target Station if CALL WAITING - TERMINATING is provided via CLASS OF SERVICE [C-15].
14. CALL WAITING - TERMINATING [C-12] at the Called Station can be restricted by CLASS OF SERVICE [C-15]. In this case, the Calling Station receives Busy Tone and can set CALL BACK [C-1].
15. ATTENDANT CAMP-ON [A-1] Tone is One-burst Tone and Call Waiting Tone from the Incoming Trunk Call is Two-burst Tones to the busy Station.
16. EXECUTIVE RIGHT-OF-WAY [E-1] is denied a Two-party Connection that has a Call Waiting.
17. STATION HUNTING [S-7, 8, 9] takes precedence over Call Waiting.
18. Priority ringing is:

0.4 seconds ON	0.2 seconds OFF	0.8 seconds ON
0.2 seconds OF	0.4 seconds ON	0.4 seconds OFF

19. These features are denied if a call is waiting at the Calling or Called Station:

ATTENDANT CAMP-ON WITH TONE INDICATION	[A-1]
ATTENDANT OVERRIDE	[A-7]
BOSS - SECRETARY OVERRIDE -D	[B-5D]
BUSY VERIFICATION	[B-3]
CALL BACK	[C-1]
CALL TRANSFER	[C-10, C-11]
CALL WAITING - TERMINATING	[C-12]
DIAL ACCESS TO ATTENDANT	[D-2]
EXCLUSIVE HOLD -D	[E-4D]
EXECUTIVE RIGHT-OF-WAY	[E-1, E-8]
NON-EXCLUSIVE HOLD -D	[N-7D]
PRIVACY RELEASE	[P-18]
THREE-WAY CALLING	[T-2]

20. CALL WAITING - ORIGINATING cannot be activated if the Target Station has accessed, entered, or is engaged in any of the following features or modes:

ATTENDANT CAMP-ON WITH TONE INDICATION	[A-1]
ATTENDANT OVERRIDE	[A-7]
BOSS - SECRETARY OVERRIDE -D	[B-5D]
BUSY VERIFICATION	[B-3, B-9]
CALL BACK	[C-1]
CALL HOLD	[C-6]
CALL TRANSFER	[C-10, C-11]
CALL WAITING - TERMINATING	[C-12]
CONSULTATION HOLD - ALL CALLS	[C-17]
DATA LINE SECURITY	[D-12]
DATA PRIVACY ON DEMAND	[D-17]
DIAL ACCESS TO ATTENDANT	[D-2, D-71]

DICTATION ACCESS	[D-3]
EXCLUSIVE HOLD -D	[E-4D]
EXECUTIVE RIGHT-OF-WAY	[E-1, E-8]
NON-EXCLUSIVE HOLD -D	[N-7D]
OFF-HOOK QUEUING	[O-7]
OUTGOING TRUNK QUEUING	[O-2]
PRIVACY -D	[P-14D]
PRIVACY RELEASE	[P-18]
THREE-WAY CALLING	[T-2]
VOICE CALL	[V-2]

21. The following features take precedence over CALL WAITING - ORIGINATING:

CALL FORWARDING - ALL CALLS	[C-5]
CALL FORWARDING - BUSY LINE	[C-3]
STATION HUNTING - CIRCULAR	[S-7]
STATION HUNTING - SECRETARIAL	[S-8]
STATION HUNTING - TERMINAL	[S-9]

22. This feature is denied while the Called Station is hearing an announcement.
23. CALL WAITING - ORIGINATING is denied if the called party connected via PHC trunk is hearing hold tone. CALL WAITING - ORIGINATING is available, however, if the called party is in tandem connection via non-PHC trunk beyond the connection via PHC trunk.
24. When Called Station activates CALL FORWARDING- BUSY LINE [C-2], and the call forwarding destination is busy:

    - When call forwarding destination is a station (including stations in FCCS network), ASYD, SYS1, Index 69, Bit 5 determines whether to set CALL WAITING - ORIGINATING to the Forwarded-from Station, or to the Forwarded-to Station.

    - When the call forwarding destination is in outside of the self office, CALL WAITING - ORIGINATING is restricted and does not work.

Interactions

1. When Station A directs the second CALL WAITING - ORIGINATING to a given Station, Station A will receive Reorder Tone.
2. A CALL WAITING - ORIGINATING Call is answered by flashing the switch hook of a Single Line Set or pressing the ANSWER or TRANSFER keys of a Desktop terminal. CALL TRANSFER - ALL CALLS [C-11] is not allowed. If the TRANSFER key on a Desktop terminal or the switch hook is flashed on a Single Line Set, the SPLITTING -D [S-4D] feature is being used.
3. CALL WAITING - ORIGINATING is denied toward a TIE Line.
4. When using this feature against a Station that is engaged in Tandem connection other than CCIS-CCIS, SFI 4 and SFI 11 require to be set to 1 for SFC 0 by using ASFC/ASFCN.
5. This feature cannot be set from SIP Handler Controlled SIP terminals. Also this feature cannot be set from different terminals to SIP Handler Controlled SIP terminals.

Programming

STEP 1: **ASYD** - SYS1, Index 69, Bit 5. CALL WAITING - ORIGINATING 0/1 = Set to Forwarded-from Station/Set to Forwarded-to Station
SYS1, Index 152. Interval for CALL WAITING - ORIGINATING. Indication Tone to be sent to the

Destination Telephone. Timer Class 3 = Two seconds. MTC × TC = Interval for CALL WAITING - ORIGINATING. MTC assigned in hexadecimal to Bit 0 to Bit 3 of this Index. Default value is ten seconds.

Example: Data 36H is assigned.

2 × 6 = 12 second intervals (TC × MTC = Timer)

SYS2, Index 10, Bit 4. Is CALL WAITING - ORIGINATING Access Code to be dialed before the Station Number? This will activate the feature automatically upon encountering a busy Station. 0/1: No/Yes.

SYS2, Index 10, Bit 6. Should CALL WAITING - ORIGINATING operation be automatic upon encountering a busy Station? If yes, no Access Code is required for operating the feature. 0/1: No/Yes. Assign on a per Tenant basis.

Note: Index 10, Bit 4 and Bit 6 are mutually exclusive. Once Bit 6 is assigned data "1", Bit 4 is not needed.

SYS2, Index 10, Bit 7. Should the Warning Tone set in SYS1, Index 152 be periodically sent? 0/1: No/Yes.

SYS3, Index 3, Bit 5. Enable Ringer Pattern 5.

SYS3, Index 7. Default data provides for four seconds ON, two seconds OFF for DISTINCTIVE RINGING [D-10].

STEP 2: **ANPD** - Reserve a Number Level for Feature Access. Select Normal Service (N), Hooking Service (H), or Busy Service (B) from the combo box for Connection Indexes. Assign NND in accordance with a predetermined Numbering Plan.

STEP 3: **ASPA** - Assign an Access Code to CALL WAITING - ORIGINATING, SRV = "SSC: Service Code", SID = 5 for Connection Status Index (CI) of "B: Busy" and "N: Normal".

STEP 4: **ASFC** - Assign to the Stations a Service Feature Restriction Class that allows SFIs 5 [Call Waiting - Originating (Calling)], 6 [Call Waiting - Originating (Called)], and 11 (Data Line Privacy = 0/ Attendant Camp-On =1). This allows the ability to originate a Call Waiting - Originating (SFI 5), to terminate a Call Waiting - Originating (SFI 6), and to receive Camp-On Tone (SFI 11).

STEP 5: **AKYD** - Optional. Assign CALL WAITING - ORIGINATING to a Line/Feature key. Assign Feature Key Call Waiting [4].

STEP 6: **ADSL** - SN = 5 (Consultation Hold) and Feature key Call Waiting [4].

This command is used for assigning the function of each Soft key.

FCCS Programming

This data setting describes only the additional or replacement commands used to activate FCCS Services. When used as an FCCS feature, these commands replace the non-network commands (for example, AAED is replaced by AAEDN). Note that the data setting for FCCS Service varies depending on the Node. **Note 1**

STEP 1: **ANPDL** - Reserve a Number Level for Feature Access. Select Normal Service (N), Hooking Service (H), or Busy Service (B) from the combo box for Connection Indexes. Assign NND in accordance with a predetermined Numbering Plan.

STEP 2: **ASPAL** - Assign an Access Code to CALL WAITING - ORIGINATING, SRV = "SSC: Service Code", SID = 5 for Connection Status Index (CI) of "B: Busy" and "N: Normal".

Note 1: When using ASYDL or ASYDN to program an FCCS feature, the basic programming of ASYD is required in addition to ASYDL or ASYDN.

C-33 Consecutive Dialing - Attendant

General Description

This feature gives the Attendant Console the ability to generate DTMF Signals while engaged in a Station/Trunk Connection. These DTMF Signals are generated via the dial keypad of the Attendant Console.

Operating Procedure

When doing a transfer from the Attendant, the Attendant presses the Start key then dials the desired number.

Service Conditions

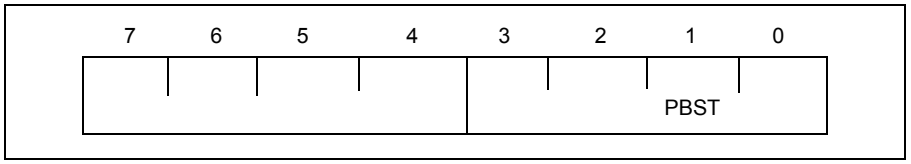
1. The ATTENDANT CONSOLE [A-3] can transmit DTMF Signals under the following conditions:
   a.) When the Attendant Console originates a Trunk Call while an Internal Station is holding, DTMF Signals can be transmitted to the Trunk or Station but not to both in a Three-way Connection.
   b.) When the Attendant Console originates a Trunk Call while another Trunk Call is on hold, DTMF Signals can be transmitted to either Trunk but not to both in a Three-way Connection.
   c.) When the Attendant Console originates an internal Station Call while holding a Station or Trunk, DTMF Signals can be transmitted to either Station or Trunk but not to both in a Three-way Connection.
   d.) When the Attendant Console originates/answers an Internal Station or Trunk Call and remains engaged in the connection. If the Start key is pressed, DTMF Signals cannot be transmitted from the Attendant Console (even when DTMF Signal sending is selected in the System Data).
2. All digits (DTMF Signals) generated by the Attendant Console will not appear on the display.
3. Additional digits can be dialed after the Trunk Soft-hold Timer has timed out or after the Trunk Answer Signal has been received.
4. If the Attendant Console overrides a connection between a Station/Trunk and a Station/Trunk, the Attendant Console may generate DTMF Signals.
5. After answering any of the following calls: the Attendant can send DTMF Signal to the Calling Party by pressing keypad while connected with the call, or the Attendant can hold the answered call and transfer it to a specific destination. Either function can be selected through the System Data.

Ring-down	Operator Call
Individual Attendant Call	Listed Directory Number
Recall	CALL FORWARDING [C-2, 3, 5] to Attendant
OFF-HOOK ALARM [O-6]	CALL FORWARDING - INTERCEPT [C-25] to Attendant
PRIORITY CALL [P-9]	CCIS Incoming Call
HOUSE PHONE [H-2] (Hotline)	

6. If the Attendant answers an Incoming Call transferred from one Attendant Console to another, by INTER-POSITION TRANSFER [I-5], DTMF Signals cannot be generated from the Attendant Console on which the Transferred Call has been answered.

Programming

ASYD - SYS1, Index 229, Bit 2. DTMF Signal sending with the keypad pressed after the Attendant has answered an Incoming Call 0/1: Out/In Service. SYS3, Index 11. Assign a Miscellaneous Timer counter for the duration of the DTMF Sender Tone. 128 milliseconds × (PBST + 1) = Duration of Sender Tone. Maximum value is 2.048 seconds.



C-43 Call Metering

General Description

This feature enables the Metering Pulses received from an Exchange Line involved in an established connection to be detected, counted, and stored as billing information. This information is output to Station Message Detail Recording (SMDR), following the release of the line.

Note: This feature is not available in North America.

Operating Procedure

No manual operation is required.

Service Conditions

1. This facility counts the Metering Pulses, received between the time speech is established and the release of the Station at the Called Office.
2. In the event that the AUTHORIZATION CODE [A-20] is activated on a route provided with the CALL METERING, the records of the Authorization Code will also be included.
3. When the Pulse Count Value exceeds 9999, all Pulse Count Records output to Station Message Detail Recording (SMDR) will indicate 9999.

Programming

STEP 1: **ARTD** - Assign the Route Class Data for the Trunk Route.

Assign CDN: 16:

SMDR2 (Detail Billing information) = 4

- Detail billing is not required for Incoming Calls.
- Detail billing is required for toll and local Outgoing Calls.
- Metering pulses are used.

Assign CDN: 39:

BC (Central Office Line Balance Condition): 0/1 = Not balanced/Balanced. (This is available in Australia only.)

Note: To change the data for existing route (RT), initialize or reboot the trunk device such as a circuit card or MG.

STEP 2: **ARTI**

Assign CDN: 40:

SPMET (Account Pulse Watch Control)

0 = -

1 = Low to High Transition

2 = High to Low Transition

3 = Low to High and High to Low Transition

Note: To change the data for existing route (RT), initialize or reboot the trunk device such as a circuit card or MG.

C-59 Called Number Display - Attendant

General Description

If a call is transferred to the ATTENDANT CONSOLE [A-3] as a result of:

DO NOT DISTURB	[D-11]
CALL FORWARDING - ALL CALLS	[C-5]
CALL FORWARDING - BUSY LINE	[C-2]
CALL FORWARDING - DON'T ANSWER	[C-3]
CALL FORWARDING - INTERCEPT / ANNOUNCEMENT	[C-25]

Information pertaining to the Called Station is displayed on the Attendant Console.

Operating Procedure

1. The Attendant answers an Incoming Call.
   a.) The DEST Lamp lights.
   b.) The Called Telephone Number is displayed.
2. The Attendant presses the SRC key.
   a.) The DEST Lamp goes OFF, and the SRC Lamp lights.
   b.) Either the Calling Party's Tenant Number is displayed, or RSC Number and Telephone Number are displayed.

Service Conditions

1. This service is effective when a Station-to-Station Call or a DID Call has been routed to an Attendant Console as a result of:

DO NOT DISTURB	[D-11]
CALL FORWARDING - ALL CALLS	[C-5]
CALL FORWARDING - BUSY LINE	[C-2]
CALL FORWARDING - DON'T ANSWER	[C-3]
CALL FORWARDING - INTERCEPT / ANNOUNCEMENT	[C-25] and is answered by the Attendant.

2. If the Attendant presses the SRC key while the Called Telephone Number is displayed, the display changes to the number of the Calling Party. The Called Telephone Number cannot be displayed again.
3. While the Called Telephone Number is displayed, the Attendant can transfer the call to a desired Station by keying the Destination Number on the keypad. In this case, the Called Telephone Number of that call cannot be displayed again.
4. While the Called Telephone Number is displayed, the Attendant can place the call on hold by pressing the HOLD key. The Called Telephone Number is displayed again when the Attendant picks up the call again.
5. While the Called Telephone Number is displayed, the Attendant can set CALL PARK [C-29] Service. The Called Telephone Number cannot be displayed again.

Programming

ASYD - SYS, Index 70, Bit 0. Called Number Display when Call Forwarding to ATTCON 0/1: Out/In Service.

C-60 Call Forwarding - Busy Line - Outside

General Description

This feature allows a call that originates from a Station or Trunk, and is destined for a Station to be routed to another Station outside the System via the appropriate Trunk Line. This feature can also be activated from a Desktop terminal Multi Line.

Operating Procedure

To set

1. Lift the handset. Receive Dial Tone.
2. Dial the CALL FORWARDING - BUSY LINE - OUTSIDE Access Code (same as CALL FORWARDING - BUSY LINE [C-2]). Receive Special Dial Tone.
3. Dial the desired Target Telephone Number (or Speed Calling Access Code and Abbreviated Code). If the ATTENDANT CONSOLE [A-3] is setting this feature, dial the Access Code, Tenant Number, Originating Telephone Number, and then the desired Outside Number. Receive Service Set Tone.

Note: A tone will not be provided when an access code is used to set this feature on SP Controlled Standard SIP terminal.

To cancel

1. Lift the handset. Receive Dial Tone.
2. Dial the CALL FORWARDING - BUSY LINE - OUTSIDE Cancel Code, (same as CALL FORWARDING - BUSY LINE [C-2] Cancel Code). Receive Service Set Tone.

To set from a Multi Line

1. Press the Multi Line key to be used for CALL FORWARDING - BUSY LINE - OUTSIDE.
2. Lift the handset or press the SPEAKER key. Receive Dial Tone.
3. Press CALL FORWARDING - BUSY LINE - OUTSIDE Feature Key. Receive Special Dial Tone.
4. Dial the desired Target Number (or Speed Calling Access Code and Abbreviated Code). Receive Service Set Tone. The LED will light. The LCD displays:

FORWARD SET **(Time Display)**

5. Replace the handset or press the SPEAKER key.
6. The LED of the associated Feature key lights on the Dterm for which this Multi Line is the My Line.

To monitor from a Multi Line

1. Press the Multi Line key of the Station CALL FORWARDING - BUSY LINE - OUTSIDE is to monitor.
2. Press CALL FORWARDING - BUSY LINE - OUTSIDE Feature key.
3. The LCD displays:

Target Number **Note 1**

FORWARD XXXXXXXX **(Time Display)**

Note 1: An entered Abbreviated Number is translated into the Forwarding Number and the first 16 digits are displayed on LCD even if a maximum of 24 digits can be stored.

To cancel from a Multi Line

1. Press the Multi Line key of the Station to which CALL FORWARDING - BUSY LINE - OUTSIDE is to be canceled.
2. Lift the handset or press the SPEAKER key. Receive Dial Tone.
3. Press the CALL FORWARDING - BUSY LINE - OUTSIDE Feature key. Receive Service Set Tone. The LED of the associated Feature key will go out. The LCD displays:

FORWARD CANCEL **(Time Display)**

4. Replace the handset or press the SPEAKER key.

Service Conditions

1. CALL FORWARDING - BUSY LINE - OUTSIDE may be set or canceled by either a user or the ATTENDANT CONSOLE [A-3].
2. Stations may be assigned this feature via CLASS OF SERVICE - INDIVIDUAL [C-15] programming in System Data.
3. CALL FORWARDING - BUSY LINE - OUTSIDE Data remains in the Data Memory, even if the system is re-initialized or in the event of a power failure.
4. A maximum of 24 digits, including Trunk Access Code, can be stored.
5. For CALL FORWARDING - BUSY LINE - OUTSIDE, Speed Call List Numbers may be programmed as the Destination Telephone Number. This permits the sent digits to exceed the 24-digit maximum.
6. When using this feature, either the Calling Station or the Forwarding Station is recorded by STATION MESSAGE DETAIL RECORDING (SMDR) [S-10] as the Originating Station of calls forwarded. The selection between the above Stations is determined through System Data.
7. CALL FORWARDING - BUSY LINE - OUTSIDE can be restricted by the RSC of the Originating Station.
8. ROUTE PEG COUNT [P-7] is used for CALL FORWARDING - BUSY LINE - OUTSIDE.
9. This feature can be restricted according to the Dterm My Line Service Feature Restriction Class. Incoming CALL FORWARDING - BUSY LINE - OUTSIDE Calls to the Dterm Multi Line cannot be transferred if the Service Feature Restriction Class of the Dterm My Line restricts this feature.

My Line Service Feature Restriction Class	Multi Line Service Feature Restriction Class	Activation from Dterm Multi Line	Remarks
Non-Restricted	Non-Restricted	Possible	
Restricted	Restricted	Not Possible	
	Non-Restricted		
	Restricted		

10. There is no limit to the number of Stations which may activate this feature simultaneously.

11. Restriction for Outgoing Connection by CALL FORWARDING - BUSY LINE - OUTSIDE are as follows:
    a.) When originating a call from a Station, Route Restriction Data Assignment is verified when this service is in effect.
    b.) When originating a call from a Trunk Restriction for Tandem Connections between the Incoming Trunk and the Outgoing Trunk selected for the call where CALL FORWARDING - BUSY LINE Service occurs.
12. When a caller places a call via OGCA and when the first Outgoing Route and the Calling Party are restricted, this service cannot be set. In this case, the service can be set via the following procedure:
    a.) Assign a Dummy Route.
    b.) Clear the restriction between the Dummy Route and all the Stations.
    c.) Assign the Dummy Route as the first Route of OGCA.
13. When a destination of Call Forwarding - All Calls - Outside has already been assigned, even if the feature is restricted by the new Office Data or changing of Day/Night mode, the assigned data can still be used or canceled.
14. This feature allows a call to terminate to a Trunk outside of the FCCS Network. Accordingly, the Access Code + Office Code or the Access Code including Office Code is required before dialing the Telephone Number to recognize the Terminating Node.
15. When the Incoming Call is forwarding to the Public Network, be sure to add the Outgoing Access Code + Destination Area Code or the Outgoing Access Code including Destination Area Code Number before dialing Subscriber Number.
16. A call can be forwarded to a Trunk in the Remote Node via FCCS.
17. Assign the System Data (ASYDL, SYS1, Index 677, Bit 3 = 1), and press the CALL FORWARDING - BUSY LINE - OUTSIDE Feature key, a maximum of 16 digits Call Forwarding Information can be displayed.

    Note: This feature is available only for the terminal that can display 24 digits on LCD.

18. A tone will not be provided when an access code is used to set this feature on SP Controlled Standard SIP terminal.
19. When a SIP Handler Controlled SIP terminal is in Multi Line Mode, the Call Forwarding setting applies to the seized line while setting. In normal case, where My Line is not used by another station, the Call Forwarding setting applies to My Line. To set the Call Forwarding applying to Sub Line, use the ACFS/ ACFS_T commands.
20. The forwarding destination for this feature can be assigned with the ACFO/ACFOL/ACFON command on a route basis. For details, see CALL FORWARDING DESTINATION PER ROUTE - TRUNK INCOMING CALL [C-209].

    Note: Data assignment on a route basis is available since FP95-114 V4 Issue 2.0 (earlier than FP95-114 V4 Issue 2.0, the destination can be assigned with the ACFO/ACFOL/ ACFON command on a tenant basis only).

Programming

STEP 1: **ASYD** - SYS1, Index 5, Bit 0. Are Access Codes for CALL FORWARDING - BUSY LINE and CALL FORWARDING - DON'T ANSWER the same or separate? 0/1: Same/Separate.
SYS1, Index 69, Bit 7, and SYS1, Index 5, Bit 1 to Bit 3 for MULTIPLE CALL FORWARDING.

STEP 2: **ASYDL** - SYS1, Index 677, Bit 3.

Display a maximum of 16 digits Call Forwarding Information on LCD? 0/1 = No (default)/Yes.

STEP 3: **ANPD** - Reserve a Number Level for Feature Access and Cancel. Select Normal Service (N), Hooking Service (H), or Busy Service (B) from the combo box for Connection Indexes. (applicable to any feature assigned to this level). Assign NND in accordance with a predetermined Numbering Plan.

STEP 4: **ASPA** - Assign an Access Code to CALL FORWARDING - BUSY LINE Entry, SID = 10 and to CALL FORWARDING - BUSY LINE Cancel, SID = 11. Assign Connection Status Index (CI) for "N: Normal" service.

STEP 5: **ASFC** - Assign the Stations to receive CALL FORWARDING - BUSY LINE a Service Feature Restriction Class that allows SFI 9 (Call Forwarding - Busy Line).

Assign either SFI 25 (Call Forward Outside - Local) or SFI 26 (Call Forward - Long Distance). And assign SFI 151 (Call Forwarding - Outside) when CALL FORWARDING - OUTSIDE is performed, RSC of the Forwarding Station is used.

Note: SFI 25 checks AMND for 11 digits or less (Local Service). SFI 26 checks AMND for 12 digits or more (Long Distance Service).

Note: SFI 151 (Call Forwarding - Outside) is not available when the System Data (ASYDL/ ASYDN, SYS1, Index 1393, Bit 5) is set to 1.

STEP 6: **ARRC** - Allow trunk-to-trunk connection between an incoming route and an outgoing route that may be used for this feature.

IC RT: External route number of a tie line for incoming calls

OG RT: External route number of a tie line for outgoing calls

ARI A-RES = 1: Alternate routing is Allowed

ARI D-RES = 1: Alternate routing is Allowed

STEP 7: **AKYD** - For Dterm sets, CALL FORWARDING - BUSY LINE may be assigned to a programmable Line/Feature key. Assign Feature key Call Forwarding - Busy Line [1].

STEP 8: **ATNR** - Allow Tenant-to-Tenant Connection for Inter and Intra-Tenant Connections. Assign Tenant Restriction Index (TRI) = 1. For this feature to be set by the Attendant Console assignment, assign TRI = 3 for Inter and Intra-Tenant Connection via the Attendant Console. This allows a Station in one Tenant to be Call Forwarded to a Station in the same or different Tenant. Also assign TRI = 0, Station-to-Station Calling.

STEP 9: **ACFO** - For Tenant-wide CALL FORWARDING - BUSY LINE of an Incoming DID and DIT Calls, assign CF = 1: Call Forwarding - Busy Line for a destination (CFI) of either the Attendant Console or Station.

FCCS Programming

This data setting describes only the additional or replacement commands used to activate FCCS Services. When used as an FCCS feature, these commands replace the non-network commands (for example, AAED is replaced by AAEDN). Note that the data setting for FCCS Service varies depending on the Node.

Note: When using ASYDL or ASYDN to program an FCCS feature, the basic programming of ASYD is required in addition to ASYDL or ASYDN.

STEP 1: **ANPDL** - Reserve a Number Level for Feature Access and Cancel. Select Normal Service (N), Hooking Service (H), or Busy Service (B) from the combo box for Connection Indexes. (applicable to any feature assigned to this level). Assign NND in accordance with a predetermined Numbering Plan.

STEP 2: **ASPAL** - Assign an Access Code to CALL FORWARDING - BUSY LINE Entry, SID = 10 and to CALL FORWARDING - BUSY LINE Cancel, SID = 11. Assign Connection Status Index (CI) for "N: Normal" service.

Note: System Timers may be set on a network basis in ASYDN.

C-62 Consecutive Speed Calling - System

General Description

This feature allows a user to call frequently dialed DID Numbers using fewer digits (Abbreviated Call Codes) than would normally be required. The stored digits would include Route Access Code, Area Code and Office Code. The caller would then manually enter the Called Telephone Number.

Operating Procedure

1. Dial the Speed Calling Access Code (one to three digits).
2. Dial the Abbreviated Call Code (maximum of four digits).
3. Dial the four-digit Telephone Number. The call is completed.

Service Conditions

1. SPEED CALLING Codes programmed in System Data are accessible to all Stations with a CLASS OF SERVICE - INDIVIDUAL [C-15] that allows CONSECUTIVE SPEED CALLING - SYSTEM.
2. Each Abbreviated Call Code can consist of a maximum of 24 digits, including Access Codes.
3. The maximum number of CONSECUTIVE SPEED CALLING - SYSTEM Codes that can be stored in the system is 10,000.

   Note: The capacity depends on the system configuration due to memory requirements.

4. Speed Calling Numbers are programmed via System Data entry on the PCPro.
5. Stations may be denied access to Speed Calling via CLASS OF SERVICE - INDIVIDUAL assignment in System Data. Speed Calling may also be allowed or denied on a System or Tenant basis.
6. TOLL DENIAL/TOLL DIVERSION [T-6] and TOLL RESTRICTION - 3/6-DIGIT [T-7] may be applied to Speed Calling, if desired.
7. LEAST COST ROUTING - 3/6-DIGIT [L-5] may be provided for Speed Calling access.
8. This feature cannot be used for an Outgoing Call that uses OUTGOING TRUNK QUEUING [O-2].
9. CONSECUTIVE SPEED CALLING - SYSTEM may not be accessed via CONSULTATION HOLD.
10. Digits dialed after the abbreviated call codes are not recorded on SMDR.

Programming

STEP 1: **ASYD** - SYS1, Index 93, Bit 3. Tenant Data 0/1: Separate/Common

SYS2, Index 1, Bit 4 and Bit 5 Toll Restriction

Bit 5	Bit 4	
0	0:	Toll Denial (ROT Connection)
0	1:	Toll Diversion (Terminated to Attendant Console)
1	0:	-
1	1:	-

STEP 2: **ANPD** - Reserve a Number Level for Feature Access. Select Normal Service (N), Hooking Service (H), or Busy Service (B) from the combo box for Connection Indexes. Assign NND in accordance with a predetermined Numbering Plan.

STEP 3: **ASPA** - Assign an Access Code to SPEED CALLING - SYSTEM, assign SRV = "SSC: Service Code", SID = 15. Program for "N: Normal. NND = Access Code plus Abbreviated Digit Code (ADC). See ASPD. For example:

6 + 010

1 digit + 3 digits = NND = 4

STEP 4: **ASFC** - Assign a Service Feature Restriction Class that allows SFI 12 (Speed Calling - System) to Stations that will have SPEED CALLING - SYSTEM.

STEP 5: **ASPD** - Enter Tenant Number (TN), Abbreviated Digit Code (ADC), and Called Number (CD). For details, see the Command Manual.

Note: Enter the necessary numbers of letter Ms instead of the digits for the Station Number.

Example: If the Destination Numbers are: 9 3300 1111 ~ 9 3300 9999

4 digits 4 digits

The required number and letter Ms to be assigned in the CD is: 9 3300 MMMM

4 digits (4 Ms)

C-63 Camp-On Busy

General Description

This feature provides a lamp and tone indication on the ATTENDANT CONSOLE [A-3] as to whether the status of the desired Station Line allows the Attendant to set CAMP-ON service.

Note: This feature is available in Australia only.

Operating Procedure

No manual operation is required.

Service Conditions

1. This feature is available in Australia only.
2. If the status of the desired Station Line allows the Attendant to set CAMP-ON service, a Busy Tone Connection is made and the Busy lamp on the Console lights up.
   Note, however, that the initial lamp and tone indications will not change if the Station Line later enters a status that does not allow CAMP-ON.
3. When the status of the desired Station Line does not allow the Attendant to set CAMP-ON, a Reorder Tone (NUT) Connection is established, and the Busy Lamp on the Attendant Console starts to flash at 60 IPM. The initial lamp and tone indications will not change if the Station Line later enters a status that allows ATTENDANT CAMP-ON WITH TONE INDICATION [A-l].
4. Refer to the specification ATTENDANT CAMP-ON WITH TONE INDICATION [A-1] for the conditions in which the Attendant is not allowed to set CAMP-ON.
5. This feature cannot be activated when CALL BACK - CCIS [C-44] has already been set.
6. This feature cannot be activated when ATTENDANT CAMP-ON WITH TONE INDICATION - CCIS [C-45] has already been set.

Programming

STEP 1: **ASYD** - SYS1, Index 7, Bit 4 (Attendant Camp-On Lamp Display): 0/1 = Busy Lamp Steady/Busy Lamp Flashing (60 IPM). Assign “1”.

C-69 Call Forwarding - All Calls - Announcement

General Description

This feature permits all calls destined for a particular Station to be routed to a Recorded Announcement. Activation and cancellation may be accomplished by either the individual user or the ATTENDANT CONSOLE [A-3].

Operating Procedure

To set and cancel from an individual Station using the same Access Code Note 1

- For setting:

1. Lift the handset. Receive Dial Tone.
2. Dial the CALL FORWARDING - ALL CALLS - ANNOUNCEMENT entry Access Code (same as CALL FORWARDING - ALL CALLS [C-5]) (**Note 2**). Receive Special Dial Tone.
3. Dial the desired Announcement Trunk Access Code (same as ANNOUNCEMENT SERVICE [A-15]). Receive Service Set Tone.

- For cancellation:

1. Lift the handset. Receive Dial Tone.
2. Dial the CALL FORWARDING - ALL CALLS - ANNOUNCEMENT entry Access Code (same as CALL FORWARDING - ALL CALLS [C-5]) (**Note 2**). Receive Special Dial Tone.

Note 1: Available from FP95-114 V4 Issue 2.0 with system data ASYDL, SYS1, Index 686, Bit 0= 1.

To set and cancel from an individual Station using different Access Codes

- For setting:

1. Lift the handset. Receive Dial Tone.
2. Dial the CALL FORWARDING - ALL CALLS - ANNOUNCEMENT entry Access Code (same as CALL FORWARDING - ALL CALLS [C-5]) (**Note 2**). Receive Special Dial Tone.

Note 2: For a Desktop terminal, press the Line/Feature key or Soft Key programmed for CALL FORWARDING - ALL CALLS [C-5].

3. Dial the desired Announcement Trunk Access Code (same as ANNOUNCEMENT SERVICE [A-15]). Receive Service Set Tone.

- For cancellation:

1. Lift the handset. Receive Dial Tone.
2. Dial the CALL FORWARDING - ALL CALLS - ANNOUNCEMENT cancel Access Code (same as CALL FORWARDING - ALL CALLS [C-5]) (**Note 3**). Receive Service Set Tone.

Note 3: For a Desktop terminal, press the Line/Feature key or Soft Key programmed for CALL FORWARDING - ALL CALLS [C-5].

Service Conditions

1. This feature requires one of the following hardware alternatives:
   a.) Central Office Trunk (COT) or TIE Line Trunk (TLT) and an Announcement Machine.
   b.) Digital Announcement Trunk (DAT).
2. Stations with this feature set may originate calls and access the CALL FORWARDING - ALL CALLS - ANNOUNCEMENT Cancellation Code.
3. Stations activating CALL FORWARDING - ALL CALLS - ANNOUNCEMENT [C-69] can be restricted from placing any calls, except to the Attendant, depending upon System Data.
4. There is no limitation to the number of Stations which can simultaneously set CALL FORWARDING - ALL CALLS - ANNOUNCEMENT [C-69].
5. Stations may be assigned this feature via CLASS OF SERVICE - INDIVIDUAL [C-15] programming in System Data.
6. By assigning the Announcement Trunk Access Code to the Announcement Trunk, you can set an announcement device as a destination of CALL FORWARDING - ALL CALLS - ANNOUNCEMENT [C-69].

   Note: An attempt to connect a call on consultation hold to a recorded announcement will fail:
   - If you press the switch hook or the TRANSFER key and dial the Announcement Trunk Access Code while placing the current call on hold.
   - If you, while placing the current call on hold, make a second call to a station where CALL FORWARDING - ALL CALLS - ANNOUNCEMENT [C-69] is enabled.
7. When a call is directed to a Forwarded Station, one burst of Ringing Signal will be sent to the Station to indicate that it is Call Forwarded.
8. Call Forwarding Data remains programmed, even if the system is re-initialized or a power failure occurs.
9. CALL FORWARDING - ALL CALLS - ANNOUNCEMENT can be provided for a maximum of 127 Trunks, which include the Announcement Trunks for:

ANNOUNCEMENT SERVICE	[A-15]
ANNOUNCEMENT SERVICE - ATTENDANT	[A-59]
CALL FORWARDING - INTERCEPT / ANNOUNCEMENT	[C-25]

10. This feature is available with the following connections:

DIRECT INWARD DIALING (DID)	[D-8]
DIRECT-IN TERMINATION (DIT)	[D-7]
STATION-TO-STATION CALLING	[S-11]
Station Calls from a TIE Line Trunk	

11. When a designation of CALL FORWARDING - ALL CALLS - ANNOUNCEMENT has already been assigned, even if the feature is restricted by the new Office Data or changing of Day/Night mode, the assigned data can still be used or canceled.
12. When SIP Handler Controlled SIP terminals are in Multiple Mode, the Call Forwarding setting applies to the seized line while setting. In normal case, where My Line is not used by another station, the Call Forwarding setting applies to My Line. To set the Call Forwarding applying to Sub Line, use the ACFS/ ACFS_T commands.

13. If the called station (Station A) is set to forward all unanswered calls to another station (Station B), on which CALL FORWARDING - ALL CALLS - ANNOUNCEMENT [C-69] is enabled, CALL FORWARDING - DON'T ANSWER [C-3] does not work. Station A will continue ringing and sending Ringback Tone (RBT) to the caller until the call is answered.
14. By registering the Call Forwarding - All Calls Access Code + the desired Announcement Trunk Access Code to the CF Setting button of a Line/Feature key on a Desktop terminal, it is possible to enable and disable CALL FORWARDING - ALL CALLS - ANNOUNCEMENT [C-69] just by pressing the CF Setting button. For a detailed explanation, see CALL FORWARDING (CF) BUTTON - ONE TOUCH [C-208].

Note: CALL FORWARDING (CF) BUTTON - ONE TOUCH [C-208] is available from FP95-114 V4 Issue 2.0.

Programming

Call Forwarding-All Calls

STEP 1: **ASYD** - SYS1, Index 4, Bit 6. One burst of ringing at Forwarding Station when CALL FORWARDING - ALL CALLS [C-5] is in service? 0/1: No/Yes. (For a Single Line Station only).

SYS1, Index 69, Bit 1. A burst of Ringback Tone to alert the person receiving a call that this is a CALL FORWARDING - ALL CALLS [C-5] Call. In service? 0/1: No/Yes. Normally assigned as data "0".

SYS2, Index 6, Bit 4. Enable Station set for CALL FORWARDING - ALL CALLS [C-5] be allowed to use their phones normally. If data "0" is assigned, the Station will only be able to call the Attendant Console. Assign on a per Tenant basis.

STEP 2: **ASYDL/N** - SYS1, Index 686, Bit 0=1.
Disabling Call Forwarding - All Calls with entry Access Code is in service.

Note: Available from FP95-114 V4 Issue 2.0.

STEP 3: **ANPD** - Reserve a Number Level for Service Feature Access and Cancel (**Note 4**). Select Normal Service (N), Hooking Service (H), or Busy Service (B) from the combo box for Connection Indexes. Assign NND in accordance with a predetermined Numbering Plan.

STEP 4: **ASPA** - Assign an Access Code to CALL FORWARDING - ALL CALLS [C-5] entry SID = 8, and to CALL FORWARDING - ALL CALLS [C-5] cancel SID = 9 (**Note 4**). Assign Connection Status Index (CI) for "N: Normal" service.

Note 4: When SYS1, Index 686, Bit 0=1 (Disabling Call Forwarding - All Calls with Access Code is in service), Call Forwarding - All calls cancel is not required because the cancellation can be performed with the entry Access Code.

STEP 5: **ASFC** - Assign the Stations to activate CALL FORWARDING - ALL CALLS [C-5] a Service Feature Restriction Class that allows SFI = 7 (Call Forwarding - All Calls).

STEP 6: **AKYD** - For Desktop terminal, CALL FORWARDING - ALL CALLS [C-5] may be assigned to a Line/Feature key. Assign Feature key Call Forwarding - All Calls [2].

STEP 7: **ADSL** - Assign SN = 1 (Originating; DT Connection) and Feature key Call Forwarding - All Calls [2].
This command is used for assigning the function for each Soft key.

STEP 8: **ATNR** - Allow Tenant-to-Tenant Connection for Inter and Intra-Tenant Connections. Assign Tenant Restriction Index (TRI) = 1. For this feature to be set by the Attendant Console, assign Inter and Intra-Tenant connection via the Attendant Console TRI = 3. Also allow TRI = 0, Station-to-Station Calling.

Announcement Service

STEP 1: **ANPD** - Reserve a Number Level for Trunk Access. Select Normal Service (N) or Hooking Service (H) from the combo box for Connection Indexes. Assign NND values in accordance with a predetermined Numbering Plan.

STEP 2: **ASPA** - Assign an Access Code to the Announcement Trunk. Assign SRV = "ANNC: Announcement Service - Single Announcement" and the EQP Number Assignment should be as assigned with AAED.

STEP 3: **ARTD** - Assign the Announcement Route as shown below.

RT:1

1-OSGS	:2	2-ONSG	:3	5-TF	:1	6-TCL	:1	7-L/T	:1
15-LSG	:4								

Note: To change the data for existing route (RT), initialize the circuit card.

STEP 4: **ATRK** - Assign the LENS. Announcement Trunk Route Number, Trunk Number, and Tenant Number. Assign one or more Trunks to a specific Route used to connect to an Announcement Machine.

STEP 5: **MBTK** - Assign the Make Idle status to the Announcement Trunk.

STEP 6: **ADSL** - Assign SN = 1 (Originating; DT Connection) and Feature key Call Forwarding - All Calls [2]. This command is used for assigning the function for each Soft key.

STEP 7: **ARRC** - If a TIE Line, DID or REMOTE ACCESS TO SYSTEM [R-2] Connection to the Announcement Service Trunk is required, allow Trunk-to-Trunk Connection using ARI-D-RES, Direct Connection.

STEP 8: **AAED** - Used to assign Announcement Equipment.

Assign:

TN:	Tenant Number
EQP:	Announcement Equipment Number
RT, TK:	Route and Trunk Number of the Trunk connected to the Announcement Equipment
C:	Duration of connection, 0/1: Disconnect in 30 seconds/the connection is held until the Station releases. See the requirements of the Announcement Equipment.
R:	Sending RBT, 0/1: Sending RBT/Not Sending RBT. Normally assign data "0".
A:	Answer Signal Sending, 0/1: No answer from Incoming Trunk/answer from Incoming Trunk. Normally assign data "0". No Answer Signal is sent to the Central Office, therefore, Calling Party will not be billed for the call.
M:	Multiple Connection 0/1: Single Connection/Multiple Connection. See the requirements of the Announcement Equipment.

C-73 Call Back - Delayed

General Description

This feature allows a Station to which CALL BACK [C-1] has been set to initiate another call, within a predetermined time period after becoming idle and before the CALL BACK [C-1] feature is activated.

Operating Procedure

No manual operation is required.

Service Conditions

1. This feature is assigned in system programming, on a system-wide basis.
2. The CALL BACK - DELAYED Timer can be set in multiples of two seconds, to a maximum of 14 seconds. (Delay Timer is unavailable for PHS/PCS/VoWLAN.)

Programming

ASYD - SYS1, Index 5, Bit 4 to Bit 6 = Timer; Bit 7 = Out of Service/In Service (No/Yes).

Timer as default is set at 00 = two seconds. Maximum delay is 14 seconds.

SYS1, Index 139. No Answer Timer for CALL BACK. Assign 00H. (RAM Data is 3FH = 30 seconds.)

SYS2, Index 0, Bit 0. Is CALL BACK enabled on a per Tenant basis? 0/1: No/Yes.

SYS2, Index 4, Bit 0. CALL BACK and TRUNK QUEUING Access Code are same or separate? 0/1: Separate/Same.

SYS3, Index 3, Bit 5. Enable Ringer Pattern 5 for CALL BACK. See DISTINCTIVE RINGING [D-10]. (Delay Timer is unavailable for PHS/PCS/VoWLAN.)

SYS3, Index 7. Assign data 42H for four seconds ON, two seconds OFF.

FCCS Programming

This data setting describes only the additional or replacement commands used to activate FCCS Services. When used as an FCCS feature, these commands replace the non-network commands (for example, AAED is replaced by AAEDN). Note that the data setting for FCCS Service varies depending on the Node.

Note: When using ASYDL or ASYDN to program an FCCS feature, the basic programming of ASYD is required in addition to ASYDL or ASYDN.

ASYDN - These data must be assigned to the NCN.

SYS1, Index 5, Bit 4 to Bit 6: Call Back Delayed Timer.

SYS1, Index 5, Bit 7=0/1: Call Back Delayed Timer is Out of Service/In Service.

C-74 Call Forwarding - Don't Answer - Outside

General Description

This feature allows a call which originates from a Station or Trunk, and is destined for a Station, to be routed to another Station outside the System via the appropriate Trunk Line. This feature can also be activated from a Desktop terminal Multi Line.

Operating Procedure

To set

1. Lift the handset. Receive Dial Tone.
2. Dial the CALL FORWARDING - DON'T ANSWER - OUTSIDE Access Code (same as CALL FORWARDING - DON'T ANSWER [C-3] Access Code). Receive Special Dial Tone.
3. Dial the desired Target Telephone Number (or Speed Calling Access Code and Abbreviated Code). If the ATTENDANT CONSOLE [A-3] is setting this feature, dial the Access Code, Tenant Number, Originating Telephone Number, and then the desired Outside Number. Receive Service Set Tone.

FORWARD SET **(Time Display)**

Note: A tone will not be provided when an access code is used to set this feature on SP Controlled Standard SIP terminal.

To cancel

1. Lift the handset. Receive Dial Tone.
2. Dial the CALL FORWARDING - DON'T ANSWER - OUTSIDE Cancel Code, (same as CALL FORWARDING - DON'T ANSWER [C-3] Cancel Code). Receive Service Set Tone.

To set from a Multi Line

1. Press the Multi Line key to be used for CALL FORWARDING - DON'T ANSWER - OUTSIDE.
2. Lift the handset or press the SPEAKER key. Receive Dial Tone.
3. Press CALL FORWARDING - DON'T ANSWER - OUTSIDE Feature key. Receive Special Dial Tone.
4. Dial the desired Target Number (or Speed Calling Access Code and Abbreviated Code). Receive Service Set Tone. The LED will light. The LCD displays:

FORWARD SET **(Time Display)**

5. Replace the handset or press the SPEAKER key.
6. The LED of the associated key lights on the Desktop terminal to which the call is forwarded.

To monitor from a Multi Line

1. Press the Multi Line key for the Station CALL FORWARDING - DON'T ANSWER - OUTSIDE is to monitor.

   Press CALL FORWARDING - DON'T ANSWER - OUTSIDE Feature key. The LCD displays:

   Target Number **Note 1**

FORWARD XXXXXXXX
(Time Display)

Note 1: A dialed Abbreviated Number is translated into the Forwarding Number and the first 16 digits are displayed on LCD even if 24 digits can be stored.

To cancel from a Multi Line

1. Press the Multi Line key for the Station CALL FORWARDING - DON'T ANSWER - OUTSIDE is to be canceled.
2. Lift the handset or press the SPEAKER key. Receive Dial Tone.
3. Press the CALL FORWARDING - DON'T ANSWER - OUTSIDE Feature key. Receive Service Set Tone. The LED of the associated Feature key will go out. The LCD displays:

FORWARD CANCEL
(Time Display)

4. Replace the handset or press the SPEAKER key.

Service Conditions

1. This feature may be set or canceled by either a Station User or the Attendant.
2. For countries except North America, when an original call is placed from a station, this feature is available only if the call is routed to the called station (station to which this feature is set) directly.
3. Stations may be assigned this feature via CLASS OF SERVICE - INDIVIDUAL [C-15] programming in System Data.
4. CALL FORWARDING - DON'T ANSWER - OUTSIDE Data remains in the Data Memory even if the system or LP is re-initialized or in the event of a power failure.
5. For this feature, Speed Call List Numbers may be programmed as the Destination Telephone Number. This permits the sent digits to exceed the 24-digit maximum.
6. When using this feature, the Forwarding Station is recorded by STATION MESSAGE DETAIL RECORDING (SMDR) [S-10] as the Originating Station of calls forwarded. (See ASYD, SYS1, Index 20, Bit 7).
7. This feature can be restricted by the RSC of the Originating Station.
8. Route PEG COUNT [P-7] is used for CALL FORWARDING - DON'T ANSWER - OUTSIDE. Every time CALL FORWARDING - DON'T ANSWER is set or canceled, it is counted in the Service PEG COUNT.

9. This feature can be restricted according to the Dterm My Line Service Feature Restriction Class. Incoming CALL FORWARDING - DON'T ANSWER - OUTSIDE Calls to the Dterm Multi Line cannot be transferred if the Service Feature Restriction Class of the Dterm My Line restricts this feature.

Dterm MY LINE Service Feature Restriction Class	Dterm MULTI LINE Service Feature Restriction Class	ACTIVATION FROM Dterm MULTI LINE	REMARKS
Non-Restricted	Non-Restricted	Possible	
	Restricted		
Restricted	Non-Restricted	Not Possible	
	Restricted		

10. Restriction for Outgoing Connection by CALL FORWARDING - DON'T ANSWER - OUTSIDE are as follows:
    a.) When originating a call from a Station: Route Restriction Data assignment is verified when this feature is in effect.
    b.) When originating a call from a Trunk: Restriction for Tandem Connections between the Incoming Trunk and the Outgoing Trunk selected for the call where this feature occurs.
11. There is no limit to the number of Stations that may activate this feature simultaneously.
12. When the Called Station does not answer for a predetermined time interval, this feature is applied according to the Route Restriction Class of the Originating Station and the Trunk used in Call Forwarding. When restricted by Route Restriction Class, CALL FORWARDING - DON'T ANSWER - OUTSIDE is not activated, and the Called Station continues to ring.
13. Call Transfer - Attendant: When the Called Station does not answer for a predetermined time interval, calls which have been transferred by the Attendant are assigned to this feature according to their Route Restriction Class. When calls are restricted by Route Restriction Class, this feature is not activated, and the Called Station continues to ring.
14. When Attendant transfers a call to a Station that has set CALL FORWARDING - DON'T ANSWER - OUTSIDE, TOLL DENIAL/TOLL DIVERSION [T-6] is not activated.
15. When a caller places a call via OGCA and when the first Outgoing Route and the Calling Party are restricted, this service cannot be set. In this case, the service can be set via the following procedure:
    a.) Assign a Dummy Route.
    b.) Clear the restriction between the Dummy Route and all the Stations,
    c.) Assign the Dummy Route as the first Route of OGCA.
16. This feature is not available for the Trunk Calls by the MF signals. In this case, the call is not forwarded to the designated Station.
17. This feature is not available for the Trunk Calls by MF signals. In this case, the call is not forwarded to the designated Station.
18. A call can be forwarded to a Trunk in the Remote Node via FCCS.
19. A maximum of 24 digits, including Trunk Access Code, can be stored.
20. Assign the System Data (ASYDL, SYS1, Index 677, Bit 3 = 1), and press the CALL FORWARDING - DON'T ANSWER - OUTSIDE Feature key, a maximum of 16 digits Call Forwarding Information can be displayed.

Note: This feature is available only for the terminal that can display 24 digits on LCD.

21. If you set Self Office Code (using ASPA SRV = "UNIF: Office Termination" or AUNE) in front of Outgoing Access Code + Station Number as a forwarding destination, this feature does not work. However, Uniform Numbering through setting Office Code of other node is available.
22. A tone will not be provided when an access code is used to set this feature on SP Controlled Standard SIP terminal.
23. When a SIP Handler Controlled SIP terminal is in Multi Line Mode, the Call Forwarding setting applies to the seized line while setting. In normal case, where My Line is not used by another station, the Call Forwarding setting applies to My Line. To set the Call Forwarding applying to Sub Line, use the ACFS/ ACFS_T commands.

Programming

STEP 1: **ASYD** - SYS1, Index 20, Bit 7. Charging to a call transferred by Call Forwarding - Outside Service 0/ 1: Charging is made to the caller of the Transferred Call/Charging is made to the Station that has set Call Forwarding - Outside Service.

STEP 2: **ASYDL** - SYS1, Index 677, Bit 3.
Display a maximum of 16 digits Call Forwarding Information on LCD? 0/1 = No (default)/Yes.

STEP 3: **ASFC** - Assign SFI 25 (Call Forward Outside - Local) or SFI 26 (Call Forward Outside - Long Distance). Also assign SFI 151 (Call Forwarding - Outside) when CALL FORWARDING - OUTSIDE is performed, RSC of the Forwarding Station is used.
See CALL FORWARDING - DON'T ANSWER [C-3].

Note: SFI 25 checks AMND for 11 digits or less (Local Service). SFI 26 checks AMND for 12 digits or more (Long Distance Service).

Note: SFI 151 (Call Forwarding - Outside) is not available when the System Data (ASYDL/ ASYDN, SYS1, Index 1393, Bit 5) is set to 1.

STEP 4: **ARRC** - Allow trunk-to-trunk connection between an incoming route and an outgoing route that may be used for this feature.
IC RT: External route number of a tie line for incoming calls
OG RT: External route number of a tie line for outgoing calls
ARI A-RES = 1: Alternate routing is Allowed
ARI D-RES = 1: Alternate routing is Allowed

C-75 Call Forwarding - Intercept / Announcement - Attendant

General Description

This feature provides for interception of Attendant-originated Calls that cannot be completed (unassigned Station, Level, etc.). These calls are automatically routed to a Recorded Announcement informing the caller that an Inoperative Number was reached.

Operating Procedure

No manual operation is required once this feature has been programmed in System Data.

Service Conditions

1. This feature requires one of the following hardware alternatives:
   a.) Central Office Trunk (COT) or 2W E&M Trunk and an Announcement Machine.
   b.) Digital Announcement Trunk (DAT).
2. The following connections are available:
   a.) Multi Connection: several Attendants can be connected to the Announcement Trunk at the same time.
   b.) Single Connection: at any given time, only one Attendant can be connected to the Announcement Trunk.
3. If the attendant forgets to cancel a call, the announcement will automatically disconnect in 30 seconds [provided this is programmed in System Data]. AAED must not have the duration of connection set to disconnect in 30 seconds. If the Duration of Connection is set to disconnect in 30 seconds, the Attendant will control disconnection of the announcement.
4. This feature is provided on a per Tenant basis.
5. This feature is only available on Attendant-originated Calls.
6. Only one message can be provided for different intercept conditions.
7. Before connection to the Announcement Machine is made, the system can be programmed to return Ringback Tone.
8. ANNOUNCEMENT SERVICE is provided to FCCS Nodes that have no Announcement Trunk in the Remote Node.
9. When CALL FORWARDING - INTERCEPT/ANNOUNCEMENT [C-25] or CALL FORWARDING - INTERCEPT/ANNOUNCEMENT - ATTENDANT [C-75] is used in an FCCS Network, the additional data shown below is required:
   a.) The Logical Route Number should be applied to the Physical Route that is provided the service.
   b.) The restriction of Route-to-Route Connection between FCCS Nodes should be released.
10. CALL FORWARDING - INTERCEPT / ANNOUNCEMENT [C-25] or CALL FORWARDING - INTERCEPT / ANNOUNCEMENT - ATTENDANT [C-75] is not provided for an Incoming Call using AUTOMATED ATTENDANT [A-82], DIRECT-IN TERMINATION [D-7], or REMOTE ACCESS TO SYSTEM [R-2].
11. Call Forwarding for an Incoming Call routed to a Dead Level or an Unused Number has priority over the ANNOUNCEMENT Service for Incoming Calls. When Attendant is assigned as the forwarded destination, the LCD on the caller's station shows the same display as Call Forwarding-Don't Answer.

12. When a call is terminated from the ISDN Trunk or MG(SIP), RBT is sent to the caller if Announcement Trunk Answer Signal is not sent.
13. When a call with Consultation Hold is terminated from a counter office, if the Announcement Trunk Answer Signal exists, announcement trunk will operate. To return to the call, press the DISC Key. For more information about it, refer to Consultation Hold Release [C-169].
    Note that pressing the hooking key disables the announcement trunk to operate although it returns to a call.

Programming

Programming of hardware into System Data is required. Refer to CALL FORWARDING-INTERCEPT / ANNOUNCEMENT [C-25]. Timers are also programmed into System Data (see VARIABLE TIMING PARAMETER [V-1]).

FCCS Programming

This data setting describes only the additional or replacement commands used to activate FCCS Services. When used as an FCCS feature, these commands replace the non-network commands (for example, AAED is replaced by AAEDN). Note that the data setting for FCCS Service varies depending on the Node.

STEP 1: **ALRTN** - This data must be assigned to the NCN.
Assign the Logical Route Number to the Physical Route.

STEP 2: **AAEDN** - This data must be assigned to the NCN.
EQUIPMENT KIND:
- DAT/Announcement Machine
- VS32

TN: Enter Tenant Number
EQP: Announcement Equipment Number
0 = Dead Level
1 = Unused Number
2-15 = Announcement Service

Duration of Connection:
- Disconnect in 30 seconds
- Connection is held until the Station releases. See the requirements of the application for the Announcement Equipment. For Dead Level applications, data 0 is assigned.

Do Not Send RBT [On/Off]

Note: Assign Do not Send RBT when Multiple Connection = Single.

Send Answer Signal [On/Off]
Multiple Connection: Select either Single or Multiple.
LGRT: Logical Route
TK: Trunk Number **Note 1**

Note 1: Not necessary for Multiple Connection = Single.

STEP 3: **ARRCN** - This data must be assigned to the NCN.
Release the restriction between the Terminating Trunk and the Announcement Trunk.

STEP 4: **ARSCN** - This data must be assigned to the NCN.
Release the restriction between the Announcement Trunk and the Station.

Note: The current command data shown is set to all Nodes using the service: ASTD, ARTD, ATRK, MBTK.

C-76 Call Waiting Lamp - UCD

General Description

This feature uses an LED indication, which is one of the programmable keys on the Desktop terminal, to indicate when a call is waiting to be answered in the UCD Queue.

Operating Procedure

No manual operation is required.

Service Conditions

1. The UCD Call Waiting Lamp displays the status of the UCD Group as follows:
   Lamp ON: One or more calls waiting in the UCD Queue.
   Lamp FLASHING: More than the pre-assigned number of calls are waiting in the UCD Queue.
   Lamp OFF: No calls are waiting in the UCD Queue.
2. The number of calls waiting in the UCD Queue that causes the UCD Call Waiting Lamp to light or flash can be assigned on a per UCD Group basis through the Maintenance Administration Terminal (PCPro).
3. The UCD Call Waiting Lamp indicates the status of the UCD Group that the Desktop terminal is assigned to, not the status of other UCD Groups.
4. The UCD Call Waiting Lamp indication lasts as long as calls are in a queue.
5. The lamp must be assigned to a Desktop terminal Line/Feature key LED through PCPro. A call cannot be answered by this key. Only the UCD Line Number key (Prime, Sub, etc.) can be used to connect a call.
6. The number of UCD Groups per system is 63.
7. When an additional Desktop terminal is allowed this service, or when a Desktop terminal Data using this service is deleted, the circuit card that accommodates the port of the Desktop terminal must be initialized.
8. This feature can be programmed in Local Data Memory (LDM).
9. When using the FCCS Network, this feature is available for UCD Stations belonging to the Node that accommodates the UCD Controlling Station.
10. When using the FCCS Network, if the number of UCD Calls in the queue is over the threshold value, the lamps of Stations in the same Node that the UCD Controlling Station belongs to, start to flash.

Interactions

1. The lamp will toggle between solid and flashing as the number of calls in the queue increases and decreases, crossing the threshold value set in AUCD.
2. If the Pilot Telephone Number of a UCD Group is set to CALL FORWARDING - ALL CALLS [C-5] to another UCD Group, UCD Hunting takes place in the second group. If all Stations in the second group are busy, queuing takes place in the second group and its Call Waiting Lamp will light.
3. ATTENDANT CAMP-ON [A-1] Call directed to the UCD Pilot Station will activate the UCD Call Waiting Lamp if all Stations in the group are busy.
4. CALL WAITING LAMP - UCD is not affected by CALL WAITING - TERMINATING [C-12].
5. CALL WAITING LAMP - UCD is only effective for the My Line Station of a Desktop terminal. Even if another UCD Group appears as Sub Lines on that Dterm, CALL WAITING LAMP - UCD will only function for the UCD Group of the My Line Telephone Number.

6. If OVERFLOW-UCD [O-13] is programmed, overflow to the second group will take place if all Stations in the first group are either busy or in the busy-out state. There will be no CALL WAITING LAMP - UCD indication to the first group.
7. If the Overflow Group is all busy, queuing takes place to the first group and the CALL WAITING LAMP - UCD will light to indicate a queuing condition at the first group.
8. When a UCD Station is assigned as the Destination Station of LDN Night Connection, this service cannot be activated.
9. For the FCCS Network, this feature is available only for UCD Stations in the same Node as the UCD Controlling Station. For more information, see UNIFORM CALL DISTRIBUTION [U-1].
10. This feature does not support LOCATION DIVERSITY [L-55].
11. Do not set ASFC, SFI 252=1 (Multi-Boss/Multi-Secretary is in service) to stations belonging to UCD groups.

Programming

STEP 1: **ASYD** - SYS1, Index 61, Bit 5. Is UCD Call Waiting Lamp - In Service? 0/1: No/Yes.

STEP 2: **ASHU/ASHUL** - Assign Stations to UCD Group.

STEP 3: **AUCD/AUCDL** - UCD Call Waiting Lamp threshold value. This command assigns the number of calls in UCD Queue before the UCD Call Waiting Lamp flashes.

TN/UGN: Tenant Number/User Group Number

STN/TELN: Assign Station Number/Telephone Number in the UCD Group. After one Station is assigned in the group, all Stations in the group will automatically have the same threshold value.

QTH ACT: UCD Queueing Peg Count Value Valid/Invalid (1/0).

QTH: UCD Queueing Peg Count Value (0-255).

MCI: Message Center Interface Data Transfer Valid/Invalid (1/0).

CWT: Call Waiting Threshold Value (1-255).

1-255: Quantity of calls in UCD Queue before the UCD Call Waiting Lamp will begin to flash.

Note: When programming the data, initialize the ELC Card which accommodates the UCD Group Station.

STEP 4: **AKYD** - The Call Waiting Lamp must be assigned to a programmable Line/Feature key to a Desktop terminal. Assign Feature key Call Waiting Lamp [47].

FCCS Programming

This data setting describes only the additional or replacement commands used to activate FCCS Services. When used as an FCCS feature, these commands replace the non-network commands (for example, AAED is replaced by AAEDN). Note that the data setting for FCCS Service varies depending on the Node.

Note: When using ASYDL or ASYDN to program an FCCS feature, the basic programming of ASYD is required in addition to ASYDL or ASYDN.

STEP 1: **ASYDN** - This data must be assigned to the NCN.

SYS1, Index 60, Bit 3: 0/1= UCD Queuing is used/not used.

STEP 2: **AUCDN** - This data must be assigned to the NCN.

UCD Call Waiting Lamp threshold value. This command assigns the number of calls in UCD Queue before the UCD Call Waiting Lamp flashes.

UGN: User Group Number

TELN: UCD Telephone Number

QTH ACT: 0/1 = UCD Queuing Peg Count Value Invalid/Valid

QTH: UCD Queuing Peg Count Value (0-255).

CWT: Call Waiting Threshold Value (1-255).

MCI: 0/1 = Message Center Interface Data Transfer is Valid/Invalid.

Note: When programming the data, initialize the ELC Card which accommodates the UCD Group Station.

C-81 Call Forwarding - I'm Here

General Description

This feature permits the CALL FORWARDING - ALL CALLS feature to be set/canceled from the Target Station.

This feature allows the user (Target Station) to overwrite the existing destination number with new number (Reconfiguration). There is no need to cancel this feature before setting new destination number. Also, Station ID Code for Privacy protection (User Authentication) can be set from Call Forwarding Station (Called Station).

Note: PS and SP Controlled SIP terminal are not supported.

Operating Procedure

To set Station ID Code from Call Forwarding Station (Called Station)

1. Lift the handset. Receive Dial Tone (DT).
2. Dial Access Code for Station ID Code Entry. Receive Special Dial Tone (SPDT).
3. Dial Station ID Code. Receive Service Set Tone (SST). **Note 1**, **Note 2**
4. Replace the handset.

Note 1: Dial a desired Station ID Code (in the range of 1-10 digits). If the number of digits for Station ID Code is less than 10, dial "#" or "*" after the Station ID Code. Dialing "#" or "*" implies the completion of Station ID Code.

Note 2: When Inter-digit timer expires, the dialed number becomes invalid (the user hears ROT).

To cancel Station ID Code from Call Forwarding Station (Called Station)

1. Lift the handset. Receive Dial Tone (DT).
2. Dial Access Code for Station ID Code Entry. Receive Special Dial Tone (SPDT).
3. Dial "#" or "*". Receive Service Set Tone (SST). **Note 3**
4. Replace the handset.

Note 3: The system data (ASYDL/ASYDN, SYS1, Index 895, Bit 4) determines whether to use "#" or "*" as a character which indicates the completion of Station ID Codes.

To set CALL FORWARDING - I'M HERE from Target Station

- When User Authentication is disabled:

1. Lift the handset. Receive Dial Tone (DT).
2. Dial CALL FORWARDING - I'M HERE Entry Access Code (from Target Station). Receive Special Dial Tone (SPDT).
3. Dial Called Station (Call Forwarding Station) Number. Receive Service Set Tone (SST).
4. Replace the handset.

- When User Authentication is enabled:

1. Lift the handset. Receive Dial Tone (DT).

2. Dial CALL FORWARDING - I'M HERE Entry Access Code (from Target Station). Receive Special Dial Tone (SPDT).
3. Dial Called Station (Call Forwarding Station) Number. Receive Special Dial Tone (SPDT).
4. Dial Station ID Code. Receive Service Set Tone (SST). **Note 4**
5. Replace the handset.

Note 4: If the number of digits for Station ID Code is less than 10, dial "#" or "*" after the Station ID Code.

To cancel CALL FORWARDING - I'M HERE from Target Station

- When User Authentication is disabled:

1. Lift the handset. Receive Dial Tone (DT).
2. Dial CALL FORWARDING - I'M HERE Cancel Code (from Target Station), receive Special Dial Tone (SPDT).
3. Dial Called Station (Call Forwarding Station) Number. Receive Service Set Tone (SST).
4. Replace the handset.

Note: Target Station and any Stations other than Forwarding Station can cancel CALL FORWARDING-ALL CALLS [C-5] set at the Forwarding Station by this operation.

- When User Authentication is enabled:

1. Lift the handset. Receive Dial Tone (DT).
2. Dial CALL FORWARDING - I'M HERE Cancel Code (from the Target Station), receive Special Dial Tone (SPDT).
3. Dial Called Station (Call Forwarding Station) Number. Receive Special Dial Tone (SPDT).
4. Dial Station ID Code. Receive Service Set Tone (SST). **Note 5**
5. Replace the handset.

To overwrite CALL FORWARDING - I'M HERE from new Target Station Note 5

- When User Authentication is disabled:

1. Lift the handset. Receive Dial Tone (DT).
2. Dial CALL FORWARDING - I'M HERE Entry Access Code (from new Target Station). Receive Special Dial Tone (SPDT).
3. Dial Called Station (Call Forwarding Station) Number. Receive Service Set Tone (SST).
4. Replace the handset.

- When User Authentication is enabled:

1. Lift the handset. Receive Dial Tone (DT).
2. Dial CALL FORWARDING - I'M HERE Entry Access Code (from new Target Station). Receive Special Dial Tone (SPDT).
3. Dial Called Station (Call Forwarding Station) Number. Receive Special Dial Tone (SPDT).
4. Dial Station ID Code. Receive Service Set Tone (SST). **Note 5**
5. Replace the handset.

Note 5: If the number of digits for Station ID Code is less than 10, dial "#" or "*" after the Station ID Code.

Note: Available when ASYDL/ASYDN, SYS1, Index 895, Bit 3=1.

To cancel CALL FORWARDING - I'M HERE from Called Station (Call Forwarding Station)

1. Lift the handset. Receive Dial Tone (DT).
2. Dial CALL FORWARDING - ALL CALLS Cancel Code (from the Called Station), receive Special Dial Tone (SPDT).
3. Replace the handset.

Service Conditions

1. The following Stations cannot set CALL FORWARDING - I'M HERE:
   The following Station can set Call FORWARDING - I'M HERE when ASYDL, SYS1, Index 895, Bit 3=1.
   a.) a Station where CALL FORWARDING - I'M HERE is being set,
   b.) a Station setting CALL FORWARDING - ALL CALLS [C-5].
2. CALL FORWARDING - I'M HERE can be set to the following Stations:
   a.) a Station setting CALL FORWARDING - I'M HERE,
   b.) a Station where CALL FORWARDING - ALL CALLS [C-5] is being set.
3. A call can be forwarded multiple times following CALL FORWARDING - I'M HERE setting, if assigned in System Data.
4. CALL FORWARDING - I'M HERE can be set or canceled from a Sub Line of a Desktop terminal. However, the Transferred Call terminates to the My Line of the Desktop terminal that has set the service.
5. CALL FORWARDING - I'M HERE can be set or canceled when the Target Station is idle.
6. CALL FORWARDING - I'M HERE can be canceled from the Forwarding Station, following the CALL FORWARDING - ALL CALLS [C-5] cancellation procedure.
7. If Call Back is set to/by the Station setting CALL FORWARDING - I'M HERE, Call Back takes precedence.
8. Other Service Conditions are the same as for CALL FORWARDING - ALL CALLS [C-5].
9. The Target Station in the Remote Node can set CALL FORWARDING - I'M HERE through FCCS.
10. CALL FORWARDING - I'M HERE does not support ISDN terminals and Attendant Consoles.
11. Terminals in logout status can also set/cancel CALL FORWARDING - I'M HERE. When both CALL FORWARDING - I'M HERE and LOGGED OUT IP STATION - CALL DESTINATION [L-53] are set at a terminal, the priority order of these two features can be determined according to Service Feature Restriction Class of the Station (ASFC/ASFCN SFI 209).
12. When setting CALL FORWARDING - I'M HERE across Tenants, use Telephone Numbers (TELN). When setting CALL FORWARDING - I'M HERE within a node, Station Number can also be used.
13. When using DIAL PLAN TO SUPPORT IP CENTREX FUNCTIONALITY [D-166], dial Station Numbers for calls within a node and Telephone Numbers (TELN) for calls between nodes.
14. When using LOCATION DIVERSITY [L-55], this feature data must be assigned with Telephone Number (TELN).
15. Terminals accommodated in Virtual LENS (PHS, SP Controlled SIP terminal) cannot be used as a transferring terminal, since User Authentication or overwriting the existing destination number are restricted on those terminals. Assign RES=0 (restricted) to SFI 143 and SFI 243 (ASFC/ASFCN).

16. When using the FCCS Network, there is no limit to the number of Stations that may access this feature simultaneously.
17. When using the FCCS Network, a Service Feature Restriction Class (SFC) is necessary for the Forwarded Destination and the Forwarding Party.
18. When using the FCCS Network, if the destination (Target Station) is a Desktop terminal, CALL FORWARDING - I'M HERE can be set or canceled from a Sub Line of a Desktop terminal. At that time, the Transferred Call terminates to the My Line of the Desktop terminal that sets the service.
19. When using the FCCS Network, the service can be set or canceled when the Forwarding Station is idle.
20. When using the FCCS Network, the Service Conditions are the same between CALL FORWARDING - DON'T ANSWER Services to be set from the Forwarding Station and the Target Station (CALL FORWARDING - I'M HERE).
21. When using the FCCS Network, CALL FORWARDING - I'M HERE cannot be set from the ISDN Terminal (ISDN Terminal cannot be the destination).
22. When using the FCCS Network, CALL FORWARDING - I'M HERE can be set from the Target Station (not ISDN Terminal) toward the ISDN Terminal (ISDN Terminal can be a Forwarding Station).
23. When using the FCCS Network, CALL FORWARDING - I'M HERE cannot be a set to the Station that has already set CALL FORWARDING - ALL CALLS (Forwarding Station). However, setting this service to the Station that operates CALL FORWARDING - ALL CALLS (destination) is available.
24. When using the FCCS Network, this service cannot be used with SPLIT CALL FORWARDING [S-99].
25. When using the FCCS Network, CALL BACK [C-1, 1D] that is set to or from the Forwarding Station has priority over CALL FORWARDING - I'M HERE (Recall or Answer Call for CALL BACK is not the object of CALL FORWARDING - I'M HERE).
26. When using the FCCS Network, if this feature is set from the Target Station in the Remote Node within an FCCS Network, dial the Telephone Number for the Forwarding Station Number.
27. When using the FCCS Network, the Telephone Number must be assigned to the Destination Station also.
28. When using the FCCS Network, the Service Restriction is decided by the ASFC Data programmed to the Station that sets the service (Destination Station).
29. The following shows service conditions concerning Station ID Code.
    a.) Up to 10 digits (0-9) can be dialed as a Station ID Code.
    b.) If the number of digits for Station ID Code is less than 10, dial "#" or "*" after the Station ID Code. Dialing "#" or "*" implies the completion of Station ID Code. If Inter digit timer expires before dialing "#" or "*", this feature is restricted.
    c.) Station ID Codes cannot be displayed or listed by PCPro. Therefore, user needs to manage their Station ID Codes by themselves.
    d.) Authorization Code Display Elimination

    - By setting SFI 137 = 1 (ASFC/ASFCN), Authorization Code Display Elimination will be allowed. The Station ID Code being dialed will be displayed as "XXXXX."

    - While dialing, character code (* or #) for end of reception will be displayed as "X," although they will not be recognized as Station ID Code. For example, if "12345" is the Station ID Code and "#" is the character code for end of reception:

      example 1: 123*45# (7 digit display: XXXXXXX)
      example 2: *12345# (7 digit display: XXXXXXX)
      example 3: 12345**# (8 digit display: XXXXXXXX)

      Station ID Code will be recognized as 12345 in all of the above.

Note: If Authorization Code Display Elimination is restricted, the dial number for the above examples 1-3 will be displayed. Furthermore, "*" and "#" will not be recognized as Station ID Code regardless of whether Authorization Code Display Elimination is allowed or restricted.

e.) Station ID Codes are shared between this feature and REMOTE CALL FORWARDING CONTROL [R-54]

f.) Even if User Authentication is enabled (ASFC/ASFCN SFI 243=1), this feature can be set without dialing Station ID Code if no Station ID Code is set in advance.

g.) If Station ID Codes already exist in the system, be sure to delete and set them again ("*" may be included in the existing Station ID Codes).

30. When SYS1, Index 69, Bit 1 [(Bit 2 - Call Forward Recall Tone). A burst of Ringback Tone to alert the person receiving a call that this is a CALL FORWARDING - ALL CALLS Call. In service? 0/1:No/Yes. Normally assigned as data "0".] is assigned, there will be no burst of Ringback Tone if the call is received by a SIP station.

31. Regarding Forwarding Party:

a.) It is possible to set Forwarding Party number by SPEED CALLING - SYSTEM [S-3].

b.) It is not possible to set Forwarding Party number by SPEED CALLING - STATION/GROUP [S-21/23].

Programming

STEP 1: Assign the CALL FORWARDING - ALL CALLS Data. (See CALL FORWARDING - ALL CALLS [C-5].)

STEP 2: **ASYD** - SYS1, Index 79, Bit 2. Assign as data "0". Disable Split Call Forwarding Service.

SYS1, Index 90, Bit 1. Backup Call Forwarding and Individual Speed Data to HDD. 0/1: No/Yes

Note: When using Routine Backup, be sure to assign as data "1".

SYS1, Index 304, Bit 0. Individual Speed Calling Data Save. 0/1 = Out of service/In service
Bit 1. Call Forwarding Data Save. 0/1 = Out of service/In service
Bit 2. PS Data Save. 0/1 = Out of service/In service

SYS1, Index 305, Bit 1. CFL (Call Forwarding-Logout) Data Save. 0/1 = Out of service/In service

STEP 3: **ASYDL** - SYS1, Index 895, Bit 3. Allows the user to overwrite the existing forwarding destination number with new one when setting CALL FORWARDING - I'M HERE. 0/1 = Out of service/In service

SYS1, Index 895, Bit 4. Determines whether to use "#" or "*" as a character which indicates the completion of Station ID Codes. 0/1 = "#" is used as a character which indicates the completion of Station ID Codes/"*" is used as a character which indicates the completion of Station ID Codes

STEP 4: **ANPD** - Reserve a Number Level for Feature Access and Cancel.

STEP 5: **ASPA** - Assign an Access Code to CALL FORWARDING - I'M HERE Entry; SID = 28 and to CALL FORWARDING - I'M HERE Cancel SID = 29. Assign a Connection Status Index (CI) for "N: Normal" service.

Assign an Access Code for Station ID Code. Set SRV = "SSCA: Service Code Appendix"; SIDA = 100.

Note: When setting with Telephone Numbers (TELN), use ASPAL/ASPAN command.

STEP 6: **ASFC** - Assign a Service Feature Restriction Class that allows SFI 99 [Call Forward I'm Here (Destination)] to the Target Stations, and SFI 100 [Call Forward I'm Here (Origination)] to the Station to be set.

Assign a Service Feature Restriction Class that allows SFI 137 [Authorization Code Display Elimination] to the Target Stations.

Assign a Service Feature Restriction Class that allows SFI 143 [Station ID Code Entry] to the Target Stations.

Assign a Service Feature Restriction Class that allows SFI 243 [User Authentication for Call Forwarding - I'm Here] to the Called Stations.

STEP 7: **MEM_HDD** - Back up the Call Forwarding data to the hard disk.

Direction Select: Memory to Hard Disk

Data Type Select: Wireless Call Forwarding, Call Forwarding, Call Forwarding-Logout and SIP Terminal Call Transfer

LP Select: Specify the LP Number.

FCCS Programming

This data setting describes only the additional or replacement commands used to activate FCCS Services. When used as an FCCS feature, these commands replace the non-network commands (for example, AAED is replaced by AAEDN). Note that the data setting for FCCS Service varies depending on the Node.

STEP 1: **ASYD** - SYS1, Index 4, Bit 6. One Burst of ringing on Call Forwarding telephone when Call Forwarding - All Calls Service has been assigned (Analog Telephones only).

0/1 = Not Required/Required

SYS1, Index 5, Bit 1 to Bit 3. Maximum number of Multiple Call Forwarding - All Calls/Busy Line occurrences:

Note: This data is valid when SYS1, Index 69, Bit 7 is assigned as "1".

SYS1, Index 69, Bit 1. Send Short Tone when a Call Forwarded via Call Forwarding - All Calls Service is answered.

0/1 = Not Required/Required

SYS1, Index 69, Bit 7. Multiple Call Forwarding - Busy Line/All Calls

0/1 = Out/In Service

SYS1, Index 79, Bit 2. Assign as data "0". Disable Split Call Forwarding Service.

SYS1, Index 90, Bit 1. Backup Call Forwarding and Individual Speed Data to HDD. 0/1: No/Yes

Note: When using Routine Backup, be sure to assign as data "1".

SYS1, Index 248, Bit 7. Tone to be sent out when the handset is off-hook at the Station on which Call Forwarding - All Calls Service is set.

0/1 = Dial Tone/Special Dial Tone

SYS1, Index 304, Bit 0. Individual Speed Calling Data Save. 0/1 = Out of service/In service
Bit 1. Call Forwarding Data Save. 0/1 = Out of service/In service
Bit 2. PS Data Save. 0/1 = Out of service/In service

SYS1, Index 305, Bit 1. CFL (Call Forwarding-Logout) Data Save. 0/1 = Out of service/In service

SYS2, Index 6, Bit 4. Call Origination Restriction of Station upon setting Call Forwarding - All Calls
0/1 = Required/Not Required

STEP 2: **ASYDN** - SYS1, Index 895, Bit 3. Allows the user to overwrite the existing forwarding destination number with new one when setting CALL FORWARDING - I'M HERE. 0/1 = Out of service/In service

SYS1, Index 895, Bit 4. Determines whether to use "#" or "*" as a character which indicates the completion of Station ID Codes. 0/1 = "#" is used as a character which indicates the completion of Station ID Codes/"*" is used as a character which indicates the completion of Station ID Codes

STEP 3: **ANPDN** - Assign the first number of the numbering level for Feature Set and Cancel Access Code.

STEP 4: **ASPAN** - Assign the Feature Access Code.

CI = N, SRV = "SSC: Service Code" and SID 28 (Call Forwarding - I'm Here; set)

CI = N, SRV = "SSC: Service Code" and SID 29 (Call Forwarding - I'm Here; cancel)

Assign an Access Code for Station ID Code. Set SRV = "SSCA: Service Code Appendix"; SIDA = 100.

STEP 5: **ASFCN** - Assign the Service Restriction Data.

Assign RES = 1 (allowed) to SFI 99 [Call Forwarding - I'm Here (Destination)]
and SFI 100 [Call Forwarding - I'm Here (Origination)]

Note: SFI 100 should be allowed at the Node that the Destination Station belongs to.

Assign a Service Feature Restriction Class that allows SFI 137 [Authorization Code Display Elimination] to the Target Stations.

Assign a Service Feature Restriction Class that allows SFI 143 [Station ID Code Entry] to the Target Stations.

Assign a Service Feature Restriction Class that allows SFI 243 [User Authentication for Call Forwarding - I'm Here] to the Called Stations.

STEP 6: **MEM_HDD_N** - Back up the Call Forwarding data to the hard disk.

Direction Select: Memory to Hard Disk
Data Type Select: Wireless Call Forwarding, Call Forwarding, Call Forwarding-Logout and SIP Terminal Call Transfer
LP Select: Specify the LP Number.

C-109 Centrex Compatibility

General Description

This feature allows a DTMF Type Telephone user engaged in a Trunk Call to send a Hooking Signal to the Called Office, enabling the call to be transferred to another Station in that Office.

Operating Procedure

Single Line Phone
Station A is connected to Station B, accommodated in another Office

1. Station A presses switch hook. Receives Special Dial Tone from System. Dial the switch hook outside Access Code. Station B is placed on hold.
2. Station A dials the desired Telephone Number. When Station A releases from the call, Station B will be connected to the dialed Station.

OR

1. Station A dials the Attendant Access Code. When Station A releases from the call, Station B will be connected to the Attendant.

Dterm
Station A is connected to Station B, accommodated in another Office

1. Station A presses the TRANSFER key. Receives Special Dial Tone of the System. Dial the switch hook Outside Access Code, or press the CAS Switch Hook Feature key. Station B is placed on hold and Station A will receive Central Office Dial Tone.
2. Station A dials the desired Telephone Number. If Station A releases from the call, Station B will be connected to the dialed Station. If Station A presses the CAS Switch Hook Feature key there will be Three-Way Conference.

OR

1. Station A dials the Attendant Access Code. When Station A releases from the call, Station B will be connected to the Attendant.

Service Conditions

1. This feature is available for DTMF Telephones TEC = 2/12 only.
2. The Trunk that passes the Hooking Signal is determined by Route Class Data.
3. A COT is required for this application.
4. A Desktop terminal instrument may have a CAS Switch Hook Feature key assigned as one of its Feature/ Line Appearance Buttons, or it can use an Access Code.
5. A Dterm may use its dedicated TRANSFER key to transfer calls internally.
6. CALL TRANSFER - ALL CALLS [C-11] to an Inter-Office Station is not available when this feature is activated.

Programming

STEP 1: **ARTD** - Assign CDN 67:HKS, Central Office Line hooking, as “1” for Central Route.

Note: To change the data for existing route (RT), initialize or reboot the trunk device such as circuit card or MG.

STEP 2: **AKYD** - Feature key CAS SHF [17] is assigned for Dterms to switch hook flash the Centrex Route.

STEP 3: **ADSL** - Assign SN = 6 (Trunk Busy) and Feature key CAS SHF [17]. This command is used for assigning the function to each Soft key.

STEP 4: **ASPA -** SRV = “SSC: Service Code”, SID = 43 Condition (CI) = “H: Hooking” for CAS switch hook outside.

C-131 Call Forwarding - Toll Restriction

General Description

This feature allows the system to be programmed to restrict Call Forwarding Destinations by determining the maximum number of digits (8-12) that can be assigned as a Destination Station Number.

Note: This feature is only available in Australia.

Operating Procedure

No manual operation is required.

Service Conditions

1. This feature is available in Australia only.
2. This service is applicable to the following Call Forwarding services.

CALL FORWARDING - ALL CALLS	[C-5]
CALL FORWARDING - ALL CALLS - CCIS	[C-45]
CALL FORWARDING - ALL CALLS - OUTSIDE	[C-28]
CALL FORWARDING - BUSY LINE	[C-2]
CALL FORWARDING - BUSY LINE - CCIS	[C-46]
CALL FORWARDING - BUSY LINE - OUTSIDE	[C-60]
CALL FORWARDING - DON'T ANSWER	[C-3]
CALL FORWARDING - DON'T ANSWER - CCIS	[C-47]
CALL FORWARDING - DON'T ANSWER - OUTSIDE	[C-74]
CALL FORWARDING - ASSIGNMENT - ATTENDANT	[C-94]

3. Whether to activate this service or not is determined by the Service Feature Restriction Class of each Station.
4. The maximum number of digits (8-12) of a Destination Station Number is determined on a system basis. When a station user tries to assign a Destination Station Number longer than the maximum the user receives a Reorder Tone.
5. When this service is not activated, the maximum number of digits of a Destination Station Number is 12.

Programming

STEP 1: **ASFC** - Assign Service Feature Restriction Class of the Station.
SFI = 111 (Call Forwarding - Toll Restriction). RES: 0/1 = Restricted/Allowed. Assign "1".

STEP 2: **ASYD** - Assign the maximum number of digits. SYS1, Index 248, Bit 0 to Bit 3 (Maximum Number of Digits for Call Forwarding - Toll Restriction)

INPUT DATA	MAXIMUM NUMBER OF DIGITS	REMARKS
0	12	
1~8	8	
9	9	
A	10	
B	11	
C~F	12	

C-133 Call Forwarding - Don't Answer To UCD Queue

General Description

This feature allows calls to a Station that has set CALL FORWARDING - DON'T ANSWER to a UNIFORM CALL DISTRIBUTION Group to be added to a UCD Queue if all Stations in the UCD Group are busy. (Originally, calls forwarded to a UCD Group would receive Reorder Tone if all Stations in the group were busy.)

Note: This feature is only available in North America.

Operating Procedure

The operating procedure is the same as for normal CALL FORWARDING - DON'T ANSWER [C-3] operation.

Service Conditions

1. This feature is available in the USA and Canada and only works in the following cases:
   a.) Station Call to a Station that has set CALL FORWARDING - DON'T ANSWER to a UCD Group.
   b.) Trunk Call to a Station that has set CALL FORWARDING - DON'T ANSWER to a UCD Group.
   c.) Attendant transfers a Station or Trunk Call to a Station that has set CALL FORWARDING - DON'T ANSWER to a UCD Group.
2. This feature is not applicable with calls on CONSULTATION HOLD [C-17] or CALL TRANSFER [C-10, C-11].
3. This feature is only available for FCCS link with CCIS.

Programming

No additional programming is required once UCD parameters are assigned.

C-146 Call Hold - Conference

General Description

This feature allows a user to establish a Conference Connection by adding a third party who is held on a Multi Line of a Desktop terminal to the existing Two-party Connection. The Held Call which is to be added may be a Station or Trunk Call.

Operating Procedure

To operate from a Desktop terminal via Access Code

1. Make a Trunk/Station Call using Line A on the Desktop terminal.
2. Place the in-progress call on hold.
3. Place or answer a call using line B on the Desktop terminal.
4. Press the TRANSFER key.
5. Enter an Access Code for Call Hold - Conference and Line Number of line A.
6. Three-party Conference is established.

To operate from a Desktop terminal via Feature key

1. Make a Trunk/Station Call using Line A on the Desktop terminal.
2. Place the in-progress call on hold.
3. Place or answer a call using Line B on the Desktop terminal.
4. Press the Call Hold Conference Feature key.
5. Press the held Line key, or dial the Line Number of Line A.
6. Three-party Conference is established.

To operate from an Analog Station

1. Make a Trunk/Station Call using line A on the Desktop terminal.
2. Place the in-progress call on hold.
3. Place or answer a call using an Analog Station that is appearing on the Desktop terminal.
4. From the Analog Station, momentarily press the SHF key of the Analog Station.
5. Enter an Access Code for Call Hold - Conference and Line Number of Line A.
6. Three-party Conference is established.

Service Conditions

1. This service is available for the general user and the Desktop terminal user (not available for the ATT and ISDN Terminal user).
2. This service is available when the call is held by the Desktop terminal HOLD key.
3. When the originally connected Party B is placed on Exclusive Hold, this service is available only when the Dterm user A engages in the call with the third Party C.
4. When the Desktop terminal user A places the Station or Trunk B on hold, including Exclusive Hold, DUAL HOLD [D-32D] cannot be activated.
5. This service is not available when Line A or B is an intercom line, Hotline or CAS line. In these cases, the service is restricted.
6. This service is available when the Parties A, C, the Lines A and B are the following:

a.) Interoffice Trunk (including ISDN Line)
b.) Central Office Line/TIE Line
c.) Station, including Virtual Stations.
In case the ISDN Terminal, Intercom Line, Hotline, or CAS Line is used, the service is restricted.

7. Data communications are not considered in this service.
8. A Desktop terminal user dials the CALL HOLD CONFERENCE Code and then dials the Telephone Number. After the CALL HOLD CONFERENCE Code is dialed, no tone is returned and the LCD shows Transferring.
9. When the Three-way Call cannot be established due to all Conference Trunks Busy, Service Restriction, etc., Reorder Tone is returned.
10. When the Held Party is the Trunk, billing is performed according to the System Data (SYS1, Index 33, Bit 6, Bit 5, Bit 4 and Bit 3). In case the Held Party is the Station, the billing stops at the time the Station is placed on hold.
11. This feature and Add-On Conference - 8 Party [A-121] are mutually exclusive.

Programming

STEP 1: **ASFC** - SFI 75 (Call Hold - Conference) must be allowed for appropriate Service Feature Restriction Classes.

STEP 2: **ANPD** - Reserve a Number Level for Feature Access.

STEP 3: **ASPA** - Assign an Access Code for Call Hold - Conference.
CI = "H: Hooking", SRV = "SSCA: Service Code Appendix", SIDA = 97, NND = Number of Digits in Access Code.

STEP 4: **ATRK** - Assign LENS in Route 909 (Digital Conference Trunk).

XX	X	XX	X
Module	Unit	Group	Level

Note that Group Number and Level Number must be one of the following: 210, 213, 216, 221, 224, 227, 232, or 235.

STEP 5: **MBTK** - Make idle the assigned Digital Conference Trunks.

STEP 6: **ARSC**: - Allow Route Restriction Data for appropriate RSCs.

STEP 7: **ARRC** - Allow Trunk-to-Trunk Connections between Routes if necessary.

STEP 8: **ASTD** - Assign the following data when a Central Office Line is involved in a Conference: STM = 0, STS = 2, and ST = 1.

STEP 9: **AKYD** - Assign FKY 116 to a key (Call Hold Conference).

C-149 CS Report

General Description

This feature provides interface between the Telephony Server and the host computer, allowing the system to output the following traffic related data. The output data is sent to the host computer connected to the system enabling the user to calculate adequate system specification. LAN interface as well as RS-232C interface is available to be output.

- Outgoing Trunk Call - Complete (Charged)
- Outgoing Trunk Call - Complete (Not Charged)
- Outgoing Trunk Call - Complete (Not Charged)
- Outgoing Trunk Call - Incomplete (Called Party No Answer)
- Outgoing Trunk All Busy - Incomplete
- Incoming Trunk Call - Complete
- Incoming Trunk Call - Incomplete (Called Station Busy, All IRTs Busy)
- Incoming Trunk Call - Incomplete (Abandoned Call)

Note: This feature is only available in Australia.

Operating Procedure

No manual operation is required.

Service Conditions

1. The maximum number of CS Report information is 47000 calls (IMG/MMG/UMGi/RNIP).

   Note: Memory area for CS Report information is shared with SMDR information.
   Buffer memory usage rate of CS Report information is determined by SYS1, Index 307 (Default value is 50%). When changing this System Data, note the following points:
   - System Initialization is not required after changing this data.
   - When the memory area for CS Report is decreased, existing CS Report information in the decreased area is deleted.
   - When the memory area for CS Report is increased, existing SMDR information in the increased area is not deleted. Consequently, new CS Report information cannot be stored until existing SMDR information is output and the memory area becomes free.
   - When the memory area for CS Report reaches full, new CS Report information cannot be stored until existing CS Report information is output and the memory area becomes free.
   - SMDR buffer overflow alarm is applicable only to the memory area for SMDR.

2. CS Report information is automatically erased when the timer assigned by ASYD, SYS1, Index 95 expires.
3. Output can be selected between RS-232C and LAN interface (ASYDL, SYS1, Index 587, Bit 0). Output to LAN and RS-232C port all together is not available.
   a.) When outputting to LAN port
   - Output port can be allocated by ASYDL (SYS1, Index 587, Bit 6 to Bit 7).
   - Maximum port number is two.

   b.) When outputting to RS-232C port

- Output port can be allocated by ASYD (SYS1, Index 308, Bit 0 to Bit 7)

- Maximum port number is two.
- If allocated port number is the same as that of MCI (ASYD, SYS1, Index 29) or SMDR (ASYD, SYS1, Index 288), CS report cannot be output (MCI or SMDR information is printed).

Programming

Basic Programming Procedure (When output to RS-232C port)

STEP 1: **ASYD** - SYS1,

Index 18, Bit 4=1 (SMDR Incomplete Message Output In Service)

Index 32, Bit 1=1 (Record Tenant Data for SMDR/CS Report), Bit 2=1 (Record Selected Trunk type for SMDR/CS Report). Bit 7=1 (SMDR/CS Report In Service)

Index 33, Bit 3 to Bit 4 (Telephone Subject to Total Billing), Bit 5=1 (Billing for transferred Incoming Call is Split), Bit 6=1 (Billing for Transferred Outgoing Call is Split)

Index 34, Bit 0=1 (SMDR RS-232C output is in service) When LAN is used, Bit 0=0

Index 58, Bit 7=1 (Specific Attendant Number is output to SMDR is Valid)

Index 67, Bit 7=0 (fixed, SMDR Spending Designation)

Index 95, Bit 0 to Bit 3: When billing data output failure has occurred to SMDR (RS-232C), the unit digit of the time in which the billing data can be saved; 0-9 (10 minutes basis) Bit 4 to Bit 7: When billing data output failure has occurred to SMDR (RS-232C), the unit digit of the time in which the billing data can be saved; 0-9 (100 minutes basis).

Note: When LAN is used, Index 95, Bit 0 to Bit 3 are not used.

SYS1, Index 296, Bit 0=1 (Output of Calling Number (ANI information) is In Service)

SYS1, Index 300, Bit 7; Output information of Calling Party intended the billing for when the received number exceeds 6 digits in CCIS/ISDN tandem connection. Bit 7=1; Route Number + Trunk Number is output.

SYS1, Index 307, Bit 0 to Bit 3: Assign Buffer memory usage rate of CS Report information.

SYS2, Index 3, Bit 0=1: SMDR is In Service

STEP 2: **AIOC** - Assign the function and attribute data of IOC port.

Note: This assignment is not needed in the case of using LAN interface.

IOC:	IOC PORT NUMBER (Assigned by ASYD, SYS1, Index 308)
TERMINAL:	SMDR
SPEED:	1200/2400/4800/9600 bps
PARITY BIT:	Non Parity
STOP BIT:	2 Bits
CHARACTER BIT:	8 Bits

STEP 3: **ASFC** - Assign 1 to SFI 14 for allowing SMDR service.

STEP 4: **ARTD** - Assign 1 to CDN 10 (SMDR). Assign appropriate data for CDN 16 (SMDR2)

Note: To change the data for existing route (RT), initialize or reboot the trunk device such as a circuit card or MG.

STEP 5: **ARTI** - CDN: 70 - Collected Information of CS Report and SMDR billing.

Note: If this data is not assigned, all the information of CS Report and SMDR billing is to be output.

0: CS Report and SMDR billing for Outgoing/Incoming Trunk Call is to be collected
1: SMDR billing for Outgoing Trunk Call is to be collected
2: SMDR billing for Incoming Trunk Call is to be collected
3: SMDR billing for Outgoing/Incoming Trunk Call is to be collected
4: CS Report for Outgoing Trunk Call is to be collected
5: CS Report and SMDR billing for Outgoing Call is to be collected
6: CS Report for Outgoing Call and SMDR billing for Incoming Call is to be collected
7: CS Report for Outgoing Call and SMDR billing for Outgoing/Incoming Call is to be collected
8: CS Report for Incoming Call is to be collected
9: CS Report for Incoming Call and SMDR billing for Outgoing Call is to be collected
10: CS Report and SMDR billing for Incoming Call is to be collected
11: CS Report for Incoming Call and SMDR billing for Outgoing/Incoming Call is to be collected
12: CS Report for Outgoing/Incoming Call is to be collected
13: CS Report for Outgoing/Incoming Call and SMDR billing for Outgoing Call is to be collected
14: CS Report for Outgoing/Incoming Call and SMDR billing for Incoming Call is to be collected
15: CS Report and SMDR billing for Outgoing/Incoming Call is to be collected

Note: To change the data for existing route (RT), initialize or reboot the trunk device such as s circuit card or MG.

STEP 6: **ASAT** - Assign the data when Attendant Console is connected.
TN: Tenant Number (1-63)
ACC: Access Code
ATNATTENDANT NUMBER (1-63, Assign the number that is assigned in ATRK - TK)
ASYDL - SYS1,
Index 584, Bit 0 to Bit 4 Assign Polling Cycle within the Self Node. [For UMGi/RNIP system]

Call/day	System Data	Call/day	System Data
~5,400,000	01	~600,000	09
~2,700,000	02	~540,000	0A
~1,800,000	03	~490,000	0B
~1,350,000	04	~450,000	0C
~1,080,000	05	~400,000	0D
~900,000	06	~385,000	0E
~770,000	07	~360,000	0F
~675,000	08	~337,000	10

Index 586, Bit 0 to Bit 7 Assign Upper limit of polling buffer rate. [For UMGi/RNIP system] **Note 1**
Index 587, Bit 0=0 (Interface type for CS Report Output is RS-232C)

Index 641, Bit 0: Designation of output numbers for SMDR ICS format. (0/1 = Physical (Station) Number/Telephone Number) **Note 2** Bit 3: Designation of Output Routes for SMDR ICS format. (0/1 = Physical Route/Logical Route)

Note 1: When the value is 00H (default), the upper limit is 50%.

Note 2: When Bit 0=1 and Telephone Number is within 6 digits, Telephone Number is to be output.

STEP 7: **ASYD** - SYS1, Index 308, Bit 0 to Bit 7. CS Report Output Port Designation. Assign 1 to the Bit corresponding to the Port Number of the IOC circuit card. **Note 3**

Note 3: This assignment should be done lastly. This assignment is not needed in the case of using LAN interface

Note: Up to 2 ports can be assigned. If more than 2 ports are assigned, CS Report data is output from the smaller numbered 2 ports.

Note: If allocated port number is the same as that of MCI (ASYD, SYS1, Index 29) or SMDR (ASYD, SYS1, Index 288), CS report cannot be output (MCI or SMDR information is output).

When output to LAN port

LAN interface as well as RS-232C interface is available to output. At first, do the Basic Programming Procedure (**1. Basic Programming Procedure**). And then go on to STEP 9.

STEP 1: **ASYDL** - SYS1,

Index 529, Bit 0 to Bit 1. Assign Parity Check Method for SMDR/MCI/CS Report with LAN Interface.
Index 584, Bit 0 to Bit 4. Assign polling cycle within the Self Node. [For UMGi/RNIP System]

Index 586, Bit 0 to Bit 7. Modify the upper limit of polling buffer usage rate (01% - 99%) if required. [For UMGi/RNIP system]
The default value is 50%. If the usage rate exceeds this limit, the Centralized Billing Office will not receive any billing information sent from local offices.

b7	b6	b5	b4	b3	b2	b1	b0
Tens				Unit			

Note: This setting applies to “Centralized Billing -CCIS [C-55]”, “Centralized Billing -FCCS (Polling Method)”, and “CS Report [For UMGi/RNIP system] [C-149]”.

SYS1, Index 587, Bit 0=1: Interface type for CS Report Output is LAN interface.
Bit 1: System Message Output for CS Report with LAN interface related to connection failure in system changeover. 0/1 = Out/In Service
Bit 6: Output equipment A for CS Report with LAN interface. 0/1=Out/In Service
Bit 7: Output equipment B for CS Report with LAN interface. 0/1=Out/In Service

SYS1, Index 641. Output Number and Route Number designation that is applied for SMDR - ICS format only.
Bit 0=0/1 (Station Number/Telephone Number) **Note 4**
Bit 4=0/1 (Physical Route Number/Logical Route Number)

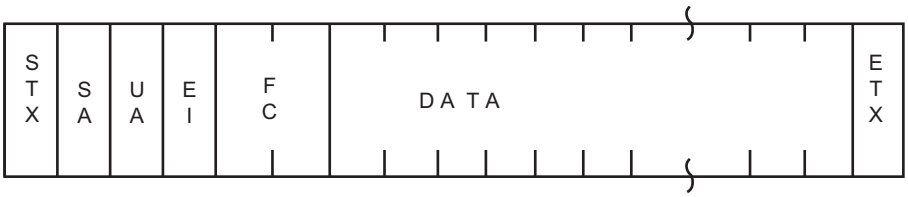
Note 4: If the number of digits of Telephone Number is within 6 digits, the Telephone Number will be output when Bit 0=1.

SYS1, Index 916/917. CS Report LAN Interface port number. When data is 00H, 60080 is used.

Message Format

1. RS-232C Interface

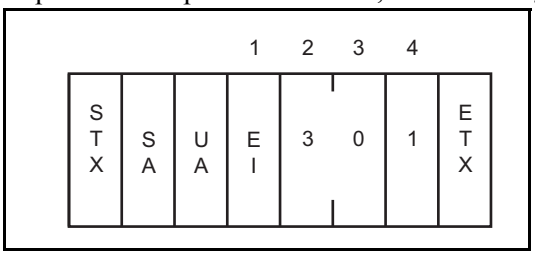
Message format via RS-232C is shown below:



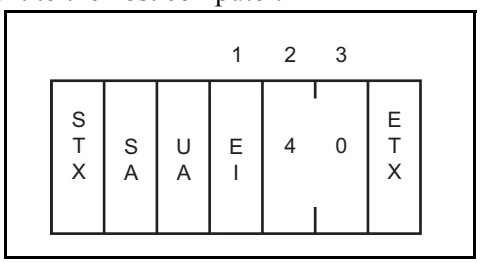
STX: Start of Text
EXT: End of Text
SA: System Address
UA: Unit Address
EI: Entry Index (P: CS Report message)
FC: Function Code **Note 5**

Note 5: The Meaning of Function Code is as follows:
30: Request for sending traffic data
40: No data
41: Outgoing Trunk Call - Complete (Charged)
42: Outgoing Trunk Call - Complete (Not charged)
43: Outgoing Trunk Call - Incomplete (Called Party No Answer)
44: Outgoing Trunk Call - Incomplete
45: Incoming Trunk Call - Complete
46: Incoming Trunk Call - Incomplete (Called Station busy, All IRTs Busy)
47: Incoming Trunk Call - Incomplete (Abandoned Call)

2. When the host computer requests CS Report information, the following message is sent to the PBX.

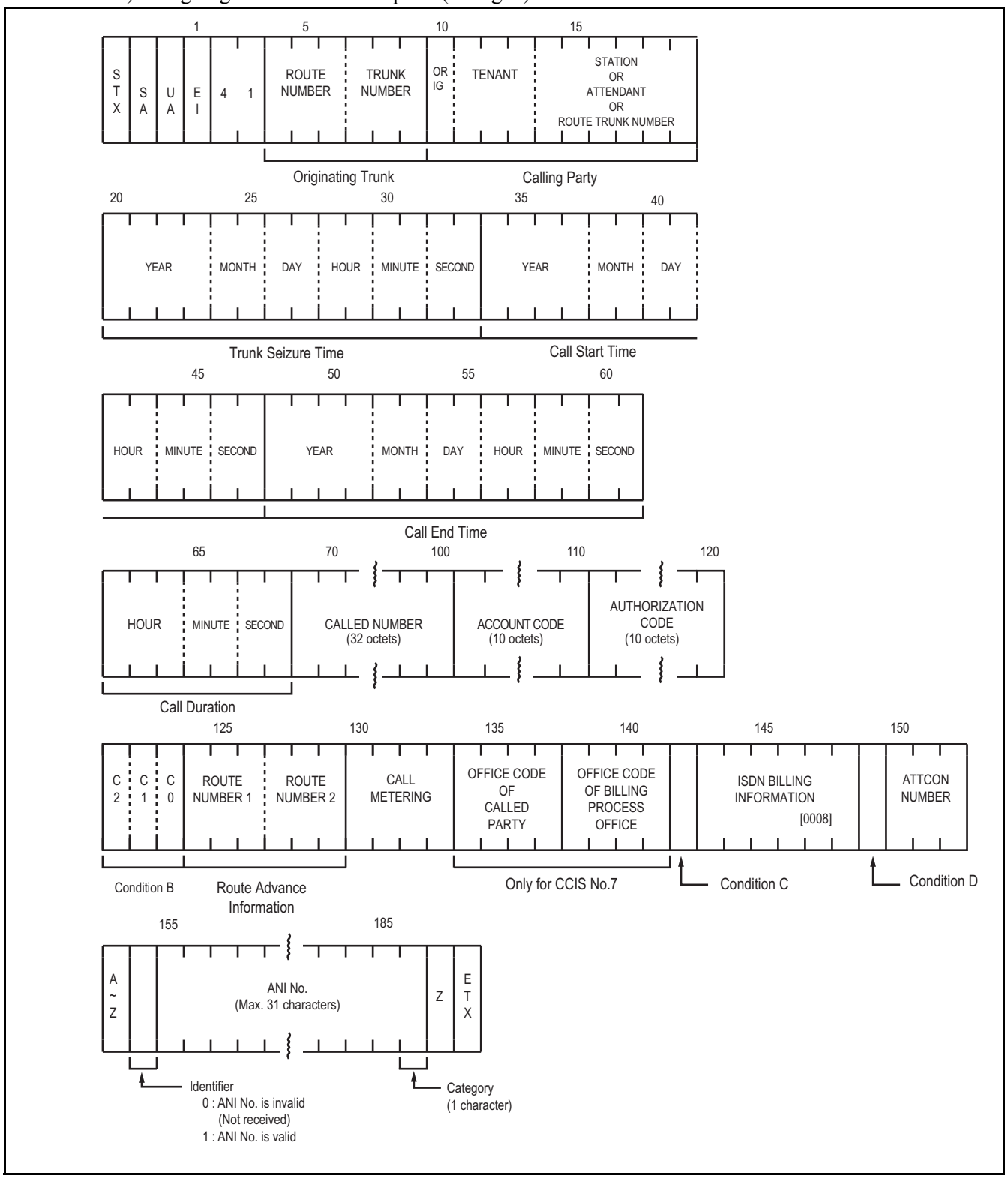


3. When the system is requested to send CS Report information, if no CS Report information is stored, the following message is sent to the host computer.

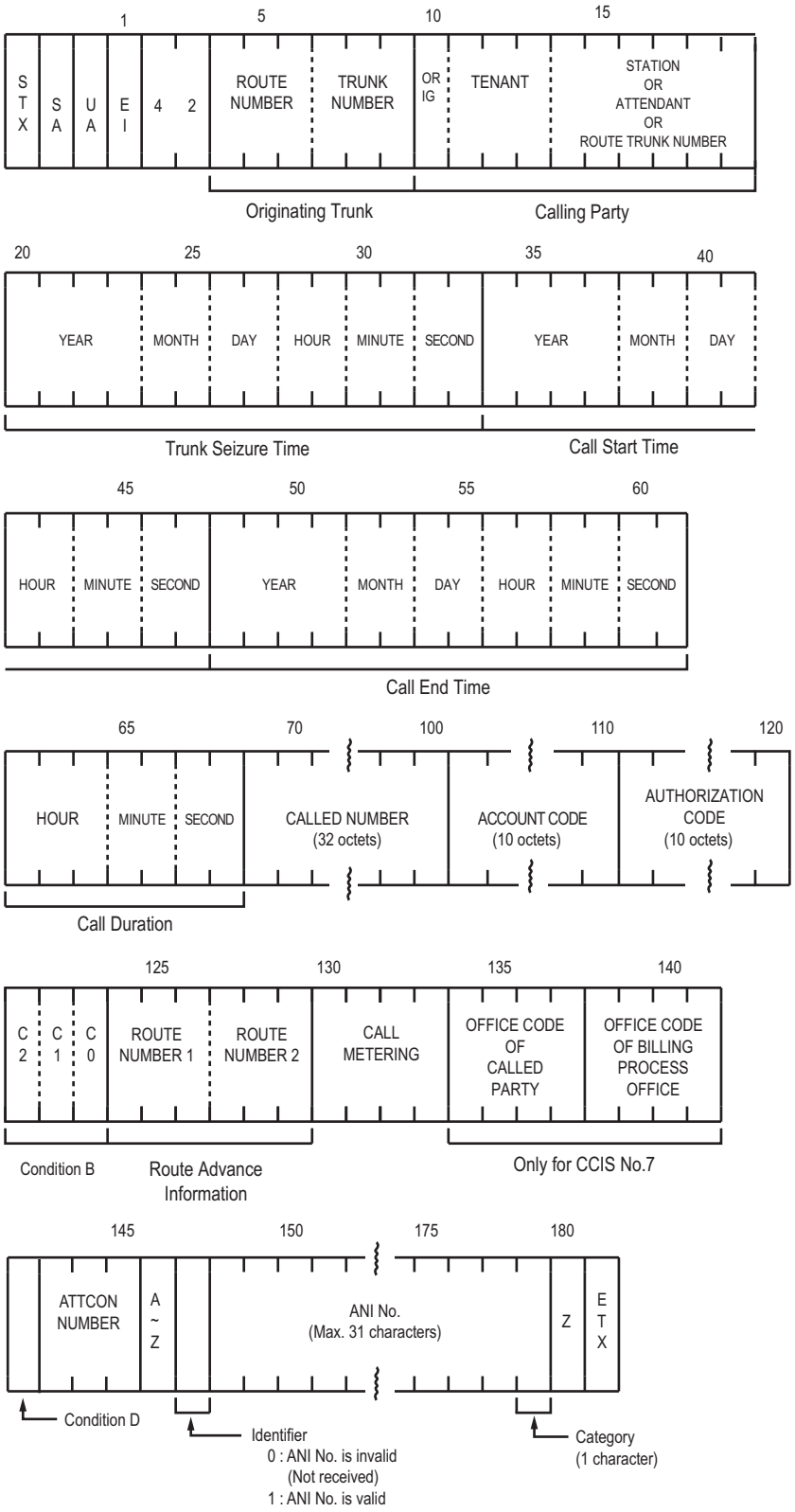


4. CS Report information sent from the PBX to the host computer is shown below:

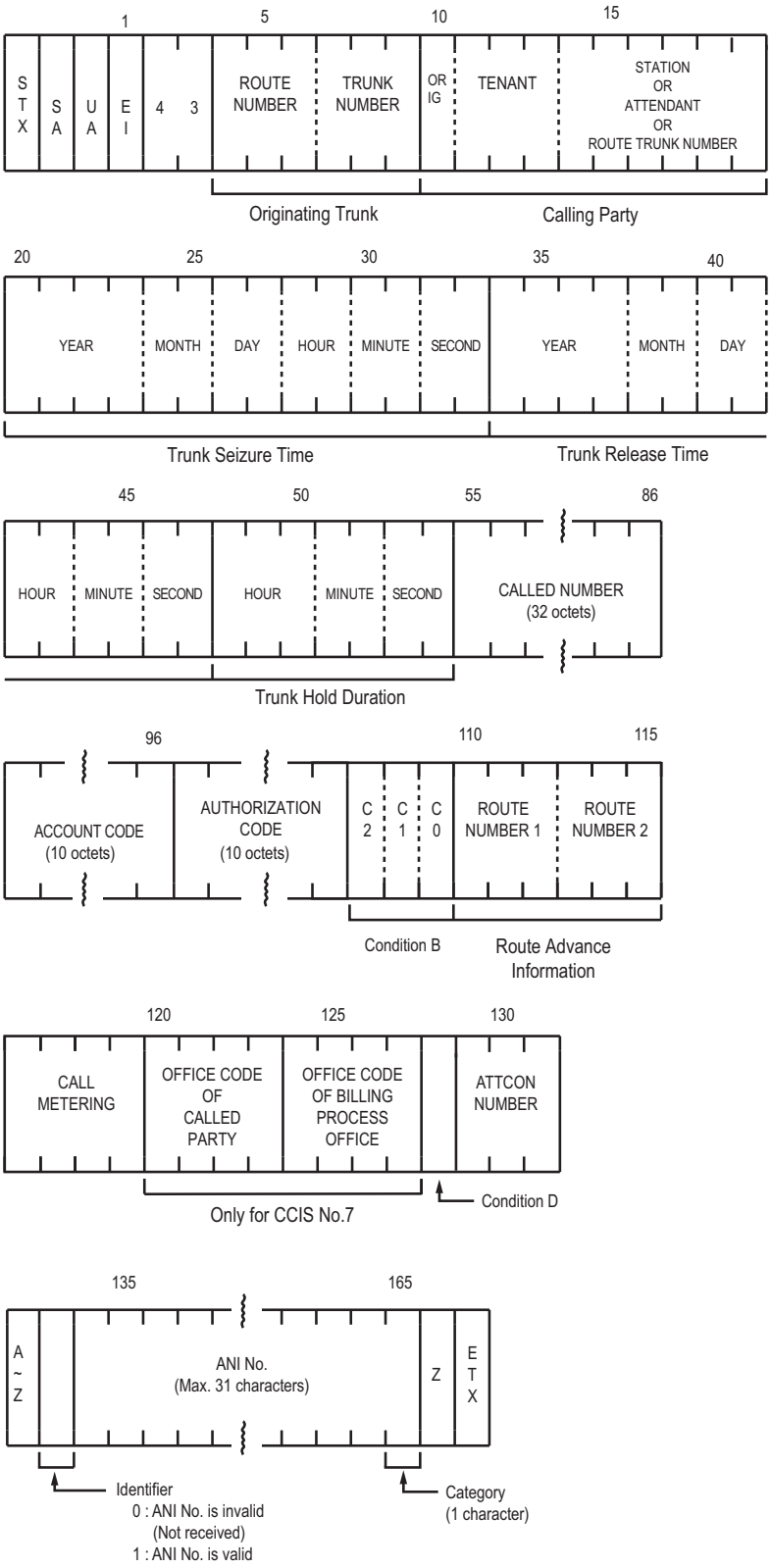
a.) Outgoing Trunk Call - Complete (Charged)



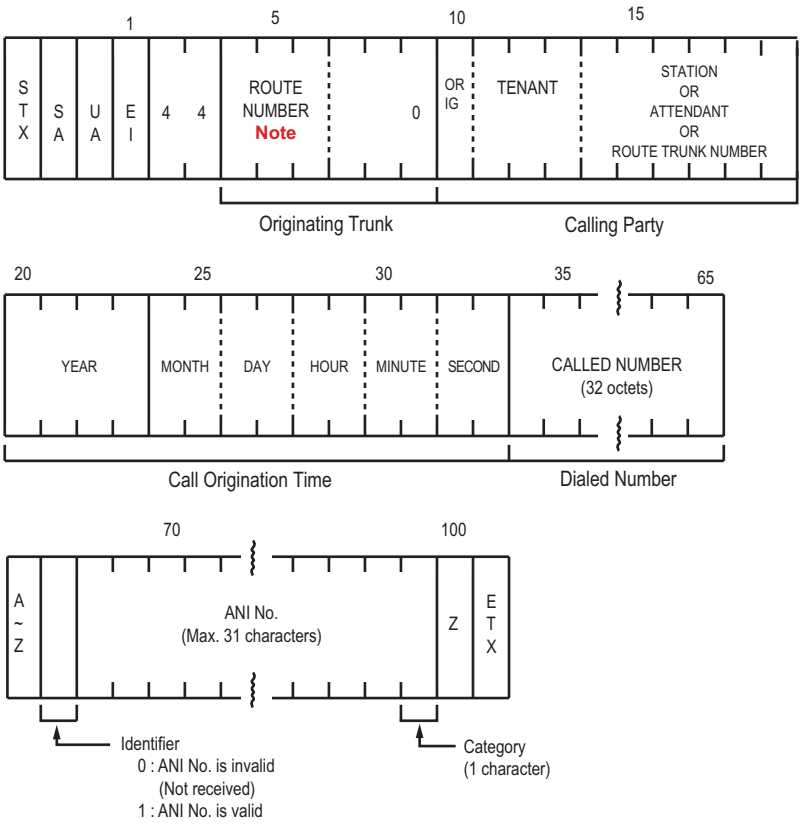
b.) Outgoing Trunk Call - Complete (Not Charged)



c.) Outgoing Trunk Call - Incomplete (Called Party No Answer)

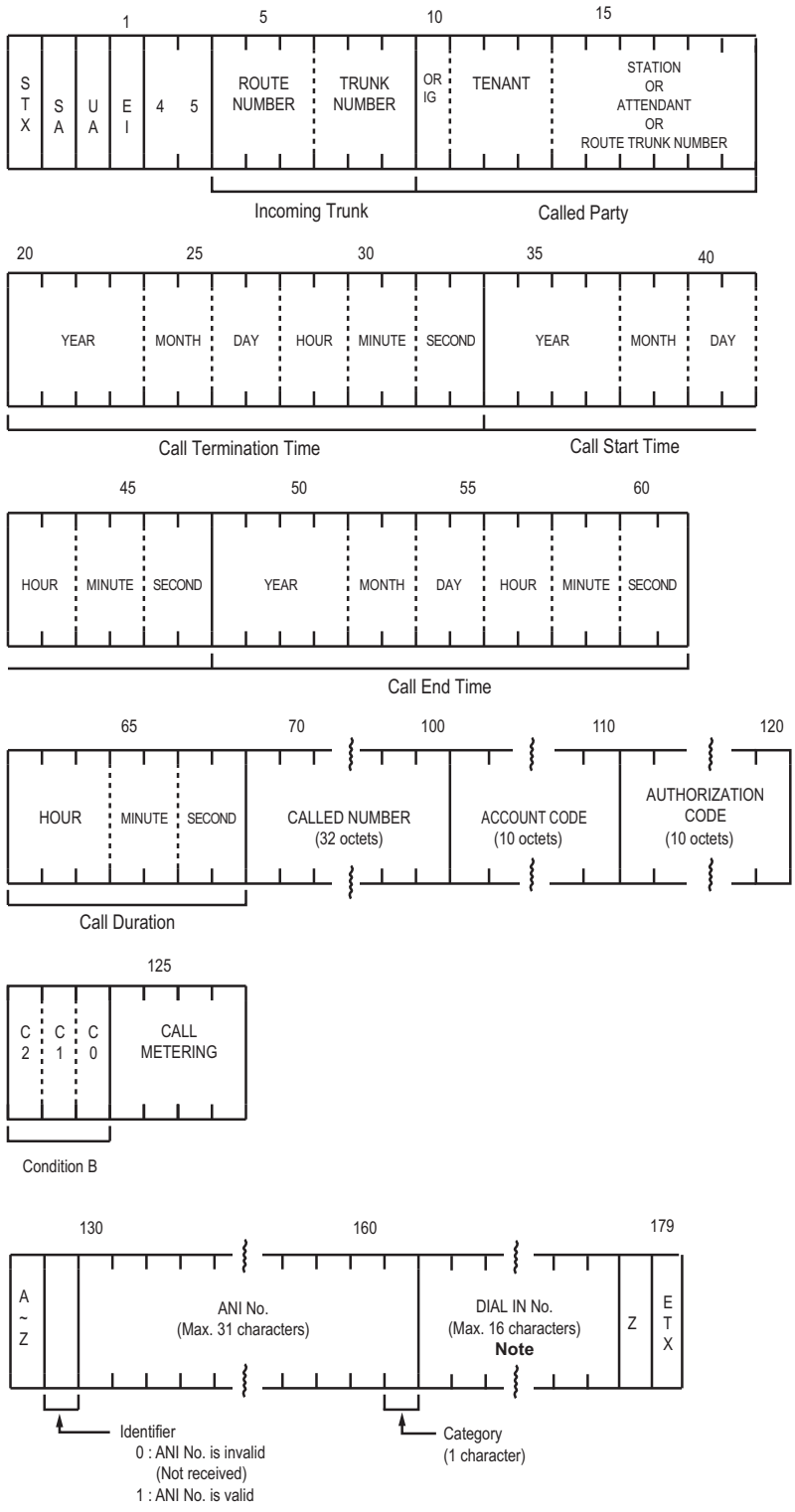


d.) Outgoing Trunk All Busy - Incomplete

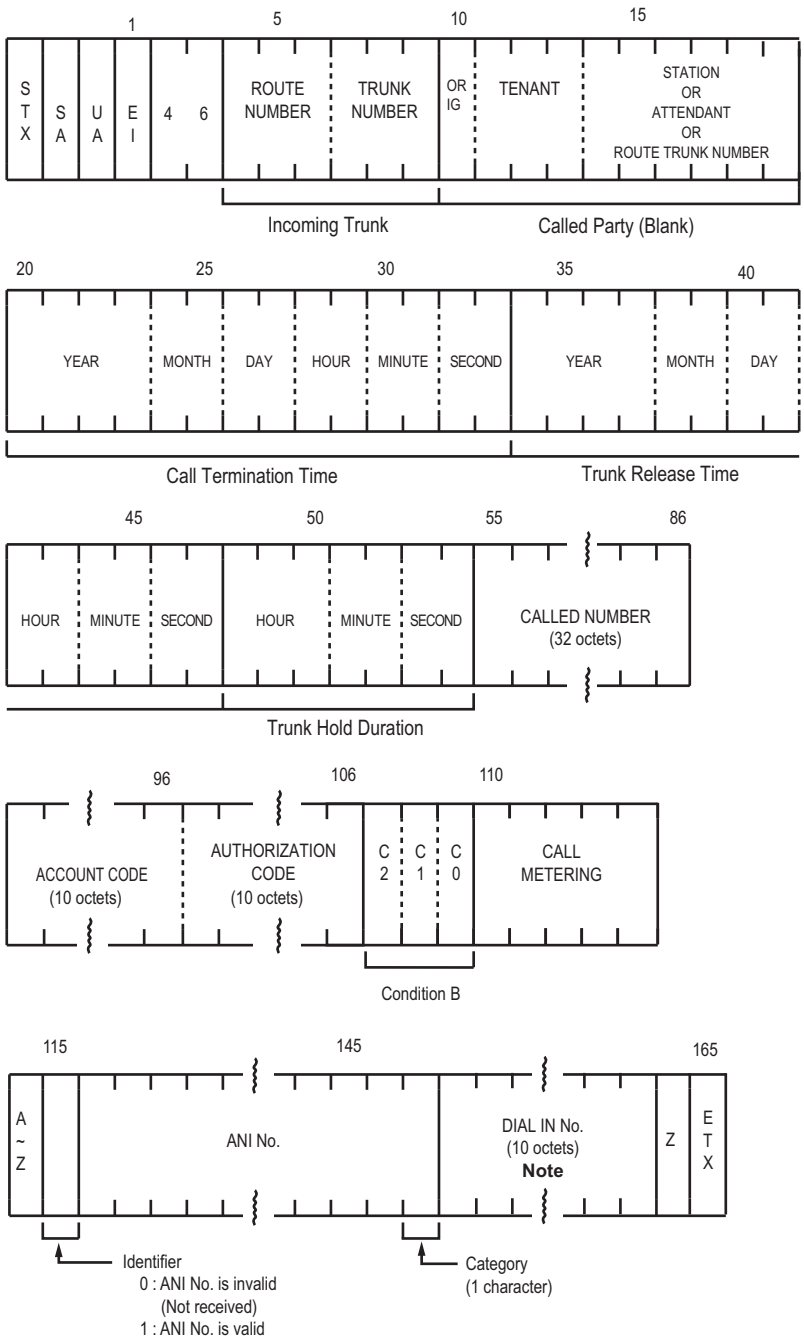


Note: In LCR/LCRS service, first selected route number is output.

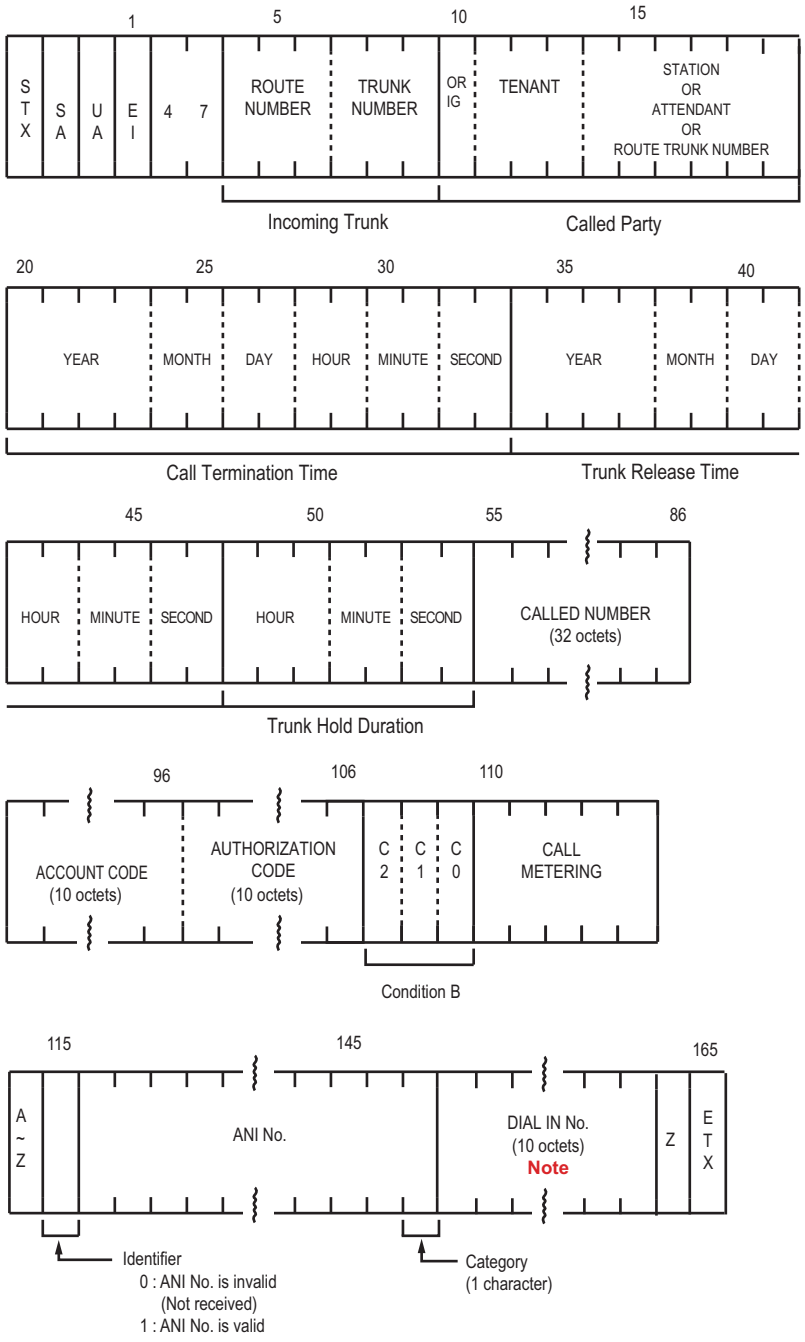
e.) Incoming Trunk Call - Complete



f.) Incoming Trunk Call - Incomplete (Called Station Busy, All IRTs Busy)



g.) Incoming Trunk Call - Incomplete (Abandoned Call)



ORIG (Originating Source Identification):

0 = Calling/Called Party is a Station
1 = Calling/Called Party is Attendant
2 = Calling/Called Party is an Outside Party

Condition B:

C0 =

0: Regular Call (Direct Dialing from a Station)
1: Regular Call (Call is made through an ATTCON)
2: Call through route advance (Direct Dialing from a Station)
3: Call through route advance (Call is made through an ATTCON)
4: Call through Least Cost Routing (Direct Dialing from a Station)
5: Call through Least Cost Routing (Call is made through an ATTCON)
6: The first 6 digits of the converted number are output to Called Number (Direct Dialing from a Station)
7: The first 6 digits of the converted number are output to Called Number (Call is made through an ATTCON)

C1 =

0: Normal Call
1: Call originated by OG Trunk Queuing
2: Call originated by dialing with Account Code
3: Call originated by OG Trunk Queuing and dialing with Account Code
4: Call originated by Call Forwarding - Outside

C2 =

0: Call is not transferred
1: Call is transferred
4: The last Station to which the call is transferred

Condition C:

0: ISDN Charge Notification Not Available
1: Charge Notification Available (Unit: 0.1 cent)
2: Charge Notification Available (Unit: 1 cent)
3: Charge Notification Available (Unit: 10 cents)
4: Charge Notification Available (Unit: 1 dollar)
?: Charge Information Error (when exceeding the maximum value)

Condition D:

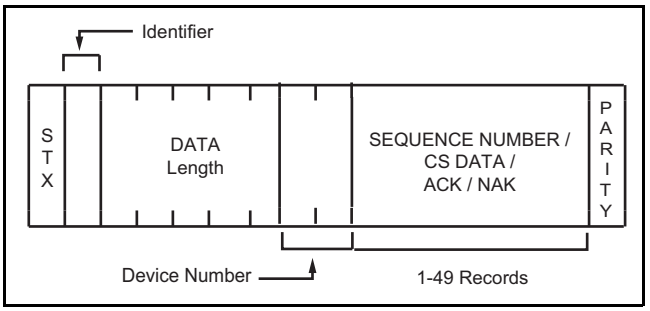
Blank: ATT Charge Notification Not Available
0: ATT Charge Notification Not Applicable
1: ATT Charge Notification Available

(2) LAN Interface

1. Interface between the Telephony Server and host computer is described below;

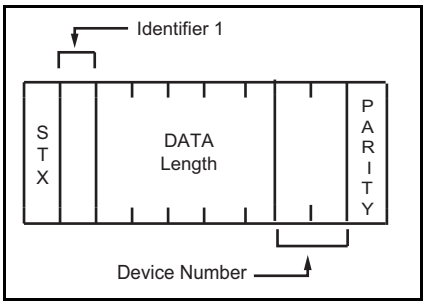
PROTOCOL	TCP/IP
INTERFACE	Ethernet
Telephony Server	SOCKET Interface
EXTERNAL DEVICE	Use the library of WinSock or Unix socket.
PORT NUMBER	60080
TRANSMISSION CODE	ASCII
CONTROL CODE	SYN: 16H Synchronous Idle STX: 02H Start of Text ETX: 03H End of Text ACK: 06H Acknowledgements NAK: 15H Negative Acknowledgements

2. Basic text format is shown below:

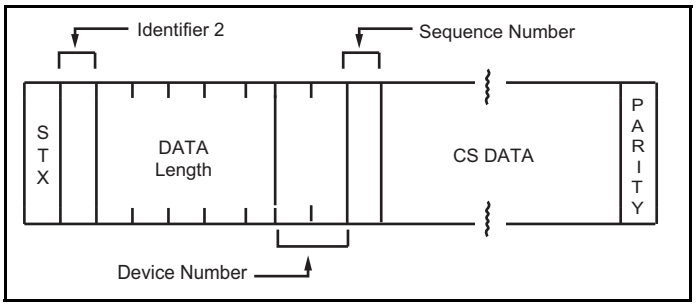


Note: The information of one record is described in (1) RS-232C.

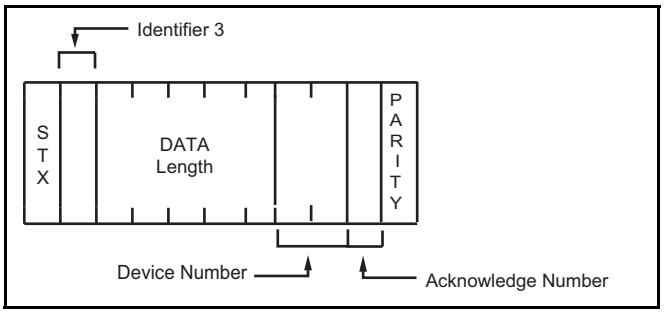
3. CS Report information is shown below.
   a.) Data Request Text: The client requests the data of server.



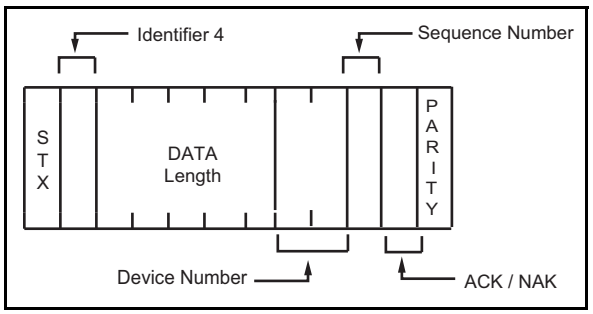
b.) Send CS Data Text: The CS data is sent to client.



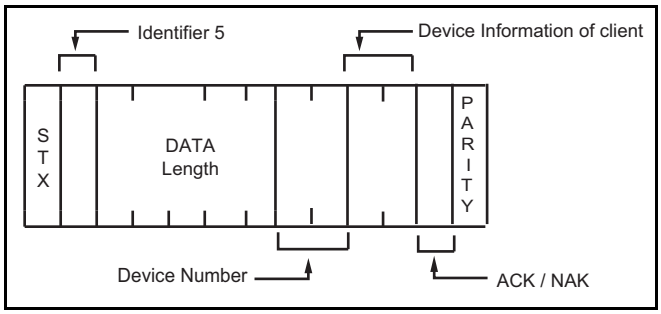
c.) Response Text: When the server has no data to send, this text is sent to client. This text is also sent to client as a response to 'Status Monitor Text'.



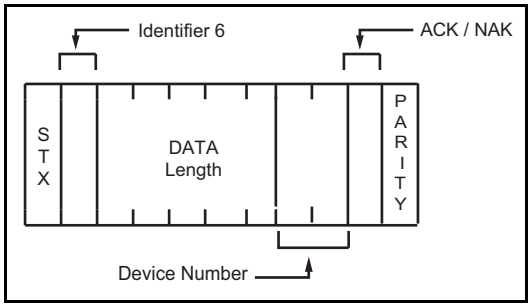
d.) Response Text: This text is sent to server as a response to 'Send CS Data Text'.



e.) Status Monitor Text: This text is the report of the status of server monitored from client side or sever side. At the same time, the condition of client is also reported. This text is sent to server. The server sends 'Response Text' as a response to this text.



f.) Request Disconnection Text: The client sends this text to disconnect from client to server. The disconnection is performed after receiving this text.



SYN: Synchronous Idle
Identifier: Identifies whether the data is sent or received
DATA Length: Size of Data
Device Number: Client sends the device number to server, and server sends client's device number to which the text is sent.
00-Client A
01-Client B
Sequence Number: Serial Number of data (Number is cyclic).
Acknowledge Number: The information of server.

0: Not used
1: Send no data as a response to Request Data from client
2: Acknowledgement Text as a response to Status Monitor Text
3: Parity Error with the received text
4: Device Number Duplication
5: Error with received text (receive invalid text)
6: The state which can not be Sent and Received
7-9: Not used

ACK/NAK:

ACK:Parity Check Result is OK
NAK:Parity Check Result is NG

Device Information of Client:

DEVICE INFORMATION OF CLIENT	ALARM	INFORMATION
00	-	Normal (Text is printed when initializing)
01	SUP	Action is stopped
02	SUP	Error when initializing
03	SUP	Error with Database when receiving data
04	SUP	Error with Database when saving data

DEVICE INFORMA-TION OF CLIENT	ALARM	INFORMATION
05	SUP	Error with Database when making report
06	SUP	Alarm to capacity over with database (more than 1.6GB)
07	SUP	Alarm to insufficient memory with Hard disk (less than 100MB)
08-50	-	Not used
51	-	Start of action
52	-	Release from error when initializing
53-99	-	Not used

C-150 Call Block

General Description

This feature is a part of Caller ID Service. This feature restricts termination of Incoming Calls from the preassigned Calling Numbers. This feature is available by designating the Calling Number by PCPro.

Call Block is used to restrict an Incoming Call that has a Restriction Number (Physical Station Number, Telephone Number or the Caller ID Number, which are preset by each station). The Restriction Number can be set or canceled at each station by using the Access Code or the Feature key. The Called station can send Busy Tone to the caller. For the Station to Station Calls, Incoming Call can be restricted by using the Access Code or the Feature key (Not by ACID command).

Note: This feature is not available in Australia.

Operating Procedure

When using Access Code

To assign the Restriction Number (the newest Incoming Call is to be registered as the Restriction Number):

1. Lift the handset.
2. Dial the Access Code (ACC for registration of Restriction Number by the last Incoming Call).
3. Receive Service Set Tone. **Note 1**
4. The newest Incoming Call Number is registered as the Restriction Number.

When using Feature key

To set the Restriction Number (the newest Calling Number is to be registered as the Restriction Number):

1. Press the CALL BLOCK Feature key when the station is idle, a call is terminating to the station, or when the station is in a two-party connection.
2. The newest Calling Number is displayed for five seconds on the LCD.
3. Press the CALL BLOCK Set key (* or # key, depending on System Data) while the number is displayed.
4. The Call Block LED lights red.
5. The newest Incoming Calling Number (the Physical Station Number or the received Calling Number) is registered as the Restriction Number.

To assign the Restriction Number by dialing an Optional Number

1. Lift the handset.
2. Dial the Access Code (ACC for registration of Restriction Number by dialing an Optional Number).
3. Receive Special Dial Tone.
4. Dial the number to be restricted.
5. Receive Service Set Tone after the preassigned Inter-digit Timer (default is six seconds). **Note 1**
6. The Dialed Number is registered as the Restriction Number.

To cancel the Registered Restriction Number

1. Lift the handset.
2. Dial the Access Code (ACC for canceling the Restriction Number).
3. Receive Service Set Tone.

To set the Restriction Number by dialing an Optional Number

1. Lift the handset.
2. Press the CALL BLOCK Feature key.
3. Receive Special Dial Tone.
4. Dial number to be restricted.
5. Receive Service Set Tone after the preassigned Inter-digit Timer (default is six seconds). **Note 1**
6. The CALL BLOCK Feature key lights red.
7. The Dialed Number is registered as the Restriction Number.

When using the PCPro command to confirm the Registered Restriction Number

1. Start the DCBD command.
2. Enter the Telephone Number or LENS.
3. Select to display the Registered Restriction Number.
4. The Restriction Number, which is registered to the line corresponding to the specified Telephone Number or LENS, is displayed.

To confirm the Registered Restriction Number

1. Press the CALL BLOCK Feature key when the station is idle (when the newest received Calling Number has been registered, the number is displayed on the LCD).
2. Continue pressing the CALL BLOCK Feature key in the same manner to confirm the other Restriction Numbers. The other Restriction Numbers are displayed in consecutive order.
3. The Registered Restriction Number is displayed for five seconds.

To cancel the registered Restriction Number

1. Confirm that the Restriction Number to be canceled is displayed on the LCD.
2. Press the * or # key for the Restriction Number to be cleared (depends on the System Data), while the corresponding number is displayed.
3. When all the Restriction Numbers are deleted, the LED of CALL BLOCK Feature key (or Soft key) goes OFF.

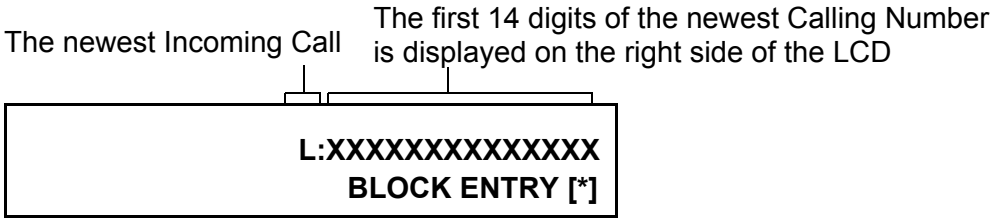
   Note: When a Desktop terminal is used as the station, the LCD displays are as shown on the next page.

4. LCD displays for Service Set, Confirmation, and Cancel are shown on the following page. Set/cancel can be selected as the "*" or "#" key, depending on System Data. Assignment of one of these keys as Cancel key, will assign the other as the Set key.

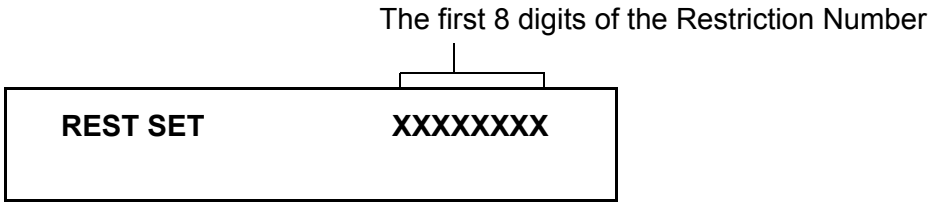
Note 1: When the Restriction Number has already registered up to the maximum number of five, Re-order Tone is heard. First cancel the former Registered Restriction Number, then set the new Restriction Number again from the beginning.

24 digit display of Desktop terminal

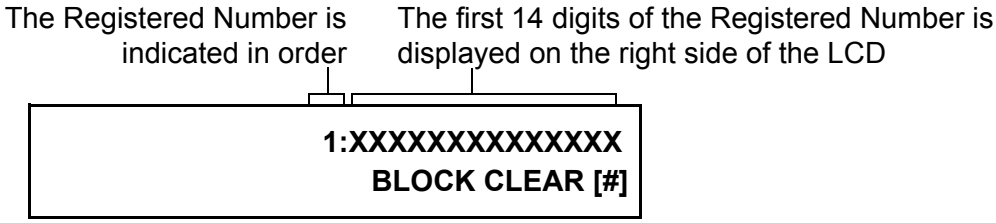
- When the last Incoming Call is displayed:



- When the Restriction Number is set:



- When confirming the Restriction Number:



- When the Restriction Number is canceled

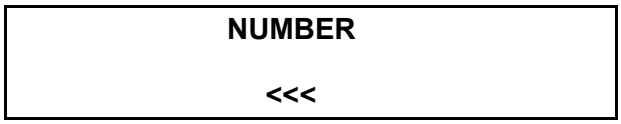


- When HELP key for CALL BLOCK of Soft key is pressed:

Page 1



Page 2



Service Conditions

When the Restriction Number is registered by PCPro

1. When the Calling Number is not received, this service is not activated.
2. The Calling Number for this service is assigned to the parameter CID of ACID. When only the first part of the Calling Number is assigned to the CID, this service is applied to any Calling Number that begins with the Assigned Number. For example, when "0471" is assigned to the CID, this service is applied to any Calling Number that begins with "0471" such as 0471-12-3456 and 0471-234-5678.
3. The following numbers cannot be assigned to the parameter CID of ACID:
   a.) The first part of the already Assigned Number
   b.) The number whose first part is the same as the already Assigned Number. For example, when "123" is already assigned to the CID, such numbers as "1", "12", "1234" cannot be assigned.
4. When the Calling Number is received via CCIS Line, this feature is available for the first 16 digits (provided that "1" is assigned to CDN:98 (CI) of ARTD).
5. This service is available only for stations (Single Line Telephone and Desktop terminal).
6. This service is available for the following cases:
   a.) When an Incoming Call is directly terminated to a station
   b.) When an Incoming Call is transferred to a station using Call Forwarding - All Calls/Busy Line/ Don't Answer
7. Call Forwarding - All Calls and Call Forwarding - Busy take precedence over this feature.
8. A maximum of 12 Calling Numbers can be assigned for a single station.
9. A CH-12COTB or PA-8COTBJ circuit card is required for Analog Caller ID:
10. This service is available only for the Stations Service Restrictions are released [SFI 165 (Call Block) = 1].
11. This service is not available for PS terminal or ISDN terminal.
12. Only one Restriction Number (a maximum of 32 digits) can be registered per station. A maximum of a 32 digit Restriction Number can be assigned up to five numbers per Desktop terminal and one number per Analog station. A maximum of 511 stations can be registered per System/IMG.
13. The Restriction Number cannot be confirmed from Single Line Telephones after registration.
14. To assign the Restriction Number, it is available to register the newest Incoming Call Number if the call is received in the Station-to-Station Calling, or a call is terminated via the Trunk which received the Calling Number. For example, if the station cannot receive the number of the last Incoming Call, the newest call that meets the conditions above is registered as the Restriction Number. Thus, it cannot be available to assign the newest Calling Number as the Restriction Number immediately after System Initialization.
15. When assigning the newest Incoming Call as the Restriction Number using Feature key from an idle station or a ringing station, the newest Calling Number to the My Line is registered as the Restriction Number (then the Feature key on the Controlling Terminal lights red). Thus, on a Desktop terminal whose Prime Line is not My Line, the Displayed Number is not registered when a call is terminated at Prime Line. The newest Calling Number terminating at My Line is registered in place of the Displayed Number.
16. When a call is transferred, the Telephone Number of the Intermediate Station is to be registered as the newest Incoming Call Number even if that station releases the connection. Only the first Terminating Number is to be the Restriction Number.
17. This service is activated corresponding to each line. In case a Desktop terminal containing some Lines, the Restriction Number is effective only for the line to be registered the Number.
18. When confirming the Registered Restriction Number, the number registered on My Line is displayed for five seconds on the LCD.

19. This service is available only when the predetermined Restriction Number and the receiving Physical Station Number/Telephone Number or the Calling Number are exactly the same.
20. In this service, a Telephone Number is used to recognize the Station Incoming Call, however, a Physical Station Number is used for the station which is not assigned the Telephone Number.
21. When an Incoming Call via CCIS is terminated, a maximum 16-digit Calling Number (Route Data CI = 1) can be received. If the Telephone Number is more than 17 digits, the Incoming Call is not restricted.
22. There is no need to register ACC for a Central Office Trunk Call (ex. "9") when assigning Restriction Number by dialing operation.
23. In case Restriction Numbers are registered up to the maximum number, a previous Restriction Number must be canceled before setting the new Restriction Number on a station. (It is impossible to overwrite the Restriction Number.)
24. Sending Tone, which is used when originating a call from a station on which Call Block is set, can be selected on the system basis (ASYD, SYS1, Index 322, Bit 5).
25. When initialization of the circuit card is activated, the Restriction Number already registered is not cleared.
26. When System Initialization is activated, the Restriction Number already registered is not cleared. However, when initialization of System Data loading is activated, the Loaded Number (former Restriction Number), not the newest Restriction Number, is to be registered as new Restriction Number.
27. Restriction Numbers, which users set from the terminals, can be checked by the DCBD command.
28. This service is activated to each line. When a Desktop terminal is used, Restriction Numbers can be registered at Multiple Lines. However, to clear the Restriction Number using the * or # key is effective for only the number registered to My Line.
29. When confirming the Registered Restriction Number, the oldest number registered on My Line is displayed for five seconds on the LCD. When the Multiple Restriction Number is assigned, press CALL BLOCK Feature key to change the display to the later Registered Number, the Displayed Number is changed in order as the key is pressed.
30. When deleting the Restriction Number with the Access Code from a Desktop terminal, the oldest Restriction Number is deleted.
31. When five Restriction Numbers are registered at a line, the assignment of the already Registered Number is available.
32. System Data decides the * key or the # key as the Restriction Number Delete key. When the Delete key is decided, the other one is used as Set key.

    Note: The Set/Delete key is available for only five seconds (while the number is displayed).

33. When the correct ELC card is not used, the indication for the operation is not displayed at the middle line of the LCD. In this case, Restriction Number is set or deleted by Set/Delete key (* or #).
34. When all Calling Numbers are deleted, the CALL BLOCK Feature key goes OFF.
35. ACID applies only to inbound Trunk Calls with ANI.
36. ACID will not light a Feature key if the Telephone is equipped with one.
37. ACID will operate via the FCCS Network for Tandem Trunk Calls.
38. Feature keys and SIDAs will operate for Physical and Logical Stations, as well as Inbound Trunk Calls with ANI. It will also light the Feature keys for both.
39. When an Incoming Call via MFC trunk is terminated, this service is not available.

40. The COT Card for receiving the Caller ID information is shown below:

CIRCUIT CARD NAME	PROGRAM	ISSUE
CH-12COTB	SP-4065 12COTB PROG-A	
PA-8COTBJ	SW-317 8COTBJ PROG-A	

41. The LC Card for notifying the Caller ID information is shown below.

CIRCUIT CARD NAME	PROGRAM	ISSUE
PA-16LCBY	SP-3396 16-LCBX PROG-A	

Interactions

1. Call Forwarding - All Calls [C-5] takes precedence over this feature.
2. If this service is set at both the station and PCPro, the station has priority.
3. When the data on a station is changed by FOLLOW PHONE [F-31] Service, the Registered Restriction Number on the station is carried over to the next user.
4. When a station is deleted, the Registered Restriction Number on the station is also deleted.
5. When activating with DUAL STATION CALL [D-156], CALL BLOCK is available only for the Main station.

Programming

When the Restriction Number is registered by PCPro

ACID - Assign the Calling Number for this service.

TYPE 1

TN: Tenant Number

STN: Station Number

CID: Calling Number (a maximum of 28 digits)

SRV: 5 (Call Block)

Note: The Restriction Numbers assigned by this command cannot be checked by DCBD.

STEP 1: **ASYD** - SYS1, Index 322, Bit 5. Assign the sending Tone when Call Block is set.

0/1: DT/SPDT

SYS1, Index 80, Bit 1. Decide the Cancel key for the Registered Restriction Number.

0/1: # key, * key

STEP 2: **ASPA** - Assign the Access Code for Call Block, SRV = "SSCA: Service Code Appendix", SIDA = 111 (ACC for the latest Incoming Call is to be the Restriction Number), SIDA = 112 (ACC for assigning the Restriction Number by dialing an Optional Number), SIDA = 113 (ACC for canceling the Restriction Number)

STEP 3: **ASFC** - Assign a Service Feature Restriction Class to allow CALL BLOCK [C-150] for the Stations. SFI 165 (Call Block), RES = 1 (to release the restriction)

STEP 4: **AKYD** - Assign the key data for CALL BLOCK on the Feature key of Desktop terminal. FKI = 1 (Feature Key) and CALL BLOCK Feature key [128]

STEP 5: **ADSL** - Assign the Soft key data of Call Block for Desktop terminal according to the following

parameter:

SKP: 0-15 (Soft Key Pattern)

SN: 0, 1, 4 (Call Situation Number)

SKN: 1-16 (Soft Key Number)

FKY: 128 (Call Block Key)

STEP 6: **ADKS** - Assign the Soft key data of Call Block for Desktop terminal according to the following parameter:

TN: Tenant Number

STN: Station Number

SKP: 1-15 (Soft Key Pattern of Call Block assigned in ADSL)

LKP: 1-2 (Location of Key Pattern)

PG: 0/1 (Soft Key/Feature Key is allocated for Page Change Key)

STEP 7: **ASYD** - Assign the routine diagnosis program start time SYS1, Index 87: Hour, Index 88: Minutes

SYS1, Index 90, Bit 1=1: Routine Backup is to be executed

SYS1, Index 304, Bit 6=1: Data Backup of Call Block

C-151 Call Return

General Description

This feature allows a Desktop terminal or an analog station to save the received Calling Number of an incoming trunk call and the received Station Number of an incoming station call. The following two types of saving features are available:

- Automatic Saving: The received Calling Number/Station Number is automatically saved when the Station rings. The Stored Number can be dialed by pressing the Feature key or dialing the Access Code.
- Manual Saving: The received Calling Number is manually saved by the Desktop terminal user while in conversation. The Stored Number can be dialed by pressing the Feature key.

Note: Call Return to Station Number of an incoming station call is not available in Hotel System.

Operating Procedure

Automatic Saving

1. Saving Calling Number:
   a.) A call is terminated to an Analog Station or My Line of the Desktop terminal.
   b.) The received Calling Number is automatically saved. When the Station is the Desktop terminal with the CALL RETURN Feature key, the received Calling Number is displayed on the LCD and the CALL RETURN Feature key lights red.

   Note: A maximum of three Calling Numbers are stored for the Desktop terminal/A maximum of one Calling Number is stored for an Analog Station.

2. Displaying Stored Calling Number (for the Desktop terminal with the CALL RETURN Feature key):
   a.) While in idle state, press the CALL RETURN Feature key.
   b.) The saved Calling Number is displayed for five seconds. When multiple numbers are stored, the display changes each time the CALL RETURN Feature key is pressed. The number is displayed from the newest to the oldest. When the indication reaches the Oldest Number, the newest number appears again by pressing the CALL RETURN Feature key.
3. Call Origination:
   - When using the CALL RETURN Feature key on the Desktop terminal
     a.) Display the desired number by pressing the CALL RETURN Feature key. (Refer to #2.)
     b.) Lift the handset and receive Dial Tone.
     c.) Press the CALL RETURN Feature key. The Displayed Number is dialed.

     Note: The Dialed Number is automatically deleted. When all the Stored Numbers are deleted, the CALL RETURN Feature key goes OFF.

   - When using Access Code
     a.) Lift the handset and receive Dial Tone.
     b.) Dial the Access Code for Call Return. The last number saved is dialed.

     Note: The Dialed Number is not deleted.

4. Deleting Stored Number (for the Desktop terminal with the CALL RETURN Feature key)
   a.) Display the desired number by pressing the CALL RETURN Feature key. (Refer to #2.)
   b.) Press the * or # key (according to System Data assignment). The Displayed Number is deleted and CANCEL is displayed on the Desktop terminal LCD.

Note: When all the Stored Numbers are deleted, the CALL RETURN Feature key goes OFF.

Note: The Set/Delete key (* or #) is available for only five seconds (while the Calling Number is displayed).

- **Manual Saving (equal to SAVE AND REPEAT -D [S-24D]**

1. Saving Calling Number:
   a.) Press the S&R Feature key while in conversation using My Line or Sub Line.
   b.) The received Calling Number is saved and the S&R Feature key lights red.

Note: One Calling Number can be saved. When a Calling Number has already saved using SAVE AND REPEAT -D, the saved Calling Number is overwritten.

2. Displaying Stored Calling Number (for the Desktop terminal with the S&R Feature key):
   a.) While in idle status, press the S&R Feature key.
   b.) The saved Calling Number is displayed for five seconds.
3. Call Origination:
   a.) Lift the handset and receive Dial Tone.
   b.) Press the S&R Feature key. The Stored Number is dialed.

Note: The Dialed Number is automatically deleted and the S&R Feature key goes OFF.

Service Conditions

1. Service Conditions for Automatic/Manual Saving:
   a.) A CH-12COTB or PA-8COTBJ circuit card is required for Analog Caller ID.
   b.) Automatic/Manual Saving is available for the Desktop terminals.
   c.) This service is not available when the Calling Number is not received.
   d.) A maximum of 32 digits can be saved. However, when the Calling Number is received via CCIS Line, this feature is available for the first 16 digits (provided that "1" is assigned to 98: CI of ARTD).
   e.) When a Trunk Access Code (ex. "9") is required for Call Origination to the Saved Number, the Access Code must be added on a Route basis by ACNP and ACND. Note - for Analog Caller ID trunks only. The added Access Code is not displayed on the Desktop terminal LCD. However, the added Code is saved.

Note: For Analog Caller ID trunks the added Access Code is not displayed on the Desktop terminal LCD. However, the added Access Code is saved even if it is not displayed. For ISDN trunks, the Access Code will be displayed on the Desktop terminal LCD.

   f.) When the Area Code is not to be sent to within the same area, the Area Code must be deleted by

AFRS, AOPR and AADC.

g.) When Station Data is changed by Follow Phone etc., Stored Number Data is not exchanged.

h.) When the system is initialized, Stored Number Data is deleted.

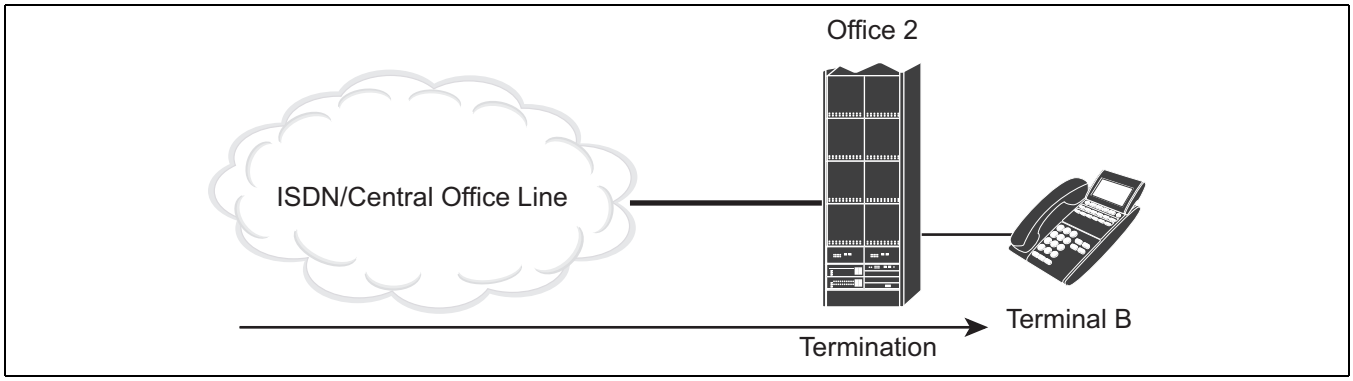
i.) When this service is used in an FCCS Network, the System Data should be set to all Nodes. However, when the Calling Number received from the Remote FCCS Node is not assigned a Telephone Number, the number of the Calling Station is not saved.

2. Service Conditions for Automatic Saving with the Feature key/Access Code:

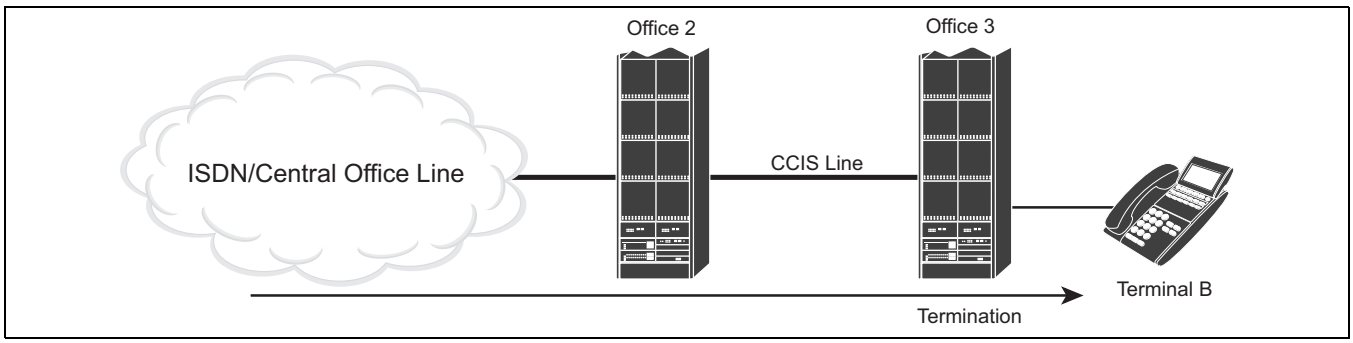
a.) Automatic Saving is available for Analog Stations allowed this service [ASFC, SFI 155 (Call Return) = 1] and the Desktop terminal with CALL RETURN Feature key.

b.) This feature is available for following calls.

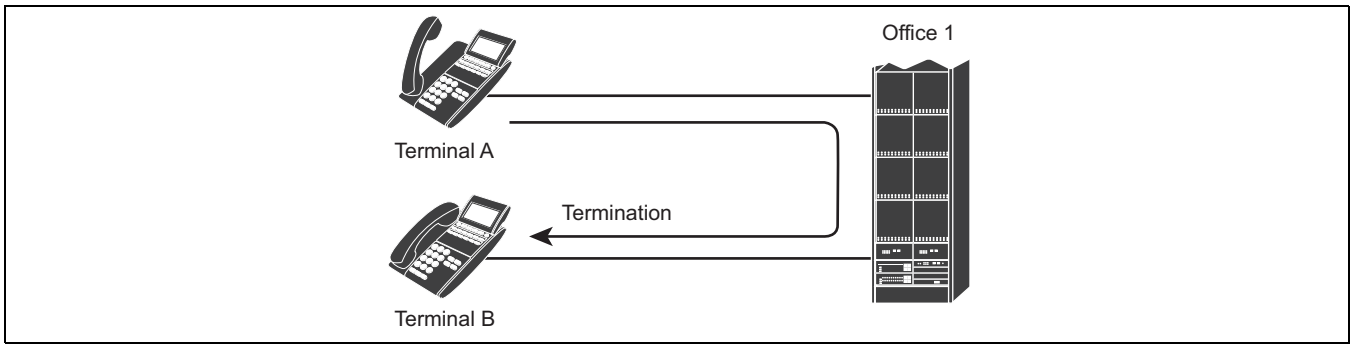
- ISDN/Central Office Call:



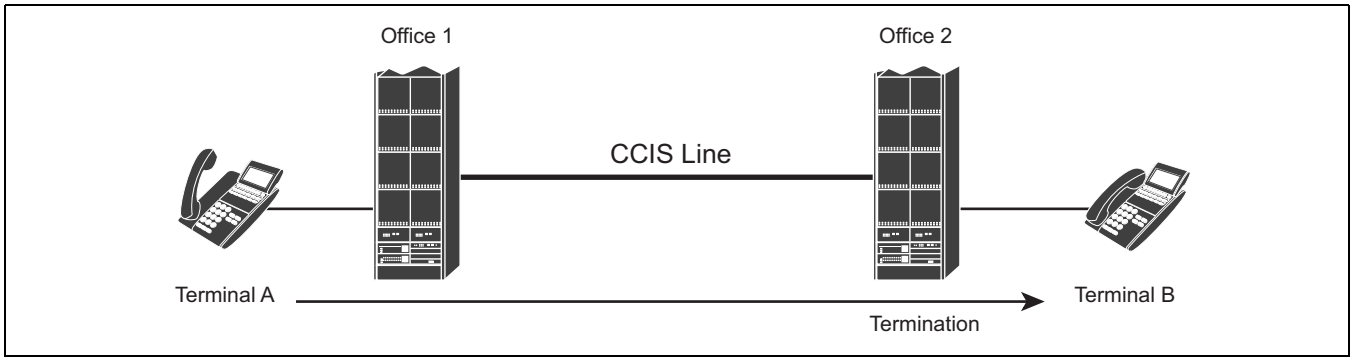
- CCIS Call via ISDN/Central Office Line:



- Station-to-Station Call:



- CCIS Call



c.) When the Station answers a call held on Multi Line, the Calling Number/Calling Station Number cannot be saved (Held Line is not My Line). When the call is terminated at the Trunk, the Calling Number is saved in the ELC Card only.

3. Sub-Service Conditions:
   a.) The newest Calling Number is saved in the Data Memory even when Call Return is not available, according to ASFC SFI 155 (Call Return) = 0. Accordingly, the office data is changed later, call origination using the number saved when the service is restricted is available.
4. The Calling Station Number is not saved in the following cases:
   a.) When answering the Station-to-Station Call is held by CALL PARK and CALL PICKUP Service.
   b.) When answering the Station-to-Station Call using CALL PICKUP Service.
5. Calling Number can be saved for Station-to-Station Call via the System Data Assignment (see Programming).

Note: When the Incoming Call is terminated via CCIS Line, the System Data assignment described above is required.

6. Service Conditions for Automatic Saving with the Feature key:
   a.) The CALL RETURN Feature key must not be assigned to the Desktop terminal which has no LCD, because Stored Number Data cannot be checked.
   b.) A maximum of three received Calling Numbers are saved with the ELC Card. If the received Calling Number is the same number as the last call, the Calling Number is not saved.
   c.) When the ELC circuit card is initialized, Stored Number Data is deleted.
   d.) The Calling Number is displayed on the Desktop terminal LCD when a call is terminated to Prime Line. Accordingly, when My Line of the Desktop terminal is not Prime Line, the displayed Calling Number is not saved.
   e.) When a call is terminated to My Line, even if the Desktop terminal user is in conversation using Sub Line, the Calling Number is saved.
   f.) When the Stored Number is displayed, the Calling Number is displayed on the upper LCD. The maximum number of digits that indicates the Calling Number is different, depending on the Desktop terminal type. When using a Desktop terminal with a 16-digit LCD, the first 16 digits are displayed. When using a Desktop terminal with a 24 digit display, the first 24 digits are displayed.
   g.) When a call is terminated while one of the Stored Numbers is displayed, the lower line of the LCD indicates the Calling Number and the upper LCD displays the Stored Number for five seconds.
   h.) When three Calling Numbers are already stored, if a call is terminated while the oldest number is displayed, the oldest number is deleted and the Received Number is saved. Consequently, call

origination cannot be made to the Displayed Number.

i.) Call origination can be made to the Stored Number while Dial Tone is heard. After seizing a Trunk, this feature is not available.

j.) When originating a call using this service, the last 24 digits are displayed on the Desktop terminal LCD.

7. Service Conditions for Automatic Saving with the Access Code:

a.) This service is available for the telephone corresponding with Caller ID Service. The Access Code equips Caller ID function or the telephone using Caller ID adapter.

b.) This service is not available when a call is terminated to an ISDN Terminal or a PS Terminal.

c.) The last Calling Number is stored in the Data Memory.

d.) Even when the Office Data makes Analog Caller ID Station effective by ASFC, SFI 146 = 1, Received Numbers are saved (Calling Number is saved for the Trunk receiving the Number and Calling Station Number is saved for the Station Call).

e.) The Operating Procedure with the Access Code is effective for call origination to the Stored Number's Party.

f.) When originating a call via the Access Code from the Desktop terminal, the LCD displays the Dialed Number and does not display Stored Number.

g.) Even if the LC Card is initialized, saved Calling Number is not deleted.

h.) The newest Calling Number is saved in the Data Memory even when CALL RETURN is not available according to the ASFC SFI 155 (Call Return) = 0. Accordingly, the Office Data is changed later, call origination using the number saved when the service is restricted is available.

8. Service Conditions for Manual Saving:

a.) Manual Saving is available for the Desktop terminal with the S&R Feature key.

b.) This service is available when a Desktop terminal is in communication with the Terminating Trunk.

c.) When the Calling Number is saved, S&R and the first 12 digits are displayed on the Dterm LCD.

d.) When the circuit card listed in item 1-b is not used, the first six digits of the Received Number are saved.

e.) Call origination can be made to the Stored Number while Dial Tone is heard. After seizing a Trunk, this feature is not available.

f.) For more information, refer to SAVE AND REPEAT -D [S-24D].

g.) When pressing the call origination key (such as Flex keys) on the station originates a call out of the office, 16 digits including an access code is stored in memory.

Programming

STEP 1: **ASYD** - SYS1, Index 80, Bit 1. 0/1 = #/* key is used to delete the stored Calling Number.
SYS1, Index 322, Bit 7. 0 = Call Return is not available for Station-to-Station Calling.
1 = Call Return is available for Station-to-Station Calling.

Note: For an Incoming Call from a CCIS line, this data is required.

STEP 2: **ASPA/ASPAL/ASPAN** - When originating a call to the Saved Number using the Access Code, assign the Access Code for this service.
SRV: "SSCA: Service Code Appendix"
SIDA: 106 (Call Return)

STEP 3: **ASFC** - When using the Access Code for this service, assign Service Feature Restriction Class that allows SFI 155 (Call Return) to the Stations activating this service.

STEP 4: **AKYD** - Assign CALL RETURN Feature key when using Automatic Saving function.
KYI: 1
FKY: 118 (Call Return)
Assign S&R Feature key when using Manual Saving function.
KYI: 1
FKY: 7 (Save and Repeat)

C-154 Call Hold Pick Up

General Description

This feature allows a user to pick up a Trunk or Station Call that is put on hold by a second user. The user can accomplish this by dialing an Access Code followed by the Telephone Number that is holding the call.

Note: This feature is only available in North America.

Operating Procedure

1. Make a Trunk or Station Call to Station A.
2. Station A places the call in progress on hold.
3. From Station B, dial the Access Code and the Station A Telephone Number.
4. Station B is connected to the Station or Trunk Call that is on hold at Station A.

Service Conditions

1. Only Stations in the same Tenant can use this feature.
2. This feature remains in effect even after hold time-out and the Station is in the re-ring state.
3. This feature also overrides the Exclusive Hold feature (a call that is put in Exclusive Hold can be picked up by using this feature).

Programming

STEP 1: **ASPA** - Assign the Access Code for this service. SRV = “SSCA: Service Code Appendix”, SIDA = 103.

STEP 2: **ASFC** - Assign Service Feature Restriction Class that allows SFI 150.

C-157 Calling Party Number - Name Assignment and Display

General Description

This feature allows a user to assign the desired name for the Calling Number, which is provided by Central Office, CCIS or ISDN feature, using PCPro command. The assigned name is displayed on the LCD of the Desktop terminal, the Desk Console and the SIP Handler Controlled SIP terminal.

Operating Procedure

No manual operation is required.

Service Conditions

1. When the Calling Number is not received, the assigned name cannot be displayed.
2. The Calling Number for this feature is assigned to the parameter CID of ACID. When only the first part of the Calling Number is assigned to the CID, the corresponding name is applied to any Calling Number that begins with the Assigned Number. For example, when "1471" is assigned to the CID, the Calling Name assigned for the number is applied to any Calling Number that begins with 1471, such as 1471-12-3456 or 1-471-234-5678.
3. The following numbers cannot be assigned to the parameter CID of ACID:
   a.) The first part of an already Assigned Number.
   b.) The number whose first part is the same as the already Assigned Number. For example, when "123" is already assigned to the CID, numbers such as "1", "12", and "1234" cannot be assigned.
4. When a Calling Number is sent via CCIS Line, this feature is available for the first 16 digits only.
5. The Name Display is available only when a call directly terminates to a Desktop terminal from the Trunk received the Calling Number. For example, the assigned name is not displayed for the Transferred Call. However, in this case, the assigned name appears after the Transferring Party (an Attendant Console or a Station) is disconnected.
6. If forwarding a call via CALL FORWARDING - BUSY LINE [C-2]/DON'T ANSWER [C-3]/ALL CALLS [C-5], the assigned name can be displayed only when the Destination Station is a Desktop terminal.
7. The assigned name cannot be displayed on the Attendant Console. Displaying the assigned name on Desk Console is available.
8. The assigned name cannot be output to the SMDR expanded area.
9. A maximum of 10,000 names for the Calling Numbers can be assigned in a system. However, this amount varies depending on the Data Capacity of CALL BLOCK [C-150] or the Data Memory Capacity of the system.
10. This feature has priority over the basic Caller ID Service.
11. The display for OAI Service has priority over the display for this service.

12. The Display Patterns are shown below:

Note: These examples below are only for Desktop Terminals and DESKCON.

Display Pattern 1 (SYS1, Index 390, Bit 0=0, Bit 1=0)

	8-digit Calling Number
24-character Assigned name	

Display Pattern 2 (SYS1, Index 390, Bit 0=0, Bit 1=1)

	Type of Call, Sub-address
24-character Assigned name	

Display Pattern 3 (SYS1, Index 390, Bit 0=1, Bit 1=0)

	8-digit Assigned Name
24-digit Calling Number	

Display Pattern 4 (SYS1, Index 390, Bit 0=1, Bit 1=1)

	Type of Call, Sub-address
24-digit Calling Number	

13. Characters of 8-bit ASCII can be displayed for this feature.
14. When the CALL ID CHANGE DISPLAY [FKY=113] Feature key is pressed, the display pattern on the LCD (assigned by ASYD) changes according to the following conditions:
    - When My Line is Prime Line, the Call ID Change Display Feature Key is effective while terminating or in conversation.
    - When My Line is not Prime Line, Call ID Change Display Feature Key is effective while in conversation.

    However, display pattern returns to the original setting when answering, holding, or transferring the call.
15. When the Received Numbers of a Calling Number and characters of the assigned name exceed the Display Field, whether to put them close to the right or left may be chosen by System Data. At this time an asterisk (*) is indicated on the exceeded digit position.
16. The Displayed Number and Characters flash when a call terminates and lights steadily during conversation.
17. The Trunk Card CH-12COTB and PA-8COTBJ are available for Analog COT.
18. When this feature is activated via FCCS, the same data should be set to all Nodes.

19. The following shows conditions when SIP Handler Controlled SIP terminals are used.

    - When a call arrives at a SIP Handler Controlled SIP terminal which supports RFC3325 and RFC5876, Calling Name can be displayed on the SIP Handler Controlled SIP terminal (in the INVITE request message, Name Display information is given in the display-name field of the From header). However, when the connected party is changed by the transferring operation or shifting from a three-way call into a two-way call due to the disconnect of one of the three parties, the display of the connected party information depends on the standard SIP terminal. For the conditions of the display, refer to SIP HANDLER - SIP TERMINAL [S-167] of Data Programming Manual - Business.

    - Name information assigned by the NAME parameter of the ACID command can be notified. Only Name Display information containing symbols and alphanumeric characters can be notified.

Programming

STEP 1: **ASYD** - SYS1, Index 241, Bit 6: Assign the data when the length of Caller ID exceeds the limit:
0/1=Only the former/latter part of data appears on the LCD.
Index 390,
Bit 0: Assign the prior data to display
0/1: Assigned Name/Calling Number
Bit 1: Assign the information displayed on the top line of the LCD
0/1: 0: Caller ID Data
1: Call Type, Sub-address
Bit 2: Enable to display Caller ID information on the Dterm to which the called line is assigned as a Sub Line.
0/1: Out/In Service

STEP 2: **ACID** - Assign the desired name to the Calling Number.
TYPE 3: Calling Party Number - Name Assignment and Display
CID: Caller ID Number
SRV3: 1: Calling Party Number - Name Assignment and Display
NAME: 24 digits maximum

STEP 3: **AKYD** - Assign the Call ID Display Change to a Line/Feature Key. (FKY=113).

STEP 4: **ADSL** - Assign the Call ID Display Change to a Soft Key.
SN: 0, 3
FKY: 113 (Call ID Change Display)

C-160 Call Trace

General Description

This feature enables the output of System Message [26-R] **Note 1**, that gives information on a Malicious Call (kind of call, number of Calling Terminal, etc.) originated from a Station/Trunk. The feature is made available when an Access Code is dialed or the CALL TRACE Feature key is pressed by the called user.

Note: This feature is not available in Australia.

Note 1: For details on the System Message [26-R], see the Operations and Maintenance Manual.

Operating Procedure

Dialing Access Code

- To output the System Message for an Abandoned Call:

1. After call termination, the Calling Party abandons the call.
2. The Called Party lifts the handset.
3. The Called Party dials the Access Code.
4. Information on the Abandoned Call is output as System Message [26-R].

- To output the System Message after the release of a call:

1. After call termination, the Called Party answers the call.
2. After speaking, the Called Party releases the call.
3. The Called Party lifts the handset.
4. The Called Party dials the Access Code.
5. Information on the Malicious Call is output as System Message [26-R].

Using Soft key or Line/Feature key

- To output the System Message for an Abandoned Call:

1. After call termination, the Calling Party abandons the call.
2. The Called Party lifts the handset.
3. The Called Party presses the CALL TRACE Feature key.
4. Information on the Abandoned Call is output as System Message [26-R].

- To output the System Message after the release of a call:

1. After call termination, the Called Party answers the call.
2. After speaking, the Called Party releases the call.
3. The Called Party lifts the handset.
4. The Called Party presses the CALL TRACE Feature key.
5. Information on the Malicious Call is output as System Message [26-R].

- To output the System Message while in speech:

1. After call termination, the Called Party answers the call.
2. While in speech, the Called Party presses the CALL TRACE Feature key.
3. Information on the Malicious Call is output as System Message [26-R].

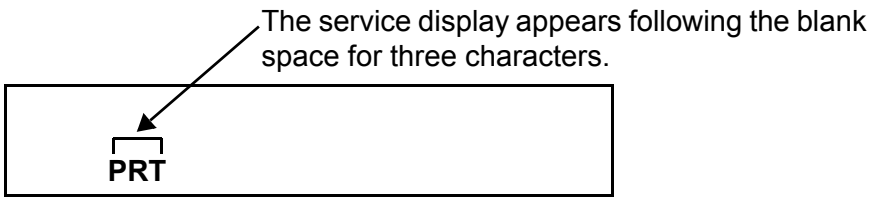
Note: Details on the LCD display (Desktop terminal only) are as follows:

24 digits display of a Desktop terminal

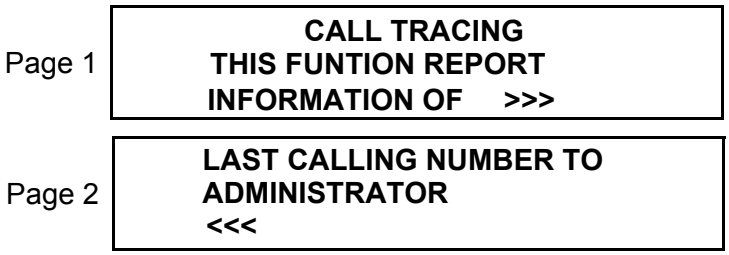
- When the CALL TRACE is set:



- When the CALL TRACE is assigned to the Soft key:



- When the HELP key (Soft key) is pressed:



Service Conditions

1. This feature is made available via the Stations, to which the following data is assigned via ASFC [SFI 170 (Call Trace), RES = 1 (Service: Allowed)].
2. This feature can be activated by Analog or Desktop terminal stations. Other Terminals such as Wireless Stations (PS), ISDN Terminals, and Attendant/Desk Consoles cannot provide this feature.
3. This feature is designed to get information on a Terminated Call (the last call terminated to a Station Line). CALL TRACE is not available for Originating Calls.
4. Because a Desktop terminal can contain Multiple Lines, the message output is executed on a line basis. Therefore, when activating this feature via an Access Code, use the line that actually handled the Target Call.
5. When a call is terminated via a transfer (the call is transferred from other Station), the System Message [26-R] shows the information only related to the immediate Station (the party who transferred the call). Information about the first originated call cannot be obtained.
6. The following are the conditions for System Message [26-R].
   a.) Each data is output in hexadecimal, giving details from the first digit.
   b.) When the Message Data contains “A”, “B”, “C” or “0”, the actual meaning is as follows:
   A: 0
   B: *
   C: #
   0: blank
   c.) Date/Time Data is based on the actual date and time when the Access Code is dialed.

d.) The maximum digits for each data are as follows:

Data	Maximum Digits
FPC	3
Tenant Number	3
Station Number	6
UGN	3
Telephone Number	16
Trunk Call Number	32

e.) When the Called Station does not have the Telephone Number, the FPC/Telephone Number/UGN Data is output as "0".

f.) When the Calling Party is the Station which does not have the Telephone Number, the Tenant Number/FPC/Telephone Number/UGN Data is output as "0". The kind of call is output as "10".

g.) If the number of the Trunk Call (Caller ID) cannot be identified, the Trunk Call Number Data is output as "0". The kind of call is displayed as "20".

7. When the call is terminated via the CCIS, the obtainable Calling Party Numbers are:
   - Maximum eight digits [when CI (Route Data) = 0]
   - Maximum 16 digits [when CI (Route Data) = 1]

   System Message is output at the Node that the Called Station (informer) belongs to.
8. Unless the message output is set on a network basis, System Message [26-R] is created at the Node which contains the Called Station (informer).
9. The default message output grade is shown below:

	MJ	MN	SUP	GR
26-R	—	—	—	2

10. To obtain the Caller ID via an Analog Trunk, use CH-12COTB or PA-8COTBJ card. Other Analog Trunk Cards cannot provide this feature.
11. This service is available with the Analog Caller ID Service.

Interactions

1. When DUAL STATION CALL [D-156] is used, a call can be terminated to a main station and a sub-station simultaneously. At this time, the Calling Party information can be obtained only by the key operation from the main station side. Information to be obtained from the sub-station is about the last call directly terminated to the sub-station.

Programming

STEP 1: **ANPD/ANPDL/ANPDN** - Assign 1st digit code (1stDC) and Number of Necessary Digits (NND) for Feature Access and Cancel. Select Normal Service (N) from the combo box for Connection Indexes.

STEP 2: **ASPA/ASPAL/ASPAN** - Assign the Access Code for CALL TRACE, CI = "N: Normal", SRV = "SSCA: Service Code Appendix", SIDA = 116 (Output of System Message [26-R]).

STEP 3: **ASFC** - Assign a Service Feature Restriction Class to allow CALL TRACE for the Stations, SFI 170, RES = 1 (Restriction: lifted).

STEP 4: **AKYD** - Optional. Assign the CALL TRACE to a Line/Feature key on the Desktop terminal, FKI = 1 (Feature Key), Feature key Call Trace [129].

STEP 5: **ADSL** - Optional. Assign the following to use a Soft key on the Desktop terminal for CALL TRACE:

SKP: 0 - 15 (Soft Key Pattern)
SN: 1 (DT), 4 (Communication in Progress) (Status Number)
SKN: 1 - 16 (Soft Key Number)
FKY: 129 (Call Trace)

STEP 6: **ADKS** - Assign the following to use a Soft key on the Desktop terminal for CALL TRACE:

TN: Tenant Number
STN: Station Number
SKP: 0 - 15 (Soft Key Pattern assigned in STEP 5)
LKP: 1 - 3 (Location of Key Pattern)
PG: 0/1 (LCD Page Change: Soft Key/Line Key)

C-169 Consultation Hold Release

General Description

This feature enables a Station, connecting to a second call while the Station has been holding (Consultation Hold) an original call (including ADD-ON CONFERENCE - 8 PARTY [A-121] member), to disconnect the second call forcibly by pressing the DISC key once for Desktop terminal, or switch hook flashing for Analog Stations/PSs/WLAN Handsets twice. After that, the second call is released, and the original call is retrieved from hold.

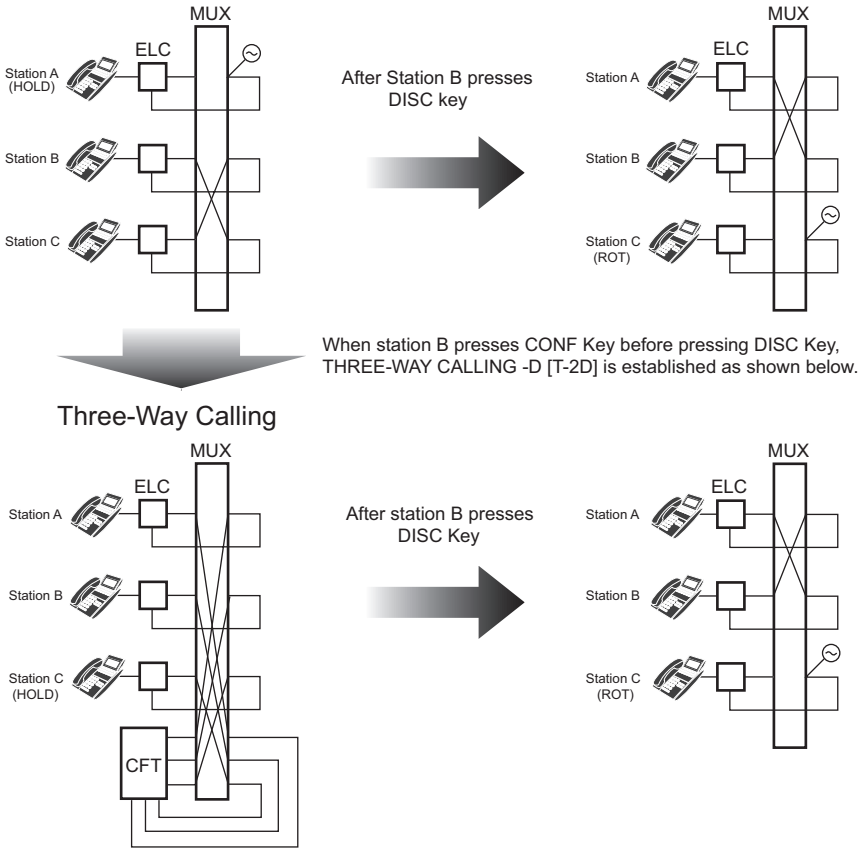
Operating Procedure

<When the operating Station for this feature is other than SP Controlled Standard SIP Terminal>

Situation 1: When Station B is talking with C while holding (Consultation Hold) an original call (Station A).

- Station A: Original Call
- Station B: Operating Station for this feature (Desktop terminal/Analog Station/PS/WLAN Handset/SIP Handler Controlled SIP terminal)
- Station C: Second call called by Station B (It can be disconnected by this feature.)

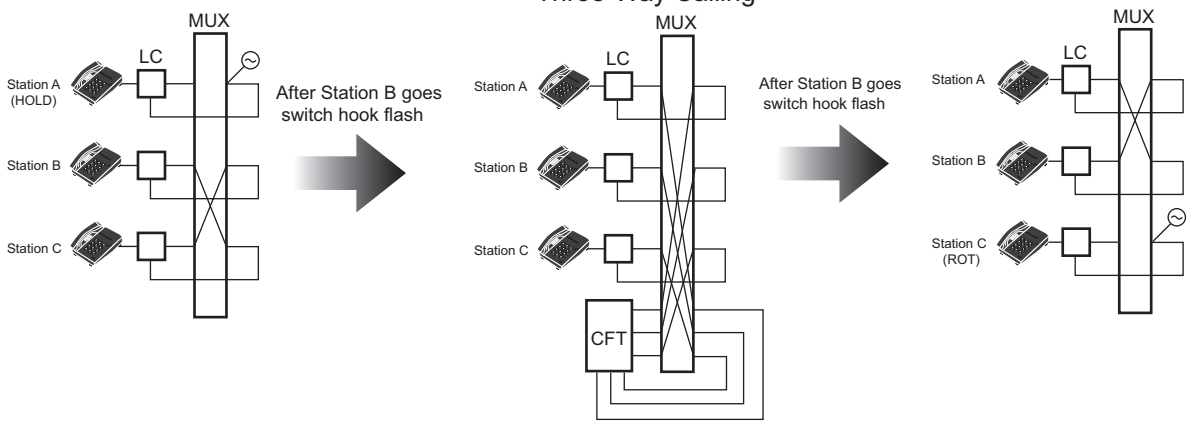
Procedure for Desktop Terminal



Note: If a Trunk for Three-Way Calling within a MUX Card is busy, this feature is not available.

Procedure for Analog Station/PS/WLAN Handset

Three-Way Calling

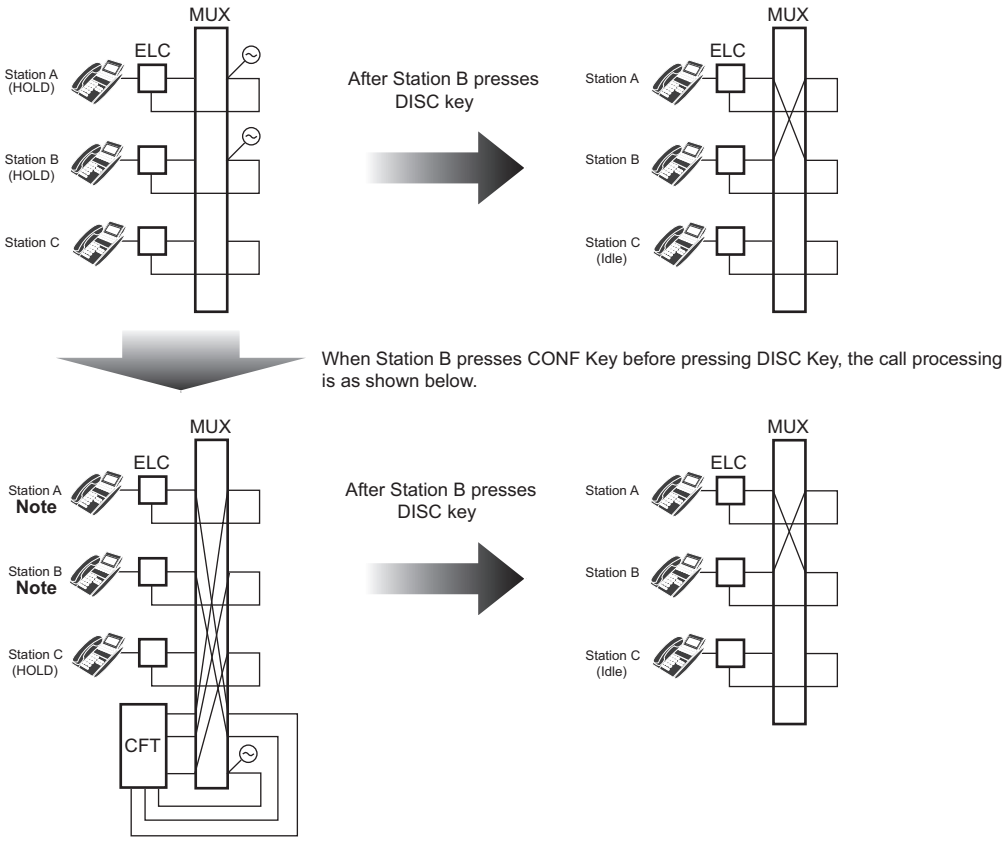


Note: If a Trunk for Three-Way Calling within a MUX Card is busy, this feature is not available.

Situation 2: When Station B is held by Station C during Situation 1 (In this case, only Desktop terminal is available.)

- Station A: Original call
- Station B: Operating Station for this feature (Desktop terminal)
- Station C: Second call called by Station B (It can be disconnected by this feature.)

Procedure for Desktop Terminal

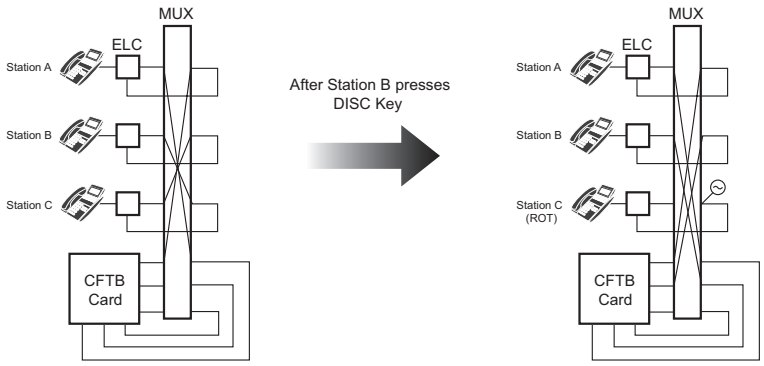


Note: Hold Tone cannot be heard.

Situation 3: When Station B is talking with Station C after Station B holds (Consultation Hold - All Calls -D [C-17D]) Station A while Station A and B in a Conference (Add - On Conference - 8 Party [A-121]). (PS/ISDN Terminal Equipment/WLAN Handset/SIP Handler Controlled SIP terminal cannot be applied to the feature.)

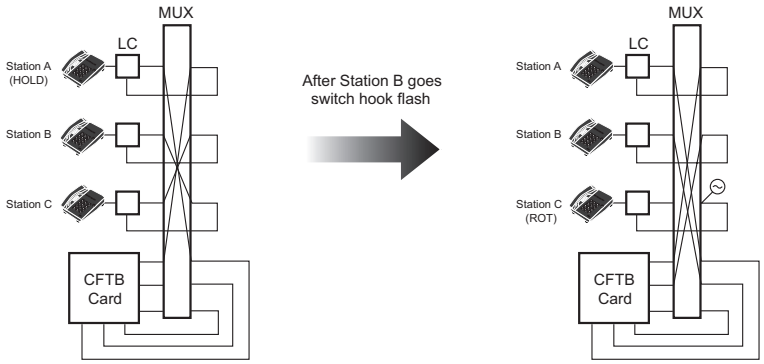
- Station A: Original call (Conference member)
- Station B: Operating Station for this feature (Conference member)
- Station C: Second call, which is not Conference member, called by Station B (it can be disconnected by this feature)

Procedure for Desktop Terminal



Note : This feature is not available when operation Station (Station B) is in Add-On Conference-8 PARTY [A-121] while holding Station C (Station/Trunk) using TRANSFER Key.

Procedure for Analog Station



Note : Only when SYS1, Index 803, Bit 2=1 is assigned, Station C can be disconnected.

Note : When ASFC, SFI 131 (Add-On Conference 8 - party) = 1 is assigned to Station B, Station C is called into Add-On Conference 8 - Party [A-121].

<When the operating Station for this feature is Standard SIP Terminal>

Refer to the manual of each Standard SIP Terminal for operating procedures.

Note: This feature is available for Standard SIP Terminal.

Service Conditions

1. This feature is available when SYS1, Index 803, Bit 1=1 is assigned.
2. When there is a call terminated at Chinese specification MFC Trunk in calls related to this feature, this feature is not available.
3. ISDN Terminal Equipment cannot be applied to Operating Station of this feature.
4. A Station assigned Emergency Call (SFI 24) cannot be applied to Operating Station of this feature.
5. The DISC key used for this feature can be assigned through the AKYD command (FKY=131).
6. When an Analog Station uses this feature while in ADD-ON CONFERENCE - 8 PARTY [A-121], assign SYS1, Index 803, Bit 2=1 also. However, the Analog Station assigned SFI 131 = 1 (Permission of calling an Add-On Conference - 8 Party) cannot be applied to Operating Station for this feature.
7. When SYS1, Index 244, Bit 6=1 is assigned, switch hook flash cannot be used while in THREE-WAY CALLING [T-2].
8. A Desktop terminal can execute this feature using the FLASH BUTTON (FKY=21) key as a substitute for the DISC key. In this case, comply with Operating Procedure for Analog Station/PS/WLAN Handset.
9. A second party (second call) is the last Terminal connected with Operation Station. Therefore, when the user changes a party using the TRANSFER key, the original party and second party are also changed.
10. When there is a call originated from Attendant Console in the calls related to this feature, this feature is not available.
11. When assigning the DISC key to a Soft key, DISC is displayed on the LCD of Desktop terminal. HELP for the DISC key is as follows:

DISC KEY PRESS THIS KEY TO RELEASE THE LAST >>>	CALLED PARTY AND RECONNECT TO THE <<< HELD PARTY

12. When a consultation call to the destination party is made via a trunk line, Consultation Hold Release is available, regardless of System Data setting (ASYDL/ASYDN, SYS1, Index 803, Bit 1).
13. Service Conditions when the operating Station for this feature is an SP Controlled SIP terminal are as follows:
    a.) This feature is available for SP Controlled Standard SIP terminals with Consultation Hold Release function. SP Controlled Standard SIP terminals must be confirmed by the connection test of NEC as well as in compliance with IETF RFC3261 and Internet-Drafts.
    b.) Holding a call must be canceled by an SP Controlled Standard SIP terminal (the intermediate party) to reconnect the original call with the first party.
    c.) If this feature becomes invalid for reasons such as when a call is transferred to an Attendant, Call Forwarding - All Calls [C-5] is executed.
14. Service Conditions when the operating Station for this feature is a SIP Handler Controlled SIP terminal are as follows. When the following conditions apply, Consultation Hold Release works with operations for putting/canceling the call on hold, but the SIP Handler Controlled SIP terminal cannot talk with the held station until the SIP Handler Controlled SIP terminal answers the held call.
    If the following conditions are not met, the SIP Handler Controlled SIP terminal is released with operations for putting/canceling the call on hold, and then the call shifts to a Two-way call between the held station and the forwarding destination.
    a.) Setting AKYD, FKY=131 (DISCONNECT key) is required.
    b.) Setting ASYDL/ASYDN, SYS1, Index 803, Bit 1=1 (Consultation Hold) is required. This setting is also required when the destination is a trunk.

15. When using this feature in the ACD/OAI system, the following system data setting is required:
ASYDL/ASYDN, SYS1, Index 1240, Bit 1=1 (Consultation Hold Release - ACD is enabled)
For more information, refer to the Data Programming Manual - ACD, "C-205A CONSULTATION HOLD RELEASE - ACD."

<For terminals other than SP Controlled Standard SIP Terminal>

Note: Refer to this section for the programming steps for SIP Handler Controlled SIP terminals. These instructions assume that the data for SIP Handler Controlled SIP terminals is already assigned.

STEP 1: **ASYDL/ASYDN** - Allow this feature. SYS1, Index 803, Bit 1: 0/1 = this feature is not allowed/allowed.

SYS1, Index 803, Bit 2: 0/1 = Analog Station cannot/can use this feature during ADD-ON CONFERENCE - 8 PARTY [A-121].

STEP 2: **ADSL/AKYD** - Assign the DISCONNECTION (DISC) key used for this feature to Soft key or Line/Feature key.

ADSL - When assigning the DISC key to a Soft key:

SKP (Soft Key Pattern):	0-15
SN (Status Number):	5 (Consultation Hold)/9 (Three-Way Calling)
SKN (Soft Key Number):	1-16
FKY (Function Key Data):	131

AKYD - When assigning the DISC key to a Line/Feature key:

KYN:	Key Number
KYI (Service Condition):	1 (Feature Key)
FKY (Feature Key Code for Flex Key):	131

<For SP Controlled Standard SIP Terminal>

The programming after the SP Controlled Standard SIP Terminal Data is assigned is as follows:

STEP 1: **ASPTL/ASPTN** - Assign SIP Terminal Data.
TERMINAL field
DETAIL field
When secondary call is disconnected, held call remains on hold:
Select the check box if this feature is used with Standard SIP Terminal.

Note: This feature has been available for Standard SIP Terminal.

Note: If the data on the TERMINAL tab is changed, the System receives a notification about the data change. If the System cannot receive the notification for some reasons such as when SP is down or when a network failure occurs, the disparity between the actual Office data and information on the System is created. At this time, the SIP Terminal is required to re-register. When SP cannot receive the notification, "There is no response from SP" message will be displayed in PCPro.

C-172 Call Forward by Service Feature Class

General Description

This feature will implement all types of Station Based Call Forwarding (Call Forwarding - Busy Line, Call Forwarding - Don't Answer, Call Forwarding - All Calls) to a predetermined number, based on the Service Feature Restriction Class (SFC) of the Called Station. This feature may be set using Common or Split Forwarding, depending on system programming.

Note: This feature is available in North America only.

Operating Procedure

No special Operating Procedures are required by the Station user to activate this feature.

Service Conditions

1. This service will have all the current limitations of the each Call Forwarding feature.
2. This feature works based on the ACFCL Data of the Service Feature Restriction Class of the Terminating Party.
3. In an FCCS Network, this feature works off the ACFCL Data that is assigned in the System of the Calling Party, for the SFC of the Called Station. Because of this, it is strongly recommended that ACFCL be assigned the same throughout the FCCS Network.
   a.) Station 1000 is in Telephony Server A SFC 1.
   b.) Station 2000 is in Telephony Server B SFC 1.
   c.) In Telephony Server A SFC 1 has Forward All to 5000 set by ACFCL.
   d.) In Telephony Server B SFC 1 has Forward All to 6000 set by ACFCL.
   - When Station 2000 calls Station 1000, the call will forward to Station 6000 in the FCCS Network. This feature reads the LDM ACFCL Data in the Calling Party's office for the Called SFC.

   e.) This feature will have the same limitation of up to five Call Forwarding - All Calls or Call Forwarding - Busy Line forwards in a row.
   f.) If ACFCL is set for SFC 1 to Call Forwarding - All Calls to 4140 and 4140 is in SFC 1 when a call is made to 4140, it will ring. If the call is placed to another Station in SFC 1, the call will forward to 4140.
   g.) Examples of Call Forwarding scenarios using ACFCL:
   SFC 1 Call Forwarding - All Calls to 4140 (4140 is programmed in SFC 2)
   SFC 2 Call Forwarding - All Calls to 4141 (4141 is programmed in SFC 3)
   SFC 3 Call Forwarding - All Calls to 4142 (4142 is programmed in SFC 4)
   SFC 4 Call Forwarding - All Calls to 4143 (4143 is programmed in SFC 5)
   SFC 5 Call Forwarding - All Calls to 4144 (4144 is programmed in SFC 6)
   SFC 6 Call Forwarding - All Calls to 4145 (4145 is programmed in SFC 7)
   SFC 7 Call Forwarding - All Calls to 4146 (4146 is programmed in SFC 8)
   SFC 8 No Data assigned
   SFC 15 No Data assigned
   - Using the assignments above, if a call is made to a Station in SFC 1, the call forwards to 4140. The call will then follow the forwarding set for each SFC. The call will forward up to five times (based on ASYD, SYS1, Index 5, Bit 1 to Bit 3) and will ring 4144.

- Using the assignments above, if 4141 sets individual Call Forwarding - All Calls to Station 5000 (SFC 15) and a call is made to 4140 the call will forward from 4140 to 4141 and then to 5000.
- Using the assignments above, if 4141 sets individual Call Forwarding - All Calls to Station 4145 (4145 has ACFCL set to 4146), a call made to 4140 will forward to 4141 (because of ACFCL), then forward to 4145 (because of individual Call Forwarding - All Calls), then forward to 4146 (because of ACFCL), it would then ring there (because no Forwarding Data is set in ACFCL for SFC 8 and no individual Call Forwarding is set).

h.) The following table illustrates the hierarchy for Call Forwarding:

PRIORITY	CALL FORWARDING TYPE	COMMAND USED
1	Call Forwarding - All Calls Individual	ACFS or as set by the individual Station
2	Call Forwarding - All Calls by SFC	ACFCL
3	Call Forwarding - Busy Line Individual	ACFS or as set by the individual Station
4	Call Forwarding - Busy Line by SFC	ACFCL
5	Call Forwarding - Busy Line by Tenant	ACFO
6	Call Forwarding - No Answer Individual	ACFS or as set by the individual Station
7	Call Forwarding - No Answer by SFC	ACFCL
8	Call Forwarding - No Answer by TN	ACFO

i.) Separate Destinations will be allowed, based on current Call Forwarding types.

- Based on ASYD, SYS1, Index 5, Bit 0=0/1
  Call Forwarding - Busy Line and No Answer Destination is separate/common.
- Based on ASYD, SYS1, Index 79, Bit 2=0/1
  Calling Party Trunk/Station Call Forward Destination is common/separate.

Note: If either of the bits above are changed while ACFCL is open, the command must be restarted. If data is already assigned in ACFCL and either of these bits is to be changed, all data should be deleted from ACFCL first.

j.) When using ACFCL Call Forwarding - Don't Answer after Blind Transfer, Service Feature Index 103 (Assignment of No Answer Timer) and Service Feature Index 104 (Blind Transfer) must be set correctly and will operate just like individual Call Forwarding - Don't Answer.

- ASFC Service Feature Index 8 (Call Forwarding - Don't Answer) should be assigned to allow Call Forwarding - Don't Answer to occur in the SFCs that use ACFCL for Call Forwarding - Don't Answer.

k.) If ACFCL for Call Forwarding - All Calls is assigned to an SFC that contains UCD members, those members are skipped in the UCD Group. This is the same as if the Call Forwarding - All Calls Individual feature is used.

- If the UCD Pilot is in an SFC that is set Call Forwarding - All Calls by ACFCL, all calls to that Pilot will forward to the Destination Station. This is the same as if the Call Forwarding - All Calls Individual feature is used.

Programming

This service can be split by Tenant Service.
This is controlled by ASYD, SYS1, Index 93, Bit 0.
If ASYD, SYS1, Index 93, Bit 0=0, this service is assigned separately on a per TN basis.
If ASYD, SYS1, Index 93, Bit 0=1, this service is common across all TNs using the data from TN 1.
This service can be split to forward, based on Day or Night mode of the System.
This is controlled by ASYD, SYS1, Index 75, Bit 1.
If ASYD, SYS1, Index 75, Bit 1=0, both D/N is common.
If ASYD, SYS1, Index 75, Bit 1=1, D/N will develop separate tables.
ACFCL - Set the following data:

D/N:	Day/Night Mode (radio button)
TN:	Tenant
SFI:	Service Feature Index

When ASYD, SYS1, Index 5, Bit 0=0 and ASYD, SYS1, Index 79, Bit 2=0
Call Forwarding - All Calls Destination
Call Forwarding - Busy Line/Call Forwarding - Don’t Answer Destination

When ASYD, SYS1, Index 5, Bit 0=1 and ASYD, SYS1, Index 79, Bit 2=0
Call Forwarding - All Calls Destination
Call Forwarding - Busy Line Destination
Call Forwarding - Don’t Answer Destination

When ASYD, SYS1, Index 5, Bit 0=0 and ASYD, SYS1, Index 79, Bit 2=1
Call Forwarding - All Calls Destination for Stations
Call Forwarding - Busy Line/Call Forwarding - Don’t Answer Destination for Stations
Call Forwarding - All Calls Destination for Trunks
Call Forwarding - Busy Line/Call Forwarding - Don’t Answer Destination for Trunks

When ASYD, SYS1, Index 5, Bit 0=1 and ASYD, SYS1, Index 79, Bit 2=1
Call Forwarding - All Calls Destination for Stations
Call Forwarding - Busy Line Destination for Stations
Call Forwarding - Don’t Answer Destination for Stations
Call Forwarding - All Calls Destination for Trunks
Call Forwarding - Busy Line Destination for Trunks
Call Forwarding - Don’t Answer Destination for Trunks

C-173 Call Redirect

General Description

This feature allows a Desktop terminal user to view on the Desktop terminal display the Station Number or Caller ID of an Incoming Call and immediately redirect the call by pressing a Feature key. The destination of the Call Redirect will be the Call Forwarding - Don't Answer Destination or the Recall Destination (if the call is transferred and no Call Forwarding - Don't Answer Destination is set).

Operating Procedure

In the case of My Line

1. Station 1001 is a Desktop terminal My Line.
2. Station 1001 sets Call Forwarding - Don't Answer via PCPro or feature to Station 4140 (VMM).
3. Station 1000 dials Station 1001.
4. Station 1001 rings and displays 1000 on its LCD.
5. Station 1001 presses the CALL REFIRECT Feature key.
6. REDIRECTING is displayed on the LCD, and the call immediately forwards to 4140.
7. In the case of Sub Line:
   a.) Station 1001 is a Desktop terminal and Station 1002 is a Sub Line appearing on 1001.
   b.) Station 1002 sets CFDNA to by PCPro or feature to Station 4140 (VMM).
   c.) Station 1000 dials Station 1002.
   d.) Station 1002 rings.
   e.) User of Station 1001 presses the line appearance of Station 1002 and then immediately presses the CALL REFIRECT Feature key.
   f.) REDIRECTING is displayed on Station 1001 and the call immediately forwards to Station 4140.
8. In the case of Call Forwarding - Don't Answer while Call Waiting - Terminating:
   a.) Station 1001 is a Desktop terminal My Line and has Call Forwarding - Don't Answer Call Waiting applied.
   b.) Station 1001 sets Call Forwarding - Don't Answer via PCPro or feature to Station 4140 (VMM).
   c.) Incoming Trunk in a Route which has Call Waiting set in Route Data calls Station 1001
   d.) Station 1001 hears Call Waiting Tone and the ANSWER key flashes.
   e.) User at 1001 presses the CALL REFIRECT key.
   f.) Trunk immediately forwards to Station 4140. In this case, if the line used is not the My Line (Station 1001), the Line key must be pressed before the Call Redirect key.
9. In the case of Automatic Recall (ASYD, SYS1, Index 69, Bit 0=0):
   a.) Station 1001 is the My Line of Desktop terminal Station 1001.
   b.) Station 1000 dials Station 1002.
   c.) Station 1002 answers the call and transfers to Station 1001 and releases the call.
   d.) Station 1001 rings.
   e.) User at Station 1001 presses the CALL REFIRECT key.
   f.) REDIRECTING is displayed on Station 1001 and the call immediately recalls to Station 1002.

Service Conditions

1. All the current limitations of normal Call Forwarding - Don't Answer also apply to this feature.
2. If the destination of a Call Redirect is in a Hunt Group, the call will follow the hunting as normal Call Forwarding - Don't Answer would. This is the same as current Call Forwarding - Don't Answer [C-3].
3. If the destination of the Call Redirect is Busy or in a state that will not accept the call, the call will continue to ring on the Station trying to invoke the Redirect feature and will display REDIRECTING. Because of this limitation, whenever possible, the destination of Call Redirect should be a UCD (VMM) Group or should have Call Forwarding - Busy Line set to a UCD. This way Call Redirect will always queue to the UCD Group if all Stations are Busy.
4. If the destination of a Call Redirect is call forwarded, the Call Redirect will follow the Route same as a normal Call Forwarding - Don't Answer [C-3] would.
5. This service is only available for Desktop terminal Stations.
6. Wireless - PHS Terminals cannot use this feature.
7. A Feature key (or Soft key) must be provided for this service.
8. This feature will operate with Call Forwarding - Don't Answer while Call Waiting - Terminate is in effect.
9. Call Redirect during Call Waiting - Terminate has all of the current Call Waiting - Terminate limitations.
10. This includes only Incoming Trunk Calls that will be redirected in a Call Waiting - Terminate state.
11. Call Redirect will not function with Call Forwarding - Don't Answer with Attendant Camp-On.
12. If the CALL REFIRECT key is pressed in a state that will not allow a Call Redirect, "REDIRECT NA" will be displayed.
13. This feature will only function on the My Line of a Dterm when ASYD, SYS1, Index 384, Bit 0=1.
14. If Call Redirect is used with Automatic Recall, the call will be redirected to the Transferring Station (this is when ASYD, SYS1, Index 69, Bit 0=0).
15. If the Prime Line and My Line of the Dterm is split, pressing the CALL REFIRECT key will redirect the My Line by default. To redirect the Prime Line or other lines, in this case the Line key must be pressed before the CALL REFIRECT Feature key.
16. For Call Redirect to be used for Call Forwarding - Don't Answer while Call Waiting - Terminate, ASYD, SYS2, Index 10, Bit 7 must be set to 1. This bit is used to send periodic Call Waiting Tone, but is also the Call Forwarding - Don't Answer Call Waiting Timer.
17. This feature will not function during Camp-On by Station feature.

Programming

Normal Call Forwarding - Don't Answer [C-3] programming applies. For Call Redirect to function during Call Forwarding - Don't Answer, Call Waiting Timer normal Call Forwarding - Don't Answer, Call Waiting Timer programming applies as described in this manual.

When used as a Feature key/Soft key, assign the following:

STEP 1: **AKYD** - FKY 132 (CALL REFIRECT Feature key)

When used as a Soft key, assign the following:

STEP 1: **ADSL** - State 0 (ringing state)

FKY 132 (CALL REFIRECT Soft key)

State 15 (Miscellaneous state)

The Soft key display will show "REDIR" on the Desktop terminal.

When programmed to State 15, this key may appear on a Soft key in a state it cannot be used. If it is pressed in that state, "REDIRECT NA" will be displayed.

If ASYD, SYS1, Index 384, Bit 0=1, Call Redirect can only be used on the My Line of a Dterm, because pressing the Line key of a Ringing Line will answer the call.

C-174 Calling Station Number Display Elimination

General Description

This feature allows calling and called station numbers to be hidden on each Desktop terminal for privacy or security purposes during a station-to-station call. Station numbers are masked with "********."

No special operating procedures are required by the station user to activate this feature.

Service Conditions

<Important Notices>

1. As for Terminal Call History, station numbers are displayed without the information masking in the outgoing call history.
2. A station number is masked with an 8-digit code "********" on each Desktop terminal by data assignment of two SFIs (SFI 179 and SFI 180) which enable the station number display elimination on calling and called stations.

   Note: This manual explains service conditions based on a case where both calling and called stations enable this feature (both SFI 179=1 and SFI 180=1 are assigned in each station).

3. Temporary cancellation and setting of this feature by terminal key operations are not available.

<Interactions with Other Features>

1. A called station number is hidden on a calling station which sets a message of Message Reminder [M-19]. When a called station user presses the MESSAGE REMINDER Feature key, a station number of the calling station which has set the message is displayed (the calling station number is not hidden). The called station user can return the call by pressing the MESSAGE REMINDER Feature key in off-hook state.
2. When Call Back [C-1] is in service, station numbers are hidden and Call Back [C-1] works normally.
3. When Save And Repeat -D [S-24D] is in service, a dialed station number is stored without the information masking, and Save And Repeat -D [S-24D] works properly.
4. When Call Return [C-151] Automatic Saving/Manual Saving is in service, "********" is stored as the station number information, and then the station attempts to call "********" during the call origination operation by the station user.
5. When a station user uses Last Number Called -D [L-6D], station numbers are displayed without the information masking, allowing the user to redial the last number called.
6. When a station user uses Call History, station numbers are hidden by this feature. The station fails to call using the Call History because it attempts to call to "********."
7. When Dial Plan to support IP Centrex Functionality [D-166] is in service, a Private Number is hidden as well as a station number.
8. When Name Display - System [N-28]/PHS Name Display [P-79] is in service, only a station number is hidden although a station name is displayed.

9. Station number of an Intermediate Station is displayed without the information masking.

   Example:

   - Station number of an Intermediate Station for Blind Transfer To Station [B-18], Call Forwarding - All Calls [C-5], Call Forwarding - Busy Line [C-2], Call Forwarding - Don't Answer [C-3], Logged Out IP Station - Call Destination [L-53]
   - Pilot Station number of Station Hunting features when an incoming call is hunted
   - Station number of a party to which an incoming call arrives before the other party picks it up by Call Pickup features
   - Station number of the party already connected when the third party tries to make a connection by Executive Right-of-Way [E-1]

10. Do not set Telephone Number to be displayed on the middle line of the LCD (ASYDL, SYS1, Index 672, Bit 0=1). Telephone Number will be displayed on the top line of the LCD regardless of the setting.
11. When a party makes a call to a busy station, the busy station's number is hidden on the calling party's LCD in six seconds. However, after six seconds eventually pass, the busy station's number is displayed without the information masking.
12. As for Terminal Call History, when an incoming call is answered or missed, station number of a calling party is not saved in the incoming call history.
    If the called party answers an incoming call arriving at a Sub Line when the called party accommodates a PS/SP Controlled SIP terminal station line as the Sub Line, a calling party number is hidden on the LCD but the number is saved without the information masking in the incoming call history.
13. To make a call to two stations simultaneously by Dual Station Call [D-156], the main station and sub-station need to be included in the same Service Feature Restriction Class (SFC).

<Hardware/Software>

1. This feature is supported by Desktop terminals such as Digital terminals, IP terminals, Soft Phones and SIP Multiple Line terminals.
2. During a connection between a PS and a Desktop terminal, the Desktop terminal station number is displayed on the PS. The PS station number can be hidden on the Desktop terminal depending on the setting of the SFIs.
3. When a Desktop terminal originates a call to an Analog Caller ID station, the Desktop terminal station number is hidden on the Analog Caller ID station. The actual display pattern depends on the type of Analog Caller ID station and the information masking may be different from "********."
4. When an Analog Caller ID station originates a call to a Desktop terminal, the Desktop terminal station number is displayed without the information masking on the Analog Caller ID station.
5. During a connection between an attendant console and a Desktop terminal, the Desktop terminal station number is displayed without the information masking on the attendant console. The attendant console's number is masked with "********" on the Desktop terminal.
6. When a Desktop terminal originates a call to an attendant console, an indication "OPR" appears on the Desktop terminal until the attendant console answers. The indication changes to "********" after the call is answered.

<Interactions with Networks>

1. This feature only works when the Desktop terminal is directly connecting to the station on which a counter station number is hidden. This feature does not work when stations are connecting through a CCIS or ISDN Trunk.
2. When this feature is used over FCCS, using ASFCN command is recommended for the data assignment. If ASFC command is used, assign SFI 179=1 (Self station number display elimination on the counter station) and SFI 180=1 (Counter station number display elimination on the self station) to the same Service Feature Restriction Class in all FCCS nodes.

<Conditions Specific to This Feature>

1. This feature functions according to the SFI data settings of the actual line, not the Prime Line for Multiple Line Operation -D [M-20D].

<Other Conditions>

None

Programming

ASYDL/ASYDN - Assign Telephone Number Display Pattern.
SYS1, Index 672, Bit 0=0 (Telephone Number is displayed on the top line of LCD)

ASFC/ASFCN - Assign the following data to the Service Feature Restriction Class (SFC) for each station.
SFI: 179 (Self station number display elimination on the counter station) **Note 1**
SFI: 180 (Counter station number display elimination on the self station) **Note 2**
RES: 0/1 (Restricted/Allowed)

Note 1: This data is valid when SFI 180=1 is set to the counter station.

Note 2: This data is valid when SFI 179=1 is set to the self station.

[Example]

The following figure shows examples of data assignment when SFC=1 is set for a calling station 4000 and SFC=2 is set for a called station 4001.

Service Feature Class (SFC): 1	Service Feature Class (SFC): 2
Calling Station: 4000	Called Station: 4001

• To hide Station Numbers on the Calling and Called Stations

LCD: ＊＊＊＊＊＊＊＊



Originates



LCD: ＊＊＊＊＊＊＊＊

SFC = 1
SFI 179 = 1 (Allow the self station number display elimination on the counter station)
SFI 180 = 1 (Allow the counter station number display elimination on the self station)

SFC = 2
SFI 179 = 1 (Allow the self station number display elimination on the counter station)
SFI 180 = 1 (Allow the counter station number display elimination on the self station)

• To hide a Calling Station Number on the Called Station

LCD: 4001 (Called Station Number)



Originates



LCD: ＊＊＊＊＊＊＊＊

SFC = 1
SFI 179 = 1 (Allow the self station number display elimination on the counter station)
SFI 180 = 0 (Restrict the counter station number display elimination on the self station)

SFC = 2
SFI 179 = 0 (Restrict the self station number display elimination on the counter station)
SFI 180 = 1 (Allow the counter station number display elimination on the self station)

• To hide a Called Station Number on the Calling Station

LCD: ＊＊＊＊＊＊＊＊



Originates



LCD: 4000 (Calling Station Number)

SFC = 1
SFI 179 = 0 (Restrict the self station number display elimination on the counter station)
SFI 180 = 1 (Allow the counter station number display elimination on the self station)

SFC = 2
SFI 179 = 1 (Allow the self station number display elimination on the counter station)
SFI 180 = 0 (Restrict the counter station number display elimination on the self station)

C-176D Call Park Group -D

General Description

This feature enables the Desktop terminal (including Soft Phone) users to make a connected call on Call Park Hold state by pressing the HOLD key, or CALL PARK GROUP -D HOLD Feature key, on the Desktop terminal. The held call can be retrieved from any member Stations of the Group with the CALL PARK GROUP -D Feature key. This feature allows the Desktop terminal users to place a call on hold without occupying a line (My Line/Sub Line) of the telephone system. Thus, Station can originate/receive another call by using a Line key while holding the call. Three options are available for retrieving a call in Call Park Group -D hold: 1) from the same line occupied by the call immediately before being held; 2) from the Prime Line; 3) from the My Line. (refer to Operating Procedure) This setting can be configured with System Data.

Note: The following keys are typically used for this feature:

HOLD key:
A Flex key. It puts a call in Call Park Group -D hold state or places a call on hold.

CALL PARK GROUP -D HOLD Feature key:
A user-defined Line/Feature key to place a call in Call Park Group -D hold state.

CALL PARK GROUP -D Feature key:
A user-defined Line/Feature key to retrieve a call which has been placed in Call Park Group -D hold state.

Operating Procedure

To hold a Central Office/TIE Line or Station Call with the CALL PARK GROUP -D HOLD Feature key

1. Press the CALL PARK GROUP -D HOLD Feature key on the Desktop terminal.
2. A call in progress is placed on hold. The lamp of the CALL PARK GROUP -D Feature key flashes (flashes green for the Operating Desktop terminal, flashes red for the other members).
3. The Operating Desktop terminal hears SERVICE SET TONE (SST).
4. Replace the handset.

Note: When you press the HOLD key (FLEX key) on the Desktop terminal, a call is placed on hold as usual (this feature is not activated).

To hold a Central Office/TIE Line or Station Call with the HOLD Feature key

1. Press the HOLD Feature key on the Desktop terminal.
2. The call in progress is placed on hold. The lamp of the CALL PARK GROUP -D Feature key flashes (flashes green for the Operating Desktop terminal, flashes red for the other members).
3. The Operating Desktop terminal hears SERVICE SET TONE (SST).
4. Replace the handset.

To retrieve a Held Call

1. Lift the handset. **Note 1, Note 2**
2. Press a the CALL PARK GROUP -D Feature key. **Note 1, Note 2**

3. The call between the operating Desktop terminal and the held call is reestablished. The lamps of the CALL PARK GROUP -D Feature keys on all members are extinguished.

Note 1: These steps can be performed in reverse order except for using Soft Phone with a USB handset. When the CALL PARK GROUP -D Feature key is pressed first, the held call is not reconnected to the Station until the handset is lifted.

Note 2: In the case of Soft Phone with a USB handset, these steps cannot be performed in reverse order. Be sure to lift the handset before pressing a CALL PARK GROUP -D Feature key; otherwise, the Held Call cannot be retrieved.

To retrieve a Held Call after Operating Desktop terminal is recalled by Party on Hold

1. When Call Back Timer for Call Park Group -D expires, the operating Desktop terminal is recalled by the party on hold.
2. Lift the handset. **Note 3, Note 4**
3. Press the CALL PARK GROUP -D Feature key. **Note 3, Note 4**
4. The call between the operating Desktop terminal and the held call is reestablished.

Note 3: These steps can be performed in reverse order except for using Soft Phone with a USB handset. When the CALL PARK GROUP -D Feature key is pressed first, the held call is not reconnected to the Station until the handset is lifted.

Note 4: In the case of Soft Phone with a USB handset, these steps cannot be performed in reverse order. Be sure to lift the handset before pressing the CALL PARK GROUP -D Feature key; otherwise, the Held Call cannot be retrieved.

Service Conditions

1. This feature is available to Desktop terminal.
2. A Station cannot belong to more than one Group.
3. The CALL PARK GROUP-D Feature key can be assigned only to keys numbered lower than 25 with the AKYD command.
4. The CALL PARK GROUP-D Feature key cannot be assigned on a DSS console in DSS mode.
5. You cannot retrieve a held call when activating Intercom, waiting for Bridge Call P-RLS key, reading NSG, using Boss-Secretary Service, making a Voice Call, and registering the SPEED CALLING-ONE TOUCH Feature key.
6. There is a difference between this feature and CALL PARK [C-29]. Thus, a call which has been placed on hold through this feature cannot be retrieved by using CALL PARK Retrieval Code.
7. When ASYDL/ASYDN, SYS1, Index 679, Bit 4 is assigned as "0", each group must consist of stations belonging to the same Module Group (MG).
8. A maximum number of Groups that can be used by this feature depends on the System Data setting (ASYDL/ASYDN, SYS1, Index 679, Bit 4):

   a.) ASYDL/ASYDN, SYS1, Index 679, Bit 4=0
   (Call Park Group across MGs/IMGs is unavailable.)
   IMG/MMG/UMGi/RNIP System: 255 Groups per Local Partition (LP) (i.e., 2 MGs (Module Groups)) **Note 1**

   Note 1: Do not assign the same Group Number to more than one Module Group (MG) in the same LP.

   b.) ASYDL/ASYDN, SYS1, Index 679, Bit 4=1
   (Call Park Group across MGs/IMGs is available.)
   IMG/MMG System: 255 Groups per System
   UMGi/RNIP System: 255 Groups per node **Note 2**

   Note 2: For UMGi/RNIP System, the stations accommodated in different nodes cannot belong to the same group.

   When enabling/disabling ASYDL/ASYDN, SYS1, Index 679, Bit 4 data setting, delete all data assigned through the APGS, APGN and ADGN commands in advance.
9. When ASYDL/ASYDN, SYS1, Index 679, Bit 4 is set to "0", all Multi Lines registered on a Desktop terminal must be accommodated in the same Module Group (MG).
10. A maximum number of member stations in a single group is 768.
11. A maximum number of the CALL PARK GROUP -D Feature keys that can be assigned to a Desktop terminal is eight.
12. The number of calls which can be placed on hold at the same time depends on the number of the CALL PARK GROUP -D Feature keys assigned to a Desktop terminal.

13. Only station numbers of My Lines can be assigned as a member of the group.

14. This feature is not available for TRUNK LINE APPEARANCE -D [T-23D]. The HOLD key is not operational during an INTERCOM Call.

15. This feature is available only when an operating station is engaged in a two-party call with another station or a trunk. This feature is not available when the operating station hears RBT, is engaged in a three-party call, or is engaged in a call with an Attendant Console.

16. A party which has been placed on hold through this feature can only release the call. Any other operations such as hooking cannot be performed.

17. When using this feature with the HOLD key, EXCLUSIVE HOLD -D [E-4D] via the HOLD key cannot be performed. EXCLUSIVE HOLD -D [E-4D] needs to be performed by dialing designated Access Code. When operating this feature with the CALL PARK GROUP -D HOLD Feature key, both EXCLUSIVE HOLD -D [E-4D] via the HOLD key and NON-EXCLUSIVE HOLD -D [N-7D] are available.

18. When Call Back Timer for Call Park Group -D specified by ASYDL, SYS1, Index 661 expires, the operating station will be recalled. At this time, the lamp of the CALL PARK GROUP -D Feature key flashes (120 millisecond interval). The setting of ASYD, SYS2, Index 12, Bit 0 determines whether Automatic Recall Ringer is required or not required.

19. When you retrieves a call on hold through this feature, the elapsed time on your LCD is reset to one second.

20. When you press the CALL PARK GROUP -D Feature key without lifting the handset, the LCD displays the information on the held party, but you cannot retrieve the call. When you press the ANSWER/SPEAKER key or go off-hook within ten seconds after pressing the CALL PARK GROUP -D Feature key, you can retrieve the call on hold (after a lapse of ten seconds, a call is made to the party on hold).

21. When all of Multi Lines on your Desktop terminal are busy, you cannot retrieve a call on hold. You can avoid this problem by assigning more than one line (other than Multi Lines in Station Hunting group) exclusively for making a call (station number which is hidden from public view).

22. The lamp of the Feature key corresponding to the line on which you hold a call in progress blinks in green until someone retrieves the call on hold. (The lamp state is not affected if another call is held with this feature on a different Desktop terminal.)

23. Pressing the CALL PARK GROUP -D Feature key during a call connects you to a held call. In this case, when Automatic Hold [A-68] is enabled, the previously active call is automatically placed on hold. If Automatic Hold is disabled, the previously active call is disconnected.

24. You cannot place an ISDN terminal on hold through this feature. An ISDN terminal placed on hold with the line in use, as usual.

25. When you press a CALL PARK GROUP -D Feature key and another Desktop terminal uses the line where you set Call PARK GROUP - D to make/receive a call, the indication on your LCD disappears.

26. The CALL PARK GROUP -D Feature key cannot be assigned to a Desktop terminal Soft key.

27. An operation key placing a call on hold can be selected either HOLD key, which is a Flex key, or CALL PARK GROUP -D HOLD Feature key which is Feature key by system data assignment (ASYDL, SYS1, Index 678, Bit 2).

(a) HOLD key (ASYDL, SYS1, Index 678, Bit 2=0) is selected;

- Pressing the HOLD key is recognized as an operation placing a call on hold for CALL PARK GROUP - D [C-176D]. The call cannot be placed on hold regularly on the operating terminal by itself.
- When a call is placed on hold by the HOLD key on the Desktop terminal without registration in a group by ADGN command, it is recognized as an operation placing a call on hold regularly on the operating terminal by itself.

(b) The CALL PARK GROUP -D HOLD Feature key (ASYDL, SYS1, Index 678, Bit 2=1) is selected;

- The CALL PARK GROUP -D HOLD Feature key (AKYD, FKY=183) needs to be assigned to a Feature Key.
- Pressing the CALL PARK GROUP -D HOLD Feature key is recognized as an operation placing a call on hold for Call Park Group - D [C-176D].
  Pressing the HOLD key is recognized as an operation placing a call on hold regularly on the operating terminal by itself.
- When a call is placed on hold by the HOLD key on the Desktop terminal without registration in a group by the ADGN command, it is recognized as an operation placing a call on hold regularly on the operating terminal by itself.
  Also, when a call is placed on hold with the CALL PARK GROUP -D HOLD Feature key, an indication meaning “Restricted” (**Note 3**) appears on the LCD and the call is maintained.
- On the Soft Phone using Soft Phone-PS [S-158] or Soft Phone WLAN Handset [S-159], CALL PARK GROUP-D [C-176D] is available only when operating with the HOLD key.
  EXCLUSIVE HOLD -D [E-4D] via the HOLD key, NON-EXCLUSIVE HOLD -D [N-7D], and CALL PARK GROUP -D [C-176D] via the CALL PARK GROUP -D HOLD Feature key are not available.

Note 3: An actual indication on the LCD is different depending on the language used on the LCD and the LCD type of Desktop terminal. As an English indication on a 24-digit LCD Desktop terminal for example, “REST” appears on the LCD.

28. In the case of Soft Phone with a USB handset, be sure to lift the handset before pressing the CALL PARK GROUP -D Feature key; otherwise, the held call cannot be retrieved.
29. 139MTC/139TC on the ADAI command (No Answer Timer determined by ASYD, SYS1, Index 139) does not take effect in this feature.
30. When the held party is an IP terminal and the connection with the party is disconnected due to the accidents such as network failure or unplugging of cables, the lamp of the CALL PARK GROUP -D Feature key on the Desktop terminal that held the call keeps flashing. If a user of this Desktop terminal tries to retrieve the call in this condition, dial tone is heard and flashing of the lamp of the CALL PARK GROUP -D Feature key will stop.

Programming

STEP 1: **ASYD** - SYS1, Index 42, Bit 7. Service Set Tone, Special Dial Tone. 0/1: Continuous/Burst.
SYS1, Index 79, Bit 0=1 (Line Preference Service via AKYD is used.)
SYS1, Index 139. No Answer Timer (Default = 30 seconds) **Note 1**

Note 1: When ASYDL, SYS1, Index 661, Bit 4 to Bit 6=03 HEX or 04 HEX, this data becomes invalid.

SYS1, Index 388, Bit 2=1 (Desktop terminal Call Hold service is used.)
SYS1, Index 388, Bit 3. Tone setting when Call Hold is completed. 0/1: Dial Tone (DT)/Service Set Tone (SST).
SYS2, Index 12, Bit 0. 0/1: Automatic Recall Ringer is required/not required for Non-Exclusive Hold.

STEP 2: **ASYDL** - SYS1, Index 661, Bit 0 to Bit 3. Call Back Timer for Call Park Group -D and Call Park Group (PS). MTC = 1 - 15 (1 HEX to F HEX) seconds. Timer value = MTC × TC.
SYS1, Index 661, Bit 4 to Bit 6. Timer Class (TC) for Call Park Group -D and Call Park Group (PS). TC3 (03 HEX) = 2 seconds, TC4 (04 HEX) = 30 seconds. **Note 2**
SYS1, Index 678, Bit 2, operation key to hold a call on a terminal in Call Park Group -D state. 0/1: HOLD key/CALL PARK GROUP -D HOLD Feature key **Note 3**

Note 2: When this data is 03 HEX or 04 HEX, ASYD, SYS1, Index 139 data becomes invalid.

Note 3: Assign this data when selecting an operation key to hold a call on a terminal in Call Park Group -D state.

STEP 3: **ASYDL/ASYDN** - SYS1, Index 1262, Bit 3=1 (Retrieving a Call in Call Park Group -D Hold from a Fixed Line is enabled.) **Note 4**
SYS1, Index 1262, Bit 4. Fixed Line Used for Retrieving a Call in Call Park Group -D Hold.
0/1: Prime Line/My Line **Note 4**, **Note 5**

Note 4: This step is required to specify a line to be used for retrieving a call placed in Call Park Group -D hold state.

Note 5: When setting this parameter, you must enable “Retrieving a Call in Call Park Group -D Hold from a Fixed Line” (ASYDL/ASYDN, SYS1, Index 1262, Bit 3=1). This setting is not required if My Line is assigned as Prime Line.

The following data setting is required when assigning the stations belonging to different MGs and LPs to the same group:
SYS1, Index 679, Bit 4=1 (Enable Call Park Group across MGs/IMGs) **Note 6**

Note 6: Delete all existing data assigned through the APGS, APGN and ADGN commands in advance.

STEP 4: **ASDT** - Assign Station Data of the Desktop terminal using this service.

STEP 5: **AKYD** - Assign the CALL PARK GROUP -D Feature key. Assign FKY 75-82 to the Desktop terminal key. **Note 7**

Assign the CALL PARK GROUP -D HOLD Feature key. Assign FKY 183 to the Desktop terminal key **Note 8**

Note 7: When only one CALL PARK GROUP -D Feature key is assigned, assign FKY 75 to the Desktop terminal key. When more than one CALL PARK GROUP -D Feature key is assigned, FKY 75 must be assigned first, and then FKY 76, FKY 77, and continuing through the remaining FKY 78 - FKY 82.

Note 8: Assign this data when using the CALL PARK GROUP -D HOLD Feature key.

STEP 6: **ASFC** - Assign Service Feature Restriction Class of the Station.

SFI = 67 (Call Park - Access and Answer)
SFI = 68 (Call Park - Called)
RES = 1 (Allowed)

STEP 7: **ADGN** - Assign the Group Number to each Station.

TN:	Tenant Number
STN:	Station Number
GROUP NO:	Group Number (1 - 255)
RINGER:	Ringer Information Not used. This parameter should be fixed to 0.

C-177 Caller ID on Call Waiting

(Series 8400 software feature)

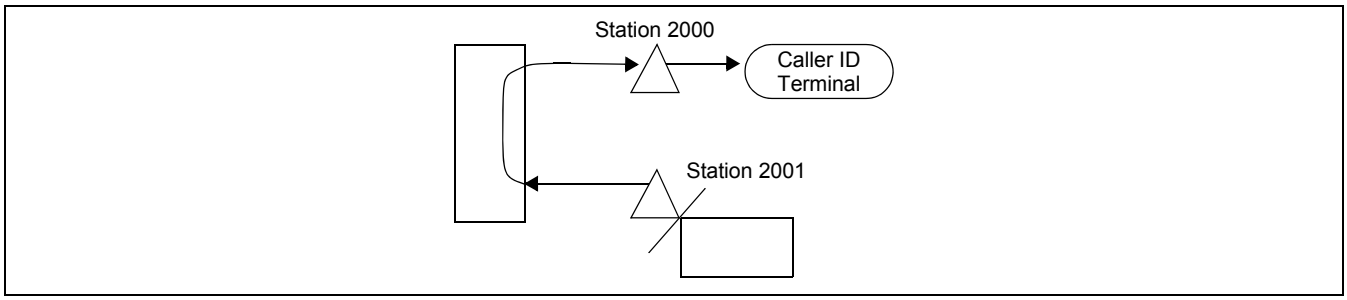
General Description

This feature allows Incoming Trunk Calls with Caller ID information to be displayed on a Caller ID device while the user is on another Station or Trunk Call. A standard Trunk Call Waiting Indication Tone is sent to the Busy Station. Caller ID information for the Calling Party that is in a Call Waiting Mode is sent to the Caller ID device. This allows the user to see the Caller ID information on the new caller and decide if they wish to use a switch hook flash to answer the second call. The user may also alternate between the two calls by switch hook flash.

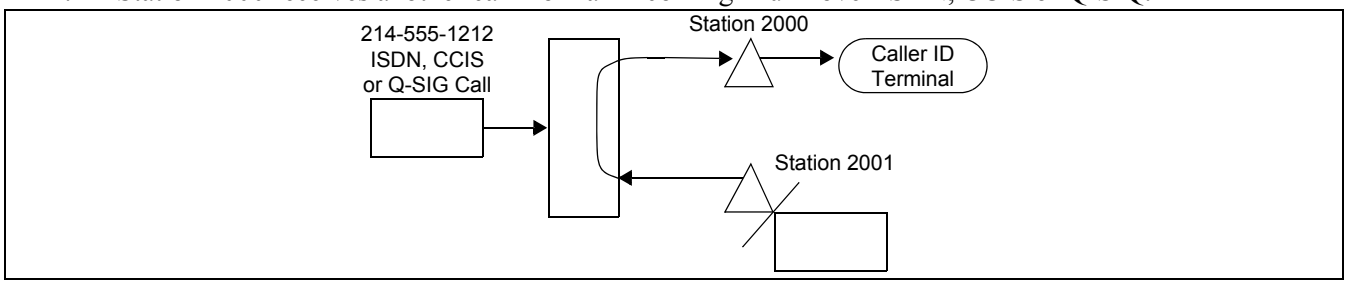
Note: This feature is only available in North America.

Operating Procedure

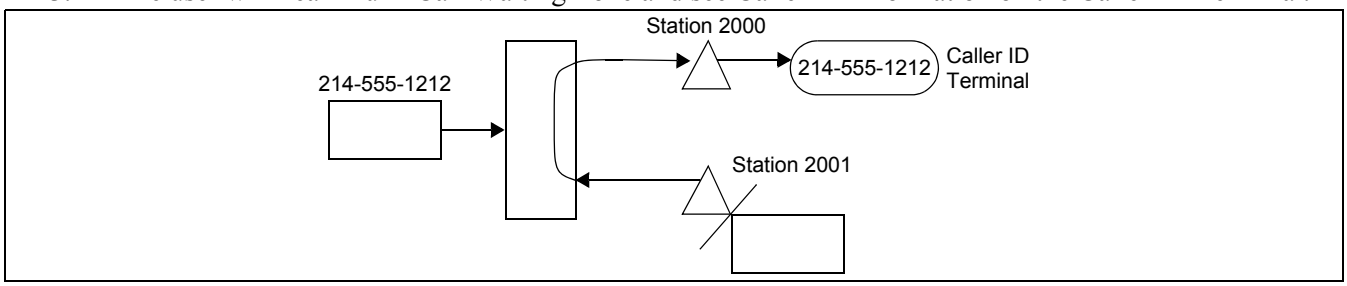
1. Station 2000 is on a call with a Station or Trunk 2001.



2. Station 2000 receives another call from an Incoming Trunk over ISDN, CCIS or Q-SIQ.



3. The user will hear Trunk Call Waiting Tone and see Caller ID information on the Caller ID Terminal.



4. No manual operation is required to activate or cancel Caller ID on Call Waiting.

Service Conditions

1. Service Conditions are the same as those of CALL WAITING - TERMINATING [C-12] for ordinary telephone sets.
2. This feature requires a telephone that is capable of Caller ID, or a Caller ID adaptor that must conform to Bellcore SR-TSV-002476.
3. This feature supports only Multiple Data Message Format (MDMF) for all data in the off-hook data format as per GR-575-CORE R3-14.
4. For Name Display from ISDN, the ISDN Interface must be an NI-2 Interface type.
5. Only TEC 3 and TEC 7 are supported for the call waiting to be answered.
6. This feature cannot be activated for an Analog Caller ID Trunk.
7. This feature requires the SPA-8RSTY-A Register Card and assigning Sender Trunk 937 to this card to send out SAS and CAS Signal to the terminal. It also requires assigning new Register Trunk 940 specified for Caller ID on Call Waiting.
8. Calling Party Caller ID data must be provided by the Central Office.
9. This feature will only provide a single Call Waiting Tone.
10. The Caller ID information is transmitted only once to the Caller ID Terminal, no attempt to re-send will be made if an error is detected (in this case, normal Call Waiting is provided).
11. As per Bellcore's GR575 Specs R3-14: During off-hook data transmission, the SPCS will send the Caller ID Call Waiting data in the Multiple Data Message Format for all Caller ID features enabled on the line.
12. System Message 36-D (Caller Waiting Caller ID Error) will be generated when an error occurs with the Call Waiting Caller ID process. This message will contain the Called Party LENS, Calling Party LENS, Sender (937) LENS, and Register (940) LENS (if available). It will also contain information designating if the error was a CAS ACK time-out, Sender time-out, or an equipment not available error.

Programming

STEP 1: **ASYD** - SYS1, Index 145, Bit 6: Leave as data "0". SYS1, Index 145. Assign the Recall Timer for a call extended by the Attendant Console that is not answered. Normally assign data 00H for 30 seconds (RAM Data 9F for 30 seconds).

STEP 2: **ASYDL** - Assign new SYS1, Index 803, Bit 6 for Call Waiting Caller ID PRINT ERR MSG flag. Bit 6=0/1 (No/Yes).

Assign new SYS1, Index 667, Bit 0 to Bit 3 for CASWT Timer. MTC * 16 milliseconds.

Assign new SYS1, Index 667, Bit 4 to Bit 7 for CASST Timer. MTC * 16 milliseconds.**Note 1**

Assign new SYS1, Index 668, Bit 0 to Bit 7 for FSWT Timer. MTC * 16 milliseconds.

SYS1, Index 669 for SAS Wait Timer. MTC * 16 milliseconds.

SYS1, Index 670 for OSI Interval. MTC * 16 milliseconds.

SYS1, Index 806 for CPE ACK Time-out.

SYS1, Index 807 for Complete Time-out Timer

SYS1, Index 808 for ACKWT Timer. MTC * 16 milliseconds.

STEP 3: **ASFC** - Set Service Feature Index 188 for Analog Phone to "1".

STEP 4: **ATRK** - Assign 937 Sender Trunk to Senders of PA-8RSTY-A Register Card. Assign 940 Register Trunk to Registers of PA-8RSTY-A Register Card.

STEP 5: **MBTK** - Assign "0" to busy bit of all 937 Sender Trunks - Top of Register Card. Assign "0" to busy bit of all 940 Register Trunks - Bottom of Sender Card.

STEP 6: **ARTD** - For CALL WAITING - TERMINATING to operate automatically for the TIE Lines and DID Routes, assign CDN 46:CW as data "1".

Note 1: Assign data 50 Hex for ASYDL, SYS1, Index 667, Bit 4 to Bit 7 when the PA-8RSTAE card is used for this feature.

Note: To change the data for existing route (RT), initialize or reboot the trunk device such as a circuit card or MG.

C-192 Called Station Name Display on CF

(Series 8700 software feature)

General Description

This feature allows the name data of a Call Forwarded Station to be displayed on the Desktop terminal target of the forward.

Note: This feature is only available in North America.

For example:
1. Station 1000 with name JOHN DOE sets Call Forwarding-All Calls to Station 1001.
2. Route 1 TRK 1 calls station 1000.
3. 1001 rings showing the name display of Station 1000.

Operating Procedure

No manual operation is required.

Service Conditions

1. The target Station must be a Desktop terminal.
2. This feature is valid for Call Forward-All Calls only.
3. The Call Forwarded Station must have Name Display Service assigned.
4. If the trunk is forwarded all multiple times, the displayed name is the first called forwarded Station.
5. This feature only functions for incoming trunk calls.
   a.) Supported Trunk types are DID, ISDN and TIE LINE.
   b.) The incoming route cannot have IND set in Route data for this feature to function.
6. The forwarded party must be a TEC 12 or TEC 18.
7. Digital Voice Mail integration will integrate with calling and called number the same when forwarded from a Station with Name Follow on CF service.

Interactions

1. This feature will override the normal display of the Route on the incoming call.
   Example: CCSA, DID

Programming

STEP 1: **ASFC** - Assign Service Feature Restriction Class of the Station.
SFI = 202 (Name Display Follow on CF)
RES = 1 (Allowed)

C-193 Call Forwarding - Don’t Answer while Call Waiting - Termination

General Description

This feature enhancement is to make Station and Trunk calls behave the same when Call Forwarding Don’t Answer with Call Waiting is used.

Note: This feature is only available in North America.

Operating Procedure

When Call Waiting - Terminating [C-12] is in service:

(A) When STN-A’s CFDNA is idle.

1. STN-A sets CFDA to STN-E.
2. STN-A talks with TRK-A (or STN-C).
3. Incoming call from STN-D to STN-A.
4. STN-A call waits.
5. CW timer (ASYD, SYS1, Index 152) expires.
6. STN-D forwards to STN-E.

(B) When STN-A is already CW, CONF, etc. And ASFC SFI 129 = 1(Refer to service condition of C-12).

1. STN-A sets CFDNA to STN-E.
2. STN-A talks with TRK-A (or STN-C).
3. STN-D calls STN-A.
4. STN-D forwards to STN-E.

(C) When STN-E(CFDNA Destination) is busy and not in a UCD group and ASFC SFI 129 =1

1. STN-A sets CFDA to STN-E.
2. STN-A talks with TRK-A (or STN-C).
3. STN-D calls STN-A
4. STN-D call waits to STN-A
5. CW timer (ASYD, SYS1, Index 152) expires.
6. Since STN-E is busy STN-D will continue to Call wait at STN-A.
7. CW timer is reset to ASYD, SYS1, Index 152.
8. When CW timer expires again the idle busy state of STN-E is checked and the call will forward if possible or the CW timer will be reset.

(D) When STN-A is already CW, CONF, etc. (Refer to service condition of C-12) and STN-E is in a busy state and not in a UCD group.

1. STN-A sets CFDA to STN-E.
2. STN-A talks with TRK-A (or STN-C).
3. STN-D calls to STN-A.

4. STN-D does not forward, STN-D shows “BUSY” and receives busy tone.

(E) When STN-E is busy and STN-E is in a UCD group.

1. STN-A set CFDA to STN-E.
2. STN-A talks with TRK-A (or STN-C).
3. STN-D calls to STN-A.
4. STN-D call waits to STN-A.
5. CW timer (ASYD, SYS1, Index 152) expires.
6. STN-D forwards to the UCD group of STN-E.

When Call Waiting - Originating [C-31] is in service:

The same operating Call Waiting - Originating procedure is required and the call will complete as above.

Service Conditions

This service requires ASYD, SYS2, Index 10, Bit 7=1 (Call Waiting Tone at intervals).

Programming

All programming for the following features must be completed for the enhancement to function. (Please see the Data Programming Manual - Business)

- Call Waiting - Terminating [C-12]
- Call Waiting - Originating [C-31]
- Call forwarding - Don’t Answer [C-3]

STEP 1: **ASYD** - SYS1, Index 152 - Interval for Call Waiting Indication Tone to Busy Station.
Default is approximately 10 seconds.
SYS1, Index 10, Bit 7=1 Call Waiting Tone at intervals.

STEP 2: **ASFC**
SFI:129=1 (CFDA while CW)
SFI:130=0/1 (ATT Transfer while CW in case CFDA busy)

FCCS Programming

All programming must be assigned in each FCCS node.

C-194 Call Park Group

General Description

This feature enables the Station Users to "Park" calls against their own Extension Numbers. Maximum of five calls can easily be parked and retrieved from any Station within the Call Park Deluxe Group by just pressing the access code or the pre-assigned Line/Feature key of a Desktop terminal (including Soft key). When retrieving a parked call from the station out of the Call Park Deluxe group, station must dial the access code of CALL PARK [C-29] (or press the Feature key of CALL PARK [C-29]) + station number of the station actually parked the call.

Operating Procedure

To set this feature from a station using the access code

1. While connected to the COT, TIE line, Station or a PS
2. Momentarily press the switch hook. Receive Special Dial Tone.
3. Dial the Call Park Deluxe Access Code. Receive Service Set Tone.
4. Replace the handset.

To set this feature from a Desktop terminal station using the Line/Feature key

1. While connected to the COT, TIE line, Station or a PS
2. Press the Line/Feature key of which Call Park Deluxe is assigned.
3. Receive Service Set Tone.
4. Call Park Deluxe has been set.

To retrieve a parked call from the originating station

1. Dial the Call Park Deluxe Retrieval Code. The Parked Call is reconnected.

To retrieve a parked call from a different station within the same Call Park Deluxe Group using the access code.

1. Dial the Call Park Deluxe Retrieval Code. The Parked Call is reconnected.

To retrieve a parked call from a different station within the same Call Park Deluxe Group using the Line/ Feature key of a Desktop terminal.

1. Press the Line/Feature key of which Call Park Deluxe Retrieval Code is assigned. The Parked Call is reconnected.

Service Conditions

1. Calls parked by a station are automatically parked on that station's extension number. The My-Line number is used for a Desktop terminal.
2. CALL PARK may be activated only during calls involving two parties.
3. A station can only park one call at a time. If a station attempts to park a second call, Reorder Tone will be heard.
4. This feature may be assigned to stations via CLASS OF SERVICE - INDIVIDUAL [C-15] programming in System Data.
5. A user can set this feature by dialing a Call Park Deluxe access code or by pressing the Line/Feature key on a Desktop terminal (Including the Soft key).
6. Stations can originate and receive calls while having a call parked.

7. The following features cannot be activated to a station that has set Call Park Deluxe:
   BUSY VERIFICATION[B-3]
   CALL WAITING - ORIGINATING[C-31]
   CALL WAITING - TERMINATING[C-12]
   EXECUTIVE RIGHT-OF-WAY[E-1]
8. A station in an ATTENDANT-CONTROLLED CONFERENCE [A-2] cannot be parked.
9. Station cannot be assigned in multiple Call Park Deluxe group (Retrieving Group).
10. A parked call will Re-call to the station that has set the Call Park Deluxe. The Re-call Timer is pre-programmed by the System Data.
11. Call Park Deluxe is denied in the following station status:
    BOSS - SECRETARY OVERRIDE -D [B-5D]
    BUSY VERIFICATION [B-3, B-9]
    CALL TRANSFER[C-10, C-11]
    CALL WAITING - ORIGINATING[C-31]
    CALL WAITING - TERMINATING[C-12]
    CONSULTATION HOLD - ALL CALLS [C-17, C-58]
    DATA LINE SECURITY [D-12, D-66]
    EXCLUSIVE HOLD -D[E-4D]
    EXECUTIVE RIGHT-OF-WAY [E-1, E-8]
    NON-EXCLUSIVE HOLD -D[N-7D]
    PRIVACY RELEASE[P-18]
    THREE-WAY CALLING [T-2]
    VOICE CALL [V-2, V-7]
12. An Attendant Console cannot be parked.
13. A Call Park Deluxe connection remains parked until retrieved or until the party disconnects.
14. A Parked Call can only be retrieved from an Idle Station.
15. An ISDN terminal user cannot use this feature.
16. An Incoming Call from the ISDN Trunk can be Parked from a Station/Attendant.
17. A station can park only one station/trunk using Call Park Deluxe feature simultaneously.
18. Call Park Deluxe is not available via FCCS network.
19. The Night Attendant Console cannot park a call.
20. A Parked Call will recall to the Station that has set the CALL PARK after a predetermined period of time assigned by the system data (ASYD, SYS1, Index 142) has elapsed (the default value is 500 seconds). If the recalling parked call is from an analog C.O line (except MC&MG-COT/MG-COT), it will be released after a predetermined period of time assigned by the system data (ASYD, SYS1, Index 139) has elapsed (the default value is 30 seconds). As for other than the analog C.O line (except MC&MG-COT/MG-COT), the recalling will be continued.
21. A parked call by the Desktop terminal is always retrieved by using the My-Line of the Desktop terminal. Even if a Desktop terminal parks a call by using the Sub Line, the parked call is retrieved by using My-Line of the Desktop terminal.

Interactions

1. A call parked using Call Park Deluxe feature is automatically parked by the CALL PARK [C-29] feature at the same time. Therefore, call also can be retrieved by the call retrieving procedure of CALL PARK [C-29] feature.

2. When a SIP Handler Controlled SIP terminal is in Multi Line Mode, the SIP terminal cannot Park the ongoing call but the SIP terminal can retrieve a call Parked from a different terminal.

Programming

STEP 1: **ASYD** - SYS1, Index 139, Bit 0 to Bit 3 (MTC), Bit 4-6 (TC). Assign No Answer Timer for Call Park Deluxe recall ringing. If a Parked Call from an analog C.O line (except MC&MG-COT/MG-COT) recalls and the set timer value (default is 30 seconds) elapses, the call will be released. SYS1, Index 142. For Call Park Deluxe Recall, the Call Park Set Timer Counter is assigned in Bit 0 to Bit 3. The Timer Class is assigned in Bit 4 to Bit 6.

SYS2, Index 1, Bit 0 and Bit 1. CONSULTATION HOLD - ALL CALLS [C-17] allowed or denied. 00: denied; 01: Originating and Terminating Calls allowed, Tandem Call denied; 11: All calls allowed.

STEP 2: **ANPD/ANPDL** - Reserve a Number Level for Feature Access and Retrieval. Select Normal Service (N), Hooking Service (H), or Busy Service (B) from the combo box for Connection Indexes. Assign NND in accordance with a predetermined Numbering Plan.

STEP 3: **ASPA/ASPAL** - Assign an Access Code to:

Assign an access code to set the Call Park Deluxe Feature:

CI="H: Hooking"

SRV="SSCA: Service Code Appendix"

a) Call Park Deluxe access code, SIDA=146 (Call Park Deluxe Hold: Hold-1)
b) Call Park Deluxe access code, SIDA=147 (Call Park Deluxe Hold: Hold-2)
c) Call Park Deluxe access code, SIDA=148 (Call Park Deluxe Hold: Hold-3)
d) Call Park Deluxe access code, SIDA=149 (Call Park Deluxe Hold: Hold-4)
e) Call Park Deluxe access code, SIDA=150 (Call Park Deluxe Hold: Hold-5)

Assign an access code to retrieve from the Call Park Deluxe Feature:

CI="N: Normal"

SRV="SSCA: Service Code Appendix"

a) Call Park Deluxe access code, SIDA=146 (Call Park Deluxe Hold: Retrieval-1)
b) Call Park Deluxe access code, SIDA=147 (Call Park Deluxe Hold: Retrieval-2)
c) Call Park Deluxe access code, SIDA=148 (Call Park Deluxe Hold: Retrieval-3)
d) Call Park Deluxe access code, SIDA=149 (Call Park Deluxe Hold: Retrieval-4)
e) Call Park Deluxe access code, SIDA=150 (Call Park Deluxe Hold: Retrieval-5)

Note: Same access code can be used for Parking and Retrieving a call by assigning the same number but making the CI to "N: Normal" for Retrieving and "H: Hooking" for Parking.

STEP 4: **AKYD** - For Desktop terminal, Call Park Deluxe may be assigned to a Line/Feature key. Assign the following parameters.

KYN: Key Number (1-40)

KYI=1 (Function Key)

a.) FKY=158 (Call Park Deluxe Hold/Retrieve-1)
b.) FKY=159 (Call Park Deluxe Hold/Retrieve-2)

c.) FKY=160 (Call Park Deluxe Hold/Retrieve-3)
d.) FKY=161 (Call Park Deluxe Hold/Retrieve-4)
e.) FKY=162 (Call Park Deluxe Hold/Retrieve-5)

STEP 5: **ADSL** - For Desktop terminal, Call Park Deluxe may be assigned to a Soft key. Assign the following parameters. This command is required only for assigning the function to each Soft key.

SN = 1 (Originating - DT connection)

a.) FKY=158 (Call Park Deluxe Hold/Retrieve-1)
b.) FKY=159 (Call Park Deluxe Hold/Retrieve-2)
c.) FKY=160 (Call Park Deluxe Hold/Retrieve-3)
d.) FKY=161 (Call Park Deluxe Hold/Retrieve-4)
e.) FKY=162 (Call Park Deluxe Hold/Retrieve-5)

SN = 4 (Communication in progress - Converse)

a.) FKY=158 (Call Park Deluxe Hold/Retrieve-1)
b.) FKY=159 (Call Park Deluxe Hold/Retrieve-2)
c.) FKY=160 (Call Park Deluxe Hold/Retrieve-3)
d.) FKY=161 (Call Park Deluxe Hold/Retrieve-4)
e.) FKY=162 (Call Park Deluxe Hold/Retrieve-5)

STEP 6: **ASFC** - Assign a Service Feature Restriction Class that allows SFI 220 (Call Park Deluxe Entry/Retrieve) to the applicable stations. This allows the station to park a call and to retrieve a parked call using Call Park Deluxe feature. Assign stations a Service Feature Restriction Class that allows SFI 68 (Call Park - Called). This allows a station to be parked by another station.

STEP 7: **ASCS** (Assignment of Call Park Group/DM)

Assign the Call Park Deluxe Group Number.

TN: Tenant Number (1-63)

STN: Station Number

GPN: Call Park Group Number (1-4095)

Note: When the station has the TELN (LDM), skip (7) and assign (8).

STEP 8: **ACPSL** (Assignment of Call Park Group Service Data/LDM)

Assign the Call Park Deluxe Group Number.

UGN: User Group Number (1-255)

TELN: Telephone Number (maximum 16 digits)

GPN: Call Park Group Number (1-4095)

C-195 Call History

General Description

This feature enables call history to be stored and displayed on the LCD. As for the system side Call History (hereinafter System Call History), key lamp or display is used to notify users of unanswered calls. Also, Call History inside of this feature is divided in system side and terminal side as explained below.

- **System Call History**

  It is the feature that can be used by enabling the ASFC/ASFCN command SFI 238 in the terminal and maintains in the system side a record of incoming calls. Outgoing call history and Incoming call history operations are explained in the following table. For a detailed explanation of this feature operation procedure and behavior, see Operating Procedure and the User's Guide of each terminal.

	Outgoing call history	Incoming call history
Terminal operation	Cursor key (Left side) **Note 1**	Feature key/Soft key

Note 1: If System Call History is used, Outgoing call history of Terminal Call History cannot be used. For that reason, use REDIAL [R-60].

- **Terminal Call History**

  It is the feature that can be used by enabling the ASFC/ASFCN command SFI 248 in the terminal and maintains in the terminal side an internal record of incoming/outgoing calls. Outgoing call history and Incoming call history operations are explained in the following table. For a detailed explanation of this feature operation procedure, see the User's Guide of each terminal.

	Outgoing call history	Incoming call history
Terminal operation	[Menu screen] - Select [History] - Select [History] - Select [Outgoing]	[Menu screen] - Select [History] - Select [Missed]/[Incoming]

Also, the pattern for Call History using My Line and Sub Line is as follows:

X: Available, N: Not available

	My Line		Sub Line Note 7		
	Answered	Unanswered	Answered		Unanswered
			My station	Other station	
System Call History					
Outgoing call history	N	N	N	N	N
Incoming call history	X	X	X Note 2	X Note 2 Note 3 Note 4 Note 5	X Note 2 Note 3 Note 4 Note 6
Terminal Call History					
Outgoing call history	X	X	N	N	N
Incoming call history	X	X	N	N	N

Note 2: Available if ASYDL, SYS1, Index 678, Bit 7=1.

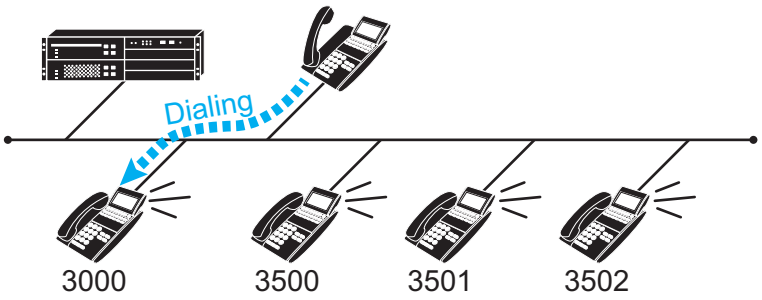
Note 3: Available from FP95-104 V4 Issue 4.0.

Note 4: Available if ASYDL, SYS1, Index 686, Bit 5=1.

Note 5: In the ASFC/ASFCN command, SFI 266 and SFI 267 need to be enabled for the target terminal.

Note 6: In the ASFC/ASFCN command, SFI 266 needs to be enabled for the target terminal.

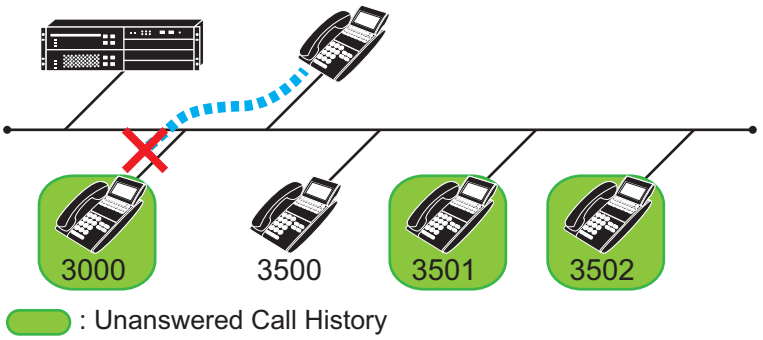
Note 7: The following are examples of how Call History is displayed when there is an incoming call to Station 3000 and the call is answered or unanswered by the My Line or the Sub Line on the terminals with different SFI settings.



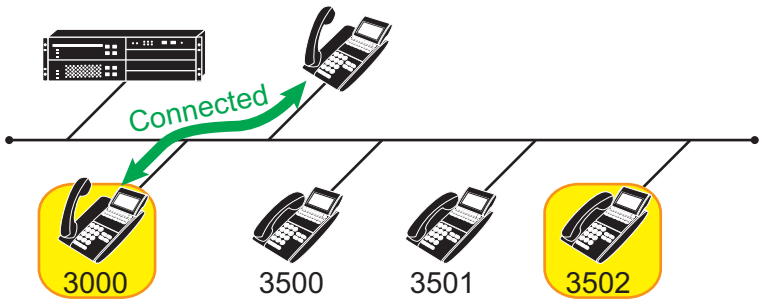
Station Number		ASFC/ASFCN	
My Line	Sub Line	SFI 266	SFI 267
3000	-	Enabled (1) or Disabled (0) *1	Enabled (1) or Disabled (0) *1
3500	3000	Disabled (0)	Enabled (1) or Disabled (0) *2
3501	3000	Enabled (1)	Disabled (0)
3502	3000	Enabled (1)	Enabled (1)

*1: The settings of SFI 266 and SFI 267 have no relation with the operations of My Line.
*2: Because SFI 266 is set to disabled, the settings of SFI 267 become irrelevant.

- Station 3000 does not answer the call:

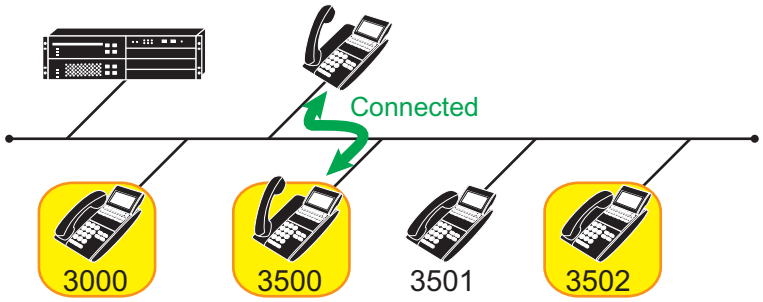


- Station 3000 answers the call:



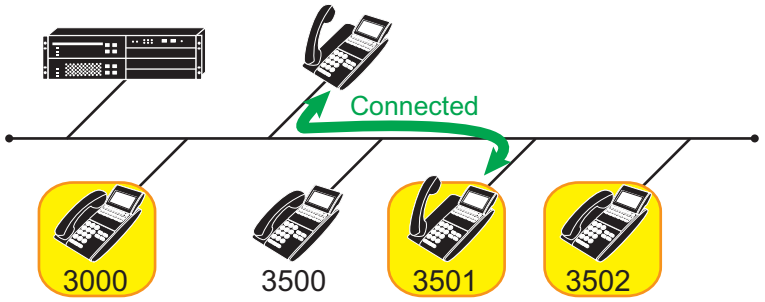
: Answered Call History

- Station 3500 answers the call with Sub Line:



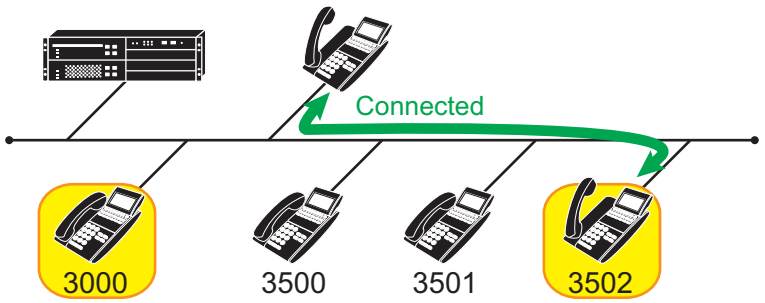
: Answered Call History

- Station 3501 answers the call with Sub Line:



: Answered Call History

- Station 3502 answers the call with Sub Line:



: Answered Call History

Besides that, System Call History can be displayed in case it is used in combination with other features. The features

in which System Call History can be displayed are as follows.

X: Available, -: Not applicable

Feature	System Call History	
	Answered	Unanswered
CALL PICKUP - GROUP [C-7] **Note 8**	X	-
CALL PICKUP - DIRECT [C-30] **Note 8**	X	-
CALL WAITING - TERMINATING [C-12] **Note 9**, **Note 10**	X	X
CALL WAITING - ORIGINATING [C-31] **Note 9**, **Note 10**	X	X
CALL WAITING-TERMINATING-PRI/PRA [C-114/116] **Note 9**, **Note 10**	X	X
EXECUTIVE RIGHT-OF-WAY [E-1] **Note 9**, **Note 10**	X **Note 11**	-
EXECUTIVE RIGHT-OF-WAY - CCIS [E-8] **Note 9**, **Note 10**	X **Note 11**	-

Note 8: The setting ASYDL/ASYDN, SYS1, Index 678 Bit 7=1 is required.

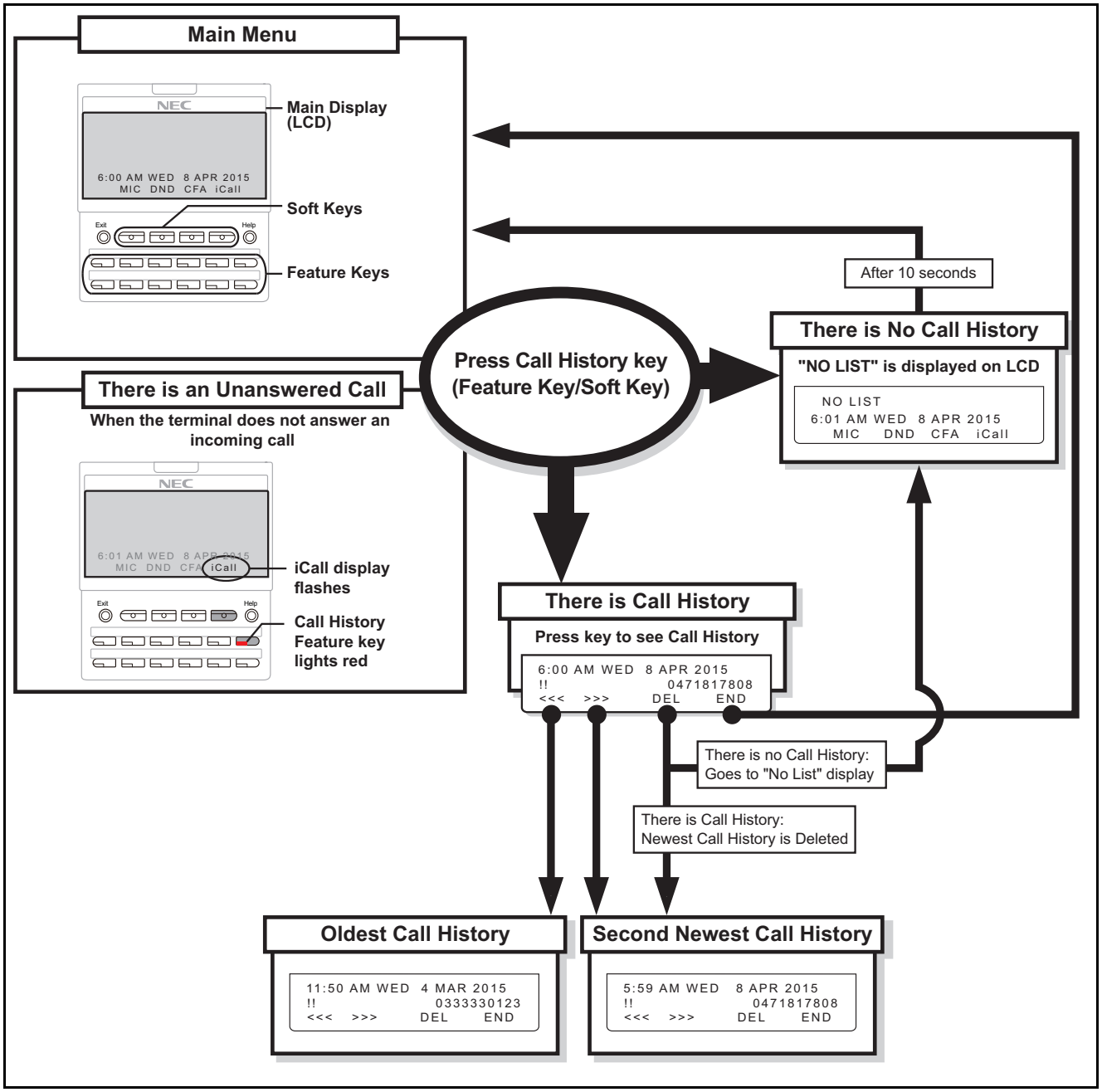
Note 9: Available from FP95-104 V4 Issue 4.0.

Note 10: The setting ASYDL/ASYDN, SYS1, Index 686 Bit 6=1 is required.

Note 11: Call History is only displayed for the overridden station.

Operating Procedure

The following diagram shows the operating procedure for using System Call History. For the operating procedure of Terminal Call History, refer to the User's Guides of DT800/DT700/DT400/DT300 Series.



Service Conditions

<Conditions for System Call History (ASFC SFI 238=1)>

1. Terminal Call History can be used in combination with System Call History through the entire system; however, Terminal Call History cannot be used in combination with System Call History in one terminal. When SFI 248=1 (Terminal Call History is allowed) and SFI 238=1 (Call History is allowed) are assigned, SFI 248=1 takes priority and Terminal Call History is effective.
2. When ASFC/ASFCN SFI 238=1 (Call History is in service) is assigned, outgoing Call History by ASFC/ASFCN SFI 248=1 (Terminal Call History is in service) is not effective. For that reason, use REDIAL [R-60].
3. This feature is available for the following terminals.

   - IP terminal **Note 12**
   - SIP Multiple Line terminal
   - Digital terminal **Note 13**, **Note 14**

   Note 12: This feature is not available on Soft Phone (Soft Client SP350/DtermSP30), since it does not have Soft keys.

   Note 13: Only available for DT400/DT300 series.

   Note 14: The ELC card (CH-16ELCA) is required.

   Note: This feature is not available with the Attendant Console.

4. The setting and terminal conditions for a terminal that uses this feature are as follows:
   a.) To use this feature, assign CALL HISTORY to a Line/Feature key or Soft key.

   Note: On DT830G/DT830/DT820/DT750/DT730G/DT730, incoming call history by this feature can be displayed by selecting [User Menu] - “1. History”. However, parameters under “1. History” menu is not available. (In the case of DT820, this condition is from firmware version 3.0.0.0.)

   b.) Do not assign CALL HISTORY Feature key if SFI = 238 is not acceptable on the terminal. If Call History is assigned to such terminal, Feature key Lamp and Soft key will be controlled but Call History will not be displayed. Key control will be cleared by pressing the CALL HISTORY Feature key.
   c.) Do not assign CALL HISTORY to a Flex key.
   d.) This feature is available only on terminals where the LCD is capable of displaying 24 digits.
   e.) For the terminals except DT Series, if the terminal has Terminal Call History capability for incoming calls, this feature cannot be used.
   f.) Depending on the type of terminal, the Soft key display can be different.

5. This terminal incoming Call History record will be cleared in the following cases.
   a.) When deleting terminal data through PCPro command, all Call History data will be deleted.
   b.) When performing System Initialization or SP Initialization, all Call History data will be deleted.

c.) When the terminal is re-registered or logged out, all Call History data will be deleted. To maintain the Call History list, activate Fixed PH Hunting by assigning System Data (ASYDL, SYS1, Index 824, Bit 0=1).

6. If there is a call to the My Line or Sub Line of a terminal that is displaying the incoming Call History record, the display will be finished and go back to the main menu.
7. Call History will store a maximum of 20 calls.
8. When Call History data exceeds 20 calls, data will be deleted, starting from the oldest data.
9. The following shows the status of Line/Feature key and Soft key display when the CALL HISTORY Feature key is assigned.

Status	Feature Key	Soft Key
Terminal has not answered an incoming call	Lights in red	Flash Display **Note 15**
After Call History is displayed	OFF	Normal Display

Note 15: When an incoming call is unanswered, Soft key maintains the Call History flashing display.

10. When receiving a second call from a number already in Call History, assign System Data to select whether to save data or overwrite the list (SYS1, Index 677, Bit 4).
11. The history of an answered call without Caller ID information is displayed as "-----" at the center of the 3-line LCD, or at the name display area of the 4-line LCD.
12. A Station can display Call History for a line which is being used by another Station.
13. Two options are available for the display format of Call History; three-line format and four-line format. You can specify the format by setting System Data (ASYDL, SYS1, Index 681, Bit 0).

Note: These options are only available for DT830/DT700/DT400/DT300 Series terminals with a 4-line LCD (4 lines for display + 1 line for icon).

14. When receiving a second call from a number already in Call History, assign System Data to select whether to save data or overwrite the list (SYS1, Index 677, Bit 4).
15. DT800/DT700/DT400/DT300 Series can originate a call using the incoming Call History with prefix added. For the operation procedure, refer to the related manuals of DT800/DT700/DT400/DT300 Series.

Note: On Desktop terminals used earlier than DT800/DT700/DT400/DT300 Series, as Call History is not compatible with prefix function, office data programming is required to add access code of incoming route.

a.) Up to 10 digits of number can be added for prefix.
b.) When prefix is added, a maximum of 32 digits of numbers can be sent out and displayed. If the number exceeds 32 digits, the overflowing last digits are deleted.
c.) In the case of incoming call termination after moving to prefix screen, process of prefix is continued during ringing. When the call is answered, process of prefix is canceled and the speech is established. The display moves to the idle state after releasing the call.
d.) The LCD cannot go back to Call History screen after moving to prefix screen.

16. The conditions for an incoming Call History to Sub Line are as follows.
    a.) When ASYDL, SYS1, Index 678, Bit 7=1 is assigned, the record of a call answered on the Sub Line of a station is kept on both stations which the call is placed to and which the call is answered on. For example, when My Line of Station B is accommodated on a Sub Line of Station A, if Station A answers a call placed to Station B by pressing the corresponding Sub Line key, the call record is stored on both Station A and Station B.

        Note: When ASYDL, SYS1, Index 678, Bit 7=0 is assigned, a call record is stored only when a call arrives on My Line. For example, in the above case, the call record is stored on Station B as an unanswered incoming call.

    b.) DSS Console Line Keys (AKYD command (KD=2)) do not support Call History to Sub Line.
    c.) Digital terminal accommodated in ELC card is not supported for unanswered Call History to Sub Line and Call History to Sub Line answered in a different terminal.
    d.) Call History is administrated per terminal. For that reason, when several terminals have registered the same Sub Line, if an unanswered call is checked in a terminal, the call remains unchecked in other terminals.
    e.) In the case of Call History to Sub Line, light and signals to inform of an unanswered call may need up to one minute to be displayed.

17. When this feature is used in combination with other features, the following conditions apply.
    a.) When ASYDL, SYS1, Index 678, Bit 7=1 is assigned, the record of a call answered through Call Pickup is kept on both stations which the call is originally placed to and which the call is answered on. When ASYDL, SYS1, Index 678, Bit 7=0 is assigned, a call answered through Call Pickup is recorded as an unanswered incoming call and kept on the station to which the call is originally placed.
    b.) If there is an incoming call to either Prime Line or My Line engaged in Call Redirect [C-173] operation while displaying Call History, the display will return to the main menu of IP terminal (except for Soft Phone) and SIP Multiple Line terminal.
    c.) When Optional Language Pack Loading [O-42] is used, Call History information for an answered/unanswered incoming call can be arbitrarily defined. A maximum of eight single-byte characters can be set. When this setting is not configured, "!!" is displayed for an unanswered call.
    d.) When Name Display - System [N-28] is in service, you can specify a preference for the type of caller information to be displayed in the Call History.
    When you assign ASYDL, SYS1, Index 678, Bit 6=1, Calling Number is preferentially displayed regardless of the setting of ASYD, SYS1, Index 390, Bit 0 (Caller ID display pattern on the LCD (Top line)). If you assign ASYDL, SYS1, Index 678, Bit 6=0, the priority is determined by the value set for ASYD, SYS1, Index 390, Bit 0.
    If the priority is set to Calling Name and a matching name is not found in the Call History file, the caller's number is displayed instead.
    e.) In the Call History screen, a Caller ID exceeding the number of digits the display can handle is truncated to the length supported by the terminal. You can specify whether to display the first or the last digits of the ID by setting System Data (ASYD, SYS1, Index 241, Bit 6).
    4-line LCD can display maximum 24 digits for Caller ID information type specified by the System Data (ASYDL, SYS1, Index 678, Bit 6 or ASYD, SYS1, Index 390, Bit 0), and 16 digits for otherwise. On the other hand, 3-line LCD supports maximum 16 digits as in previous versions.

f.) When receiving a call through CALL TRANSFER - ALL CALLS [C-11]/CONSULTATION HOLD - ALL CALLS [C-17], the number of the intermediate party (the first terminal to receive the call) will be saved in Call History.
g.) When the terminal is placed under the control of SR-MGC, Call History data will not be carried over.
h.) When the terminal is placed under the control of the Backup Node, Call History data will not be carried over. (In cases such as when using LOCATION DIVERSITY [L-55], SR-MGC, etc.)
i.) When Software Multiple My Line Appearance [S-143] is in service, records of calls to the Same Number Sub Line (a Sub Line assigned to the same My Line number) are kept as long as they are answered. (Unanswered calls are not recorded.) If a call arriving on the Same Number Sub Line is answered through Call Pickup, the record is kept only on the answering station and not on the station to which the call is originally placed.
j.) The call history data of this feature (both System side and Terminal side) are not cleared by Guest Station Call Record Automatic Clear [G-27]. However, when you use DT700 Series with firmware versions earlier than 2.2.0.0, only Terminal Call History data (terminal side) can be cleared (Terminal Call History data cannot be cleared when you use DT700 Series (except DT730G) with firmware 2.1.0.0 or later versions).

<Conditions for Terminal Call History (ASFC SFI 248=1)>

Terminal Call History allows the dialed number to be stored to the terminal as outgoing Call History; the received number to be stored to the terminal as incoming Call History. The conditions for Terminal Call History are as follows.

1. This feature can be used only by the following DT Series terminals.

- DT830G/DT830
- DT430 **Note 16**
- DT750/DT730G/DT730
- DT330 **Note 16**

Note: When SFI 248 (Terminal Call History) setting is changed;
- DT830/DT700 Series requires to request re-registration. (Make-busy and make-busy cancel operation by MBST command is also available.)
- DT400/DT300 Series requires ELC circuit card to perform initialization. (Make-busy and make-busy cancel operation by MBPM/MBST command is also available.)
- ASYDL, SYS1, Index 673, Bit 4=0/1 (inform only Calling Number to call history/inform Calling Number and Calling Name to call history) needs to be assigned.
  Data change does not turn effective without re-registration and circuit card initialization.

Note 16: The PA-16ELCN (SP-3956 16ELCN PROG-A or later) is required.

Outgoing Call History:

1. Regardless of using line, the outgoing Call History is stored to operating terminal.
2. When outgoing Call History data exceeds the following number of calls, data will be deleted, starting from the oldest data.

   DT830/DT700 Series: 50
   DT400/DT300 Series (DT430/DT330): 10

3. Data stored as outgoing Call History are;

   - DT830/DT700 Series:

     The dialed number and the dialed number that completely matches the registered name in the Terminal Phone book database are stored as outgoing Call History. Calling Name notified from the system is not stored.

   - DT400/DT300 Series (DT430/DT330):

     The dialed number is stored as outgoing Call History. Calling Name notified from the system is not stored.

4. Conditions for dialed number stored as outgoing Call History to the terminal are;

   - The number dialed until the call is released from the call origination:

     Such as the dialed number during an ongoing call or the dialed number for the Call Forwarding operation. However, the dialed number after holding a line is not stored.

   - The number dialed until the call is released after an incoming call is answered:

     Such as the dialed number during an ongoing call or the dialed number during the Call Forwarding operation. However, the dialed number after holding a line is not stored.

   Note: Up to 24 digits dialed number is stored. When the number exceeds 24 digits, the first 24 digits are stored.

   Note: The dialed number mentioned above does not include the Secret Code (the default is "*" or "#") displayed when Dialed Number Privacy Mode [D-165] is activated.

5. The outgoing Call History is cleared in the following conditions.

   - "Delete All" function of the terminal is executed by the user.
   - The terminal is turned off caused by a condition such as removed user cable (modular, LAN, etc).
   - The terminal is released from the status terminal is turned off because of a failure.
   - The terminal is turned off temporarily by the HUB reset when the power is supplied by the HUB. [For DT800/DT700 Series]
   - The station data is changed or deleted. [For DT800/DT700 Series] **Note 17**

- The terminal is in logout status. [For DT800/DT700 Series] **Note 17**

Note 17: Call history is cleared for DT730 Series with firmware version prior to 2.2.0.0.

6. Outgoing Call history shows incorrect outgoing time for up to one minute (the cycle of sending the time information) immediately after the initialization of the system or the card ("0/0 00:00" is displayed).

Incoming Call History:

1. Two types of incoming Call History such as answered incoming Call History and unanswered incoming Call History are stored.
2. When incoming Call History data exceeds the following number of calls, data will be deleted, starting from the oldest data.

   DT830/DT700 Series: 50
   DT400/DT300 Series (DT430/DT330): 10

3. Whether only Calling Number or Calling Number and Calling Name can be selected per system as the incoming call history information sent from the system to the terminal.
4. Conditions of Calling Number sent from the system are:

   - The station number is sent from the system when the Calling Party is a station. On the other hand, the Calling Number is sent from the system when the Calling Party is a trunk.
   - Up to 24 digits Calling Number is sent. When the number exceeds 24 digits, the exceeded digits will be deleted by system data setting (ASYD, SYS1, Index 241, Bit 6). In this case, "*" mark which indicates the deleted number is not displayed.
   - When the Calling Number is not existed, "Withhold", "Public Telephone", or "No Notification" (**Note 18**) is stored in Terminal Phone book on DT800/DT700 Series as a reason Calling Number is not existed.
     For DT400/DT300 Series (DT430/DT330), "Withhold" is displayed as incoming call history.

Note 18: "No Notification" means beyond the limits to display.

5. When the destination is a station, or Name Display - CCIS [N-37] applies to a station, registered name assigned by Name Display feature (ANDD) is stored. When the destination is a trunk, registered name assigned by Caller ID feature (ACID) is stored.
6. The calling name of the number completely matches the registered name in the Phone book is stored as the incoming call history.
7. When an incoming call is answered by "Terminal B" in which My Line of "Terminal A" is accommodated, incoming call history is stored into both "Terminal A" and "Terminal B".
8. When a call is answered by CALL PICKUP - GROUP [C-7] or CALL PICKUP - DIRECT [C-30], incoming call history is stored into the terminal which is accommodated the call terminating My Line. Also the incoming call history is also stored into the answered terminal.
9. When a call is answered by Direct Inward Dialing (DID) [D-8], incoming call history will not be stored.
10. When a forwarded call is terminated, Calling Number stored as incoming Call History is as follows.

    - Call Transfer - All Calls [C-11]:
      The number of Intermediate station (the first terminal to receive the call)

- Blind Transfer To Attendant [B-17]/Blind Transfer To Station [B-18]:
  The number of originally calling station

11. The incoming Call History is cleared in the following conditions.
    - "Delete All" function of the terminal is executed by the user.
    - The terminal is turned off caused by a condition such as removed modular cable.
    - The terminal is released from the status terminal is turned off because of a failure.
    - The terminal is turned off temporarily by the HUB reset when the power is supplied by the HUB. [For DT830/DT700 Series]
    - The station data is changed or deleted. [For DT830/DT700 Series] **Note 19**
    - The terminal is in logout status. [For DT830/DT700 Series] **Note 19**

    Note 19: Call history is cleared for DT730 Series with firmware version prior to 2.2.0.0.

12. Because the missed call notification icon of the Bluetooth Handset of Cordless Terminal (DT430/DT330 (12BT)) is synchronized with the call history of the Bluetooth Cradle, browse the incoming call history on Bluetooth Cradle to turn off the icon.
13. While the Bluetooth Cradle of the Cordless Terminal (DT430/DT330 (12BT)) displays Incoming/ Outgoing call history, Dynamic Dial Pad cannot be executed from the Bluetooth Handset.
14. Bluetooth Cradle of the Cordless Terminal (DT430/DT330 (12BT)) can display only in English and Japanese.

For DT800/DT700 Series only:

1. In the case of a call origination using Phone book, display of Phone book data is exited when dialing the number is finished. Then the LCD changes over to the display informed from the system automatically. During a call termination or answering a call, the LCD keeps displaying Phone book data.
2. When the Calling Station performs operations such as pressing the SPEAKER key or Headset key, or going off-hook while displaying Phone book data, the call is originated toward the Phone book data on display. If a call terminates Prime Line of the station, operations such as pressing the SPEAKER key or Headset key, or going off-hook are processed as answering call operation. Call origination toward the Phone book data on display is not available.
3. For the prefix function, refer to the related manuals of DT800/DT700 Series.
4. On the terminal's configuration menu, the parameters under [History] menu is not available when this [C-195] Call History (ASFC SFI: 238) is in service.
5. Even though Terminal Call History is set as out of service (ASFC SFI: 248=0), the parameters under [History] menu will be displayed on the terminal's configuration menu regardless of SFI: 248 data setting.
6. DT830/DT700 Series user has an option between Call History and Terminal Call History according to the setting of the ASFC/ASFCN command (SFI 238=1: Call History, SFI 248=1: Terminal Call History).
7. Phonebook information can be displayed in 4-line with a DT800/DT700 series terminal that has 4-line (except the line for icon display) LCD. By displaying Phonebook information in 4-line, you can see the name and the number in the search result display or the deletion confirmation display, whereas in 3-line, you can only see the name.

If you wish to display Phonebook in 3-line as before, assign the system data (ASYDL, SYS1, Index 681, Bit 1=1).

Note: The search result with 4-line display is only available for Local Phonebook and System Phonebook. Deletion confirmation with 4-line display is only available for Local Phonebook.

<Conditions Specific to combination of Call History and Phonebook>

The following shows the conditions when DT800/DT700/DT400/DT300 Series are used.

X: Available, C: Conditionally available, N: Not available

Kind of Call History	DT800/DT700 Series					DT400/DT300 Series	
	DT830G/ DT830	DT820	DT750	DT730G/ DT730	DT710	DT430/ DT330	DT410/ DT310
CALL HISTORY [C-195]							
System Call History	X	C Note 21	X	X	C Note 20	X	C Note 20
Terminal Call History Note 22	X	N	X	X	N	X	N
PHONEBOOK [P-83]							
Local Phonebook	X	X	X	X	C Note 20	X	C Note 20
System Phonebook	X	X	X	X	C Note 20	X	C Note 20
Terminal Phonebook	X	N	X	X	N	N	N
Corporate Phonebook	X	N	X	X	N	N	N

Note 20: Only 6-button type terminal is supported.

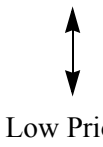
Note 21: From firmware version 3.0.0.0, incoming call history by this feature can be displayed by selecting [User Menu] - “1. History”. However, parameters under “1. History” menu are not available.

Note 22: Terminal Call History is targeted for DT Series only. When DT Series operate in Dterm Series *i* mode, SFI 248 (Terminal Call History) settings take no effect.

1. For DT800/DT700 Series, you can add a prefix before the displayed number when originating a call by Terminal Call History. For DT400/DT300 Series, prefix function is not supported.

   Note: For using prefix function, register the prefix with user setting on the terminal beforehand. For details, refer to the related manuals of DT800/DT700 Series.

2. The priority order to display the name on incoming call termination and call history is as follows.



- Registered name on the selected phone book (Local Phone book/ Terminal Phone book)
- Registered name on System Phone book
- Registered name assigned by Name Display feature (ANDD) and Caller ID feature (ACID)

Note: In the following cases, contact names are not displayed in the above priority order.
- The combination of Terminal Phone book and Call History is used:
- Contact name of priority order I is not displayed. Contact name of priority order II or priority order III is displayed.
- The combination of Local Phone book and Terminal Call History when ASYDL, SYS1, Index 673, Bit 4=0 is assigned (inform only Calling Number to call history):
- Contact name is not displayed as the call history.
- The combination of Terminal Phone book and Terminal Call History when ASYDL, SYS1, Index 673, Bit 4=0 is assigned (inform only Calling Number to call history):
- Contact name of the priority order II and III are not displayed as the call history.

3. Phone book information can be registered from call history generated by Call History [C-195] to Local Phone book on DT830/DT700 Series. For more information, refer to Phonebook [P-83]. Also Phone book information can be registered from call history generated by Terminal Call History to Terminal Phone book on DT830/DT700 Series. For more information, refer to the related manuals of call DT830/DT700 Series.
4. When either Call History or Terminal Call History is selected by **Call History** → **Incoming** menu displayed by pressing Menu button, the selected call history can be displayed.
5. When you disable Terminal Call History, the missed call icon must be cleared before changing the office data.

Programming

Note: When you want to change the display information on an answered/unanswered incoming call, the data setting for Optional Language Pack Loading [O-42] is required before setting the following data. For details, see Optional Language Pack Loading [O-42].

< System Call History >

STEP 1: **ASYD** - System Data
SYS1, Index 241, Bit 6
When Caller ID digits exceed the number of digits able to be displayed by terminal.
0: Display from Caller ID's first digit to display limit
1: Display from Caller ID's last digit to display limit

SYS1, Index 390, Bit 0
Specify which Caller ID data is given priority for display
0/1: Name Display is given priority/Number Display is given priority

STEP 2: **ASYDL** - System Data for LDM
SYS1, Index 824
Assign Fixed PH Hunting method
Bit 0 = 0/1: No/Yes

STEP 3: **ASYDL/ASYDN** - System Data for LDM/NDM
SYS1, Index 677, Bit 4
When receiving an incoming call from a station whose Call History data is already on the list.
0/1 = Save the Call History data on the list/Overwrite the call history data on the list

SYS1, Index 678, Bit 6 **Note 23**
Preference for the type of caller information to be displayed in the Call History.
0/1 = Follows the setting of ASYD, System Data 1, Index 390, Bit 0/Displays Calling Number

SYS1, Index 678, Bit 7 **Note 24**, **Note 25**
Sub Line Call History (answered calls) and Call History used in combination with Call Pickup features.
0/1 =Kept only on the original destination station (My Line)/Kept on both original destination station (My Line) and answering station

SYS1, Index 681, Bit 0
Number of lines displayed for Call History.
0/1 = 4 lines/3 lines

SYS1, Index 686, Bit 5 **Note 26**
Sub Line Call History (unanswered calls and calls answered in other station).
0/1 = Out of service/In service

SYS1, Index 686, Bit 6 **Note 26**
Call History used in combination with Call Waiting features and Executive Right-of-Way features.
0/1 = Out of service/In service

Note 23: If you assign ASYDL, SYS1, Index 678, Bit 6=1, Calling Number of call history is preferentially displayed regardless of the setting of ASYD, SYS1, Index 390, Bit 0 (Caller ID display pattern on the LCD (Top line)). Otherwise, the display follows the setting of ASYD, SYS1, Index 390, Bit 0.

Note 24: When ASYDL, SYS1, Index 678, Bit 7=1 is assigned, the records of the following calls are kept on both stations which the call is originally placed to and which the call is answered on:

- Calls answered through Call Pickup feature
- Calls answered on the Sub Line of a station (where My Line of the original destination station is accommodated on a Sub Line

Note 25: If Sub Line Call History (unanswered calls and calls answered in other station) is used, this Bit needs to be set to 1.

Note 26: Available from FP95-114 V4 Issue 4.0.

STEP 4: **ASFC** - Assign a Service Feature Restriction Class that allows the following

SFI 238 (RES=1) to the station using Call History

SFI 266 (RES=1) to the station using Call History for unanswered Sub Line calls **Note 27**, **Note 28**

SFI 267 (RES=1) to the station using Call History for Sub Line calls answered in another station **Note 27**, **Note 28**

Note: When SFI 238=1 (Call History is in service) is assigned, assign also SFI 248=0 (Terminal Call History is out of service).

Note: When SFI 248=0 (Terminal Call History is out of service) is assigned, outgoing call history on Terminal side will be also out of service.
On DT400/DT300 Series (DT430/DT330), **[Menu]** → **Call History** and **[Shortcut Menu]** → **1 Missed Calls** will not be operated.

Note: IP terminal/SIP Multiple Line terminal requires to request re-registration (Make-busy and make-busy cancel operation by MBST command is also available).
Digital terminal requires ELC circuit card to perform initialization (Make-busy and make-busy cancel operation by MBPM/MBST command is also available).

Note 27: Available from FP95-114 V4 Issue 4.0.

Note 28: The relation between SFI 266 and SFI 267 is as follows:

Operation	SFI 266= Disabled	SFI 266= Enabled	
		SFI 267= Disabled	SFI 267= Enabled
Call History for unanswered Sub Line calls	Call History is not kept	Call History is kept	Call History is kept
Call History for Sub Line calls answered in another station	Call History is not kept	Call History is not kept	Call History is kept

STEP 5: **ADSL/ADSLN**- Assign a programmable Soft key as Call History to the following parameter:
SKP: 0-15 (Soft Key Pattern)
SN: 0 = idle
FKY: 178 = Call History

STEP 6: **ADKS** - Assign each Soft Key Pattern data to the desired IP terminals (except for Soft Phone) and SIP Multiple Line terminals.
TN: Tenant Number
STN: Station Number
SKP: 1-15 (Soft Key Pattern assigned by ADSL)
LKP: Assign the Key Pattern of the IP terminal (except for Soft Phone) and SIP Multiple Line terminal.
1 = 16 Line/Feature Buttons + DSS keys
2 = 24 Line/Feature Buttons + DSS keys
3 = All Line/Feature Buttons
PG: Specify the Page Scroll key allocation
0 = Page change key appears on a software key (>>>)
1 = Page change key disappears on a software key

STEP 7: **AKYD** - Assign FKY=178 to a Feature key.

STEP 8: **ACNP -** Assign a Calling Number Pattern.
In Call History, when saving a number with an access code attached, assign the caller's Calling Number Pattern.
IC/OG: Incoming/Outgoing Call
IC: Incoming Call
OG: Outgoing Call
RT: External Route Number [1-255]
CNP: Calling Number Pattern [1-255]

STEP 9: **ACND** - Assign a Calling Number Data.
In Call History, when saving a number with an access code attached, assign the caller's Calling Number Data (Access Code).
CNP: Calling Number Pattern [1-255]
SKIP: Number of Skip Digits [0-24]
ADD: Number of Digits of the Additional Number [0-24]
DC: Additional Digit Information [Maximum 24 digits]

< Terminal Call History >

STEP 1: **ASYD**
SYS1, Index 241, Bit 6
When Caller ID digits exceed the number of digits able to be displayed by terminal.
0: Display from Caller ID's first digit to display limit
1: Display from Caller ID's last digit to display limit

STEP 2: **ASYDL**
SYS1, Index 673, Bit 4
Incoming calling history display pattern on the LCD.
0/1 = Calling Name only/Calling Name and Calling Number

STEP 3: **ASFC** - Assign a Service Feature Restriction Class that allows the following:
SFI 248 (RES=1) to the station using Terminal Call History

Note: To activate Terminal Call History on Digital terminal, assign also SFI 247=0 (Terminal Phone book is out of service).
If SFI 247=1 (Terminal Phone book is in service) is assigned, System Phone book, which is controlled by the system, will be activated when **[Menu]** → **2 Directory** is selected or Right-Cursor key is pressed on the DT400/DT300 Series (DT430/DT330) in idle state.

Note: Re-registration of terminal or circuit card initialization are required after changing data assignment.

STEP 4: **ACNP -** Assign a Calling Number Pattern.
In Call History, when saving a number with an access code attached, assign the caller's Calling Number Pattern.
IC/OG: Incoming/Outgoing Call
IC: Incoming Call
OG: Outgoing Call
RT: External Route Number [1-255]
CNP: Calling Number Pattern [1-255]

STEP 5: **ACND** - Assign a Calling Number Data.
In Call History, when saving a number with an access code attached, assign the caller's Calling Number Data (Access Code).
CNP: Calling Number Pattern [1-255]
SKIP: Number of Skip Digits [0-24]
ADD: Number of Digits of the Additional Number [0-24]
DC: Additional Digit Information [Maximum 24 digits]

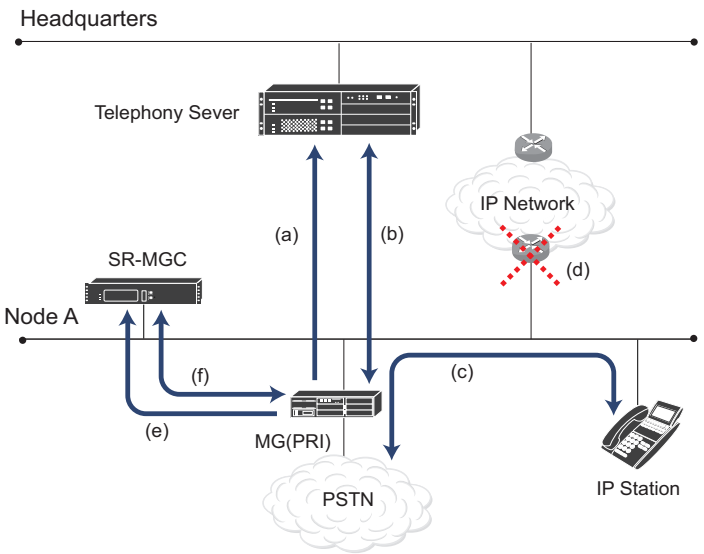
C-203 Call Protected Re-Registration

General Description

By this feature, while talking via the MG(PRI), if an ongoing call is disconnected by a network failure between the MG(PRI) and the Telephony Server, the MG(PRI) with the retained ongoing call can be re-registered to the SR-MGC. Therefore, all the channels of the MG(PRI) except the channel retaining the ongoing call can be used under the SR-MGC.

If a network failure occurs, the MG(PRI) with the retained ongoing call re-registers to the SR-MGC as follows.

Note: This feature is different from the MG Based Call Retention feature, details of which are found in "14.9 MG Based Call Retention" in Chapter 2.



(a) When the MG(PRI) is activated, it registers to the Telephony Server.

(b) Health Check between the Telephony Server and MG(PRI) starts right after the registration.

(c) A call is originated from the IP station via the MG(PRI) and the call is answered. (Example: channel 0 is used and becomes busy state.)

(d) A network failure occurred between the Telephony Server and MG(PRI) and a Health Check time-out is detected.

(e) The MG(PRI) re-registers to the SR-MGC. In this case, channel 0 in busy state is not reset and the ongoing call can be retained. Also, all the channels except channel 0 are available for a call origination/termination after the re-registration.

(f) SR-MGC periodically checks the status of channel 0 retaining the ongoing call. When the ongoing call is disconnected, the SR-MGC resets channel 0. Channel 0 becomes available for a call origination/termination again.

Operating Procedure

No operation is required.

Service Conditions

1. This feature supports the following MGs.
   - MG(PRI) (1.5M) [SCA-24PRIA/MG-24PRIA] (SP-3884 MG PRI PROG-A Issue 10 or later)
   - MG(PRI) (2M) [SCA-30PRIA/MG-30PRIA] (SP-3884 MG PRI PROG-A Issue 10 or later)
   - UG50 (PRT 1.5M) [Proprietary Protocol]
   - UG50 (PRT 2M) [Proprietary Protocol]
   - UG50-24PRIA [Proprietary Protocol]
   - UG50-30PRIA [Proprietary Protocol]

   Note: This feature supports proprietary protocol only. SIP is not supported.
2. This feature is available for SR-MGC.
3. This feature is available for FCCS Cluster. The ongoing call can be retained when moving to backup node as well.
4. The software version of the Telephony Server and office data assigned by AMGIL command must be common for all devices (all nodes). If there is a device for which this feature is not valid, the MG(PRI) will not register to the device when a network failure occurred.
5. Use RLMG command to disconnect forcefully the channel retaining the ongoing call only. For other channel, use RLST command.
6. Use AMGIL command to reset the MG(PRI) supporting this feature. When resetting, all the ongoing call will be disconnected even if this feature is valid.
7. There is no limitation of time to retain the ongoing call. The ongoing call is disconnected in the following cases.
   - When RTP packet stopped in such the case that the equipment of connected party is rebooted.
   - When the opposite party released the call.
   - When RLMG command is executed from PCPro.
   - When a reset is executed by the AMGIL command.

   Note: For the detail of call retention function that each terminals have individually, refer to System Description - CHAPTER 3 IP Equipment - Operation of Each IP Equipment/ Terminal When a Fault Occurs During a Conversation.

   Note: The time-out value to determine that the ongoing call is disconnected is assigned by the Configuration Command (RTP_STOP_DETECT).

8. When the network failure between the Telephony Server and MG(PRI) is recovered, the control of MG(PRI) changes over to the Telephony Server from SR-MGC. In this case, the MG(PRI) with the retained ongoing call will re-register to the Telephony Server.
9. For the changeover from SR-MGC, assign Automatic Changeover (Forceful Switchover is not used). Otherwise, the ongoing call cannot be retained.
10. The retained ongoing call can be retained until it is released, regardless of the number of changeovers to SR-MGC/Telephony Server.
11. Use RLMG command on the registering device to check which channel is retaining the ongoing call.
12. When using SPTS command on the registering device to check the channel retaining the ongoing call, the channel state is make-busy.
13. When the channel retaining the ongoing call remains seized although the terminal using the channel is released, the channel can be released forcefully by RLMG command.
14. There is no limitation for the number of channels that can retain an ongoing call. An ongoing call can be retained although all the channels are busy. However in this case, call origination/termination from the telephony server/SR-MGC of registration destination is not available until any channel is released.
15. Billing information will be output from the device to which the MG(PRI) had registered just before a network failure occurred. At this time, the billing information is output at the instant of the network failure detection, assuming the ongoing call is released. Therefore, the time length of talk will be different from the actual time length.
16. If ARTD command is used for the PAD control of MG(PRI) while operating, the MG(PRI) needs to be turned off and on or needs to be reset by using AMGIL command to apply the changed PAD value.
17. There is no limitation for the number of MG(PRI)s that can use this feature at the same time.
18. This feature will be available even if MG(PRI) uses a DHCP server.
19. A call origination may fail if the call reference value used by the channel retaining the ongoing call and the call reference value used when making a new call are different.

    Note: In this case, make a new call again. When making a new call again, a new call reference value is used.

20. This feature cannot retain the ongoing call if the status of the call is changed.
21. A call cannot be retained when ringing or held.
22. A call cannot be retained when rebooting from the local console of MG(PRI).
23. A call cannot be retained in the following cases.

    - If a patrol function is inactivated by using RLST command while talking.
    - If disconnected by the trunk ineffective hold detect of routine diagnosis.

Programming

1. To use Call Protected Re-Registration
   Assign the office data for the MG(PRI). The procedure here describes just the data setting to use this feature. Refer to Peripheral Equipment Description (IP Devices) for the configuration of MG(PRI) or Peripheral Equipment Description (UG50) for the following UG50 PRT cards:

   - UG50 (PRT 1.5M) [Proprietary Protocol]
   - UG50 (PRT 2M) [Proprietary Protocol]

- UG50-24PRIA [Proprietary Protocol]
- UG50-30PRIA [Proprietary Protocol]

AMGIL - Select the checkbox of "CALL HOLD (MG-PRI Call Hold Service)" to enable CALL PROTECTED RE-REGISTRATION [C-203].

Note: When this feature is used on the already registered MG(PRI) or deleted, the MG(PRI) will be reset and the ongoing call will be disconnected.

Note: When this feature is enabled, it overrides MG Based Call Retention configured with the maintenance command of MG(PRI).

[Call Control Takeover by SR-MGC]

ASYDL - SYS1
Index 847, Bit 0 to Bit 3 (Timer Counter for forceful switchover) = F (Forceful switchover is not executed while the active calls remain.)
Index 852, Bit 5 (IP terminal registration changeover from SR-MGC to MGC) = 1 (Automatic Changeover)

2. To reset MG(PRI).

AMGIL - Enter the MG-ID of the corresponding MG(PRI) to reset.

MG-ID: ID of MG [00-FF (Hex)]
Assign the MAC Address of MG (Media Gateway).

FUNCTION
Reset: To reset the MG-PRI

Note: When resetting, the ongoing call cannot be retained.

3. When resetting the channel retaining an ongoing call.

RLMG - Resets the channel retaining an ongoing call.

LENS: Line Equipment Numbers of MG(PRI) [six digits]
Allocate the head channel of HW (Highway Block) to which Virtual Speech/Signal Channel is assigned.

Channel Number to reset: Forceful Switchover Channel Number (0-31)
Assign the channel number to switch over forcefully.

Note: Channel 0 indicates the LENS to which the corresponding MG(PRI) is registered by AMGIL command.

For example, in the case of MG(PRI) assigned to "LENS: 022120" by AMGIL command.

The relation of the channel number and LENS is as follows.

CH0: 022120
CH1: 022121
:
CH7: 022127
CH8: 022130
:
:
CH31: 022157

4. For displaying the information of the Media Gateway.

DIMG - Display Information of Media Gateway

Input Data
ISDN Trunk (PRI) Information: Information of MG(PRI) belonging to the Telephony Server.

Note: The route information here is the information of virtual signaling channel (Dch) assigned by AMGIL command.

C-204 Caller ID Display - Sub Line

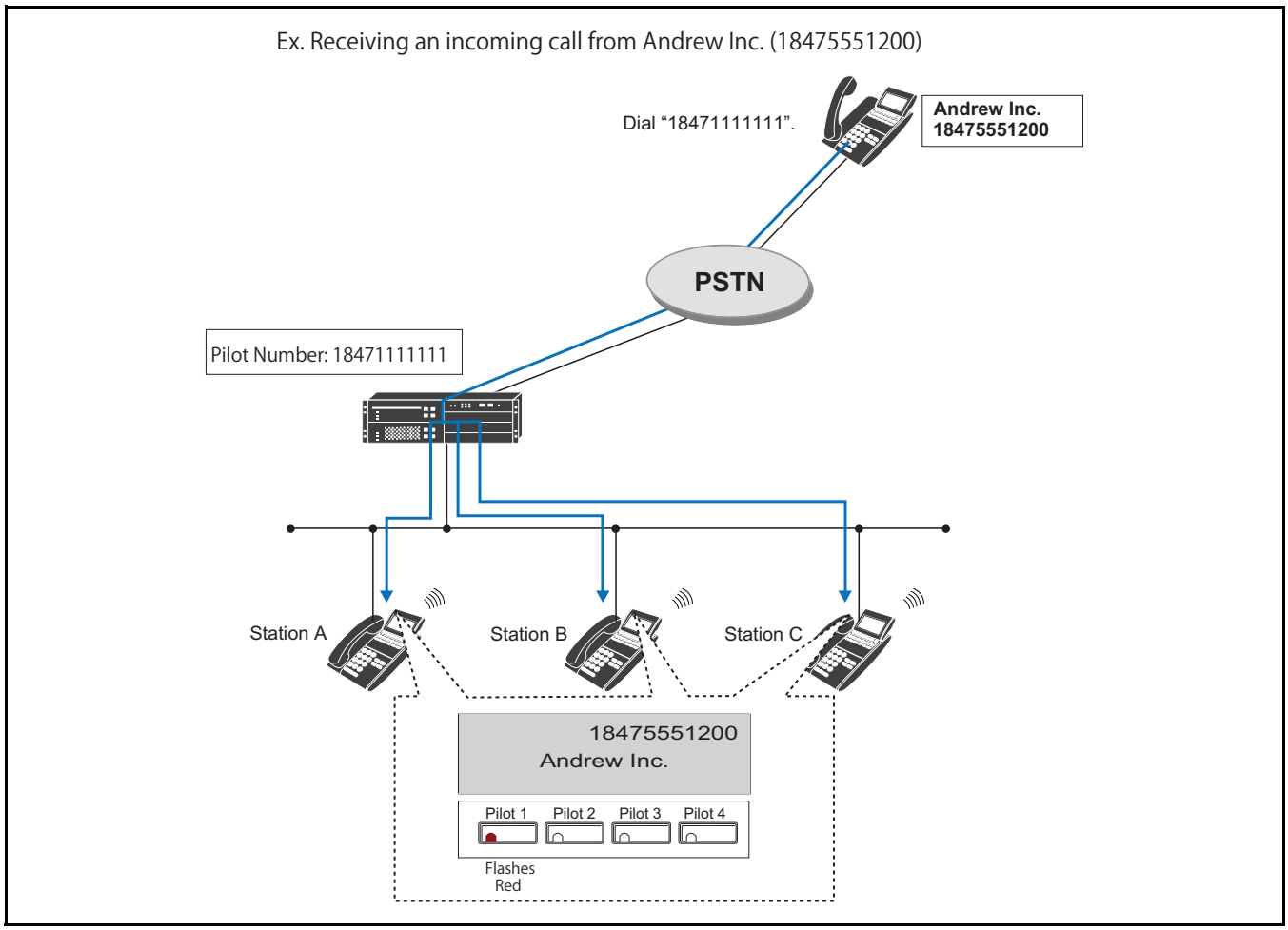
General Description

This feature displays Caller ID (calling party number and the registered name) on the LCD of the Desktop terminal when Sub Line receives an incoming call. A user can answer the incoming call which is currently displayed on the LCD.

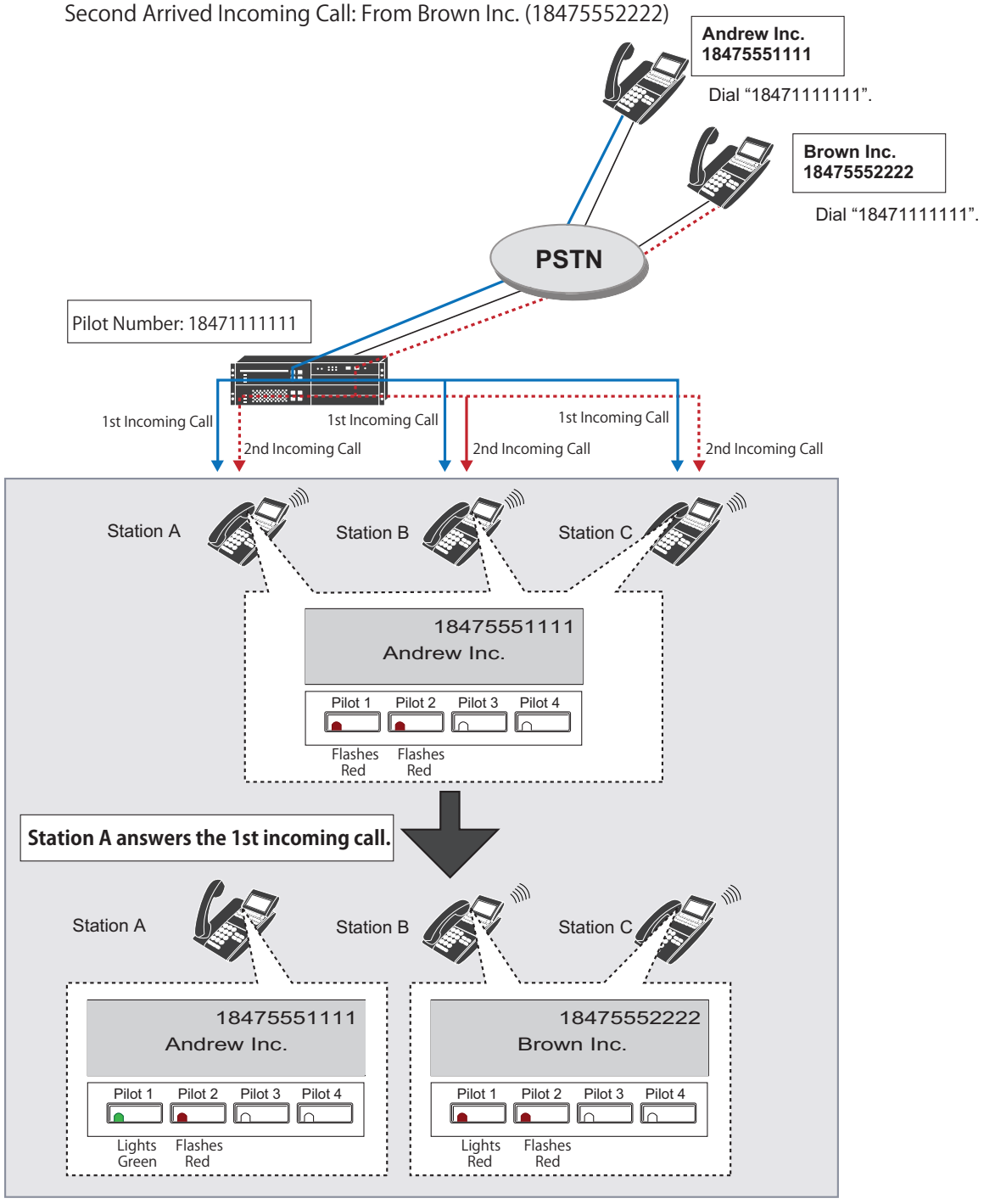
Note: After the data of Caller ID Display - Sub Line (ASYDL/ASYDN, SYS1, Index 1248, Bit 0) and Mixed Display of Caller ID Display-Subline (ASYDL/ASYDN, SYS1, Index 1248, Bit 4) are changed, a system needs to be initialized. For terminals connected to an ELC card, the card needs to be initialized.

Note: After changing data for the LN PRE parameter of AKYD command, initialize the ELC card.

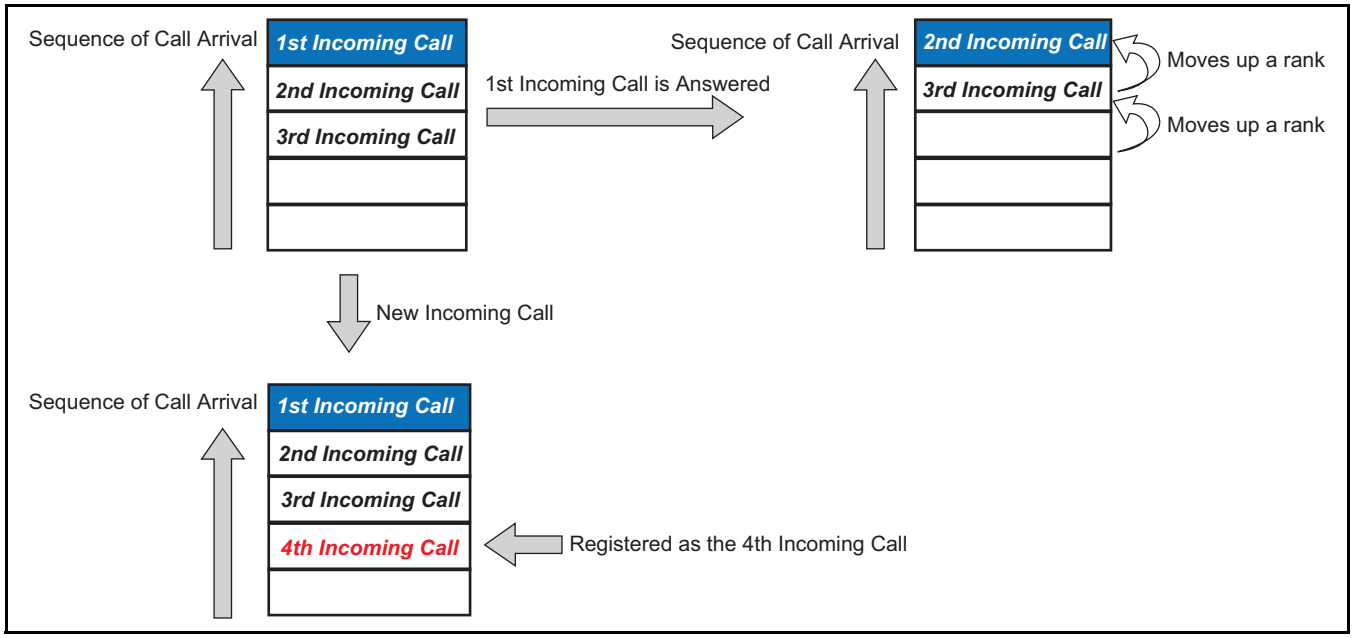
Note: In this feature description, it is assumed that Prime Line and My Line are set as the same number, and both are referred to as Prime Line. For more information about multiple lines, refer to 5.1 Multiple Lines of CHAPTER 2.



Ex. First Arrived Incoming Call: From Andrew Inc. (18475551111)
Second Arrived Incoming Call: From Brown Inc. (18475552222)

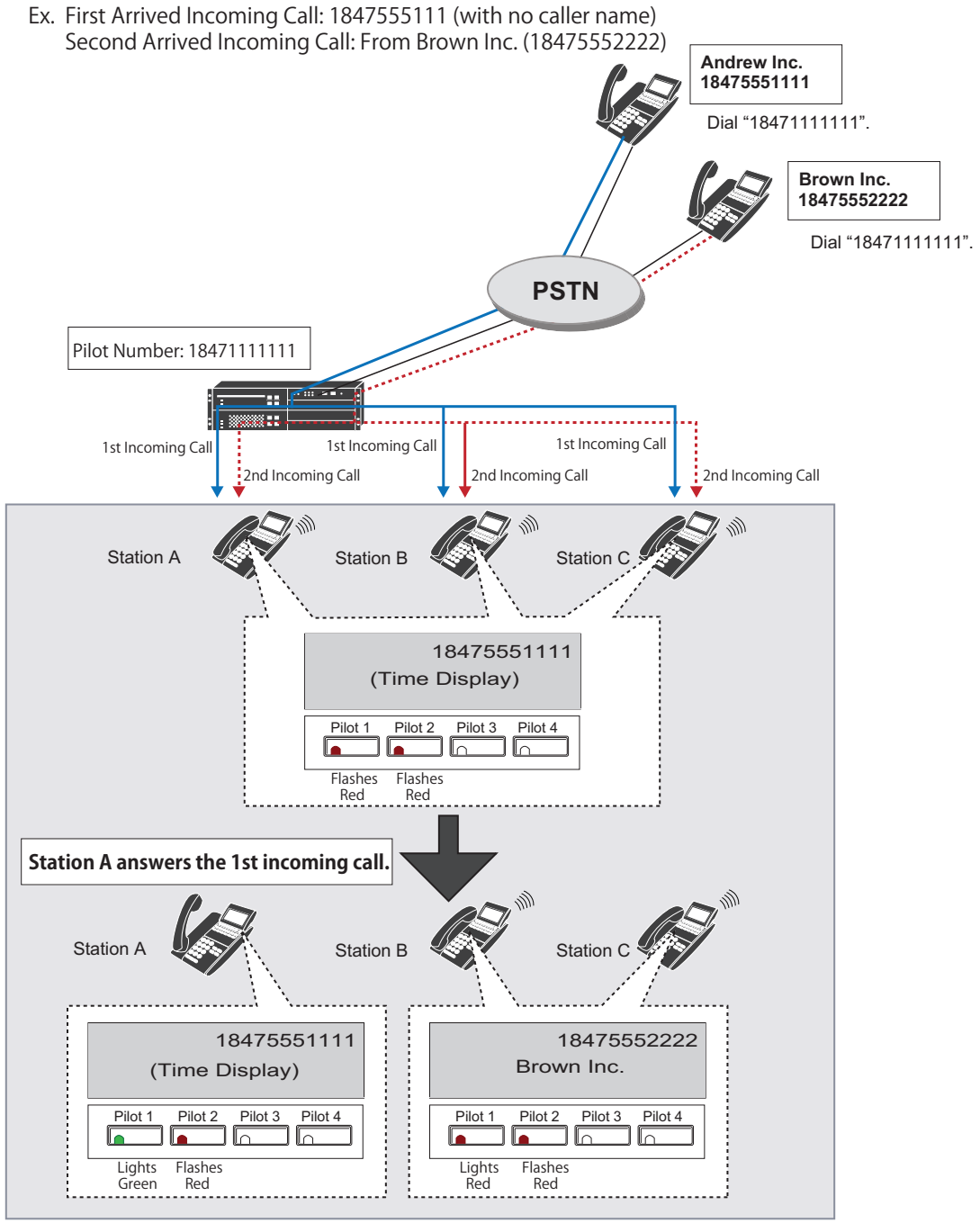


Note: When multiple Sub Lines receive incoming calls respectively, Caller ID of the first arrived call is displayed. When the first arrived call is answered, Caller ID of the second arrived call is displayed on the other stations' LCDs.



When multiple lines receive calls, an LCD shows the information of the Caller ID and the caller name that are corresponding to each other.

When the first incoming call does not have the caller name but the second incoming call has the caller name, Station A displays only the Caller ID of the first incoming call that is being answered.



Operating Procedure

None.

Service Conditions

<Important Notices>

1. The information to be displayed for incoming calls on Sub Line is the same as for calls on Prime Line.

   Note: When the terminal type or setting of the language of Prime Line and Sub Line differs, the information which is applied in each terminal condition is displayed.

2. When name is not registered, current time is displayed.

   Top line : Calling Party Number

   Middle Line : Calling Party Name

   Bottom line : Not used in this feature

3. A maximum of 40 Caller IDs can be accumulated per telephone (Prime Line is included, therefore, 39 lines are available as Sub Line). LCD displays Caller IDs for the lines which are registered to KYNs 1-40 of AKYD command.
4. When Station A accommodates Prime Line of Station B using Phonebook [P-83] - Local Phonebook as its Sub Line, and Prime Line of Station B receives an incoming call, Local Phonebook data of Station B is displayed on the LCD of Station A. In such a case, do not use Phonebook [P-83] - Local Phonebook at Station B.

<Interactions with Other Features>

1. When this feature is used in conjunction with Multiple Line Operation, a station number can be shared among up to 20 SIP Multiple Line telephones (set as their Sub Lines). If an incoming call is terminated to a station whose number is shared among more than 20 SIP Multiple Line telephones, the traffic of IP packets sent from SIP server to the SIP telephone is momentarily increased. This restriction does not apply to Legacy terminals such as DT300 Series.
2. When you use Call History [C-195], Call History data for Prime Line is stored regardless of whether or not the call has been answered; however, the data for Sub Line cannot be stored (displayed on the LCD but not stored).
3. This feature is not available for Line keys on the DSS console (ADM mode). Even when the Line key on the console receives an incoming call, LCD does not display the Caller ID.
4. When you use Phonebook [P-83], LCD displays name registered to Phonebook. Also, the LCD displays numbers notified from the system.
5. Intermediate Station Number Display - D [I-10D] is still available when you use this feature.
6. For an incoming call from a station with Telephone Number set, LCD on the called station displays the Telephone Number as Caller ID.
7. Enable 24-digit format by assigning ASYDL, SYS1, Index 672, Bit 1=1.
8. LCD does not display Caller ID when CLI Suppression–CCIS is allowed (ASFC, SFI 242 = 1).
9. This feature is not available in OAI/ACD system.
10. This feature is available for the Hotel system.
11. When your system enables Line Pre-selection-D [L-9D], this feature executes as follows:
    - LCD displays Caller ID with higher priority when multiple Sub Lines receive incoming calls through this feature.
    - LCD displays Caller ID for a certain period of time by pressing a Line key when multiple Sub Lines receive incoming calls. You can answer the displayed call.

12. When the conversation of the first call ends, in case a station receives the second call during the Elapsed Time Display for the first call, Calling Party Number (for the second call) will be displayed after the Elapsed Time Display ends.

<Hardware/Software>

1. One of the ELC cards listed below is required for this feature.

ELC Card	Firmware Requirements
CH-16ELCA	SP-4035 16ELCA PROG-A Version 1 or later.
PA-16ELCN **Note 1**	SP-3981 16ELCN PROG-C Version 1 or later.
PA-16ELCN-A **Note 1**	SP-4013 16ELCN-A PROG-A Version 1 or later.

Note: Use SP-4034 16ELCN-A PROG-B Issue 1A or later to stop showing information of the call that is terminated to another line while you are talking on or placing a call.

Note 1: This feature does not work except when the ELC card switch setting (SW11-5, 6) is set to standard (both SW11-5, 6 are turned on). For more details about the ELC card switch setting, refer to each ELC card description in CHAPTER 4 Cards in 8U-PIR of Circuit Card Description.

2. DtermIP INASET do not support this feature.

Note: SIP/IP terminal starts to accumulate Caller IDs for incoming calls after the telephone boots up. Once you reboot the telephone, the accumulated Caller IDs are cleared. However, when PH Fixed Hunting is enabled, the accumulated Caller ID information is retained even if you reboot the telephone.

Note: The following system data determines PH Fixed Hunting: ASYDL, SYS1, Index 824, Bit 0.

Note: When the following command data is assigned or changed, Caller ID information is cleared by a restart of the telephone even if PH Fixed Hunting is used in the system.

- ADKS
- AKYD
- AKYD_T
- ASCL
- ASCL_T
- ASTN
- MBLE
- MBST
- MBST_T

3. (Conditions for PA-16ELCJ-B/PA-16ELCN/PA-16ELCN-A) This feature only works when the Terminal Mode Setting of Switch 11 (SW11 5-6) on ELC circuit card is set to ON/ON (Standard Setting). For details on the switch settings of the circuit cards, see sections related to ELC cards in the Circuit Card Description.

4. When calls concurrently arrive at both of My Line and Sub Line of a SIP Handler Controlled SIP terminal in Multi Line Mode, the SIP Handler Controlled SIP terminal receives the calling station information such as name and number from both calling stations. The calling station information to be displayed depends on the specification of the SIP Handler Controlled SIP terminal.

<Interactions with Networks>

None.

<Conditions Specific to This Feature>

1. When a station receives multiple incoming calls, and the first arrived call is answered by another station or abandoned, the displayed Caller ID changes to that of the second incoming call.
2. When a station receives multiple incoming calls, LCD displays Caller IDs as follows:
   - When Prime Line receives an incoming call, LCD displays Caller ID for Prime Line.
   - When Prime Line does not receive an incoming call, LCD displays Caller IDs in order of arrival.
3. When a station becomes idle (from busy status) while receiving multiple incoming calls, LCD displays Caller ID of the next priority call.
4. When you answer an incoming call, LCD displays Caller ID of the answered call. Even though Prime Line receives another call, the LCD displays Caller ID of the answered call.
5. When multiple Sub Lines receive incoming calls, you can answer the displayed call by pressing the ANSWER key. An incoming call received on Prime Line has higher priority than other calls.
6. When multiple stations answer incoming calls at the same time, there may be differences between the displayed Caller ID and that of the actually answered call. In that case, check the Caller ID displayed after answering the call.
7. When multiple Sub Lines receive incoming calls, you cannot identify which call (call terminated to Sub Line) is currently displayed on the LCD.
8. When you use Terminal Call History (terminal side call history), only the history data of answered call will be saved. (history data of unanswered call is not saved even if the Caller ID is displayed).

Note: History data for Prime Line can be saved regardless of whether the call is answered or not answered; however, when using Soft Phone, history data is not saved in the following situation.
- A call is not answered.
- A call via Soft Phone WLAN Handset or SOFT PHONE - PS is answered.
- A call via Desktop terminal with call history feature is answered.

9. This feature has the following limitations on the Line Pickup Preference function controlled by the LN PRE parameter of the AKYD command:

   - When LN PRE is set to 0 (Standard)

     Incoming Line Preference (Button Location Ascending Order) does not work when the ANSWER button is pressed.

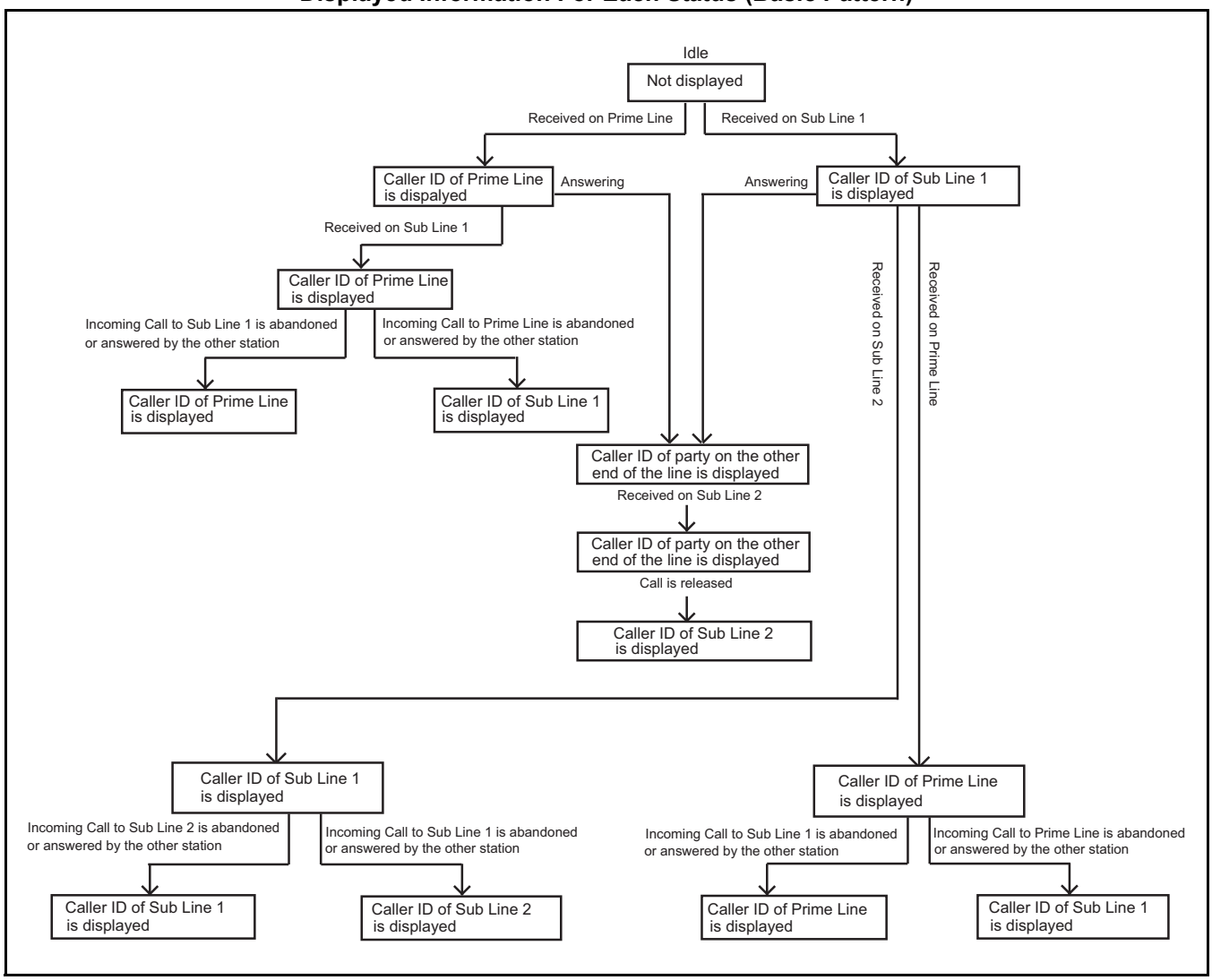
   - When LN PRE is set to 1 (Customize; Line Preference data is set by ANSWER button and ORIGINATING button.)

     The ANS parameter (Line Preference For Answer Button or Handset/Headset) cannot be set to 2 (Ringing Line Preference (CO Incoming Call takes precedence)). Since this option gives preference to CO incoming calls over other incoming calls, the CALLER ID DISPLAY - SUB LINE [C-204] feature does not operate properly.

10. If a call is terminated to any line while you are talking on or placing a call, LCD does not display the information of the incoming call until you finish talking or placing the call.

Note: If you need to see incoming call information while you are talking or placing a call, assign ASYDL/ASYDN, SYS1, Index 1248, Bit 4=1 (Mixed Display of Caller ID Display-Subline is enabled).

Displayed Information For Each Status (Basic Pattern)



Note: The above flowchart shows an example of displayed information.

11. The following conditions are applied when you use Terminal Phonebook.
    a.) LCD displays as follows:

When a Sub Line is being called	Telephone number of a called party is only displayed
When a call terminated on a Sub Line is being answered	Information registered on Terminal Phonebook is displayed

    b.) When you also use a name display service by system settings such as Name Display data corresponding to each station number (assigned by the ANDD/ANDD_T command), Caller ID data (assigned by the ACID command) or System Phonebook data (assigned by the APBKN command), LCD displays as follows:

When a Sub Line is being called	Caller name by system settings such as the ANDD/ANDD_T, ACID or APBKN command data is displayed
When a call terminated on a sub line is being answered	Information registered on Terminal Phonebook is displayed

Note: For the other conditions concerning Terminal Phonebook, see Phonebook [P-83].

<Other Conditions>

None.

Programming

STEP 1: **ASYD** - System Data
SYS1, Index 390, Bit 2=1 (Caller ID is displayed on the bottom line of the LCD)

STEP 2: **ASYDL** - System Data for LDM
SYS1, Index 672, Bit 1=1 (24-digit format is enabled)

STEP 3: **ASYDL/ASYDN** - System Data (LDM/NDM)
SYS1, Index 1248, Bit 0=1 (Caller ID Display - Sub Line is enabled)
SYS1, Index 1248, Bit 4=0 (Mixed Display of Caller ID Display-Subline is disabled)

Note: Initialize the terminal (for IP Multiple Line terminal/SIP Multiple Line terminal) or ELC card (for Digital Multiple Line terminal) for the change of this system data to take effect.

STEP 4: **AKYD** - Key Data for Desktop Terminal
TN: Tenant Number
STN: Station Number (Assign Station Number of My Line)
PRI: Form of Line Preference
PL TN: Prime Line Tenant
PL STN: Prime Line Station Number
MWD: Message Waiting Data Display Selection
0: MW data display on the top line
1: MW data display on the bottom line
LN PRE: Line Preference **Note 2**, **Note 3**
0: Standard
1: Customize (Line Preference data is set by ANSWER button and ORIGINATING button)

Note 2: For details about Line Pickup Preference setting, see “5.4 Line Pickup Preference” in CHAPTER 2.

Note 3: When LN PRE is set to 1, do not assign 2 (Ringing Line Preference (CO Incoming Call takes precedence)) to the ANS parameter of this command.

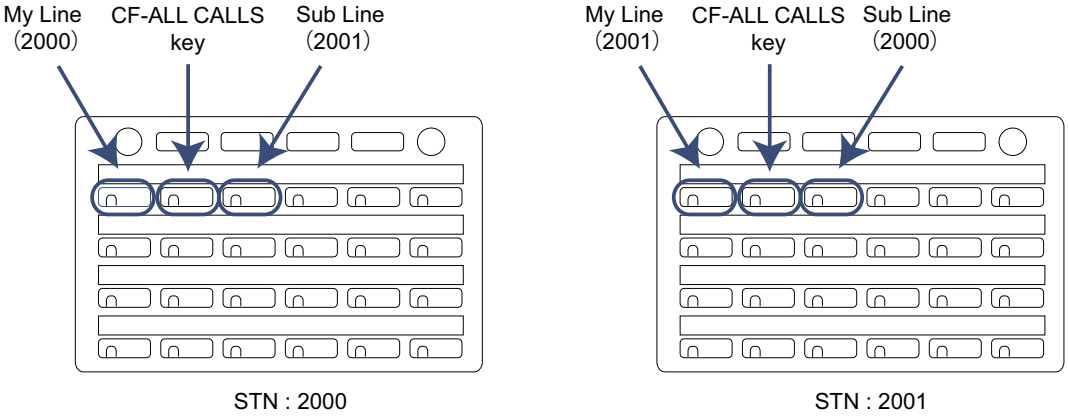
C-206 Call Forward Line Lamp Indication

General Description

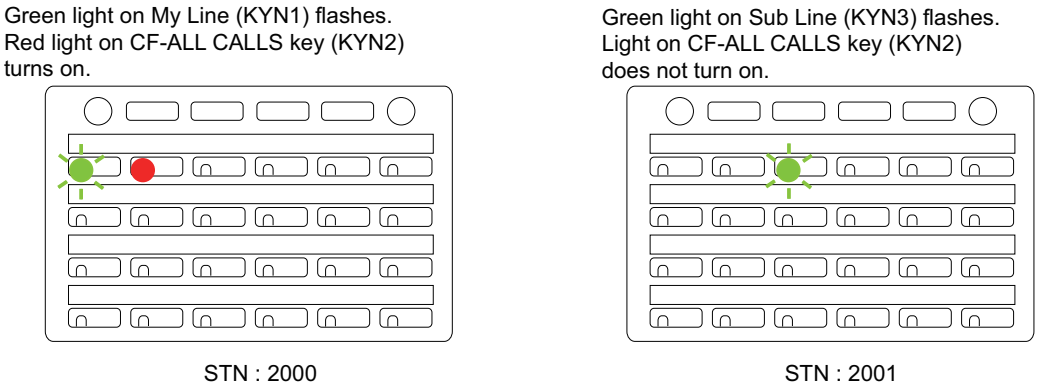
A Line key lamp turns on or flashes (hereinafter both status are described as “flash”) by enabling this feature when Call Forwarding-All Calls features are set to My Line as well as Sub Lines. You can set the indication pattern of the Line key lamp by assigning system data.

The following shows the behavior of Line key lamps when this service is used with the settings outlined below.

Key Number	Settings	
	STN : 2000	STN : 2001
1	My Line (2000)	My Line (2001)
2	Call Forwarding - All Calls	Call Forwarding - All Calls
3	Sub Line (2001)	Sub Line (2000)



<Call Forwarding - All Calls is set to STN: 2000>



Operating Procedure

None.

Service Conditions

<Important Notices>

1. After changing the system data, perform one of the following operations for each system data to reflect the setting.

   [Index 679, Bit 3 and Index 681, Bit 3]

   - Setting and canceling Make-Busy for a station by the MBST command

   [Index 1257, Bit 3, Bit 4 to Bit 6 and Bit 7]

   - System Initialization

<Interactions with Other Features>

1. If the Prime Line key is flashing due to this feature, Dynamic Dial Pad located in SELECTIVE DIALING [S-162] cannot be used.
2. If Call Forwarding-All Calls features are set for My Line on the terminal where Software Multiple My Line Appearance [S-143] is set, the Line key lamp of the Same Number Sub Line does not flash.
3. When using this feature and Message Lamp Control with Message Center Interface [M-26] and Message Waiting Lamp Setting - Station [M-30] at the same time, the priority of these features needs to be set by the system data, Line Lamp Priority for Call Forward Line Lamp Indication (ASYDL, SYS1, Index 681, Bit 3).
4. This feature cannot be used on Line keys that are accommodated by PS Line Appearance On Dterm [P-71].
5. This feature is not available for OAI systems.

<Hardware/Software>

1. For DT800 Series/DT700 Series, when the Prime Line key is flashing due to this feature, a call cannot be originated from the Terminal Phonebook.

   Note: A small LCD type of DT700 Series does not have Terminal Phonebook.

2. For DESI-less terminal, an icon is displayed in accordance with the indication pattern while the Line key lamp is flashing. The Exclusive Hold icon is displayed with the default setting of the system data (ASYDL, SYS1, Index 1257, Bit 4 to Bit 6).

   Note: When Call Forward Line Lamp Indication Pattern (ASYDL, SYS1, Index 1257, Bit 4 to Bit 6) is set to 1 or 7, an icon turns on. Other than this setting, an icon flashes (the flashing patterns cannot be set).

3. For DT700 Series (a large LCD type), an icon can be displayed instead of the Line key lamp indication. When using this function, set the system data (ASYDL, SYS1, Index 1257, Bit 7=1).

   Note: When Call Forward Line Lamp Indication Pattern (ASYDL, SYS1, Index 1257, Bit 4 to Bit 6) is set to 1 or 7, an icon turns on. Other than this setting, an icon flashes (the flashing patterns cannot be set).

4. When using this feature on Softphone, you need to assign Call Forward Line Lamp Indication for Softphone (ASYDL, SYS1, Index 1257, Bit 3=1). In this case, the Softphone display shows an icon on the action screen in accordance with the indication pattern (the default is the "HOLD" icon). When Call Forward Line Lamp Indication Pattern (ASYDL, SYS1, Index 1257, Bit 4 to Bit 6) is set to 1 (Red light turns on) or 7 (Green light turns on) and Call Forwarding-All Calls is set, once you lift the handset you will hear no tone and you will not be able to cancel Call Forwarding-All Calls, therefore, avoid to set to "Red light turns on" or "Green light turns on" patterns.

5. When using this feature on Softphone, if you want to cancel Call Forwarding-All Calls, lift the handset and press the Feature Key for Call Forwarding-All Calls. Using the forwarding set screen of Softphone you can set Call Forwarding-All Calls but you cannot cancel it.

6. This feature is not available for the following terminals.
   - DtermSP30 (Proprietary Protocol/SIP)

7. When you use DSS Console, the following restrictions are applied at the time of terminal re-registration.

   - ADM mode

     This feature can be set for the top 24 Line keys on a DSS console.

   - DSS mode

     This feature cannot be set for the DSS keys on a DSS console.

8. This feature is not available for a terminal that does not have lamps on Line keys.

9. This feature is not available for a PA-16ELCJ-B card.

<Conditions Specific to This Feature>

1. Do not load the Call Forwarding data to a memory during system operation. If you do so, the Desktop terminal LCD may not indicate the correct Call Forwarding status.
   To correct this, you can do either of the following:

   - Setup or cancellation of the Call Forwarding feature

   - System Initialization

2. When Call Forward Line Lamp Indication Pattern (ASYDL/ASYDN, SYS1, Index 1257, Bit 4 to Bit 6) is set to 1 (Red light turns on) or 7 (Green light turns on) and Call Forwarding-All Calls is set, you can originate a call by lifting the handset or pressing the Speaker key after pressing the lighting Line key.

<Other Conditions>

None.

Programming

ASYDL/ASYDN - System Data (LDM/NDM)

System Data 1

Index 679
Bit 3=1: Call Forward Line Lamp Indication is enabled **Note 1**

Index 681
Bit 3: Line Lamp Priority for Call Forward Line Lamp Indication **Note 1**

0 = Message Waiting Lamp Setting - Station [M-30] has priority over Call Forward Line Lamp Indication [C-206]

1 = Call Forward Line Lamp Indication [C-206] has priority over Message Waiting Lamp Setting - Station [M-30]

Index 1257
Bit 3: Call Forward Line Lamp Indication for Soft Phone

0 = Call Forward Line Lamp Indication is not enabled on Soft Phone

1 = Call Forward Line Lamp Indication is enabled on Soft Phone

Bit 4-6: Call Forward Line Lamp Indication Pattern

: Green light is on : Red light is on : Light is off
The number (ms) displays how long the lights turn on/off.

Setting	Status	Pattern
0	Green light flashes	640ms, 128ms, 128ms, 128ms
1	Red light turns on	
2	Red light flashes	128ms, 128ms, 128ms, 128ms, 128ms, 128ms, 128ms, 128ms
3	Red light flashes	512ms, 512ms
4	Green light flashes	128ms, 128ms, 128ms, 128ms, 128ms, 128ms, 128ms, 128ms
5	Green light flashes	512ms, 512ms
6	Green light flashes	640ms, 128ms, 128ms, 128ms
7	Green light turns on	

Note: When using this feature on Softphone (SYS1, Index 1257, Bit 3=1), set the Bit 4 to Bit 6 to 0 or 2~6.

Bit 7: Call Forward Line Lamp Indication for DT700 Series (Large LCD Type)

0 = Call Forward Line Lamp Indication is not enabled on DT700 Series (Large LCD Type)

1 = Call Forward Line Lamp Indication is enabled on DT700 Series (Large LCD Type)

Note 1: To apply the change, setting and canceling Make-Busy for a station by the MBST command are required.

C-207 Call Pickup - One Touch Key

General Description

This feature allows an incoming call directed to a Call Pickup Group to be answered using a Line/Feature key assigned to a Desktop terminal. Desktop terminal users can pick up the call by pressing the Feature key corresponding to the Call Pickup Group.
If multiple Call Pickup Groups are assigned to Feature keys, the lamp for the corresponding Call Pickup Group key flashes when an incoming call arrives to a station.

Operating Procedure

To pick up an incoming call

1. A station belonging to Call Pickup Group A receives an incoming call.
2. The Lamp for the Feature key that is assigned to Call Pickup Group A flashes red.
3. Lift the handset or press the SPEAKER key.
4. Press the Feature key corresponding to the Call Pickup Group. You can now start talking with the calling party.

Service Conditions

<Important Notices>

None

<Interactions with Other Features>

1. A Call Pickup Group for this feature is not the same as a Call Pickup Group assigned for CALL PICKUP - GROUP [C-7] feature.
2. This feature is available in combination with the following features:
   CALL PICKUP - GROUP [C-7]
   Expanded Call Pickup - Group
   CALL PICKUP - DIRECT [C-30]
3. This feature is counted as CALL PICKUP - GROUP [C-7] in the SERVICE FEATURE PEG COUNT [S-27]. This feature is not counted individually.
4. Location Diversity feature is not available with this feature.
5. As for DIAL PLAN TO SUPPORT IP CENTREX FUNCTIONALITY [D-166], note the following restrictions:
   - Assignment of the data by the ACPKN command is available for the same site in the same node.
   - Assignment of the data by the ACPDN command is available not only for the same site but also for the different site in the same node.
6. The CALL PICKUP - ONE TOUCH Feature key cannot be used to answer a call from the paged individual through Paging Answer Access Code in the Delay System of PAGING TRANSFER [P-10].

<Hardware/Software>

1. When an incoming call arrives to a Call Pickup Group, the lamp for the Feature key assigned to the Call Pickup Group flashes red.
2. When multiple incoming calls arrive to a Call Pickup Group, the Feature key continues to flash until all the calls are picked up. When a single incoming call arrives to a Call Pickup Group, the Feature key stops flashing after the call is picked up.
3. Once Ringer Tone is enabled for a Call Pickup Group, the ringing can be enabled/disabled per terminal by assigning the RG parameter in AKYD/AKYD_T command.

Note: For FP95-114 V4 or earlier, when an incoming call arrives to a Call Pickup Group, the lamp for the Line/Feature key flashes but there is no Ringing Tone.

a.) You can set whether to enable/disable the Ringing Tone per Call Pickup Group.
b.) The ringer pattern of an incoming call can be assigned by ASYDL/ASYDN, SYS1, Index 685, Bit 0-2. Also, when multiple incoming calls arrive to the Call Pickup Group, the priority of the ringer pattern for terminals other than those belonging to the group, will be as follows:

INDEX 685 bit 0-2	Ringer Pattern Name	Ringer Pattern Setting (default)	Ringing Tone	Priority of when multiple incoming calls arrive
0	No tone	-	No tone	7 (Low)
1	Ringer Pattern 1	ASYD, SYS3, Index1 (0x21)	1 second ON/2 seconds OFF (When Index3, bit 1=1, the tone is 0.4 seconds ON/0.2 seconds OFF/0.4 seconds ON/2 seconds OFF)	6
2	Ringer Pattern 0	ASYD, SYS3, Index0 (0x21)	1 second ON/2 seconds OFF (When Index3, bit 0=1, the tone is 0.4 seconds ON/0.2 seconds OFF/0.4 seconds ON/2 seconds OFF)	5
3	-	-	0.5 seconds ON/0.5 seconds OFF	4
4	Not used (This setting is restricted)			
5	Ringer Pattern 5	ASYD, SYS3, Index7 (0x42)	2 seconds ON/4 seconds OFF (When Index3, bit 5=1, the tone is 0.4 seconds ON/0.2 seconds OFF/0.8 seconds ON/0.2 seconds OFF/0.4 seconds ON/4 seconds OFF)	3
6	Ringer Pattern 6	ASYD, SYS3, Index8 (0x00)	1 second ON/2 seconds OFF	2
7	Ringer Pattern 7	ASYD, SYS3, Index9 (0x11)	1 second ON/1 second OFF	1 (High)

For example:
If an incoming call arrives to the Call Pickup Group that is assigned Ringer Pattern 0, and the My Line is assigned Ringer Pattern 1, the ringing tone changes to Ringer Pattern 0.

c.) Depending on the settings of the RG parameter in the AKYD command and system data (ASYDL/ASYDN, SYS1, Index 685, bit 0-2), the terminal operation when an incoming call arrives to a Call Pickup Group is as follows:

RG Parameter of AKYD Command	SYS1 INDEX 685 Bit 0-2	Call Indicator Lamp	Ringing Tone
0-6	0	No lamp indication	No tone
	0-7	No lamp indication	No tone
7	0	No lamp indication	No tone
	1/2/3/4/5/6/7	Flashes red	Ringer pattern is 1/2/3/4/5/6/7

Note: The flashing pattern of the call indicator lamp cannot be changed.

d.) For the terminals accommodated in a UG50, the ringer patterns for incoming calls are different from those of the terminals accommodated in the Telephony Server. For details, see 7.5 Limitations on Voice and Tone in the CHAPTER 1 BEFORE INSTALLATION on Peripheral Equipment Description (UG50).

e.) When RG=7 is assigned to the group pickup key on AKYD/AKYD_T command, if an incoming call arrives at a busy state terminal that is in Call Pickup Group, the ringing tone volume will be smaller than one used in on-hook status (approximately 20dB smaller than the setting).

f.) When using a headset, ringing tone volume will be lower for the incoming call except during a conversation. The volume for the ringing tone can be set not to change by assigning SFI: 263=1 in ASFC/ASFCN command. For details, see RING VOLUME IN HEADSET MODE [R-62].

4. As for Soft Phones, flashing of a Line/Feature key Lamp can be shown by displaying Line Lamp Indication.
5. A Call Pickup Group can be assigned to Line/Feature keys of Desktop terminals but it cannot be assigned to Soft keys.
6. The following terminals can belong to a Call Pickup Group, but cannot pick up calls arriving to the Call Pickup Group.

- Analog station
- SIP Handler Controlled SIP terminal

<Interactions with Networks>

1. For the system using FCCS network, the stations assigned to the same Call Pickup Group must be accommodated in the same Node.

<Conditions Specific to This Feature>

1. When using CALL PICKUP - DIRECT [C-30], you can answer a specific incoming call for the Call Pickup Group using the access code assigned for Call Pickup - Direct + the station number that is receiving the incoming call you want to answer.

   Note: Available since FP95-115 V5. For FP95-114 V4 or earlier, call pickup with access code is not available.

2. When an incoming call to a station that is a member of Group A is picked up and another new incoming call arrives for a different station that is also a member of Group A, the latter call can also be picked up by pressing the Feature key of Group A assigned to the other Desktop terminals.
3. Telephone Numbers assigned in the ALGSN command are used as the station numbers for the Call Pickup Group members.
4. This feature is available only to the member stations accommodated in the same Node. An incoming call to the group that is accommodated in another Node cannot be picked up.
5. A maximum number of 1000 Call Pickup Groups can be assigned per System.
6. A maximum number of 50 stations can be assigned per Call Pickup Group.
7. A maximum number of five Call Pickup Groups can be assigned per Desktop terminal.
8. The same Call Pickup Group can be assigned to up to 100 Desktop terminals.
9. A station cannot be assigned to two or more different Call Pickup Groups.
10. The same Call Pickup Group cannot be assigned to a Desktop terminal more than once.
11. The conditions for accommodating member stations of Call Pickup Group and Desktop terminals are as follows.

System Configuration	Conditions for Accommodation
IMG/MMG System	within IMG0 - IMG3 (including virtual lines)
UMGi/RNIP System	Not supported

12. If multiple incoming calls arrive to stations assigned to the same Call Pickup Group, the order for answering the incoming calls depends on system data. The following two patterns will determine the order that is used to answer the incoming calls.

    Pattern 1:
    The calls are picked up in the order they are assigned to the group in the ACPKN command (ASYDL/ASYDN, SYS1, Index 685, bit 7=0).

    Pattern 2:
    The incoming calls are picked up in the order of their arrival (ASYDL/ASYDN, SYS1, Index 685, bit 7=1).

    Note: For FP95-114 V4 or before, only Pattern 1 is supported.

13. For Pattern 2 above, the conditions are as follows:
    a.) When an incoming call arrives to a Call Pickup Group, if the incoming call is forwarded to a station within the same group or to a different group via Call Forwarding - All Calls/Busy Line/ Logout, the order of the incoming call will be last in the forwarding destination group. After that, calls to the forwarding destination group, are answered in the order of the incoming calls to the forwarding destination group including the forwarded call.
    b.) When an incoming call arrives to a Call Pickup Group, if the incoming call is forwarded to a station within the same group or to a different group via Call Forwarding - Don't Answer, the order of the incoming call will be last in the forwarding destination group (the order of the incoming call before being forwarded will not be maintained). After that, calls to the forwarding destination group, are answered in the order of the incoming calls to the forwarding destination group including the forwarded call.

Note:

- If a call to a Call Pickup Group of the forwarding destination is picked up before the call is forwarded via Call Forwarding - Don't Answer, the call is answered in the order of the forwarding destination group if there are other incoming calls to the forwarding destination group. If there are no incoming calls to the forwarding destination group, the pickup will be restricted.
- After a call has been forwarded via Call Forwarding - Don't Answer, when you pick up a call to the Call Pickup Group that the call has been forwarded to, if there are other incoming calls on the forwarded destination group, the call is answered in order of the forwarded destination group except the forwarded call. If there are no incoming calls to the forwarding destination group, the pickup will be restricted.

    c.) When an incoming call arrives to a Call Pickup Group, if the incoming call is transferred to a station within the same group or to a different group by Call Transfer - All Calls, the order of the incoming call will be last in the transferring destination group. After that, calls to the transferring destination group, are answered in order of the incoming calls to the transferring destination group including the transferred call (The call has already been answered. Therefore, the order of the incoming call before being transfered will not be maintained. The call will be regarded as the new incoming call to the transferring destination group without relating the order after the call transferred.)
    d.) When the setting for answering calls in order of arrival for Call Pickup - One Touch Key (ASYDL/ASYDN, SYS1, Index 685, Bit 7) is changed from disable to enable, calls can be picked up in the order of arrival only if the incoming calls arrived to the pickup group while the system data is being set to enabled. Calls arriving to the pickup group when the system data is still set to disabled will be last.
    e.) When the setting for answering calls in order of arrival for Call Pickup - One Touch Key (ASYDL/ASYDN, SYS1, Index 685, Bit 7) is changed from enable to disable, calls can be picked up in the order they are assigned to the Call Pickup Group in the ACPKN command.
    f.) Change the data of ACPKN command only when all lines for the terminals are in an idle state and when there are no incoming calls in the pickup group.
14. When incoming calls to a Call Pickup Group are answered, the pickup information can be not displayed on the LCD by ASYDL/ASYDN, SYS1, Index 685, Bit 4.

15. When an incoming call is arriving at a Call Pickup Group, when the assignment is changed for the CALL PICKUP - ONE TOUCH Line/Feature keys on each terminal in the ACPDN command, the following are conditions for the lamp indication for the Line/Feature keys.

    Note: For FP95-114 V4 or before, be sure to change the setting in the ACPDN command only when all lines of the terminals are in an idle state and when there are no incoming calls to the pickup group. If you change the group number when an incoming call is arriving to the Pickup Group, the lamp for the feature key for the Call Pickup Group might not flash or turn off.

    a.) When you are changed from a group that is receiving a call to a group that is not receiving a call or if deleted from a group that is receiving a call, the lamp is turned off immediately.
    b.) When you are changed from a group that is not receiving a call to a group that is receiving a call or if creating a new group, the lamp for the new incoming calls arriving to the changed group, the lamp will flash red. Also, you can answer the incoming call when the lamp is turned off.
    c.) When you are changed from a group that is receiving a call to a group that is receiving a call, the lamp is turned off immediately. If a new incoming call arrives to the changed group, the lamp flashes red. Also, you can answer the incoming call for the changed group when the lamp is turned off.
    d.) When the assignment of a Line/Feature key is changed from one group, that is receiving an incoming call, to another groups that is also receiving an incoming call, the lamp will be turned off immediately. If a new incoming call arrives to the group, the lamp for the changed Line/Feature key will flash red.
    You can answer the incoming call by pressing the Line/Feature key for the changed group, but you cannot answer the incoming call by pressing the Line/Feature key before it is changed; the lamp status will not be changed.
    e.) When a new group is being assigned via the ACPDN command while an incoming call is arriving to the Call Pickup Group, the lamp for the Line/Feature key will be off. When a new incoming call arrives after the group is added, the lamp will flash red. Also, you can answer the incoming call even if the lamp is turned off.

<Other Conditions>

None

Programming

STEP 1: **ASYDL/ASYDN** (Assignment of System Data (LDM/NDM))

SYS1
Index 679
Bit 5=1 (Call Pickup - One Touch Key = Enabled)

Index 685 **Note 1**
Bit 0-2=1-7 (Ringing Tone for Call Pickup - One Touch Key (1-7))
Bit 3=1 (Call Pickup - One Touch Key Enhancement = Enable)
Bit 4=0/1 (Call Pickup information display on LCD = 0 (Enable)/1 (Disable))
Bit 7=0/1 (Answering a call in order of incoming calls for Call Pickup - One Touch Key = 0 (Disable)/1 (Enable))

Note 1: Note the following conditions:

- Available since FP95-115 V5.
- For the Bit 3, assign **Enable** to use the following commands:
  - ASYDL/ASYDN, SYS1, Index 685, Bit 0-2
  - ASYDL/ASYDN, SYS1, Index 685, Bit 4=1
  - RG=7 setting for AKYD/AKYD_T command

STEP 2: **ACPKN** (Assignment of Call Pickup Key Group for NDM)
Assign Telephone Numbers (up to 50 station numbers) to the corresponding Call Pickup Group.

TYPE: Select either of the radio buttons.
Telephone Number
Call Pickup Group Number

When Telephone Number is selected:
BASE UGN: User Group Number (1 - 255)
BASE TELN: Telephone Number (up to 16 digits)
BASE CPG: Call Pickup Group Number (1 - 1000)
TELN: Telephone Number (up to 16 digits)

When Call Pickup Group Number is selected:
BASE CPG: Call Pickup Group Number (1 - 1000)
UGN: User Group Number (1 - 255)
TELN: Telephone Number (up to 16 digits)

STEP 3: **AKYD/AKYD_T** (Assignment of Desktop Terminal Data/ Assignment of Desktop Terminal Data (Telephone Number))
Assign up to five Line/Feature keys for Call Pickup - One Touch keys to a Desktop terminal.
FKY=185 - Call Pickup Key Group-1
FKY=186 - Call Pickup Key Group-2
FKY=187 - Call Pickup Key Group-3
FKY=188 - Call Pickup Key Group-4
FKY=189 - Call Pickup Key Group-5

To provide a ringing tone when an incoming call arrives to the Call Pickup Group, assign RG (Ringing information for each multi-line (0-7)) = 7 to the target group pickup key. **Note 2**

Note 2: The conditions are as follows.
- Available since FP95-115 V5.
- When you assign 1-6 to RG, the terminal works the same as the parameter 0 (No tone).
- RG cannot be specified to fix key. Be sure not to assign FKY=185-189 to fix key.
- For AUPDN command and AKDTN command, RG cannot be specified on FKY=185-189.
- For ADM mode of DSS console, be sure not to assign FKY=185-189 to the key data of DSS console by AKYD/AKYD_T command.

STEP 4: **ACPDN** (Assignment of Call Pickup Key Group Dterm for NDM)

Assign Call Pickup Group Numbers to Call Pickup - One Touch keys.
UGN: User Group Number (1 - 255)
TELN: Telephone Number (up to 16 digits)
CPG: Call Pickup Group Number (1 - 1000)

Note: To delete/change the data, perform the above procedure in reverse order.
Deletion/change of the data through the ACPDN command should be performed when no incoming call arrives to a Call Pickup Group.
If the data is deleted/changed while an incoming call arrives to a Call Pickup Group, there is a possibility that Feature key lamps might not flash/turn off.

C-208 Call Forwarding (CF) Button - One Touch

General Description

By registering the Call Forwarding - All Calls (**Note 1**) Access Code + a desired Forwarding Destination Number to a CF Setting button that is assigned to one of the Line/Feature keys on a Desktop terminal, this feature allows the setting and canceling of the Call Forwarding - All Calls feature just by pressing a CF Setting button.

Note: This feature is available from FP95-114 V4 Issue 2.0.

Note 1: Regarding this feature, Call Forwarding - All Calls is used to refer to the following features:

- CALL FORWARDING - ALL CALLS [C-5]
- CALL FORWARDING - ALL CALLS - ANNOUNCEMENT [C-69]
- CALL FORWARDING - ALL CALLS - OUTSIDE [C-28]
- MULTIPLE CALL FORWARDING - ALL CALLS [M-44]
- SPLIT CALL FORWARDING [S-99] (only the Call Forwarding - All Calls related content)
- CALL FORWARDING - ALL CALLS - CCIS [C-45]
- MULTIPLE CALL FORWARDING - ALL CALLS - CCIS [M-40]
- CALL FORWARDING-ALL CALLS- PRI/PRA [C-95/111]
- CALL FORWARDING-ALL CALLS- ANNOUNCEMENT - PRI/PRA [C-123/124]
- CALL FORWARDING-ALL CALLS - OUTSIDE - PRA [C-139]
- MULTIPLE CALL FORWARDING - ALL CALLS - PRI/PRA [M-75/83]

To register the desired Forwarding Destination Number

(A) Registration with Feature key

1. Press the Feature key located among the Flex keys.
2. Press a CF Setting button to register.
3. Dial the Call Forwarding - All Calls entry Access Code + the desired Forwarding Destination Number.
4. Press the Feature key.

(B) Registration with SPD setting key **Note 2**

1. Press the SPD setting key registered to one of Line/Feature keys or Soft keys.
2. Press the CF Setting button to register.
3. Dial the Call Forwarding - All Calls Access Code + the desired Forwarding Destination Number.

Note 2: For details on the SPD setting key, see SPEED CALLING - ONE TOUCH [S-26D] feature in this manual.

To set the Call Forwarding - All Calls feature

1. Press the CF Setting button to which the desired Forwarding Destination Number is registered.

To cancel the Call Forwarding - All Calls feature

1. Press any CF Setting button where Call Forwarding - All Calls feature is assigned.

Service Conditions

<Important Notices>

1. The CF Setting button can be used also as the button for Speed Calling - One Touch. However, in that case, the lamp control is only for Call Forwarding - All Calls.
2. If there is a Line/Feature key set with Call Forwarding - All Calls using the AKYD command, the lamp of such key also turns on and off. The lamp patterns for turning on and off of the CF Setting button (AKYD command FKY= 201 to 204) and the Call Forwarding - All Calls Line/Feature key (AKYD command FKY= 2) are as follows:

When Call Forwarding Feature is Set

Operation	Line/Feature key Lamp	
	CF Setting button	Call Forwarding - All Calls button
CF Setting button registered with Access Code (Call Forwarding - All Calls) + desired Forwarding Destination Number is pressed	Light	Light
Key registered with Speed Calling - One Touch and Access Code (Call Forwarding - All Calls) + desired Forwarding Destination Number is pressed	Not Light	Light
Access Code (Call Forwarding - All Calls) + desired Forwarding Destination Number are dialed	Not Light	Light
Call Forwarding - All Calls Line/Feature key is pressed + desired Forwarding Destination Number is dialed	Not Light	Light
Call Forwarding - All Calls data is set with the ACFS command	Not Light	Light
Call Forwarding - All Calls feature is set in the OAI service	Not Light	Light
CALL FORWARDING - I'M HERE [C-81] is set	Not Light	Light
REMOTE CALL FORWARDING CONTROL [R-54] is set	Not Light	Light

When Call Forwarding Feature is Canceled

Operation	Line/Feature key Lamp	
	CF Setting button	Call Forwarding - All Calls button
CF Setting button registered with Access Code (Call Forwarding - All Calls) + desired Forwarding Destination Number is pressed	Not Light	Not Light
Key registered with Speed Calling - One Touch and Access Code (Call Forwarding - All Calls) + desired Forwarding Destination Number is pressed	Not Light	Not Light
Access Code (Call Forwarding - All Calls) is dialed	Not Light	Not Light
Cancel Code (Call Forwarding - All Calls) is dialed	Not Light	Not Light
Call Forwarding - All Calls Line/Feature key is pressed	Not Light	Not Light
Call Forwarding - All Calls data is deleted with the ACFS command	Not Light	Not Light
Call Forwarding - All Calls feature is canceled in the OAI service	Not Light	Not Light
CALL FORWARDING - I'M HERE [C-81] is canceled	Not Light	Not Light
REMOTE CALL FORWARDING CONTOL [R-54] is canceled	Not Light	Not Light
Call Forwarding All Clear is set	Not Light	Not Light

3. For Digital terminals, register the CF Setting button with the AKYD command in the range of KYN= 1 to 32. In the range of KYN= 33 to 40, the CF Setting button cannot be used.
   SIP Multiple Line terminals can use the range KYN= 1 to 40 if 16LK Unit is used. Also, if ADM mode is used, the valid range is KYN= 1 to 16.
4. Several CF Setting buttons can be set with different desired Forwarding Destination Numbers. However, more than one CF Setting button cannot be set at a time.
5. If the operations described below are performed when a CF Setting button is set, the lamp of the CF Setting button turns OFF. Therefore, before performing such operations, cancel the CF Setting button and verify that the lamp turns OFF.

   - Changing/deleting a phone number that is assigned to the CF Setting button with the Feature key or the SPD Setting key.
   - Writing One-Touch Speed Calling data from HD to the memory with the BOSD command.
   - Replacing an ELC card.

6. Backup of the CF Setting button lamp state can be performed with the MEM_HDD command by selecting "One Touch Speed Calling for DtermIP". This way, when performing an Office Data Load & System Initialize, the state that is backup with the MEM_HDD command for CF Setting button lamp, One-Touch Speed Call Data (**Note 3**) and Call Forwarding data is used.

Note 3: In the case of Digital terminals, Speed Calling - One Touch Data is saved in ELC card and backup is not possible with the MEM_HDD command. For that reason, perform the backup and writing for this data with the BOSD command.

7. If station data is deleted, the data set for the CF Setting button is also deleted. When necessary, perform a backup with the MEM_HDD command or the BOSD command.
8. CF Setting button is available for DT800/DT700 Series and DT400/DT300 Series.
9. If the CF Setting button is used with a Digital terminal, an ELC card with the following firmware is required:

    - SP-4035 16ELCA PROG-A Issue 4 or later.

<Interactions with Other Features>

1. The following applies when using SPLIT CALL FORWARDING [S-99]:
    a.) A different CF Setting button can be set for an internal Station incoming call access code and an Outside Party (Central Office TIE, DID, etc.) incoming call access code. Therefore, setting and canceling operations can be performed also individually for each CF Setting button.
    b.) The CF Setting button lamp control can be different for the internal Station incoming call and the Outside Party incoming call.
2. If SOFTWARE MULTIPLE MY LINE APPEARANCE [S-143] is used, the desired Forwarding Destination Number can be set only for My Line.
3. CF Setting button cannot use Speed Calling - One Touch key with Page Changeover. Even if a page is designated when a desired Forwarding Destination Number is assigned, the same number is used to both pages.
4. For CALLING PARTY RECOGNITION SERVICE (CALL FORWARDING-All Calls/Busy Line/Don't Answer) [C-103] it is not possible to cancel the Call Forwarding - All Calls feature with the CF Setting button or by dialing the CALL FORWARDING - ALL CALLS entry ACCESS CODE (**Note 4**). Do not set the CF Setting button with this feature.

    Note 4: Call Forwarding - All Calls can be canceled with the entry Access Code with system data ASYDL/N, SYS1, Index 686, Bit 0= 1).

5. CF Setting button cannot be used if the features below are in use. Do not set the CF Setting button.

    - LOCATION DIVERSITY [L-55].
    - SPEED CALLING - STATION [S-21]
    - SPEED CALLING - GROUP [S-23]
    - SPEED CALLING - STATION-PRI/PRA [S-84/102]
    - SPEED CALLING - GROUP-PRI/PRA [S-85/103]

<Interactions with Networks>

None.

<Hardware/Software>

1. Display (LCD) Indications on DESI-Less terminals show the same data as Speed Calling - One Touch. You can register data to be displayed on LCD Indications for CF Setting buttons as well. For details, see Speed Calling - One Touch - D on DESI-Less Phone of Interactions section in DESI-LESS TERMINAL [D-169]. Note that there is the following condition on LCD Indications for Speed Calling - One Touch.

    - When CF Setting button is in service (ASYDL, SYS1, Index 1249, Bit 4=1), registering information to be displayed on LCD Indications to the 13th to 16th SPEED CALLING - ONE TOUCH buttons (**Note 5**) on the page 1 is not available.

    Note 5: This does not apply to Line/Feature keys where Speed Calling - One Touch (FKY= 49 in the AKYD command) is assigned.

    - When DESI-Less icon display is in service (ASYDL/ASYDN, SYS1, Index 684, Bit 1=0), DESI-Less icon is displayed in synchronization with lamp status.

2. CF Setting button cannot be assigned to Line/Feature keys on the DSS console.
3. For the Cordless Terminal used in DT400/DT300 Series, if the CF Setting button is used in the Bluetooth Handset, assign KYN= 1-8 with the AKYD command.

<Conditions Specific to This Feature>

1. If a telephone using this feature is re-registered with the same station number, regardless of the settings of PH Fixed Hunting method, the CF Setting button set/cancel state and lamp state do not change.
2. Pressing any of the CF Setting buttons cancels the Call Forwarding - All Calls feature that is set on any of the CF Setting buttons.
3. The lamp control of the CF Setting button when setting/canceling the Call Forwarding - All Calls feature is only for My Line. If My Line is not the Primary Line, it is necessary to preset My Line as Primary Line before using the CF Setting button. If CF Setting button is used with a Sub Line, the setting/canceling of the Call Forwarding - All Calls feature is performed for such Sub Line and the CF Setting button lamp control does not happen.
4. If a CF Setting button is pressed, and then another CF Setting button is pressed while the setting or canceling of a function that is set to the first CF Setting button is still being processed, the operation of the second CF Setting button will not be processed.

<Others>

None.

Programming

Note: It is assumed that access code for Call Forwarding - All Calls has been assigned.

STEP 1: **ASYDL/ASYDN** - Assign the following System Data
SYS1, Index 686, Bit 0=1 (Disabling Call Forwarding - All Calls with entry Access Code) **Note 6**
SYS1, Index 1249, Bit 4=1 (CF Setting button is in service) **Note 6**

STEP 2: **AKYD** - Assign CF Setting button for Line/Feature key of the Desktop terminal. **Note 8**
KYN: Key number
KYI= 1
FKY= 201 to 204 (CF Setting buttons 1 to 4) **Note 6**, **Note 7**

Note 6: Available from FP95-114 V4 Issue 2.0.

Note 7: Assign four CF Setting buttons with four FKY (201-204) one by one.

Note 8: For Registration with SPD setting key, assign the following as well:
KYN: Key number
KYI= 1 (Function Key)
FKY= 199 (SPD setting)

C-209 Call Forwarding Destination Per Trunk Route - Incoming Call

General Description

This feature allows an incoming trunk call to be forwarded to a destination on an incoming route basis by specifying a destination with the ACFO/ACFOL/ACFON command.

Note: Please note the following conditions:

- Earlier than FP95-114 V4 Issue 2, only a Call Forwarding destination on a tenant basis could be assigned with the ACFO/ACFOL/ACFON command.
- This feature is available since FP95-114 V4 Issue 2.0.

Operating Procedure

None.

Service Conditions

<Important Notices>

1. This feature requires a vacant tenant (the maximum number of tenants set with ASYD, SYS1, Index 8 must be two or more).
2. To enable this feature, for the Call Forwarding data set with the ACFO command, Tenant Data table development must be separated by the system data (ASYD, Index 93, Bit 1). For the Call Forwarding data set with the ACFOL/ACFON command, Tenant Data table development is separated without condition. When the ACFO command is used, set the following to separate Tenant Data table development:

   ASYD, SYS1, Index 93, Bit 1=0
   (Separate Tenant Data table development for the Call Forwarding data (ACFO command))

3. When the Trunk Application data (ARTI/ARTIN command) via DM and NDM coexist in a system, the data registered on NDM (ARTIN command) takes priority over the data registered on DM (ARTI command).
4. When the Call Forwarding data (ACFO/ACFOL/ACFON command) via DM and LDM/NDM coexist in a system, the priority is:

   NDM (ACFON command) → LDM (ACFOL command) → DM (ACFO command)

5. When Tenant Data table development for the Station Data (ASDT, AAST, AGST, ALDN, ASAT commands) is separated (ASYD, SYS1, Index 92, Bit 3=0), a forwarding destination station cannot be set with the ACFO command. In this case, the forwarding destination station (Telephone Number) needs to be set with the ACFOL/ACFON command.

   Note: This feature uses vacant tenants. No station data such as station numbers are not assigned to the vacant tenants.

<Interactions with Other Features>

1. This feature does not support Trunk Answer from Any Station (TAS).
2. Setting a Call Forwarding destination from an individual station or attendant console, UCD and busy station services overrides the setting with the ACFO/ACFOL/ACFON command.
3. If Call Forwarding - Busy Line (CF=1) is set together with Call Forwarding - Don't Answer (CF=2), when a call arrives at the destination station of Call Forwarding - Don't Answer and the timeout of its timer occurs, the destination station is regarded as in busy state. Then the call is forwarded to another destination station of Call Forwarding - Busy Line (CF=1).
4. The table below shows relationship between incoming trunk calls and Call Forwarding features.

Supported: X, Not Supported: -, Supported with Conditions: C

Incoming Trunk Call		Call Forwarding Feature									
		CALL FORWARDING - BUSY [C-2] Note 1		CALL FORWARDING - DON'T ANSWER [C-3] Note 2		CALL FORWARDING - INTERCEPT/ ANNOUNCEMENT [C-25] Note 3		CALL FORWARDING - PS INCOMING INCOMPLETE [C-155]		LOGGED OUT IP STATION - CALL DESTINATION [L-53]	
		ATT	STN/ TELN	ATT	STN/ TELN	ATT	STN/ TELN	ATT	TELN	ATT	STN/ TELN
Feature Used When A Trunk Call Arrives	AUTOMATED ATTENDANT [A-82] (Via COT)	X	X	X	X	X	X	X	X	X	X
	DIRECT-IN TERMINATION (DIT) [D-7] (Via COT)	- Note 4	-	X	X	-	-	X	X	-	-
	DIRECT INWARD DIALING (DID) [D-8] (Via COT)	X	X	X	X	X	X	X	X	X	X
	AUTOMATED ATTENDANT [A-82] (From ISDN)	X	X	X	X	-	-	X	X	X	X
	DIRECT-IN TERMINATION (DIT) [D-7] (From ISDN)	- Note 4	-	X	X	-	-	X	X	-	-
	DIRECT INWARD DIALING (DID) [D-8] (From ISDN)	X	X	X	X	X	X	X	X	X	X
	MG-SIP (From SIP network)	X	X	X	X	X	X	X	X	X	X
	MG-SIP TIE LINE [M-122] (From SIP Tie line)	X	X	X	X	X	X	X	X	X	X
	CCIS TANDEM (From C.O. line)	X	X	X	X	-	-	X	X	X	X

Supported: X, Not Supported: -, Supported with Conditions: C

Incoming Trunk Call		Call Forwarding Feature									
		CALL FORWARDING - BUSY [C-2] Note 1		CALL FORWARDING - DON'T ANSWER [C-3] Note 2		CALL FORWARDING - INTERCEPT/ ANNOUNCEMENT [C-25] Note 3		CALL FORWARDING - PS INCOMING INCOMPLETE [C-155]		LOGGED OUT IP STATION - CALL DESTINATION [L-53]	
		ATT	STN/ TELN	ATT	STN/ TELN	ATT	STN/ TELN	ATT	TELN	ATT	STN/ TELN
Feature Used After A Trunk Call Arrives	BLIND TRANSFER TO STATION [B-18] (After Intermediated station' disconnection)	-	-	C Note 5	C Note 5	-	-	-	-	-	-
	CALL FORWARDING - BUSY [C-2] Note 1	X	X	-	-	-	-	-	-	-	-
	CALL FORWARDING - DON'T ANSWER [C-3] Note 2	-	-	X	X	-	-	-	-	-	-
	CALL FORWARDING - INTERCEPT/ANNOUNCEMENT [C-25] Note 3	-	-	-	-	X	X	-	-	-	-
	CALL FORWARDING - PS INCOMING INCOMPLETE [C-155]	-	-	-	-	-	-	X	X	-	-
	LOGGED OUT IP STATION - CALL DESTINATION [L-53]	-	-	-	-	-	-	-	-	X	X
	STATION HUNTING [S-7/S-8/S-9] (When all stations are busy)	C Note 4	C Note 6	-	-	-	-	-	-	-	-
	UNIFORM CALL DISTRIBUTION (UCD) [U-1] (When all stations in the UCD group are busy)	- Note 4	C Note 7	-	-	-	-	-	-	-	-

Note 1: Same conditions apply to the following features:
- CALL FORWARDING - BUSY LINE - CCIS [C-46]
- CALL FORWARDING - BUSY LINE - OUTSIDE [C-60]
- CALL FORWARDING - BUSY LINE - PRI/PRA [C-96/113]
- CALL FORWARDING - BUSY LINE - D - PRA [C-113D]

Note 2: Same conditions apply to the following features:
- CALL FORWARDING - DON'T ANSWER - CCIS [C-47]
- CALL FORWARDING - DON'T ANSWER - PRI/PRA [C-129/130]

Note 3: Same conditions apply to CALL FORWARDING - INTERCEPT/ANNOUNCEMENT - PRI/PRA [C-101/115].

Note 4: When a call arrives at a tenant where the attendant console/the night attendant console belongs by using DIRECT-IN TERMINATION (DIT) [D-7], the call needs to be received by the attendant console/the night attendant console according to the system data setting (ASYD, SYS2, Index 11 Bit 7). Thus this system data setting takes priority over the data assigned with the ACFO/ACFOL/ACFON command.

Note 5: When a call is received via ISDN by using DIRECT INWARD DIALING (DID) [D-8], the call cannot be transferred by using BLIND TRANSFER TO STATION [B-18].

Note 6: When a call is received by using DIRECT-IN TERMINATION (DIT) [D-7], the Call Forwarding feature is not available for the call.

Note 7: The Call Forwarding feature is available only for the incoming call from the analog trunk. It is not available for the incoming call using CCIS NETWORKING VIA IP (Peer-to-Peer connection) [C-163] and incoming call from the ISDN trunk.

<Hardware/Software>

None.

<Interactions with Networks>

None.

<Conditions Specific to This Feature>

1. Supporting patterns are as follows:

Supported: X, Not Supported: -

Call Forwarding Features (CF parameter set with ACFO/ACFOL/ACFON command)	Call Forwarding Destination	Support Status
1: Call Forwarding - Busy Line	Attendant Console (ATT) STN/TELN	X
2: Call Forwarding - Don't Answer	Attendant Console (ATT) STN/TELN	X
3: Call Forwarding - Intercept	Attendant Console (ATT) STN/TELN	X
4: Night Connection Outside	Abbreviated Digit Code (ADC)	-
5: Call Forwarding - Logout	Attendant Console (ATT) STN/TELN	X
6: Call Forwarding - Incomplete Number **Note 8**	Attendant Console (ATT) TELN Monitor	-
7: Call Forwarding - Not Reach **Note 9**	Attendant Console (ATT) TELN	X

Note 8: **6: Call Forwarding - Incomplete Number** is only displayed in the ACFOL/ACFON command.

Note 9: **7: Call Forwarding - Not Reach** is only displayed in the ACFON command.

<Other Conditions>

None.

Programming

This feature supports features mentioned below.

For the programming procedure, see explanations of each related feature in this manual.

- AUTOMATED ATTENDANT [A-82]
- BLIND TRANSFER TO STATION [B-18]
- CALL FORWARDING - BUSY [C-2]
- CALL FORWARDING - DON'T ANSWER [C-3]
- CALL FORWARDING - INTERCEPT/ANNOUNCEMENT [C-25]
- CALL FORWARDING - BUSY LINE - OUTSIDE [C-60]
- DIRECT-IN TERMINATION (DIT) [D-7]
- DIRECT INWARD DIALING (DID) [D-8]
- LOGGED OUT IP STATION - CALL DESTINATION [L-53]
- STATION HUNTING - CIRCULAR [S-7]
- UNIFORM CALL DISTRIBUTION (UCD) [U-1] **Note 10**

Note 10: Call Forwarding is available when the following condition applies:
When a call is received by using [D-7] DIRECT-IN TERMINATION (DIT) or an analog line call is received by using [D-8] DIRECT INWARD DIALING (DID) and the call overflows because all stations in the UCD group are busy. Also the following system data are assigned:

- ASYD, SYS1, Index 60, Bit 3=1 (UCD Queuing is not required (BT))
- ASYD, SYS2, Index 11, Bit 6=0 (Trunk Answer from Any Station (TAS) is out of service)

For the programming procedure, see explanations of each related feature in Data Programming Manual-CCIS.

- CALL FORWARDING - BUSY LINE - CCIS [C-46]
- CALL FORWARDING - DON'T ANSWER - CCIS [C-47]

For the programming procedure, see explanations of each related feature in Data Programming Manual-ISDN.

- CALL FORWARDING - BUSY LINE - PRI/PRA [C-96/113]
- CALL FORWARDING-BUSY LINE-D-PRA [C-113D]
- CALL FORWARDING - INTERCEPT/ANNOUNCEMENT - PRI/PRA [C-101/115]
- CALL FORWARDING - DON'T ANSWER - PRI/PRA [C-129/130]
- OVERSEAS COLLECT CALL ISDN DIVERSION [O-40]

For the programming procedure, see explanations in Data Programming Manual-PCS/PHS.

- CALL FORWARDING - PS INCOMING INCOMPLETE [C-155]

To set this feature, creating vacant tenants is required. The following explains the procedure how to change the maximum number of tenants coexisting in the system, create vacant tenants, and set this feature to the vacant tenants.

Note: For details of Tenant Development, see Tenant Data section in CHAPTER 2 ASSIGNMENT OF BASIC OFFICE DATA.

[When ASYD, SYS1, Index 93, Bit 1=0 (Separate Tenant Data Table Development for Call Forwarding Data (ACFO command))]

STEP 1: **ASYD** - System Data
SYS1
Assign the maximum number of tenants coexisting in the system.
Index 8 (Number of Tenants: 2-63 (02-3F (HEX))

Note: When the existing data is **01**, set a value equal to the number of Call Forwarding destinations to separate (**02** or more).

Separate Tenant Data Table Development for Call Forwarding Data (ACFO command).
Index 93, Bit 1=0 (Separate)

STEP 2: **MEM_HDD/MEM_HDD_N** - Data Control Between Memory and HDD
Back up the Call Forwarding data in the memory to the hard disk.

STEP 3: **SINZ/SINZN** - System Initialize
Execute one of these operations:

- System Initialize
- Office Data Load & System Initialize
- Restart

Note: It is possible to use Telephony Server Maintenance Menu in this step. In the Telephony Server Maintenance Menu, select **Reboot** in the SYSTEM CONTROL field (for a dual CPU system, perform the same operation on both systems).

STEP 4: Go to [Setting A Call Forwarding Destination on A Route Basis].

[When ASYD, SYS1, Index 93, Bit 1=1 (Common Tenant Data Table Development for Call Forwarding Data (ACFO command))]

Note: When ASYD, SYS1, Index 93, Bit1=1 (common Tenant Data table development for the Call Forwarding data (ACFO command)) is active on the system with the Call Forwarding feature data assignments, delete the existing Call Forwarding data assigned with the ACFO command at first. Add more vacant tenants, and then assign the Call Forwarding data again with the ACFO command.

STEP 1: **LCFO** - Listup of Call Forwarding Data
As a preparation to re-registration of ACFO command, list up all the current data assigned with the ACFO command.

STEP 2: **ACFO/ACFOL/ACFON** - Assignment of Call Forwarding Data
Delete the current ACFO/ACFOL/ACFON command data.

STEP 3: **ASYD** - System Data
SYS1, Index 8 (Number of Tenants: 2-63 (02-3F (HEX))

Note: When the existing data is **01**, set a value equal to the number of Call Forwarding destinations to separate (**02** or more).

Separate Tenant Data Table Development for Call Forwarding Data (ACFO command).
Index 93, Bit 1=0 (Separate)

STEP 4: **MEM_HDD/MEM_HDD_N** - Data Control Between Memory and HDD
Back up the Call Forwarding data in the memory to the hard disk.

STEP 5: **SINZ/SINZN** - System Initialize
Execute one of these operations:

- System Initialize
- Office Data Load & System Initialize
- Restart

Note: It is possible to use Telephony Server Maintenance Menu in this step. In the Telephony Server Maintenance Menu, select **Reboot** in the SYSTEM CONTROL field (for a dual CPU system, perform the same operation on both systems).

STEP 6: **ACFO/ACFOL/ACFON** - Assignment of Call Forwarding Data
Re-register Call Forwarding data which was saved in STEP 1 to all the operating tenants.

STEP 7: Go to [Setting A Call Forwarding Destination on A Route Basis].

[Setting A Call Forwarding Destination on A Route Basis]

STEP 1: **ARTI/ARTIN** - Assignment of Trunk Application Data
Specify a vacant tenant number to set the data with the ACFO command.
CDN: 112 (ACFO_TN): 1-63

Note: The following are conditions for this step:

- If ACFO_TN is not programmed, the ACFO/ACFOL/ACFON uses the tenant number which the existing call forward destination station belongs to.
- The tenant number to be specified here should be a new tenant number. If a number being used is specified, the existing Call Forwarding feature which is on a tenant basis will be valid (calls will be forwarded to the tenant which the existing call forwarding destination station belongs to).

STEP 2: **ACFO/ACFOL/ACFON** - Assignment of Call Forwarding Data
Assign the Call Forwarding features to the tenant number specified with the ARTI/ARTIN command.
TN: 1-63
CF: Call forwarding service (1-7) **Note 11**

1: Call Forwarding - Busy Line
2: Call Forwarding - Don't Answer
3: Call Forwarding - Intercept
5: Call Forwarding - Logout
7: Call Forwarding - Not Reach **Note 12**

CFI: Call forwarding destination

Attendant Console (ATT)
STN (Station)
TELN **Note 13**

STN: Station number

When CFI=STN (Station) is set with the ACFO command, assign a Station Number of a forwarding destination.

UGN: User Group Number

When CFI=TELN is set with the ACFOL/ACFON command, assign a User Group Number of a forwarding destination.

TELN: Telephone Number

When CFI=TELN is set with the ACFOL/ACFON command, assign a Telephone Number of a forwarding destination.

Note 11: This feature does not support **4: Night Connection Outside** and **6: Call Forwarding - Incomplete Number**.

Note 12: 7: Call Forwarding - Not Reach is only displayed in the ACFON command.

Note 13: TELN is only displayed in the ACFOL/ACFON command.

[Example Data Assignment]

To forward an incoming trunk call to a destination on a route basis, assign the data as shown in the table below. The table shows example data assignments and destinations of incoming trunk calls forwarded by a busy station (Station A) when it receives trunk calls.

Example Data Assignments and Destinations for Call Forwarding-Busy Line

Route	Called Party	ARTI Data ACFO_TN (CDN: 112)	ACFO Data	Call Forwarding Destination
C.O. Line RT=3	TN: 1 (Station A in busy state receives a call)	ACFO_TN=0 (No data)	TN=1 CF=1: Call Forwarding - Busy Line CFI=STN (Station) STN=Station Number of Station B	Station B
Tie Line RT=5	TN: 1 (Station A in busy state receives a call)	ACFO_TN=2	TN=1 CF=1: Call Forwarding - Busy Line CFI=STN (Station) STN=Station Number of Station C	Station C
Tie Line RT=7	TN: 1 (Station A in busy state receives a call)	ACFO_TN=4	TN=4 (No data for Call forwarding destination)	Calls are not forwarded. **Note 14**

Note 14: In the system with the data assignments of the Call Forwarding on a tenant basis (assigned with the ACFO/ACFOL/ACFON command), to disable the data assignments of the Call Forwarding to a specific route, set a vacant tenant number to CDN=112 (ACFO_TN) with the ARTI/ARTIN command. If no Call Forwarding data is set to the vacant tenant with the ACFO/ACFOL/ACFON command, calls are not forwarded.

C-210 Call Recording

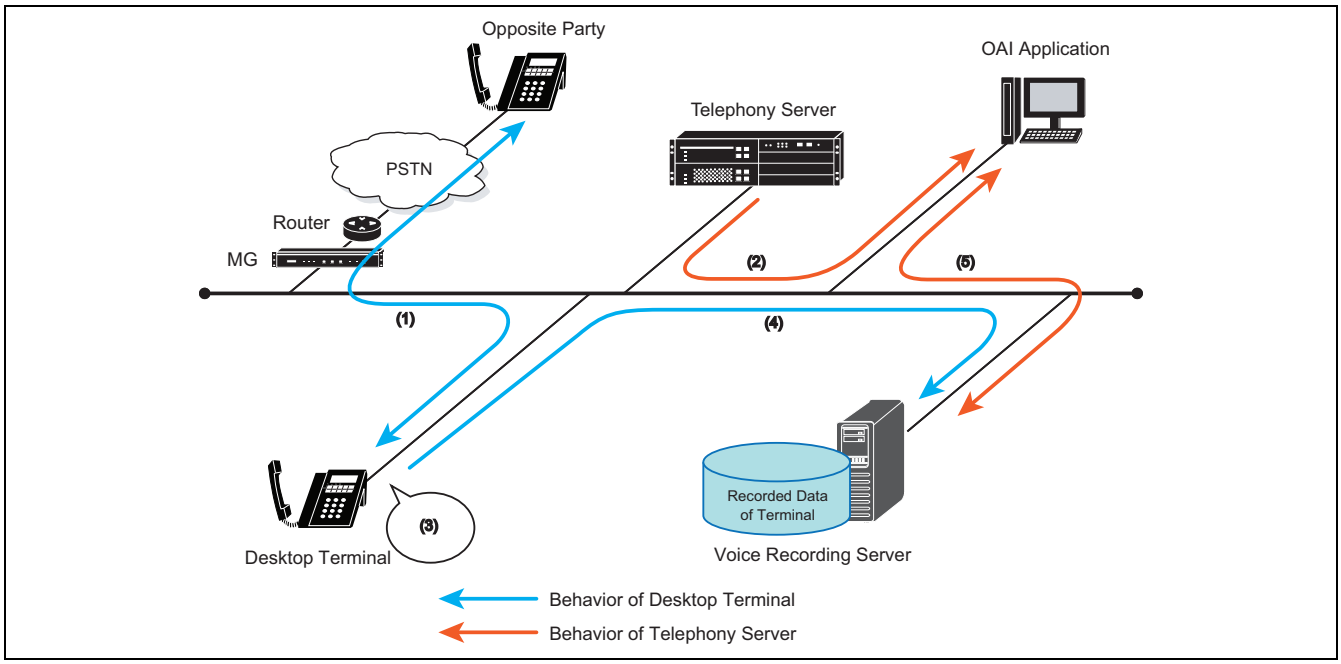
General Description

Note: Available since FP95-115 V5.

You can record the voice call data to a voice recording server (hereafter recording server).
The voice call data is divided into transmitted and received data and sent separately from the terminal to the recording server.
The recording server obtains the call information (such as call start/end time, recording start/end time and call partner information) from the notified RTCP data by the terminal and the recorded sent/received RTP data, and the recorded data can be managed in conjunction with an OAI application.
The methods for starting the call recording that can be chosen on the terminal configuration are as follows:

- Automatic start recording
  A call recording is started automatically when the call is started.
- Manual start recording (**Note 1**)
  A call recording is started when pressing the Call Recording key.

Note 1: For manual start recording, assigning FKY=194 (Call Recording) to Line/Feature keys by the AKYD or AKYD_T command is required.
For automatic start recording, the record key can be used for pausing/resuming the recording if assigned. Depending on the recording server, the recording data during pausing time may be no voice.



(1) A call is started.
(2) Telephony Server sends call information of the terminal to an OAI application by using SMFN.
(3) A recording is started on the terminal.
(4) The terminal sends voice recorded data and the call information to the recording server.

(5) The recorded data is managed with the OAI application.

Operating Procedure

There are two operations as follows:

<Automatic start recording>

<Manual start recording>

<Automatic start recording>

A call recording is started automatically when the call is started. The other use is as follows:

Example 1: Skipping a call recording for a call and then using call recording for the next call.

STEP 1: Press the Call Recording key before a call.
The LED of the Call Recording key lights in green (the recording standby state) is turned off (the recording stop state).
STEP 2: Make a call (or answer an incoming call).
The LED of the Call Recording key is still off.
STEP 3: End the call.
Green LED light is turned on (the recording standby state).
STEP 4: Make a call (or answer an incoming call).
Red LED light is turned on (the recording start state). The call recording is started.
STEP 5: End the call.
Green LED light is turned on (the recording pause state). The call recording stops.

Example 2: Pausing and resuming operation during recording.

STEP 1: Press the Call Recording key during the call.
Green LED light is turned on (the recording pause state). The call recording is paused.

Note: When the Call Recording key is assigned, the recording is paused/resumed by pressing the Call Recording key even if the automatic start recording is selected.

STEP 2: Press the Call Recording key.
Red LED light is turned on (the recording resuming state). The call recording is resumed.

<Manual start recording>

A call recording is started when pressing the Call Recording key.

STEP 1: Press the Call Recording key during a call.
Red LED light is turned on (the recording start state). The call recording is started.
STEP 2: Press the Call Recording key.

Green LED light is turned on (the recording pause state). The call recording is paused.
STEP 3: Press the Call Recording key again.
Red LED light is turned on (the recording resuming state). The call recording is resumed.
STEP 4: End the call.
The LED is off (the recording stop state). The call recording stops.

Note: The call recording stops regardless of the Call Recording key being assigned or not.

Example: When handling the call recording from the beginning of the call (similar behavior to automatic start recording).

STEP 1: Press the Call Recording key before a call.
Green LED light is turned on (the recording standby state).
STEP 2: Make a call (or answering an incoming call).
Red LED light is turned on (the recording start state). The call recording is started.
STEP 3: Press the Call Recording key during the call recording.
Green LED light is turned on (the recording pause state). The call recording is paused.
STEP 4: Press the Call Recording key.
Red LED light is turned on (the recording resuming state). The call recording is resumed.
STEP 5: End the call.
The LED is off (the recording stop state). The call recording stops.

Service Conditions

<Important Notices>

1. The following firmware version is required on each terminal for this feature.

DT830G/DT830	:	FW 5.2.0.0 or later
DT820	:	FW 3.2.0.0 or later

2. Assigning the Telephone Number to stations that use this feature is required.
3. When the call recording failed with any errors between terminals and the recording server, Telephony Server will not recognize or acknowledge the failure.
4. A number of call recording licenses that is greater than a number of stations allowed a call recording is required for releasing the restriction of this feature by the ATSRL/ATSRN commands.
   When a number of call recording licenses is less than a number of stations, the restriction of this feature cannot be released.
   Restarting the terminal is required in the following cases:
   - After the restriction of this feature is released by the ATSRL/ATSRN commands.
   - When changing the restriction data by the ATSRL/ATSRN commands during the terminal in operation.
5. The call information (such as call start/end time, recording start/end time and call partner information) is notified from Telephony Server by using SMFN of OAI. The conditions of SMFN notification for each service follows the conditions of OAI.

6. Handling of SMFN of OAI and the information of recorded data on the terminal depends on the specification of a recording server.

<Interactions with Other Features>

1. This feature cannot be used with USB Call Recording.
2. The conditions for PEER TO PEER BANDWIDTH CONTROL [P-75] are as follows:

   (a) A speech signal between a terminal and a recording server is not included in the calculation of used bandwidth. Also a bandwidth restriction between a terminal and a recording server cannot be controlled.

   (b) The payload type configured by system data is applied to the call only. The payload type between a terminal and a recording server uses one of the following payload types configured by the terminal side.

   LinearPCM (8kHz, 16bit), LinearPCM (16kHz, 16bit) (**Note 3**), G.711μ, G.711A

   Note 3: LinearPCM (16kHz, 16bit) is invalid for DT820.

3. The conditions for VoIP ENCRYPTION [V-27] are as follows:

   (a) Using this feature, a control signal between a call recording activated terminal and Telephony Server is encrypted.

   (b) Using this feature when an RTP packet encryption is activated, the RTP packet between the call recording activated terminal and an opposite party is encrypted, but the RTP packet between the terminal and a recording server cannot be encrypted.

4. For RTP INFORMATION OUTPUT [R-58], the information between a terminal and a recording server is not included in the output call log information. The information only between the terminal and an opposite party is included in it.

5. The conditions for MONITORING - ACD SUPERVISOR - ACD [M-28A] are as follows:

   (a) When a call recording activated terminal is an agent's terminal:
   When Silent Monitor (Multi-path Monitor) is enabled, the call only between the agent terminal and an opposite party is recorded. The voice path of monitoring for a supervisor is not recorded.

   (b) When a call recording activated terminal is a supervisor's terminal:
   When Silent Monitor (Multi-path Monitor) is enabled, the monitoring received call between the agent terminal and an opposite party is recorded.

6. For PAD/EC Control IP/PHC Support, a voice that has applied PAD/EC control settings is recorded.
7. The conditions for ToS Control are as follows:

   (a) A ToS value that is used in SIP, RTP and RTCP packets sent from a terminal to recording equipment such as a recording server can be set in the terminal configuration. The value cannot be set on the Telephony Server side.

   (b) The ToS value in the packets sent from the recording equipment depends on the specification of the recording equipment.

8. The conditions for SURVIVABLE REMOTE MGC [S-141] are as follows:

   (a) This feature is available when a terminal is registered to SR-MGC. Communicating between the terminal and a recording server is required in this case.

   (b) Assigning the call recording licenses to the Telephony Server side beforehand and releasing the restriction by the ATSRL/ATSRN commands are required. The call recording license is not required on the SR-MGC side.

   (c) To change the data by the ATSRL/ATSRN commands is not available for SR-MGC. For this reason, the restriction of this feature cannot be enabled/disabled while the terminal is registered to SR-MGC.

   (d) OAI-SMFN notification during operating SR-MGC depends on the operation on the OAI application side. (Monitor request in the SR-MGC is required.)
   Other conditions of OAI follow the conditions on “SR-MGC for OAI” in Data Programming Manual - OAI.

9. The conditions for Geographic Redundancy are as follows:

   (a) This feature is available when a terminal is registered to GR-Node or SR-Node. Communicating between the terminal and a recording server is required in this case.

   (b) For Primary Node, a number of call recording licenses that is greater than a number of stations using call recording is required.

   (c) OAI-SMFN notification on GR-Node or SR-Node depends on the operation on the OAI application side. (Monitor request in the backup side is required.)
   Other conditions of OAI follow “Interactions to Other Features” on “Geographic Redundancy in Stand Alone OAI System” or “Geographic Redundancy in OAI System with FCCS Networking via IP” in Data Programming Manual - OAI

<Hardware/Software>

1. A Call Recording key cannot be assigned to DSS Console or 8LK Unit.
2. The conditions for a recording server are as follows:

   (a) A redundant recording server is available.

   (b) As the following ports are consumed on a terminal in the communication with the recording server, the Firewall settings must be configured carefully.

Recording Session	:	One port (Default: 5080) The port number follows the terminal settings.
Recorded Media Stream	:	96 ports (Default: 21050~21146) The port number follows the terminal settings.

(c) When a destination port number of recorded Tx data and recorded Rx data from the terminal to the recording server is the same, treat both data separately on the recording server side.

Note: Distinguishing both data is performed as follows:
- When the sender IP address and the IP address set as CNAME in RTCP are the same address
  = Recorded Tx data
- When the sender IP address and the IP address set as CNAME in RTCP are the different address
  = Recorded Rx data

(d) Call information that a recording server accepted is as follows:

- Call start time
- Call end time
- Recording start time
- Recording stop time
- Recorded TX data
- Recorded RX data
- Call partner number information **Note 5**
- Call partner name information **Note 5**
- Phonetic transcription in kana with a call partner name **Note 5**
- Company name information **Note 5**
- Department information **Note 5**

Note 5: The recording server may not be notified of this information depending on the system or the terminal operation.

3. The conditions for a terminal are as follows:

(a) A recording target is from the start to the end of the call session including held.

(b) BGM is not recorded even if "All" is selected at "Reg Segments" in a terminal configuration.

(c) For a terminal with Bluetooth Connection Adapter (BCA-Z UNIT), the call on the mobile phone side is not recorded when using a mobile phone provided with Bluetooth functions.

(d) When upgrading the terminal firmware version from 5.0.0.0 or earlier to 5.1.0.0 or later for DT830 or the terminal firmware version from 3.0.0.0 or earlier to 3.1.0.0 for DT820, the Rec Tone value in the terminal configuration is back to the default setting and the behavior of the Rec Icon displaying may be irregular.

(e) When using this feature, the call information sent from a terminal to a recording server is not encrypted. To prevent an information leak to a third person, install and operate the terminal and the recording server in secured guaranteed network. If not, it is not supported.

(f) The recording works even if RTP in a call session on Dynamic Mode (SIP control) is already encrypted. However, the data sent to a recording server is not encrypted.

(g) When a terminal cannot receive an RTP data from an opposite party because of network failure and so on, the terminal sends the received RTP data (Rx) with no voice to a recording server. The transferred RTP data (Tx) is sent to the recording server even if network failure occurs.

(h) When a terminal cannot receive an RTCP Reception Report (**Note 9**), displaying of online monitoring is still default value ("---"). The follows are likely to be factors.
- RTCP packets sent from a recording server cannot reach to the terminal.
- No packets of RTP or RTCP sent from the terminal reach to the recording server.

Note 9: An RTCP Reception Report is one of RTCP packets.

(j) It takes several seconds to be recording ready of this feature after a terminal starts up.

(k) When an event notified from Telephony Server is the same with the event information stored in a recording server, the recording may not work.

(l) For Dynamic mode (with SIP control), it may take several times to complete sending a recording data after operation of recording stop on a terminal because of recording session control with a recording server. When the duration from recording stop to recording start is shorter, the recording may not work.

(m) A volume control change operation on the terminal is not reflected to the recorded contents.

(n) When using recording servers in redundant configuration, a terminal sends the recorded data to both recording servers.

(o) Playing recorded data on a recording server is not supported. For playing the recorded data, refer to a recording server manual.

(p) For DT820, assign a Call Recording key to Line/Feature keys by the AKYD or AKYD_T command because displaying Rec Icon to LCD is not supported.

<Interactions with Networks>

1. The conditions for FCCS are as follows:

(a) For NCN, a number of call recording licenses that is greater than a number of stations using a call recording is required.

(b) For LN, the call recording license is not required.

2. LOCATION DIVERSITY [L-55] is not supported.

<Conditions Specific to This Feature>

1. The conditions in UMGi system are as follows:

(a) This feature is available for CCN or RN in Survival Mode that is controlling a terminal (because of a network failure, etc). Communicating between the terminals and a recording server is required in this case.

(b) For CCN, a number of call recording licenses that is greater than a number of stations using a call recording in the all RN is required.

(c) For RN, the call recording license is not required. It follows a number of call recording licenses in CCN.

(d) OAI-SMFN notification does not work in survival mode (based on the OAI conditions).

2. The following tones heard on a terminal side are recorded:
   - PBX tone (such as DT, RBT, BT and Internal Music-on-hold)
   - RTP warning tone **Note 5**
   - Short tone when each service is used
   - PB tone when a terminal receives DTMF **Note 6**

   It is possible that the tones above are not recorded by using "Rec Segments" in the terminal configuration. However, the tones heard during a call (RTP warning tone, short tone and PB tone when a terminal receives DTMF) are recorded even if the "Talk Segments" in "Rec Segments" is selected in the terminal configuration.
   Also, a part of the PBX tone may be included in recorded data depending on a timing of switching a call.

   Note 5: Available when RTP warning tone is enabled.

   Note 6: Available when DTMF playback is enabled.

3. For conferences (**Note 7**), the conference data is recorded to each call recording that terminal joined the conference.

   Note 7: The following conferences are indicated:
   - Conference - 3 party with conference trunk (CFT, VS32)
   - Conference - over 4 party with CFT circuit card and VS32

Programming

For the settings on a terminal side, see “Call Recording” on Chapter 3 in Peripheral Equipment Description (Multi-Line Telephone).

Note: Assign the station data and Telephone Number to the terminal allowing Call Recording before the following settings.

STEP 1: **ATSRL/ATSRN** - Release the restriction of Call Recording for Telephone Number. The assigned data is deleted automatically when the Telephone Number is deleted by the ALGSL/ALGSN, AUPDN and ASPTL/ASPTN commands.
UGN=User Group Number
TELN=Telephone Number (Up to 16 digits)
SVN=1 (Call Recording)
STATUS=1: Used

Note: When a number of call recording licenses is less than a number of stations, the restriction of this feature cannot be released.

When using the recording start manually, the following step is also required. For using the recording start automatically, the Call Recording key can be used for pausing/resuming the recording if assigned by the following step.

STEP 2: **AKYD/AKYD_T** - Assign a Call Recording key to the Line/Feature keys of a terminal.
KYN=Key Number (1 to 24)
KYI=1 (Function Key)
FKY=194 (Call Recording)

Note: Be sure to assign this key data to Line/Feature keys within KYN=1 to 24. For DSS keys, assign this key data within KYN=1 to 16. However, this key data cannot be assigned to the optional line key units (such as 8LK Unit or DSS console).

D-2 Dial Access To Attendant

General Description

This feature allows users to access the ATTENDANT CONSOLE [A-3] by dialing the Operator Call Code.

Operating Procedure

To access the Attendant:

1. The user dials the Operator Call Code. Receives Ringback Tone.
2. The ATND Lamp for the INCOMING CALL IDENTIFICATION [I-2] flashes. Attendant answers the call.

Service Conditions

1. The Operator Call Code will be maximum six digits. Normally "0" should be assigned as the Operator Call Code.
2. All Stations, including fully restricted Stations, can originate a call to the Attendant.
3. If TENANT SERVICE [T-12] is provided, each Tenant Group may access their own ATTENDANT CONSOLE [A-3] via the same code.
4. This feature allows any user or dial repeating TIE Line Trunk to reach the ATTENDANT CONSOLE [A-3] by dialing an Operator Call Code, usually but not necessarily a "0". The ATTENDANT CONSOLE [A-3] may complete these calls to Trunk facilities if allowed by the Station's CLASS OF SERVICE [C-15]. There is no restriction on the Attendant Console [A-3] extending a Station Call to another Station.
5. The Fusion Point Code (FPC) of the Node which accommodates the Called Attendant Console must be assigned after System Data assignment.

Interactions

1. Stations will terminate to the Attendant Console located in the same Tenant when the Station dials "0".
2. Stations in Tenant A will terminate to the Attendant Console of Tenant B if Tenant A is not equipped with an Attendant Console. Tenant B's Attendant Console must be programmed to accept calls from Tenant A.
3. Stations may dial the ASAT Number to terminate at a specific Attendant Console. See INDIVIDUAL ATTENDANT ACCESS [I-6].
4. When the system is in the Night mode, a Station dialing "0" can be answered by the assigned ATTENDANT NIGHT TRANSFER [A-19] Station, if provided. However, when the ASAT Number is dialed in the Night mode, Reorder Tone is returned. No connection to the ATTENDANT NIGHT TRANSFER [A-19] is allowed.

Programming

STEP 1: **ANPD** - Reserve a Number Level for Feature Access and Cancel. Select Normal Service (N) or Hooking Service (H) from the combo box for Connection Indexes. Assign the NND in accordance with a predetermined numbering scheme. Usually Level 0 is assigned for the Attendant Console.

STEP 2: **ASPA** - For Access Code "0", assign SRV = "SSC: Service Code" SID = 2 for Operator Call. Assign for a Connection Index (CI) of "N: Normal" and "H: Hooking".

STEP 3: **ATNR** - Allow Tenant-to-Tenant Connection for Inter and Intra-Tenant Connections. TRI = 0 and TRI = 4.

STEP 4: **ASAT** - If required, assign a specific number to the Attendant Console. This number must be equivalent to a Station Number.

FCCS Programming

This data setting describes only the additional or replacement commands used to activate FCCS Services. When used as an FCCS feature, these commands replace the non-network commands (for example, AAED is replaced by AAEDN). Note that the data setting for FCCS Service varies depending on the Node.

STEP 1: **ASPAL** - This data must be assigned to the Node calling the Attendant.

SRV: "SSC: Service Code"

SID: 2 = Operator Call

FPC: FPC of the Node that accommodates the Called Attendant Console

D-3 Dictation Access

General Description

This feature permits users to access customer provided Dictation Equipment, and in some instances allows them to maintain telephone dial control of normal dictation System features.

Operating Procedure

1. Dial the DICTATION ACCESS Code.
2. Proceed according to the dictation instructions provided by your organization.

Service Conditions

1. One Trunk Circuit is required for each DICTATION ACCESS.
2. Dictation Equipment must be equipped to receive DTMF Tone Signals if dial control is desired.
3. Dictation Equipment must be customer provided.
4. Dictation Trunks can be used with Rotary Dial Signals.
5. The dictation Trunk can be accessed from a Station, Attendant, TIE Line, or REMOTE ACCESS [R-2] Trunk.
6. OUTGOING TRUNK QUEUING [O-2]: If OUTGOING TRUNK QUEUING is provided and the user dials a busy dictation Trunk Group, the Station is placed in queue and is called back when the group becomes available.
7. THREE-WAY CALLING [T-2]: An attempt to access the DICTATION ACCESS Trunk from a Two-party Connection is allowed.
8. FULLY RESTRICTED STATION: With this feature assigned, only Station-to-Station and ATTENDANT CONSOLE [A-3] dial “0” calls are permitted. Any attempt to access the DICTATION ACCESS Trunk is routed to Reorder Tone.
9. MISCELLANEOUS TRUNK RESTRICTIONS [M-3]: This feature prevents certain Stations from using Miscellaneous Service Trunks. Any attempt to access the DICTATION ACCESS Trunk from a Station with this feature assigned is routed to Reorder Tone.

Interactions

1. OUTGOING TRUNK QUEUING [O-2]: If OUTGOING TRUNK QUEUING [O-2] is provided and the user dials a busy Trunk Route, the Station is placed in queue and is called back when a Trunk in the Route becomes available. If the Trunk Route is part of a ROUTE ADVANCE [R-6] Group, all Trunks in the ROUTE ADVANCE [R-6] Group must be busy before queuing takes place. OUTGOING TRUNK QUEUING [O-2] is available on DICTATION ACCESS.
2. THREE-WAY CALLING [T-2]: An attempt to Conference the DICTATION ACCESS Trunk from a Two-party Connection may be allowed. This depends on assignments in ARSC.
3. MISCELLANEOUS TRUNK RESTRICTION [M-3]: This feature prevents certain Stations via Route CLASS OF SERVICE - INDIVIDUAL [C-15] from using various Routes in the system. If a Station is restricted, any attempt by a Station to access the DICTATION ACCESS results in Reorder Tone.

Note: This feature may be installed on the Station side of the switch. In that case, the PA-16LC may be used. However, when installed on the Station side, Route Restrictions are not available to the system programmer. All Stations within the system (Tenant Restrictions excluded), as well as Stations in other Telephony Servers connected via a TIE Line, will have access to the Dictation Equipment.

4. If the Station side is used to connect DICTATION ACCESS, STATION HUNTING [S-7, 8, 9] will provide capabilities to connect multiple Dictation Machines to the system.

Programming

STEP 1: **ANPD** - Reserve a Number Level for Station Access. Select Normal Service (N) or Hooking Service (H) from the combo box for Connection Indexes. Assign NND in accordance with a predetermined Numbering Plan.

STEP 2: **ASPA** - Assign an Access Code to the DICTATION ACCESS Trunk, SRV = "OGC: Outgoing Call". Correlate Access Code to Route. One Trunk per Announcement Route. Assign CI: of "N: Normal" and "H: Hooking".

STEP 3: **ARTD** - Assign the DICTATION ACCESS Route as follows:

RT:1

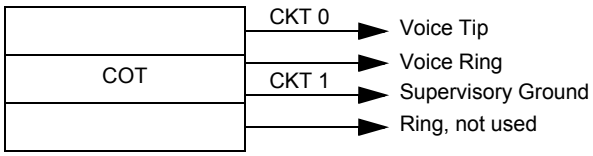
1-OSGS	:2	2-0NSG	:3	5-TF	:1	6-TCL	:10	7-L/T	:1
8-RLP	:2	44-PRV	:1						

Note: To change the data for existing route (RT), initialize the circuit card.

STEP 4: **ATRK** - Assign the LENS, DICTATION ACCESS Trunk Route Number, Trunk Number, and Tenant Number.

STEP 5: **MBTK** - Assign the Make Idle status to the DICTATION ACCESS Trunk.

Note: If supervision is required, use the first two circuits of the COT circuit card. Move the switch on COT to PGT.



STEP 6: **ARRC** - If TIE Line or Remote Access Connection to DICTATION ACCESS Trunk is required, allow Trunk-to-Trunk Connection using ARI-D-RES, Direct Connection.

Note: The DICTATION ACCESS Equipment may be installed on the Line Side of the System. The connection would then be considered STATION-TO-STATION CALLING [S-11].

STEP 7: **ARSC** - Allow RRI 3 to the RSCs allowed access to Dictation.

D-5 Digital Display - Station

General Description

This feature provides the Attendant with a visual display (via the ATTENDANT CONSOLE [A-3]) of the Telephone Number, its Trunk Route Restriction, CLASS OF SERVICE - INDIVIDUAL [C-15] and TENANT SERVICE [T-12] Number during Attendant-to-Station Connection.

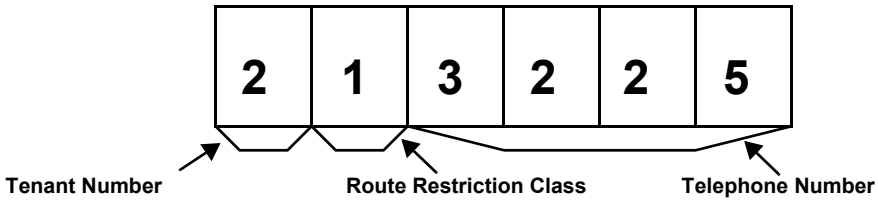
Operating Procedure

The DIGITAL DISPLAY STATION is automatically activated whenever the Attendant answers an Incoming Call from a Station, or originates a call to a Station.

Service Conditions

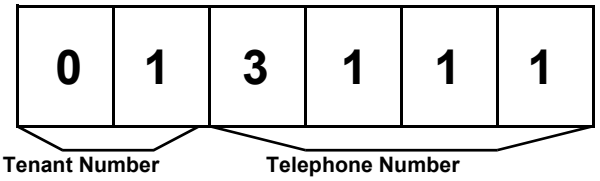
1. The Telephone Number is displayed on the left side of the ATTENDANT CONSOLE [A-3] digital display. Trunk Route Restriction CLASS OF SERVICE - INDIVIDUAL [C-15] and TENANT SERVICE [T-12] number are simultaneously displayed on the left. See Example 1 below.

   Example 1: Single-digit Trunk Route Restriction CLASS OF SERVICE - INDIVIDUAL [C-15] and TENANT SERVICE [T-12] Number (four-digit Telephone Number).

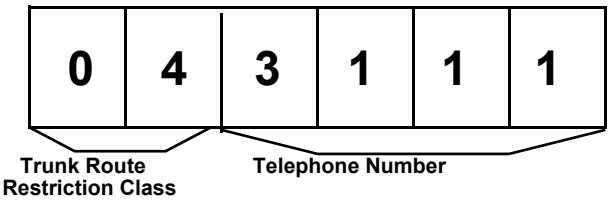


2. If the Trunk Route Restriction Class and/or TENANT SERVICE [T-12] Number are two digits, only the Telephone Number may be displayed. Refer to Example 2.
3. If a Station dialed by the Attendant is busy or in the LINE LOCKOUT [L-3] mode, only the Telephone Number will be displayed.
4. The ATTENDANT CONSOLE [A-3] digital display is limited to a six-digit numeric display. Since the Class of Service and Trunk routing display is in numeric form, a list must be provided to the Attendant explaining the restrictions of each Class of Service Number and Trunk Identification Number.
5. CALL FORWARDING [C-2, 3, 5]: When this feature is activated by (or for) the Called Station, the Terminating Telephone Number is indicated on the digital display rather than the Called Telephone Number when the Attendant extends a call.
6. STATION HUNTING [S-7, 8, 9]: When this feature is applied because of the Called Station being busy and the Station belonging to a Hunt Group, the Terminating Telephone Number is indicated on the digital display rather than the Called Telephone Number when the Attendant extends a call.
7. In the case of a three or four-digit Station-to-Station Numbering Plan:
   a.) One digit is used to identify the Tenant (in the case where two digits are required for Class of Service, the Tenant Number is not identified).
   b.) One or two digits are used to identify the Station Class of Service.
   c.) Four digits are used to display the Telephone Number.
8. The DIGITAL DISPLAY - STATION feature will occur whenever the Attendant:
   a.) Extends an Incoming Call to a user.
   b.) Originates a call to a user.

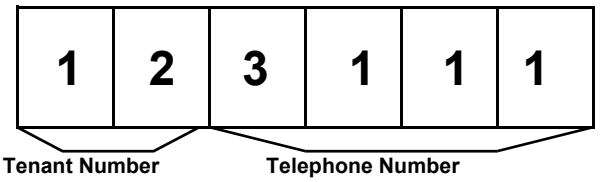
c.) Answers a Recall or Call Forwarded Call.
Example 2: Two-digit TENANT SERVICE [T-12] Number, four-digit Telephone Number.



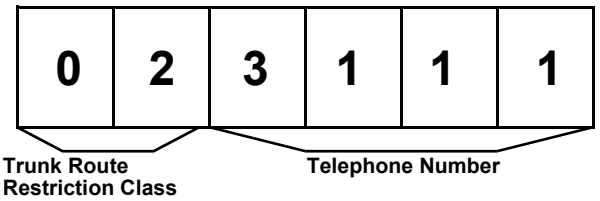
Example 3: Two-digit Trunk Route Restriction Class, four-digit Telephone Number.



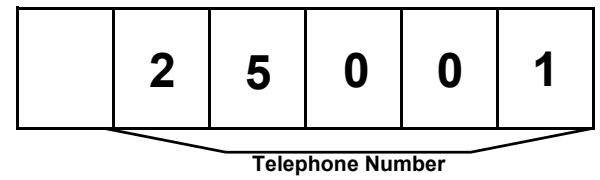
Example 4: Single digit TENANT SERVICE [T-12] number, five-digit Telephone Number.



Example 5: Single digit Trunk Route Restriction Class, five-digit Telephone Number.



Example 6: When five-digit Station Numbering is used, only the Telephone Number is displayed.



9. For the FCCS Network, when a Telephone Number has been assigned for a Station, it is displayed on the Attendant Console. However, if a Telephone Number is seven digits or more, the Physical Station Number is displayed.

Programming

ASYD - SYS1, Index 82, Bit 0 to Bit 3. Assign a hexadecimal value according to the choice.
HEXADECIMAL - Station information

0: TN + CLS + Four-digit Station Number
1: TN + Four-digit Station Number
2: RSC + Four-digit Station Number
3: TN + Five-digit Station Number
4: RSC + Five-digit Station Number
5: Station Number

Note: This above only works on the PA-CS33 circuit card if SW 16 Element 2 is OFF (Desk Console Expanded LCD is not available).
Element 2 is ON (Desk console Expanded LCD is available) displays as follows:
TN OO CLS OO
NO: OOOO
Element 2 is Off, ASYD 82 takes effect.

D-6 Digital Display - Trunk

General Description

This feature provides the Attendant with a visual indication of Incoming and Outgoing Trunk Calls on the ATTENDANT CONSOLE [A-3]. Trunk Identification Number, Trunk Route Number, and TENANT SERVICE[T-12] Number, or Central Office Trunk Code, are displayed in a numerical readout.

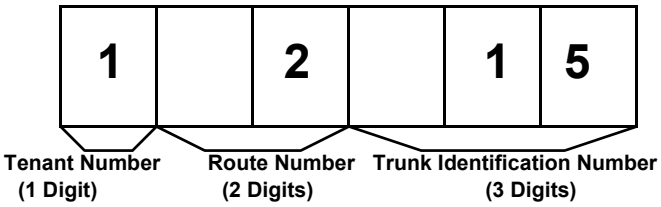
Operating Procedure

The DIGITAL DISPLAY TRUNK is automatically activated whenever the Attendant answers an Incoming Trunk Call or originates a Trunk Call.

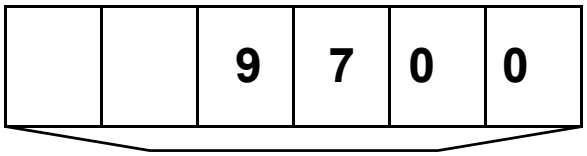
Service Conditions

1. By pressing the Loop key of a call that is in the ATTENDANT CAMP-ON [A-1] mode and using the DEST key, the Attendant can determine both Trunk and Telephone Number onto which the call is camped (Telephone Number and Trunk Number cannot be displayed simultaneously).
2. The Trunk Identification Number is displayed on the right side of the ATTENDANT CONSOLE [A-3] display while TENANT SERVICE [T-12] and Trunk Route Numbers are simultaneously displayed on the left.

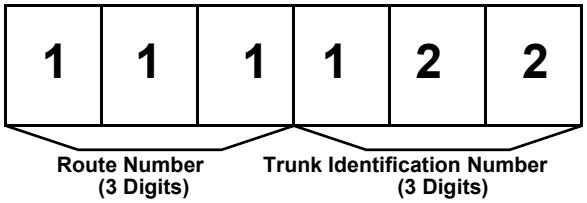
   Example 1:



   Example 2:



   Example 3:



3. If the Trunk Route Number is three digits, the Trunk Identification Number and Trunk Route Number are displayed.
4. The ATTENDANT CONSOLE [A-3] digital display is limited to a six-digit numeric display. Since the Class of Service and Trunk routing display is in numeric form, a list must be provided to the Attendant explaining the restrictions of each Class of Service Number and Trunk Identification Number.

5. The DIGITAL DISPLAY - TRUNK feature displays Trunk Routing information or a four-digit Central Office Line Number. The method used can be optionally selected on a per Trunk Group basis.
6. The DIGITAL DISPLAY - TRUNK feature will occur whenever the Attendant answers/originates:
   a.) An Incoming Central Office (DID, Bothway, Oneway, FX, WATS) type call.
   b.) An Incoming TIE Line Call.
   c.) An Intercept Call.
   d.) An Outgoing Central Office (DOD, WATS, FX) Type Call.
   e.) An Outgoing TIE Line Call or re-enters a Held Trunk Call.
7. An Attendant Console also indicates the Central Office Code assigned by System Data at each Node in the FCCS Network.
8. Regardless of the value of the data assignment DPLY (CND:50) for the ARTD command, a received Telephone Number will be displayed.
9. Trunk Number cannot be displayed during an incoming call.

Programming

STEP 1: **ASYD** - SYS1, Index 68, Bit 1. Type of INDIVIDUAL TRUNK ACCESS, 0/1: Route and Trunk Number/Central Office Code.

Note: If SYS1, Index 68, Bit 1 is assigned for the Individual Trunk Access to be the ACOC Code, then all of the Trunks in all of the Routes in the system must have the ACOC Code assigned.

STEP 2: **ARTD** - In CND 25:R/L, if data "0" is assigned, then Tenant, Route Number, and Trunk Number are displayed. If data "1" is assigned, then Central Office Code is displayed. ACOC must be assigned.

Note: For display purposes only, Route and Trunk Numbers and Central Office Codes may be used simultaneously in the same system and are applied on a per Route basis, CDN 25:R/L.

Note: To change the data for existing route (RT), initialize or reboot the trunk device such as a circuit card or MG.

Route and Trunk Number

STEP 3: **ATRK** - The route and Trunk Number assigned here provides the system with specific information on types of Routes and their associated Trunks, LENS, number of Tenants, and any CLASS OF SERVICE - INDIVIDUAL [C-15] Route Restriction information.

Central Office Code

STEP 4: **ACOC** - Each Trunk must be assigned a Central Office Code.

FCCS Programming

This data setting describes only the additional or replacement commands used to activate FCCS Services. When used as an FCCS feature, these commands replace the non-network commands (for example, AAED is replaced by AAEDN). Note that the data setting for FCCS Service varies depending on the Node. **Note 1**

STEP 1: **ARTDN** - In CND 25:R/L, if data "0" is assigned, then Tenant, Route Number, and Trunk Number are displayed. If data "1" is assigned, then Central Office Code is displayed. ACOC must be assigned.

STEP 2: **ACOCN** - Each Trunk must be assigned a Central Office Code.

Note 1: When using ASYDL or ASYDN to program an FCCS feature, the basic programming of ASYD is required in addition to ASYDL or ASYDN.

D-7 Direct-In Termination (DIT)

General Description

This feature automatically routes Incoming Public Network Calls directly to a Pre-selected Station without Attendant assistance. The call can then be processed by the Called Party. THREE-WAY CALLING [T-2], CALL TRANSFER [C-10], etc., are handled in the same manner as any Normal Trunk Call.

Operating Procedure

The Calling Party, outside the system, dials the desired number as usual. However, the call is answered directly at a predetermined Station, bypassing the ATTENDANT CONSOLE [A-3].

Service Conditions

1. Any groups of Trunks can be used, up to the maximum capacity of the System.
2. Bothway Trunks can be used for DIRECT-IN TERMINATION, but incoming-only Trunks are recommended.
3. It is not necessary to have special DID Trunk Connections to the Public Network as is required with DIRECT INWARD DIALING [D-8]. However, care should be exercised in the application of Central Office Trunk Hunting.
4. If the DIT Station is busy, the Incoming Call is kept ringing until the DIT Station becomes idle, or forwarded to the ATTENDANT CONSOLE [A-3].
5. If there is no answer at a Station, the Calling Party will continuously receive Ringback Tone. There is no AUTOMATIC RECALL [A-8] to the Attendant in this case.
6. If the Station is assigned to a STATION HUNTING [S-7, 8, 9] Group, the Incoming Call will follow the Hunt Group Station assignment. If the Station is in a CALL FORWARDING [C-2, 3, 5] mode, the Incoming Call is connected to the CALL FORWARD Target Station.
7. This feature is normally used where Direct-in Service is required on a limited basis. The number of Stations serviced is effectively limited to the number of Trunks available for DIRECT-IN TERMINATION.
8. DIRECT-IN TERMINATION is mutually exclusive with NIGHT CONNECTION - FIXED [N-1].
9. DIT Trunks can be programmed to terminate elsewhere in the Night mode.
10. When the system is in the Night mode of operation and the DIRECT-IN TERMINATION (DIT) Station is busy, the call will divert to TAS if provided, otherwise, the call will wait on the busy Station until it becomes idle.
11. More than one Trunk can be programmed for DIRECT-IN TERMINATION to a particular Station Line.
12. The DIT Call can terminate to a DO NOT DISTURB -D [D-11D] Station.
13. When a DIT Call terminates to a restricted Station or a Station setting CALL FORWARDING - ALL CALLS [C-5]/BUSY LINE [C-2], or MULTIPLE CALL FORWARDING - ALL CALLS [M-44]/BUSY LINE [M-24] to a restricted Station, the call is routed to the Attendant and terminates at the ICPT key on the ATTENDANT CONSOLE [A-3].

    Note: This service can be activated in System Data.

14. After a call is answered, it can be processed by the Called Station like any normal Trunk Call.

15. Only the Telephone Number is available for the Destination Station assignment when activating this feature via FCCS.
16. When activating this feature via the FCCS Network, this feature is activated by data in the LDM of the Node accommodating the Terminated COT.
17. When activating this feature via the FCCS Network, when the Telephone Number of the Destination Station is deleted:
    a.) When the Attendant Console is used, the DIT Call terminates to the Attendant Console.
    b.) When the Attendant Console is not used, or if the Attendant Console is in Night mode, the caller hears Ringback Tone.
18. When activating this feature via the FCCS Network, a DIT Call cannot be queued while all the FCCS Links are busy. Calling Party receives Ringback Tone at that time and after the FCCS Link is restored to idle.
19. When activating this feature via the FCCS Network, the different destination for Central Office Incoming Call (Ringdown) cannot be set at each Tenant Group.

Interactions

1. Night DIT Service is NIGHT CONNECTION - FIXED [N-1]. The NIGHT CONNECTION - FIXED/FLEXIBLE [N-1, 2] feature can be used to simulate a DIT application.
2. When using NIGHT CONNECTION - FIXED/FLEXIBLE [N-1, 2], if the Station is busy, the call will divert to TRUNK ANSWER ANY STATION [T-8], if provided. If not provided, the call will queue at the busy Station until it becomes idle.
3. The DIRECT-IN TERMINATION Trunk is limited to DDD, FX, WATS, and CCSA.
4. All CALL FORWARDING [C-2, 3, 5] features may be used in conjunction with DIRECT-IN TERMINATION. In the case of CALL FORWARDING - DON'T ANSWER [C-3], the call must physically be ringing the DIT Station. Call Forwarding is only available from a ground start DIT Trunk.
5. STATION HUNTING [S-7, 8, 9] and UNIFORM CALL DISTRIBUTION (UCD) [U-1] may be used with DIRECT-IN TERMINATION.

Programming

STEP 1: **ASYD** - SYS2, Index 11, Bit 7. Will the DIT Call to a busy Station be routed to the Attendant Console, or will it queue to the busy Station? 0/1: Attendant Console/Queue. Assign on a per Tenant basis. If the call goes to the Attendant Console, it will ring in on the BUSY or ICPT key.

SYS1, Index 147. DIT Supervisory Timer for a Busy Station. How often will the CPU look at the Idle/Busy status of the Station for rerouting the DIT Call to the destination? This index works when SYS2, Index 11, Bit 7 is 1.

STEP 2: **ARTD** - Assign the DIRECT-IN TERMINATION Route as shown below:

GROUND START:

RT: 1

1-OSGS	:2	2-ONSG	:3	3-ISGS	:1	4-INSG	:3	5-TF	:2
6-TCL	:1	7-L/T	:1	8-RLP	:2	15-LSG	:1		

LOOP START:

RT:1

1-OSGS	:2	2-ONSG	:3	3-ISGS	:1	4-INSG	:3	5-TF	:2
6-TCL	:1	7-L/T	:1	8-RLP	:2				

Note: To change the data for existing route (RT), initialize or reboot the trunk device such as a circuit card or MG.

STEP 3: **ATRK** - Assign the LENS, DIT Trunk Route Number, Trunk Number, and Tenant Number.

STEP 4: **MBTK** - Assign the Make Idle status to the DIT Trunk.

STEP 5: **ARSC** - The DIT Target Station receiving the call must be assigned a Route Restriction Class (RSC), allowing the Station to be connected to the Trunk. The DIT Route must be allowed via Route Restriction Indexes (RRI) 0 and 1.

STEP 6: **ACFR** - Designate the type of Incoming Call Class Indexes (CCI) that require a Transfer Service Feature Index (TSFI) = 3, DIRECT-IN TERMINATION. The type of CCI must match CDN 6 of the DIT Route. For Night DIT (NIGHT CONNECTION - FIXED [N-1]), allow the CCI a TSFI = 2, NIGHT CONNECTION - FIXED [N-1].

For Day and Night DIT Service - Same Station

ACSI - Assign the External Route Number, Trunk Number, and Connection Service Index = 3 for DIT. Then assign the Tenant Number and Station Number to serve as the DIT Station.

For Day DIT Service

ACSA - Assign the External Route Number, Trunk Number, Connection Service Index A = 2 for DIT. Then assign the Tenant Number and Station Number to serve as the DIT Station.

For Day and Night DIT Service - Different Stations

ACSI - Assign the External Route Number, Trunk Number, Connection Service Index = 4 for NIGHT CONNECTION - FIXED (DIT) [N-1]. Then assign the Tenant Number and Station Number to serve as the DIT Station.

Night DIT Only

ACSI - Assign the External Route Number, Trunk Number, Connection Service Index = 4 for NIGHT CONNECTION - FIXED (DIT) [N-1]. Then assign the Tenant Number and Station Number to serve as the DIT Station.

FCCS Programming

This data setting describes only the additional or replacement commands used to activate FCCS Services. When used as an FCCS feature, these commands replace the non-network commands (for example, AAED is replaced by AAEDN). Note that the data setting for FCCS Service varies depending on the Node.

Note: When using ASYDL or ASYDN to program an FCCS feature, the basic programming of ASYD is required in addition to ASYDL or ASYDN.

STEP 1: **ACSIL** - This command must be assigned to the Node accommodating the COT. Assign External Route Number/Logical Route Number (RT/LGRT), Trunk Number (TK), and Connection Service Index (CSI) = 3 for DIT activated in Day and Night mode or 4 for DIT activated in Night mode. Then assign the User Group Number (UGN) and Telephone Number (TELN) to serve as the DIT Station.

STEP 2: **ACSAL** - This command must be assigned to the Node accommodating the COT, when setting the DIT Service in Day only or different DIT Station is used in Day and Night. Assign External Route Number/Logical Route Number (RT/LGRT), Trunk Number (TK), and Connection Service Index A (CSIA) = 2 for DIT activated in Day mode. Then assign the User Group Number (UGN) and Telephone Number (TELN) to serve as the DIT Station.

Note: The following data should also be set in addition to the above-mentioned commands: The commands to be set to the Node accommodating the COT: ASYD/ARTD/ATRK/MBTK/ARSC/ACFR.

D-8 Direct Inward Dialing (DID)

General Description

This feature provides for all Incoming Calls from the Public Network (except FX or WATS) to reach any System Station without Attendant assistance.

Operating Procedure

The Calling Party, outside the system, dials the Telephone Number as usual. However, the call is answered directly at a predetermined Station, bypassing the Attendant.

Service Conditions

1. This feature requires that special DID Trunks and Telephone Number be obtained through the local Telephone Company.

   Note: Not all Telephone Company Central Offices are capable of providing this service.

2. If the Central Office Numbering Plan differs from the user's Station Numbering Plan, addition or deletion of digits can be implemented to coincide with the existing Numbering Plan.
3. This feature is normally used when DIRECT-IN Service is desired on an extensive or system basis.
4. DID is also frequently referred to as CENTREX Service.
5. If the Called Station is non-existent, the DID Call is automatically routed to the Attendant or another predesignated Station. Calls to invalid numbers can be routed to an Attendant or to a Recorded Announcement. See CALL FORWARDING - INTERCEPT/ANNOUNCEMENT [C-25].
6. Wink Start or Delay Start Signaling is recommended for use in conjunction with this feature.
7. The DID Trunk contains a switchable 2db transmission pad that is switched in or out via software control.
8. If DID Trunk is immediate start, dial pulse is required.
9. Incoming Calls have priority over Outgoing Calls. When a Register is assigned as an Outgoing Register Trunk (ORT), it will also act as an Incoming Register Trunk (IRT) if no IRTs are available. Therefore, ORTs can act as IRTs, but if IRTs are programmed, they will not act as ORTs.
10. If immediate Start DID Trunks are required, Incoming Register Trunks must be programmed in the same local processor as the DID Trunks.

Interactions

1. CALL FORWARDING [C-2, 3, 5]: When a Station has activated CALL FORWARDING - ALL CALLS [C-5], CALL FORWARDING - BUSY LINE [C-2], or CALL FORWARDING - DON'T ANSWER [C-3], the DID Call is forwarded to a designated Station rather than to the Specific Station dialed.
2. STATION HUNTING [S-7, 8, 9]: If the Called Station is programmed for this feature and is busy, the call follows any prescribed Hunting Pattern.
3. CALL WAITING - TERMINATING [C-12]: If the Called Station has the Call Waiting - Terminating feature, Call Waiting Ringback Tone to the Calling Party is provided when the Called Station is busy. The Called Station hears a Two-burst Call Waiting Tone. Call Waiting is allowed in CDN 46:CW, of ARTD.
4. On an Incoming Call to a busy Station, the Call Forwarding feature takes precedence over the Station Hunting feature, and the Station Hunting feature takes precedence over the Call Waiting feature.
5. The PA-8TLT circuit card provides interface capabilities for eight DID Circuits.

Programming

STEP 1: **ASYD** - SYS, Index 16, Bit 3. Enable the four-digit Station Number. It is normal to enable all options regarding length of Station Numbering. Assign data 1FH.

SYS1, Index 65. Assign the number of Routes in the system.

SYS1, Index 70, Bit 0. Is Calling Number for a DID Incoming to the Attendant in service? 0/1: No/Yes.

STEP 2: **ARTD** - Assign the DIRECT INWARD DIALING Route as shown below:

RT:1

3-ISGS	:7	4-INSG	:3	5-TF	:2	6-TCL	:5*	7-L/T	:1
8-RLP	:2	15-LSG	:8	28-ANS	:1	46-CW	:1		

Note: *This can be assigned as data 1, DDD.

Note: To change the data for existing route (RT), initialize or reboot the trunk device such as circuit card or MG.

STEP 3: **ATRK** - Assign the LENS, DID Trunk Route Number, Trunk Number, and Tenant Number.

Assign the LENS and Incoming Register Trunk Route Number (903). The LENS must be for Levels 0-3 for any 8RST circuit card (see Interactions).

STEP 4: **MBTK** - Assign the make idle status to the DID and Incoming Register Trunks.

STEP 5: **ANPD** - Reserve a Number Level for Station Access. Select Normal Service (N) or Hooking Service (H) from the combo box for Connection Indexes. Number of Necessary Digits (NND) is usually four, and the Busy Lamp Field is activated.

STEP 6: **ASPA** - Assign the Access Code as assigned in ANPD above. Type of Service, SRV = “STN: Station Number”. The number of digits usually required is four.

STEP 7: **ARSC** - The DID Station receiving the call must be assigned a Route Restriction Class (RSC) that will allow the Station to be connected to the Incoming DID Route. The DID Route must be allowed via Route Restriction Indexes (RRI) 0 and 1.

FCCS Programming

This data setting describes only the additional or replacement commands used to activate FCCS Services. When used as an FCCS feature, these commands replace the non-network commands (for example, AAED is replaced by AAEDN). Note that the data setting for FCCS Service varies depending on the Node.

Note: To change the data for existing route (RT), initialize or reboot the trunk device such as circuit card or MG.

STEP 1: **ARTDN** - Assign the DIRECT INWARD DIALING Route as shown below:

RT:1

3-ISGS	:7	4-INSG	:3	5-TF	:2	6-TCL	:5	7-L/T	:1
8-RLP	:2	15-LSG	:8	28-ANS	:1	46-CW	:1		

STEP 2: **ANPDN** - Reserve a Number Level for Station Access. Select Normal Service (N) or Hooking Service (H) from the combo box for Connection Indexes. Number of Necessary Digits (NND) is usually four, and the Busy Lamp Field is activated.

STEP 3: **ASPAN** - Assign the Access Code as assigned in ANPD above. Type of Service, SRV = "TELN: Telephone Number". The number of digits usually required is four.

STEP 4: **ARSCN** - The DID Station receiving the call must be assigned a Route Restriction Class (RSC) that will allow the Station to be connected to the Incoming DID Route. The DID Route must be allowed via Route Restriction Indexes (RRI) 0 and 1.

Note: When using ASYDL or ASYDN to program an FCCS feature, the basic programming of ASYD is required in addition to ASYDL or ASYDN.

D-9 Direct Outward Dialing (DOD)

General Description

This feature permits any user to gain access to the Public Network without the assistance of the Attendant, by dialing an Access Code and receiving a Second Dial Tone. The user may then proceed to dial the desired Public Network Number.

Operating Procedure

To place an Outside Call:

1. Go off-hook. Receive Dial Tone,
2. Dial the Trunk Access Code (usually “9”).
3. Receive Second Dial Tone from Central Office.
4. Dial the desired Telephone Number.
5. If the call is denied for any reason, Reorder Tone is heard.

Service Conditions

1. Outgoing Connection Restriction can be assigned on an individual Station basis. Refer to CLASS OF SERVICE - INDIVIDUAL [C-15].
2. Various types of Trunks (FX, WATS, TIE, Central Office, etc.) can be directly accessed using this feature.
3. The Trunk Group Access Code is limited to a maximum of three digits (only when the SPEED CALLING [S-3, 3D, 21, 23] Service is required).
4. Use of the DOD feature can be denied on a Trunk Group basis when the following restrictions are active on the Originating Station Line.
    - FULLY RESTRICTED STATIONS: Direct Outward Dialing attempts are routed to Reorder Tone when the Station is Fully Restricted.
    - RESTRICTION FROM OUTGOING CALLS: A Station assigned this feature is denied the ability to dial-access preselected Trunk Groups.
    - HOTLINE [H-1]: Exchange Network Call Completion using this feature is not permitted. The Originating Station can only be routed to the ATTENDANT CONSOLE [A-3] or another Station when going off-hook.
    - MISCELLANEOUS TRUNK RESTRICTION [M-3]: Preselected Station Lines are denied access to preselected miscellaneous Trunk Groups, the Station is routed to Reorder Tone when the Station attempts to originate a call on the Trunk Group.
    - TOLL RESTRICTION: Level of this feature Restricts unauthorized Station Lines’ ability to complete Outgoing Central Office or Foreign Exchange Calls to specified areas or Office Codes within an area.
    - TOLL DENIAL: A Station with Toll Denial is routed to Reorder Tone when a 0 or 1 is dialed as the first digit after the Trunk Access Code has been dialed.
5. The Outgoing Call using a Trunk in the Remote Node via FCCS link is available. For more detailed information, see the Data Programming Manual - FCCS.

Interactions

1. The COT circuit card is used for all Ringdown Trunks. This circuit card is equipped to interface with Central Office circuits and may be used for either Loop Start or Ground Start Trunks.
2. TOLL RESTRICTION - 3/6-DIGIT [T-7] may be applied to DIRECT OUTWARD DIALING.

Programming

STEP 1: **ASYD** - SYS1, Index 65. Assign the number of Routes in the system.

STEP 2: **ANPD** - Reserve a Number Level for Outgoing Trunk Access. Select Normal Service (N) or Hooking Service (H) from the combo box for Connection Indexes. Assign NND in accordance with a predetermined numbering scheme.

STEP 3: **ASPA** - Assign the Access Code as assigned in ANPD above. Type of Service, SRV = "OGC: Outgoing Call" for Outgoing Trunk. Assign the Route Number associated with the Access Code.

STEP 4: **ARTD** - Assign the DOD Route as shown below:

GROUND START:

RT: 1

1-OSGS	:2	2-ONSG	:3	5-TF	:1	6-TCL	:1	7-L/T	:1
8-RLP	:2	15-LSG	:1						

LOOP START:

RT:1

1-OSGS	:2	2-ONSG	:3	5-TF	:1	6-TCL	:1	7-L/T	:1
8-RLP	:2								

Note: To change the data for existing route (RT), initialize the circuit card.

STEP 5: **ATRK** - Assign the LENS, DOD Trunk Route Number), Trunk Number, and Tenant Number.

STEP 6: **MBTK** - Assign the Make Idle status to the DOD Trunks.

STEP 7: **ARSC** - The Route Restriction Class must be constructed via RRI: 3, direct access, to allow the Station access to the involved Routes.

FCCS Programming

This data setting describes only the additional or replacement commands used to activate FCCS Services. When used as an FCCS feature, these commands replace the non-network commands (for example, AAED is replaced by AAEDN). Note that the data setting for FCCS Service varies depending on the Node.

Note: To change the data for existing route (RT), initialize the circuit card.

STEP 1: **ANPDL** - Reserve a Number Level for Outgoing Trunk Access. Select Normal Service (N) or Hooking Service (H) from the combo box for Connection Indexes. Assign NND in accordance with a predetermined numbering scheme.

STEP 2: **ASPAL** - Assign the Access Code as assigned in ANPD above. Type of Service, SRV = "OGC: Outgoing Call" for Outgoing Trunk. Assign the Logical Route Number associated with the Access Code.

STEP 3: **ARTDN** - Assign the DOD Route as shown below:

GROUND START:

RT: 1

1-OSGS	:2	2-ONSG	:3	5-TF	:1	6-TCL	:1	7-L/T	:1
8-RLP	:2	15-LSG	:1						

LOOP START:
RT:1
1-OSGS :2 2-ONSG :3 5-TF :1 6-TCL :1 7-L/T :1
8-RLP :2

STEP 4: **ARSCN** - The Route Restriction Class must be constructed via RRI: 3, direct access, to allow the Station access to the involved Routes.

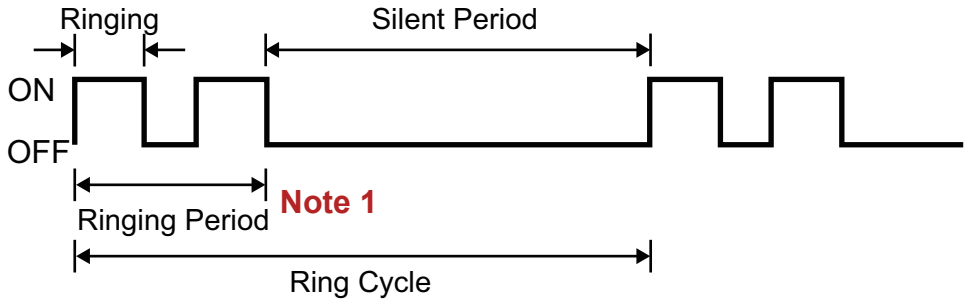
Note: When using ASYDL or ASYDN to program an FCCS feature, the basic programming of ASYD is required in addition to ASYDL or ASYDN.

D-10 Distinctive Ringing

General Description

This feature provides distinctive station ringer patterns so that the user can distinguish between internal and External Incoming Calls.
Ringer patterns are as follows:

[Definition of Terms]



Note 1: Ringing Period consists of a single ringing or group of ringings.

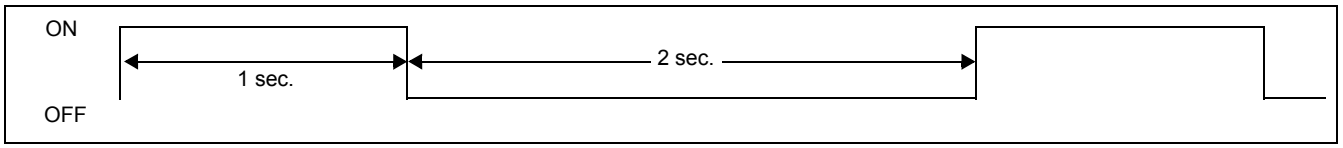
[Ringer Patterns]

(1) Ringer Pattern 0: Used for an incoming call from a central office line.

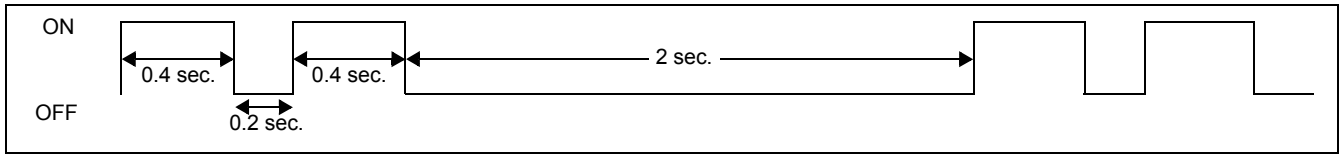
Recommended Setting of Ringer Pattern 0

System Data	Description	Recommended Value
ASYD, SYS3, Index 0	Duration of the Ring Cycle (timer value for ringer used for incoming trunk calls)	21 (Hex)
ASYD, SYS3, Index 3, Bit 0	Multiple Ringings per Ringing Period for Ringer Pattern 0	Bit 0=0 (Disabled)

<Recommended Ring Cycle>



<Ring Cycle When Multiple Ringings per Ringing Period is Enabled
(ASYD, SYS3, Index 3, Bit 0=1)>

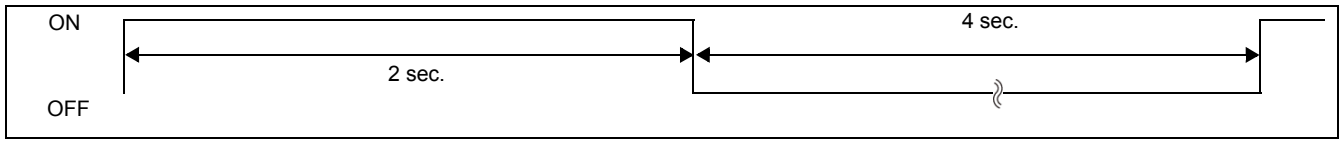


(2) Ringer Pattern 1: Used for an incoming call from a station or TIE line trunk.

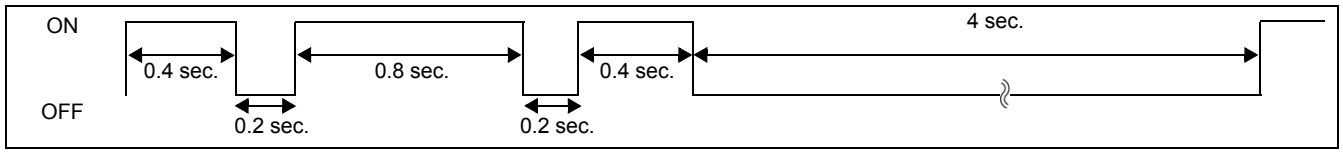
Recommended Setting of Ringer Pattern 1

System Data	Description	Recommended Value
ASYD, SYS3, Index 1	Duration of the Ring Cycle (timer value for ringer used for station calls)	42 (Hex)
ASYD, SYS3, Index 3, Bit 1	Multiple Ringings per Ringing Period for Ringer Pattern 1	Bit 1=0 (Disabled)

<Recommended Ring Cycle>



<Ring Cycle When Multiple Ringings per Ringing Period is Enabled
(ASYD, SYS3, Index 3, Bit 1=1)>

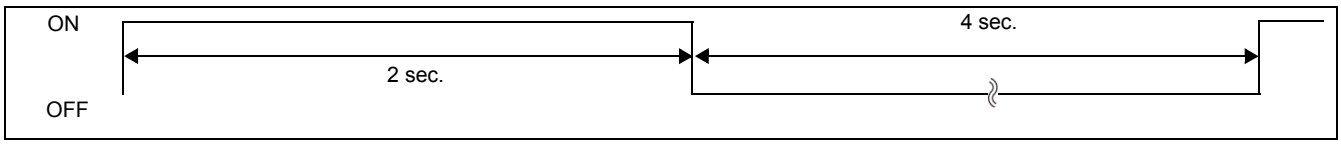


(3) Ringer Pattern 5: Used for an incoming call from a station or TIE line trunk to the Personal Number, CALL BACK [C-1], OUTGOING TRUNK QUEUING [O-2], Priority Ringing in CALL WAITING [C-12], AUTOMATIC/MANUAL INTERCOM [A-30], and DIAL INTERCOM [D-87].

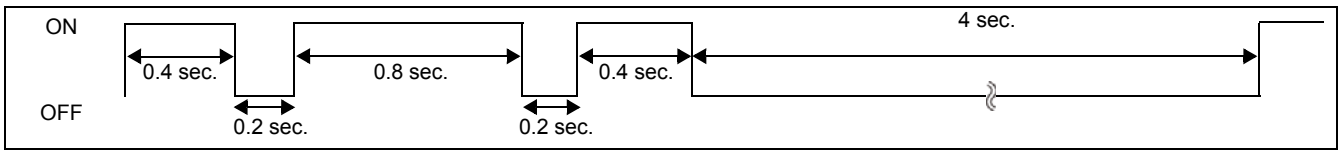
Recommended Setting of Ringer Pattern 5

System Data	Description	Recommended Value
ASYD, SYS3, Index 7	Duration of the Ring Cycle (timer value for ringer for Call Back, OG Trunk Queuing, Priority Ringing in Call Waiting, Auto/Manual Intercom, and Dial Intercom)	42 (Hex)
ASYD, SYS3, Index 3, Bit 5	Multiple Ringings per Ringing Period for Ringer Pattern 5	Bit 5=0 (Disabled)

<Recommended Ring Cycle>



<Ring Cycle When Multiple Ringings per Ringing Period is Enabled
(ASYD, SYS3, Index 3, Bit 5=1)>



Operating Procedure

No manual operation is required.

Service Conditions

1. The following Ringer Patterns are recommended. For the ringer patterns for the Personal Number, refer to PERSONAL RINGER [P-59].

Note: Personal Number means the Telephone Number allowed Personal Ringer function.

Note: Only Desktop terminals can use Personal Ringer function. (This function is not available for analog telephones and Software Line).

- Ringing for Incoming Internal Call:

a.) When a Service Feature Restriction Class that allows Personal Ringer (ASFC, SFI 148) is assigned to a station:
Ringer Pattern 1 - Ringing for Incoming Internal Call to Pilot Number:
A two-second ringing followed by a four-second silence.
Ringer Pattern 5 - Ringing for Incoming Internal Call to Personal Number (ex. My Line):
A two-second ringing followed by a four-second silence

b.) When a Service Feature Restriction Class that restricts Personal Ringer (ASFC, SFI 148) is assigned to a station:
Ringer Pattern 1 - A two-second ringing followed by a four-second silence.

- Ringing for Incoming External Call:

a.) Ringer Pattern 0 - A one-second ringing followed by a two-second silence.

- Recall Ringing for CALL BACK [C-1], CALL WAITING - ORIGINATING [C-31], or OUTGOING TRUNK QUEUING [O-2]:

a.) Ringer Pattern 1 - A two-second ringing followed by a four-second silence
Or
Ringer Pattern 5 - A two-second ringing followed by a four-second silence

Note: These timing parameters can be changed via the end users requirements (1-15 seconds).

Availability of Customization for Ringer Patterns 0 to 7

X: Available N: Not Available –: Not applicable

Ringer Pattern	Customization of Ringer Pattern	Customization of Ring Cycle (Note 2) Duration through System Data		Recommended Ring Cycle Duration (Ringing/Silent) (sec.)	Used for
		ELC	LC	ELC	
Ringer Pattern 0	X	X	X	1/2	Incoming Trunk Calls
Ringer Pattern 1	X	X	X	2/4	Station Calls
Ringer Pattern 2	N	N	N	-	Fax
Ringer Pattern 3	N	N	N	-	Wake Up Calls
Ringer Pattern 4	N	N	N	-	Not used.

Availability of Customization for Ringer Patterns 0 to 7

X: Available N: Not Available –: Not applicable

Ringer Pattern	Customization of Ringer Pattern	Customization of Ring Cycle (Note 2) Duration through System Data		Recommended Ring Cycle Duration (Ringing/Silent) (sec.)	Used for
		ELC	LC	ELC	
Ringer Pattern 5	X	X	X	2/4	Call Back, Outgoing Trunk Queuing, Call Waiting, Auto/Manual Intercom, and Dial Intercom
Ringer Pattern 6	X	X	N	1/2	-
Ringer Pattern 7	X	X	N	1/1	-

Note 2: The Ring Cycle consists of a ringing period and a silent period.

2. This feature may be assigned per Trunk Route via System Data programming.
3. If an Off-Premises Station (OPX) or denied station is assigned to a station line, an option is available for that Class of Service to eliminate burst Ringing Signals.
4. The DISTINCTIVE RINGING interruption rates can be optionally selected on a per system basis. In this case, System Initialization may be required.
5. An option is available on a per Trunk Group basis to provide one-burst ringing on Terminating Calls.
6. When calling a third station (THREE-WAY CALLING [T-2], CONSULTATION HOLD [C-17], etc.), the Ringing Signal to the called station depends on the type of facility (station or trunk) on hold.

Interactions

1. CALL HOLD [C-6]: Recall ringing for CALL HOLD [C-6] can be distinguished between a held station call (two-burst ringing) from a held trunk call (one-burst ringing).
2. CONSULTATION HOLD [C-17]: Station A places Party B on CONSULTATION HOLD [C-17] and calls Station C. If Party B is a Trunk, Station C hears DISTINCTIVE RINGING for a Trunk Call. If Party B is a station, Station C hears Distinctive Ringing for a station call.
3. When Personal Ringer function is set to a station, if an Incoming Internal Call is terminated to the Personal Number and the call is transferred by CALL FORWARDING - DON'T ANSWER [C-3], the Ringer Pattern of the transferred station is Ringer Pattern 3 for an analog telephone, and Ringer Pattern 5 for a Dterm.

Type Of Transferred Station	Ringer Pattern
Analog Telephone	1
Dterm	5

4. The ringer pattern options provided by this feature are not available on telephones connected to UG50, where those patterns are translated into corresponding ringer patterns. For details, see Peripheral Equipment Description (UG50), Chapter 1, "Limitations on Voice and Tone."

Programming

STEP 1: **ASYD** - SYS1, Index 72, Bit 6. Is DISTINCTIVE RINGING applied to calls handled via the *Intermediate Party (Station or Attendant Console)*? 0/1: No/Yes. Otherwise, these calls will ring as Station-to-Station Calls.

Note: This system data is also available in the following case.
- Trunk line → CCIS → Intermediate Party (Station or Attendant Console) → Station

Ringer Pattern 0 - Normally used for central office calls.

SYS3, Index 0. Duration of the Ring Cycle (Timer Value) for Ringer Pattern 0. Normally, data 21 hexadecimal (one second-ON, two seconds-OFF) should be assigned to this Index.

SYS3, Index 3, Bit 0. Two Ringings (0.4 seconds.ON, 0.2 seconds.OFF, 0.4 seconds.ON) per Ringing Period for Ringer Pattern 0. 0/1: Disabled/Enabled.

Ringer Pattern 1 - Used for Station Calls.

SYS3, Index 1. Duration of the Ring Cycle (Timer Value) for Ringer Pattern 1. Normally, data 42 hexadecimal (two seconds-ON, four seconds-OFF) should be assigned to this Index.

SYS3, Index 3, Bit 1. Three ringings (0.4 seconds-ON, 0.2 seconds-OFF, 0.8 seconds-ON, 0.2 seconds-OFF, 0.4 seconds-ON) per Ringing Period for Ringer Pattern 1. 0/1: Disabled/Enabled.

Ringer Pattern 5 - Used for Call Back, OG Trunk Queuing, Call Waiting, and Auto/Dial Intercom.

SYS3, Index 7. Duration of the Ring Cycle (Timer Value) for Ringer Pattern 5.

SYS3, Index 3, Bit 5. Three ringings (0.4 seconds-ON, 0.2 seconds-OFF, 0.8 seconds-ON, 0.2 seconds-OFF, 0.4 seconds-ON) per Ringing Period for Ringer Pattern 5. 0/1: Disabled/Enabled.

Note: For information on how to customize the Ringer Pattern, refer to Command Manual.

Note: When the System Data is changed, circuit card (ELC) initialization is required.

SYS1, Index 417, Bit 0. Recall Ringer Pattern for CALL BACK [C-1], CALL WAITING - ORIGINATING [C-31], and OUTGOING TRUNK QUEUING [O-2].

0/1: Ringer Pattern 5/Ringer Pattern 1.

STEP 2: **ARTD** - To provide a distinctive ring to specific Routes, assign data "1" to CDN 12: DR (follows the ringer pattern assigned by ASYD, SYS3, Index 0, Bit 0 and ASYD, SYS3, Index 3, Bit 0).

Note: To change the data for existing route (RT), initialize or reboot the trunk device such as circuit card or MG.

STEP 3: **ASFC** - Assign a Service Feature Restriction Class that allows SFI 48 (Burst Ringing), to PA-16LC Port Station that will not operate using DISTINCTIVE RINGING. With this feature, the system does not look at ASYD, SYS3, Index 3, Bit 0 and Bit 1 to determine how this telephone will ring. Telex, Fax Machines, and OPX Stations may require a longer Ringing Signal. Digital terminals will not respond to this feature.

Assign a Service Feature Restriction Class that allows SFI 148 (Personal Ringer) to the stations that activates Personal Ringer function. Personal Ringer function changes ringer pattern for Incoming Internal Call according to the Called Number as shown below:

Incoming Internal Call terminated to Pilot Number: Ringer Pattern 1

Incoming Internal Call terminated to Personal Number (ex. My Line): Ringer Pattern 5

D-11D Do Not Disturb -D

General Description

This feature allows a user to establish DO NOT DISTURB status. Incoming Calls will be denied access to My Line, while DO NOT DISTURB status is in effect.

Operating Procedure

To set

1. Press the DO NOT DISTURB (DND) Feature key. The associated LED will light. The LCD displays:

DND SET
(Time Display)

To cancel

2. Press the DO NOT DISTURB (DND) Feature key. The LED will go out. The LCD displays:

DND CANCEL
(Time Display)

Service Conditions

1. This service can only be set or canceled while the Dterm is idle.
2. Lines programmed to ring will not do so while this service is set.
3. While DO NOT DISTURB is set, Outgoing Calls may be made. The user may answer other Lines, excluding My Line.
4. DND is not displayed on the Calling Station's Dterm when the Called Station is in the DO NOT DISTURB condition. Restriction (REST) appears on the display in place of DND. Reorder Tone is heard, and no indication will appear in the DO NOT DISTURB condition when the Calling Station is a HOTLINE [H-1].

REST	**XXXXX**
(Time Display)	

5. A DND Station in a HUNT GROUP [D-7, 8, 9] is skipped.
6. CALL FORWARDING - BUSY LINE [C-2] can be activated for an idle Station that has set DND.
7. The following features are denied to the DND Station:

ATTENDANT CAMP-ON WITH TONE INDICATION	[A-1]
ATTENDANT OVERRIDE	[A-7]
BOSS - SECRETARY OVERRIDE -D	[B-5D]
BUSY VERIFICATION	[B-3]
EXECUTIVE RIGHT-OF-WAY	[E-1]
VOICE CALL	[V-2]

8. When the user sets DO NOT DISTURB while engaged in a call, privacy is set. In this case, the following services will be restricted:

CALL WAITING - TERMINATING	[C-12]
EXECUTIVE RIGHT-OF-WAY	[E-1]

9. When the terminal performs re-registration while DO NOT DISTURB is set, the associated LED of the DND Feature key will light and the Soft key will flash. However, this is only available for IP terminals and SIP Multiple Line terminals and will not be available for Digital terminals.

Interactions

1. DO NOT DISTURB may be listed in some literature as available by a Line/Feature key. A Desktop terminal ACD Station is not equipped with the DND Feature key. If DO NOT DISTURB is to be made available to this Station, it must be programmed to a Line/Feature key.
2. DATA INTERFACE - AUTOMATIC ANSWER may be activated if the Dterm Station is in the DO NOT DISTURB mode.

Programming

STEP 1: **AKYD** - Assign the following Key Data for the Desktop terminal:

TN: 1
STN: Station Number
TP: Type of Dterm (0-3)
KYN: Key Number (1-40)
KYI: Service Index
0: Key not used
→ 1: Feature Key
2: Multi Line Key
KD: Not assigned for Feature key
FKY: Feature Key Number
FKY: 50 (Do Not Disturb)

STEP 2: **ADSL** - Assign SN = 0 (Idle) and Feature key Do Not Disturb [50].
This command is used for assigning the function to each Soft key.

STEP 3: **ASFC** - Select an indication type (REST/DND) on the display of a Dterm when it terminates to a Station that has been set DO NOT DISTURB.

SFI: 114, [Calling Party DND Indication (Dterm)]
0/1: REST/DND.

D-13D Dial Monitor -D

General Description

This feature provides for dialed digits to appear on the Desktop terminal LCD.

Operating Procedure

1. Lift the handset or press the SPEAKER key. Receive Dial Tone.
2. Dial the desired number. The digits will appear in sequence on the LCD.

Service Conditions

1. Digits appear from the right and move to the left as additional digits are dialed. A maximum of 16 digits can be displayed. If more than 16 digits are dialed, the first dialed digits will be dropped and the last dialed digit appears. This process continues until all digits have been dialed.
2. Digits also appear on the display when implementing SPEED CALLING - ONE TOUCH -D [S-26D], LAST NUMBER CALLED -D [L-6D] and SAVE AND REPEAT -D [S-24D].

Programming

No programming is required.

D-15 Day/Night Class of Service

General Description

This feature permits any Station to be assigned one CLASS OF SERVICE - INDIVIDUAL [C-15], for Day and another for Night. System Data can be programmed so that once the ATTENDANT CONSOLE [A-3] has entered the NIGHT CONNECTION [N-1, 2] mode, a Station's CLASS OF SERVICE - INDIVIDUAL [C-15] will be automatically changed when required. Class may be upgraded or downgraded, and Trunk Groups normally controlled by the Attendant in the Daytime may be opened to Station control in the Night mode.

Operating Procedure

This feature is controlled entirely via the ATTENDANT CONSOLE [A-3]. Pressing the Night key or removing the handset changes the Service Class. Releasing the Night key or replacing the handset restores normal CLASS OF SERVICE - INDIVIDUAL [C-15].

Service Conditions

1. DAY/NIGHT CLASS OF SERVICE may be assigned on a per Tenant and per Station basis.
2. DAY/NIGHT CLASS OF SERVICE may be assigned to different Trunk Restriction and Service Feature Restriction Classes.
3. Refer to CLASS OF SERVICE - INDIVIDUAL [C-15] for further details.
4. Night mode Restriction Classes can be set at any time by entering the related command on the PCPro.
5. When TENANT SERVICE [T-12] is provided with MULTIPLE CONSOLE OPERATION [M-4], the system can be programmed for either all ATTENDANT CONSOLEs [A-3] or primary ATTENDANT CONSOLE [A-3] to control activation of DAY/NIGHT CLASS OF SERVICE.
6. When DAY/NIGHT mode Changeover occurs, Tenant Number, Day/Night mode of the Tenant, status of the External Switch, and cause of changeover are printed out on the Alarm Printer and PCPro Printer. This service is determined by System Data.
7. Day/Night information of the unused Tenant is not printed out.
8. A maximum of 14 Tenants' information can be printed per message.
9. This feature will output a 17-H informational System Message as per System Data programming (see System Message 17-H below for details).
10. When the Attendant's Night key or External Switch is turned ON and OFF repeatedly, the messages printed out and the operation may not coincide, but the System Message will represent the final Day/Night mode.
11. In the following cases, Day/Night mode is not changed when the External Switch is turned ON, and only the System Message is printed out:
    a.) When neither the ATTENDANT NIGHT TRANSFER [A-19] or TAS [T-8] is available, SYS2, Index 13, Bit 1=1 (no check by Attendant at the time of Day/Night mode Changeover), and the Night key of Attendant indicates Day.
    b.) When neither ATTENDANT NIGHT TRANSFER [A-19] or TAS [T-8] is available, SYS2, Index 13, Bit 1=0 (check by Attendant at the time of Day/Night mode Changeover), and the Night key of Attendant indicates Night.
12. In the same Tenant, it is not available to use both the Attendant's Night key and the External Switch for Day/Night mode Changeover. Select either the Night key or External Switch per Tenant.

13. When Day/Night mode Changeover is activated by jack at the time System Data is SYS1, Index 77, Bit 0=1 (Day/Night mode Changeover by jack available), the cause of changeover is indicated as the Night key (= 1).
14. This feature is to be controlled by the Attendant Console accommodated in the Terminating Node for the FCCS Network.
15. Day/Night mode is changed by the Destination Node of the Operator Call assigned System Data for the FCCS Network.

    Note: The following Attendant Console Day/Night mode Change methods are determined by System Data: By the Primary Attendant Console/By all Attendant Consoles.

16. The Node which does not accommodate the Attendant Console controls DAY/NIGHT mode of the Self Node using the External key for the FCCS Network.
17. The user can assign only one Destination Node from a Node, even when the system is organized by more than one Tenant.
18. When activating this feature by using the External key, only the Self Node which accommodates the External key can be controlled.

Interactions

1. When assigning Station data, assign the Restriction Classes desired to cause the Station to upgrade, downgrade, or remain unchanged status of Restriction Class when going from Day to Night. If DAY/NIGHT CLASS OF SERVICE is not required, assign the commands normally.
2. The DAY/NIGHT CLASS OF SERVICE arrangements are based on the Day/Night Operating Status of the Tenant.
3. DAY/NIGHT CLASS OF SERVICE affects the following restrictions:
   a.) Route Restriction Class (ARSC/ARSCN)
   b.) Service Feature Restriction Class (ASFC/ASFCN)
   c.) Toll Restrictions (ATDP/ATDPL/ATDPN)
   d.) Tenant Restrictions (ATNR)
4. When TRUNK ANSWER FROM ANY STATION (TAS) [T-8] is changed with set or applied, a Station's Night RSC must allow access to the IC Trunks. Also, the Night SFC must allow TAS, SFI 13.

Programming

STEP 1: **ASYD** - SYS1, Index 75, Bit 0. Are Route Restriction Class Pattern (ARSC/ARSCN) Tables to be developed for both Day and Night? 0/1: No/Yes.

SYS1, Index 75, Bit 1. Are the Service Feature Restriction Class Pattern (ASFC/ASFCN) Tables to be developed for both Day and Night? 0/1: No/Yes.

SYS1, Index 75, Bit 2. Are the Tenant Restriction Data Pattern (ATNR) Tables to be developed for both Day and Night? 0/1: No/Yes.

SYS1, Index 75, Bit 3. Are the Toll Restriction Data (ATDP/ATDPL/ATDPN) Tables to be developed for both Day and Night? 0/1: No/Yes.

SYS1, Index 77, Bit 0. Is Day/Night mode change accomplished by removing the handset from the ATTCON? 0/1: No/Yes.

SYS1, Index 229, Bit 3. System Message printout on changeover of the Day/Night mode? 0/1: Out/In Service.

SYS2, Index 2, Bit 2 to Bit 3. ATTCON Day/Night mode change is:

Bit 3	Bit 2	
0	0	= Not required (External Switch effective)
0	1	= by Primary Attendant Console
1	0	= by all Attendant Consoles

STEP 2: **ASFC/ASFCN** - Assign this command for both the Day and Night Tables.

STEP 3: **ARSC/ARSCN** - Assign this command for both the Day and Night Tables.

STEP 4: **ATNR** - Assign this command for both the Day and Night Tables.

STEP 5: **ATDP/ATDPL/ATDPN** - Assign this command for both the Day and Night Tables.

STEP 6: **ASID** - If ATTENDANT NIGHT TRANSFER [A-19] is needed, assign this command.

FCCS Programming

This data setting describes only the additional or replacement commands used to activate FCCS Services. When used as an FCCS feature, these commands replace the non-network commands (for example, AAED is replaced by AAEDN). Note that the data setting for FCCS Service varies depending on the Node.

Note: When using ASYDL or ASYDN to program an FCCS feature, the basic programming of ASYD is required in addition to ASYDL or ASYDN.

STEP 1: **ASYDL** - This data must be set to all the Nodes. SYS1, Index 640. Assign the Node of Attendant Console.

STEP 2: **ASPAL** - These data must be set to the Node that calls the Attendant.

SRV: "SSC: Service Code"

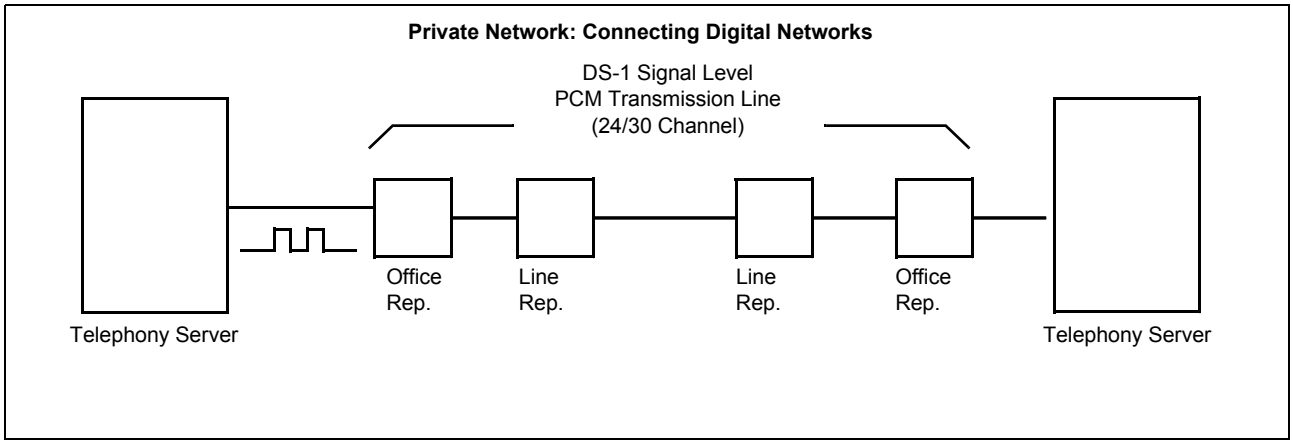
SID: 2 = Operator Call

FPC: FPC of the Node accommodates the Attendant Console

D-16 Direct Digital Interface

General Description

This feature provides the capability to connect Trunks from the Telephony Server directly to T1/E1 Carrier Links via either Private or Public Network.



Operating Procedure

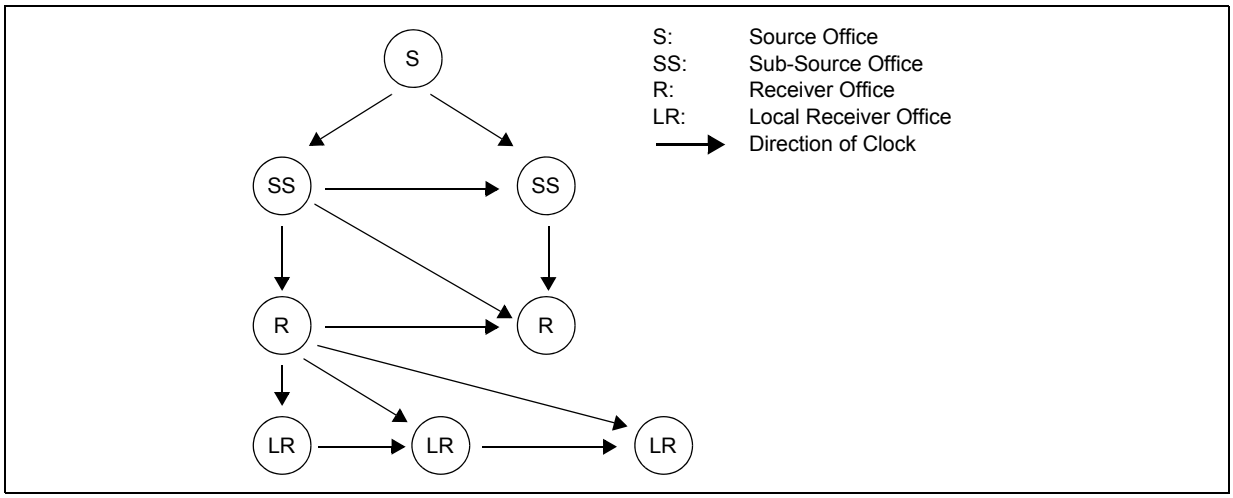
No manual operation is required.

Service Conditions

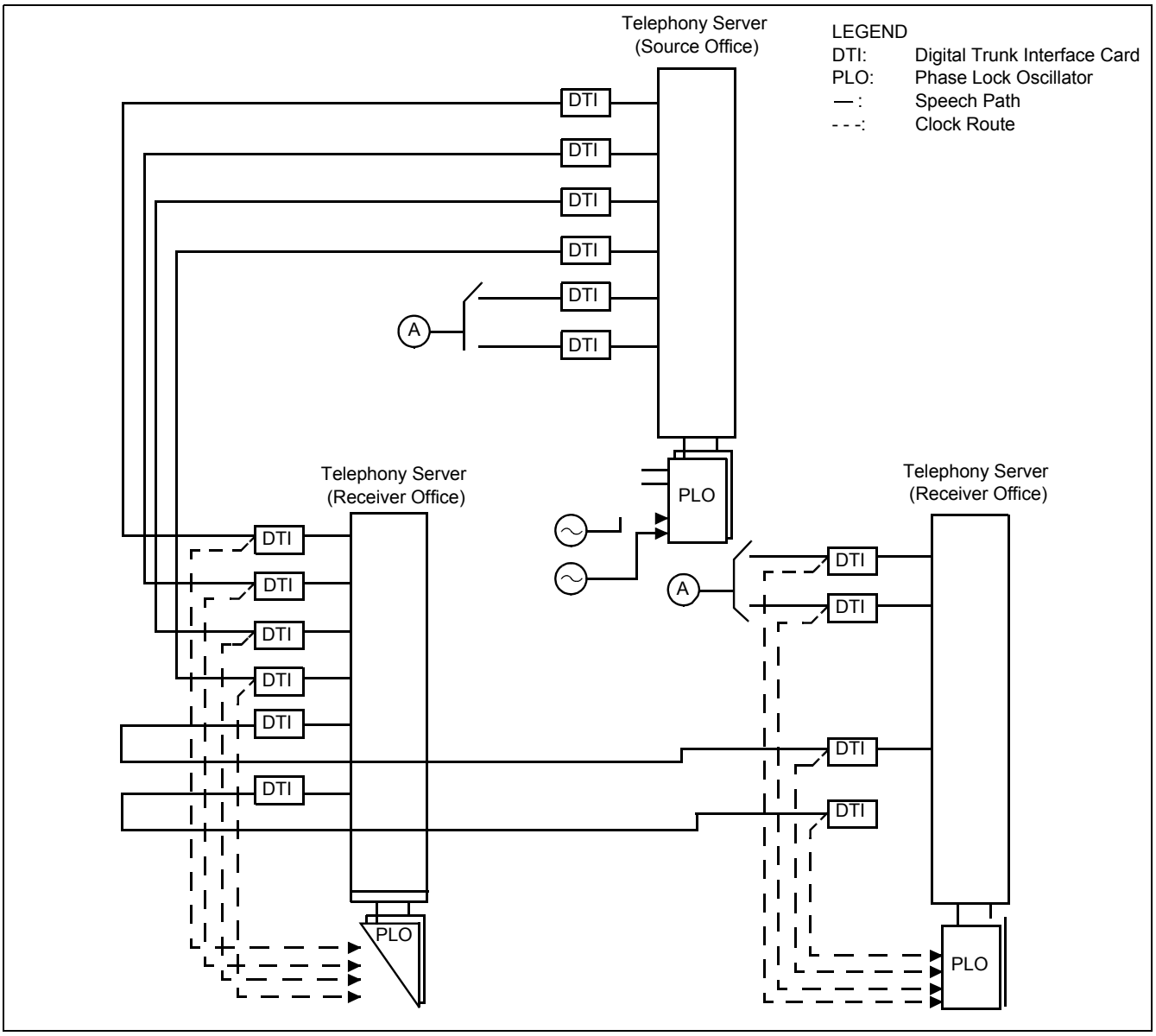
1. The DTI provides Signalling Interface, Bipolar/Unipolar Conversion, Frame Synchronization, Insertion/ Extraction and Alarm Detection.
2. The Digital Interface Module can be provided with circuit cards to interface with T1/E1 Carrier and/or CCITT Standard Networks.
3. The Digital Interface Module (DTI) is mounted in a Port Interface Rack (PIR).
4. Source-Receiver System (Source) or Source-Receiver System (Receiver) may be used for network synchronization:
5. Each Office hierarchy is defined as follows:
   a.) Source Office: One center will operate as the Source Office. This location has two highly stabilized Source Oscillators, and distributes the Source Clock to all the systems through the Digital Interface Lines.
   b.) Sub-Source Office: This Office operates using a Phase Lock Oscillator (PLO) to synchronize with the clock at the Source Office. If the Source Clock has some failure, the Office can operate using its own backup oscillator.
   c.) Receiver Office: This Office is programmed so it will have at least two Clock Routes, one for Receiver and the other for Standby. Synchronization Clock is derived from Incoming PCM Bit Stream from higher hierarchy Offices.
   d.) Local Receiver Office: This is the End Office in a Digital Network arrangement. This Office will not be provided with a backup Route for the PLO, because this Office is the only one influenced in an event of trouble occurrence.

6. Each Telephony Server can be equipped with a duplicate Phase Lock Oscillator used for network synchronization. Each PLO has four inputs that clocks from the Source Oscillator or Digital Interface Module connect to.

Network Clocking Hierarchy



Network Clock Synchronization



Programming

STEP 1: **ASYD** - SYS1, Index 91, Bit 4 to Bit 7. If one Phase Lock Oscillator (PLO) is mounted, assign data "1" to Bit 4 and Bit 6. If two PLO's are mounted, assign all four Bits as data "1".

STEP 2: **ARTD** - Assign the DIRECT DIGITAL INTERFACE Route as follows. This is for Second Dial Tone TIE Lines.

RT: 1

1-OSGS	:2	2-ONSG	:3	3-ISGS	:2	4-INSG	:3	5-TF	:3
6-TCL	:4	7-L/T	:1	8-RLP	:2	13-AC	:1	15-LSG	:5
30-PAD	:7								

LCR FLAG ROUTE:

RT: 31

5-TF	:3	6-TCL	:1	7-L/T	:1	8-RLP	:2	9-TQ	:1
13-AC	:1								

Note: To change the data for existing route (RT), initialize the circuit card.

STEP 3: **ATRK** - Assign the LENS, DIRECT DIGITAL INTERFACE Trunk Route Number, Trunk Number, and Tenant Number.

STEP 4: **MBTK** - Assign the make idle status to the DIRECT DIGITAL INTERFACE Trunks.

STEP 5: **ANPD** - Reserve a Number Level for Register Sender Least Cost Routing. This level should correspond to the Station Numbering Plan of the Opposite Office. Select Normal Service (N) or Hooking Service (H) from the combo box for Connection Indexes. Number of Necessary Digits will be the length of the digits in the Opposite Office's Station Numbering Plan.

STEP 6: **ASPA** - Assign the Access Code of the LCR Flag Route. Type of Service, SRV = "LCR: Least Cost Routing", LEAST COST ROUTING - 3/6 DIGIT [L-5]. Assign Connection Index, CI = "N: Normal", "H: Hooking". Assign the LCR Flag Route Number associated with this Access Code.

STEP 7: **ARNP** - Assign the Physical Route Numbers as assigned in ARTD, and relate these Routes to the LCR Access Code as assigned in ASPA for SRV = "LCR: Least Cost Routing". The LCR Access Code is the first digit of the Opposite Offices' Station Numbers.

Note: These same Routes may also be listed in ASPA for SRV = "OGC: Outgoing Call". In that case, these Routes may also be available via direct dial Access Codes, but are not assigned for LCR in ARNP.

STEP 8: **AMND** - Assign the Maximum Necessary Digits the system must translate for correct Route selection. In this case, assign the first digit of the other system's Station Numbering and the number of digits in the Station Numbering Plan.

STEP 9: **AFRS** - List the dialing patterns in groups that are served by a specific sequence of Routes. Use only the DTI Routes and the first digit of the Station Numbering. Assign this dialing sequence to an OPR.

STEP 10: **AOPR** - AFRS designates the dialing sequence and the OPR to be selected. The AOPR defines the Routes and in what order these Routes will be selected. In this case, only the DTI Routes will be assigned. Assign the following parameters:

TDC Number (TDPTN) - Time-of-Day Pattern Change. Usually, this parameter does not apply, and data "0" must be entered. For an AFRS assignment, however, if different OPRs will be required as a function of time, designate which Time-of-Day Pattern Change Number (TDPTN) OPR is assigned (0 -7).

OPR Number (OPR) - Outgoing Pattern Route, the number referenced in AFRS consisting of a series of routes to be chosen based on the Dialed Number.

Route Advance Number (RA) - The order in which a Route will be selected (0-7). Usually only one or two Routes are involved with DTI TIE Lines.

Route Advance End Display (E) - The last Route in the Route Advance Group. Indicate with data "0". All other Routes are indicated by data "1".

Route Number (RT) - The Physical Route Number to be selected, as programmed in ARTD.

Skip Digits (SKIP) - Do not skip any digits. Usually the LCR Access Code is the opposite system's Station Level. This digit must be sent to the Opposite System.

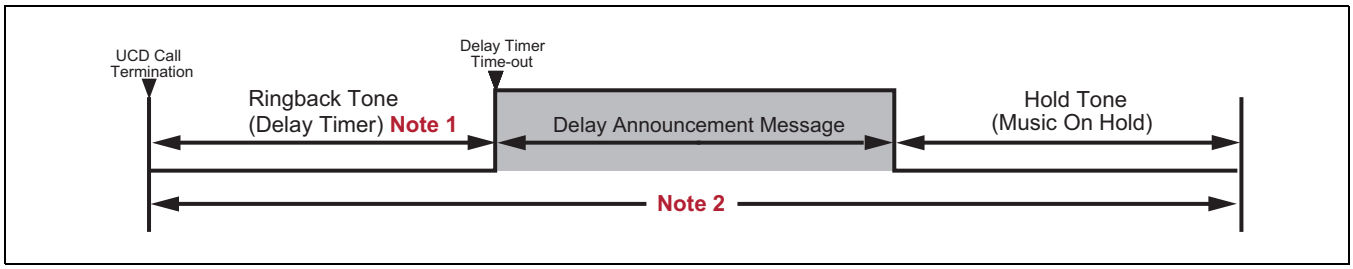
Pattern Number Location (PNL) - If digits are to be added after translating the dialed digits in AFRS, use AADC. This is usually not applicable.

Overflow Tone (OVFT) - Assign if a Tone is wanted for the Last Choice Route. If so, assign data "1" for DDD. Usually not applicable.

D-31 Delay Announcement - UCD

General Description

This feature provides recorded Announcement Service for an Incoming Call directed to a busy UNIFORM CALL DISTRIBUTION (UCD) [U-1] Group and placed in a Waiting Queue. The Recorded Announcement occurs after Ringback Tone is sent for a predetermined time. Refer to the figure below.



Note 1: Delay Time depends on DLYT parameter of AUAD.

Note 2: When a UCD group Station becomes idle, a Queued Call is connected to the UCD Group Station.

This feature can provide Announcement Service by using VS32 Conference.

Operating Procedure

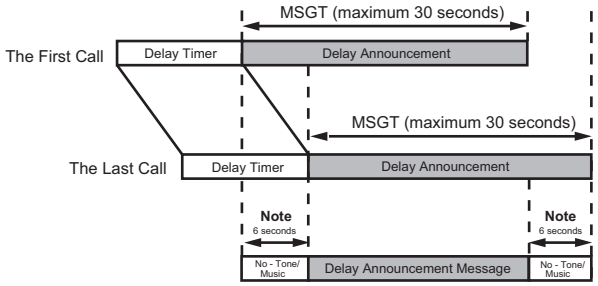
No manual operation is required.

Service Conditions

1. UNIFORM CALL DISTRIBUTION [U-3] must first be provided.
2. Up to eight Announcement Machines can be assigned per UCD [U-1] Group.
3. After the announcement is played, MUSIC ON HOLD [M-7] is activated.
4. Ringback Tone is sent to the Queued UCD [U-1] Call before the first Delay Announcement begins.
5. If programmed, the DELAY ANNOUNCEMENT can be repeated while the Incoming Call is waiting in the queue. Between each announcement, MUSIC ON HOLD [M-7] will be heard.
6. When a UCD [U-1] Station becomes idle, the Incoming Call is connected immediately to the idle Station, even if the Incoming Call is connected to the DELAY ANNOUNCEMENT - UCD.
7. If the Incoming Call comes from a Central Office/ISDN Trunk, billing begins with the start of the Delayed Announcement.
8. If an Incoming Call encounters all Stations busy in a UCD [U-1] Group and overflows to another UCD [U-1] Group, and if all Stations are busy in the OVERFLOW - UCD [O-13] Group, the Calling Party will be placed in the queue for the originally Called UCD [U-1] Group, and DELAY ANNOUNCEMENT - UCD will be invoked, if programmed.
9. The following figure shows Delayed Announcement.

Time Chart for Delay Announcement

The first duration of six seconds after Delay Announcement plays, calls can be connected to the Announcement Trunk (Multiple Connection is available). Calls which are not connected to the Trunk are connected to another Announcement Trunk or received RBT until the Trunk or UCD Group Station becomes idle.



Note: When sending the announcement to all connected calls from the outset, record the no-Tone or music before and after the Delay Announcement Message.

10. This feature can be programmed in Local Data Memory (LDM).
11. When a UCD Station is assigned as the destination of LDN Night Connection this service cannot be activated.
12. Delay Announcement Trunk (DAT) in the Node accommodating the UCD Controlling Station must be used for this feature in the FCCS Network.
13. When all the IPPADs are placed into busy state during the Announcement, Ringback Tone is sent instead of Hold Tone (Music On Hold) after the Announcement.
14. When this feature is activated for CCIS incoming call, the call duration count is renewed every time the line is connected to the announcement trunk.
15. Do not set ASFC, SFI 252=1 (Multi-Boss/Multi-Secretary is in service) to stations belonging to UCD groups.

When using VS32 Conference Server:

1. Announcement patterns can be specified per trunk.
2. Announcement message is sent from the VS32 Conference Server in the System having the UCD group data.
3. This feature is available for an incoming call via PHC.
4. When using this feature via FCCS network, announcement message is sent from the VS32 Conference Server in the System having the UCD Control Station.
5. Multi-connection is not supported.
6. A maximum number of channels that can be used as Announcement Trunk of VS32 at a time is as follows:
   - When Payload Type = 10 msec: 16 channels
   - When Payload Type = 20/30/40 msec: 32 channels

Interactions

1. When a UCD Station is assigned as the destination of LDN Night Connection, this service cannot be activated.
2. This feature does not support LOCATION DIVERSITY [L-55].

Programming

STEP 1: **ASYD** - SYS1, Index 70, Bit 6. Are DELAY ANNOUNCEMENT - UCD common or separate for all UCD Groups assigned by ASHU? 0/1: Common/Separate.
ASYDL - SYS1, Index 804, Bit 4. Are DELAY ANNOUNCEMENT - UCD common or separate for all UCD Groups assigned by ASHUL? 0/1: Separate/Common

STEP 2: **ARTD** - Assign the Announcement Route as follows:

1-OSGS	:2	2-ONSG	:3	3-ISGS	:2	4-INSG	:3	5-TF	:3
6-TCL	:4	7-L/T	:1	8-RLP	:2	15-LSG	:5	28-ANS	:0/1

Note: To change the data for existing route (RT), initialize the circuit card.

STEP 3: **ATRK** - Assign the LENS, Announcement Route Number, Trunk Number, and Tenant Number.

STEP 4: **MBTK** - Assign the Make Idle status to the Announcement Trunks.

STEP 5: **AUAD/AUADL** - This command is used to assign the DELAY ANNOUNCEMENT - UCD.

Assign the following information:

TN/UGN: Tenant Number/User Group Number of UCD Control Station

STN/TELN: Station Number/Telephone Number of UCD Control Station

Equipment Kind: Select “DAT/Announcement Machine”. (only for AUADL)

DLYT: Delay Timer (0 - 30 seconds in 2 second even numbered intervals. 32 - 120 seconds in 8 second even numbered intervals).

MSGT: Message Timer (0 - 30 seconds in 2 second even numbered intervals).

RP: Announcement Output Reply

0: Only once

1: Replay

INT: Replay Interval (0 - 30 seconds in 2 second even numbered intervals. 32 - 120 seconds in 8 second even numbered intervals).

When RP = 0 is assigned, INT (Replay Interval) value is “0”.

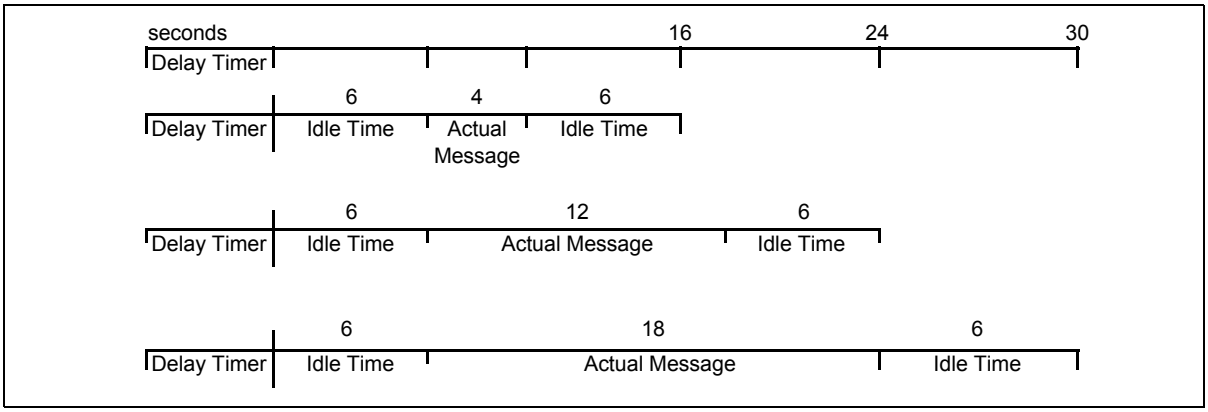
CNT: Number of Announcement Trunks (0-30 seconds; two seconds, even number increments)

RT: Route Number

TK: Trunk Number

The Delay Timer, DLYT, is the time that the Delay Announcement Trunk will ring and collect other Incoming Calls that will listen to the DELAY ANNOUNCEMENT - UCD.

The Message Timer, MSGT, should include both the DELAY ANNOUNCEMENT - UCD time plus idle time before and after the announcement. The following patterns are available (example).



When using VS32 Conference Server:

STEP 1: **ASYD** - SYS1, Index 60, Bit 3=0. UCD Queuing is used.

STEP 2: **ASYDL** - SYS1, Index 804, Bit 4 (UCD Announcement Information): 0/1 = Individual setting for each UCD group/Common in all the UCD groups. This data is set for UCD group created by the Telephone Numbers (ASHUL/ASHUN).

STEP 3: **ARTD/ARTDN** - Assign the route class data of the Announcement trunk.
CDN28 (ANS): Fixed to 0 (No answer signal is given.from the destination office).

Note: To change the data for existing route (RT), reboot the VS32.

Note: When using logical route number, assign the following data. At this time, the remainder of the data must be set by using the NDM commands (ASHUN, AUADN, ARRCN, ARSCN).

STEP 4: **ALRTN** - Assign the logical route number to the route number set by ARTD.

STEP 5: **AIVSL** - Assign the announcement trunk data.

- Basic Data

CH: Enter Channel Number of VS32.
KIND: Select “ANNC”.
RT: Route Number
TK: Trunk Number
LENS: Line Equipment Numbers
TN: Tenant Number

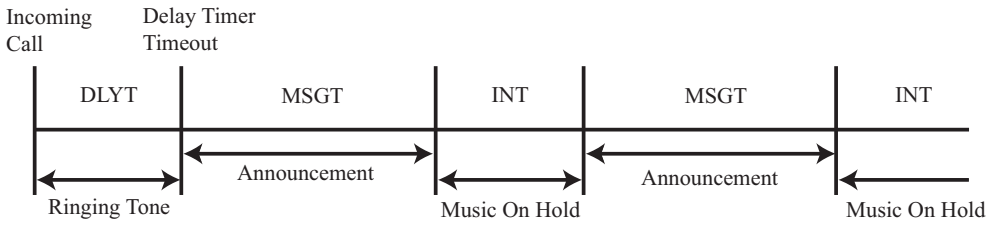
STEP 6: **ASHUL/ASHUN** - Assign the UCD Group data.
UGN: User Group Number
TELN: Telephone Number of UCD Control Station
[Station List in Group]
TELN: Telephone Number of UCD Member Station

STEP 7: **AUADL/AUADN** - Assign the Delayed Announcement data.
UGN: User Group Number
TELN: Telephone Number of UCD Control Station

Equipment Kind: Select “VS”.
DLYT: Delay Timer Value (0 - 30 seconds in 2 second even numbered intervals. 32 - 120 seconds in 8 second even numbered intervals).
MSGT: Message Timer Value (0 - 30 seconds in 2 second even numbered intervals).
RP: Only Once/Replay
INT: Repetition Period (0 - 30 seconds in 2 second even numbered intervals. 32 - 120 seconds in 8 second even numbered intervals).
CNT: Number of Announcement Trunks (1-8)
RT/LGRT: Route/Logical Route Number of Announcement Trunk
TK: Announcement Trunk Number
ANP: Assign the source of announcement to be played.
0-254: Wave file 0 to 254
255: External Input Interface
REP: Assign repeat pattern of the announcement (0-3).
0: Play it once only
1: Play the wave file specified by the ANP parameter.
2: Play four wave files specified by the ANP parameter.
(Example: 00(wave file specified by ANP)–01–02–03–End)
3: Play four wave files to the last, and then repeat the third and the forth by rotations.
(Example: 00–01–[02–03] Repeat)
4-255: Not used

Note: The INT parameter appears when RP = 1.

Note: The ANP and REP parameters appear when Equipment Kind = VS.



Note: Repeat Pattern in the MSGT is assigned by the REP.

STEP 8: **MBTK** - Release the make-busy of Announcement Trunk.
RT: Route Number of Announcement Trunk
TK: Announcement Trunk Number
MB: Assign “0” (Make Idle).

STEP 9: **ARRC/ARRCN** - Allows the tandem connection restriction between Office Line/TIE Line Trunk and Announcement Trunk.

ICRT: Route Number of Office Line/TIE Line Trunk
OGRT: Route Number of Announcement Trunk
ARI A-RES: Assign "1" (Not restrict).
ARI D-RES: Assign "1" (Not restrict).

STEP 10: **ARSC/ARSCN** - Release the restriction between trunks to be used for the connection between Station and Announcement Trunk.

D/N: Day/Night
TN: Tenant Number
RT: Route Number of Announcement Trunk
PRI: Route Restriction Index (0-3)
RSC: Route Restriction Class
RES: Assign "1" (Not restrict).

FCCS Programming

This data setting describes only the additional or replacement commands used to activate FCCS Services. When used as an FCCS feature, these commands replace the non-network commands (for example, AAED is replaced by AAEDN). Note that the data setting for FCCS Service varies depending on the Node.

Note: When using ASYDL or ASYDN to program on an FCCS feature, the basic programming of ASYD is required in addition to ASYDL or ASYDN.

STEP 1: **ASYDL** - SYS1, Index 804, Bit 4.
Are DELAY ANNOUNCEMENT-UCD common or separate for all UCD groups assigned by ASHUL command? 0/1 = Separate/Common.

STEP 2: **AUADN** - The following data must be assigned to the NCN.

UGN: User Group Number

TELN: Telephone Number

Equipment Kind: Select “DAT/Announcement Machine”.

DLYT: Delay Timer (0 - 30 seconds in 2 second even numbered intervals. 32 - 120 seconds in 8 second even numbered intervals).

MSGT: Message Timer Value (0 - 30 seconds in 2 second even numbered intervals).

RP: Announcement Replay
0/1=Only one/Replay

INT: Replay Interval (0 - 30 seconds in 2 second even numbered intervals. 32 - 120 seconds in 8 second even numbered intervals).

Note: When RP=0 is assigned, INT (Replay Interval) value is “0.”

CNT: Number of Announcement Trunks

LGRT: Logical Route Number

TK: Trunk Number

Note: The following data should also be set in addition to the above-mentioned commands. The commands to be set to the Node that accommodates the UCD Controlling Station only: ATRK, MBTK

D-32D Dual Hold -D

General Description

This feature allows two Desktop terminals to simultaneously be placed on hold. This allows the Held Parties to answer or originate a call from a Sub Line Appearance or the idle Prime Line.

Operating Procedure

To place a call in a DUAL HOLD -D condition

1. When two Desktop terminals are connected, either Desktop terminal user can press the pre-assigned HOLD key and place the other Desktop terminal on hold.
2. If the held Desktop terminal user wishes to originate or answer another call from a Sub Line Appearance, the held Desktop terminal user presses the pre-assigned HOLD key and places the connection in a DUAL HOLD -D condition.

To place back to the original connection

1. The Desktop terminal user goes off-hook and presses the Held Line.
2. The held Desktop terminal hears MUSIC ON HOLD [M-7] if the originally connected Desktop terminal is still holding the call.
3. The held Desktop terminal comes back to the original call if the originally connected Desktop terminal has already returned to the original call.

Service Conditions

1. The DUAL HOLD -D feature can be activated by pressing the HOLD key during a call between any two Desktop terminals.
2. While a call is being held, Automatic Recall is initiated after a preprogrammed time on the Desktop terminal that placed the call on Hold.
3. For STATION MESSAGE DETAIL RECORDING (SMDR) [S-10] information between extensions, a call is regarded as completed when the HOLD key is pressed to hold the call. When both Stations answer the Held Call, it is processed as a new call.
4. Even in the case of a call between two Desktop terminals, DUAL HOLD -D cannot be initiated between Boss and Secretary after the Boss - Secretary Service, or between Overriding and Overridden Parties after the PRIVACY RELEASE [P-18] Service.
5. This feature is not restricted by the Class of Service for the Stations in the network.
6. The following services are available to either the holding line or the line placed on hold when a call is held by pressing the HOLD key in a DUAL HOLD -D situation:

CALL BACK	[C-1]
MESSAGE REMINDER	[M-19]

7. The following services are not available to either the holding line or the held line in a DUAL HOLD -D situation:

ATTENDANT CAMP-ON WITH TONE INDICATION	[A-1]
BUSY VERIFICATION	[B-3]
CALL WAITING - ORIGINATING	[C-31]
CALL WAITING - TERMINATING	[C-12]
EXECUTIVE RIGHT-OF-WAY	[E-1]
PRIVACY RELEASE	[P-18]

8. DUAL HOLD -D cannot be activated for calls established by:

BOSS - SECRETARY Service	[B-5D, 8, 11]
PRIVACY RELEASE	[P-18]
BUSY VERIFICATION	[B-3]
ATTENDANT CAMP-ON	[A-1]
EXECUTIVE RIGHT-OF-WAY	[E-1]
CALL WAITING - ORIGINATING	[C-31]
CALL WAITING - TERMINATING	[C-12]

9. In FCCS network, this feature is available within the self node only.

Programming

ASYD - SYS1, Index 144. Recall Timer for EXCLUSIVE, NON-EXCLUSIVE HOLD. Assign data 00H for 30 seconds. (RAM default data is 3FH.)

SYS2, Index 12, Bit 0. Is Automatic Recall Ringing provided on a per Tenant basis? 0/1: Yes/No

D-63 Day/Night Change - Outside Key Operation

General Description

This service allows users to switch the DAY/NIGHT mode by operation of an External key connected to the system.

Note: This feature is not available in North America and Australia.

Operating Procedure

To switch the DAY/NIGHT mode:

1. Switch the mode by External key connected to the system.

Service Conditions

1. An NCU is required to connect the External key.
2. The same package for this feature is also used for POWER FAILURE TRANSFER [P-3] (a maximum of 12 circuits per package).
3. Single tenant or multi tenant control by one point of contact can be selected by the system data.
4. Even when the system switches to night mode by use of an External key, all ordinary night time services such as Day/Night Class of Service will operate.
5. External Key Box must be provided by the customer.

Interactions

1. When this feature is enabled, the Day/Night switching feature of DAY-NIGHT MODE STATUS -D - II [D-164D - II] is not available.
   On the other hand, the Day/Night mode status display feature of DAY-NIGHT MODE STATUS -D - II [D-164D - II] is available.
2. This feature is not available when DAY-NIGHT MODE STATUS -D - II [D-164D - II] is enabled.

Programming

No programming is required.

D-80 Denied Tone Classification

General Description

This feature provides the Caller (Station, Attendant Console or Exchange Line), who is not connected with the Called Party, with an appropriate Distinctive Tone Connection, identifying the status (Busy, Restricted, Blocked) in which the Called Party is denying the Calling Party.

Note: This feature is only available in Australia.

Operating Procedure

No manual operation is required.

Service Conditions

1. This feature is available in Australia only.
2. Tone are classified as follows:
   a.) Number Unobtainable Tone (NUT): Dead Level, Dead Number, Restriction.
   b.) Congestion Tone (CGT): Trunk Block (not sent to the Distant Office).

      Note: CGT is also sent out where the OUTGOING TRUNK QUEUING [O-2] service is available. However, if OGTQ is automatically set, Service Set Tone is sent out instead.

   c.) Busy Tone (BT): Sent out in cases other than l and 2 above.

Interactions

There are no known Interactions.

Programming

ASYD - SYS1, Index 6, Bit 3 (Tone for Outgoing Connection): 0/1=NEC Standard/Australian (Second Dial Tone on LCR). Assign “1”.

D-81 Double Seizure Protection

General Description

This feature is provided to prevent the occurrence of double-line seizure, or the collision of Outgoing and Incoming Calls on the same TIE Lines.

Note: This feature is available in Australia only.

Operating Procedure

No manual operation is required.

Service Conditions

1. This feature is available in Australia only.
2. Double Seizure is identified if an Incoming Call is detected within 120 milliseconds after the TIE Trunk Seizure by an Outgoing Call.
3. If Double Seizure is detected, Busy Tone is sent to the Calling Station and the line is monitored for release. When the Calling Station is a Dterm, the LCD shows “BUSY”, together with the dialed number flashing at 60 IPM.
4. This feature does not operate with an Outgoing Connection from a Data Terminal.

Programming

ASYD - SYS1, Index 7, Bit 3 (Double Seizure Protection): 0/1=Out/In Service. Assign “1”.

D-87 Dial Intercom

General Description

This feature gives multiple Desktop terminal users the ability to simultaneously call any other member of the Intercom Group, regardless of their idle or busy status. An Intercom Group member can override another Intercom Station in a Two-party Intercom Connection. This service is referred to as an Intercom Bridge.

Operating Procedure

To originate

1. The Intercom Station presses the DICM key and goes off-hook within five seconds.
2. Dial the desired INTERCOM Telephone Number. Receive Ringback Tone. The Calling Desktop terminal displays:

Dialed Intercom Number

ICM	XX
(Time Display)	

3. The desired INTERCOM Station answers the call and is connected to the Calling Station.

To answer

1. The DICM key will flash, indicating an Incoming DIAL INTERCOM Call. The Called Station may also hear distinctive three-burst ringing. The Called Desktop terminal displays:

Calling Intercom Number

ICM	XX
(Time Display)	

2. If the Called Station is already engaged in a conversation with a Non-Intercom Call, the Station may press the HOLD key and answer the DIAL INTERCOM Call by pressing the DICM key. Ringing will stop.

To bridge

1. When two Stations are engaged in a DIAL INTERCOM Call, a third Station, belonging to the same DIAL INTERCOM Group, presses the DICM key and goes off-hook (**Note 1**). The Called and Calling Desktop terminals display:

Intercom Number

OVERRIDE	XX
(Time Display)	

Note 1: When ASYD, SYS1, Index 384, Bit 0=0 is set, you can skip off-hook operation to initiate Automatic Intercom Call or originate Manual Intercom Call. However, off-hook operation cannot be skipped to bridge into an Automatic Intercom Call or a Manual Intercom Call regardless of the ASYD, SYS1, Index 384, Bit 0 data setting.

2. A Three-party Conference is established. The Called Stations may hear a Warning Tone. The option for hearing the Tone is preassigned in System Data. The Called and Calling Desktop terminals display:



3. If Intercom Bridge Service is denied, the Desktop terminal displays:



Prime Line Pickup -D [P-13D]

The Desktop terminal may be programmed to seize the Incoming Call on the AUTOMATIC, MANUAL, or DIAL INTERCOM Lines before the Prime Line. If two Incoming Calls terminate simultaneously to both the Prime Line and the DIAL INTERCOM Line, when the user goes off-hook, PRIME-LINE-PICKUP -D [P-13D] will seize the DIAL INTERCOM Line before the Prime Line. If only the Prime Line is ringing when the user goes off-hook, then PRIME-LINE-PICKUP -D [P-13D] will seize the Prime Line. The programmable options and ringing priority are illustrated in the following table:

Ringing Priority	Programmable Options				
	PRI 0	PRI 1	PRI 2	PRI 3	PRI 4
1	Prime Line	Dial Intercom	Automatic Intercom	Manual Intercom	Automatic Intercom
2	--	Prime Line	Prime Line	Prime Line	Manual Intercom
3	--	--	--	--	Prime Line

Service Conditions

1. Only Desktop terminals can be assigned as a member of a DIAL INTERCOM Group.
2. All members of the same DIAL INTERCOM Group must be assigned in the same Module Group.
3. Up to 64 Stations may be assigned to one Dial Intercom group when two digits (00-63) Dial Intercom Station is selected via PCPro. Each Station in the DIAL INTERCOM Group can be assigned as an Automatic, Manual and a DIAL INTERCOM member of three different groups.
4. No more than one connection can be established at the same time in one DIAL INTERCOM Group.
5. A Three-party Conference can be established in a DIAL INTERCOM Group if Intercom Bridge Service is available. If a fourth party tries to activate Intercom Bridge Service, the connection is denied.
6. The maximum number of DIAL INTERCOM Groups in the System is 50 per Module Group.
7. It is possible to simultaneously assign one group of Stations as an Automatic/Manual Intercom Group, as well as a DIAL INTERCOM Group.
8. The Distinctive Ringing (three-burst ringing) sent from an Incoming DIAL INTERCOM Station takes priority over the ringing of Multi Line Appearances.
9. Features normally available to Multi Line Appearances are not available to DIAL INTERCOM Lines.
10. DIAL INTERCOM Lines cannot be held by pressing the HOLD key.

11. When a Station is moved from one DIAL INTERCOM Group to another via system commands, the DICM key on the associated Desktop terminal must be deleted before the change is made.
12. Port traffic and Station PEG COUNTS [P-7] reports will record the use of the DIAL INTERCOM feature under "other".
13. Dedicated Desktop terminal Feature keys cannot be used during a DIAL INTERCOM Call. Only the SPEAKER key for hands-free operation can be used.
14. If there are simultaneous Incoming Calls to the Prime Line, Automatic Intercom Line, the Manual Intercom Line, and/or the DIAL INTERCOM Line, the Prime Line will normally be selected when the Station goes off-hook. The selection order can be changed for each Desktop terminal via Office Data assignment.
15. DIAL INTERCOM Group Numbers are either one or two digits and can be set for each DIAL INTERCOM Group:

    One digit: 0-9
    Two digits: 00-63
16. The display on the Desktop terminal is as follows:

    a.) Calling

    Dial Intercom Number of the Calling Station
    (appears for ten seconds when the DICM Line is preselected)

    Called Station | **ICM** **XX** / **(Time Display)** | 60 IPM flashing

    Dial Intercom Number of the Calling Station

    Calling Station | **ICM** **XX** / **(Time Display)**

    b.) Called Station Answers

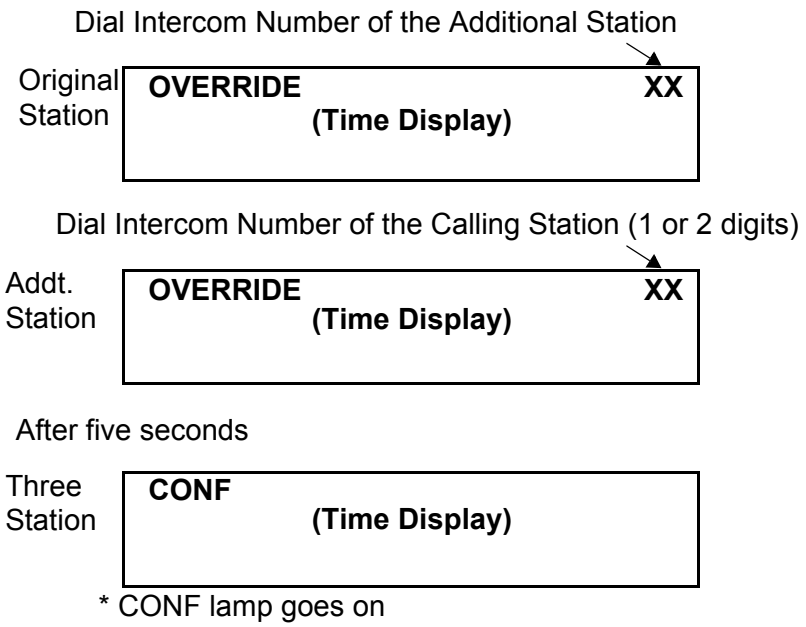
    Dial Intercom Number of the Calling Station

    Called Station | **ICM** **XX** / **(Time Display)**

    Dial Intercom Number of the Answering Station

    Calling Station | **ICM** **XX** / **(Time Display)**

c.) Intercom Bridge



* CONF lamp goes on

d.) Intercom Bridge Restriction



* When the DICM key is pressed for the bridge, after originating a call, the above is shown for five seconds.

17. To answer a call terminated to the INTERCOM Line while the Prime Line is busy, press an INTERCOM Line key.
18. The DIAL INTERCOM ringing by an Incoming Call is interrupted while the MANUAL SIGNALING [M-47] Signal is being sent out.
19. While a call is in progress on a Multi Line, the AUTOMATIC INTERCOM Line, or MANUAL INTERCOM Line, pressing the DICM key can disconnect the call to allow a DIAL INTERCOM Call origination or answer.
20. This feature does not work in FCCS network.

Programming

STEP 1: **ASYD** - SYS1, Index 59, Bit 7. Is AUTOMATIC/MANUAL and DIAL INTERCOM in service? 0/1: No/Yes.

SYS1, Index 69, Bit 6. Is a Warning Tone required when creating a three-way AUTOMATIC/MANUAL INTERCOM Call? 0/1: No/Yes.

SYS1, Index 78, Bit 0 and Bit 1. Assign both Bits as data "1" to enable full Desktop terminal display operation.

SYS3, Index 3. Assign data 20H for Multiple Ringings per Ringing Period, if necessary.

SYS3, Index 7. Assign the Ringing Pattern if necessary.

STEP 2: **ADIM** - Assign the following data:

MG: Module Group

G-ID: Group ID Number per MG

N: Number of Digits of Dial Intercom Group (0/1)
- 0: One digit
- 1: Two digits

BCR: Bridge Call Restriction (0/1)
- 0: Restricted
- 1: Allowed

TN: Tenant Number

ICMCT Number of Intercom Stations (2-64) (actual number to dial)

ICM STN: Intercom Station Number

STEP 3: **AKYD** - Assign an Intercom Line to the Station according to the following parameters:

TN: 1

STN: Station Number

TP: Type of Desktop terminal (0-3)

PRI: Form of Line Preference (0-3)

KYN: Key Number (1-40)

KYI: Service Index
- 0: Key Not Used
- 1: Feature Key
- → 2: Multi Line Key

KD: Kind of Line
- 0: Multi Line
- → 1: Intercom Line
- 2: DSS Console Key
- 3: -
- 4: Multiple Line Appearance

ICM: Kind of Intercom
- 0: Automatic Intercom
- 1: Manual Intercom
- → 2: Dial Intercom.

G-ID: Group ID Number

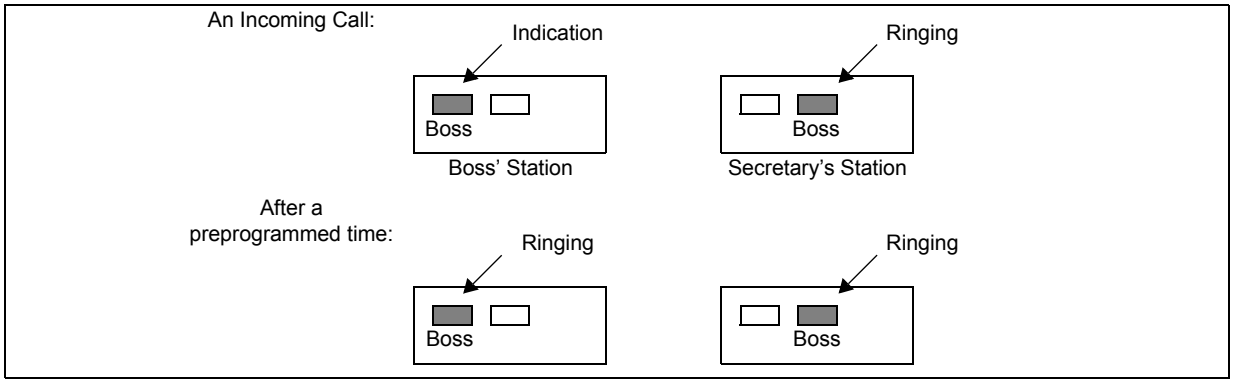
As programmed in ADIM

D-90D Delayed Ringing -D

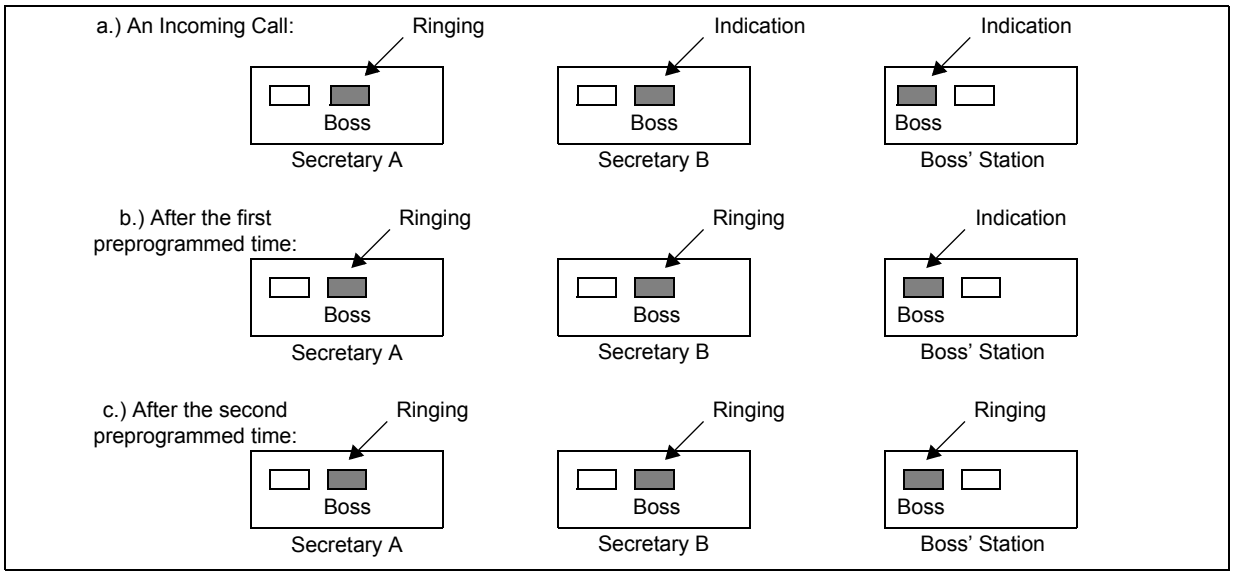
General Description

This feature allows a Desktop terminal to receive an Incoming Ringing Signal after a predetermined time.

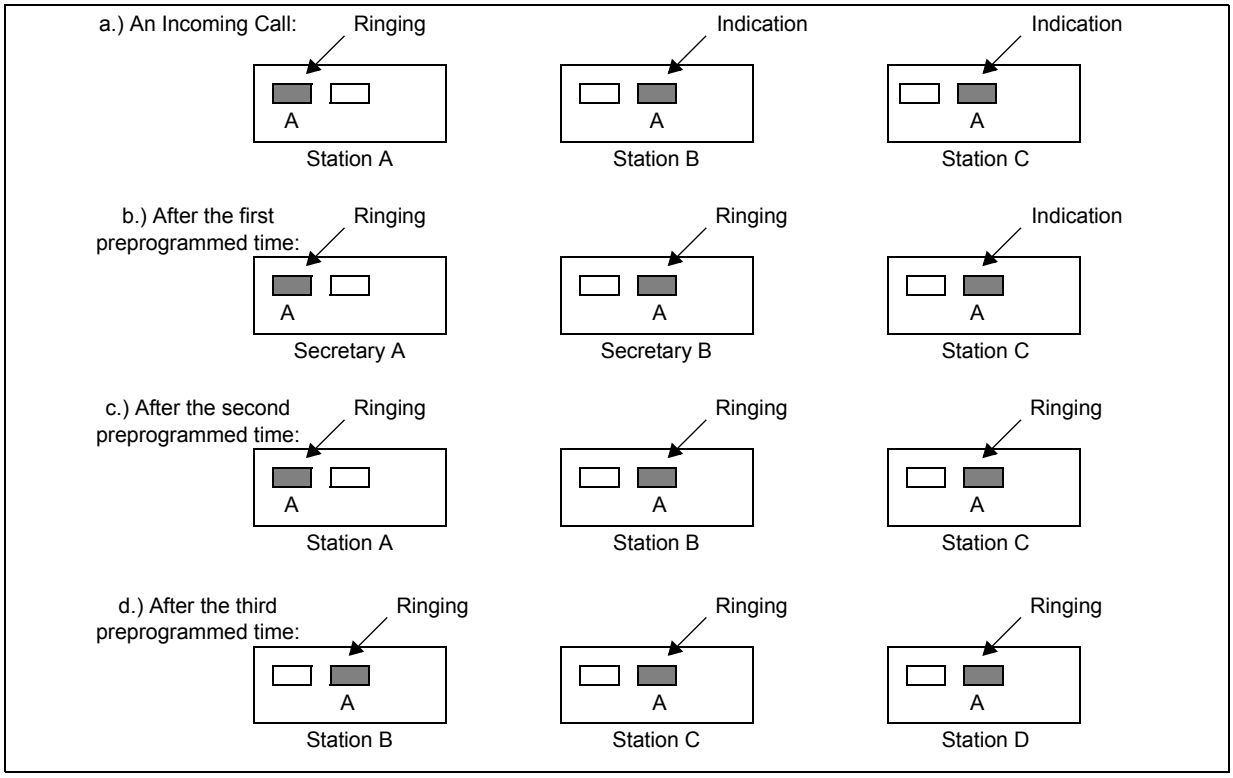
1. In Boss - Secretary Service, a call terminating on the Boss' Line first rings at the Secretary's Station. If the Secretary does not answer within a preprogrammed time, the Boss' Station will also ring. This service is convenient when many Lines terminate on a Secretary's telephone, and the Secretary is engaged in a call, or is temporarily away from the telephone, and cannot answer the call terminated to the Boss' Station.



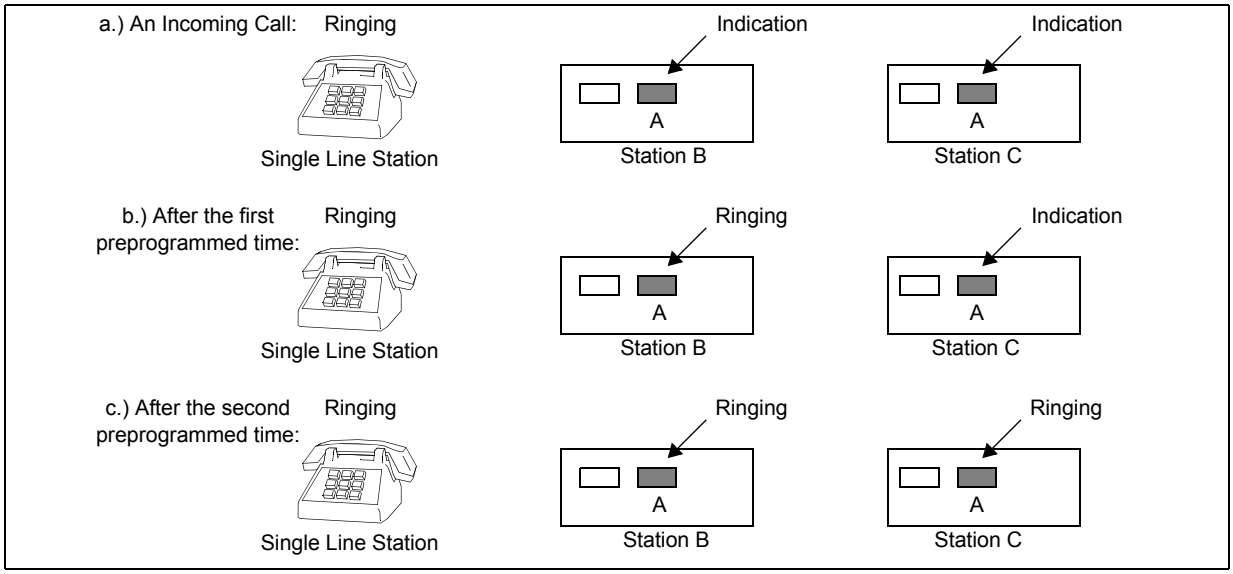
2. Calls terminated to the Boss' Station may be routed to multiple secretaries by assigning timing delays for ringing to each Secretary's Station.



3. When a call to the Boss' Station is received, it rings on Secretary A's Station. If it is not answered in a preprogrammed time, it will also ring on Secretary B's Station. If Secretary B does not answer within the second preprogrammed time, the call will also ring on Boss' Station.
4. This feature may also be used in a similar way to CALL FORWARDING - DON'T ANSWER [C-3] or STATION HUNTING [S-7, 8, 9].



5. A call rings on Station A. If Station A does not answer within a preprogrammed time, the call also rings on Station B. If the call is not answered within the second preprogrammed time, the call also rings on Station C. If the call is not answered within the third preprogrammed time, the call also rings on Station D.
6. DELAYED RINGING may also be used for calls terminated to Single-Line Appearances on a Desktop terminal. Calls to Single Line Appearances that are not answered within a preprogrammed time will ring on the next Station in sequence.



7. A call rings on Single Line Station A. If Station A does not answer within a preprogrammed time, the call also rings on Desktop terminal Station B. If the call is not answered within the second preprogrammed time, the call also rings on Desktop terminal Station C.

Operating Procedure

No manual operation of the feature is required after it has been programmed in System Data.

Service Conditions

1. This feature is available for Multi Lines on Desktop terminals.
2. Delay timing for each line on the Desktop terminal may be programmed through PCPro.
3. Delay timing may be 10, 20, or 30 seconds.

Interactions

1. Off-hook ringing on a Desktop terminal will follow the assigned DELAYED RINGING -D [D-90D] Timer the same as on-hook ringing.
2. Stations that have Multi Line Appearances with DELAYED RINGING -D assigned must be aware of the interaction with CALL FORWARDING - DON'T ANSWER [C-2]. For example, Station A appears on Desktop terminal B with a delayed ringing of 30 seconds (RG = 4); and also on Desktop terminal C with a delayed ringing time of 20 seconds (RG = 5). If Station A sets CALL FORWARDING - DON'T ANSWER [C-2] and the CALL FORWARDING - DON'T ANSWER [C-2] Timer is set at 24 seconds, when an Incoming Call comes into Station A, it will begin to ring. At 20 seconds Desktop terminal C will begin to ring also. At 24 seconds the call will forward to Station A's Target Destination and Station A and Desktop terminal C will stop ringing. Desktop terminal B never rings.
3. After a call has been answered by a Secretary, the activation of the Boss - Secretary feature by the Secretary to a Single Line Boss Station will not reactivate the DELAYED RINGING -D cycle.

Programming

AKYD - Assign the following parameters to Stations that will have delayed ringing:

TN: Tenant Number
STN: Station Number
TP: Type of Dterm (0-3)
PL TN STN: Prime Line Tenant and Station Number
KYN: Key Number (1-40)
KYI: Service Index
0: Key Not Used
1: Feature Key
→ 2: Multi Line Key
KD: Kind of Line
→ 0: Multi Line
1: Intercom Line
2: DSS Console Key
3: -
4: Multiple Line Appearance
TN STN: Tenant Number and Station Number of Multi Line Appearance
RG: Ringing Assignment of Multi Line Appearance
0: No Ringing
1: Ringing Day Only
2: Ringing Night Only
3: Not Used
4: 30-second delay time before ringing
5: 20-second delay time before ringing
6: 10-second delay time before ringing
7: Ringing Day and Night (immediate)

D-114 Delay Announcement - Attendant

General Description

This feature provides a Recorded Announcement to the Incoming Trunk Caller waiting to be answered by the Attendant. This feature is divided into the following two types, depending on the time when the call is connected with the Announcement Trunk.

1. The call is connected to Announcement Trunk immediately after terminating to the Attendant Console. The announcement continues until the Attendant answers, or the Trunk side abandons the call.
2. The call is connected to the Announcement Trunk after the Ringback Tone Connection.

Operating Procedure

When the Announcement Service is provided immediately after the call terminates

1. When an Incoming Call terminates to an Attendant Console, which is handling calls, the Calling Party receives an announcement from the Digital Announcement Trunk (DAT).
2. When the Attendant completes call handling, ringing starts at the Attendant Console.
3. The Attendant presses the ANSWER key. The Calling Party and the Attendant talk with each other.

When the Announcement Service is provided after a fixed time

1. When an Incoming Call terminates to an Attendant Console, the LDN key lamp starts to flash and ringing starts at the Attendant Console, and the Calling Party receives the Ringback Tone.
2. When the Attendant has not answered in the designated time (SYS1, Index 242), the call is connected to the Announcement Trunk automatically (the lamp keeps flashing and the ringing continues at the Attendant Console).
3. If the Attendant does not answer the call when the first announcement completes, the Trunk Call disconnects from the announcement automatically and the caller hears the music on hold.
4. The Attendant presses the ANSWER or LDN key. The Calling Party and the Attendant talk with each other.

Service Conditions

When the Delay Announcement Service is provided immediately after the call terminates

1. This service requires a DAT card or VS32.
2. This feature is available for the following Incoming Trunk types:
   a.) Ringdown
   b.) LDN (DID, TIE, ISDN)
   c.) Call Forwarding - All Calls/Busy Line/Don't Answer (DID, TIE, ISDN)
   d.) Call Forwarding - Intercept - Attendant (DID, TIE, ISDN)
   e.) Central Office Line, TIE Line Incoming Call via CCIS network.

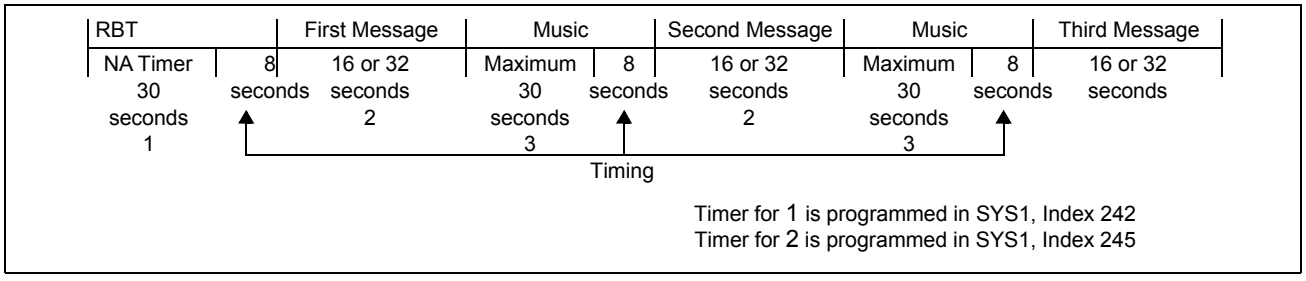
   Note: This feature is not available for the following Incoming Trunk types transferred to the Attendant Console by Call Forwarding-All Calls/Busy Line:
   - Direct Inward Dialing (DID)
   - Direct-In Termination (DIT)

3. If IPPAD is all busy while the system sends announcement, the attendant can answer when IPPAD becomes available.

4. This feature is not available for TAS Incoming Calls.
5. When the system is in NIGHT mode, this service is not available.
6. At the time of DAY/NIGHT mode Changeover, the call connected to the announcement will be released, and the Calling Party receives Reorder Tone. A call will be released as follows, according to the Incoming Trunk type:
   a.) When a call terminates from a Trunk where Trunk release cannot be supervised, the Trunk will be released forcefully (Night Transfer Station cannot be assigned).
   b.) When a call terminates from a Trunk where Trunk release can be supervised, the Trunk will not be released forcefully, but will be released on its own accord (Night Transfer Station can be assigned).
7. This service is not available for Incoming Calls to an Individual Attendant.
8. When a Central Office Line Incoming Call terminates to a Trunk where Call Clear cannot be confirmed after the call has been answered, the Incoming Trunk and Announcement Trunk remain connected until Attendant answers and releases the Trunk.
9. When a call terminates to an Incoming Trunk, for which this service is not required, the Connection Route between the Incoming Trunk and the Announcement Trunk is restricted.
10. The Announcement Trunk remains connected until the Incoming Trunk is released (Timer Control Release is not used).
11. In case of Multiple Connection:
    a.) Caller is not connected at the beginning of the announcement.
    b.) The Announcement Trunk is not released by Attendant upon answering or releasing the Incoming Trunk.
12. For Single Connections, this service is not available when the Announcement Trunks all are busy.
13. When an External Announcement Machine is used, the announcement starts from the time the System acknowledges the Announcement Trunk answer, and ends at the time of the Announcement Trunk release.
14. Whether to use an Announcement Trunk for Single or Multiple Connection is determined by the System Data on a Tenant basis.

When the Announcement is provided after the Ringback Tone has been heard

1. This service requires a DAT card. VS32 is not supported for this feature.
2. When the caller is hearing the message or music, the call is still indicated on the ICI key on the Attendant Console. Therefore, the Attendant can answer the call anytime by pressing the ANSWER key or the corresponding ICI key.
3. The message is sent in the following sequence. Whether to send the second message and after can be determined in the System Data. When the other messages are not sent after the first announcement, MUSIC ON HOLD [M-7] continues.



Note: Separate timers are used for No Answer Timer and Music On Hold.

4. Each channel of the DAT can record a single message. Recording time for a single message is 16 or 32 seconds.
5. When Multiple Playback is selected, additional DAT Channel is required for the second announcement, even if the second announcement has the same contents as the first one.
6. A maximum of ten Incoming Trunk Calls can be connected to a single DAT Trunk at one time for sending a prerecorded message from the beginning to the Called Parties.
7. The maximum programmable Channels of DAT for this feature are five Channels for the first announcement, and five Channels for the second announcement.
8. When a call encounters Announcement Trunks all busy, the call waits in a queue until a Trunk becomes available. The latter Trunk Call hears Ringback Tone until the announcement to the former terminated call completes.
9. When providing the first message, the Answer Signal is sent to the Opposite End (the billing starts here). In the case of Central Office Trunks, where Trunk Release cannot be detected after the call has been answered, the display on the Attendant Console continues even after the Calling Party has gone on-hook (the Trunk does not release). Therefore, the Incoming Call should always be answered by the Attendant.
10. In the case of billing for the Incoming Call, billing starts when the caller is answered by the Attendant, not when connected to a message.
11. When the call terminates to the Attendant Console at the time of DAY/NIGHT mode Changeover, this service is activated as follows:

DAY/NIGHT Mode Change	Call Processing
Attendant Console in Day mode to Night Attendant Console	The Announcement Service is provided after the call is transferred to the Night Attendant Console and the Night Attendant Console answers the call.
Attendant Console in Day mode to Attendant Console of Night Connection - Fixed [N-1]/Flexible [N-2]	The call stays at the Attendant Console in Day mode and the Announcement Service is provided. The Attendant Console answers the call.
Night Attendant Console to Attendant Console of Night Connection - Fixed [N-1]/ Flexible [N-2]	The call stays at the Night Attendant Console and the Announcement Service is provided. The Night Attendant Console answers the call.

12. This feature is not available for the following calls:
    a.) AUTOMATIC RECALL [A-8] Calls.
    b.) calls from an Attendant (ATTENDANT-TO-ATTENDANT CALLING [A-60]/INTER-POSITION TRANSFER [I-5]).
    c.) calls from a Station (DIAL ACCESS TO ATTENDANT [D-2]/OFF-HOOK ALARM [O-6]/ PRIORITY CALL [P-9]).
    d.) Calls transferred to the Attendant due to unused number dialing.
    e.) Calls transferred to the Night Station via ATTENDANT NIGHT TRANSFER [A-19].
    f.) Calls via CALL FORWARDING - ALL CALLS [C-5]/BUSY LINE [C-2]/DON'T ANSWER [C-3]
    g.) Incoming calls via CCIS/TIE Line.
    h.) TAS Incoming Calls.
    i.) DIRECT INWARD DIALING (DID) [D-8] Calls.
    j.) DID ADDRESSING [D-123] Calls.
13. This service is not available for Incoming Calls to an Individual Attendant.

14. This service is not activated for the Attendant of NIGHT CONNECTION - FIXED [N-1] or NIGHT CONNECTION - FLEXIBLE [N-2].
15. The message is sent only once. If the Attendant does not answer by the end of the message, the caller will be disconnected from the Announcement Trunk automatically and will hear MUSIC ON HOLD [M-7].
16. CALL QUEUING [C-9] and CALL WAITING LAMP [C-13] Services are available when this service is activated.
17. The Calling Party cannot receive the message if the restriction between the Incoming Trunk and the Announcement Trunk is not cleared.
18. When this service is allowed in System Data, the ATTENDANT OVERFLOW [A-105] feature cannot be activated.
19. When the External Announcement Machine is used, the announcement starts from the time when the System acknowledges the Announcement Trunk answer, and ends at the time of the Announcement Trunk release.
20. When activating this feature in an FCCS Network, the announcement device and the Called Attendant Console must be accommodated in the same Node.

Programming

When the Delay Announcement Service is provided immediately after the call is terminated

STEP 1: **ASYD** - SYS1, Index 244, Bit 5=0 (announcement is provided immediately after the Trunk Call is terminated to the Attendant Console)

STEP 2: **ARTD** - Assign the Route Data of the Announcement Trunk

1-OSGS	:2	2-ONSG	:3	5-TF	:1	6-TCL	:5	7-L/T	:1
8-RLP	:2	15-LSG	:5	28-ANS	:0 **Note 1**	44-PRV	:1		

Note 1: When the External Announcement Equipment is used, assign “1” in ANS parameter.

Note: To change the data for existing route (RT), initialize or reboot the trunk device such as circuit card or VS32.

STEP 3: **ATRK** - Assign the Trunk Data of the Announcement Trunk.

STEP 4: **MBTK** - Cancel the make-busy of the Trunk assigned by ATRK.

STEP 5: **AAED** - Assign the data to the Announcement Unit Number and the Route Data, etc. assigned by ATRK.

EQP: 32
C: 1
R: 1
A: 0
M: 0
SS: 0
RT: Route Number

STEP 6: **ASTD** - Allow Tandem Connection between COT and the Announcement Trunk. STM = 0, STS = 2, ST = 1

STEP 7: **ARRC** - Cancel the restrictions on the Trunk Route for Incoming Trunk and the Route for the Announcement Trunk.

ICRT: Route Number for Incoming Trunk

OGRT: Route Number for Announcement Trunk
ARI-D-RES: 1

STEP 8: **ALDN** - Assign the Listed Directory Number [L-24] of the Console if using DID, TIE, or ISDN.

When the Delay Announcement Service is provided after the Ringback Tone:

STEP 1: **ASYD** - SYS1, Index 242. Timer for Delay Announcement - Attendant (Duration of RBT)
Timer for duration of RBT = TC × MTC. TC = 3.2 seconds. MTC = 1-15
SYS1, Index 244. Bit 5=1 (announcement is provided immediately after the Ringback Tone)
SYS1, Index 245. Timer for Delay Announcement - Attendant (Duration of Music on Hold)
Timer for duration of RBT = TC × MTC. TC=3.2 seconds. MTC = 1-15
SYS1, Index 246. Bit 2 (Attendant Delay Announcement): 0/1= Multiple Playback/Single Playback

STEP 2: **ARTD** - Assign the Route Data of the Announcement Trunk

1-OSGS	:2	2-ONSG	:3	5-TF	:1	6-TCL	:5	7-L/T	:1
8-RLP	:2	15-LSG	:5	28-ANS	:0 **Note 2**			44-PRV	:1

Note 2: When External Announcement Equipment is used or Multiple Playback is selected by key setting on the DAT Card, set 1 to CDN 28 (ANS).

Note: To change the data for existing route (RT), initialize or reboot the trunk device such as circuit card or VS32.

Note: For the Incoming Route, CND 6-TCL must be assigned “1”.

STEP 3: **ATRK** - Assign the Trunk data of the Announcement Trunk.

STEP 4: **MBTK** - Cancel the make-busy of the Trunk assigned by ATRK.

STEP 5: **AAED** - Assign the data to the Announcement Unit Number and the Route Data, etc. assigned by ATRK.
EQP = 49, 50, 51, 52, 53 (for the first announcement)
= 54, 55, 56, 57, 58 (for the second announcement)
RT: Route Number
TK: Trunk Number
C: 1
R: 1
A: 1
M: 1 (Multiple Connection)

STEP 6: **ASTD** - Allow Tandem Connection between COT and the Announcement Trunk. STM - 0, STS = 2, ST = 1

STEP 7: **ARRC** - Cancel the restrictions on the Trunk Route for Incoming Trunk and the Route for the Announcement Trunk.
ICRT: Route Number for Incoming Trunk
OGRT: Route Number for Announcement Trunk
ARI-D-RES: 1

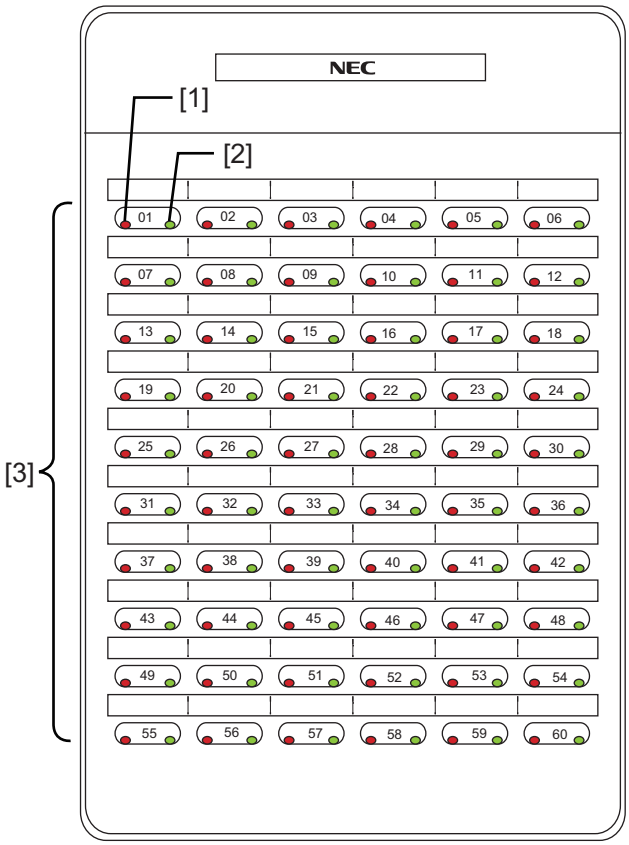
STEP 8: **ALDN** - Assign the Listed Directory Number [L-24] of the Console if using DID, TIE, or ISDN.

D-149 Direct Station Selection (DSS) Console

General Description

Direct Station Selection (DSS) Console has 60 keys, each of which has two lamps. DSS Console can be used in combination with a digital terminal or a SIP multiple line terminal.

Keys, Lamps and Functions of Direct Station Selection (DSS) Console



[1] Red Lamp (Total 60 Lamps)

- When Multiple Lines are set:
  OFF (Idle) **Note 1** / ON (The other station is talking) / Flash (On hold)
- When the Feature keys are set:
  OFF (The feature set to the key is invalid.) / ON (The feature set to the key is valid. / Flash (The feature set to the key is valid.)

[2] Green Lamp (Total 60 Lamps)

- When Multiple Lines are set:
  OFF (Idle) **Note 1** / ON (Talking) / Flash (On hold)
- When Feature keys are set:
  OFF (The feature set to the key is invalid.) / ON (The feature set to the key is valid. / Flash (The feature set to the key is valid.)

[3] 01 - 60 keys

(key numbers are not actually imprinted.)

- Used as:
  The programmable Line keys
  The programmable Feature keys
- The programmable Line keys for internal calls and external calls, and the Feature keys can coexist in one DSS Console.

Note 1: The key is in idle state only when both red and green lamps are OFF.

DSS Console has following two operating modes:

- Direct Station Selection (DSS) / Busy Lamp Field (BLF) Mode

  Connecting DSS Console to a Desktop terminal adds more Line keys to the terminal. A Desktop terminal that is attached to DSS Console can transfer a call by pressing a key on the console that is assigned to a desired station number. This is called Direct Station Selection (DSS) mode.
  Each key on DSS Console has a pair of LED lamps to indicate whether the station that is assigned to that key is BUSY or IDLE. This is called Busy Lamp Field (BLF) Mode.

  Note: For a SIP multiple line terminal, DSS/BLF mode is not supported for FP95-114 V4 or earlier. DSS/BLF mode is supported since FP95-114 V4 Issue 2.0.

- Add-On Module (ADM) Mode

  Connecting DSS Console to a Desktop terminal can practically add up to 60 Line/Feature Access keys to the terminal. Available lines keys include Multi-Line keys and Feature keys.

Either mode is assigned to each console depending on connected LENS for DSS Console and the Office Data.

For other information such as operation procedures or service conditions of terminals, refer to the following headings:

Direct Station Selection (DSS) Console for SIP Multiple Line terminal

Direct Station Selection (DSS) Console for Digital Terminal

Direct Station Selection (DSS) Console for SIP Multiple Line terminal

This section explains service conditions and programming procedure for Direct Station Selection (DSS) Console used in combination with a DT800/DT700 Series.

Operating Procedure

To originate a call using SPEED CALLING - ONE TOUCH:

1. User A lifts the handset or presses the SPEAKER key on a terminal.
   User A hears the dial tone.
2. User A presses a key on DSS Console where a station number of User B to whom User A wants to call is assigned.
   User A hears the ringback tone from the terminal. A terminal of User B starts ringing.
3. User B answers the call.
   They can start a conversation.

To transfer the call to the Station assigned on the DSS key:

1. While User A and User B are engaged in a call, User A goes on-hook. **Note 2**
2. User A presses a key on DSS Console where a station number of User C to whom User A wants to transfer the call is assigned.
   User A hears a ringback tone from the terminal. User B is placed on hold. A terminal of User C starts ringing.
3. User C answers the call.
   User B and User C can start a conversation.

Note 2: Going on-hook is not a required operation. You can still transfer a call by pressing a key on DSS Console without going on-hook first.

Service Conditions

<Important Notices>

1. DSS/BLF mode is available since FP95-114 V4 Issue 2.0. For FP95-114 V4 or earlier, ADM mode only is available.
2. Make sure that a DT800/DT700 Series and DSS Console are turned off when connecting them to each other. After the connection process is completed, connect the AC adapter to DSS Console, and then perform the registration of the DT800/DT700 Series on the Telephony Server. To connect DSS Console to a DT800/DT700 Series, refer to Peripheral Equipment Description (Multi-Line Telephone).
3. Power must be provided to DSS Console as well as a DT800/DT700 Series (Power supply by AC adapter).
4. To use 60 keys as DSS Console, two lines of key data needs to be assigned by the AKYD command. Refer to STEP 3: for DSS/BLF mode and STEP 2: for ADM mode of the AKYD command.
5. When DSS Console calls a station on which Call Forwarding is set, if the forwarding destination is a PS that is roaming at that moment, the incoming call will not arrive at the PS. Instead of it, out-of-zone features such as Automatic Announcement - Called PS Out of Zone [A-129] will be performed.
6. For ADM mode, up to 16 Line/Feature keys can be used on the DT800/DT700 Series side. This setting is assigned by the ADKS command. Refer to STEP 3: the ADKS command.

7. In DSS/BLF mode, Call Forwarding via trunks is available on DSS Console by pressing a key where a station whose forwarding destination for Call Forwarding - All Calls - Outside, Call Forwarding - Busy Line - Outside, and Logged Out IP Station - Call Destination is an outside number.

   Call Forwarding conditions are as follows:

   - The system data (ASYDL/ASYDN, SYS1, Index 686, Bit 2=1) is required.
   - When a forwarding destination of the station is an attendant console, Call Forwarding is restricted.
   - When a forwarding destination number is Uniform Numbering, Call Forwarding is restricted.
   - LCR, LCRS, OGC, and OGCA are supported for the originating methods.
   - When a forwarding destination is a station, Call Forwarding - All Calls, Call Forwarding - Busy Line, and Call Forwarding - Don't Answer are supported. However, it is not supported when the forwarding destination is an attendant console.
     When a forwarding destination is a trunk line, calls for Call Forwarding - All Calls - Outside, Call Forwarding - Busy Line - Outside, and Logged Out IP Station - Call Destination can be forwarded when:

     - a transferor is hearing a dial tone

     - a transferor is on a call

     - a transferor goes on-hook while on a call

   - As Split Call Forwarding is not available, the call is transferred to the forwarding destination of Call Forwarding for trunk calls.

Note: Call Forwarding via trunks in DSS/BLF mode is available since FP95-114 V4 Issue 2.0.

<Interactions with Other Features>

1. For ADM mode, do not assign the following parameters to Function Key Number (FKY) when the Line/Feature keys are assigned on DSS Console by the AKYD command.
   - FKY = 6 (Message Reminder)
   - FKY = 49 (Speed Calling - One Touch)
   - FKY = 75 to 82 (Call Park Group-D key)
   - FKY = 118 (Call Return)
   - FKY = 163 (Privacy key)
2. When LOCATION DIVERSITY [L-55] is in active, DSS Console can be used in a Primary Node.
   If the terminal performs re-registration to a Backup Node, DSS Console which is used in combination with a DT800/DT700 Series is not included in service.
3. When the user of the station with DSS Console calls the station assigned on the DSS key by SPEED CALLING - ONE TOUCH, or transfers the call to the station assigned on the DSS key, and the station assigned on the DSS key is the member of the UCD [U-1] or Station Hunting Group, the call follows the UCD or the Hunting Operation. (If the MCI is valid, the message is sent out.)
4. When the station assigned on the DSS key that becomes the call destination or the forwarding destination is not in an idle state but in a call-block state, a lockout state or a make-busy state, BUSY is displayed on the originating terminal.
5. For DSS/BLF mode, the following features do not work after a call is originated by using DSS keys.
   - CALL BACK [C-1]
   - CALL WAITING - TERMINATING [C-12]
   - CALL WAITING - ORIGINATING [C-31]
   - EXECUTIVE RIGHT-OF-WAY [E-1]
   - MESSAGE REMINDER [M-19]

- STEP CALL [S-13]
- VOICE CALL [V-2]

<Hardware/Software>

1. Multiple DT800/DT700 Series cannot share one DSS Console.
2. DSS Console can be used in combination with the following DT800/DT700 Series.

Terminals		Firmware	Connection Condition
DT710	2-button/6-button Type	-	Not supported
	8-button Type (DESI-LESS)		Not supported
DT730 **Note 3**	12-button Type	FW4.0.0.0 or later	Supported
	24-button Type **Note 4**		Supported
	8-button Type (DESI-LESS)		Supported
DT730G	12-button Type (12DG/12CG)	No conditions	Supported
	24-button Type (24DG/24CG) **Note 4**		Supported
DT750 **Note 3**		FW4.0.0.0 or later	Supported
DT770G		-	Not supported
DT820	6-button Type	-	Not supported
	8-button Type (DESI-LESS)		Not supported
DT830 **Note 3**	12-button Type	For DSS/BLF mode: FW4.0.0.0 or later For ADM mode: FW3.0.0.0 or later	Supported
	24-button Type **Note 4**		Supported
	8-button Type (DESI-LESS)		Supported
DT830G	12-button Type (12DG/12CG)		Supported
	24-button Type (24DG/24CG) **Note 4**		Supported

Note 3: You cannot use DSS Console in combination with the following terminals.
- The terminal with ACD button unit attached
- The terminal with Hotel button unit attached

Note 4: 8LK Unit is also available.

<Interactions with Networks>

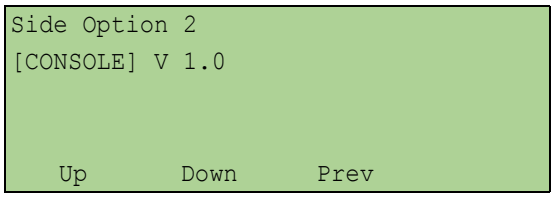
None

<Conditions Specific to This Feature>

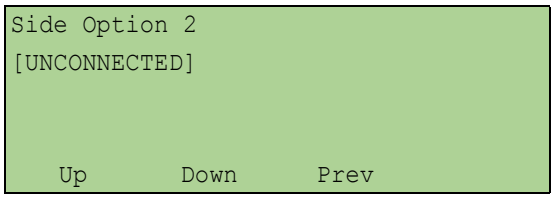
1. The IPAN command (Display of IP Machine Status Display Function) does not show registration information and version on DSS Console side.

2. To view the version of DSS Console on the DT800/DT700 Series LCD screen, access the Terminal Information Menu, following the steps presented below:
   [1] Hold down the HELP key on the DT800/DT700 Series. The Online Monitoring Menu will open.
   [2] Press the digit key "2" to go to the System Information Menu.
   [3] Press the digit key "5" to display the Terminal Information Menu.
   [4] Use Soft keys for "Up" or "Down" to move the page up or down, and check the version of DSS Console on "Side Options 2".

   Example 1: Case where DSS Console is attached to the [SIDE2] connector on the back of DT800/DT700 Series



   Example 2: Case where no device is attached to the [SIDE2] connector



Note: The following terminal information is available:

Menu	Item	Description
Terminal Information Menu	Hardware Version	Hardware version of the DT800/DT700 Series
	MAC Address	MAC address of the DT800/DT700 Series
	Firmware Version	Firmware version of the DT800/DT700 Series
	Bottom option	ADA Unit version connected to the DT800/DT700 Series
	Side Option 1	Version of the optional device connected to the Option 1 connector on the back of the DT800/DT700 Series *e.g. 8LK Unit
	Side Option 2	Version of the optional device connected to the Option 2 connector on the back of the DT800/DT700 Series *e.g. DSS Console

3. When you originate a call to DSS Console station number, you will hear Reorder Tone (ROT). If a Call Forwarding feature is set to DSS Console station number, Telephony Server operates in the following way regardless of DSS Console state.

Call Forwarding	Operation
CALL FORWARDING - ALL CALLS [C-5]	Forwarded

Call Forwarding	Operation
CALL FORWARDING - DON'T ANSWER [C-3]	Not Forwarded **Note 5**
CALL FORWARDING - BUSY LINE [C-2]	Forwarded
LOGGED OUT IP STATION - CALL DESTINATION [L-53]	Forwarded

Note 5: Calling Party's display shows a busy connection state.

4. When a user of a DT800/DT700 Series paired with DSS Console is already logged in to the system, another IP/SIP Multiple Line terminal user who is attempting to log in by using the station number of DSS Console will fail and return to the login screen. (Login inhibited.)
   If MAC authentication is implemented, the IP/SIP Multiple Line terminal will repeat registrations.
5. When an IP/SIP Multiple Line terminal user is already logged in to the system by using the station number of DSS Console, if another user of a DT800/DT700 Series paired with DSS Console logs in, the console cannot be used.
6. When a user of a DT800/DT700 Series paired with DSS Console is already logged in to the system, another user who needs to use the station number currently used by the mentioned terminal must perform either of the following steps before connection:
   a.) Log out the current user of the station number by using the LOGOUT key.
   b.) Disconnect the LAN cable from the IP/SIP Multiple Line terminal and turn the power off. This causes the system to detect Health Check NG. (System message 33-S will be output.)
   c.) Set the terminal to the Configuration mode.
7. If a currently logged-in user of a DT800/DT700 Series paired with DSS Console makes a change with the AKYD command, the DT800/DT700 Series performs re-registration, and **Line Busy** is displayed.
   This re-registration also occurs when a change is made with the ASLG command in such a situation. However, **Line Busy** is not displayed.
8. To remove DSS Console, use the AISTL command to delete data from both the terminal that is attached to the console and console itself. The order of deletion is irrelevant. If you wish to keep the data on the terminal, delete the data on DSS Console, and register the terminal again.
9. To connect DSS Console to a terminal while the terminal is operating, add the data of DSS Console to Telephony Server, and register the terminal again.
10. A SIP multiple line terminal with DSS Console in any mode spends one client license.
11. For ADM mode, My Line/Prime Line must be assigned within 16 Line/Feature keys. (However, when the 12-button type terminal is used, My Line/Prime Line must be assigned within 12 Line/Feature keys.)

<Other Conditions>

None

Programming

<For DSS/BLF mode>

<For ADM mode>

<For DSS/BLF mode>

This section describes how to assign DSS/BLF mode to DSS Console and DT800/DT700 Series that is attached to the console. The data programming is explained using the example below:

Note: Note the following conditions:
- Available since FP95-114 V4 Issue 2.0.
- Set the level of LENS for DT800/DT700 Series between level 0 and level 5, and then accommodate the two LENS for DSS Console from there.

Example of Data Programming

TERMINAL	LINE EQUIPMENT NUMBERS (LENS)	STATION NUMBER (STN)
DSS Console-2	060102	3099
DSS Console-1	060101	3098
DT800/DT700 Series	060100	3000

Perform the following procedure. Note that X indicates a value that should be tailored to the existing system.

STEP 1: **ASYDL/ASYDN** - SYS1, Index 686
Bit 1=1: DSS/BLF mode for DT800/DT700 Series is available.
Bit 2=1: Call Forwarding - Outside of a station assigned to the DSS key is available.

Note: Available since FP95-114 V4 Issue 2.0.

STEP 2: **AISTL** - Assign a station number of DT800/DT700 Series and DSS Console.

[Assignment of the station number of DT800/DT700 Series (3000)]
TN: X
IP STN: 3000 ← Station Number of DT800/DT700 Series
TEC: 12 ← Fixed
KIND: SIP MultiLine TEL
RSC: X
SFC: X
LENS: 060100 **Note 6**

[Assignment of the station number of DSS Console-1 (3098)]
TN: X
IP STN: 3098 ← Station Number of DSS Console-1

TEC: 12 ← Fixed
KIND: SIP MultiLine TEL
RSC: X
SFC: X
LENS: 060101 **Note 7**

[Assignment of the station number of DSS Console-2 (3099)]
TN: X
IP STN: 3099 ← Station Number of DSS Console-2
TEC: 12 ← Fixed
KIND: SIP MultiLine TEL
RSC: X
SFC: X
LENS: 060102 **Note 8**

Note 6: Set the level of LENS for DT800/DT700 Series between level 0 and level 5.

Note 7: Set the LENS next to DT800/DT700 Series.

Note 8: Set the LENS next to DSS Console-1.

STEP 3: **AKYD** - Assign the station numbers to the Line/Feature Access keys for DT800/DT700 Series (STN=3000), DSS Console-1 (STN=3098) and DSS Console-2 (STN=3099) which are assigned by AISTL command.

Note: **Line Busy** is displayed when the data assignment, but the data is assigned correctly.

a.) Data Assignment of the DT800/DT700 Series for DSS Console

To assign information of these keys, use the port of the DT800/DT700 Series that is paired with DSS Console.
TN: X
STN: 3098/3099 ← Station Number of DSS Console
PRI: 0
PL TN: X
PL STN: 3000
S: 0 (Off-hook Suppression is OFF)
MWD: X (Message Waiting Data Display Selection)
LN PRE: 0 (Prime Line Pickup)
TP: 1

[Data Assignment for KYN=1]
KYN: 1 (Assign DT800/DT700 Series Prime Line.)
KYI: 2 (Multi-line)
KD: 0 (LINE)

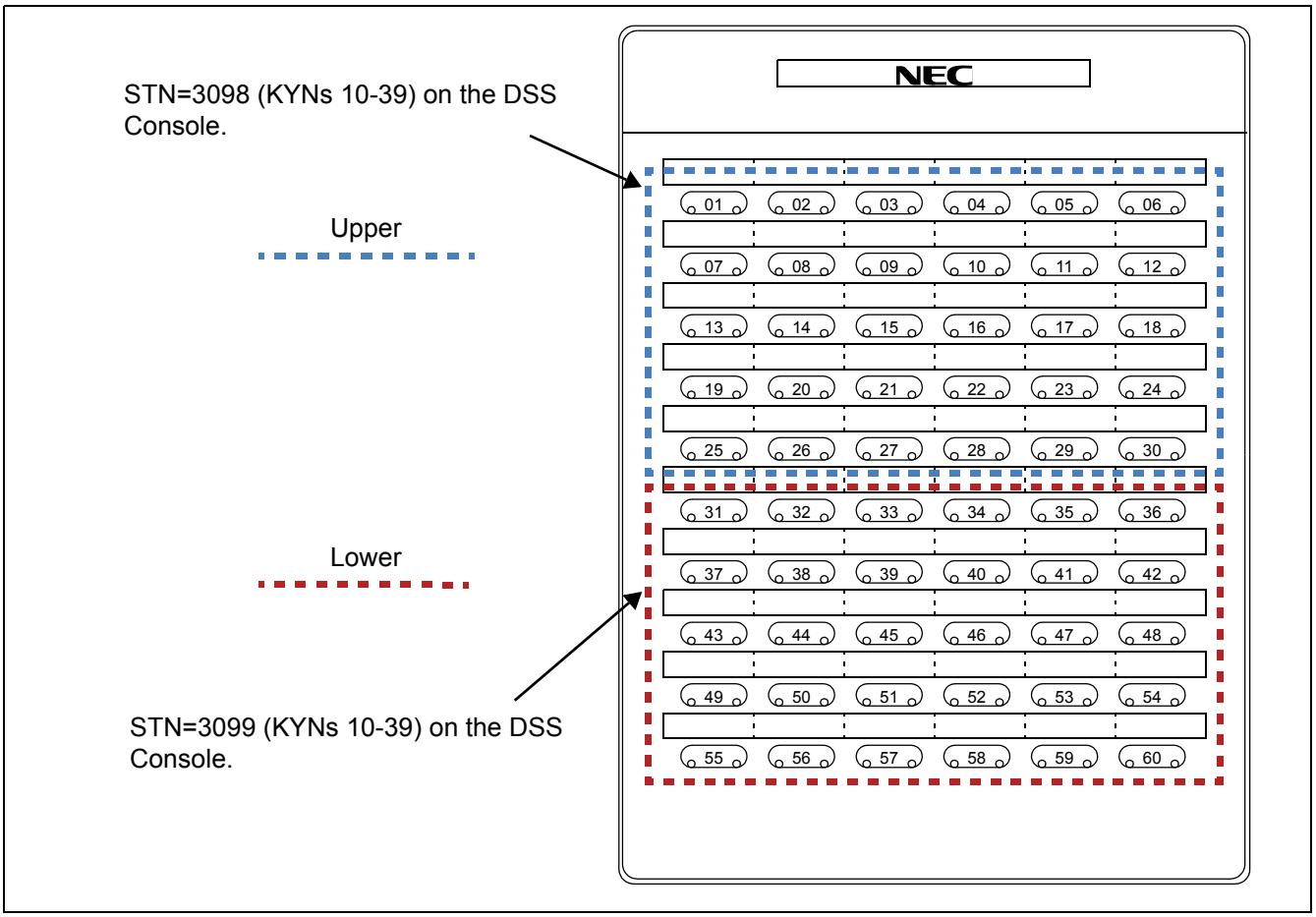
ML TN: Multi-line Tenant Number
ML STN: 3098/3099 (Station Number of DSS Console)
RG: 0-7 (Ring Information for Each Line)

[Data Assignment for KYN=2]
KYN: 2 (Assign DSS Console station number.)
KYI: 2 (Multi-line)
KD: 0 (LINE)
ML TN: Multi-line Tenant Number
ML STN: 3000 (DT800/DT700 Series station number paired with DSS Console)
RG: 0-7 (Ring Information for Each Line)

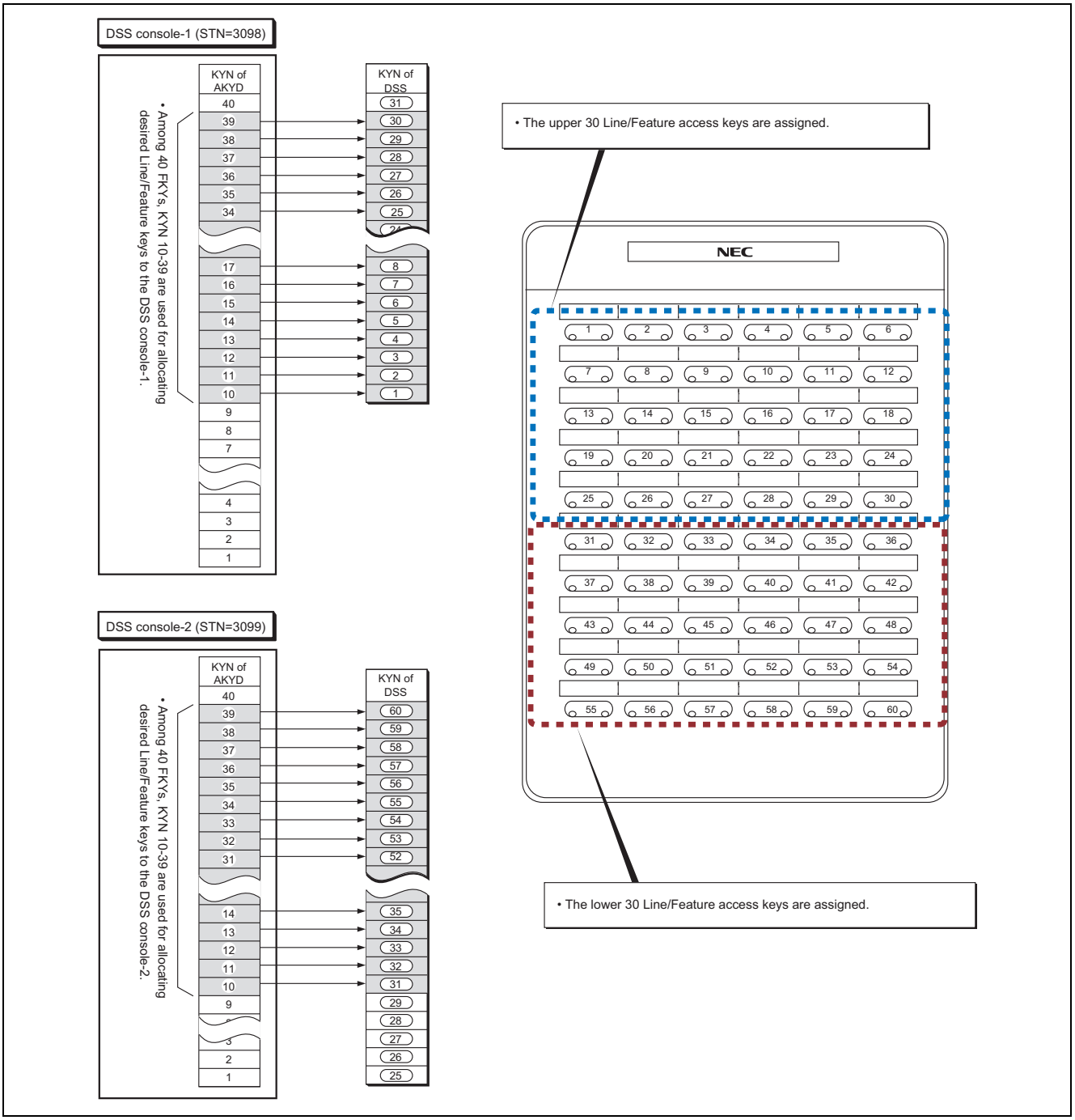
[Data Assignment for KYN=10 to 39]
KYN: 10-39 (Assign KYNs 10-39 keys on DSS Console.)
KYI: 2 (Multi-line)
KD: 2 (DSS Console Key)
ML TN: Multi-line Tenant Number
ML STN: Station number for use
RG: 0-7 (Ring Information for Each Line)

Note: Execute the AKYD command to both station numbers 3098 and 3099 of DSS Console that was assigned with the AISTL command.

DSS Key Assignment



Assignment for all of 60 keys on DSS Console are not required. The setting image of the AKYD command is shown in the figure below.



b.) Data Assignment of the DT800/DT700 Series for the SIP Multiple Line Terminal

TN: X
STN: 3000
PRI: 0
PL TN: X
PL STN: 3000
S: 0 (Off-hook Suppression is OFF)
MWD: X (Message Waiting Data Display Selection)
LN PRE: 0 (Prime Line Pickup)
TP: X (Type of SIP Multiple Line Terminal)

Example List of Data Programming

The following table provides all of steps for Data Programming.

ASYDL/ASYDN

	INDEX	BIT		REMARKS
SYS1	686	Bit 1	1	DSS/BLF mode is available.
		Bit 2	1	Call Forwarding via trunks is enabled when a station assigned to a key on DSS Console is assigned Call Forwarding via trunks.

AISTL

TENANT NUMBER (TN) 1-63	STATION NUMBER (IP STN) MAXIMUM SIX DIGITS	KIND OF IP STATION (KIND)	LINE EQUIPMENT NUMBERS (LENS)				TELE. EQUIP. CLASS (TEC) 1-31	ROUTE REST. CLASS (RSC) 0-15	Service Feature Restriction Class (SFC) 0-15	REMARKS
			MG	U	G	LV				
X	3000	SIP MultiLine TEL	0 6	0	1 0	0	12	X	X	For DT800/ DT700
	3098	SIP MultiLine TEL	0 6	0	1 0	1	12	X	X	For DSS Console-1
	3099	SIP MultiLine TEL	0 6	0	1 0	2	12	X	X	For DSS Console-2

AKYD

For Line/Feature Access Keys: KYNs 1-24 (DT800/DT700 Series)

MY LINE		TYPE OF DTERM (TP) 0-3	RING DATA (RING) 0-3	FORM OF LINE PREFERENCE (PRI) 0-3	LINE PREFERENCE (LN PRE) 0/1	LINE PREFERENCE FOR SPEAKER BUTTON (SPK) 0
TENANT NUMBER (TN) 1-15/63	STATION NUMBER (STN) MAXIMUM SIX DIGITS					
*X	3000 (=DT800/ DT700)	1	X	0	0	

LINE PREFERENCE FOR ANSWER BUTTON (ANS) 0-2	LINE PREFERENCE FOR ORIGINATING BUTTON (ORG) 0/1	MESSAGE WAITING DATA DISPLAY SELECTION (MWD) 0/1	Prime Line		OG FROM PRIME LINES 0/1
			TENANT NUMBER (TN) 1-63	STATION NUMBER (STN) MAXIMUM FIVE DIGITS	
		X	X	3000 (=DT800/ DT700)	0

KEY NUMBER (KYN) 1-16	SERVICE KEY INDEX (KYI) 0-2	FEATURE KEY NUMBER (FKY) 1-65	MULTI LINE			INTERCOM	
			LINE IDENTITIY (KD) 0-3	TENANT NUMBER (TN) 1-15	STATION NUMBER (STN) MAXIMUM FIVE DIGITS	KIND OF INTERCOM (ICM) 0-2	GROUP ID NUMBER (G-ID) 1-50
**1XXXXXX	2 (Multi Line)	**1XXXXXX	0 (Line)	**1XXXXXX	3000	**1XXXXXX	**1XXXXXXX
**2	X	X	X	X	X	X	X
**3	X	X	X	X	X	X	X
**4	X	X	X	X	X	X	X
**5	X	X	X	X	X	X	X
**6	X	X	X	X	X	X	X
**7	X	X	X	X	X	X	X
**8	X	X	X	X	X	X	X
**9	X	X	X	X	X	X	X
**10	X	X	X	X	X	X	X
**11	X	X	X	X	X	X	X
**12	X	X	X	X	X	X	X
**13	X	X	X	X	X	X	X
**14	X	X	X	X	X	X	X
**15	X	X	X	X	X	X	X
**16	X	X	X	X	X	X	X
17	X	X	X	X	X	X	X
18	X	X	X	X	X	X	X
19	X	X	X	X	X	X	X
: :	: :	: :	: :	: :	: :	: :	: :
38	X	X	X	X	X	X	X
39	X	X	X	X	X	X	X
40	X	X	X	X	X	X	X

Note: *Buttons 1-40 (1-32 buttons for DT800) are assignable / **Assignment of Line/Feature Access keys (KYNs 1-16) on DT800/DT700 Series 3000/ **Assignment of Line/Feature Access keys (KYNs 1-24) on DSS Consoles

For Line/Feature Access keys: KYNs 25-60 (DSS Console-1)

MY LINE		TYPE OF DTERM (TP) 0-3	RING DATA (RING) 0-3	FORM OF LINE PREFERENCE (PRI) 0-3	LINE PREFERENCE (LN PRE) 0/1	LINE PREFERENCE FOR SPEAKER BUTTON (SPK) 0
TENANT NUMBER (TN) 1-15/63	STATION NUMBER (STN) MAXIMUM SIX DIGITS					
*X	3098 (DSS Console-1)	1	X	0	0	

LINE PREFERENCE FOR ANSWER BUTTON (ANS) 0-2	LINE PREFERENCE FOR ORIGINATING BUTTON (ORG) 0/1	MESSAGE WAITING DATA DISPLAY SELECTION (MWD) 0/1	Prime Line		OG FROM PRIME LINES 0/1
			TENANT NUMBER (TN) 1-63	STATION NUMBER (STN) MAXIMUM FIVE DIGITS	
		X	X	3000 (=DT800/ DT700)	0

KEY NUMBER (KYN) 1-16	SERVICE KEY INDEX (KYI) 0-2	FEATURE KEY NUMBER (FKY) 1-65	MULTI LINE			INTERCOM	
			LINE IDENTITY (KD) 0-3	TENANT NUMBER (TN) 1-15	STATION NUMBER (STN) MAXIMUM FIVE DIGITS	KIND OF INTERCOM (ICM) 0-2	GROUP ID NUMBER (G-ID) 1-50
**1	2 (Multi Line)		0 (Line)	X	3098 (DSS Console-1)		
2	2 (Multi Line)		0 (Line)		3000		
3							
4							
5	X	X	X	X	X	X	X
6	X	X	X	X	X	X	X
7	X	X	X	X	X	X	X
8	X	X	X	X	X	X	X
9	X	X	X	X	X	X	X
10	X	X	X	X	X	X	X
: :	: :	: :	: :	: :	: :	: :	: :
***32	X	X	X	X	X	X	X
***33	X	X	X	X	X	X	X
***34	X	X	X	X	X	X	X
***35	X	X	X	X	X	X	X
***36	X	X	X	X	X	X	X
***37	X	X	X	X	X	X	X
***38	X	X	X	X	X	X	X
***39	X	X	X	X	X	X	X
***40	X	X	X	X	X	X	X

For Line/Feature Access keys: KYNs 25-60 (DSS Console-2)

MY LINE		TYPE OF DTERM (TP) 0-3	RING DATA (RING) 0-3	FORM OF LINE PREFERENCE (PRI) 0-3	LINE PREFERENCE (LN PRE) 0/1	LINE PREFERENCE FOR SPEAKER BUTTON (SPK) 0	
TENANT NUMBER (TN) 1-15/63	STATION NUMBER (STN) MAXIMUM SIX DIGITS						
*X	3098 (DSS Console-2)	1	X	0	0		
LINE PREFERENCE FOR ANSWER BUTTON (ANS) 0-2	LINE PREFERENCE FOR ORIGINATING BUTTON (ORG) 0/1	MESSAGE WAITING DATA DISPLAY SELECTION (MWD) 0/1		Prime Line		OG FROM PRIME LINES 0/1	
				TENANT NUMBER (TN) 1-63	STATION NUMBER (STN) MAXIMUM FIVE DIGITS		
		X		X	3000 (=DT800/ DT700)	0	

KEY NUMBER (KYN) 1-16	SERVICE KEY INDEX (KYI) 0-2	FEATURE KEY NUMBER (FKY) 1-65	MULTI LINE			INTERCOM	
			LINE IDENTITY (KD) 0-3	TENANT NUMBER (TN) 1-15	STATION NUMBER (STN) MAXIMUM FIVE DIGITS	KIND OF INTERCOM (ICM) 0-2	GROUP ID NUMBER (G-ID) 1-50
**1	2 (Multi Line)		0 (Line)	X	3099 (DSS Console-2)		
2	2 (Multi Line)		0 (Line)		3000		
3							
4							
5	X	X	X	X	X	X	X
6	X	X	X	X	X	X	X
7	X	X	X	X	X	X	X
8	X	X	X	X	X	X	X
9	X	X	X	X	X	X	X
10	X	X	X	X	X	X	X
: :	: :	: :	: :	: :	: :	: :	: :
***32	X	X	X	X	X	X	X
***33	X	X	X	X	X	X	X
***34	X	X	X	X	X	X	X
***35	X	X	X	X	X	X	X
***36	X	X	X	X	X	X	X
***37	X	X	X	X	X	X	X
***38	X	X	X	X	X	X	X
***39	X	X	X	X	X	X	X
***40	X	X	X	X	X	X	X

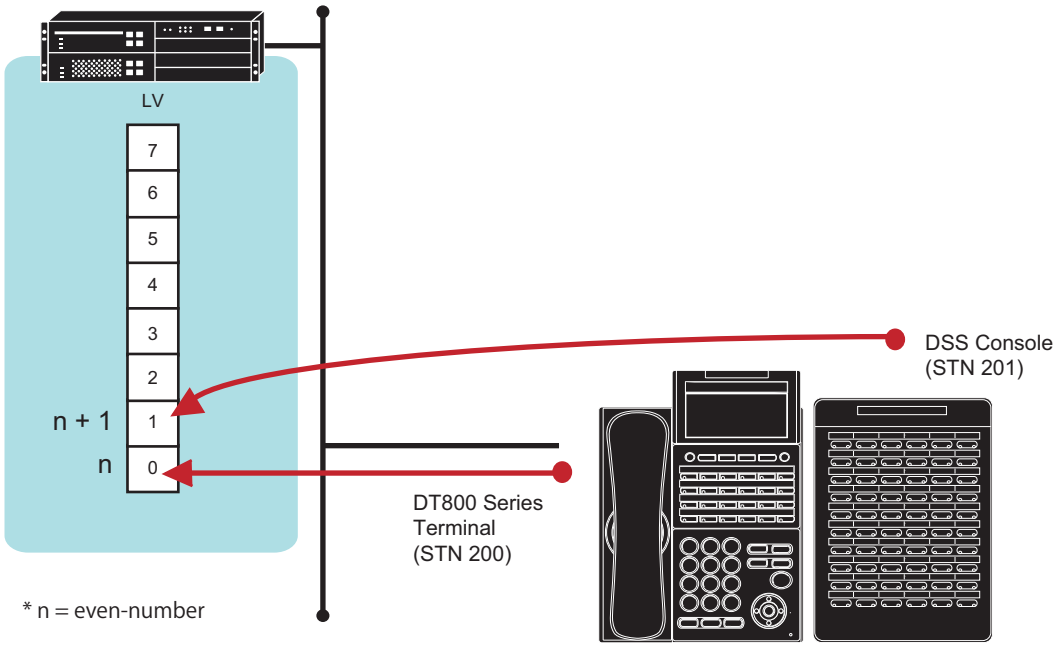
Note: *Buttons 1-40 (1-32 buttons for DT800) are assignable / **Assignment of Line Appearance Data for DSS Console 3098 or 3099 / ***Assignment of Line/Feature Access keys (KYNs 25-60) on DSS Console 3098 or 3099

<For ADM mode>

This section describes how to assign ADM mode to DSS Console and DT800/DT700 Series that is attached to the console. For ease in understanding, the data programming is explained using the example below:

Example of Data Programming

TERMINAL	LINE EQUIPMENT NUMBERS (LENS)	STATION NUMBER (STN)
DSS Console	000101	201 **Note 9**
DT800/DT700 Series	000100	200



Note 9: Station Number "201" is used only for assigning Line/Feature Access keys to the DSS Console. The number is not used in actual operations.

Perform the following procedure. Note that X indicates a value that should be tailored to the existing system.

STEP 1: **AISTL** - Assign a DT800/DT700 Series to an even-numbered LENS level. (In this example, "LV 0" is used.) Assign DSS Console (ADM) to the subsequent odd-numbered LENS level. (In this example "LV 1" is used.)

[An even-numbered level of LENS for DT800/DT700 Series paired with DSS Console]

TN: X
IP STN: 200 ← Station Number of DT800/DT700 Series
TEC: 12 ← Fixed
KIND: SIP MultiLine TEL
RSC: X

SFC: X
LENS: 000100 ◄—— Even-numbered LENS level

[An odd-numbered level of LENS for DSS Console]
TN: X
IP STN: 201 ◄—— Station Number of DSS Console
TEC: 12 ◄—— Fixed
KIND: SIP MultiLine TEL
RSC: X
SFC: X
LENS: 000101 ◄—— Odd-numbered LENS level

STEP 2: **AKYD** - Assign the station numbers to the Line/Feature Access keys for DT800/DT700 Series (STN=200) and DSS Console (STN=201) which are assigned by AISTL command.
To STN=200, assign 16 keys on the DT800/DT700 Series and the upper 24 (KYNs 1-24) Line/Feature Access keys on DSS Console that belong to the paired DT800/DT700 Series.
To STN=201, assign the remaining lower 36 (KYNs 25-60) Line/Feature Access keys belong to DSS Console. Also assign DT800/DT700 Series Prime Line to KYN=1 of DSS Console (STN=201), and assign the station number of DSS Console itself (STN=201) to KYN=2.

[Data Assignment of the DT800/DT700 Series and Upper 24 Keys on DSS Console]
To assign information of these keys, use the port of the DT800/DT700 Series that is paired with DSS Console.
TN: X
STN: 200 ◄—— Station Number of DT800/DT700 Series
PRI: 0
PL TN: X
PL STN: 200
S: 0 (Off-hook Suppression is OFF)
MWD: X (Message Waiting Data Display Selection)
LN PRE: 0 (Prime Line Pickup)
TP: 1 or 3
KYI: X (Service Index)
0: Key is not used
1: Function Key
2: Multi-line
KYN: 1-16 (For DT800/DT700 Series)
17-40 (Correspond to KYNs 1-24 on DSS Console)
KD: 0-2 (Kind of Line)
Line TN: Multi-line Tenant Number
STN: Multi-line Station Number
RG: 0-7 (Ring Information for Each Line)
ICM: 0-2 (Kind of Intercom)
G-ID: 1-50 (Group ID Number)

[Data Assignment of the Lower 36 Keys on DSS Console]

To assign information of these keys, use the port of DSS Console (ADM).

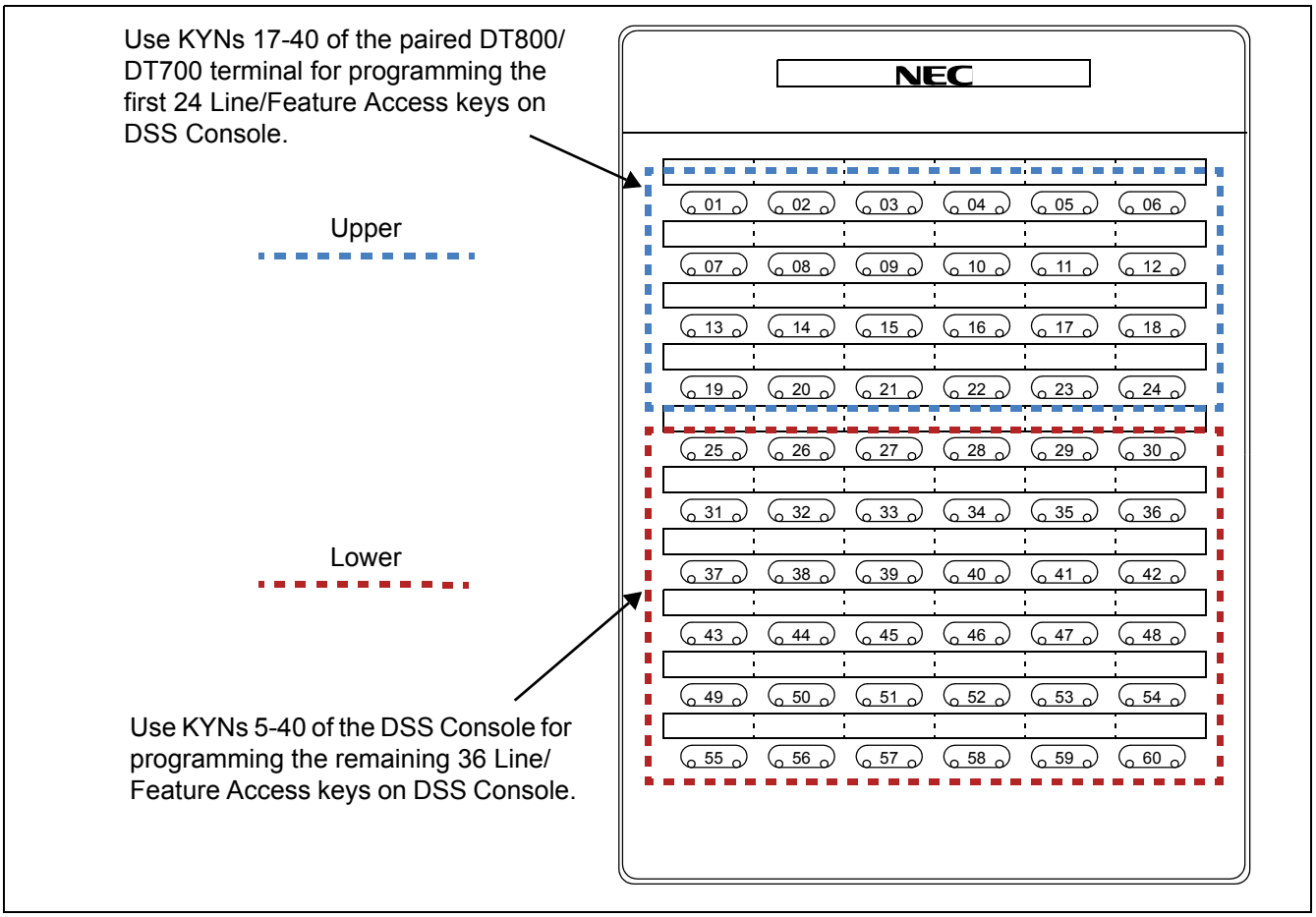
TN: X
STN: 201 ← Station Number of DSS Console
PRI: 0
PL TN: X (Prime Line Tenant Number)
PL STN: 200 ← Enter the paired DT800/DT700 Series Prime Line as a PL STN
S: 0 (Off-hook Suppression is OFF)
MWD: X (Message Waiting Data Display Selection)
LN PRE: 0 (Prime Line Pickup)
TP: 1

KYN=1: Assign DT800/DT700 Series Prime Line.
KYN=2: Assign DSS Console station number.
KYI: 2 (Multi-line)
KD: 0 (Line)
Line TN: Multi-line Tenant Number
STN: 200/201

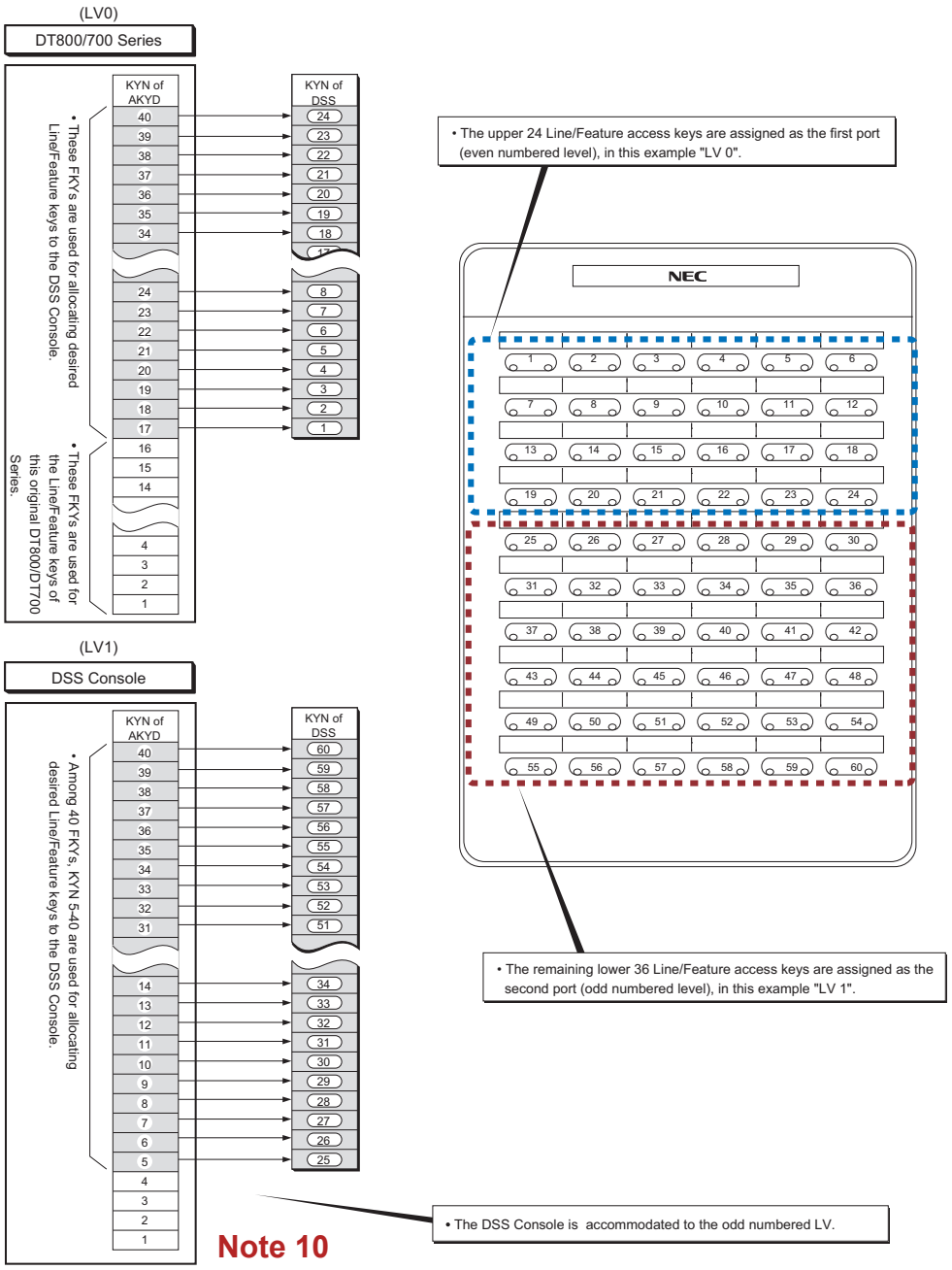
KYN	KYI	KD	LINE TN	STN	Remarks
1	2	0	1	200	DT800/DT700 Series Prime Line
2	2	0	1	201	DSS Console station number

KYN=5-40: Assign KYNs 25-60 keys on DSS Console.
KYI: X (Service Index)
0: Key is not used
1: Function Key
2: Multi-line
FKY: 1-320 (Function Key Number)
KD: 0-2 (Kind of Line)
TN: X
STN: XXX
RG: 0-7 (Ring Information for Each Line)
ICM: 0-2 (Kind of Intercom)
G-ID: 1-50 (Group ID Number)

DSS Key Assignment



To designate the meaning of the upper 24 keys on DSS Console, FKYs 17-40 of a DT800/DT700 Series that works in combination with DSS Console are used. The remaining 36 keys on DSS Console are assigned using FKYs 5-40 of the original port of DSS Console. The relationship between KYN, that appears in the AKYD command, and 60 keys on DSS Console are shown in the figure below. This figure also shows DSS Console connected to LV 1.



Note 10: Note the following conditions:

- To KYN=1, assign the Prime Line of DT800/DT700 Series paired with the DSS Console.
  In the case of Data Programming Procedure in this manual: AKYD, KYI=2, KD=0, STN=200.
- To KYN=2, assign the DSS Console station number which is accommodated in the second port (odd numbered level).
  In the case of Data Programming Procedure in this manual: AKYD, KYI=2, KD=0, STN=201.
  On the 8-button Digital terminal, KYN=9-16 cannot be used.

STEP 3: **ADKS** - Assign Digital terminal Key Status Data to the station number for DT800/DT700 Series (STN=200) which is assigned by AISTL command.

TN: X
STN: 200 ← Station Number of DT800/DT700 Series
SKP: 0-15 (Soft Key Pattern data)
LKP: 1 (16 Line/Feature buttons + DSS keys)
PG: 0/1 (Soft key/Feature key is allocated for the PAGE CHANGE key)

Example of Data Programming

The following table provides all of steps for Data Programming.

AISTL

TENANT NUMBER (TN) 1-63	STATION NUMBER (IP STN) MAXIMUM SIX DIGITS	KIND OF IP STATION (KIND)	LINE EQUIPMENT NUMBERS (LENS)				TELE. EQUIP. CLASS (TEC) 1-31	ROUTE REST. CLASS (RSC) 0-15	Service Feature Restriction Class (SFC) 0-15	REMARKS
			MG	U	G	LV				
X	200	SIP MultiLine TEL	0 0	0	1 0	*0	12	X	X	For DT800/ DT700
	201	SIP MultiLine TEL	0 0	0	1 0	**1	12	X	X	For DSS Console

Note: *0 = Even-numbered LENS level / **1 = Odd-numbered LENS level

AKYD

For Line/Feature Access Keys: KYNs 1-24 (DT800/DT700 Series)

MY LINE		TYPE OF DTERM (TP) 0-3	RING DATA (RING) 0-3	FORM OF LINE PREFERENCE (PRI) 0-3	LINE PREFERENCE (LN PRE) 0/1	LINE PREFERENCE FOR SPEAKER BUTTON (SPK) 0	
TENANT NUMBER (TN) 1-15/63	STATION NUMBER (STN) MAXIMUM SIX DIGITS						
*X	200 (=DT800/ DT700)	1	X	0	0		
LINE PREFERENCE FOR ANSWER BUTTON (ANS) 0-2	LINE PREFERENCE FOR ORIGINATING BUTTON (ORG) 0/1	MESSAGE WAITING DATA DISPLAY SELECTION (MWD) 0/1		Prime Line		OG FROM PRIME LINES 0/1	
				TENANT NUMBER (TN) 1-63	STATION NUMBER (STN) MAXIMUM FIVE DIGITS		
		X		X	200 (=DT800/ DT700)	0	
KEY NUMBER (KYN) 1-16	SERVICE KEY INDEX (KYI) 0-2	FEATURE KEY NUMBER (FKY) 1-65	MULTI LINE			INTERCOM	
			LINE IDENTITIY (KD) 0-3	TENANT NUMBER (TN) 1-15	STATION NUMBER (STN) MAXIMUM FIVE DIGITS	KIND OF INTERCOM (ICM) 0-2	GROUP ID NUMBER (G-ID) 1-50
**1XXXXXX	2 (Multi Line)	**1XXXXXX	0 (Line)	**1XXXXXX	200	**1XXXXXX	**1XXXXXXX

**2	X	X	X	X	X	X	X
**3	X	X	X	X	X	X	X
**4	X	X	X	X	X	X	X
**5	X	X	X	X	X	X	X
**6	X	X	X	X	X	X	X
**7	X	X	X	X	X	X	X
**8	X	X	X	X	X	X	X
**9	X	X	X	X	X	X	X
**10	X	X	X	X	X	X	X
**11	X	X	X	X	X	X	X
**12	X	X	X	X	X	X	X
**13	X	X	X	X	X	X	X
**14	X	X	X	X	X	X	X
**15	X	X	X	X	X	X	X
**16	X	X	X	X	X	X	X
17	X	X	X	X	X	X	X
18	X	X	X	X	X	X	X
19	X	X	X	X	X	X	X
: :	: :	: :	: :	: :	: :	: :	: :
38	X	X	X	X	X	X	X
39	X	X	X	X	X	X	X
40	X	X	X	X	X	X	X

Note: *Buttons 1-40 (1-32 buttons for DT800) are assignable / **Assignment of Line/Feature Access keys (KYNs 1-16) on DT800/DT700 Series 200/ **Assignment of Line/Feature Access keys (KYNs 1-24) on DSS Console 201

For Line/Feature Access keys: KYNs 25-60 (DSS Console)

MY LINE							
TENANT NUMBER (TN) 1-15/63	STATION NUMBER (STN) MAXIMUM SIX DIGITS	TYPE OF DTERM (TP) 0-3	RING DATA (RING) 0-3	FORM OF LINE PREFERENCE (PRI) 0-3	LINE PREFERENCE (LN PRE) 0/1	LINE PREFERENCE FOR SPEAKER BUTTON (SPK) 0	
*X	201 (DSS Console)	1	X	0	0		
LINE PREFERENCE FOR ANSWER BUTTON (ANS) 0-2	LINE PREFERENCE FOR ORIGINATING BUTTON (ORG) 0/1	MESSAGE WAITING DATA DISPLAY SELECTION (MWD) 0/1		Prime Line		OG FROM PRIME LINES 0/1	
				TENANT NUMBER (TN) 1-63	STATION NUMBER (STN) MAXIMUM FIVE DIGITS		
		X		X	200 (=DT800/ DT700)	0	
KEY NUMBER (KYN) 1-16	SERVICE KEY INDEX (KYI) 0-2	FEATURE KEY NUMBER (FKY) 1-65	MULTI LINE			INTERCOM	
			LINE IDENTITY (KD) 0-3	TENANT NUMBER (TN) 1-15	STATION NUMBER (STN) MAXIMUM FIVE DIGITS	KIND OF INTERCOM (ICM) 0-2	GROUP ID NUMBER (G-ID) 1-50
**1	2 (Multi Line)		0 (Line)	X	201 (DSS Console)		

2	2 (Multi Line)		0 (Line)		200		
3							
4							
5	X	X	X	X	X	X	X
6	X	X	X	X	X	X	X
7	X	X	X	X	X	X	X
8	X	X	X	X	X	X	X
9	X	X	X	X	X	X	X
10	X	X	X	X	X	X	X
: :	: :	: :	: :	: :	: :	: :	: :
***32	X	X	X	X	X	X	X
***33	X	X	X	X	X	X	X
***34	X	X	X	X	X	X	X
***35	X	X	X	X	X	X	X
***36	X	X	X	X	X	X	X
***37	X	X	X	X	X	X	X
***38	X	X	X	X	X	X	X
***39	X	X	X	X	X	X	X
***40	X	X	X	X	X	X	X

Note: *Buttons 1-40 (1-32 buttons for DT800) are assignable / **Assignment of Line Appearance Data for DSS Console 201/ ***Assignment of Line/Feature Access keys (KYNs 25-60) on DSS Console 201

ADKS

TENANT NUMBER (TN) 1-63	STATION NUMBER (IP STN) MAXIMUM SIX DIGITS	SOFTWARE KEY PATTERN DATA (SKP) 0-15	KEY PATTERN DATA (LKP) 1-3	LCD PAGE (PG) 0/1
X	200		1	

Direct Station Selection (DSS) Console for Digital Terminal

This section explains service conditions and programming procedure for Direct Station Selection (DSS) Console used in combination with a Digital terminal.

Operating Procedure

The following shows examples of the operating procedure when the 60 keys are used as the Line keys, allowing the user to activate SPEED CALLING - ONE TOUCH and Call Transfer.

To originate a call using SPEED CALLING - ONE TOUCH:

1. User A lifts the handset or presses the SPEAKER key on a Digital terminal.
   User A hears the dial tone.
2. User A presses a key on DSS Console where a station number of User B to whom User A wants to call is assigned.
   User A hears the ringback tone from the terminal. A terminal of User B starts ringing.
3. User B answers the call.
   They can start a conversation.

To transfer the call to the Station assigned on the DSS key:

1. While User A and User B are engaged in a call.
2. User A presses a key on DSS Console where a station number of User C to whom User A wants to transfer the call is assigned.
   User A hears a ringback tone from the terminal. User B is placed on hold. A terminal of User C starts ringing.
3. User C answers the call.
   User B and User C can start a conversation.

Note: When the a Digital terminal user switch hook flashes during a ringback tone, the user returns to the Held Party, and the third party (the station assigned on the DSS key) stops ringing.

Note: If the Digital terminal user replaces the handset before the third party answers, Blind Transfer is activated. In this case, if the third party does not answer for approximately 30 seconds, the user is recalled and upon answering, will be connected to the Held Party.

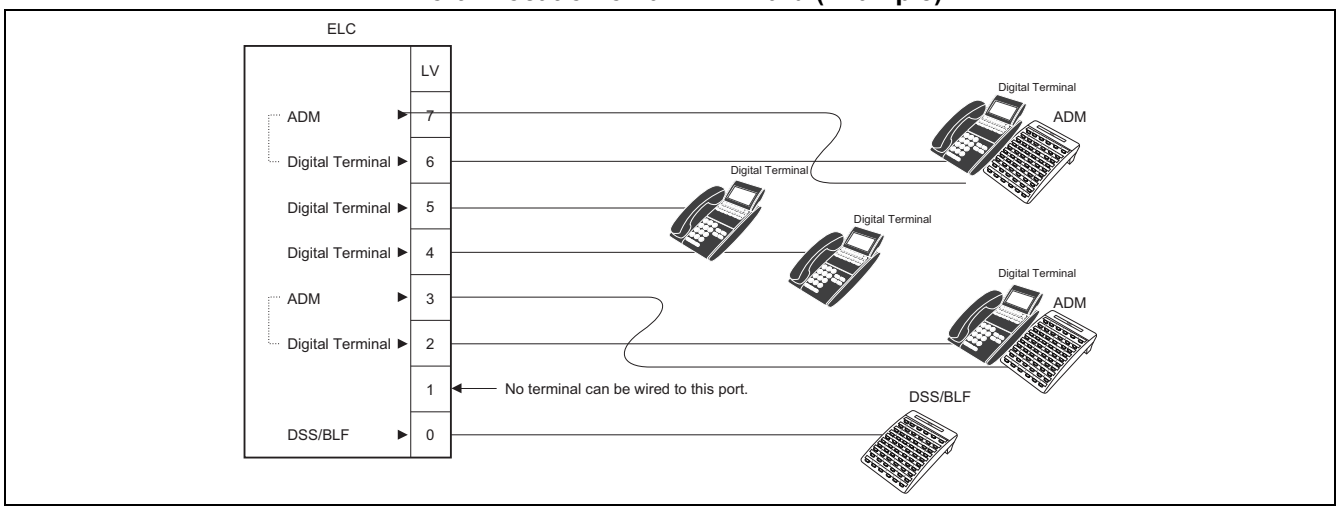
For more information, refer to the User Guide.

Service Conditions

Common Condition

1. DSS Console (DSS/BLF) and DSS Console (ADM) can co-exist on one circuit card. The figure below shows an example of a port allocation on an ELC circuit card.

Port Allocation on an ELC Card (Example)



2. When the station assigned on the DSS key that becomes the call destination or the forwarding destination is not in an idle state but in a call-block state, a lockout state or a make-busy state, BUSY is displayed on the originating terminal.

For DSS Console (DSS/BLF)

<Important Notices>

1. One DSS Console (DSS/BLF) requires two consecutive ports on an ELC card.
2. To use 60 keys on DSS Console (DSS/BLF), key data for two consecutive ports need to be assigned by the AKYD command. Refer to STEP 3: AKYD command.
3. Since FP95-114 V4 Issue 2.0, Call Forwarding via trunks is available on DSS Console by pressing a key where a station whose forwarding destination for Call Forwarding - All Calls - Outside, Call Forwarding - Busy Line - Outside, and Logged Out IP Station - Call Destination is an outside number is assigned.
   Call Forwarding conditions are as follows:
   - The system data (ASYDL/ASYDN, SYS1, Index 686, Bit 2=1) is required.
   - When a forwarding destination of the station is an attendant console, Call Forwarding is restricted.
   - When a forwarding destination number is Uniform Numbering, Call Forwarding is restricted.
   - LCR, LCRS, OGC, and OGCA are supported for the originating methods.
   - When a forwarding destination is a station, Call Forwarding - All Calls, Call Forwarding - Busy Line, and Call Forwarding - Don't Answer are supported. However, it is not supported when the forwarding destination is an attendant console.
     When a forwarding destination is a trunk line, calls for Call Forwarding - All Calls - Outside, Call Forwarding - Busy Line - Outside, and Logged Out IP Station - Call Destination can be forwarded when:

     - a transferor is hearing a dial tone

     - a transferor is on a call

     - a transferor goes on-hook while on a call

- As Split Call Forwarding is not available, the call is transferred to the forwarding destination of Call Forwarding for trunk calls.

For FP95-114 V4 or earlier, a station cannot make/transfer a call by using a DSS key when the target station of the DSS key sets following forwarding features.

- CALL FORWARDING to Attendant Console (including ALL CALLS, BUSY LINE, DON'T ANSWER)
- CALL FORWARDING - OUTSIDE (including ALL CALLS, BUSY LINE, DON'T ANSWER)
- MULTIPLE CALL FORWARDING (including ALL CALLS, BUSY LINE, DON'T ANSWER)
- SPLIT CALL FORWARDING

<Interactions with Other Features>

1. When the Station user calls the station assigned on the DSS key by SPEED CALLING - ONE TOUCH, or transfers the call to the station assigned on the DSS key, and the station assigned on the DSS key is the member of the UCD [U-1] or Station Hunting Group, the call follows the UCD or the Hunting Operation. (If the MCI is valid, the message is sent out.)
2. When the user calls the station assigned on the DSS key by SPEED CALLING - ONE TOUCH, or transfers the call to the station assigned on the DSS key, following features set by the station assigned on the DSS key are not available:
   - BUSY VERIFICATION [B-3]
   - CALL BACK [C-1]
   - CALL WAITING [C-12, C-31]
   - EXECUTIVE RIGHT-OF-WAY [E-1]
   - MESSAGE REMINDER [M-19]
   - STEP CALL [S-13]
3. The following features are not used when DSS Console (DSS/BLF) encounters a busy station.
   - CALL BACK
   - CALL WAITING
   - EXECUTIVE RIGHT-OF-WAY
   - STEP CALL
4. The following features do not work after a call is originated by using DSS keys.
   - CALL BACK [C-1]
   - CALL WAITING - TERMINATING [C-12]
   - CALL WAITING - ORIGINATING [C-31]
   - EXECUTIVE RIGHT-OF-WAY [E-1]
   - MESSAGE REMINDER [M-19]
   - STEP CALL [S-13]
   - VOICE CALL [V-2]

<Hardware/Software>

1. One Digital terminal can use multiple DSS Console.
2. Multiple Digital terminals cannot share one DSS Console.
3. DSS Console can accommodate IP Stations (IP terminals/SIP Multiple Line terminals) as Multiple Lines, but cannot be paired with IP Stations expect for DT800/DT700 Series. For more information, refer to Direct Station Selection (DSS) Console for SIP Multiple Line terminal.
4. ACD Agent/Supervisor terminals do not support this feature.

<Interactions with Networks>

None

<Conditions Specific to This Feature>

1. When DSS Console is used as DSS Console (DSS/BLF), assign and wire DSS Console (DSS/BLF) to a specific LENS whose level (LV) is an even number (LV = 0, 2, 4, 6). Since DSS Console (DSS/BLF) requires two consecutive ports for programming the DSS keys, the subsequent port of the designated port, to which DSS Console (DSS/BLF) is wired, must be reserved.
2. A Digital terminal, attached DSS Console, and stations that are assigned to the DSS keys must belong to the same Module Group.
3. DSS Console and the attached Digital terminal can be accommodated in the different circuit cards, but should belong to the same Module Group. Also, the stations assigned on the DSS keys should belong to the same LP with DSS Console.
4. When the user calls the station assigned on the DSS key using SPEED CALLING - ONE TOUCH and the station assigned on the DSS key is either busy, not available, or restricted, the user receives a Reorder Tone.
5. When the user transfers the call to the station assigned on the DSS key which is either busy, not available, or restricted, the user receives a Reorder Tone. The user returns to the original call by switch hook flash.
6. The user can recognize the idle/busy status of the accommodated Stations by the lamps.
   The left lamp (red) lights when busy and goes out when the accommodated Stations are in idle status. Also the lamp shows status such as having an incoming call, or a call on hold. The right lamp (green) shows status such as I-Used, I-Hold.

<Other Conditions>

None

For DSS Console (ADM)

<Important Notices>

1. One DSS Console (ADM) occupies one port on an ELC card.
2. To use 60 keys on DSS Console (ADM), key data for two consecutive ports need to be assigned by the AKYD command. Refer to STEP 2: AKYD command.
3. Up to 16 Line/Feature keys on Digital Multiline terminal can be used in ADM mode. This setting is assigned by the ADKS command. Refer to STEP 3: ADKS command.

<Interactions with Other Features>

1. Do not assign the following parameters to Function Key Number (FKY) when the Line/Feature keys are assigned on the DSS Console (ADM) by the AKYD command.
   - FKY = 6 (Message Reminder)
   - FKY = 49 (Speed Calling - One Touch)
   - FKY = 75 to 82 (Call Park Group-D key)
   - FKY = 118 (Call Return)
   - FKY = 163 (Privacy key)
2. When the Station user calls the station assigned on the DSS key by SPEED CALLING - ONE TOUCH, or transfers the call to the station assigned on the DSS key, and the station assigned on the DSS key is the member of the UCD [U-1] or Station Hunting Group, the call follows the UCD or the Hunting Operation. (If the MCI is valid, the message is sent out.)

<Hardware/Software>

1. Multiple Digital terminals cannot share one DSS Console.
2. Either of the following circuit card is required.
   - PA-16ELCJ-B
   - PA-16ELCN

<Interactions with Networks>

None

<Conditions Specific to This Feature>

1. For DSS Console (ADM), assign DSS Console to a specific LENS whose Level (LV) is an odd number (LV = 1, 3, 5, 7). The Digital terminal that is used in combination with DSS Console (ADM) must be assigned to the preceding even numbered LENS level (LV = 0, 2, 4, 6).
2. A Digital terminal, attached DSS Console (ADM), and stations that are assigned to the DSS keys must belong to the same Module Group.
3. The Digital terminal to be attached to DSS Console should be accommodated in the same circuit card as DSS Consoles on the even-numbered port, using a number smaller than the one where DSS Console is accommodated.
4. On the 25th key and following keys, My Line/Prime Line for Digital terminal cannot be assigned.
5. A maximum of 48 line appearances and 12 Feature keys can be programmed for DSS Console (ADM).
6. The 48th key and following keys can be used only as the Feature keys, not as the Line keys. These keys do not have the green lamps, therefore, I-Used or I-Hold cannot be indicated.

<Other Conditions>

None

Programming

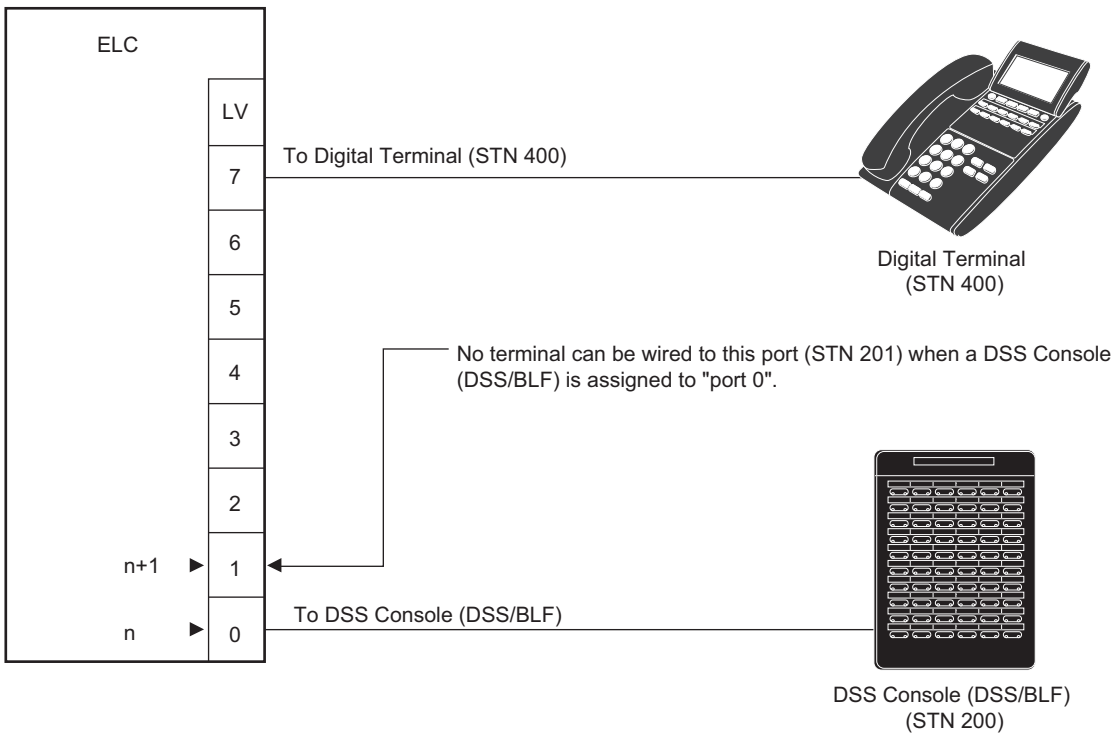
This section describes how to program Office Data to pair Digital terminal with DSS Console (DSS/BLF).

DSS Console (DSS/BLF)

This section describes how to assign DSS Console (DSS/BLF). The data programming is explained using the example below:

Example of Data Programming for DSS Console (DSS/BLF)

TERMINAL	LINE EQUIPMENT NUMBERS (LENS)	STATION NUMBER (STN)
DSS Console (DSS/BLF) **Note 11**	000100	200
	000101	201
Digital terminal	000107	400



Note 11: Two Station Numbers are required only for programming DSS keys. Accordingly, these numbers are not used in actual operations of the DSS Console (DSS/BLF).

When assigning DSS Console (DSS/BLF) Data, perform the following procedure. Note that X indicates a value that should be tailored to the existing system.

STEP 1: **ASYDL/ASYDN** - SYS1, Index 686
Bit 2=1: Call Forwarding - Outside of a station assigned to the DSS key is available.

Note: Available since FP95-114 V4 Issue 2.0.

STEP 2: **ASDT** - Assign DSS Console (DSS/BLF) to an even numbered LENS level. (In this example **LV 0** is used.)

TN: X
STN: 200 ← Station Number of Digital terminal
LENS: 000100 ← Even-numbered LENS level
TEC: 12 ← Fixed
RSC: X
SFC: X

Assign a Station which is used for programming DSS keys (KYNs 31-60) to the subsequent LENS (In this example **LV 1** is used.)

TN: X
STN: 201
LENS: 000101 ← To the subsequent LENS
TEC: 12 ← Fixed
RSC: X
SFC: X

STEP 3: **AKYD** - Assign key data for 60 DSS keys. The upper 30 DSS keys (KYNs 1-30) are assigned using the first port (**LV 0** in this example) and the remaining lower 30 DSS keys (KYNs 31-60) are assigned using the second port (**LV 1** in this example). Be sure to assign the paired Digital terminal Station Number as the Station Number of Prime Line.

TN: X
STN: 200/201
TP: 1 or 3 ← Buttons 1-40 assignable
RING: X
PRI: 0
PL TN: X
STN: 400 ← Enter the paired Digital terminal as a PL STN
S: 0 (Off-hook Suppression is OFF)
LN PRE: 0 (Prime Line Pickup)
MWD: X

KYN 1 must be programmed as DSS Console station which has been assigned in ASDT.

KYN: 1
KYI: 2 (Multi Line)
KD: 0 (Line)
TN: X
STN: 200/201 ← Station Number of DSS Console

KYN 2 must be programmed as the paired Digital terminal station.

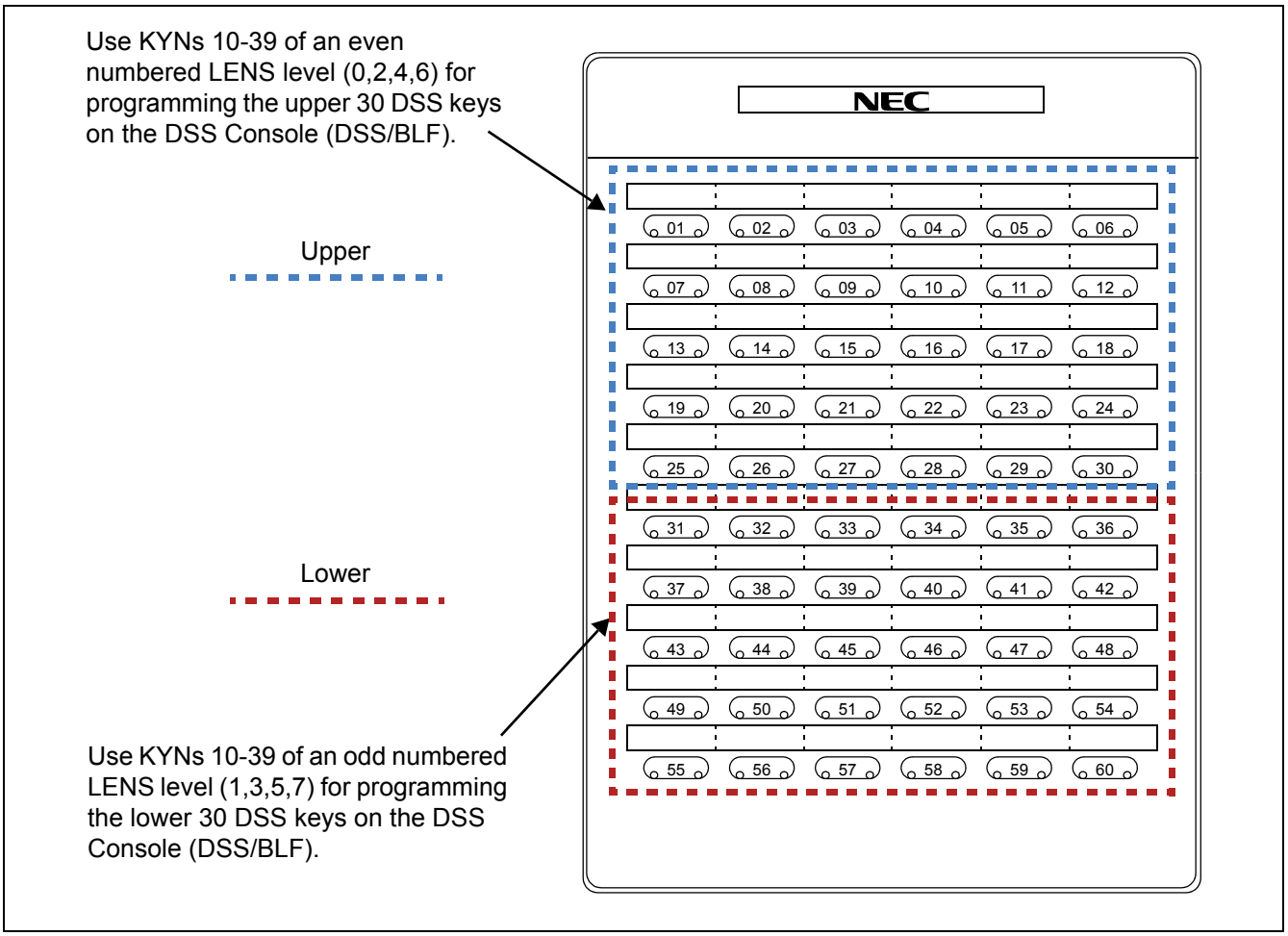
KYN: 2
KYI: 2 (Multi Line)
KD: 0 (Line)
TN: X
STN: 400 ← The paired Digital terminal

Assign desired Station Numbers to the corresponding DSS keys on the Console. In this instance, KYNs 10-39 are used. The upper and lower 30 keys are paired with even-numbered port and odd-numbered port respectively as shown in the following figure.

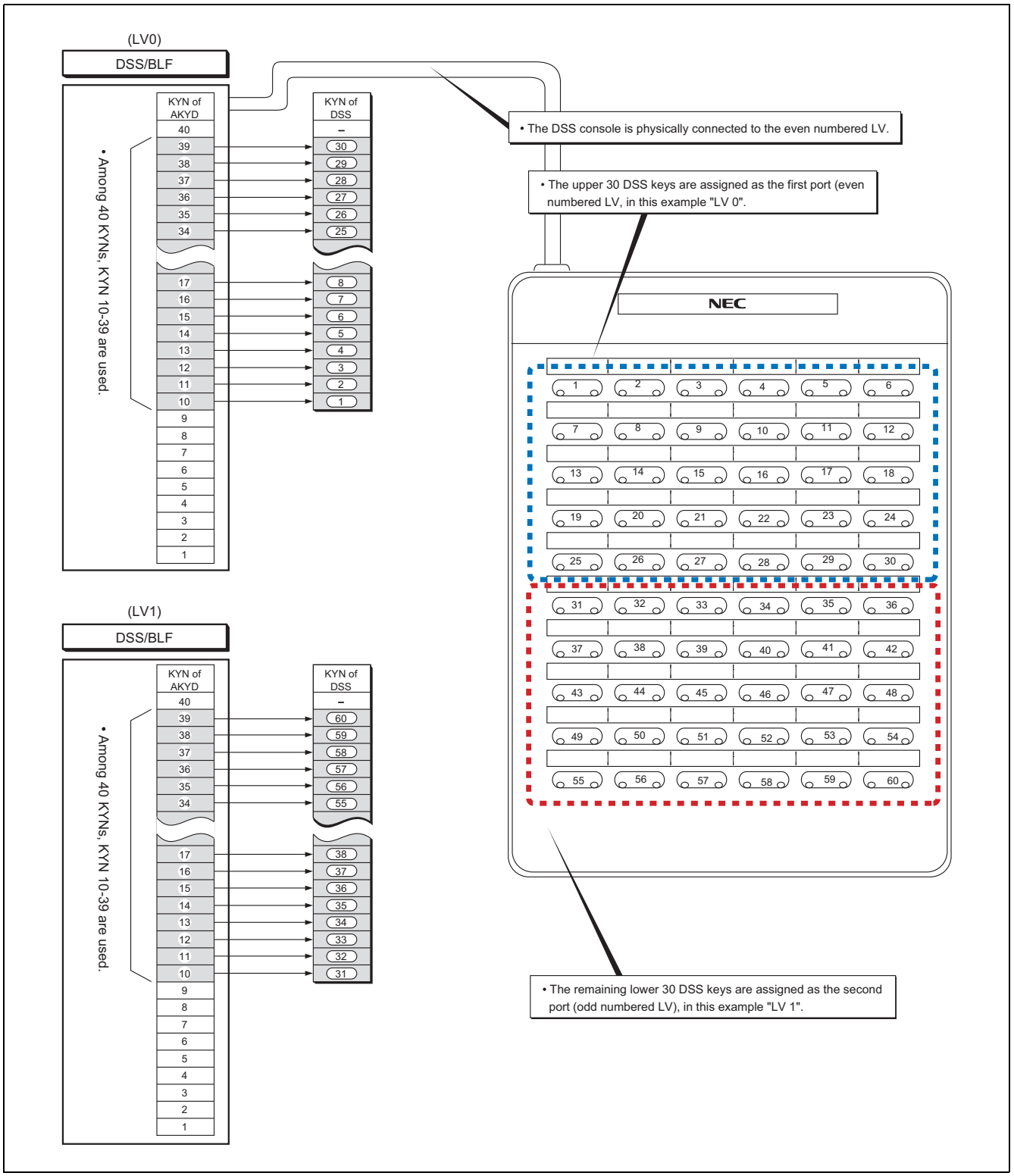
KYN: 10-39
KYI: 2 (Multi Line)
KD: 2 (For DSS Console key)
TN: X
STN: XXXXX ← Desired Station Number (maximum five digits)

Note: Assign AKYD command data for both STN 200 and STN 201 registered by the ASDT command.

DSS Key Assignment for DSS Console (DSS/BLF)



Key Numbers (KYN: 1-40) that appear in the AKYD command, correspond to 60 DSS keys. This figure also shows an example of DSS key allocation on DSS Console (DSS/BLF) when it is connected to LV 0.



Example of Data Programming for DSS Console (DSS/BLF)

The following table provides all of steps for Data Programming.

ASYDL/ASYDN

	INDEX	BIT		REMARKS
SYS1	686	Bit 2	1	Call Forwarding via trunks is enabled when a station assigned to a key on DSS Console is assigned Call Forwarding via trunks.

ASDT

TENANT NUMBER (TN) 1-63	STATION NUMBER (STN) MAXIMUM SIX DIGITS	LINE EQUIPMENT NUMBERS (LENS)				TELE. EQUIP. CLASS (TEC) 1-31	ROUTE REST. CLASS (RSC) 0-15	Service Feature Restriction Class (SFC) 0-15	REMARKS
		MG	U	G	LV				
X	200	0 0	0	1 0	*0	12	X	X	For DSS Console (DSS/BLF)
	201	0 0	0	1 0	**1	12	X	X	For DSS Console (DSS/BLF)
	400	0 0	0	1 0	7	12	X	X	For Paired Digital terminal

Note: *0 = Even-numbered LENS level / **1 = Odd-numbered LENS level

AKYD

For even-numbered LV (2n)

MY LINE						
TENANT NUMBER (TN) 1-15/63	STATION NUMBER (STN) MAXIMUM SIX DIGITS	TYPE OF DTERM (TP) 0-3	RING DATA (RING) 0-3	FORM OF LINE PREFERENCE (PRI) 0-3	LINE PREFERENCE (LN PRE) 0/1	LINE PREFERENCE FOR SPEAKER BUTTON (SPK) 0
*X	200	1	X	0	0	0

LINE PREFERENCE FOR ANSWER BUTTON (ANS) 0-2	LINE PREFERENCE FOR ORIGINATING BUTTON (ORG) 0/1	MESSAGE WAITING DATA DISPLAY SELECTION (MWD) 0/1	Prime Line		OG FROM PRIME LINES 0/1
			TENANT NUMBER (TN) 1-63	STATION NUMBER (STN) MAXIMUM FIVE DIGITS	
		X	X	400 (=Digital terminal)	0

			Multi Line			Intercom	
KEY NUMBER (KYN) 1-16	SERVICE KEY INDEX (KYI) 0-2	FEATURE KEY NUMBER (FKY) 1-65	LINE IDENTITY (KD) 0-3	TENANT NUMBER (TN) 1-15	STATION NUMBER (STN) MAXIMUM FIVE DIGITS	KIND OF INTERCOM (ICM) 0-2	GROUP ID NUMBER (G-ID) 1-50
**1	2 (Multi Line)		0 (Line)	X	200		
**2	2 (Multi Line)		0 (Line)	X	400 (Digital terminal)		
**3							
**4							
**5							
**6							

**7							
**8							
**9							
**10	2 (Multi Line)		2 (DSS)	X	X	X	X
**11	2 (Multi Line)		2 (DSS)	X	X	X	X
**12	2 (Multi Line)		2 (DSS)	X	X	X	X
**13	2 (Multi Line)		2 (DSS)	X	X	X	X
"	"	"	"	"	"	"	"

Note: *Buttons 1-40 are assignable / **Assignment of DSS Keys (KYNs 1-30) using even-numbered LENS level.

31	2 (Multi Line)		2 (DSS)	X	X	X	X
32	2 (Multi Line)		2 (DSS)	X	X	X	X
33	2 (Multi Line)		2 (DSS)	X	X	X	X
34	2 (Multi Line)		2 (DSS)	X	X	X	X
35	2 (Multi Line)		2 (DSS)	X	X	X	X
36	2 (Multi Line)		2 (DSS)	X	X	X	X
37	2 (Multi Line)		2 (DSS)	X	X	X	X
38	2 (Multi Line)		2 (DSS)	X	X	X	X
39	2 (Multi Line)		2 (DSS)	X	X	X	X

For odd-numbered LV (2n + 1)

MY LINE						
TENANT NUMBER (TN) 1-15/63	STATION NUMBER (STN) MAXIMUM SIX DIGITS	TYPE OF DTERM (TP) 0-3	RING DATA (RING) 0-3	FORM OF LINE PREFERENCE (PRI) 0-3	LINE PREFERENCE (LN PRE) 0/1	LINE PREFERENCE FOR SPEAKER BUTTON (SPK) 0
*X	201	1	X	0	0	0

LINE PREFERENCE FOR ANSWER BUTTON (ANS) 0-2	LINE PREFERENCE FOR ORIGINATING BUTTON (ORG) 0/1	MESSAGE WAITING DATA DISPLAY SELECTION (MWD) 0/1	Prime Line		OG FROM PRIME LINES 0/1
			TENANT NUMBER (TN) 1-63	STATION NUMBER (STN) MAXIMUM FIVE DIGITS	
		X	X	400 (=Digital terminal)	0

			Multi Line			Intercom	
KEY NUMBER (KYN) 1-16	SERVICE KEY INDEX (KYI) 0-2	FEATURE KEY NUMBER (FKY) 1-65	LINE IDENTITY (KD) 0-3	TENANT NUMBER (TN) 1-15	STATION NUMBER (STN) MAXIMUM FIVE DIGITS	KIND OF INTERCOM (ICM) 0-2	GROUP ID NUMBER (G-ID) 1-50
1	2 (Multi Line)		0 (Line)	X	201		
2	2 (Multi Line)		0 (Line)	X	400 (Digital terminal)		
3							
4							

5							
6							
7							
8							
9							
10	2 (Multi Line)		2 (DSS)	X	X	X	X
11	2 (Multi Line)		2 (DSS)	X	X	X	X
12	2 (Multi Line)		2 (DSS)	X	X	X	X
13	2 (Multi Line)		2 (DSS)	X	X	X	X
“	“	“	“	“	“	“	“
**31	2 (Multi Line)		2 (DSS)	X	X	X	X
**32	2 (Multi Line)		2 (DSS)	X	X	X	X
**33	2 (Multi Line)		2 (DSS)	X	X	X	X
**34	2 (Multi Line)		2 (DSS)	X	X	X	X
**35	2 (Multi Line)		2 (DSS)	X	X	X	X
**36	2 (Multi Line)		2 (DSS)	X	X	X	X
**37	2 (Multi Line)		2 (DSS)	X	X	X	X
**38	2 (Multi Line)		2 (DSS)	X	X	X	X
**39	2 (Multi Line)		2 (DSS)	X	X	X	X

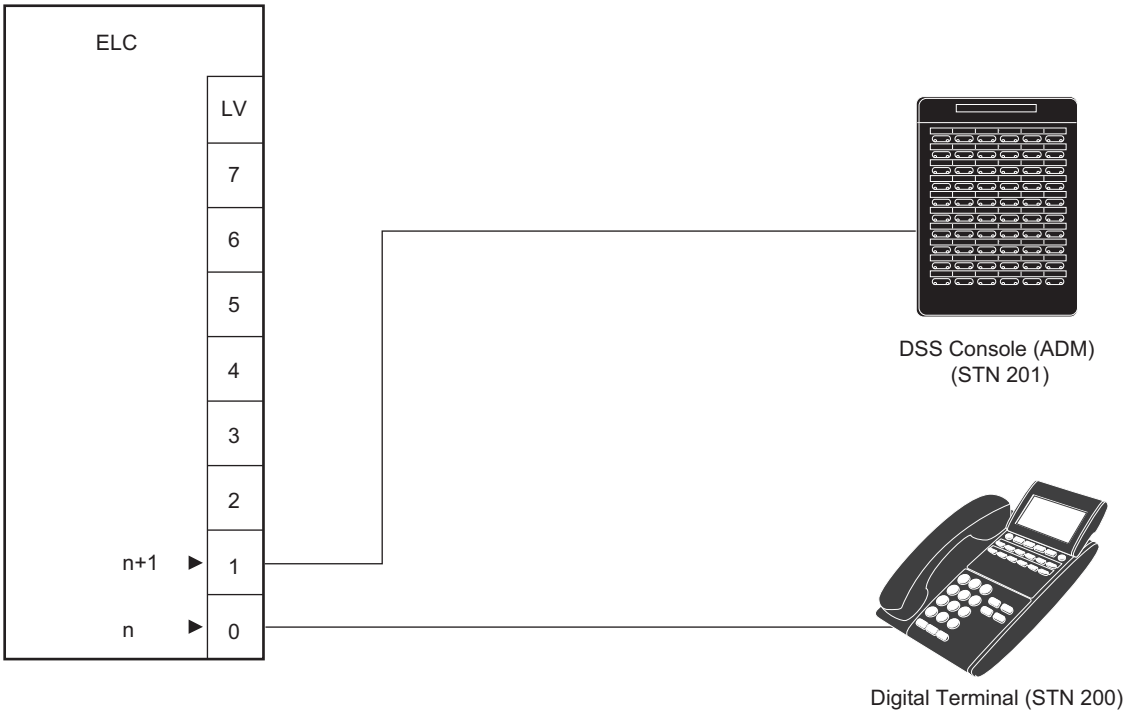
Note: *Buttons 1-40 are assignable / **Assignment of DSS keys (KYNs 31-60) using odd-numbered LENS level.

DSS Console (ADM)

This section describes how to assign DSS Console (ADM) and the paired digital terminal. For ease in understanding, the data programming is explained using the example below:

Example of Data Programming for DSS Console (ADM)

TERMINAL	LINE EQUIPMENT NUMBERS (LENS)	STATION NUMBER (STN)
DSS Console (ADM)	000101	201 **Note 12**
Digital terminal	000100	200



Note 12: Station Number 201 is used only for assigning Line/Feature Access keys to the DSS Console (ADM). The number is not used in actual operations of the DSS Console (ADM).

When assigning DSS Console (ADM) data, perform the following procedure. Note that X indicates a value that should be tailored to the existing system.

STEP 1: **ASDT** - Assign a Digital terminal to an even-numbered LENS level. (In this example, **LV 0** is used.) Assign DSS Console (ADM) to the subsequent odd-numbered LENS level. (In this example **LV 1** is used.)

[An even-numbered LENS level of LENS for Digital terminal paired with DSS Console]
TN: X
STN: 200

LENS: 000100 ← Even-numbered LENS level
TEC: 12 ← Fixed
RSC: X
SFC: X

[An odd-numbered LENS level of LENS for DSS Console]
TN: X
STN: 201
LENS: 000101 ← Odd-numbered LENS level
TEC: 12 ← Fixed
RSC: X
SFC: X

STEP 2: **AKYD** -Assign the station numbers to the Line/Feature Access keys for Digital terminal (STN=200) and DSS Console (STN=201) which are assigned by ASDT command.
To STN=200, assign 16 keys on the Digital terminal and the upper 24 (KYNs 1-24) Line/Feature Access keys on DSS Console that belong to the paired digital terminal.
To STN=201, assign the remaining lower 36 (KYNs 25-60) Line/Feature Access keys belong to DSS Console.
Also assign Digital terminal Prime Line to KYN=1 of DSS Console (STN=201), and assign the station number of DSS Console itself (STN=201) to KYN=2.

[Data Assignment of the Digital terminal and the upper 24 keys on DSS Console (ADM)]
To assign information of these keys, use the port of the Digital terminal that is paired with DSS Console.
TN: X
STN: 200 ← Station Number of Digital terminal
RING: X
PRI: 0
PL TN: X
PL STN: 200
S: 0 (Off-hook Suppression is OFF)
MWD: X (Message Waiting Data Display Selection)
LN PRE: 0 (Prime Line Pickup)
TP: 1 or 3 ← Buttons 1-40 assignable
KYI: X (Service Index)
0: Key is not used
1: Function Key
2: Multi-line
KYN: 1-16 (for Digital terminal)
17-40 (Correspond to KYNs 1-24 on DSS Console (ADM))
KD: 0-2 (Kind of Line)
Line TN: Multi-line Tenant Number
STN: Multi-line Station Number

RG: 0-7 (Ring Information for Each Line)
ICM: 0-2 (Kind of Intercom)
G-ID: 1-50 (Group ID Number)

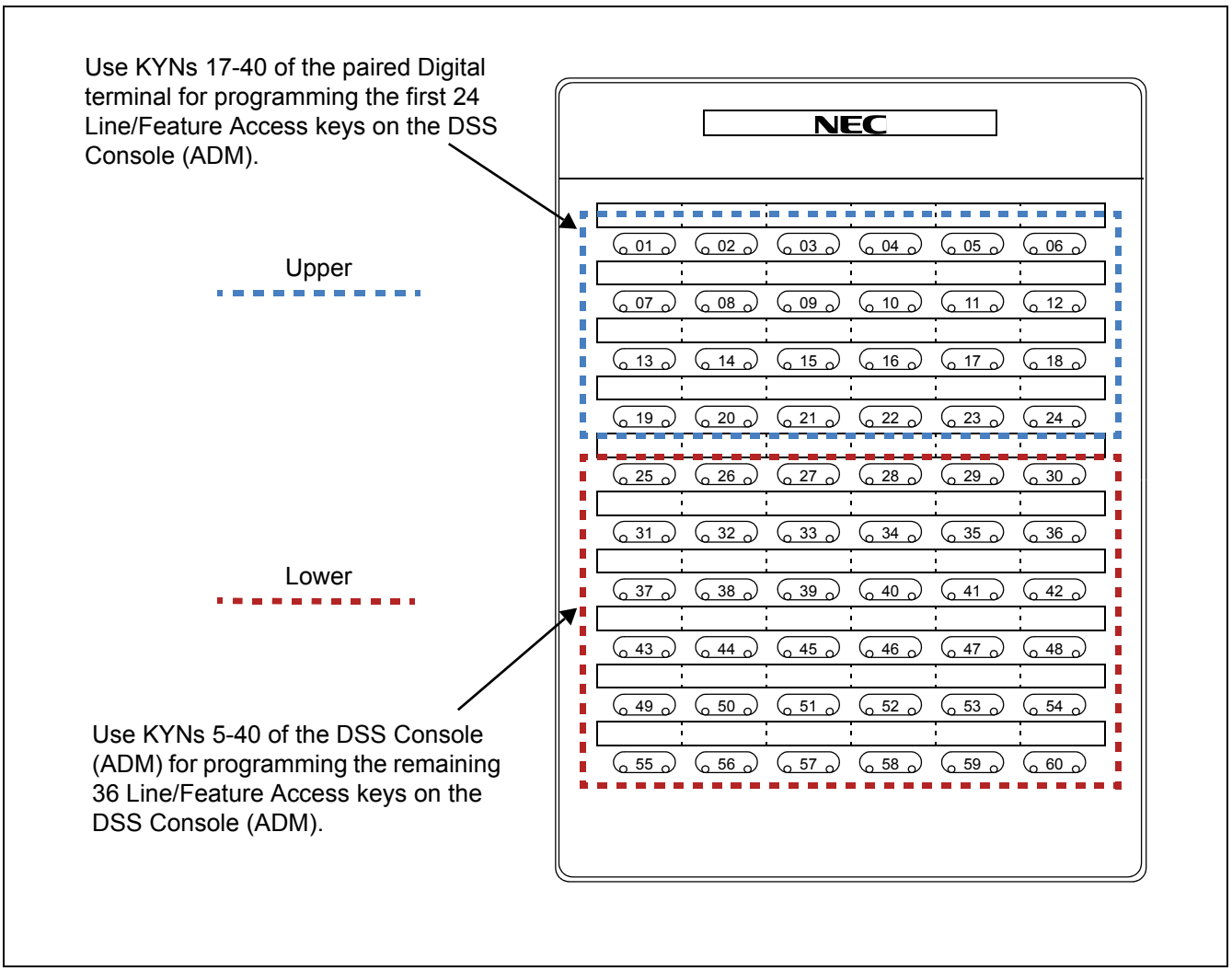
[Data Assignment of the lower 36 keys on DSS Console (ADM)]
To assign information of these keys, use the port of DSS Console (ADM).
TN: X
STN: 201 ← Station Number of DSS Console (ADM)
PRI: 0
PL TN: X (Prime Line Tenant Number)
PL STN: 200 ← Enter the paired Digital terminal Prime Line as a PL STN
S: 0 (Off-hook Suppression is OFF)
MWD: X (Message Waiting Data Display Selection)
LN PRE: 0 (Prime Line Pickup)
TP: 1 ← Buttons 1-40 assignable

KYN=1: Assign Digital terminal Prime Line.
KYN=2: Assign DSS Console station (ADM) number.
KYI: 2 (Multi-line)
KD: 0 (Line)
TN: X
STN: 200/201

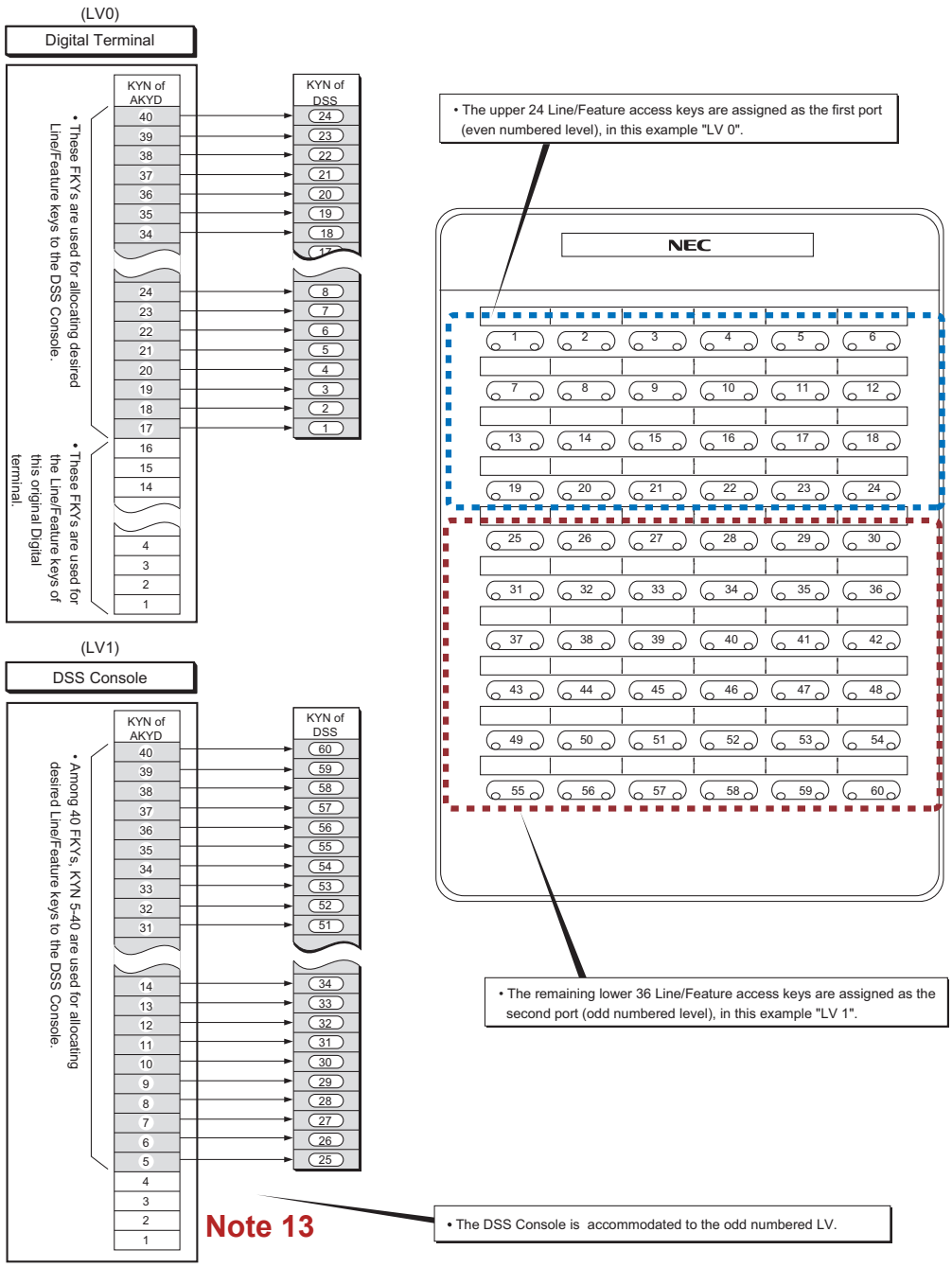
KYN	KYI	KD	LINE TN	STN	Remarks
1	2	0	1	200	Digital terminal Prime Line
2	2	0	1	201	DSS Console station number

KYN=5-40: Assign KYNs 25-60 keys on DSS Console (ADM).
KYI: X (Service Index)
0: Key is not used
1: Function Key
2: Multi-line
FKY: 1-320 (Function Key Number)
KD: 0-2 (Kind of Line)
TN: X
STN: XXX
RG: 0-7 (Ring Information for Each Line)
ICM: 0-2 (Kind of Intercom)
G-ID: 1-50 (Group ID Number)

DSS Key Assignment for DSS Console (ADM)



To designate the meaning of the upper 24 keys on DSS Console (ADM), FKYs 17-40 of a Digital terminal that works in combination with DSS Console are used. The remaining 36 keys on DSS Console (ADM) are assigned using FKYs 5-40 of the original port of DSS Console (ADM). The relationship between KYN, that appears in the AKYD command, and 60 keys on DSS Console (ADM) are shown in the figure below. This figure also shows an example of DSS key allocation on DSS Console (ADM) when it is connected to LV 1.



Note 13: Note the following conditions:

- To KYN=1, assign the Prime Line of Digital terminal paired with the DSS Console.
  In the case of Data Programming Procedure in this manual: AKYD, KYI=2, KD=0, STN=200
- To KYN=2, assign the DSS Console station number which is accommodated in the second port (odd numbered level).
  In the case of Data Programming Procedure in this manual: AKYD, KYI=2, KD=0, STN=201

STEP 3: **ADKS** - Assign Digital terminal Key Status Data to the station number for Digital terminal (STN=200) which is assigned by ASDT command.

TN: X
STN: 200 ← Digital station number
SKP: 0-15 (Soft Key Pattern Data)
LKP: 1 (16 Line/Feature buttons + DSS keys)
PG: 0/1 (Soft key/Feature key is allocated for the PAGE CHANGE key)

Example of Data Programming for DSS Console (ADM)

The following table provides all of steps for Data Programming.

ASDT

TENANT NUMBER (TN) 1-63	STATION NUMBER (STN) MAXIMUM SIX DIGITS	LINE EQUIPMENT NUMBERS (LENS)				TELE. EQUIP. CLASS (TEC) 1-31	ROUTE REST. CLASS (RSC) 0-15	Service Feature Restriction Class (SFC) 0-15	REMARKS
		MG	U	G	LV				
X	200	0 0	0	1 0	*0	12	X	X	For Digital terminal
	201	0 0	0	1 0	**1	12	X	X	For DSS Console (ADM)

Note: *0 = Even-numbered LENS level / **1 = Odd-numbered LENS level

AKYD

For Line/Feature Access keys: KYNs 1-24 (Digital terminal)

MY LINE		TYPE OF DTERM (TP) 0-3	RING DATA (RING) 0-3	FORM OF LINE PREFERENCE (PRI) 0-3	LINE PREFERENCE (LN PRE) 0/1	LINE PREFERENCE FOR SPEAKER BUTTON (SPK) 0
TENANT NUMBER (TN) 1-15/63	STATION NUMBER (STN) MAXIMUM SIX DIGITS					
*X	200 (=Digital terminal)	1	X	0	0	

LINE PREFERENCE FOR ANSWER BUTTON (ANS) 0-2	LINE PREFERENCE FOR ORIGINATING BUTTON (ORG) 0/1	MESSAGE WAITING DATA DISPLAY SELECTION (MWD) 0/1	Prime Line		OG FROM PRIME LINES 0/1
			TENANT NUMBER (TN) 1-63	STATION NUMBER (STN) MAXIMUM FIVE DIGITS	
		X	X	200 (=Digital terminal)	0

KEY NUMBER (KYN) 1-16	SERVICE KEY INDEX (KYI) 0-2	FEATURE KEY NUMBER (FKY) 1-65	MULTI LINE			INTERCOM	
			LINE IDENTITIY (KD) 0-3	TENANT NUMBER (TN) 1-15	STATION NUMBER (STN) MAXIMUM FIVE DIGITS	KIND OF INTERCOM (ICM) 0-2	GROUP ID NUMBER (G-ID) 1-50
**1XXXXXX	2 (Multi Line)	**1XXXXXX	0 (Line)	**1XXXXXX	200	**1XXXXXX	**1XXXXXXX

**2	X	X	X	X	X	X	X
**3	X	X	X	X	X	X	X
**4	X	X	X	X	X	X	X
**5	X	X	X	X	X	X	X
**6	X	X	X	X	X	X	X
**7	X	X	X	X	X	X	X
**8	X	X	X	X	X	X	X
**9	X	X	X	X	X	X	X
**10	X	X	X	X	X	X	X
**11	X	X	X	X	X	X	X
**12	X	X	X	X	X	X	X
**13	X	X	X	X	X	X	X
**14	X	X	X	X	X	X	X
**15	X	X	X	X	X	X	X
**16	X	X	X	X	X	X	X
17	X	X	X	X	X	X	X
18	X	X	X	X	X	X	X
19	X	X	X	X	X	X	X
“	“	“	“	“	“	“	“

Note: *Buttons 1-40 are assignable / **Assignment of Line/Feature Access keys (KYNs 1-16) on Digital terminal 200/ **Assignment of Line/Feature Access keys (KYNs 1-24) on DSS Console (ADM) 201

38	X	X	X	X	X	X	X
39	X	X	X	X	X	X	X
40	X	X	X	X	X	X	X

For Line/Feature Access keys: KYNs 25-60 (DSS Console (ADM))

MY LINE		TYPE OF DTERM (TP) 0-3	RING DATA (RING) 0-3	FORM OF LINE PREFERENCE (PRI) 0-3	LINE PREFERENCE (LN PRE) 0/1	LINE PREFERENCE FOR SPEAKER BUTTON (SPK) 0	
TENANT NUMBER (TN) 1-15/63	STATION NUMBER (STN) MAXIMUM SIX DIGITS						
*X	201 (DSS Console (ADM))	1	X	0	0		
LINE PREFERENCE FOR ANSWER BUTTON (ANS) 0-2	LINE PREFERENCE FOR ORIGINATING BUTTON (ORG) 0/1	MESSAGE WAITING DATA DISPLAY SELECTION (MWD) 0/1		Prime Line		OG FROM PRIME LINES 0/1	
				TENANT NUMBER (TN) 1-63	STATION NUMBER (STN) MAXIMUM FIVE DIGITS		
		X		X	200 (=Digital terminal)	0	
KEY NUMBER (KYN) 1-16	SERVICE KEY INDEX (KYI) 0-2	FEATURE KEY NUMBER (FKY) 1-65	MULTI LINE			INTERCOM	
			LINE IDENTITY (KD) 0-3	TENANT NUMBER (TN) 1-15	STATION NUMBER (STN) MAXIMUM FIVE DIGITS	KIND OF INTERCOM (ICM) 0-2	GROUP ID NUMBER (G-ID) 1-50

**1	2 (Multi Line)		0 (Line)	X	201 (DSS Console (ADM))		
2	2 (Multi Line)		0 (Line)		200		
3							
4							
5	X	X	X	X	X	X	X
6	X	X	X	X	X	X	X
7	X	X	X	X	X	X	X
8	X	X	X	X	X	X	X
9	X	X	X	X	X	X	X
10	X	X	X	X	X	X	X
“	“	“	“	“	“	“	“
***32	X	X	X	X	X	X	X
***33	X	X	X	X	X	X	X
***34	X	X	X	X	X	X	X
***35	X	X	X	X	X	X	X
***36	X	X	X	X	X	X	X
***37	X	X	X	X	X	X	X
***38	X	X	X	X	X	X	X
***39	X	X	X	X	X	X	X
***40	X	X	X	X	X	X	X

Note: *Buttons 1-40 are assignable / **Assignment of Line Appearance Data for DSS Console (ADM) 201/ ***Assignment of Line/Feature Access keys (KYNs 25-60) on DSS Console (ADM) 201

ADKS

TENANT NUMBER (TN) 1-63	STATION NUMBER (IP STN) MAXIMUM SIX DIGITS	SOFTWARE KEY PATTERN DATA (SKP) 0-15	KEY PATTERN DATA (LKP) 1-3	LCD PAGE (PG) 0/1
X	200		1	

D-153 Distinctive Ringing - Caller ID

General Description

This feature is a part of CALLER ID Service. This feature provides a Distinctive Ringer Pattern (0, 1, 5, 6 or 7) according to the received Calling Number.

Note: This feature is available in North America only.

Note: For the definition of terms related to ringer patterns, see DISTINCTIVE RINGING [D-10].

Operating Procedure

No manual operation is required.

Service Conditions

1. ON-OFF timing of Ringer Patterns 0, 1, 5, 6 and 7 can be changed using ASYD. System Data assigned for Ringer Patterns 6 and 7 cannot be applied to analog telephones.
2. Ringer Patterns 6 and 7 can be used for ELC cards and the LC card. With regard to other LC cards, Ringer Patterns 0, 1, 5 are available for this feature.
3. When the Calling Number is not received, this service is not available (the Ringer Pattern is determined according to the parameter 12:DR of ARTD).
4. The Calling Number for this service is assigned to the parameter CID of ACID. When only the first part of the Calling Number is assigned to the CID, this feature is applied to any Calling Number that begins with the Assigned Number. For example, when 1471 is assigned to the CID, this feature is applied to any Calling Number that begins with 1471 such as 1471-12-3456 and 1-471-234-5678.
5. The following numbers cannot be assigned to the parameter CID of ACID:
   - The first part of the already Assigned Number
   - The number whose first part is the same as the already Assigned Number

   For example, when “123” is already assigned to the CID, such numbers as “1”, “12” or “1234” cannot be assigned.
6. When the Calling Number is received via CCIS Line, this feature is available for the first 16 digits (provided that “1” is assigned to 98: CI of ARTD).
7. This service is available only for stations (Single Line telephones and Desktop terminals).
8. This service is available when an Incoming Call is directly terminated to a station and/or an Incoming Call is transferred to a station using Call Forwarding - All Calls/Busy Line/Don’t Answer.
9. This service is not available when a call is terminated to an ISDN terminal.
10. Fax Ringer overrides this feature.
11. This feature takes precedence over the Personal Ringer function.
12. A maximum of 12 Calling Numbers can be assigned for a single station.
13. Ringer Patterns 2, 3 or 4 cannot be assigned for this feature. When one of those Ringer Patterns is assigned from PCPro, recommended value Ringer Pattern 6 is automatically set.
14. When SYS3 is changed, circuit card (LC, ELC) initialization is required.
15. A CH-12COTB or PA-8COTBJ circuit card is required for Analog Caller ID.

16. When multiple incoming calls are simultaneously terminated to the Multi Line on the Desktop terminal, the priority of Ringer Pattern is as follows:

Priority	1st	2nd	3rd	4th	5th
Ringer Pattern	RG 7	RG 6	RG 5	RG 0	RG 1

17. When no value is specified for either Ringer Pattern 6 or 7 (ASYD, SYS3, Index 8/Index 9=0), default (Recommended) values of Ringer Patterns 0 to 7 are as follows:

Ringer Patterns 0 to 7

X: Available N: Not Available –: Not applicable

Ringer Pattern	Customization of Ringer Pattern	Customization of Ring Cycle (Note 1) Duration through System Data		Recommended Ring Cycle Duration (Ringing/Silent) (sec.)	Used for
		ELC	LC	ELC	
Ringer Pattern 0	X	X	X	1/2	Incoming Trunk Calls
Ringer Pattern 1	X	X	X	2/4	Station Calls
Ringer Pattern 2	N	N	N	-	Fax
Ringer Pattern 3	N	N	N	-	Wake Up Calls
Ringer Pattern 4	N	N	N	-	Not used.
Ringer Pattern 5	X	X	X	2/4	Call Back, Outgoing Trunk Queuing, Call Waiting, Auto/Manual Intercom, and Dial Intercom
Ringer Pattern 6	X	X	N	1/2	-
Ringer Pattern 7	X	X	N	1/1	-

Note 1: The Ring Cycle consists of a ringing period and a silent period.

Programming

STEP 1: **ASYD** - SYS1, Index 417, Bit 0. Recall Ringer Pattern for CALL BACK [C-1], CALL WAITING - ORIGINATING [C-31], and OUTGOING TRUNK QUEUING [O-2].

0/1 = Ringer Pattern 5/Ringer Pattern 1.

SYS3, Index 0: Duration of the Ring Cycle (Timer Value) for Ringer Pattern 0.

SYS3, Index 1: Duration of the Ring Cycle (Timer Value) for Ringer Pattern 1.

SYS3, Index 3, Bit 0: Two Ringings (0.4 seconds.ON, 0.2 seconds.OFF, 0.4 seconds.ON) per Ringing Period for Ringer Pattern 0.

0/1 = Disabled/Enabled.

SYS3, Index 3, Bit 1: Three ringings (0.4 seconds-ON, 0.2 seconds-OFF, 0.8 seconds-ON, 0.2 seconds-OFF, 0.4 seconds-ON) per Ringing Period for Ringer Pattern 1.

0/1 = Disabled/Enabled.

SYS3, Index 3, Bit 5: Three ringings (0.4 seconds-ON, 0.2 seconds-OFF, 0.8 seconds-ON, 0.2 seconds-OFF, 0.4 seconds-ON) per Ringing Period for Ringer Pattern 5.

0/1 = Disabled/Enabled.

SYS3, Index 7: Duration of the Ring Cycle (Timer Value) for Ringer Pattern 5.

SYS3, Index 8: Duration of the Ring Cycle (Timer Value) for Ringer Pattern 6.

SYS3, Index 9: Duration of the Ring Cycle (Timer Value) for Ringer Pattern 7.

STEP 2: **ACID** - Assign the Calling Number for this service.

TYPE: 1

TN: Tenant Number

STN: Station Number

CID: Calling Number (maximum 28 digits)

SRV: 6 (Distinctive Ringing - Caller ID)

RGP: Ringer Pattern (0, 1, 5 - 7)

D-156 Dual Station Call

General Description

This feature allows a user to simultaneously call two Stations (referred to as a main station and sub-station), which are accommodated in two different Nodes within an FCCS Network, by dialing one Telephone Number of the main station. At that time, the call can be answered from either the Main or the sub-station. Additionally, when this service is used with BOSS - SECRETARY Service, the main station can forward a call automatically from a Secretary Station.

This service requires the data setting of FCCS. For more information on FCCS Data, see the Data Programming Manual - FCCS.

Operating Procedure

To activate this service

When DUAL STATION CALL is activated without BOSS - SECRETARY Service

1. To call two Desktop terminals from the Station:
   a.) Lift the handset.
   b.) Receive Dial Tone.
   c.) Dial the Telephone Number of a main station.
   d.) Main station and sub-station ring.
2. To pick up a call:
   a.) A Station answers the call.
   b.) The caller is connected with the Station which picked up the call (the other Station stops ringing).

Note: If the call is abandoned, both Desktop terminals stop ringing.

When DUAL STATION CALL is activated with BOSS - SECRETARY Service (B-MOVE Feature key is not set)

Station A ⟶ Main station (Boss Station)
Station B ⟶ Sub-station
Station C ⟶ Secretary Station which accommodates Sub Line to be assigned the main station's Number

1. To call two Desktop terminals from a Station:
   a.) Lift the handset.
   b.) Receive Dial Tone.
   c.) Dial Station A's Telephone Number.
   d.) Station A, Station B and Station C ring.
2. To pick up a call via Station C:
   a.) Station C answers the call.
   b.) The caller is connected with Station C.
   c.) Station A and Station B stop ringing.
   d.) Station C presses Station A's Line.
   e.) The caller hears MSC.
   f.) Station C calls Station A.

When DUAL STATION CALL is activated with the BOSS - SECRETARY Service (B-MOVE Feature key is set)

Station A ⟶ Main station (Boss Station)
Station B ⟶ Sub-station (the destination for the B-MV Feature key)
Station C ⟶ Secretary Station which accommodates Sub Line to be assigned the main station's Number

1. To call two Desktop terminals from a Station:
   a.) Lift the handset.
   b.) Receive Dial Tone.
   c.) Dial Station A's Telephone Number.
   d.) Station A, Station B and Station C ring.
2. To answer a call via Station C:
   a.) Station C answers.
   b.) The caller is connected with Station C.
   c.) Station A and Station B stop ringing.
   d.) Station C presses Station A's Line.
   e.) The caller hears MSC and Station C's user hears Ringback Tone.
   f.) Station C calls Station B as assigned Station A's destination for the B-MV Feature key. Station B rings.

Note: When Station B is busy, Station C hears Busy Tone. Station C can perform a switch hook flash to connect with the Calling Party again. This operation is the same as CONSULTATION HOLD - ALL CALLS [C-17].

Note: When a sub-station is programmed as BOSS Station, the current BOSS - SECRETARY Service is activated.

Note: When the destination for NUMBER SHARING is set by pressing the PS key, the Secretary Station cannot transfer a call to the sub-station.

To set/cancel the B-MV Feature key

1. To set the B-MV Feature key:
   a.) When a Desktop terminal is idle, press the B-MV Feature key.
   b.) Lamps of the B-MV Feature key accommodated in the main station and sub-station light. (When the PS key is assigned to the Desktop terminal, PS key lights also.) The LCD displays as shown below:

FORWARD SET

2. To cancel the B-MV Feature key:
   a.) Press the B-MV Feature key while the Desktop terminal is idle.
   b.) Lamps of the B-MV Feature key accommodated in the main station and sub-station go OFF. (When the PS key is assigned to the Desktop terminal, PS key also goes OFF.) The LCD displays as shown below:

FORWARD CANCEL

Service Conditions

1. A Telephone Number must be assigned to main station and sub-station.
2. Only the Desktop terminal can be programmed as main station and sub-station.
3. When using ANSDN command, only stations within the FCCS Network can be assigned as main station and sub-station.
4. Main station or sub-station assignment cannot be duplicated for another pair.
5. Only one Telephone Number as sub-station can be assigned to a main station.
6. B-MV Feature key can only be set when the Station is idle.
7. The Calling Party's LCD will display the Telephone Number of a main station no matter which Station picks up the call. Also, when sub-station picks up a call from CCIS Line, the Telephone Number of the main station is sent to the Calling Party.
8. MESSAGE REMINDER [M-19] is available for the Terminating Station Service (all Stations).
9. MESSAGE WAITING Service set by the CMWL is available for the preassigned Station.
10. CALL BACK [C-1] is available for only the Station to be set CALL BACK.
11. This service is activated in a Hotel/Motel System, but not activated with Hotel/Motel Application Service.
12. When a sub-station is in communication, the Telephone Number of the sub-station is sent to OAI or MFC as Calling Number Notification.
13. When a sub-station is busy, a call terminates to the main station only. However, when a main station is busy, a call is not terminated to an idle sub-station.
14. When the Telephone Number of a sub-station is dialed, a call is terminated to the sub-station only.
15. When this Service is activated with BOSS - SECRETARY Service and a sub-station is assigned as a BOSS Station, the BOSS - SECRETARY service is activated as usual by pressing Boss's Line accommodated in Secretary Station's Sub Line.
16. When a sub-station is called through FCCS:
    - FLEXIBLE ROUTING - FCCS [F-35] is not available, even though all FCCS are busy.
    - If all Connection Trunks are busy when the sub-station tries to answer the call, the call cannot be established and Busy Tone is sent.

Interactions

1. When any of these services is set to a main station, the service has priority over DUAL STATION CALL:

CALL FORWARDING - BUSY LINE	[C-2]
CALL FORWARDING - ALL CALLS	[C-5]
DO NOT DISTURB	[D-11]

2. When CALL FORWARDING - DON'T ANSWER is set to a main station, it is activated depending on the time-out data programmed in the main station.
3. When CALL BLOCK is set to a main station, which is in the middle of being called for DUAL STATION CALL, CALL BLOCK has priority over this feature. (CALL BLOCK cannot be set to a sub-station while DUAL STATION CALL is activated.)
4. The LCD for My Line displays each Station's Telephone Number.

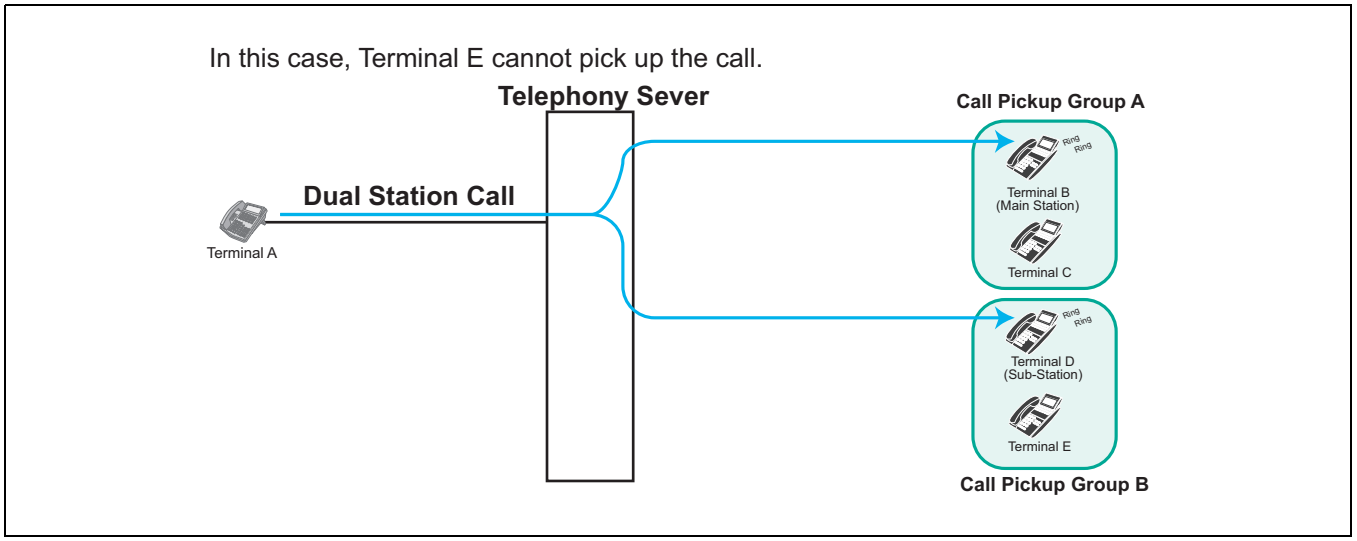
5. When a main station is assigned as a member of a UCD Group or a STATION HUNTING Group, the priority order of these services is shown in the table below:

Main Station		Controlling/Pilot Station		The Other Member Station	
B-MV Key		None	Set	None	Set
Group	Station Hunting-Pilot	1	1	1	2
	Station Hunting-Circular	1	1	1	2
	UCD	1	2	1	2

Legend:

1 = Both main station and sub-station are called (DUAL STATION CALL is activated).
2 = UCD/STATION HUNTING is activated.

6. When CALL FORWARDING-ALL CALLS/-BUSY LINE/-DON'T ANSWER is set to a sub-station:
   a.) DUAL STATION CALL is activated by dialing the main station's Telephone Number.
   b.) The services shown above are activated by dialing the sub-station's Telephone Number.
7. A Station cannot be used simultaneously for DUAL STATION CALL and NUMBER SHARING because the data of these services are used in common. For more detailed information on NUMBER SHARING, refer to the PCS Features and Specifications.
8. When a Calling Party sets VOICE CALL [V-2] or performs switch hook flash while calling a main station and sub-station, the sub-station stops ringing.
9. This service is not activated when a call is terminated via INTERCOM CALLING [I-9], AUTOMATIC/ MANUAL INTERCOM [A-30], DIAL INTERCOM [D-87] or MANUAL SIGNALING [M-47] (only the main station rings).
10. This service is not activated while CALL BACK [C-1], CALL BACK DELAYED [C-73], CALL BACK - CCIS [C-44], ATTENDANT CAMP-ON WITH TONE INDICATION [A-1] or ATTENDANT CAMP-ON WITH TONE INDICATION - CCIS [A-45] is activated.
11. When a Main or sub-station accommodates PS key and B-MV Feature key, both keys light by pressing any of these keys.
12. When DO NOT DISTURB [D-11] is set to a sub-station, the Incoming Call routed to a main station is terminated to the main station only.
13. When a sub-station is set CALL BLOCK [C-150], the Incoming Call routed to a main station terminates to both the main station and sub-station.
14. When DO NOT DISTURB [D-11] or CALL BLOCK [C-150] is set to a sub-station, the Incoming Call routed to the sub-station is restricted.
15. When a main station and sub-station belong to different Call Pickup - Groups, a Dual Station Call cannot be picked up from Terminal in Pickup Group that the sub-station belongs to. For details, refer to the following figure.



16. When a sub-station is called through FCCS, FLEXIBLE ROUTING - FCCS [F-35] is not available.
17. When a main station does not pick up a CALL FORWARDING - DON'T ANSWER Call within the pre-assigned time via System Data, the sub-station stops ringing.
18. The B-MV Feature key does not work on the SIP Handler Controlled SIP terminal.

Programming

STEP 1: **ASYDL** - SYS1, Index 513: Assign 01 (hexadecimal) as the LDM's Memory Block Number
SYS1, Index 514: Assign 01 (hexadecimal) as the NDM's Memory Block Number

STEP 2: **ANSDL/ANSDN** - Assign the related data of Dual Station Call.
MAIN UGN, MAIN TELN:
SUB UGN, SUB TELN:
OD: 0 = Only Main Telephone Number is used for SMDR and MCI Mailbox Number
1 = Both Main Telephone Number and Sub Telephone Number are used for SMDR and MCI Mailbox Number

Note: When OD = 0, the mailbox for the sub-station must be deleted.

MW (This data is available when OD = 0)
0 = Message Waiting Control for Main and Sub Telephone Numbers
1 = Message Waiting Control for Main Telephone Number only

Note: When the data value for MW changes from 0 to 1, make sure there are no messages left (MWL is OFF). If MWL is ON at that time, the old lamp indication stays after the data is changed.

LM: 0 = Number Sharing
1 = Dual Station Call

STEP 3: **AKYD** - Assign Sub Line Appearance on Secretary Telephone of main station and sub-station. Assign KYI 1 (Feature Key) and Feature key B-MV [127].

STEP 4: **ADSL** - Assign the Soft Key Pattern Data.
SN: 0
FKY: 127 (B-MV key)

STEP 5: **ADKS** - Assign the B-MV key to the Soft key of each Station. **Note 1**
TN (Tenant Number), STN (Station Number)
SKP (Soft Key Pattern):1~15
LKP (Key Pattern Data):
1 = 16 Line/Feature buttons + DSS keys
2 = 24 Line/Feature buttons + DSS keys
3 = ALL Line/Feature buttons
PG: 0 = Page change key appears on a software key(>>>)
1 = Page change key disappears on a software key

Note 1: When the Soft key data is used on a system basis (SKP = 0 is assigned by ADSL), this data is not necessary.

FCCS Programming

Refer to the Data Programming Manual - FCCS.

D-157 Dialed Number Primary Restriction

(Series 8400 software feature)

General Description

This feature provides the ability to block Dialed Numbers up to 24 digits long. This function is very similar to Primary Code Restriction via APCR. The main difference is that the number of restrictions has been expanded.

Operating Procedure

1. No special operating procedures are required by the Station user to activate this feature.
2. The user dials an Outside Number and is restricted or allowed based on the number dialed. The following are examples using this feature with the North American Numbering Plan:
   The number 1-800-444-2222 is a Toll Free Number that is answered and then transferred to a per minute charged number. The cost of this call can be from $1.00 to $25.00 per minute after the transfer is made. By use of this feature, the Telephony Server administrator could assign 918004442222 in ADNPRL to stop users from dialing the original Number.
   An Outgoing Call Center makes calls to numbers to solicit products and services to the public. If a person at 972-555-1234 requests not to be called by the Call Center any more, by law, they cannot call this number again for one year. If they do call the number again with in a one year time period, the Call Center may have a fine imposed upon them by the Federal Communications Commission.
   By using this feature, the Telephony Server administrator can assign 919725551234 in ADNPRL and stop calls from being made to the number.

Service Conditions

1. This service is based on the number dialed, not the number sent on the Trunk.
2. The Access Code is included in the number to be restricted.
3. This service is limited to a maximum of 24 digits.
4. ADNPRL cannot be used as an Allow Table to override any other restriction. It can only be used to restrict.
5. This feature only functions on Outgoing Calls to External Trunk Routes from Stations.
6. Station Numbers cannot be restricted using this service.
7. Tandem Calls cannot be restricted by this service, except RSC mate-check with CCIS.
8. This service can be used while a Station has a call on Consultation Hold, the RSC of the Transferring Party will be used in this case.

9. The following shows the Restriction Order of Tables checked:
   a.) The quantity of numbers that may be entered by this service is limited to the amount of available Local Data Memory. Currently there is 6 Mb of LDM available for use by all LDM features, DDMO may be used to calculate the amount of free LDM.
   b.) The Attendant Console is not affected by this service.
   c.) This service supports the use of Authorization Codes with the following limitations:
      - For this service to work with Authorization Codes, the code must be assigned as a Restriction Type 1 with a specific RSC and SFC assigned, the RSC value must be assigned a value between 1 and 15.
      - If Restriction Type 2 is used, calls using that Authorization Code will be restricted, based on any information assigned in ADNPRL, regardless of which RSC it is assigned to.
      - When restriction via OAI is required when using SCF 1, 3, 4, and 7, the Notify Field in the SCF message must be set to three or four, depending on the application. Please see the restriction for Class Error Service Feature Specification for more information on this service.
      - If the number 9972 is entered in to ADNPRL to be restricted for all SFCs, and 99725185555 is dialed, the call will be restricted (all calls beginning with 9972 will be restricted).
      - This feature is only supported when LCR routing is used. LCRS, OGC, OGCA, or other types of call routing are not supported with this feature.

Programming

STEP 1: **ADNPRL** - Assignment of Dialed Number Primary Restriction LDM. This command is used to assign and delete numbers to be restricted.

TN: Tenant

PDC: The number to be restricted

RSC: Route Restriction Class (0 - 15)

Because the ADNPRL Table is a Restrict Table, allowing all RSCs (RSC 0 ~ 15 = 1) tells the system to remove that entry from the table.

D-158 Digital Announcement Trunk Recording via Remote Access

(Series 8400 software feature)

General Description

This feature allows a user to record/playback an announcement on DAT via remote access to the System using an external Central Office Trunk. This is required by any user who needs to re-record or to replay (playback) DAT Announcements because of severe weather closure of Office/Call Center. This feature provides the following four functions:

- Recording/Playback by designating Route and Trunk
- Recording/Playback by designating Route
- Recording/Playback of an announcement for a UCD Group
- Recording/Playback of an announcement for ACD

Note: This feature is only available in North America.

1. Recording DAT via remote access:
   a.) The outside user dials the Remote Access Number to connect to the System. Receive Ringback Tone through Public Network.
   b.) If there is an Announcement Service, it will be heard. Receive Service Set Tone.
   c.) Dial the Authorization Code. Receive the Dial Tone.
   d.) Dial the DAT Recording Access Code. Receive Special Dial Tone.
   e.) For recording via Route and Trunk Number:
   Enter Route Number and Trunk Number. The Special Dial Tone stops.
   Or
   For recording via Route Number (all Trunks in Route are recorded):
   Enter Route Number and enter #. The Special Dial Tone stops.
   Or
   For recording via UCD Group:
   Enter * and enter the UCD Controlling Station Number. The Special Dial Tone stops.
   Or
   For recording via ACD Message Number:
   Enter # and enter the Message Number that has been assigned to ACD. The Special Dial Tone stops.
   f.) Receive Service Set Tone after completion of entry of the Route and Trunk Number, Route Number, UCD Group Number or ACD Message Number.

   Note: The Service Set Tone continues for three seconds.

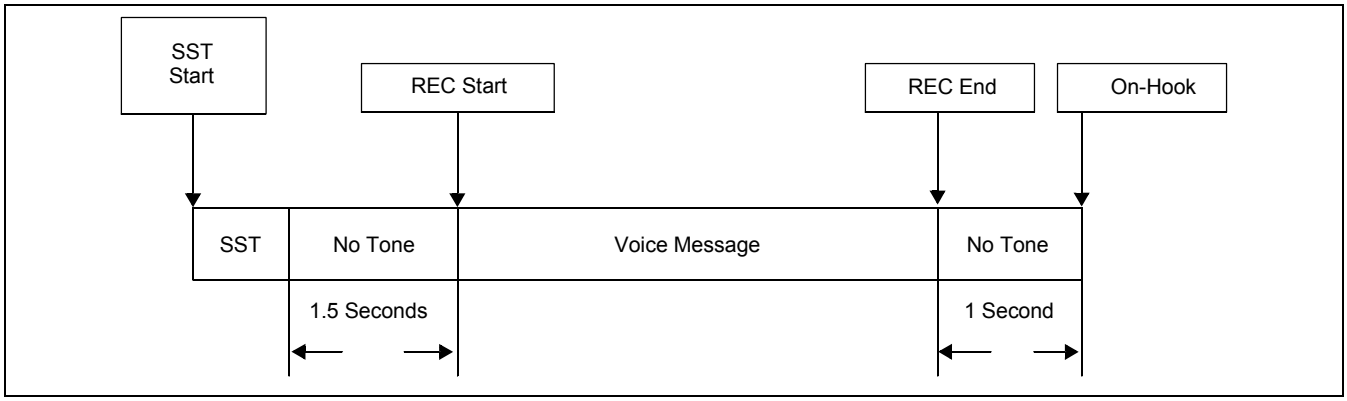
   g.) Start your recording when the Service Set Tone stops.
   h.) The outside user releases the call to finish recording.
2. Playback DAT via remote access:
   a.) The outside user dials the Remote Access Number to connect to the System. Receive Ringback Tone through Public Network.
   b.) If there is an Announcement Service, it will be heard. Receive Service Set Tone.
   c.) Dial the Authorization Code. Receive the Dial Tone.

d.) Dial the DAT Playback Access Code. Receive Special Dial Tone.
e.) For playback by Route and Trunk Number
Enter Route Number and Trunk Number. The Special Dial Tone stops.
Or
For playback by Route Number:
Enter Route Number and enter #. The Special Dial Tone stops.
Or
For playback by UCD Group:
Enter * and enter the UCD Controlling Station Number. The Special Dial Tone stops.
Or
For playback by ACD Message Number:
Enter # and enter the Message Number that has been assigned to ACD. The Special Dial Tone stops.
f.) Enter Route Number and Trunk Number. The Special Dial Tone stops.
g.) A recorded message of the designated Trunk is automatically played back.
h.) The outside user releases the call to finish playback of the recorded message.

Service Conditions

1. The maximum values for RT, TK, and Message Number for ACD are:
   RT: 001 - 899
   TK: 001 - 255
   Message Number: 01 - 58
2. The maximum recording length is 60 seconds (30 seconds in the case of UCD).
3. This feature is not effective for Trunks currently being dubbed or that are placed in busy status.
4. When a RT, TK, UCD Controlling Station, or Message Number that has not been assigned is dialed, the caller will be connected to Reorder Tone.
5. When all DAT Trunks are busy, the outside user (via Incoming Trunk) will hear a Busy Tone after dialing RT/TK/UCD Controlling Station Number/ACD Message Number.
6. An outside user cannot record a message to an Announcement Trunk in which a recording/dubbing is being made.
7. When the system encounters a Busy Trunk in an attempt to make a dubbing of the message to the remaining Trunks, the Trunk is retried after a 16 second interval. Each Trunk is attempted a maximum of 16 times. If the final attempt is rejected because of busy, the system does not make a dubbing to the Trunk.
8. After completion of dialing RT/TK/UCD Controlling Station Number/ACD Message Number, the outside user hears Service Set Tone. In this instance, if the designated Trunk is making a dubbing or being dubbed, the dubbing is stopped. When the outside user goes on-hook during the Service Set Tone, the dubbing is also stopped.
9. A maximum of four TRKs can be simultaneously dubbed when recording is initiated by dialing RT Number or UCD Control Station Number or ACD Message Number.
10. When in Dubbing mode, the current message will override the previous message.
11. All TRKs in same RT will be dubbed by dialing RT Number without TRK Number.
12. All TRKs (assigned by AUAD) will be dubbed by dialing UCD Control Station Number.
13. All TRKs (assigned by AADT) will be dubbed by dialing ACD Message Number.
14. In the dubbing message processing, if the TRK being dubbed is not a DAT, it will try to dub the next assigned TRK after two seconds.
15. The TRK being used as the primary for dubbing is not available while recording to the other TRKs.

16. A message is recorded in the following sequence. As illustrated in the following figure, 1.5 seconds after the Service Set Tone and one second before releasing, messages are not recorded. This is to insure that no Service Set Tone or Reorder Tones are recorded.



17. Refer to the following table to obtain the approximate time for dubbing.

DUBBING TIME	NUMBER OF TRUNK					
	1	**10**	**15**	**20**	**63**	**1**
6 SECONDS	8	80	120	160	504	8n
16 SECONDS	18	180	270	360	1134	18 × 1
30 SECONDS	32	320	480	640	2016	32 × 1
46 SECONDS	48	480	720	960	3024	48 × 1
60 SECONDS	62	620	930	1240	3906	62 × 1

Note: The dubbing time is calculated by the following equation:
(Dubbing Time) = (Recording Time + 2 seconds) × (Number of Trunks)

18. When accessing the Remote Access DAT Recording feature, a 36-B System Message will output by way of the PCPro/system printer. The 36-B message is generated to the PCPro/system printer upon successful validation of an Authorization/Forced Account Code and a RT/TK, RT, ACD or UCD Access Code. The purpose of this System Message is to inform the technician/System Administrator that an External Trunk attempted to record or playback a message to or from a Digital Announcement Trunk. The 36-B System Message does not guarantee that the operation was successful, only that an access attempt was made. The message is an informational message only and will not generate a Top Unit Alarm.
19. The Remote Access DAT Recording feature will not generate or output an SMDR record.

Interactions

This section describes the interaction between this feature and MULTI CHANNEL RECORDING - DAT [M-48]. The functionality of this feature is exactly the same as that of the MULTI CHANNEL RECORDING - DAT [M-48]. The only difference between these two features is that the feature's originator is different. The originator of this feature is an Incoming Trunk (ICT), while the originator of MULTI CHANNEL RECORDING - DAT [M-48] is a Station.
In function, they will override each other. In short, the current message will override the previous message. For example, one DAT's Trunk was accessed and recorded with Message "A" by a Station using MULTI CHANNEL RECORDING - DAT [M-48]. After that, the same DAT's Trunk is accessed and recorded with Message "B" by an Incoming Trunk using Remote Access DAT Recording [D-158]. The result will be that the Message "B" will override Message "A". Message "B" is recorded on this DAT's Trunk. Basically whichever method is used to record the message last will be the message recorded.

System Message

Upon accessing Remote Access DAT Recording, a 36-B System Message will output by way of the PCPro/system printer. The message is an informational message only, and will not generate a Top Unit Alarm.

The following information is in the System Message

Route - Maximum of three digits (001 - 899)

Trunk - Maximum of three digits (001- 255)

ACD (AADT) Announcement Number - Maximum of three digits (000 - 058)

UCD Controlling Station Number (AUAD) - Maximum of five digits

Automatic Number Identification (ANI)/Calling Party Number (CPN) - Maximum 16 digits (Trunk must provide ANI/CPN)

Authorization Code/Forced Account Code - Maximum of ten digits

System Message details

1: Route + Trunk Number
2: Route Number only
3: ACD (AADT) Announcement Number
4: UCD Controlling Station Number (digit code entered in AUAD)
5: Authorization/Forced Account Code (digit code entered in AATC)
6: ANI/CPN Number [provided by the Incoming Trunk (ISDN, Caller ID, etc.)].

Details of 36-B System Message for Route and Trunk

DFTD 00/05/30 16:16 XXXX (System Name) 3-FPC7
SYSTEM MESSAGE 36-B REMOTE ACCESS DAT RECORDING REPORT
XXXX (System Name) 3-FPC7 May 25 07:58
LP00-0-ACT
1:**AA 10 AA** 20 00 00 00 00 2:00 00 00 00 00 00 00 00 3:00 00 00 00 00 00 00 00
4:00 00 00 00 00 00 00 00 5:98 76 54 32 1A 00 00 00 6:97 25 18 4A 5A 00 00 00
7:00 00 00 00 00 00 00 00 8:00 00 00 00 00 00 00 00

Example of Route and Trunk printout

1: **AA 10 AA 2**0 00 00 00 00 = **RT 001 Trunk 002**
1 2 3 4 5 6

1. First digit of Route Number = 0
2. Second digit of Route Number = 0
3. Third digit of Route Number = 1
4. First digit of Trunk Number = 0
5. Second digit of Trunk Number = 0
6. Third digit of Trunk Number = 2

Example of Authorization/Forced Account Code details

Note: "0" = not used, "A" = digit "0".

5: **98 76 54 32 1A** 00 00 00 = Authorization/Forced Account Code of **9876543210**
12 34 56 78 910

1. First digit Authorization/Forced Account Code = 9
2. Second digit Authorization/Forced Account Code = 8
3. Third digit Authorization/Forced Account Code = 7
4. Fourth digit Authorization/Forced Account Code = 6
5. Fifth digit Authorization/Forced Account Code = 5
6. Sixth digit Authorization/Forced Account Code = 4
7. Seventh digit Authorization/Forced Account Code = 3
8. Eight digit Authorization/Forced Account Code = 2
9. Ninth digit Authorization/Forced Account Code = 1
10. Tenth digit Authorization/Forced Account Code = 0

Example of ANI/CPN

Note: "0" = not used, "A" = digit "0".

6: **97 25 18 4A 5A** 00 00 00 = ANI/CPN **9725184050**
12 34 56 78 910

1. First digit of ANI/CPN = 9
2. Second digit of ANI/CPN = 7
3. Third digit of ANI/CPN = 2
4. Forth digit of ANI/CPN = 5
5. Fifth digit of ANI/CPN = 1
6. Sixth digit of ANI/CPN = 8
7. Seventh digit of ANI/CPN = 4
8. Eight digit of ANI/CPN = 0
9. Ninth digit of ANI/CPN = 5
10. Tenth digit of ANI/CPN = 0

36-B Message detail

Route: **001**
Trunk: **002**
ACD: -
UCD: -
Auth/Forced Account Code - **9876543210**
ANI/CPN: **9725184050**

Detail of System Message for Route only

Note: "0" = not used, "A" = digit "0".

DFTD 00/05/30 16:16 XXXX (System Name) 3-FPC7
SYSTEM MESSAGE 36-B REMOTE ACCESS DAT RECORDING REPORT
XXXX (System Name) 3-FPC7 May 25 07:58
LP00-0-ACT
1:00 00 00 00 00 00 00 00 2:**AA 1**0 00 00 00 00 00 00 3:00 00 00 00 00 00 00 00
4:00 00 00 00 00 00 00 00 5:**98 76 54 32 1A** 00 00 00 6:**97 25 18 4A 5A** 00 00 00
7:00 00 00 00 00 00 00 00 8:00 00 00 00 00 00 00 00

Example of Route only System Message printout

2: **AA 1**0 00 00 00 00 00 00
1 2 3

1. First digit Route Number = 0
2. Second digit Route Number = 0
3. Third digit Route Number = 1

36-B Message detail

Route: **001**
Trunk: -
ACD: -
UCD: -
Auth/Forced Account Code - **9876543210**
ANI/CPN - **9725184050**

Detail of System Message for ACD Announcement

Note: "0" = not used, "A" = digit "0".

DFTD 00/05/30 16:16 XXXX (System Name) 3-FPC7
SYSTEM MESSAGE 36-B REMOTE ACCESS DAT RECORDING REPORT
XXXX (System Name) 3-FPC7 May 25 07:58
LP00-0-ACT
1:00 00 00 00 00 00 00 00 2:00 00 00 00 00 00 00 00 3:**A5 10** 00 00 00 00 00 00
4:00 00 00 00 00 00 00 00 5:**98 76 54 32 1A** 00 00 00 6:**97 25 18 4A 5A** 00 00 00
7:00 00 00 00 00 00 00 00 8:00 00 00 00 00 00 00 00

Example of ACD Announcement printout

3: **A5 1**0 00 00 00 00 00 00
12 3

1. First digit of ACD Announcement = 0
2. Second digit of ACD Announcement = 5
3. Third digit of ACD Announcement = 1

Route: Not Used
Trunk: Not Used
ACD Announcement:**051**
UCD: Not Used
Auth/Forced Account Code:**9876543210**
ANI/CPN: **9725184050**

Detail of System Message for UCD Announcement

Note: "0" = not used, "A" = digit "0".

DFTD 00/05/30 16:16 XXXX (System Name) 3-FPC7
SYSTEM MESSAGE 36-B REMOTE ACCESS DAT RECORDING REPORT
XXXX (System Name) 3-FPC7 May 25 07:58
LP00-0-ACT
1:00 00 00 00 00 00 00 00 2:00 00 00 00 00 00 00 00 3:00 00 00 00 00 00 00 00
4:**48 96** 00 00 00 00 00 00 5:**98 76 54 32 1A** 00 00 00 6:**97 25 18 4A 5A** 00 00 00
7:00 00 00 00 00 00 00 00 8:00 00 00 00 00 00 00 00

Example of UCD Controlling Station printout

4: **48** **96** 00 00 00 00 00 00
12 34

1. First digit of UCD Controlling Station (AUAD) = 4
2. Second digit of UCD Controlling Station (AUAD) = 8
3. Third digit of UCD Controlling Station (AUAD) = 9
4. Fourth digit of UCD Controlling Station (AUAD) = 6

36-B System Message detail

Route: Not Used
Trunk: Not Used
ACD: Not Used
UCD Announcement: **4896**
Auth/Forced Account Code: **9876543210**
ANI/CPN: **9725184050**

Programming

The following commands are used to set this feature in service. For this feature, the programming procedures are same as MULTI CHANNEL RECORDING - DAT [M-48]. Also, the Service Code Appendix (SSCA) and the Service Feature Restriction Class (SFC) are identical to that of MULTI CHANNEL RECORDING - DAT [M-48]. To use this feature, REMOTE ACCESS TO SYSTEM [R-2] and AUTHORIZATION CODE [A-20] both must be assigned.

Programming for MULTI CHANNEL RECORDING - DAT [M-48]

STEP 1: **ANPD** - Assign Number of Necessary Digits (NND) for the first digit of the Access Code of the DAT RECORDING/PLAYBACK.

STEP 2: **ASPA** - Assign Access Codes for the following Service Code Appendix (SSCA).
SRV "SSCA: Service Code Appendix"

SIDA 66 (Multi Channel Recording <REC>)
68 (Multi Channel Recording <RPY>)

STEP 3: **ASFC** - For this feature, Service Feature Restriction Data must be entered on a Service Feature Restriction Class (SFC) basis.
SFI 87 (Multi Channel Recording <REC>) and SFI 90 (Multi Channel Recording <REPLAY>)

STEP 4: **ARTD** - Assign Route Class Data for an Announcement Trunk.

Note: To change the data for existing route (RT), initialize the circuit card.

STEP 5: **ATRK** - Assign the Announcement Trunk Data.

STEP 6: **MBTK** - Assign the Make Idle Status to the Announcement Trunk.

STEP 7: **ASYD** - SYS1, Index 390, Bit 0=1 for Caller ID CPN to appear in System Message 36-C.

STEP 8: **MBTK** - Assign the Make Idle Status to the Announcement Trunk.

STEP 9: **ASYD** - SYS1, Index 390, Bit 0=1 for Caller ID CPN to appear in System Message 36-C.

To delete this feature, the steps are shown as follows

STEP 1: **ATRK** - Delete the Announcement Trunk Data.

STEP 2: **ARTD** - Delete Route Class Data for an Announcement Trunk.

STEP 3: **ASFC** - Delete Service Feature Restriction Class (SFC). For this feature, Service Feature Restriction Data must be entered on a SFC basis.
SFI 87 (Multi Channel Recording <REC>) and SFI 90 (Multi Channel Recording <REPLAY>)

STEP 4: **ASPA** - Delete Access Code for the following Service Code Appendix (SSCA).
SRV "SSCA: Service Code Appendix"
SIDA 66 (Multi Channel Recording <REC>)
68 (Multi Channel Recording <RPY>)

STEP 5: **ANPD** - Delete Number of Necessary Digits (NND) for the first digit of the Access Code of the DAT RECORDING/PLAYBACK.

Programming for Remote Access to System [R-2]

This feature allows a user outside of the System to access the system via an Exchange Network Connection without Attendant or Station assistance. The outside user may originate calls over any or all System facilities.

Programming Conditions

Authorization [A-20] or Forced Account Code [F-7] must be used when Remote Access to System is used in conjunction with Remote Access DAT Recording. For additional Service Conditions see Programming for Remote Access to System [R-2].

STEP 1: **ASYD** - SYS1, Index 43, Bit 0 must =0 - Authorization or Forced Account Code required.

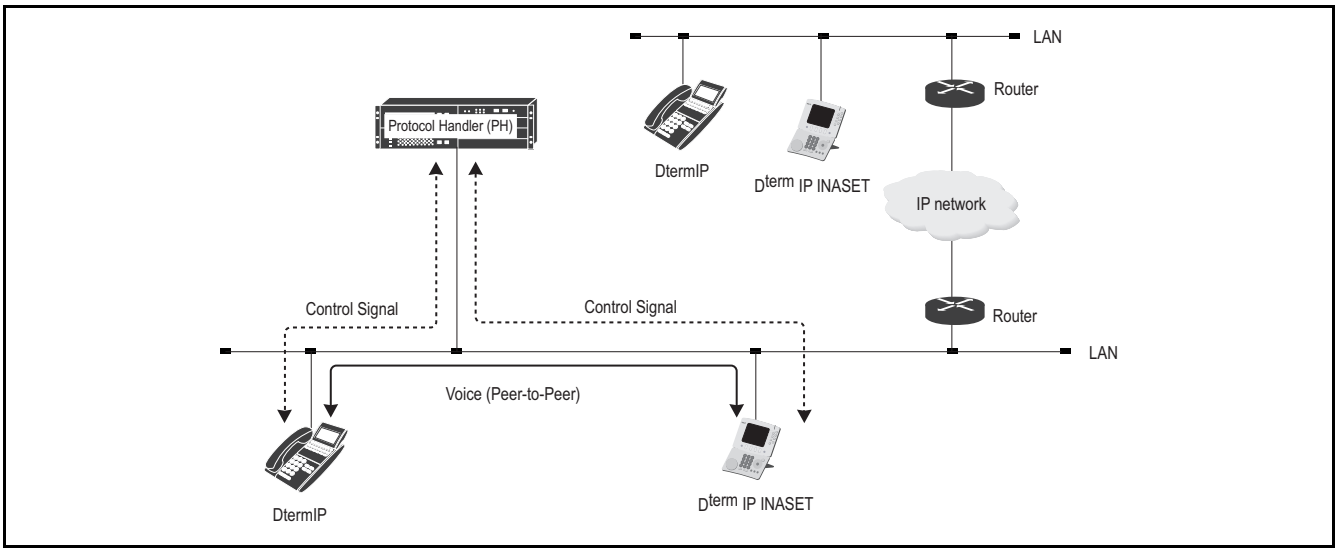
STEP 2: **AATC** - Assign Tenant Number (TN) and Digit Code (DC).
Authorization Code Restriction (ACR) must be 1.
The RSC entered must be allowed to the DAT Announcement Routes.
The SFC entered must be allowed SFI 87 (DAT Record) and/or SFI 90 (DAT Playback).
Only Service Feature Restriction Classes 1 - 15 are valid. SFC "0" is not valid.
For additional Programming Conditions see Programming for Remote Access to System [R-2].

D-159 DtermIP

General Description

This feature allows the system to use the Voice Over IP-capable telephone as a Station.
DtermIP and DtermIP INASET terminals have two LAN connectors (on the back panel) and the internal switching HUB. One of the LAN connectors may be used for the LAN and the other can be used for PC.

SP-PHD controls the registration of all the IP terminals and SIP Multiple Line terminals. For details on SIP Multiple Line terminal, see SIP MULTIPLE LINE OPERATION [S-160].



[DtermIP]

In addition to the existing Digital terminal features, web-based management can be performed on DtermIP, DtermIP INASET, IP Enabled Dterm, and Soft Phone (hereinafter called DtermIP in this feature).

DtermIP Operation Mode

DtermIP operates in one of the following two modes:

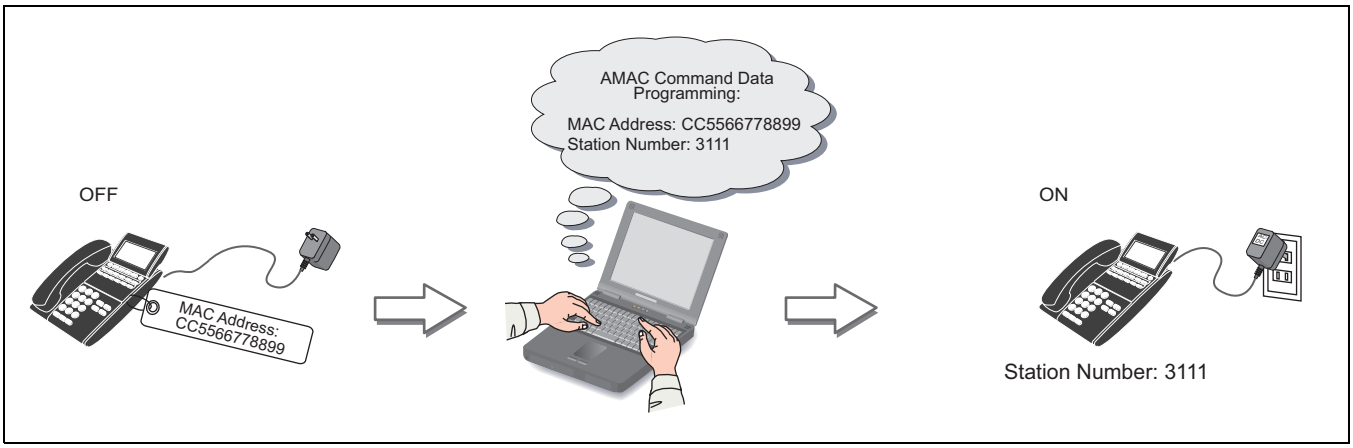
- MAC Authentication Mode
- Login Mode

MAC Authentication Mode

This mode enables each DtermIP to start up automatically, upon being connected to the LAN (with its power ON). Once this mode is set, the terminal does not require the “Login” operation by the terminal user. “Login” operation is necessary only when pressing the LOGOUT Feature key, in a temporary attempt to use the terminal as the one with a different Station Number. For example, assume that the user of IP Station 2000 is now in a distant location and tries to make a call from IP Station 2021. At this time, the user can use IP Station 2021 temporarily as Station 2000 by pressing the LOGOUT Feature key. When this key is pressed, a Login prompt appears on the LCD and the user can log in to the Terminal using the Station Number (2000).

Note: This mode is available as a result of “MAC Address Authentication” by the System. For this reason, be sure to assign the MAC Address Data of the DtermIP as well as its Station Number, SFC/RSC and others. For details on MAC Address Authentication, refer to “MAC Authentication” in this section.

MAC Authentication Mode

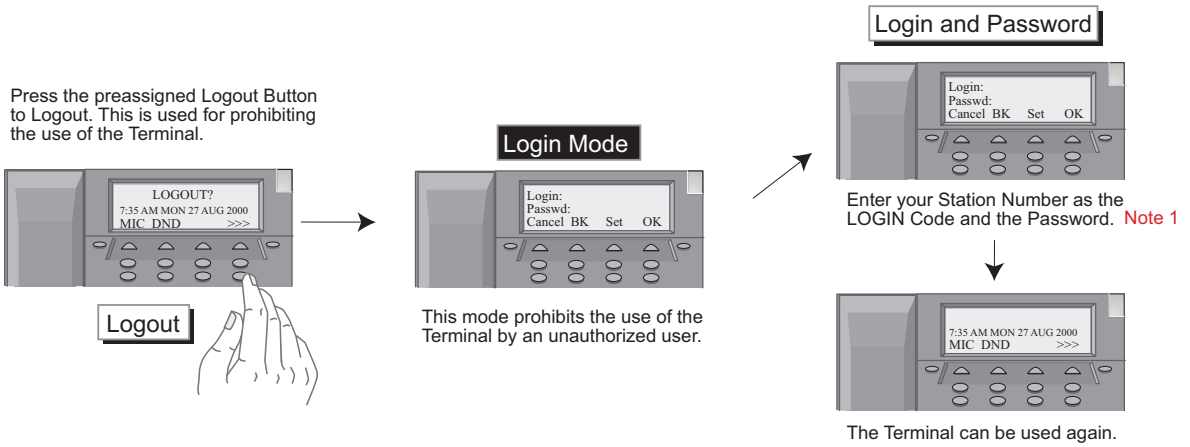


Note: Assignment of MAC address can also be registered from DtermIP. In this case, you don't need to assign the MAC address from PCPro in advance.

Login Mode

In this mode, DtermIP requests the user to enter “Login Code” and “Password” whenever the Terminal power is turned on. When the user enters the Login Code and the Password, the Terminal becomes operational as a Station with the number specified by Login Code. This mode is useful for the user to prohibit the use of the Terminal by unauthorized user while he or she is absent. When the user returns to the desk, the Terminal can be used again by entering the Login Code and Password.

Login Mode

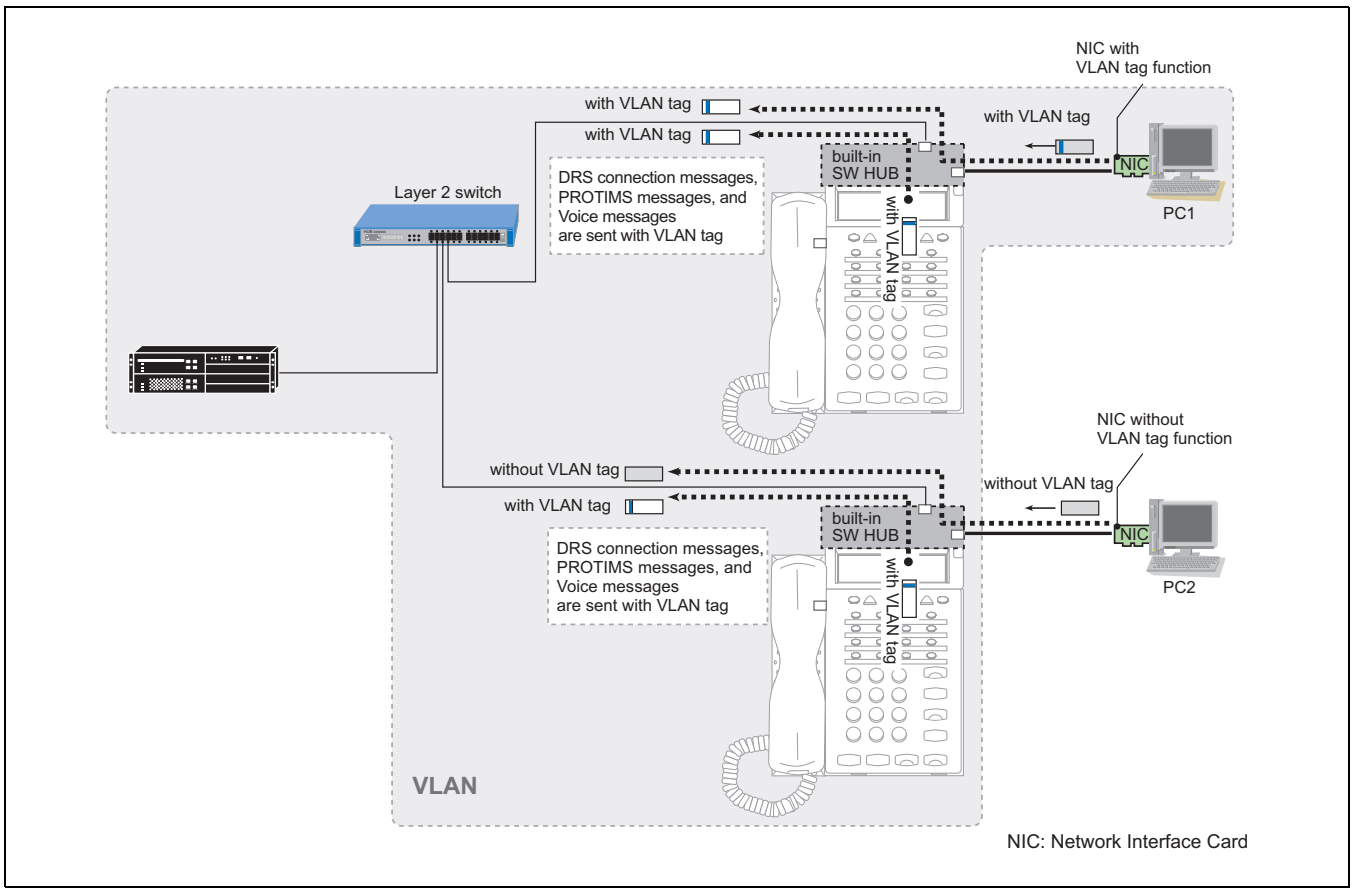


Note 1: If an arbitrary number is assigned as Login Code by ALGCL/ALGCN command, enter the Login Code instead of the Station Number.

Tag VLAN for DtermIP

In the following example, DtermIPs and the System belong to the same VLAN. Each DtermIP embeds the VLAN tag in the frame with MAC Address, PROTIMS messages (PRW), Voice messages (RTP). A PC can be connected to an DtermIP using the PC port. To set VLAN tag to the MAC frames of the PC, a NIC Card having VLAN function is required on the PC. In this example, PC2, not having VLAN tag function, cannot set the VLAN tag.

VLAN for DtermIP



Built-In SW-HUB

The following shows a listing of function of Built-in SW HUB.

Function of Built-in SW HUB

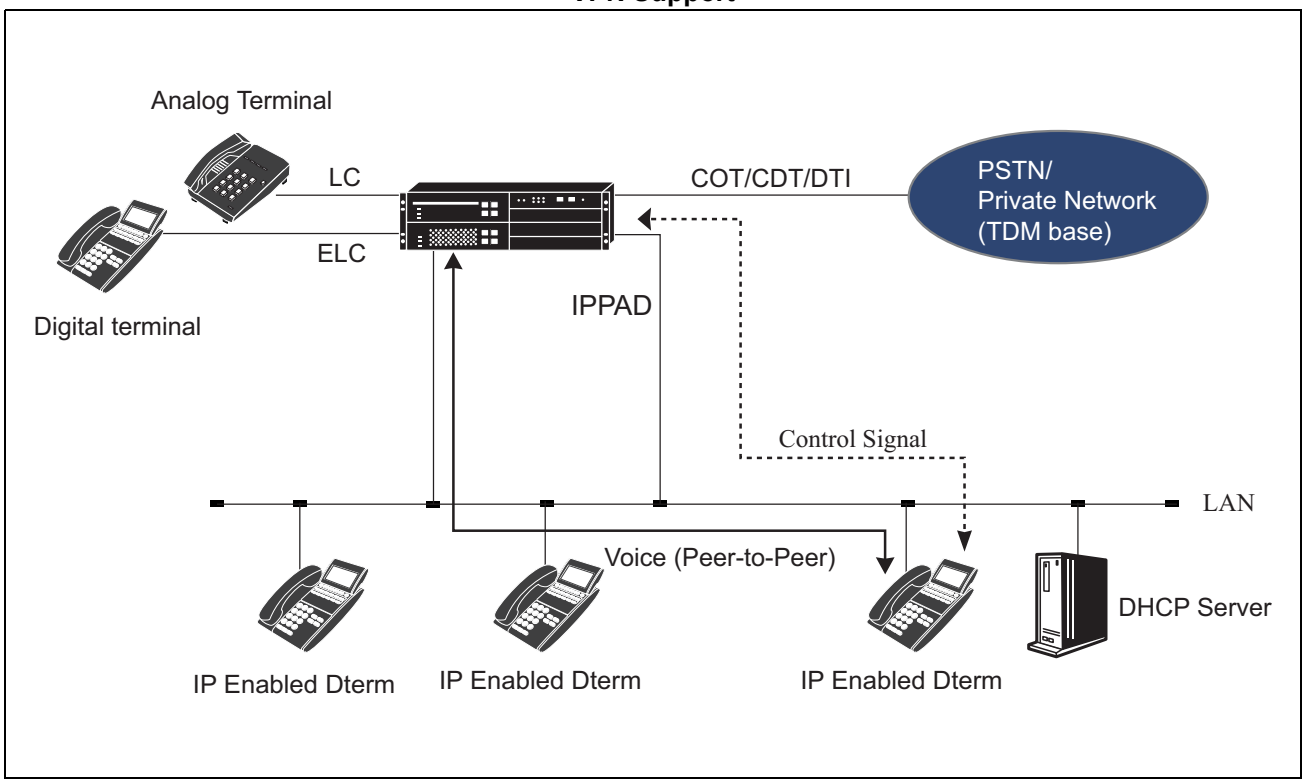
Function		Remarks
Auto Negotiation	x	
10/100 BASE support	x	
Spanning Tree	-	
Remote Maintenance	-	
Port VLAN	-	
Tag VLAN	*	SW HUB passes through the MAC frames.
MAC VLAN	-	
GARP	-	
GMRP	-	
GVRP	-	

x: available
-: not available
*: available with condition

Virtual Private Network (VPN) Support

This feature allows the Telephony Server to notify private IP Address that is assigned by VPN server to the VPN client. Thus, the IP Telephone application can communicate with Station/Trunk accommodated in the Telephony Server. In the following example, a home PC uses the global IP Address that is temporarily given by ISP. When the PC user attempts to communicate with terminals accommodated in the Telephony Server, IP packets are sent and received through VPN tunnel. However, the IP Telephone Application cannot know the private IP Address assigned to the VPN tunnel by the VPN Server. By using this feature, the Telephony Server can notify the private IP address to the IP Telephone Application, accomplishing communications between IP telephone application and terminals accommodated in the Telephony Server.

VPN Support



Note: No Data Programming is required for this feature.

[DtermIP INASET]

DtermIP INASET enables you to perform the following three application features in addition to the DtermIP features.

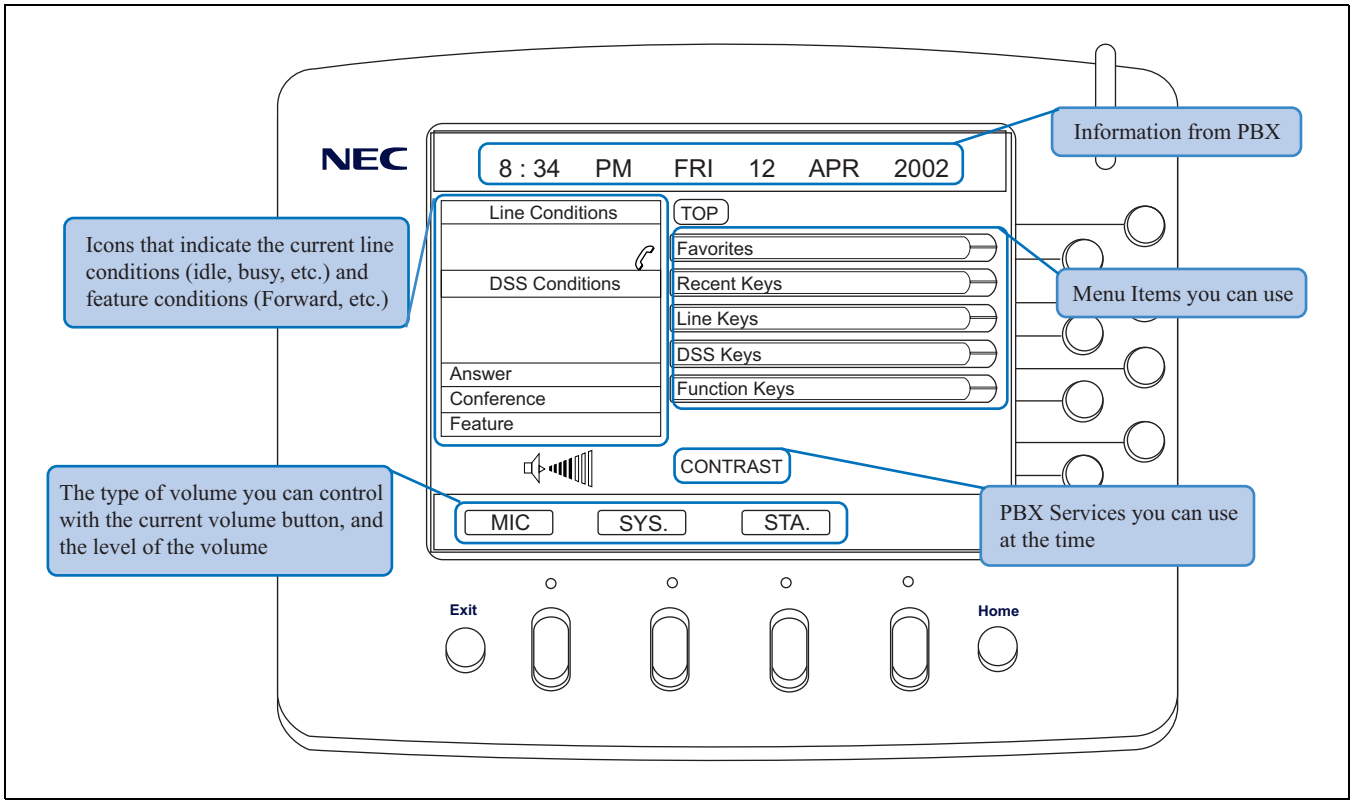
- Home Application
- Browser Application
- Directory Application

Home Application

The Home Application enables a user to perform operations of telephones, such as placing or receiving calls by much the same way as the conventional Dterm.

To give most of the same LCD displays as the conventional Dterm, the Home Application allows you to use the many functions supported by the conventional Dterm.

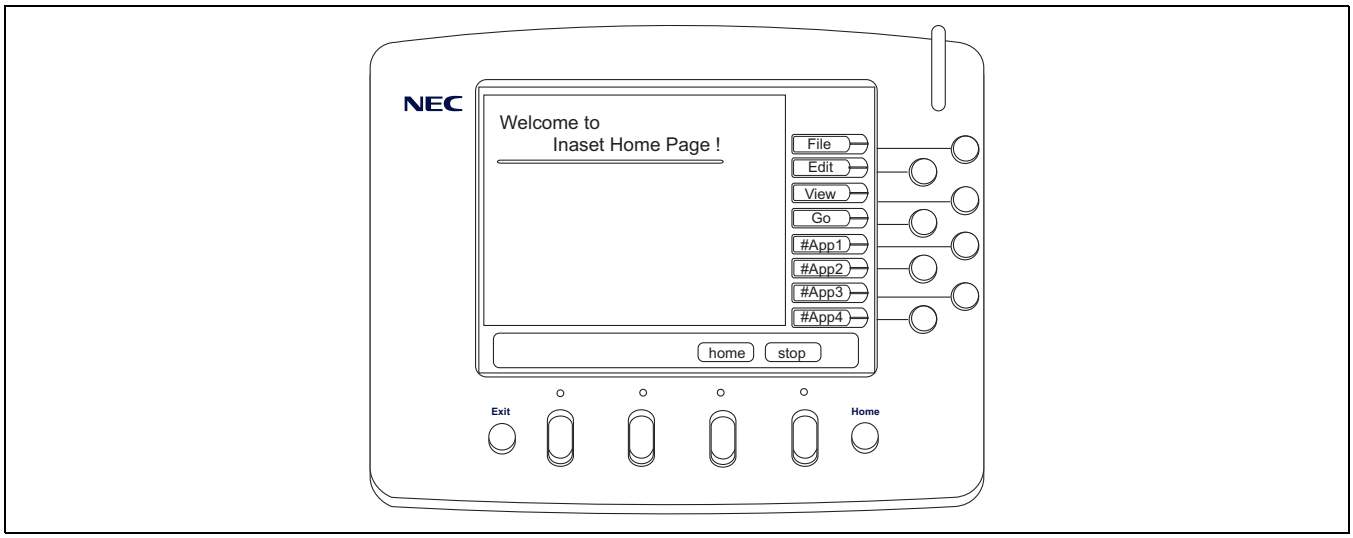
Home Application Window



Browser Application

The Browser Application enables a user to view the DtermIP INASET web page provided on the Internet. You can get various services by accessing the web page where you can find DtermIP INASET applets registered.

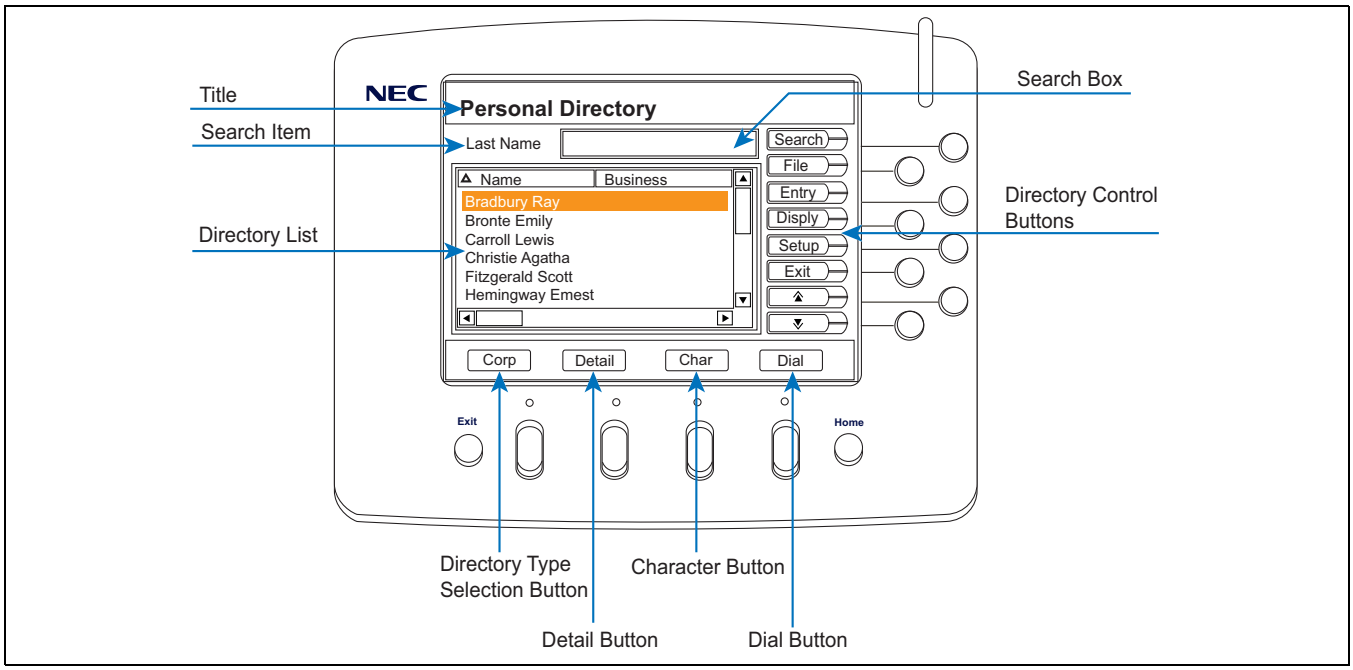
Browser Application Window



Directory Application

The Directory Application can search the registered directory data and originate a call based on the data. In addition, when a call is terminated, the Caller Name can be displayed on the screen (the Caller's Telephone Number and Name must be registered in advance).

Directory Application Window



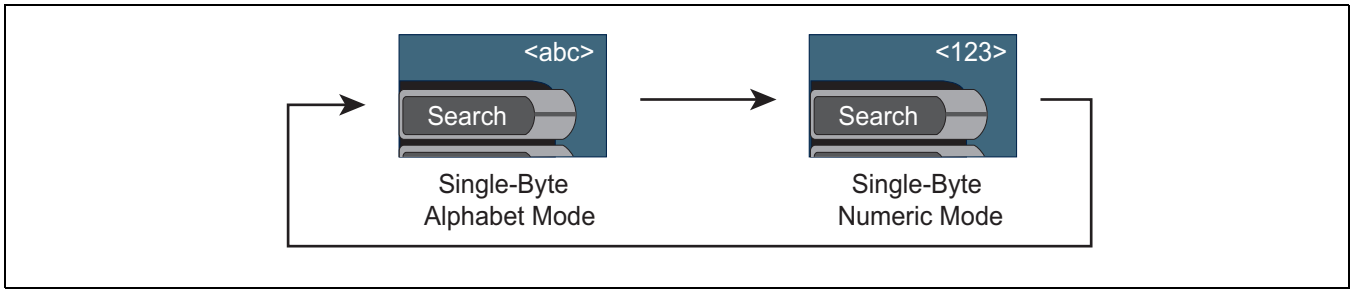
Title
The title shows a directory type. Three directory types are available: Personal, Corporate and Group.

Search Item
When you search by entering a character in a search box, you can select the search condition. The three types of search conditions are keyword, Last Name and company.

Search Box
Enter characters here. The searching Target Data predetermined by the Search Item will be searched for in the Directory List, and the cursor will move to the piece of data where a forward match occurs.

Directory List
The Directory List is a quick reference list of the registered data. Each of the three types of directory lists allow registration of up to 200 items.

Character Entry Mode
The Character Entry Mode shows the type of characters that can now be entered in the directory data search mode. The type of characters that can be used in the search mode can be selected by the Character button. The character <123> indicates single-byte numeric characters, and <abc> indicates single-byte alphabet.
Each time you press the Character Button, the character types will change in the following sequence.



Directory Control Buttons
The Directory Control Buttons are used to perform eight different types of directory control procedures.

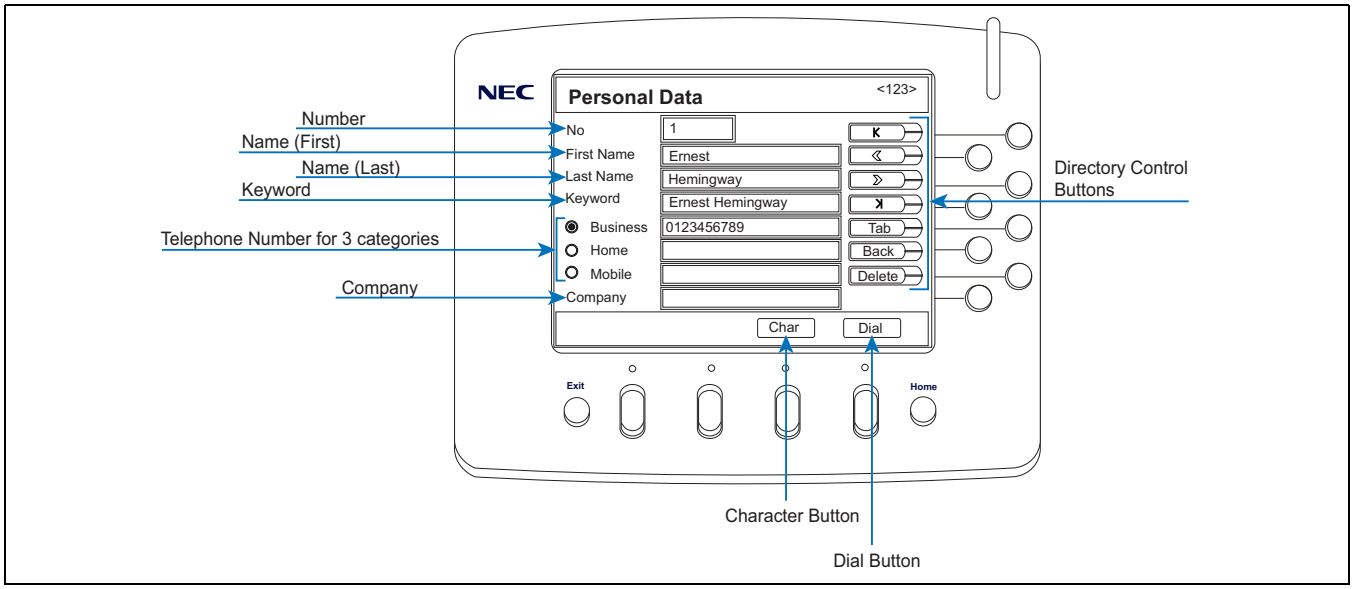
Directory Type Selector Button
Use the button to select a directory type. Each time you click the button, the directory types will change to Corporate to Group to Personal (when you start the procedure with the Personal Data List on the screen).

Detail Button
The Detail window of selected Directory Data will open. When you have selected two or more pieces of data, the details of the first piece of data will be displayed.

Character Button
The focus will move onto the Search Box, and control will enter the Directory Data Search mode. Each time you click the button, the Character Entry Mode will change to Single-Byte Numeric to Single-Byte Alphabet.

Dial Button
Click the button to place a call to the first selected piece of data on the Directory List without specifying a Line Number. When selecting a piece of Directory Data on the initial window shown above and clicking the Detail Button, the following Detail window will open.

Directory Application - Detail Window



No.
Shows a Directory Data Number. You can choose any number from 1 through 200.

Name (First)
Shows the first name of a person.

Name (Last)
Shows the last name of a person.

Keyword
Shows the keyword to use for data search.

Business, Home, Mobile
Shows registered directory numbers. You can register three directory numbers per person. You can place a call by selecting any of the radio buttons Business, Home, Mobile and then clicking the Dial button. You can change the item names Business, Home, Mobile to different names.

Company
Shows the name of the company to which a person belongs.

Directory Control Button
The Directory Control Buttons are used to perform seven different types of Directory Control Procedures for a detail window.

Add Button
Use the button to add the entered data to the Directory List. Use the button when registering a new piece of data.

OK Button
Use the button to validate the entered data. Use the button when modifying data.

Character Button
Each time you click the button the Character Entry Mode will change. As in the Directory List Window, you can choose any of the two types of characters: Single-Byte Numeric, Single-Byte Alphabet.

Dial Button
Use the button to place a call to the selected Directory Number.

[Improved UMG System Durability]

For UMG system, IP terminals can be controlled even if the System Processor (SP) is down. This is accomplished by storing registry information in both SP and CMP.

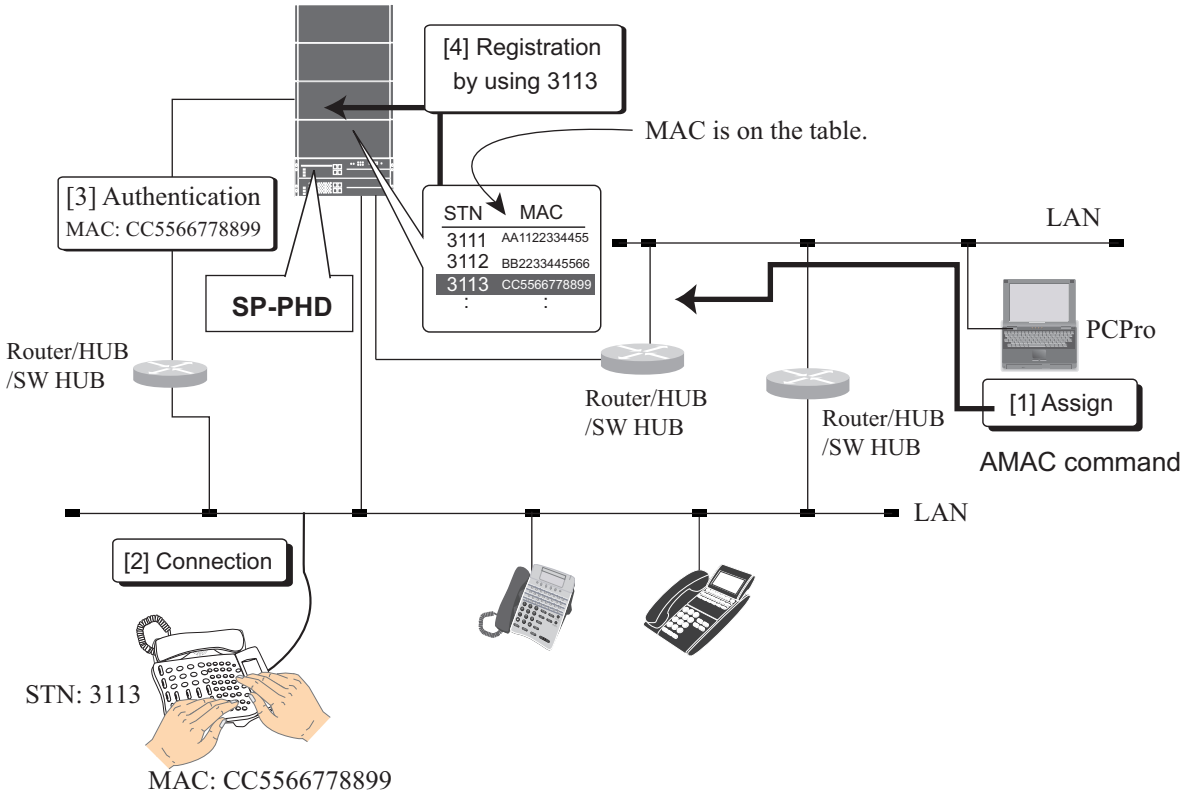
Operating Procedure

Note: Refer to the related User's Guide for the operation on each terminal.

<MAC Authentication>

This feature allows the system to perform authentication of DtermIP users by checking the MAC address of the terminal. Registration is accepted when the MAC address of the terminal has been assigned on the System. For this feature, Station Number and MAC address must be assigned in advance by using the PCPro command. By connecting the terminal to the LAN, the authorized user can log in to the system by the preassigned Station Number.

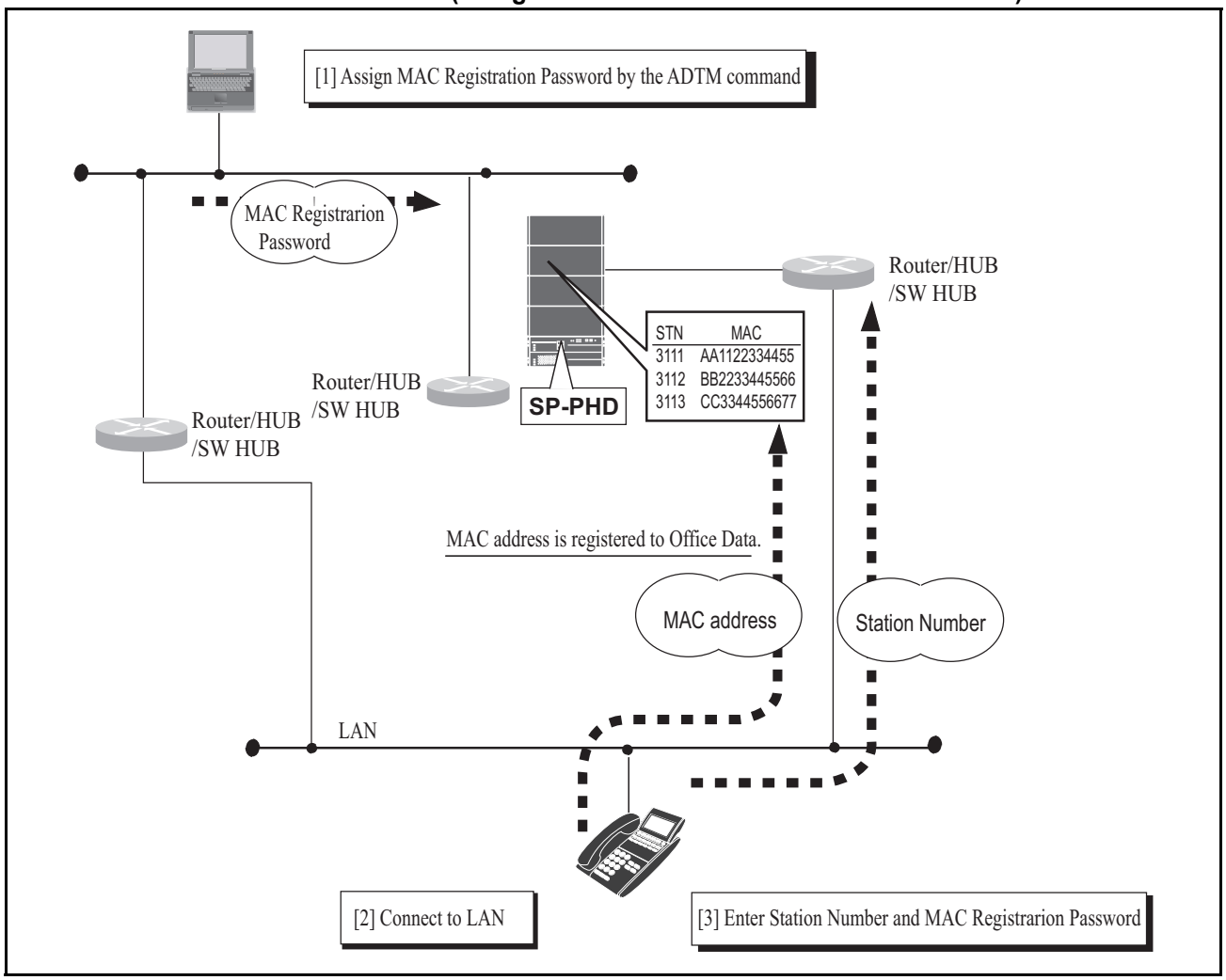
MAC Authentication (Assignment of MAC Address from PCPro)



1. Assign the MAC address of the DtermIP and the Station Number by using AMAC command.
2. Connect the DtermIP to the LAN.
3. MAC authentication is automatically performed.
4. The DtermIP is registered by the assigned Station Number.

The MAC address can also be assigned from DtermIP when initially connecting the terminal to the LAN. By entering MAC Registration Password (pre-assigned by the ADTM command) from the terminal, the MAC address is automatically registered to the MGC Office Data. In this case, you need not assign MAC address from the PCPro (AMAC command) in advance. For information on how to register MAC address onto the Telephony Server data from the terminal, refer to the Peripheral Equipment Description (Multi-Line Telephone).

MAC Authentication (Assignment of the MAC Address from DtermIP)



Note: When you register MAC address from your DtermIP, MAC Registration Password must be assigned by ADTM command in advance. Once this assignment is performed, after that you don't need to assign any more.

1 Connect the DtermIP to the LAN: The packet (including MAC address) is sent to the System.
2 Enter Station Number and MAC Registration Password (pre-assigned by the ADTM command) from DtermIP.
3 MAC address and Station Number is registered to the Office Data: IP terminal becomes available status automatically with pre-assigned Station Number.

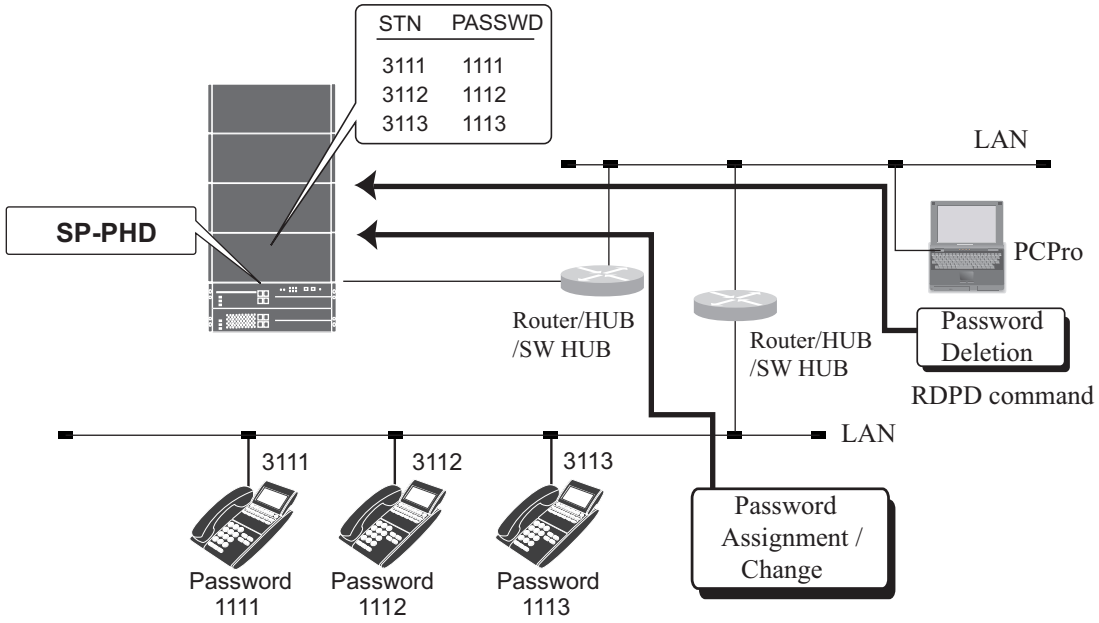
<Login>

This feature allows a user to log in to the system by using password stored in the Telephony Server memory. The password is assigned or changed from DtermIPs. To delete password PCPro is used.

Note: The password used by this feature is not for the line to be used, but for the terminal itself.

DtermIP Login Mode

Passwords set from DtermIPs are stored in the Telephony Server memory.



Service Conditions

1. SP-PHD can handle a maximum of 4000 DtermIPs.
2. Station Number can be composed of a maximum of 16-digit numerics including "*", "#".
3. DtermIP cannot be accommodated in Group 24 or larger Group.
4. DtermIP controlled by SP-PHD can be used on any point of LAN (Intranet). However, restrictions related to LENS is also applied to DtermIP as Dterm. (i.e. Multi Line)
5. Be sure to initialize the SP-PHD after assigning the IP data by using the ALIDL command
6. DtermIP can be allocated to the LENS (physical LENS) of vacant slot on which the Station/Trunk is not mounted. (IP Station cannot be assigned to the already registered Station in duplicate.) IP Stations can also be allocated to the virtual-LENS in addition to the physical LENS.

7. DtermIPs with duplicate Station Number cannot operate in a system. When attempting "Registration" (**Note 1**) using the Station Number that is already used, "Override?" indication appears on the overriding DtermIP LCD, and then the user can determine whether to override the existing Station or not (use the different Station Number). The overridden DtermIP becomes inoperative.
"Override" function can be performed under the following conditions.

   - This function is available between IP terminals.
   - DtermIP cannot use the Station Number that is already used by other IP Stations connected to the Analog MC.
   - Stations connected to the Analog MC cannot use this function.

   Note 1: "Registration" is the function that supports the establishment of control signal (IP Packet) path between DtermIP and the System, the information such as IP address of each device must be notified each other.

8. The default number of UDP port used for Registration by MGC is "3456." When changing this port number by the ADTM command, the existing data settings of DtermIP must be modified.

9. By installing a Registration Proxy server on the network having multiple System(s), the Registration Proxy server can handle the Registration Requests from DtermIPs on the network, as a substitute for the System(s).

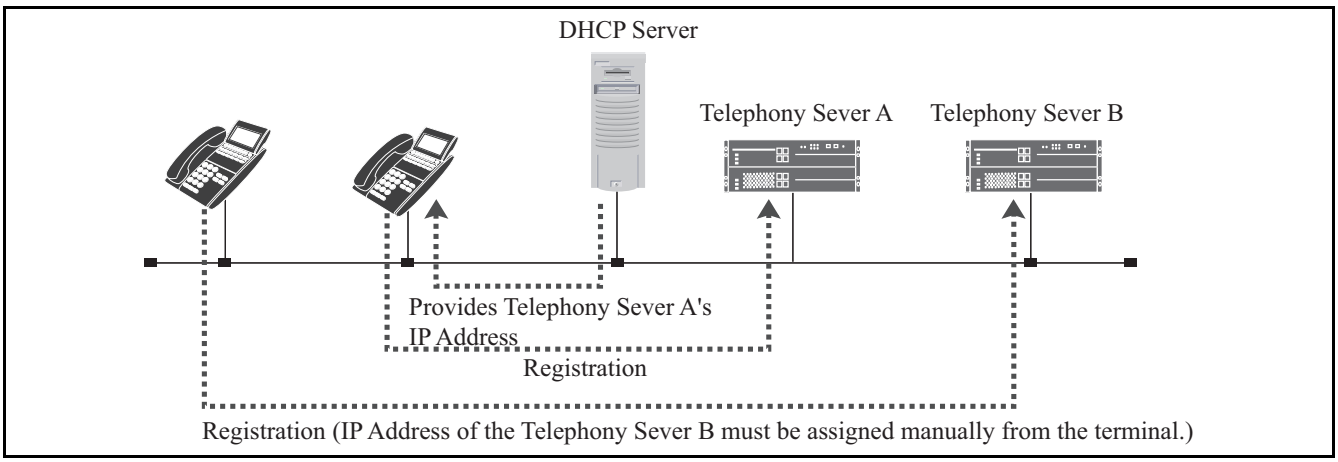
10. When SP-PHD is reset while in operation, MGC sends Software Reset Packet to IP terminals connected to the SP-PHD. At this time, the terminals that are reset can re-perform Registration to the MGC which would hunt the other SP-PHD available for the terminals.

11. When the System is initialized, all the IP terminals must re-perform Registration.

12. When communication between IP terminals and PH (Protocol Handler) is released while in operation, communication failure notification is provided under the following conditions.

    - When the communication failure is detected by PH side, the System stores the fault information received from PH for a certain period of time, and then notifies the fault information to PCPro if required.
    - When the communication failure is detected by IP terminals, the fault notification packet is sent to the Telephony Server from IP terminals, and then the Telephony Server stores the information received from IP terminals for a certain period of time and notifies the fault information to PCPro if required.

13. When multiple Telephony Server(s) exist on the same LAN, the DtermIP can select a target Telephony Server of the registration per a terminal basis, by setting the each IP address of the target System.
For System having DHCP server: For IP addresses of the target Telephony Server(s) not to be assigned at the DHCP Server, they must be assigned manually from each DtermIP.



14. IP Stations can also be allocated to the virtual-LENS in addition to the physical LENS. The virtual-LENS requires no Physical PIM (Port Interface Module). Consequently, the system can accommodate a maximum number of the IP Stations (*For 1-IMG/4-IMG system: Max. 4,000 IP Stations. For UMGi system: Max. 16,000 IP Stations.*) with a minimum of hardware configuration. For more details, refer to the Virtual PIR Data section in Chapter 2 of this manual.

15. The Health Check function can be provided for the IP terminals (such as DtermIP, DtermIP INASET, Soft Phone, etc.) controlled by the SP-PHD. By providing the Health Check, the system can monitor the state of the IP terminals in real-time and figure out the cause of failure immediately. The cycle of periodic Health Check can be determined by the System Data, ASYDL, SYS1, Index 959 (see the Programming section).

16. If the data of your IP terminal is changed by AKYD command while your line is busy, the new data is passed to your terminal when your line is released.

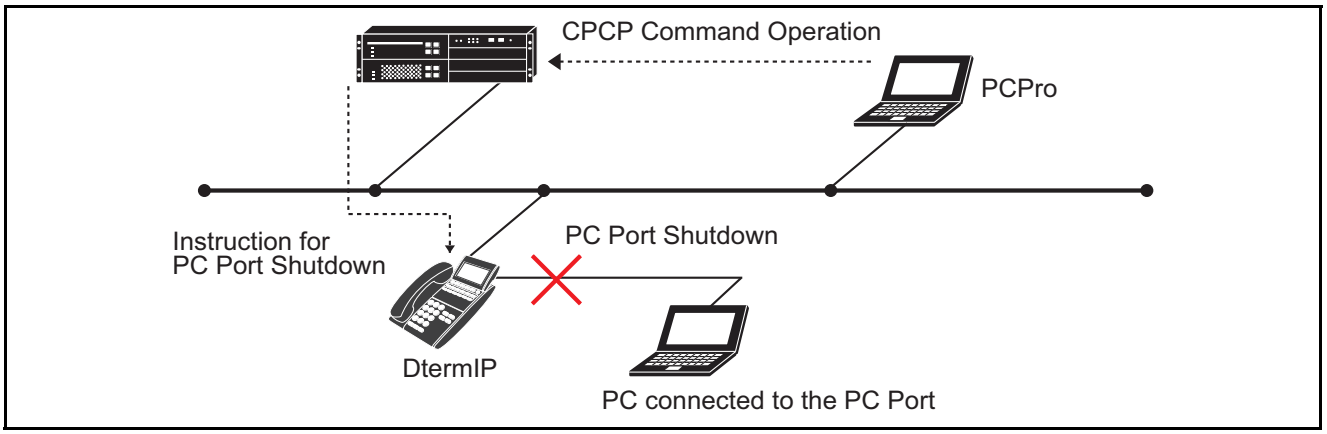
17. If you change the data of AKYD/ADKS command, re-login operation (Login and Password) is not necessary.

18. First, Registration Request is sent to the same network. If the available internal PH/SP-PH are not found (or can not be connected), it will be transferred to another network. In the case of Analog MC, it will be registered to internal PHE.

19. The maintenance personnel can disconnect/connect the PC port of DtermIP from/to the network by using the CPCP command. For example, this feature allows the maintenance personnel to quickly disconnect the PC (that is connected the PC port of DtermIP) from the network when virus is detected on the network. For details, see "IP Device PC Port Remote Control" in Operations and Maintenance Guide.

Note: Available firmware version is as follows:
Dterm Series i: Ver. 3.06 or later
DtermIP (Proprietary Protocol): Ver. 3.08 or later

Note: Condition only for DtermIP (Proprietary Protocol):
PC Port setting of this feature is canceled when DtermIPs (Proprietary Protocol) perform Registration. Therefore, DtermIPs (Proprietary Protocol) needs to be placed in "PC Port shutdown" status (by the CPCP command) after each Registration process.



20. When an IP terminal is in a make busy state with the MBST command, if there is a network failure between the Telephony Server and the IP terminal, even if you try to put the IP terminal in a make idle state with the MBST command after the network is restored, the IP terminal continues being in a make busy state. To disable the make busy state, restart the IP terminal by turning manually OFF then ON again.

<Conditions for MAC Authentication mode>

1. You can use Mac Authentication mode with a DtermIP that has a MAC address.

2. MAC address data is assigned per DtermIP by the AMAC command. MAC address data can also be assigned from each DtermIP.

3. A single Station Number can be shared among multiple MAC addresses. However, the DtermIP that came on-line last, is the terminal that will be operative. Multiple DtermIP Stations assigned to the same Station Number cannot be simultaneously used.

4. When DtermIPs to which no AMAC command data is assigned are connected to the LAN, the DtermIPs start up in Login Mode.

5. The following conditions must be considered when MAC address data is assigned from DtermIP.
   a.) MAC Registration Password is pre-assigned by the ADTM command.
   b.) When you assign MAC address from DtermIP, you can use Physical Station Number (STN) only. Telephone Number (TELN) cannot be used as your Station Number. When you want to allocate a Telephone Number with your MAC address, use the AMAC command. You can use either Physical Station Number (STN) or Telephone Number (TELN) when assigning Station Number through this feature. In a FCCS network, you cannot use Telephone Numbers allocated to Stations of the nodes other than the node to which your DtermIP registers.
   c.) If you would delete a MAC address that has already assigned, use the AMAC command. Enter the MAC Address and then press the DELETE key.
   d.) By pressing the LOGOUT Feature key, DtermIP operating in MAC Authentication mode can be forcibly changed to “Login Mode” and requests the user to enter “Login Code” and “Password.” At this time, when the user does not enter the Login code (Station Number) and password within 10 minutes, the DtermIP is automatically returned to the MAC Authentication mode.
   e.) Confirmation of Station Number, MAC address and date and hour that is registered could be performed by ADTM command (Option - Retrieve Information - Log Information). To save Log Information on PCPro, you should select “Save to disc"”.
   f.) A single Station Number can be shared among multiple MAC addresses, but only one IP terminal

is available at the same time. If terminal (A) connects to the LAN when terminal (B) is already connected to the LAN with the same Station Number as terminal (A), terminal (A) would be displayed “Override?” If user of terminal (A) presses the “Yes” key, terminal (A) becomes available status, and terminal (B) becomes not available status. Terminal (B) would display the message “Override” in 90 seconds.

g.) If the DtermIP MAC Address Registration Password (assigned in the ADTM command), is the same as the Password used by a terminal for Login Mode, the Login Mode will take precedence and the station cannot use this feature.

h.) If you attempt to register an incorrect Station Number, you couldn't register MAC address to the Office Data. Your DtermIP would display the message “Unauthorized”.

i.) When the number of digits for the Station Number is greater than the number of digits for the password, MAC address needs to be assigned by the AMAC command.

6. If you change the following data settings, delete the AMAC command data for the corresponding Station Numbers.

   - IP station data (AISTL data) for stations whose Physical Station Number is associated with AMAC is deleted by AISTL command.
   - Telephone Number data (ALGSL/ALGSN data) for stations whose Telephone Number is associated with AMAC is deleted.
   - Physical Station Numbers or Telephone Numbers associated with AMAC are reassigned to stations other than DtermIPs.
   - Telephone Numbers associated with AMAC are reassigned to stations of the nodes other than the node having the AMAC data.

   Note: You need to delete the MAC address data assigned from DtermIP as in the case of AMAC command data.

7. When Physical Station Number Calling Restriction [P-87] is activated by the system data (ASYDL/ASYDN, SYS1, Index 1144, Bit 0=1), only login with a Telephone Number is available.

8. Do not use Mac Authentication mode with a Soft Phone.

<Conditions for Login mode>

1. Login can be restricted per a Service Feature Restriction Class (SFC) basis by assigning "1" to SFI186 (IP Enabled Dterm Login).
2. This feature is available for DtermIPs.
3. If terminals other than DtermIP attempt this feature (dials "DtermIP Password Entry" Access Code):
   Analog Telephone: Receive ROT
   Dterm: Receive ROT with "RST" indication on the LCD
4. When attempting Login for the first time, use the Station Number as the password. Then the password can be newly assigned/modified by dialing "Password Entry" Access Code (refer to "New Password Assignment" in Peripheral Equipment Description (Multi-Line Telephone)).

   Note: If an arbitrary number is assigned as Login Code by the ALGCL/ALGCN command, enter registered Login Code instead of Station Number for a password.

5. The number of digits that can be used as a password is 4 through 10. The maximum number of digits is determined by the AMND command.
6. Password can be deleted from the Telephony Server memory only by using the RDPD command.
7. Password is not displayed or listed up on the PCPro display.
8. Backup copies of assigned password is created on the hard disk by using the MEM_HDD command Data Type: Call Forwarding from PCPro or by routine diagnosis.
9. While setting or changing password "6-sec timer" is automatically set for the 1st digit. If the timer expires, the password setting becomes NG with "ERROR" indication on the LCD with BT.
10. While setting or changing password, if the inter-digit timer expires, the password setting becomes NG with "ERROR" indication on the LCD with BT.
11. Available digits for password are: 0 - 9, # and *.
12. Password set from the DtermIP is not displayed on the LCD. (The LCD indicates the input password as "*****".)
13. When the system is initialized without loading the Office Data, the passwords before the System Initialization are effective.
14. When the system is initialized with loading the Office Data, the passwords stored on the hard disk become effective. If passwords are changed in between "Office Data Backup Completion" and "System Initialization with loading the Office Data," the previous passwords become effective.
15. In order to keep the password (the password entered by the DtermIP) secret, the encrypted password will be sent to the System from the DtermIP (operating in Login mode). The password to be sent to the System is encrypted with MD5 algorithm as default. The algorithm can be changed to Proprietary Protocol by assigning the System Data (ASYDL, SYS1, Index 848, Bit 7).
16. When Physical Station Number Calling Restriction [P-87] is activated by the system data (ASYDL/ASYDN, SYS1, Index 1144, Bit 0=1), only login with a Telephone Number is available.

<Conditions for DtermIP INASET>

1. This feature provides Peer-to-Peer Connection.
2. A Tag VLAN-capable Two-port Switching HUB is embedded in the DtermIP INASET.
3. Available Voice Compression Types for DtermIP INASET are G.711, G.729a and G.723.1.
4. DtermIP INASET can be powered through the AC Adapter, Local Power Supply and the Power Feeding Patch Panel (SN1604PWEMS), or Center Power Supply.
5. Each of three types of directory lists (Personal, Corporate and Group), allows registration of up to 200 Telephone Numbers. The Directory Lists can be created in Comma Separated Value (CSV) format on the FTP Server. This CSV file can be shared between more than one DtermIP INASET Operator by downloading the file from each Terminal.
6. All Digital terminal features except [I-21] INTERNAL ZONE PAGING are available for DtermIP INASET.
7. Using DtermIP INASET in MAC Authentication Mode is recommended.

<Conditions for Improved UMGi/RNIP System Durability>

1. While the CCN is down, the system does not accept any registration. When an IP terminal logs out, it cannot log in again while the CCN is down.
2. IP terminals that have been registered before the CCN failure can continue to be serviceable.
3. If any SPEED CALLING - ONE TOUCH -D [S-26D] data is set after a RN changes to survival mode, the data will be used. However, when the RN reverts normal mode, also the data will revert to its original settings in normal mode.

Note: For the conditions other than above, refer to UMGi Configuration and Maintenance Guide.

<Conditions for Improved UMG System Durability>

1. While SP is down, the system does not accept any registration. When an IP terminal logs out, it cannot log in again while the SP is down.
2. IP terminals that have been registered before SP failure can continue to be serviceable.
3. If any SPEED CALLING - ONE TOUCH data has been set during SP failure, the data can be used temporarily. When the terminal performs registration again, however, the data will return to the original data.
4. When the failed SP recovers, it does not accept any registration until it completes loading registry information from the CMP. The loading takes a few seconds. On completion of the loading, registration will be resumed.

Interactions

The passwords used for this feature are shared with [P-57] PAD LOCK. The following shows conditions related to PAD LOCK.

1. When PAD Lock ID Code has been assigned before performing password entry for this feature, use the already assigned password for this feature.
2. After performing PAD Lock ID Code assignment, if the password is changed by this feature, the changed password takes effective.
3. After performing password entry by this feature, if PAD Lock ID code is set, the assigned password becomes effective.

4. After canceling PAD Lock ID Code, the PAD Lock becomes ineffective. However, the password is valid.
5. The following shows the relationship between memory status and the commands.

command / memory status	DtermIP Login Function			PAD LOCK		
	Assign/ Change	Delete		Assign/ Change	Delete	
		DtermIP	PCPro		DtermIP	PCPro
Not Assigned	PASSWD only assigned	-	NG	PASSWD + RSC + SFC	receives BT	NG
PASSWD	PASSWD changed	-	all deleted	PASSWD + RSC + SFC	receives BT	all deleted
PASSWD + RSC + SFC	PASSWD only changed	-	all deleted	PASSWD + RSC + SFC	RSC + SFC only deleted	all deleted

Programming

This section describes the necessary data assignments for MAC Authentication mode and Login/Logout mode. The following must be assigned in addition to the basic Station data, the basic Desktop terminal data, and the basic IP Station data. For the assignment of Station Accommodation Data for the DtermIP, AISTL command (for IP Station) must be used instead of ASDT command (for the conventional Station) command.

- For the basic Station data, see the Assignment of Station Data section in Chapter 2.
- For the basic Desktop terminal data, see the Assignment of Desktop Terminal Station Data section in Chapter 2.
- For the basic IP Station data, see the IP Station Data section in Chapter 2.

Note: For the assignment of the Configuration data, refer to Peripheral Equipment Description (IP Devices).

<Data Assignment - MAC Authentication>

STEP 1: **ASYDL** - SYS1, Index 811: Tone Code by countries. For more details, see Command Manual.

STEP 2: **AMAC** - Assign MAC address associated with each Station Number assigned in AISTL command.

MAC ADDRESS:Enter the fixed MAC Address of each DtermIP terminal in Hexadecimal notation.
TYPE: Select either “STN” or “TELN”.
When TYPE=STN is selected:
TN: Tenant Number
STN: Enter Station Number associated with MAC address assigned in the MAC ADDRESS parameter.
When TYPE=TLEN is selected:
UGN: User Group Number
TELN: Enter Telephone Number associated with MAC address assigned in the MAC ADDRESS parameter.

Note: If this data is assigned, the DtermIP operates in MAC Authentication mode. If not, it operates in Login mode. This data is not required when MAC address is assigned from each terminal.

STEP 3: **ADTM** - Assign MAC Registration Password when you assign MAC address from DtermIP.

Option -DtermIP MAC Address Registration Password
Password: Enter the Password to assign MAC Registration Password.
Confirmation input: Enter the Password again.

<Data Assignment - Login/Logout>

STEP 1: **ASYD** - SYS1, Index 93, Bit 6. Are the AMND tables Separate or Common for tenant in the system. Assign data "0" (= Separate).

STEP 2: **ASYDL** - SYS1, Index 848, Bit 0 to Bit 3 Output the system message (36-E) when the password entered from the DtermIP is invalid. **Note 1**

0: System message is not output.
1-15: The System will generate a system message if login failed consecutively and the number of failure read has to this.

Note 1: For SIP Multiple Line terminal, the system message (36-E) is not output even though a wrong password is entered in the terminal.

STEP 3: Assign the following data related to the Registration Password Encryption function, by using either ASYDL or ADTM command.

(In the case of ASYDL command)
SYS1, Index 848, Bit 7=0 (Registration Password Encryption Function (0 = MD5))

(In the case of ADTM command)
Password Encryption: Select either of the algorithms for Password Encryption

- MD5 (default)
- Proprietary (NEC Standard Procedure)

STEP 4: **ANPD** - Reserve a Number Level for DtermIP Password Entry.

TN: Tenant Number
1stDC: 1st Digit Code
CI: Select Normal Service from the combo box for Connection Indexes.
NND: Number of Necessary Digits
BLF: 0 (Out of Service)

STEP 5: **ASPA** - Assign the Access Code for DtermIP password entry.

TN: Tenant Number
ACC: Access Code
CI: "N: Normal"
SRV: "SSCA: Service Code Appendix", SIDA = 123 (DtermIP Password Entry)

STEP 6: **ASFC** - Restrict DtermIP LOGIN.

SFC: Service Feature Restriction Class
SFI: 186 (DtermIP LOGIN)
RES: 0 (Allowed)/1 (Restricted)

STEP 7: **AMND** - Assign the MND data for the passwords.

TN: 0
DC: Destination Code (Assign the 1st digit of all passwords.)
MND: Maximum Necessary Digits (Assign the MND of passwords.)

STEP 8: **AKYD** - Assign the key data of DtermIP.

FKY 142 = LOGOUT (for DtermIP)

STEP 9: **ADSL/ADKS** - Assign the Software key data of DtermIP.

FKY 142 = LOGOUT (for DtermIP)

D-162 Distinctive Ringing List

(Series 8800 software feature)

General Description

This feature allows a Station the ability to set 16 individual CPNs in bins, up to 16 digits each. These numbers will then ring the station with a Distinctive Ringing Pattern.

Note: This feature is available in North America only.

Operating Procedure

To set a Distinctive Ringing List entry by Bin Number

1. Lift the handset and dial the Distinctive Ringing List (DRL) Feature Set Access Code.
2. Receive Stutter Dial Tone.
3. Dial the individual Bin Number 00-15.
4. Dial the exact digits to distinctively ring (a maximum of 16 digits).
5. Wait for Service Set Tone.
6. Replace the handset.

To remove a Distinctive Ringing List entry by Bin Number

1. Lift the handset and dial the Distinctive Ringing List (DRL) SET Code.
2. Dial the individual Bin Number 00-15.
3. Wait for Service Set Tone.
4. Replace the handset.

Service Conditions

1. This feature can only be set/canceled by an individual station using the appropriate Access Code.
2. The PCPro command DDRL will only display the list, it cannot modify it.
3. Individual Call Block can only be used if the incoming Station/Trunk supplies a Calling Party Identification Number.
4. When the System is initialized with data memory load, the Distinctive Ringing List (DRL) feature will be in the last state saved to the System hard drive.
5. This feature is only available for stations with Service Feature Restrictions Released (SFI 191 = 1).
6. This feature is not available for BRI or Wireless Terminals even if Individual Call Block Service Feature Restriction Class SFI 165 is allowed.
7. This feature can be used with any trunk or station supplying a valid Calling Party Number Identification to the System (Caller ID/ISDN or MF).
8. This feature is not supported in a Hotel/Motel system.
9. ASYD, SYS1, Index 160, Bit 0 and Bit 1 must=0.

Programming

STEP 1: **ASFC** - Assign a Service Feature Restriction Class to allow DR LIST for stations. SFI 191 RES = 1 to release restriction.

STEP 2: **ANPD/ANPDL/ANPDN** - Assign a number level for Individual Call Block.

STEP 3: **ASPA/ASPAL/ASPAN** - Assign the following Access Codes for CI = “N: Normal”
Distinctive Ringing List - Set/Cancel - SRV = “SSCA: Service Code Appendix”, SIDA = 133.

To delete

The deletion procedure is the reverse order of assigning the feature.

D-163 Distinctive Ringing for Toll Calls

(Series 8500 software feature)

General Description

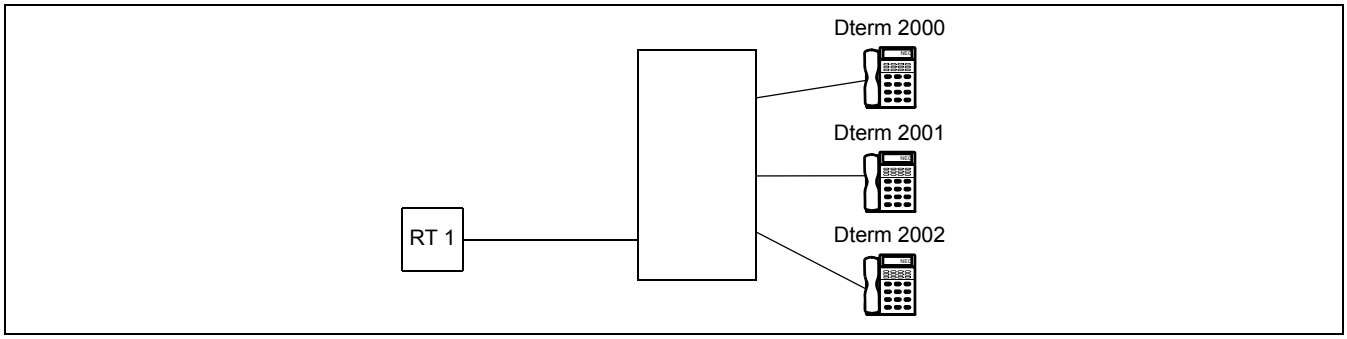
This feature allows Incoming Calls from outside of a defined Local Area to ring with Ringer Pattern 5, while Incoming Calls from the defined Local Area will ring with Ringer Pattern 1.

Note: This feature is only available in North America.

Operating Procedure

There will be no change in the Operating Procedure for the user.
Example of operation:

1. The ALCAL command has the area codes 972 and 214 defined as local.
2. Station 2000 is assigned to SFC 1 and SFC 1 has new SFI = 1.
3. RT 1 is a PRI with CPN data.
4. ASYD, SYS3, Index 1 is set to 42 (4 on/2 off) (Ringer Pattern 1).
5. ASYD, SYS3, Index 7 is set to 73 (7 on/3 off) (Ringer Pattern 5).



6. Station 2000 receives a call from CPN 972-555-1234, the call rings with Ringer Pattern 1.
7. Station 2000 receives a call from CPN 316-555-5678, the call rings with Ringer Pattern 5.

Service Conditions

1. This service is only active for Incoming Calls on PRI, Feature Group DMF, and Analog Caller ID, and Trunks when at least ten digits of CPN are provided.
2. If no CPN is provided, or not enough digits are provided, the call will ring as it normally would have.
3. This feature is mutually exclusive with Personal Ringer Service (SFI 148). If both features are on, Personal Ringer Service will have priority.
4. This feature cannot be used with Distinctive Ring on Phantom Extensions.
5. Call Back, Recall and other features may also use Ringer Pattern 5.
6. After the Ringer Timers are changed in system data 3, the line package must be reset before the new Ringer Timers will go in to effect.
7. A maximum of 16 Area Codes per TN can be assigned as Local in ALCAL.

8. This service is available on Incoming Calls that are Tandem from CCIS, as long as the requirements in #3 are met. CI in ARTD of the CCIS Route must be set to receive more than eight digits on CCIS Tandem Calls.
9. To split this command data by TN, use ASYD, SYS1, Index 93, Bit 0.
10. To split this command data by Day/Night, use ASYD, SYS1, Index 75, Bit 1.
11. This feature is not supported in a Hotel/Motel system.
12. ASYD, SYS1, Index 160, Bit 0 and Bit 1 must=0 for this feature to operate.

Programming

To assign

STEP 1: **ASYD** - Assignment of System Data. SYS3, Index 1 must be set for Ringer Pattern 1. SYS3, Index 7 must be set for Ringer Pattern 5.

STEP 2: **ASFC** - Assignment of Service Feature Restriction Class.

TN: X
SFC: X
SFI: SFI 190 (Distinctive Ringing for Toll Calls) must be set to "1" for this service to be in effect.

STEP 3: **ALCAL** - Assignment of Local Calling Area for LDM.

DN: Day/Night
TN: Tenant
Local Area Codes: 0-15
Assign Local Codes up to 16 entries.

To delete

STEP 1: **ALCAL** - Assignment of Local Calling Area for LDM.

DN: Day/Night
TN: Tenant
Local Area Codes: 0-15
Delete this data.

STEP 2: **ASFC** - Assignment of Service Feature Restriction Class.

TN: X
SFC: X
SFI: SFI 190 (Distinctive Ringing for Toll Calls) must be set to "0" for this service to be removed.

STEP 3: **ASYD** - Assignment of System Data. SYS3, Index 1 must be set for Ringer Pattern 1. SYS3, Index 7 must be set for Ringer Pattern 5. Set to Normal data.

D-164D Day-Night Mode Status -D

General Description

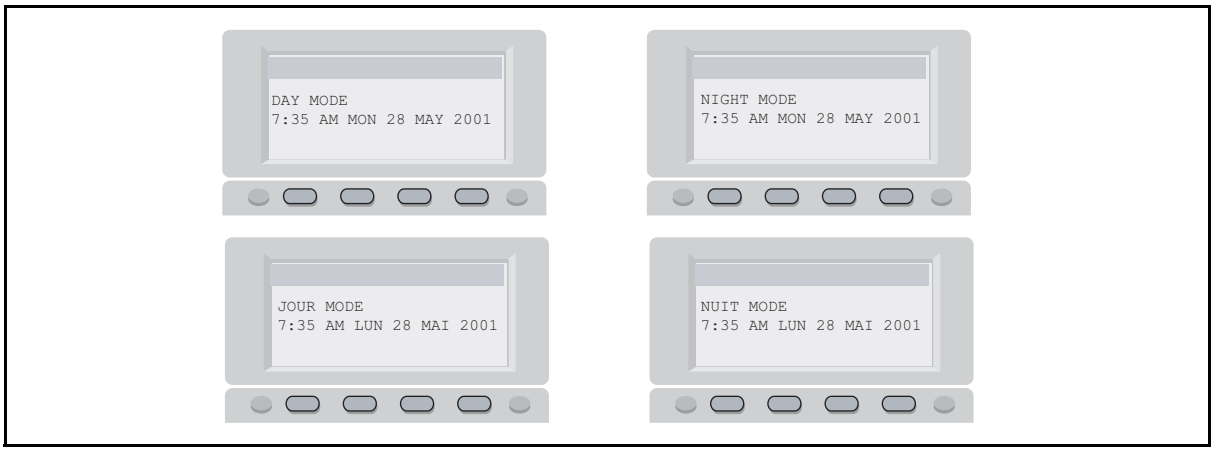
This feature allows a Desktop terminal the ability to display the current System status and change the System into Day or Night using either the "*" or "#" key.

Note: This feature is available only in North America.

Operating Procedure

Day/Night Mode Status Display on LCD of Desktop Terminal
(Not effected by ASFC)

1. Lift the handset or press the SPEAKER key. Receive Dial Tone.
2. Press the DAY/NIGHT MODE Feature key. Receive the Service Set Tone.
3. Display the current Day/Night status of the System.
4. Place handset in cradle or press the SPEAKER key.



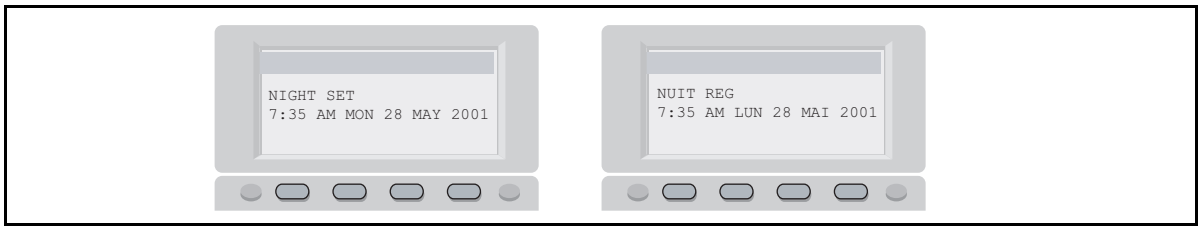
Day/Night Mode Status Display on LED of DAY/NIGHT MODE Feature key
(Not effected by ASFC/ASFCN)

The LED lamp status of DAY/NIGHT MODE Feature key on Desktop terminal indicates the day/night mode status as follows.

LED Status	Day/Night Mode Status
Turned OFF	Day Mode
Lights red	Night Mode

Change from Day mode to Night mode from Desktop terminal
(Effected by ASFC)

1. Lift the handset or press the SPEAKER key. Receive Dial Tone.
2. Press the DAY/NIGHT MODE Feature key. Receive burst Special Dial Tone.
3. Display the current Day/Night status of the System.
4. Dial "#" to set Night mode. Receive the Service Set Tone.
5. Place handset in cradle or press the SPEAKER key.



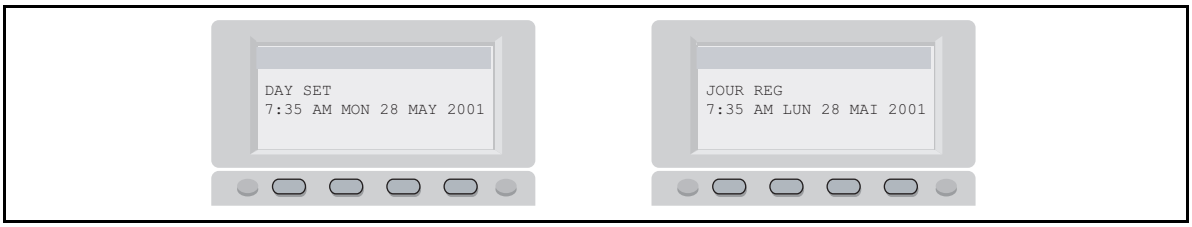
Change from Night mode to Day mode from Desktop terminal
(Effected by ASFC)

1. Lift the handset or press the SPEAKER key. Receive Dial Tone.
2. Press the DAY/NIGHT MODE Feature key. Receive burst Special Dial Tone.
3. Display the current Day/Night status of the System.
4. Dial "*" to set Day mode. Receive the Service Set Tone.
5. Place handset in cradle or press the SPEAKER key.



This feature is restricted or not available

1. Lift the handset or press the SPEAKER key. Receive Dial Tone.
2. Press the DAY/NIGHT MODE Feature key. Receive the Reorder Tone.



Service Conditions

1. This feature is available for the following Desktop terminals.
   - Digital terminal
   - IP terminal (Proprietary Protocol/SIP)
   - Soft Phone (Proprietary Protocol/SIP)
2. This feature is controlled by Service Feature Restriction Class and available by assigning the DAY/NIGHT MODE Feature key (FKY = 153) and Service Feature Index 214 unrestricted (SFI 214 = 1).

   Note: The day/night mode status display on the LCD of Desktop terminal is available when "0" is assigned to SFI 214 as well.

3. Any Desktop terminal Line appearance may be used to perform the day/night status change. However, only the Tenant of the Desktop terminals My Line will be effected. (For example, a Tenant 3 Line appearance on a Tenant 1 Desktop terminal can not change day/night mode of Tenant 3, only Tenant 1).
4. The DAY/NIGHT MODE Feature key LED will never light during display, setting or completion of this feature, therefore the Desktop terminal key LED does not indicate the DAY or NIGHT mode status.
5. This feature will not update the Night Status Lamp on the ATTCON/DESKCON.
6. ATTCON/DESKCON PCB is provided:
   a.) TAS service is available as per system programming.
   b.) If the DESKCON circuit card is reset, the day/night mode will be determined by switch settings on the PCB (For details, refer to the Circuit Card Description).
   c.) In case of a System system reset, the day/night mode will be determined by the switch settings on the DESKCON PCB (For details, refer to the Circuit Card Description).
   d.) When a ATTCON terminal is provided and the ATTCON circuit card is reset the day/night condition of the System will be determined by the position of the ATTCON terminal Night Key.
7. DESKCON(s) or ATTCON(s) terminal provided:
   a.) In the case of a night to day status change via this feature, the Deskcon(s) will not have incoming call indication when the night LED is illuminated on the terminal(s).
8. ATTCON or DESKCON PCB is not provided:
   a.) TAS service is not available.
   b.) In the event of system initialization the Telephony Server will be in Night mode.
9. For changing of DAY-NIGHT MODE STATUS -D Special Dial Tone is fixed to burst tone.
10. This feature can control the day/night mode of the same tenant in remote nodes as specified in programming for a FCCS network.
11. This feature can control the day/night mode of remote nodes as specified in programming for a CCIS network.
12. This feature setting is not available in consultation hold condition.
13. In the case of DAY-NIGHT MODE STATUS -D [D-164D] in a non-controlling node of a FCCS network, the local System can change day/night status. However if the network is controlled by a master day/night node the local System will change day/night status as per FCCS programming.
14. For CCIS network DAY-NIGHT MODE STATUS -D [D-164D] can be assigned however, if the network is controlled by a master day/night node the local System will change day/night status as per [C-56] Centralized Day/Night Mode Change - CCIS.

15. Night Attendant Console [(NATT) ATRK = 915] can not be used in conjunction with DAY-NIGHT MODE STATUS -D [D-164D]. These features are mutually exclusive.
16. DAY-NIGHT MODE STATUS -D [D-164D] can not be used with an External key Unit (NCU). These features are mutually exclusive.
17. DAY-NIGHT MODE STATUS -D [D-164D] will output a 17-H informational System Message as per System Data programming. (see System Message 17-H below for details).
18. The DAY/NIGHT MODE Feature key cannot be assigned as a Soft key.

Interactions

1. Refer to the Service Conditions and Programming pertaining to the following features:
    - ATTENDANT NIGHT TRANSFER [A-19]
    - CENTRALIZED DAY/NIGHT MODE CHANGE - CCIS [C-56]
    - DAY NIGHT CLASS OF SERVICE [D-15]
    - LDN NIGHT CONNECTION [L-28]
    - LDN NIGHT CONNECTION - CCIS [L-29]
    - LDN NIGHT CONNECTION OUTSIDE [L-30]
    - NIGHT CONNECTION - FIXED [N-1]
    - NIGHT CONNECTION - FLEXIBLE [N-2]
    - NIGHT CONNECTION - FIXED - CCIS [N-15]
    - NIGHT CONNECTION - FLEXIBLE - CCIS [N-16]
    - NIGHT CONNECTION OUTSIDE - SYSTEM [N-17]

Programming

STEP 1: **ASFC** - Assign to the Dterm telephone a Service Feature Restriction Class that allows SFI 214. Data must be "1".

Note: The day/night mode status display on the LCD of Desktop terminal is available when "0" is assigned to SFI 214 as well.

STEP 2: **AKYD** - Assign DAY-NIGHT Mode STATUS -D to a Line/Feature key. Assign KYI =1, FKY = 153.

STEP 3: **ASYD** - SYS1, Index 229, Bit 3. 17-H System Message output of Day/Night Change: 0/1= Out of service/In service

STEP 4: **ASYDL** - SYS1, Index 1131, Bit 0=1. The current day/night mode LED status of day/night mode: 0/1=Out of service/In service

(FCCS Programming)

STEP 1: **ASYDL** - SYS1, Index 640. Assign Fusion Point Code of Day/Night Controlling Node (1-255).

SYS1, Index 704 to Index 735. Assign Fusion Point Code of the nodes (1-255) to be controlled by the Day/Night Controlling Node (assigned by the ASYDL, SYS1, Index 640).

(CCIS Programming)

STEP 1: **ASYD** - SYS1, Index 18, Bit 6=1. Assign for satellite Office to follow Day/Night Mode at Main Controlling Node (1-15).

SYS1, Index 19, Bit 0 to Bit 3. Assign Tenant (TN = 1 - 15) of the Main Office where the Satellite Office belongs.

D-164D - II Day-Night Mode Status -D - II

General Description

This feature allows a Desktop terminal to display the current day/night mode status and change the System into Day or Night mode using either the "*" or "#" key.

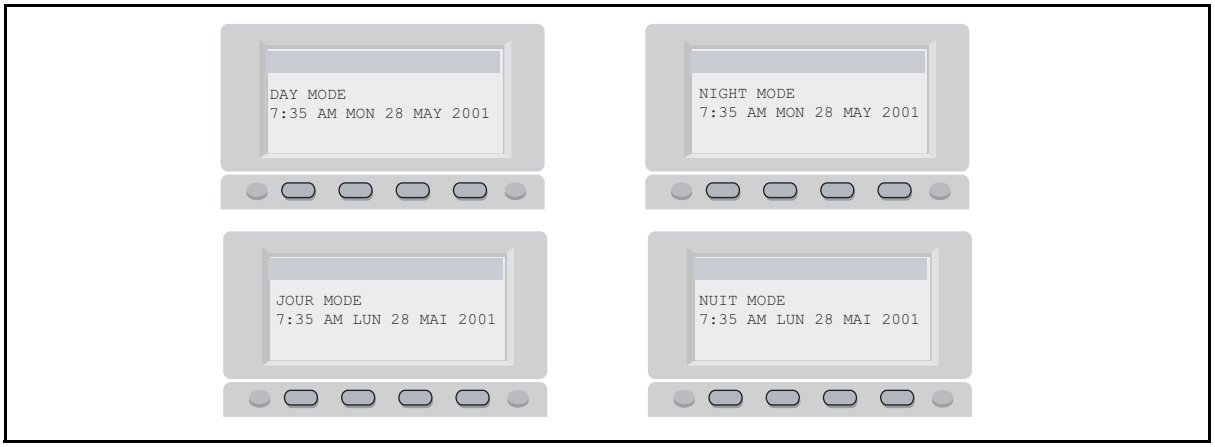
Note: If this feature is used in an environment where an attendant console is accommodated, day/night mode cannot be changed from Desktop terminal.

Note: This feature is not available in North America.

Operating Procedure

Day/Night Mode Status Display on LCD of Desktop terminal
(Not effected by ASFC/ASFCN)

1. Lift the handset or press the SPEAKER key. Receive Dial Tone.
2. Press the DAY/NIGHT MODE Feature key. Receive the Service Set Tone.
3. The current day/night status of the System is displayed.
4. Place handset in cradle or press the SPEAKER key.



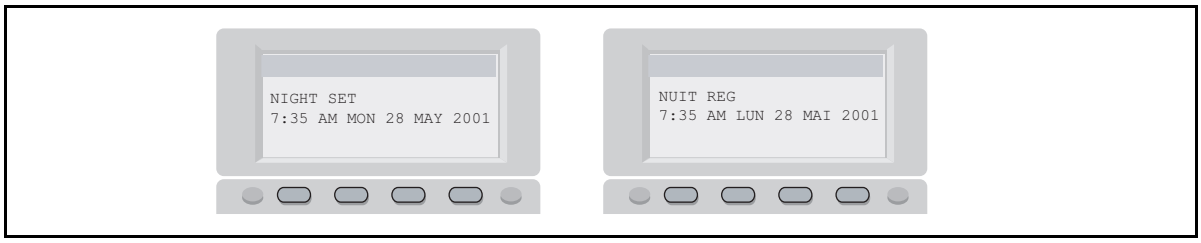
Day/Night Mode Status Display on LED of DAY/NIGHT MODE Feature key
(Not effected by ASFC/ASFCN)

The LED lamp status of DAY/NIGHT MODE Feature key on Desktop terminal indicates the day/night mode status as follows.

LED Status	Day/Night Mode Status
Turned OFF	Day Mode
Lights red	Night Mode

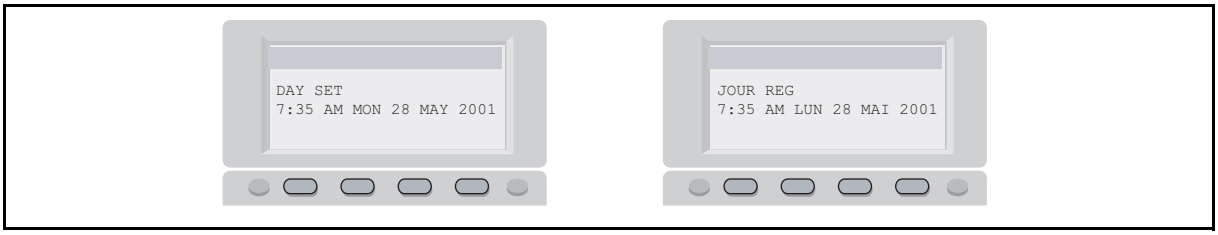
Change from Day mode to Night mode from Desktop terminal
(Effected by ASFC/ASFCN)

1. Lift the handset or press the SPEAKER key. Receive Dial Tone.
2. Press the DAY/NIGHT MODE Feature key. Receive burst Special Dial Tone.
3. The current day/night status of the System is displayed.
4. Dial "#" to set Night mode. Receive the Service Set Tone.
5. Place handset in cradle or press the SPEAKER key.



Change from Night mode to Day mode from Desktop terminal
(Effected by ASFC/ASFCN)

1. Lift the handset or press the SPEAKER key. Receive Dial Tone.
2. Press the DAY/NIGHT MODE Feature key. Receive burst Special Dial Tone.
3. The current day/night status of the System is displayed.
4. Dial "*" to set Day mode. Receive the Service Set Tone.
5. Place handset in cradle or press the SPEAKER key.



Change the language for display

1. To change the language for display, the data assignment is required. For the detail, refer to Programming in [M-117] Multilingual Display.

Service Conditions

1. This feature is available for the following Desktop terminals.
   - Digital terminal
   - IP terminal (Proprietary Protocol/SIP)
   - Soft Phone (Proprietary Protocol/SIP)
2. The current day/night mode status can be checked by the LCD of Desktop terminal, or the LED status of DAY/NIGHT MODE Feature key by assigning the system data (ASYDL, SYS1, Index 1131, Bit 0=1).
3. It may take a maximum of two minutes to change the LED status of DAY/NIGHT MODE Feature keys on all the corresponding Desktop terminals. Also, when activating the system by night mode, it may take a maximum of two minutes after the system activation to light red the LED of DAY/NIGHT MODE Feature keys on all the corresponding Desktop terminals.
4. This feature is controlled by Service Feature Restriction Class and available by assigning the DAY/NIGHT MODE Feature key (FKY = 153) and Service Feature Index 214 for MGC (SFI 214 for MGC = 1) or Service Feature Index 244 for SR-MGC (SFI 244 for SR-MGC = 1).

   Note: The day/night mode status display on the LCD of Desktop terminal and the LED of DAY/NIGHT MODE Feature key are available when "0" is assigned to SFI 214 for MGC or SFI 244 for SR-MGC as well.

5. ASYD - SYS1, Index 9 (Number of Attendant Consoles) must be "00" hex for this feature. Otherwise, this feature except "Day/Night Mode Status Display on LCD of Desktop terminal" and "Day/Night Mode Status Display on LED of DAY/NIGHT MODE Feature key" is not available.
6. Any Desktop terminal Line appearance may be used to perform the day/night status change, however only the Tenant of the Desktop terminals My Line will be effected. (For example, a Tenant 3 Line appearance on a Tenant 1 Desktop terminal can not change day/night mode of Tenant 3, only Tenant 1).
7. When the day/night mode change is executed from multiple terminals, the mode is changed by each terminal.
8. The day/night mode change by this feature indicates the mode change from Day mode to Night mode 3 or from Night mode 3 to Day mode.
9. For changing day/night mode, Special Dial Tone is fixed to burst tone.

   Note: When using Softphone, the burst Special Dial Tone will not be provided for changing day/night mode after DAY/NIGHT MODE Feature key is pressed. In this case, Dial Tone will be provided, and the Day-Night mode can be changed from this state.

10. The day/night mode in day/night non-controlling nodes can be changed from the day/night controlling node of the same tenant only as specified in programming for a FCCS network. The day/night data is transferred to the day/night non-controlling nodes from the day/night controlling node in 30 seconds cycle, regardless of the day/night mode change execution.
11. When this feature is used in a FCCS network, "Day/Night Mode Status Display on LED of Desktop terminal" is available for the day/night controlling node only.
12. This feature must not be assigned to a Line/Feature key of terminals under the day/night non-controlling nodes.
13. This feature setting is not available in consultation hold condition.

14. This feature can not be used with External Key Unit (NCU). These features are mutually exclusive.
15. This feature will output a 17-H informational System Message as per System Data programming. (see System Message [17-H] in Operations and Maintenance Manual for details).
16. This feature cannot be assigned to a Soft key.
17. The day/night mode cannot be changed by an access code.
18. At the time of changeover from MGC to SR-MGC, the day/night mode (fixed to Night Mode 3 at this time) can be changed from station that is assigned the SFC that has the following system data.

    - ASFC/ASFCN command: SFI 244 for SR-MGC = 1 (Day/night mode change (only for SR-MGC) is allowed.)

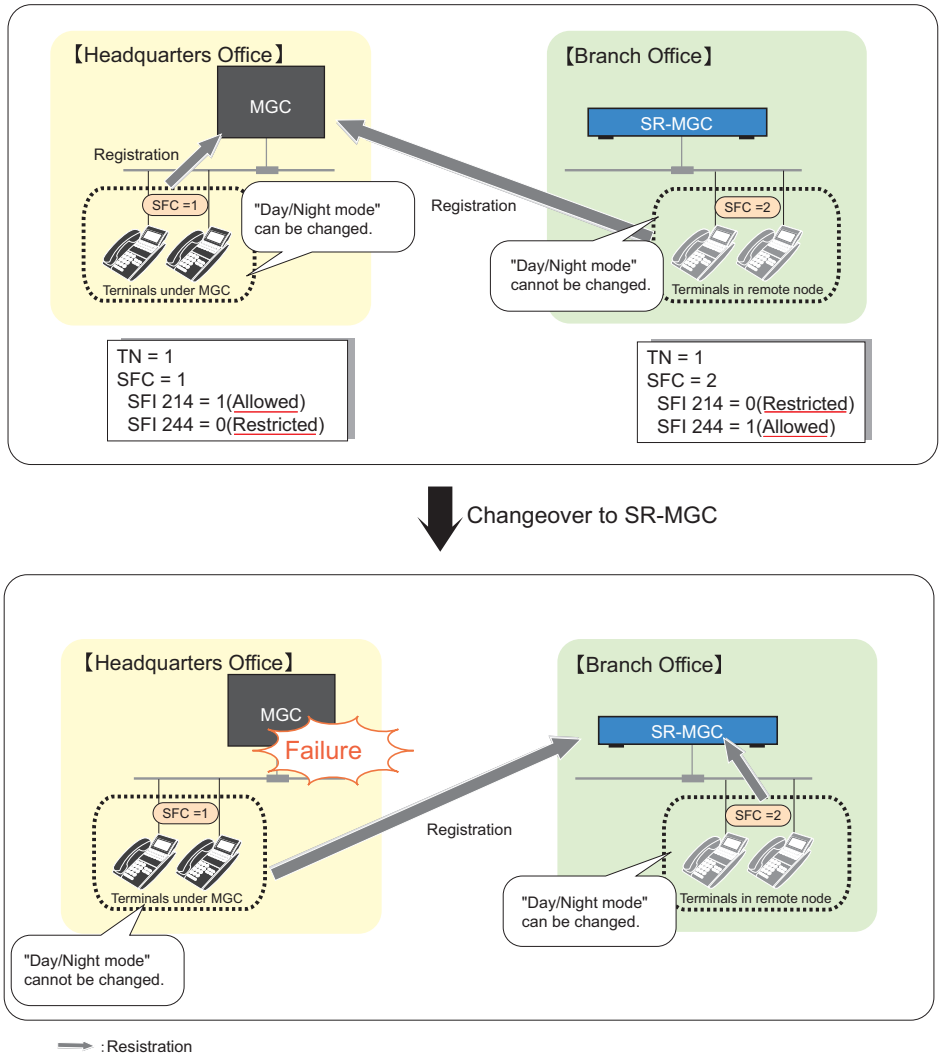
    Note: SFI = 244 is valid only when registered in SR-MGC. (SFI = 214 is valid only when registered in MGC)

    At the time of changeover from MGC to SR-MGC, the day/night mode can be changed from SR-MGC as well, by dividing the SFC for the terminal under MGC (hereinafter called headquarters office in the following example) and terminal under SR-MGC (hereinafter called branch office in the following example).

[Example]

The day/night mode can be changed from the terminal on headquarters side only when being registered in MGC (At the time of changeover from MGC to SR-MGC, the day/night mode cannot be changed), and from the terminal on branch office side at the time of changeover from MGC to SR-MGC only (it cannot be changed unless the changeover is executed).

Terminal's belonging	Tenant	SFC	Mode Change on MGC side	Mode Change on SR-MGC side
Headquarters Office	TN = 1	SFC = 1	Available (SFI 214 for MGC = 1)	Not available (SFI 244 for SR-MGC = 0)
Branch Office	TN = 1	SFC = 2	Not available (SFI 214 for MGC = 0)	Available (SFI 244 for SR-MGC = 1)



19. TAS service is not available.
20. Current day/night mode can be retained after the system initialization by assigning System data - ASYDL, SYS1, Index1120, Bit 0 to Bit 5 for Day/Night Data Automatic Backup. For the detail, refer to the Day/Night Data Retention section in Chapter 2 of this manual.
21. Assign a destination number for a COT incoming call. If not assigned, the system cannot answer COT incoming calls. For more information, refer to [D-7] Direct-In Termination (DIT) or [D-8] Direct Inward Dialing.

Interactions

1. Refer to the Service Conditions and Programming pertaining to the following features:
    - Day/Night Data Retention
    - DAY-NIGHT MODE STATUS -D [D-164D]
    - MULTILINGUAL DISPLAY [M-117]
2. When DAY-NIGHT MODE STATUS -D - II [D-164D - II] is enabled, neither NIGHT CONNECTION - FIXED [N-1] nor NIGHT CONNECTION - FLEXIBLE [N-2] is available.
3. The Day/Night switching feature of DAY-NIGHT MODE STATUS -D - II [D-164D - II] is not available when one of the following services is enabled:
    - NIGHT CONNECTION - FIXED [N-1]
    - NIGHT CONNECTION - FLEXIBLE [N-2]
    - DAY/NIGHT CHANGE - OUTSIDE KEY OPERATION [D-63]

On the other hand, the Day/Night mode status display feature of DAY-NIGHT MODE STATUS -D - II [D-164D - II] can be used in conjunction with the above services.

Programming

STEP 1: **ASFC** - Assign to the Dterm under MGC a Service Feature Restriction Class that allows SFI 214 for MGC and, assign to the Dterm in remote nodes an other Service Feature Restriction Class that allows SFI 244 for SR-MGC. By assigning "1" to the each SFI, day/night mode can be changed from Dterm at both MGC and SR-MGC if a changeover is executed. **Note 1**

Note 1: The day/night mode status display on the LCD of Desktop terminal and the LED of DAY/NIGHT MODE Feature key are available when "0" is assigned to SFI 214 for MGC or SFI 244 for SR-MGC as well.

STEP 2: **AKYD** - Assign DAY/NIGHT MODE to a Line/Feature key. Assign KYN (1-40), KYI =1, FKY = 153.

STEP 3: **ASYD** - SYS1, Index 9=00. Number of Attendant Consoles **Note 2**
SYS1, Index 229, Bit 3. 17-H System Message output of Day/Night Change: 0/1= Out of service/In service
SYS2, Index 11, Bit 6=0. Trunk Answer from Any Station (TAS) **Note 3**

Note 2: This data must be "00" hex for this feature. Otherwise, this feature except "Day/Night Mode Status Display on LCD of Desktop terminal" and "Day/Night Mode Status Display on LED of DAY/NIGHT MODE Feature key" is not available.

Note 3: For the tenant where this feature is used, this data must be “0”. If this data is changed from “1” to “0”, the following procedure will be required.
- Backup office data (select all Data Type) by using MEM_HDD command.
- Delete “GPCM04.DAT (Day/Night Data)” by using DHDF command if it exists.
- Execute “Office Data Load & System Initialize” by using SINZ command.

STEP 4: **ASYDL** - SYS1, Index 1131, Bit 0. Day/Night Mode Status Display on LED of DAY/NIGHT MODE Feature key: 0/1 = Out of service/In service.

(FCCS Programming)
STEP 1: **ASYDL**

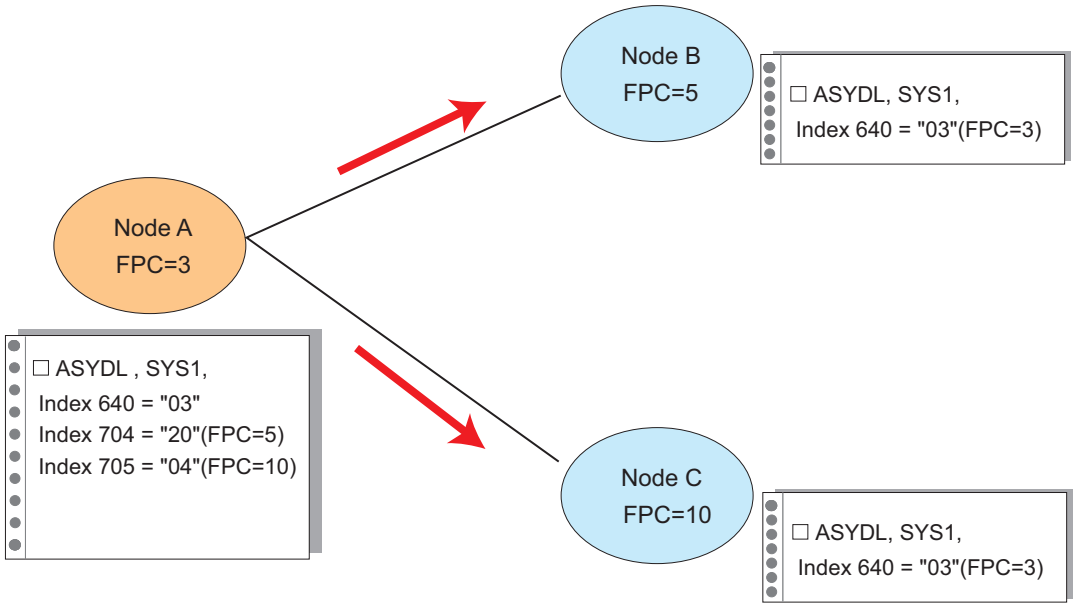
[For all nodes]

SYS1, Index 640. Assign Fusion Point Code of Day/Night Controlling Node (1-255).

[For Day/Night Controlling Node]

SYS1, Index 704 to Index 735. Assign Fusion Point Code of the Day/Night Non-Controlling Nodes (1-255) to be controlled by the Day/Night Controlling Node (assigned by the ASYDL, SYS1, Index 640).

[Example: Day/night data is transferred from Node A to Node B/Node C]



STEP 2: **ASFCN** - Assign to the Desktop terminal under MGC a Service Feature Restriction Class that allows SFI 214 for MGC and, assign to the Desktop terminal in remote nodes an other Service Feature Restriction Class that allows SFI 244 for SR-MGC. By assigning "1" to the each SFI, day/night mode can be changed from Desktop terminal at both MGC and SR-MGC if a changeover is executed.

Note: The day/night mode status display on the LCD of Desktop terminal and the LED of DAY/NIGHT MODE Feature key are available when "0" is assigned to SFI 214 for MGC or SFI 244 for SR-MGC as well.

D-165 Dialed Number Privacy Mode

General Description

This feature allows the Desktop terminal user to change the Dialed Number Privacy mode at any time. The following three modes are available:

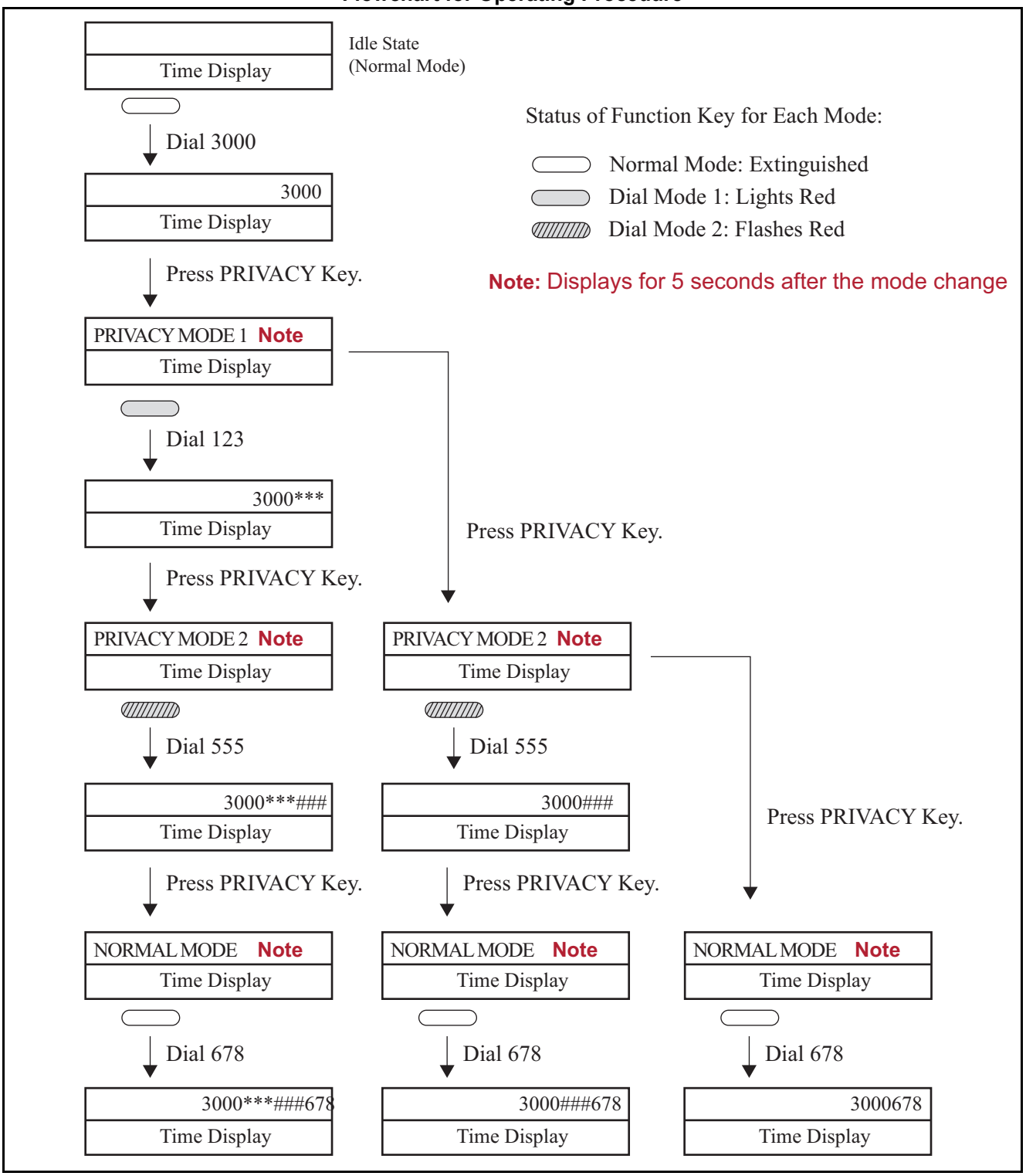
Normal Mode:
Dialed Number is displayed on the LCD of Desktop terminal as usual.
In this mode, Last Number Called -D [L-6D] is available.

Privacy Mode 1:
The Secret Code (the default is "*") is displayed on the LCD of Desktop terminal instead of the dialed number.
In this mode, Last Number Called -D [L-6D] is not available.

Privacy Mode 2:
The Secret Code (the default is "#") is displayed on the LCD of Desktop terminal instead of the dialed number.
In this mode, Last Number Called -D [L-6D] is available.

Operating Procedure

Flowchart for Operating Procedure



Service Conditions

1. This feature is available for the following terminals:
   Dterm Series *i*
   Softphone
   DtermIP INASET
   DtermIP
   DT800 Series/DT700 Series/DT400 Series/DT300 Series
2. This feature doesn't conceal the Calling Party Number and Name during the conversation, ringing and call termination.
3. All the dialed numbers, including Service Access Codes, can be concealed by this feature.
4. System Initialization and the ASDT data modification/deletion place the Stations in the Normal mode.
5. Except in case of the above (4), the existing mode is maintained until it is changed manually, regardless of the status of the terminal.
6. When the terminal is reset (AKYD, ASCL, ASTN, ADKS, MBST, MBLE or DtermIP Registration):
   If the PRIVACY Feature key (FKY: 163) is programmed to the terminal, the existing mode is preserved even after the terminal reset.
7. The Secret Code to be used for this feature can be changed by the System Data (ASYDL, SYS1, Index 675 and Index 676). If an invalid code is set by the System Data, the default code (Privacy Mode 1: "*", Privacy Mode 2: "#") will be used.
8. When the Secret Code is changed by the System Data, new Secret Code is valid after the next mode change.
9. By assigning ASYDL, SYS1, Index 674, Bit 7=1, this feature can be activated with DESI-Less Phone under the following conditions.
   a.) The icon on DESI-Less Phone indicates each dial mode as follows:
   - Normal Mode: No information
   - Privacy Mode 1: ■ (steady)
   - Privacy Mode 2: ■ (flashing)

   b.) When ASYDL, SYS1, Index 674, Bit 6=1 (Speed Calling number is displayed on the LCD of SPD key), Speed Calling number display can act together with this feature depending on the system data setting (**Note 1**).
   <Example>
   When "12345" is registered to SPD key and ASYDL, SYS1, Index 674, Bit 7=1 (Dial mode works on Speed Calling number display):
   - Normal Mode: "12345" is displayed.
   - Privacy Mode 1, 2: "SPD" is displayed.

Note 1: Regardless of the ASYDL, SYS1, Index 674, Bit 7 setting, the feature name "SPD" is displayed when ASYDL, SYS1, Index 674, Bit 6=0.

Interactions

1. When using this feature, Last Number Called -D [L-6D] is activated under the following conditions:

   Note: "***" indicates the dialed numbers while the terminal is in Privacy Mode 1.
   "###" indicates the dialed numbers while the terminal is in Privacy Mode 2.

   a.) When verifying the stored number in the memory, the number indication is dependent on the mode at the time it was dialed.
   Example: "123###" was dialed:
   When verifying the above stored number, "123###" is displayed, regardless of the present mode.
   b.) Dialed Number while the terminal is in Privacy Mode 1 cannot be stored in the memory.
   c.) When the user originates a call by Last Number Called -D [L-6D], the Digit Codes are sent under the following conditions:
   Example: The following tables shows the display indication when "123456789" is redialed in each mode.
   "***" indicates the Digit Codes sent when the user redials in Privacy Mode 1.
   "###" indicates the Digit Codes sent when the user redials in Privacy Mode 2.

Dialed Number	Mode when the user redials			Remarks
	Normal Mode	Privacy Mode 1	Privacy Mode 2	
123###***	123###	******	######	"123456" is sent as Digit Codes.
123***###	123	***	###	"123" is sent as Digit Codes.
###456***	###456	******	######	"123456" is sent as Digit Codes.
###***789	###	***	###	"123" is sent as Digit Codes.
***456###	-	-	-	Redial is not available.
***###789	-	-	-	Redial is not available.

   d.) If the user redials while the terminal is in Mode 1 or 2, the numbers are stored in the memory under the following conditions:
   Example: Redial the stocked number "123###":
   (In the case of Privacy Mode 1)
   The redialed number "******" is not stored in the memory, and "123###" remains in the memory.
   (In the case if Privacy Mode 2)
   The redialed number "######" is stored in the memory, instead of the stored number "123###."
   e.) When verifying the stored number in the memory after the Secret Code is changed by the System Data, the stored numbers are indicated with the new Secret Code.
2. This feature (Privacy Mode 1 or 2) has precedence over the other concealment function such as Authorization Code Display Elimination and DtermIP Password Encryption. When using Normal mode, the other concealment function works as usual ("X" will be displayed).
3. This feature (Privacy Mode 1 or 2) is also available for the call origination by using Speed Calling - One Touch -D [S-26], Call Return [C-151], and Message Reminder [M-19].
4. When a call is originated by Speed Calling - Station [S-21], Speed Calling - Group [S-23], or Speed Calling - System [S-3], only the Abbreviated Call Code is concealed. The destination number cannot be concealed.

5. When using this feature, Speed Calling - One Touch -D [S-26] is activated under the following conditions:
   - If the destination number is programmed to the SPEED CALLING-ONE TOUCH key while in the terminal is in Privacy Mode 1 or 2, "*" or "#" is displayed on the LCD of Desktop terminal.
   - If the PRIVACY Feature key is pressed to change the Privacy Mode during the SPEED CALLING-ONE TOUCH key programming, the programming is canceled.
   - When verifying the destination number programmed to the SPEED CALLING-ONE TOUCH key, the LCD display is dependent on the present mode.
   - This feature can conceal the number only.
6. When using this feature, Save And Repeat -D [S-24D] is activated under the following conditions:
   - The number dialed in Normal Mode or Privacy Mode 2 can be saved. The number dialed in Privacy Mode 1 cannot be saved. At this time, the call origination is restricted.
   - The Calling Party Number can be saved regardless of the Mode.
   - When the saved number is re dialed, the LCD display is dependent on the present mode.
7. When this feature is activated with PA-16ELCJ-B, the following firmware is required: SP-3797 PROG-H or later
   This feature is not available for the speech log panel (by pressing call logs button) of Soft Phone.

Programming

STEP 1: **ASYDL/ASYDN** -

SYS1, Index 675: Assign Secret Code for Privacy Mode 1. [The default (00 hexadecimal) is "*."]
SYS1, Index 676: Assign Secret Code for Privacy Mode 2. [The default (00 hexadecimal) is "#."]

Programmable Secret Codes

		Higher order 4 bits (hex.)															
		0	1	2	3	4	5	6	7	8	9	A	B	C	D	E	F
Lower order 4 bits (hex.)	0				0	@	P	`	p								
	1			!	1	A	Q	a	q								
	2			"	2	B	R	b	r								
	3			#	3	C	S	c	s								
	4			$	4	D	T	d	t								
	5			%	5	E	U	e	u								
	6			&	6	F	V	f	v								
	7			'	7	G	W	g	w								
	8			(	8	H	X	h	x								
	9			)	9	I	Y	i	y								
	A			*	:	J	Z	j	z								
	B			+	;	K	[	k	{								
	C			,	<	L	¥	l	|							
	D			-	=	M	]	m	}								
	E			.	>	N	^	n									
	F			/	?	O	_	o									

STEP 2: **AKYD** - Assign the PRIVACY Feature key data to a programmable Line/Feature key on the desired Dterm. (This command data is necessary when using a Line/Feature key as the PRIVACY key.)
FKI: 1 (Function Key)
FKY: 163 (PRIVACY Key)

STEP 3: **ADSL** - Assign the PRIVACY key data to a Soft key on the desired Dterm. (This command data is necessary when using a Soft key as the PRIVACY key.)
SKP: 0-15 (Soft Key Pattern)
SN: Assign each Status Number for "SKP." (0-15)
SKN: 1-16 (Soft Key Number)
FKY: 163 (PRIVACY key)

STEP 4: **ADKS** - Register each Soft Key Pattern data to the desired Dterm(s). (This command data is necessary when using a Soft key as the PRIVACY key.)
TN: Tenant Number
STN: Station Number
SKP: 1-15 (Soft Key Pattern assigned by ADSL)
LKP: Assign the Key Pattern of the Dterm.
1 = 16 Line/Feature keys + DSS keys
2 = 24 Line/Feature keys + DSS keys
3 = ALL Line/Feature keys
PG: Specify the page scroll key allocation
0 = Page change key appears on a software key (>>>)
1 = Page change key disappears on a software key

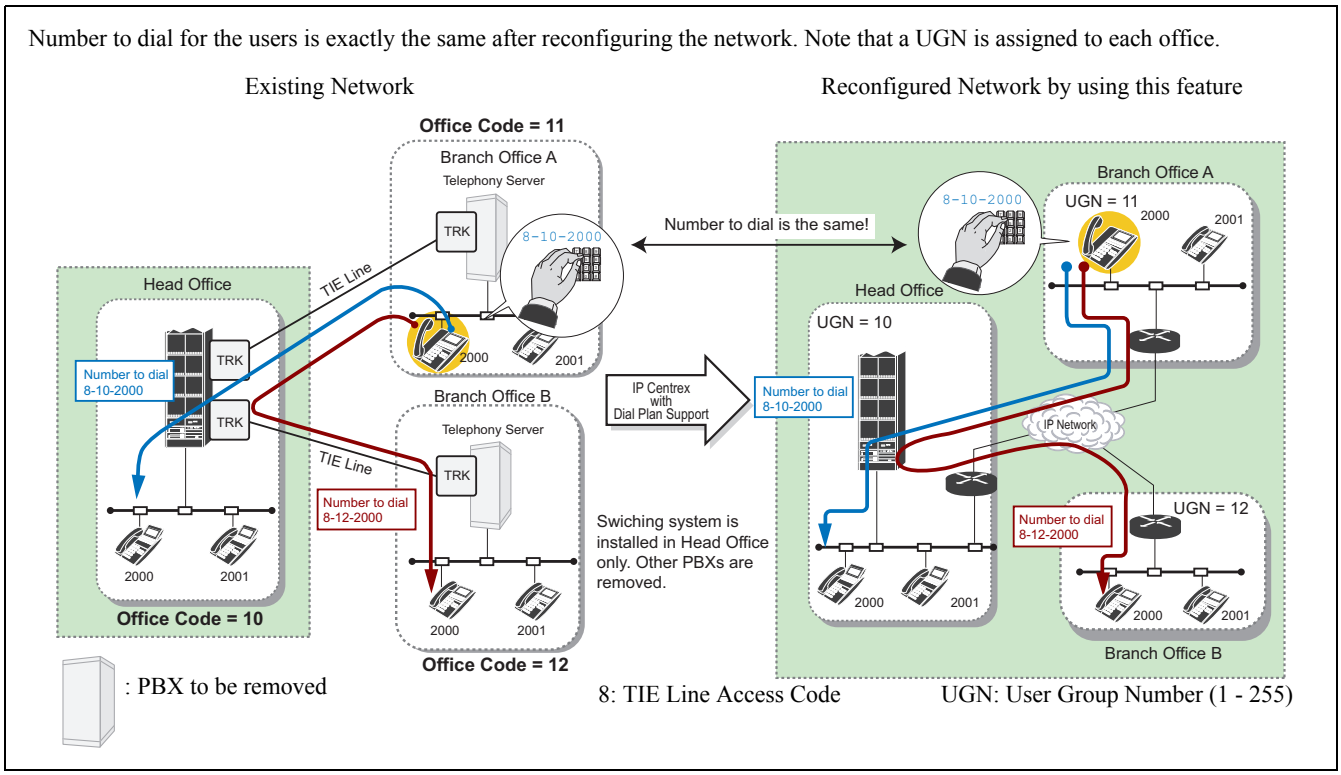
<Data Programming for Speed Calling on DESI-Less Phone>

STEP 1: **ASYDL** - SYS1, Index 674, Bit 7=1: Dialed Number Privacy Mode works with Speed Calling number display.

D-166 Dial Plan to support IP Centrex Functionality

General Description

This feature is useful when changing an existing network to IP-based network. In the following example, two switching systems installed in branch offices are removed. Telephones in the branch office are accommodated in the Head Office. And then the users can use the existing telephone services without their own switching system (Centrex Functionality). Thus this feature can bring about the cost reduction of telecommunications. In addition, the telephone users can use the previous numbering plan to which they have been accustomed - Dial Plan is supported.



Operating Procedure

Number Definition

The following number is used for this feature.

Extension Number: Used for a Station-to-Station call within a node.
example: `77033` (Telephone Number)

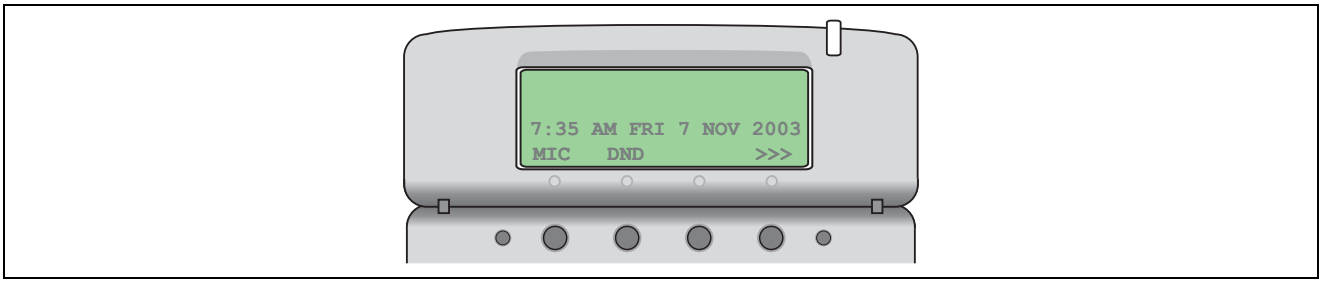
Private Number: Used for a call over nodes. This number includes the access code.
example: `8 + 26 + 77033` (Access Code + Office Code + Extension Number)

Public Number: Used for an outgoing call to PSTN.
example: `9 + 04 + 71420230` (Access Code + Toll Code + Local Number)

Login Operation from a DtermIP

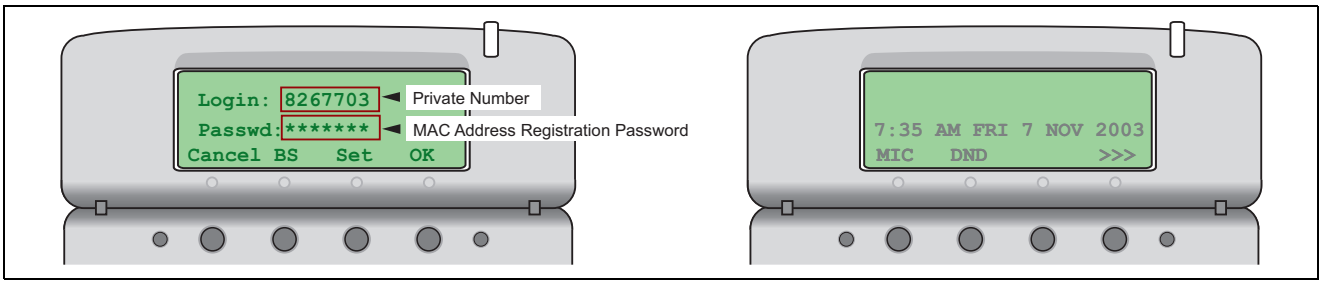
[MAC Authentication Mode by AMAC]

1. Assign the MAC address of DtermIP you want to use together with the Extension Number by AMAC.
2. Turn on the DtermIP.
3. After a while, the current time is appeared on the LCD, indicating that the DtermIP is in service.



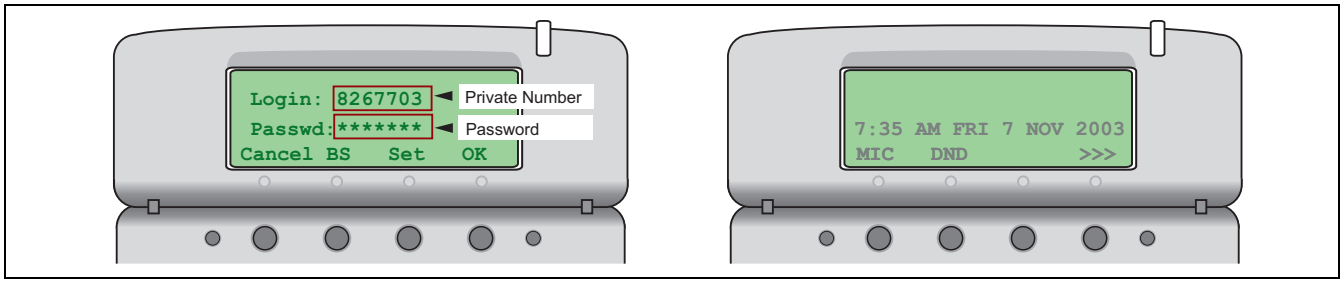
[MAC Authentication Mode by MAC Address Registration Password]

1. Assign the MAC Address Registration Password by ADTM.
2. Turn on the DtermIP.
3. To log in, enter the Private Number as Login Code at Login prompt. And enter the MAC Address Registration Password that has been assigned by ADTM at Password prompt.
4. After a while, the current time is displayed on the LCD, indicating the DtermIP is in service. (When reconnecting the DtermIP to the LAN, the terminal is in service without entering the Login Code and Password.)

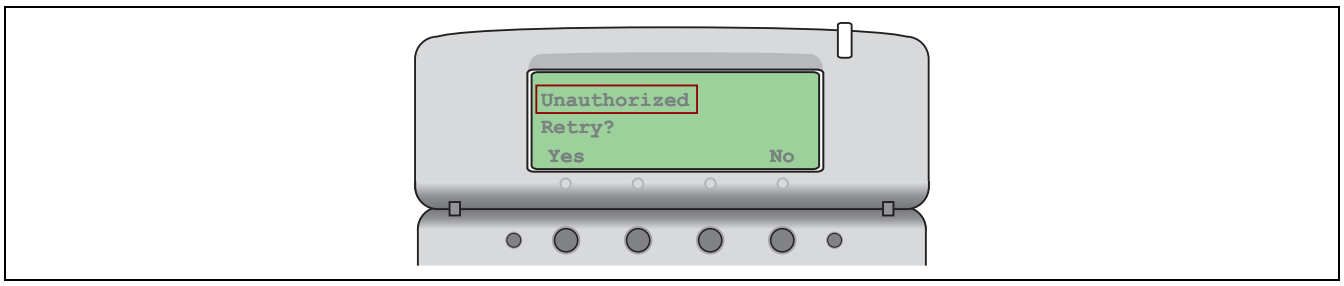


[Login Mode]

1. Turn on the DtermIP.
2. To log in, enter the Private Number as Login Code at the login prompt. And then enter the password at the Passed prompt. If the password has not been assigned, enter the Private Number as the password.
3. After a while, the DtermIP can be used. (When the user logs off or reconnects the DtermIP to the LAN, the terminal can be operational after entering the Login Code and Password.)

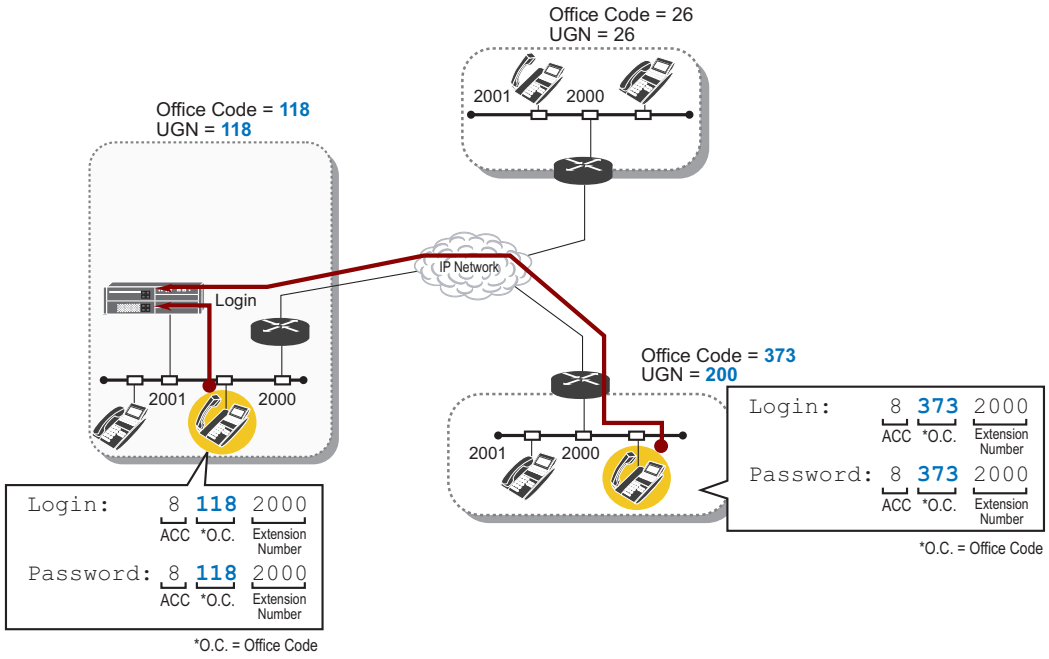


Note: In login mode, the LCD displays “Unauthorized” if Private Number is not entered at Login prompt or a wrong Password is entered.



Login Code / Password entry Example

The following shows an example of Login Code/Password entry from the DtermIP at each office. Note that Private Number is used to log into the system. When the Password has not been assigned, enter the Private Number as the Password.



UGN: User Group Number ACC: Access Code

[Password Change]

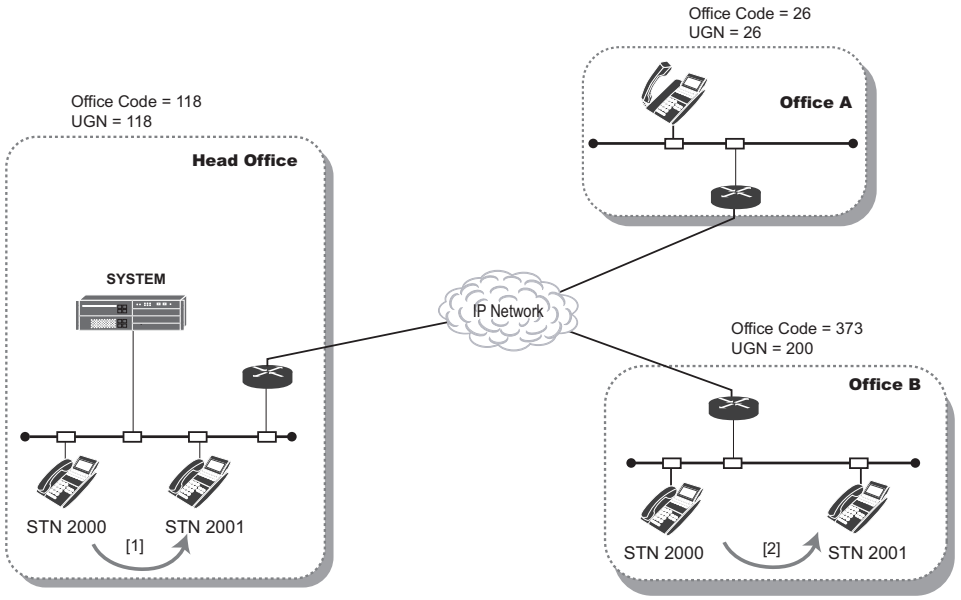
1. Lift the handset and dial “IP Enabled Dterm/DtermIP Password Entry” access code; receive SPDT. The LCD indicates “OLD PASSWORD” prompt.
2. Enter the current password; receive SPDT.
3. Enter the new password at “NEW PASSWORD” prompt; receive DT.
4. “REPEAT” prompt appears. Enter the new password again to confirm your entry; receive SST.
5. Replace the handset.

Service Conditions

1. Basic connection conditions are as follows.

[Internal Node Connections]

- Number is managed by Telephone Number assigned on a UGN basis.
- Dialed number is logically developed by using the UGN of the calling party.



[1] When STN 2000 of Head Office dials 2001, the number is logically developed using UGN = 118, Telephone Number = 2001. Then the call terminates to STN 2001 of Head Office.

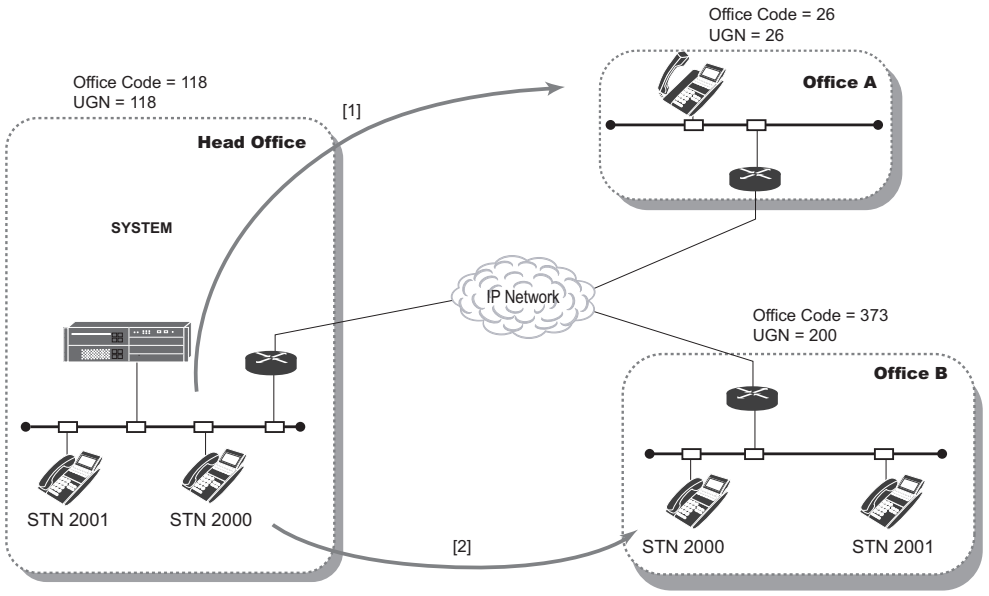
[2] When STN 2000 of Office B dials 2001, the number is logically developed using UGN = 373, Telephone Number = 2001. Then the call terminates to STN 2001 of Office B.

- Calling Party Information

Station-to-Station Call	Number to dial	Calling Party Information to be displayed
STN 2000 (UGN:118) => STN 2001 (UGN:118)	2001	2000
	8–118–2001	2000

[Inter-UGN Call]

- The relationship between UGN and Private Number is managed by AUGNN and AUPVN commands.
- UGN and Telephone number are decided by dialed number. And it is logically developed.
  (example) Dialed number: Access Code (8) + Office Code (26)* + Telephone Number (77033)
  * The underlined part corresponds to AUGNN command.



[1] When STN 2000 of Head Office dials 8-26-2000, the number is logically developed using UGN = 26, Telephone Number = 2000. Then the call terminates to STN 2000 of Office A. By using the LENS of the calling party, the UGN and Telephone Number of the calling party can be decided. Then the Private Number (81182000) can be obtained that is indicated on the LCD of the called party by using the UGN (118) and Telephone Number (2000).

[2] When STN 2000 of Head Office dials 8-373-200, the number is logically developed using UGN = 200, Telephone Number = 2000. Then the call terminates to STN 2000 of Office B. By using the LENS of the calling party, the UGN and Telephone Number of the calling party can be decided. Then the Private Number (81182000) can be obtained that is indicated on the LCD of the called party by using the UGN (118) and Telephone Number (2000).

- Calling Party Information is obtained by the following procedure.

  - UGN, Telephone Number are decided by using the LENS of the calling party.
  - Access Code, Office Code are determined by the UGN and Telephone Number.
  - The Access Code and Office Code are added to the Telephone Number.
  - The obtained Number is displayed on the LCD of the called party.

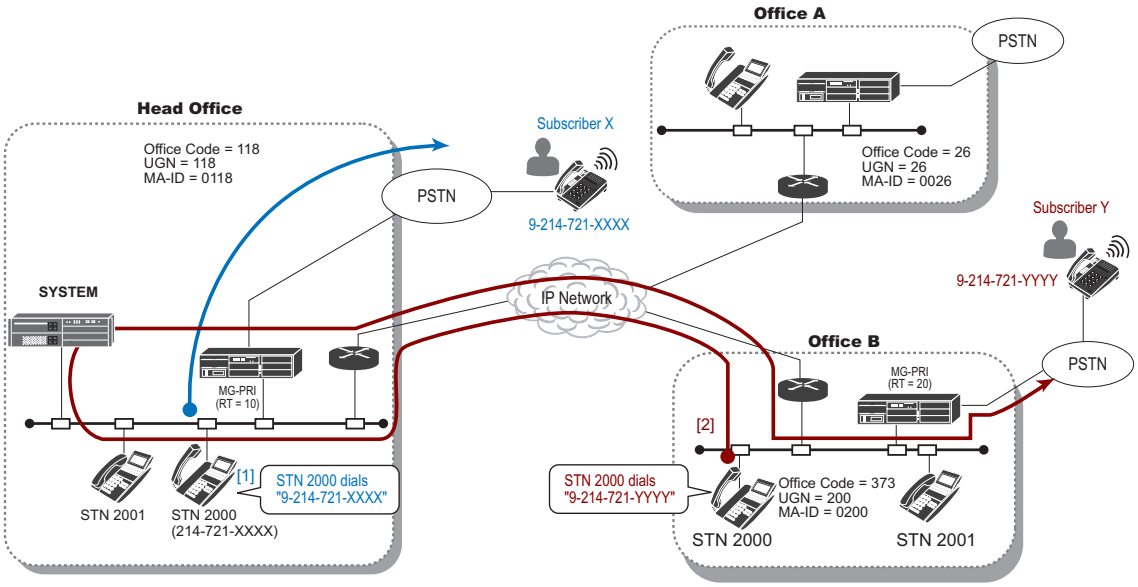
- Calling Party Information

  Before using this feature, this call was treated as a trunk call. Therefore, the trunk type and elapsed time indication were displayed. After using this feature, the information is not displayed. Only the following information is displayed.

Inter-UGN Call	Number to dial	Calling Party Information to be displayed
STN 2000 (UGN:118) => STN 2001(UGN:26)	8-26-2000	8-118-2000

[Central Office Outgoing Call]

- Routing by using MG in each node is performed by LEAST COST ROUTING FOR MG-PRI [L-54].



[1] When STN 2000 of Head Office dials 9-214-721-XXXX, MG-PRI (RT 10) is selected by LEAST COST ROUTING FOR MG-PRI for placing the call to PSTN.

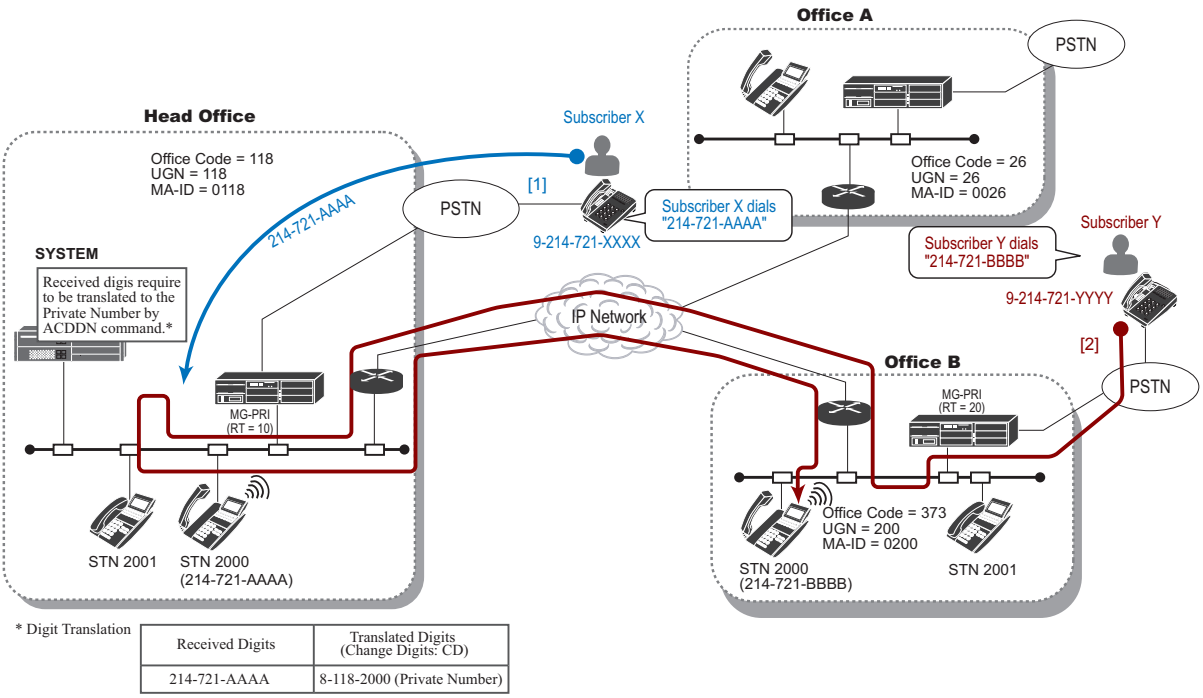
[2] When STN 2000 of Office B dials 9-214-721-YYYY, MG-PRI (RT 20) is selected by LEAST COST ROUTING FOR MG-PRI for placing the call to PSTN.

- Calling Party Information

Central Office Outgoing Call	Number to dial	Calling Party Information to be displayed
STN 2000 (UGN:118) => STN 2001 (UGN:26)	8-26-2000	8-118-2000

[Central Office Incoming Call]

- Received number is translated to the corresponding Private Number by ACDDN command.
- Be sure to assign Private Number to the Change Digit Code (CDC) of ACDDN command. If not, the call cannot terminate to the destination.



* Digit Translation

Received Digits	Translated Digits (Change Digits: CD)
214-721-AAAA	8-118-2000 (Private Number)

[1] When Subscriber X dials 214-721-AAAA, the received digits are translated to 81182000 by ACDDN data, terminating to MG-PRI of Head Office. The call is logically developed by Private Number 81182000.

[2] When Subscriber Y dials 214-721-BBBB, the received digits are translated to 83732000 by ACDDN data, terminating to MG-PRI of Office B. The call is logically developed by Private Number 83732000.

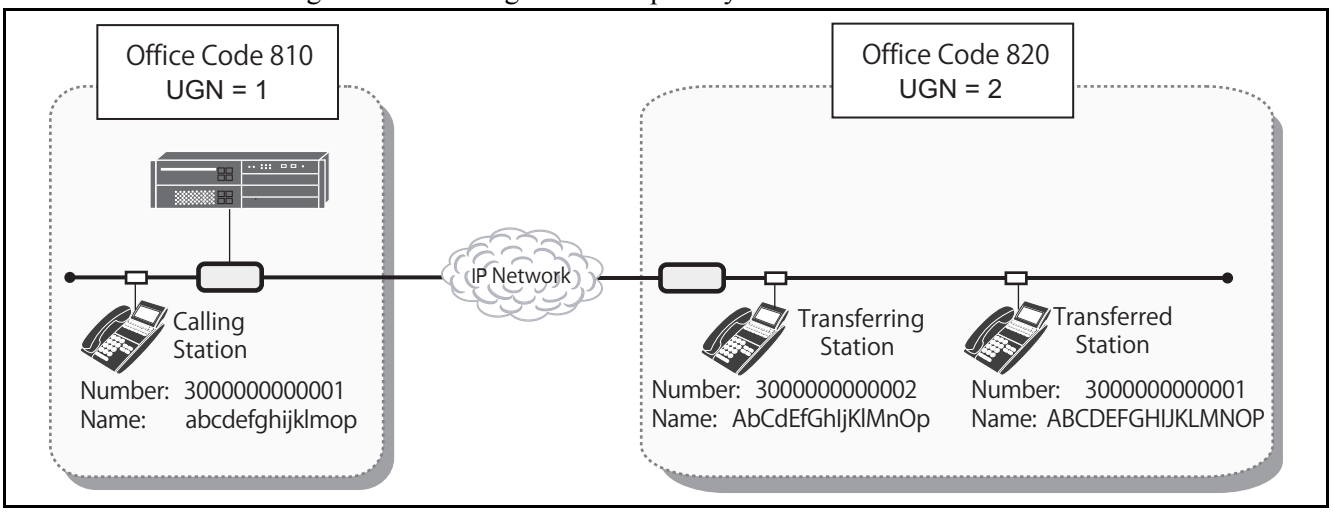
2. User Group Number (UGN) is used to realize Identical Number Processing.
3. Extension Number (Telephone Number), Private Number, and Public Number can be used by telephone users. The maximum digits for each number is as follows:

   - Extension Number (Telephone Number) : 14 digits **Note 1**
   - Private Number : 16 digits (inclusive of Access Code and Office Code) **Note 1**
   - Public Number : 16 digits

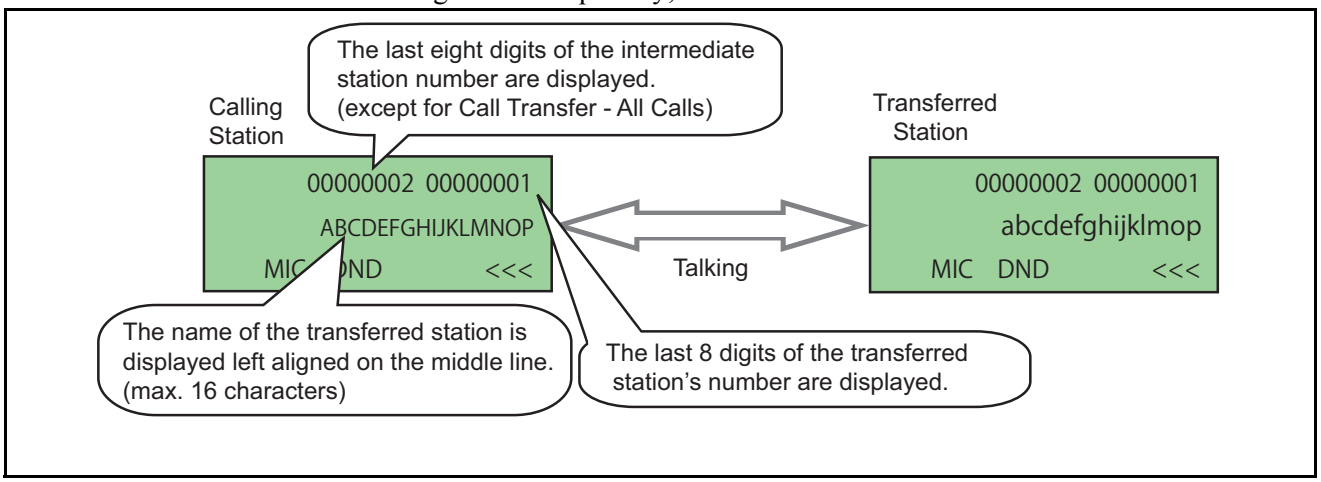
   Note 1: Only when the following system data is enabled: ASYDN, SYS1, Index 818, Bit 3; if not, up to five digits for an Extension Number and up to eight digits for a Private Number.
4. The following are the service conditions when more than 9-digit Private Number is used in Inter-UGN calls.

- If a call is transferred (except when a call is transferred by Call Transfer - All Calls), the last eight digits of the intermediate station number are displayed.
- When using Name Display - System [N-28], the LCD display varies depending on the following system data setting. The following shows an example.

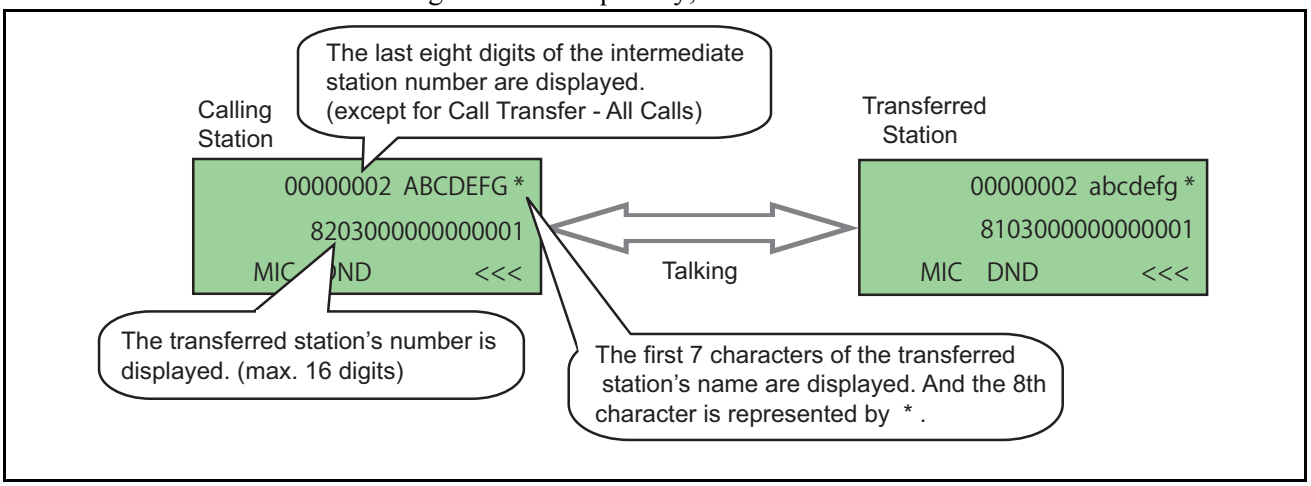
ASYDN, SYS1, Index 390, Bit 0 Caller ID display pattern on the LCD
0/1 = Calling Number/Calling Name has priority.



- Bit 0=0: When Calling Name has priority;

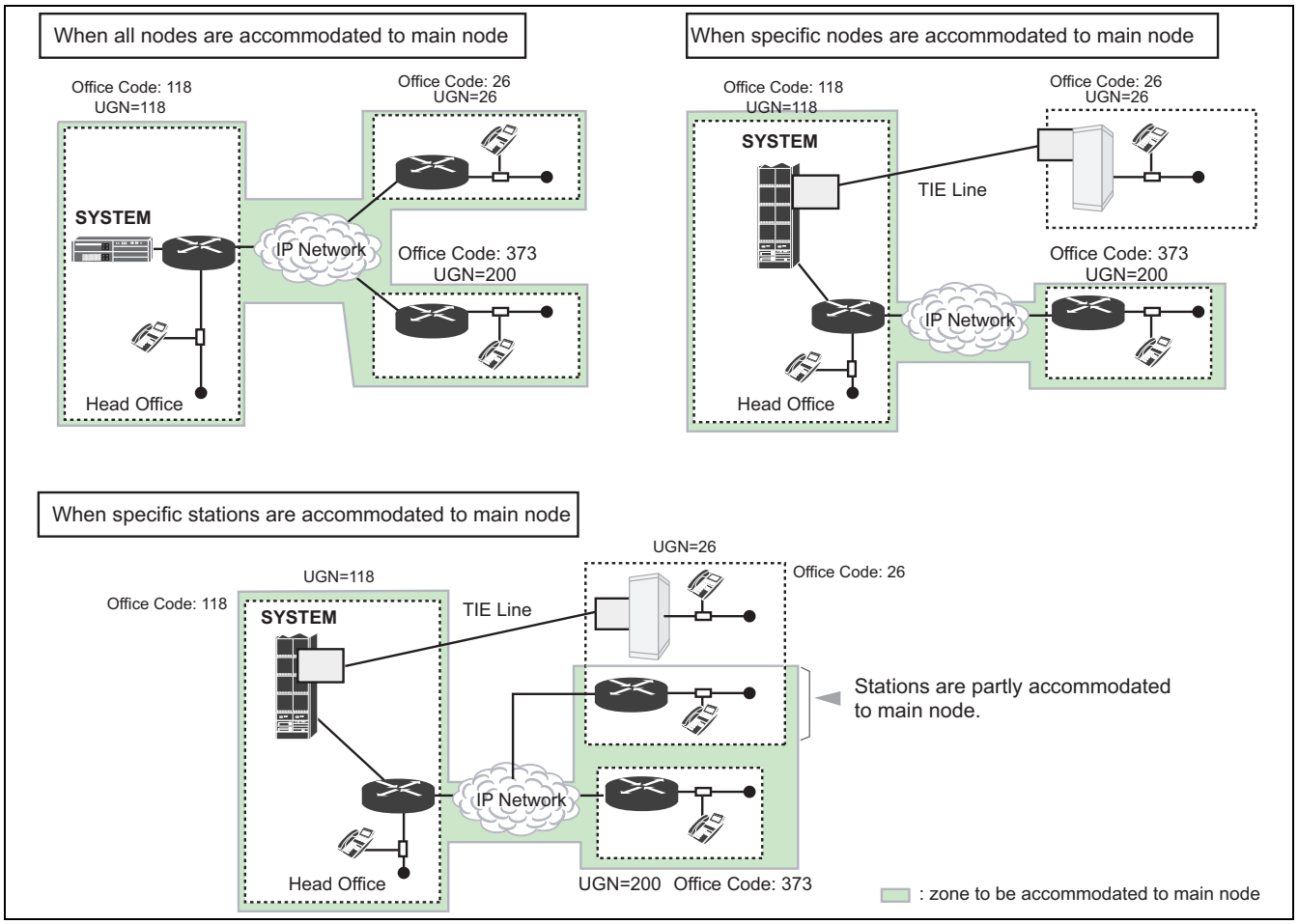


- Bit 0=1: When Calling Number has priority;



- If the numbers have more than 16 digits, only the first 16 digits are displayed.
- As for DT800/DT700 Series, refer to Name Display - System [N-28].

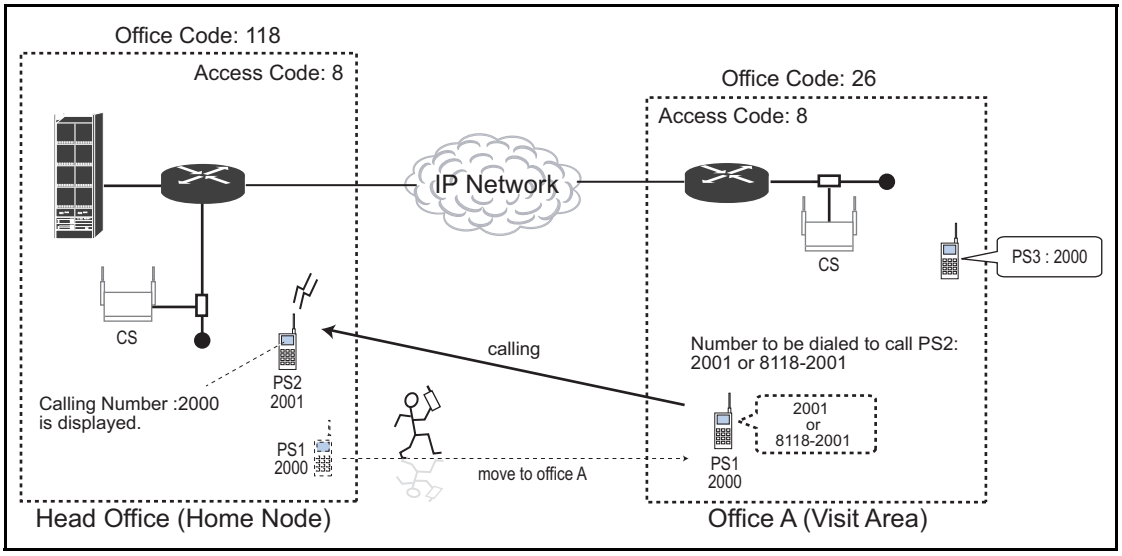
5. The following three types of system configuration are available.



6. An arbitrary number between 1 and 255 can be used as a UGN. Note that limitations may be imposed by the number of offices to be used after re-configuration and internal resources.
7. Tenant, Service Feature Restriction Class (SFC), and Route Restriction Class (RSC) can be used as usual. Separate/Common Tenant Data table is also available. Outgoing Call restriction, Incoming Call restriction, Service Feature restriction can be set on a tenant basis. These restrictions are applied on a node basis. Therefore, the Tenant Number must be separated on a node basis. Accordingly, a maximum of 62 nodes can be accommodated to the main node since the maximum tenant number is 63. (One tenant number is used at the main node. The remaining nodes are 62 nodes.)
8. Route Number ranges from 1 to 899.
9. In a configuration where TIE line(s) is remained after stations are partially accommodated to the main node, all digits (of Private Number) must be sent out from the call originating node. If the node sends out the Extension Number only, the call can not reach the destination. When all digits (of Private Number) cannot be sent out depending on the numbering plan, assign the Additional Digits by AISPN/AOSPN command for the terminating node to receive all digits (of Private Number).
10. Service Access Codes used for the previous node are not inherited to the newly configured network.
11. In login mode, the LCD displays “Unauthorized” if Extension Number only is entered at Login prompt.

12. This feature is not available for Hotel systems.
13. This feature is not available for ACD systems.
14. This feature is available for OAI systems. For details, refer to Data Programming Manual - OAI.
15. Conditions for PS movement are as follows:
    a.) When moving to a different User Group Number (UGN).

    - Even if a PS moves out the PHS/PCS home node, it follows the numbering plan of the home node.
    - For Nodes with Dial Plan to support IP Centrex Functionality, assign the same settings for CS System ID (SYS-ID).

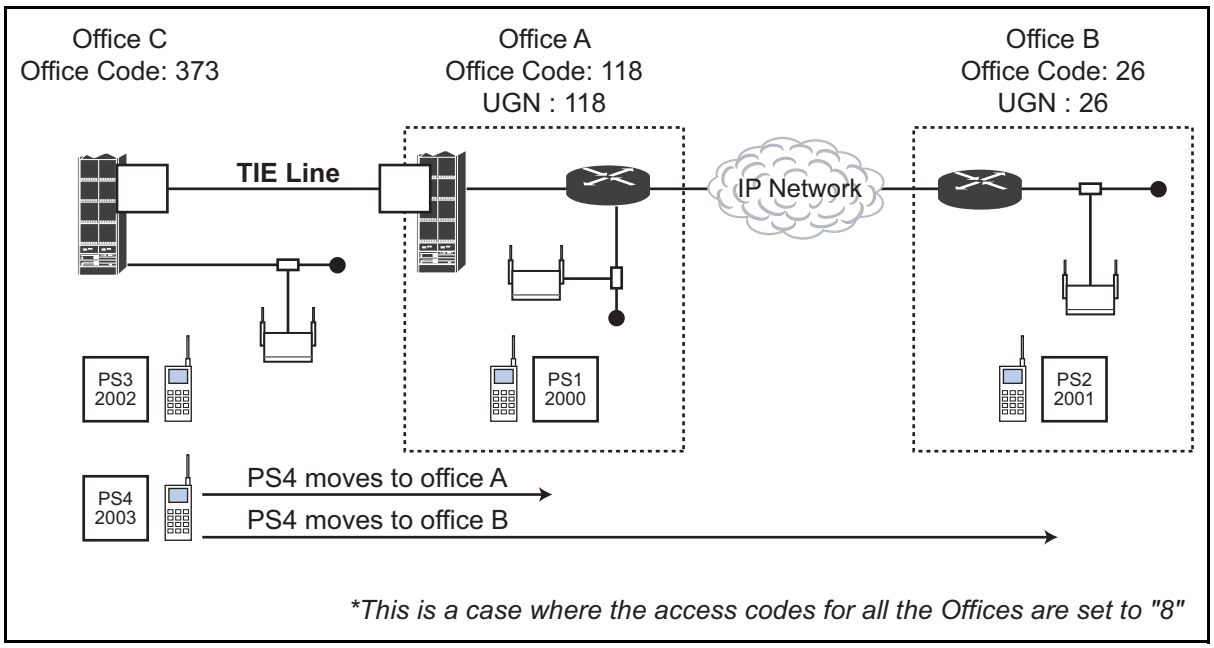


When PS1, which has CS of Head Office as its Home Node, moves to Office A:

Originating a call from PS1 to PS2	Dial Number	2001 or 81182001
	Calling number displayed on PS2	2000
Originating a call from PS1 to PS3	Dial Number	8262000
	Calling number displayed on PS3	81182000
Originating a call from PS2 to PS1	Dial Number	2000 or 81182000
	Calling number displayed on PS1	2001
Originating a call from PS3 to PS1	Dial Number	81182000
	Calling number displayed on PS1	8262000

b.) When a PS moves in a configuration where a TIE line is included.

When roaming across a TIE line, Roaming is only possible to only one User Group Number (UGN):



While it is possible to Roam to Office A, if PS4, which has Office C's CS as its Home Node, moves to Office A or Office B:

Originating a call from PS4 to PS1	Dial Number	2000 or 81182000
	Calling number displayed on PS1	83732003
Originating a call from PS4 to PS2	Dial Number	8262001
	Calling number displayed on PS2	83732003
Originating a call from PS4 to PS3	Dial Number	83732002
	Calling number displayed on PS3	83732003
Originating a call from PS1 to PS4	Dial Number	83732003
	Calling number displayed on PS4	81182000
Originating a call from PS2 to PS4	Dial Number	83732003
	Calling number displayed on PS4	8262001
Originating a call from PS3 to PS4	Dial Number	2003
	Calling number displayed on PS4	83732002

Note: Even when moving to Office B, it will be the same status as moving to Office A.

c.) For Nodes with Dial Plan to support IP Centrex Functionality, register the private number code of the Roaming destination Node (UGN) to the Roaming Self Node number. Roaming Self Node number is one setting per PHS community number (PCN). Since PCN is one setting per PBX, settings may not be made per Node (UGN).

d.) For Nodes with Dial Plan to support IP Centrex Functionality, set the Node for private number code (UGN) as the virtual station registered for Roaming PS (Visiting Area PS).

e.) For Nodes with Dial Plan to support IP Centrex Functionality, assign the same settings for CS System ID (SYS-ID). Also, SYS-ID must be different from the Node across a TIE line.

16. SMDR - When a node is accommodated into the main node, calls that were output as Outgoing/Incoming Calls in the previous configuration are output as Station-to-Station Calls.
17. MCI - UGN data is output in the UGN field of the extended format. Therefore, the external terminal requires to support the extended format.
18. SIP terminals (WLAN Handset (MH Series) and Standard SIP Terminal) that are accommodated in the System can use this feature under the following conditions.

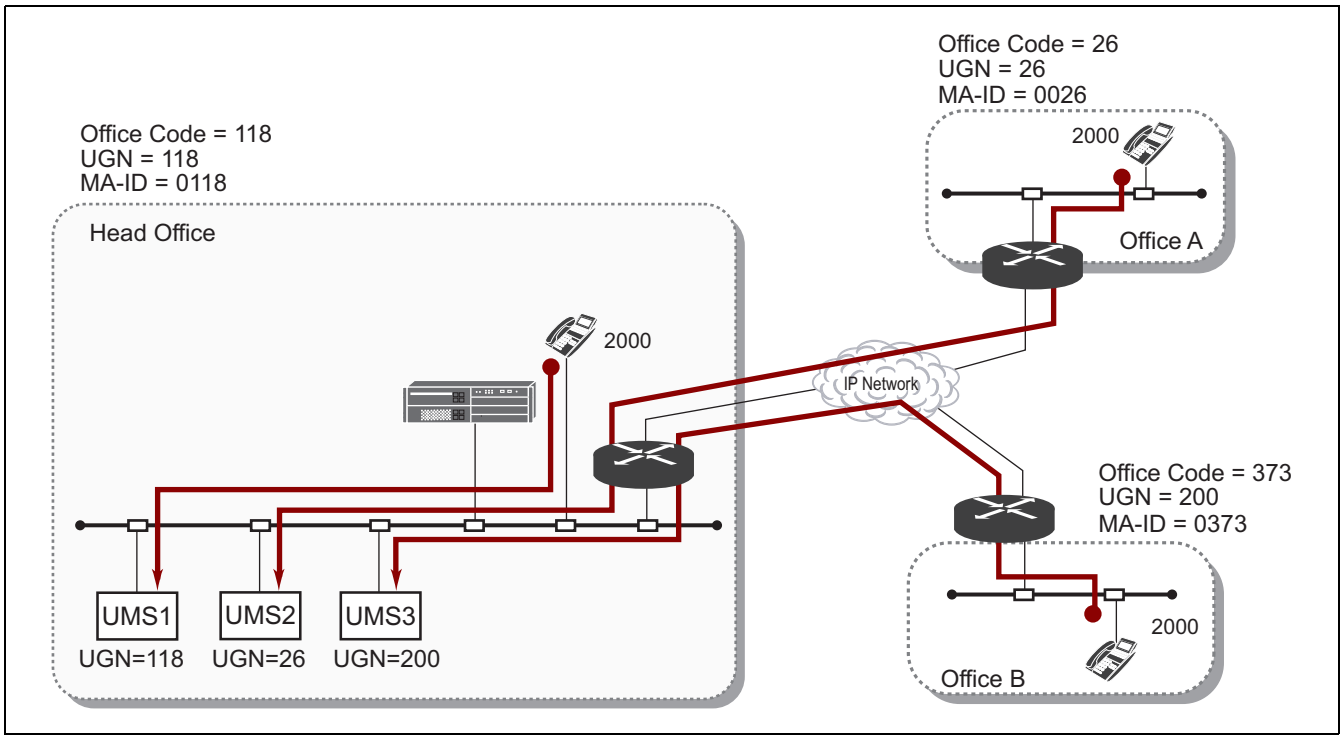
a.) The same Station number can be assigned to multiple SIP terminals accommodated in different nodes (User Group Numbers). For example, both SIP terminal A in Node B and SIP terminal C in Node D can use “2000” as their Station numbers.

b.) Dial Plan to support IP Centrex Functionality feature must be enabled.

c.) Each SIP terminal must be given a unique number as Private Number.

19. See the examples below when Unified Messaging System (UMS) is configured.

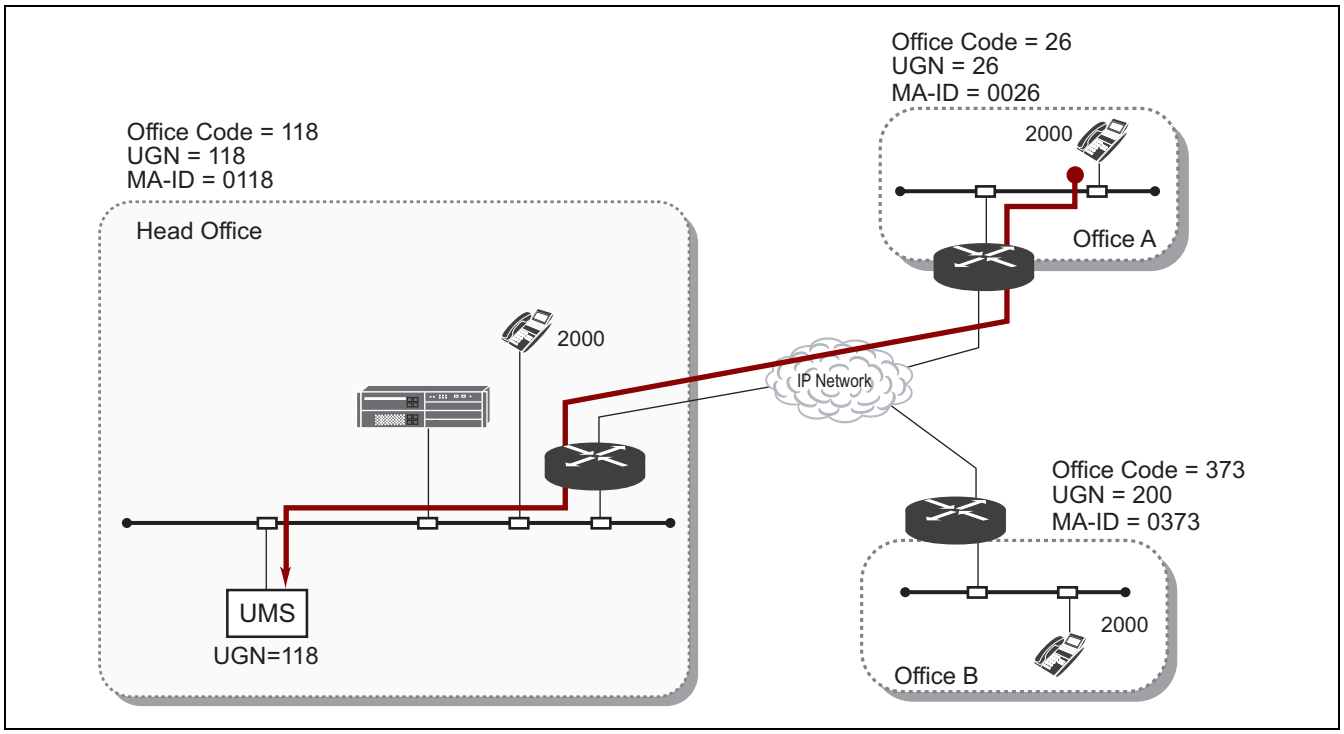
[When UMS is accommodated on each node]



- Extension Number is informed to each UMS from each office.
- Assign Extension Numbers when registering user IDs to UMS.

  <UMS user ID registration example>
  UMS1 (UGN=118) User ID: 2000 (UGN=118)
  UMS2 (UGN=26) User ID: 2000 (UGN=26)
  UMS3 (UGN=200) User ID: 2000 (UGN=200)

[When UMS is accommodated on Head Office]



- Private Number is informed from Office A/B to UMS, and Extension Number is informed in Head Office.
- Assign Extension Numbers when registering user IDs to UMS in Head Office (in the same UGN). Assign Private Numbers to the user IDs for Office A/B (in other UGN).

  <UMS user ID registration example>
  UMS1 (UGN=118)
  User ID: 2000 (UGN=118)
  User ID: 8262000 (UGN=26)
  User ID: 83732000 (UGN=200)

20. Conditions on caller identification number:

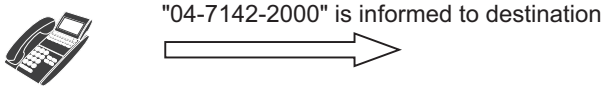
ARTDN		ARTIN	ACDDN (Note 2)	ACNDN	INDICATION
(INT)	(NET)	(INTD)			
4	1	–		×	ACNDN + Private Number Note 2
				–	Private Number
4 10	0 –	– 2	×	×	ACNDN + ACDDN

×: Required to assign

Note: Assign Private Number to “CDC (Change Digit Code)” parameter of ACDDN command.

Note 2: Private Number is displayed as caller number identification when INT=3 and NET=1 is assigned to the route class data of ISDN route.

<Example setting>
To inform “04-7142-2000” as the caller number for the route class of INT=3 and NET=1:



Extension Number: 2000
Private Number: 8118200

ACNPN: IC/OG = OG: Outgoing Call
LGRT = ISDN route
CNP = 1

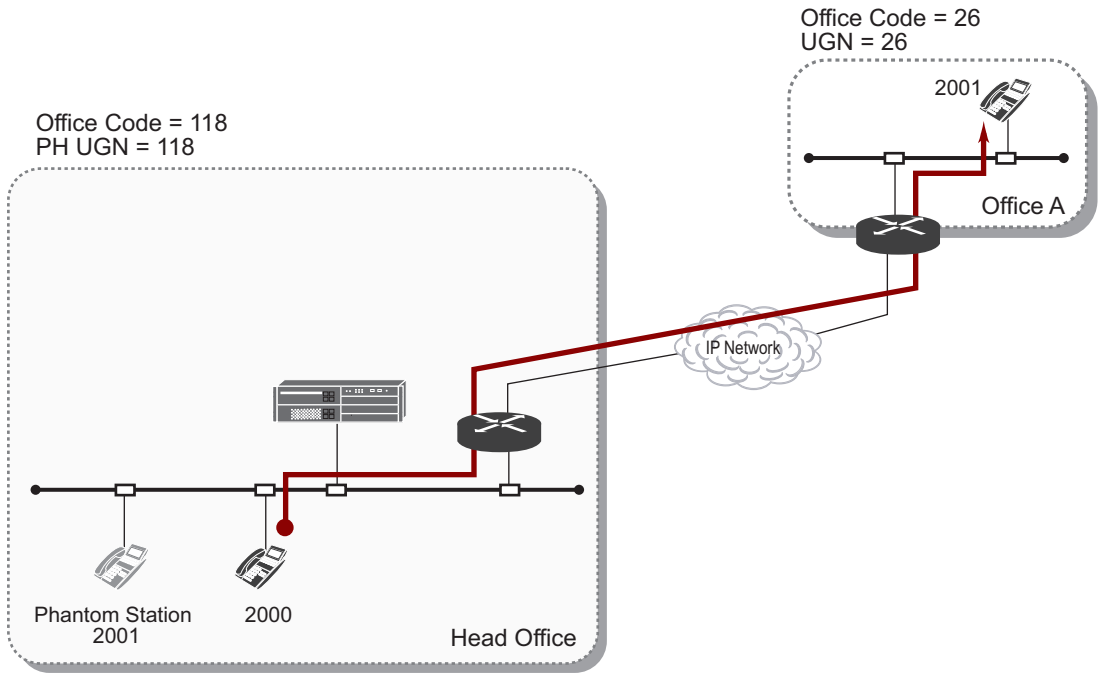
ACNDN: CNP =1
SKIP = 4
ADD = 6
DC = 047142

21. Phantom Number can be assigned across different UGNs. In the numbering development below, calls terminate to extensions in other UGN by dialing Extension Number only.

<Example setting>

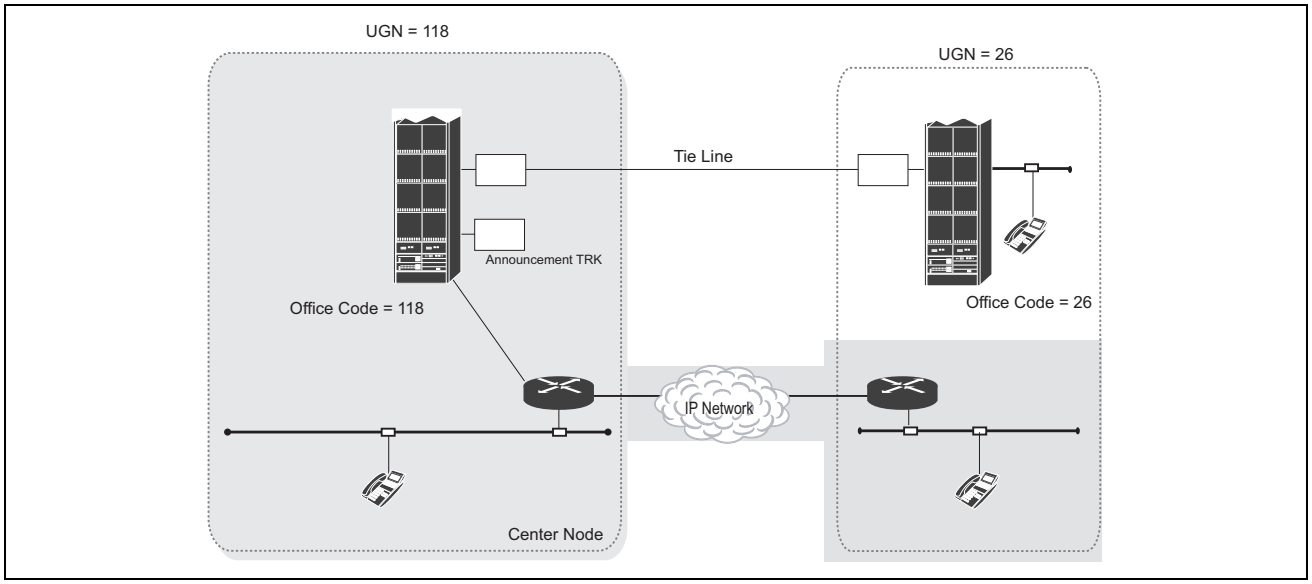
APHNN: UGN = 26
TELN = 2001
PH = 0
PH UGN = 118
PH TELN = 2001

When Extension 2000 dials "2001" in Head Office (PH UGN=118), the call terminates to Extension 2001 in Office A (UGN=26).

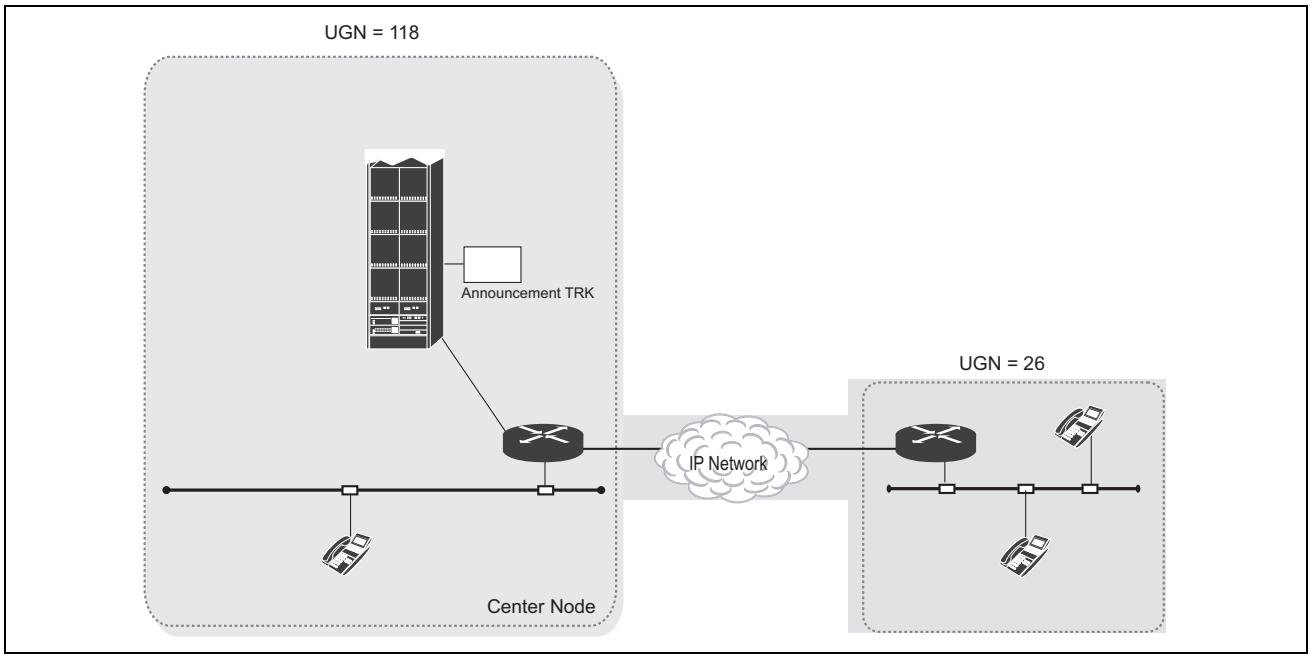


22. When using Unused Number/Dead Level Number-Announcement

When some station numbers are accommodated in a center node, Unused Number/Dead Level Number-Announcement feature the center node provides is not available.



When all the station numbers are accommodated in a center node, if all the data assigned on AFRS(L/N), AOPR(L/N), AUNE(L) are deleted, Unused Number/Dead Level Number-Announcement feature the center node provides will be available.



23. When you use PSs as stations, assign the PS data under the following.
    a.) When you assign Individual PS Numbers (IND-PS) and PS Station Numbers (PS NO) by the APSD command, use Private Numbers (Access Code + Office Code + Station Number) rather than Telephone Numbers because they must be unique in a system (among all nodes in a FCCS network).
    To use Individual PS Numbers with less than 5 digits, Individual PS Numbers must be mutually unique in a system. In that case, when you attempt to display your own telephone number on PS, some types of PSs may display the corresponding Individual PS Numbers instead of the Station Numbers.
    b.) If PHS Inter-Node Roaming is used, PS Station Numbers must be numbers which can receive a call from a station in Visitor Node when the PSs are in Home Node. If the PS moves to another node across a Tie Line, assign PS Station Numbers (PS NO) as Private Numbers (Self Node office code=Access Code + Office Code).

    Note: When you change Individual PS Numbers (IND-PS) and PS Station Numbers (PS NO) by the APSD command, you need to download the new data to PS again.

24. By changing the system data, you can reduce the amount of registered data for the AFRSN command, and the office data consumption. For details, see 2.3 Reduction of Data Consumption of the AFRSN command, CHAPTER 5 DATA PROGRAMMING of Data Programming Manual - FCCS.

25. The following tables show service feature availability and conditions for this feature.

Note: When using Dial Plan to support IP Centrex Functionality [D-166], User Group Number (UGN) can be assigned in the range of 1 to 255.

OG Call Features Availability

SERVICE FEATURE NAME	Provided	MAXIMUM	REMARKS
Direct Outward dialing	×		
Restriction from Outgoing Calls	×		
Toll Denial/Toll Diversion	×		
Toll Restriction - 3/6-Digit	×		
Speed Calling - Station	×	A maximum of 1400 blocks can be stored per system. (One block can be used for 10 numbers.)	AEIA command is assigned by station number.
Speed Calling - System	×	A maximum of 10000 codes can be stored per system.	
Outgoing Trunk Queuing	×		
Off-Hook Queuing	×		
Alternate Routing	×		
Pushbutton-To-Rotary Conversion	×		

×: Available N: Not Available -: Not Applicable C: Conditionally Available

IC Call Features Availability

SERVICE FEATURE NAME	Provided	MAXIMUM	REMARKS
Attendant Console	–		
Direct-In Termination (DIT)	×		
Direct Inward Dialing	×		
Remote Access to System	×		Additional Number is dialed by Private Number.
Night Connection - Fixed	–		
Music-On-Hold	×		
Multiple Music-On-Hold	×	A maximum of 16 music sources can be assigned.	
Distinctive Ringing	×		
Tenant Service	×	A maximum of 63 tenants can be used per system.	
Slumber Time - Do Not Disturb	×		

×: Available N: Not Available -: Not Applicable C: Conditionally Available

Station Service Features Availability

SERVICE FEATURE NAME	Provided	MAXIMUM	REMARKS
Station Hunting	×	Station Number: 100 Stations/Group Telephone Number: 100 Stations/ Group There is no limitation for the number of Groups. Number of Secretaries Terminal: 255 Circular: 255 Priority: 4096 Priority with Switchback: 4096	Cannot be set over UGNs.
Call Pickup	×	Station Number: 100 Stations/Group Telephone Number: 100 Stations/ Group There is no limitation for the number of Groups.	Cannot be set over UGNs.
Selectable Station Call Pickup	×		Cannot be set over UGNs.
Call Pickup - Direct	×		Cannot be set over UGNs.
Step Call	×		
Call Back	×		
Automatic Call Back Cancel	×		
Call Waiting - Terminating / Originating	×		
Executive Right of Way	×		
Privacy	×		
Call Forwarding - All Calls	×		Can be set over UGNs.
Call Forwarding - Busy Line	×		Can be set over UGNs.
Call Forwarding - Don’t Answer	×		Can be set over UGNs.
Call Forwarding - All Calls - Outside	×		Can be set over UGNs.
Call Forwarding - Busy Line - Outside	×		Can be set over UGNs.
Call Forwarding - Don’t Answer - Outside	×		Can be set over UGNs.
Split Call Forwarding	×		The incoming call is transferred to the target station for station calling when this feature has been set over UGNs.
Call Forwarding All Call - Cancel by Routine Diagnosis	×		
SPDT is sent out when lifting the handset of terminal on which C.F. - All Call has been set.	×		
Call Forwarding - Busy Line - Override	×		

Call Forwarding - I'm Here	×		Cannot be set over UGNs.
Remote Call Forwarding Control	N		
Call Transfer - All Calls	×		
Camp On by Station	×		Can be set over UGNs.
Boss - Secretary	×		
Three - Way Calling	×		
Station - Controlled Conference / Preset Conference	×		
Call Hold	×		
Recall	×		
Exclusive Hold	×		
Hot Line	×		
Priority Call	×		
Station-to-Station Calling	×		
Class of Service - Individual	×		
Day/Night Class of Service	×		
Timed Reminder / Automatic Wake-Up	×		
Call Forwarding - All Calls - Announcement	×		
Howler Tone Sending	×		
Line Lockout	×		
Off-Hook Alarm	×		
Periodic Time Indication Tone	×		
Immediate Ringing	×		
Pushbutton Calling	×		
Number Portability	×		
Flexible Numbering of Stations	×		
Line Circuit Reverse Relay Control	×		
Emergency Call	×		
Station 5 dB PAD	×		
One Digit Hooking	×		
Dual Station Call	×		Cannot be set over UGNs.
Number Sharing	×		Cannot be set over UGNs.
Last Number Call	×		

×: Available N: Not Available -: Not Applicable C: Conditionally Available

Other Service Features Availability

SERVICE FEATURE NAME	Provided	MAXIMUM	REMARKS
Call Park	×		Cannot be set over UGNs.
Uniform Call Distribution (UCD)	×		Cannot be set over UGNs.
Delay Announcement - UCD	×		Cannot be set over UGNs.
Long Line Station	×		
Announcement Service	×		
Paging Transfer	×	Maximum 6 routes can be used for paging trunk.	
Radio Paging	×	A maximum of 20 calls can be placed in queue for Paging/Paging Transfer/ Radio Paging per system. Maximum 6 routes can be used for radio paging trunk.	
Unused Number/Dead Level Number- Announcement	×		Some Restrictions (Refer to Service Conditions.)
Router Restriction - Announcement	×		
Direct Station Selection (DSS) Console	×		
Authorization Code	×		
Multiple Music on Hold Source	×		
Account Code	×		
Message Center Interface (MCI)	×		

×: Available N: Not Available -: Not Applicable C: Conditionally Available

Dterm Service Features Availability

SERVICE FEATURE NAME	Provided	MAXIMUM	REMARKS
Multiple Line Operation -D	×		Must be in the same server.
Prime Line Pickup -D	×		
Soft key	×		
Non-square Line Assignment -D	×		
Hands-free Answer Back -D	×		
Intercom Calling	×		Must be in the same server.
Voice Call -D	×		Can be set over UGNs.

Internal Zone Paging	×	A maximum of 99 Internal Zone Paging groups can be set per tenant. A maximum of 24 terminals can be used per group. A maximum of 255 groups can be used per system. A maximum of four terminals can be set in one group (Level 0 - Level 7).	Cannot be set over UGNs.
Manual Signaling	–		
Hands-free Dialing/Monitoring -D	×		
Selective Dialing	×		
Last Number Call -D	×		
Speed Calling - One Touch key with Page Changeover	×		
Brokerage Hot Line	×		
Pause Entry	×		
Save and Repeat -D	×		
Off-Hook Line Number Display -D	×		
Name display - System	×		Can be set over UGNs.
Intermediate Station Number Display -D	×		
Elapsed Time Display -D	×		Cannot be displayed for a call over UGNs.
Service Display -D	×		
Time Display -D	×		
Called Station Status Display -D	×		
Message Reminder -D	×		
Hands Free Call	×		
Ringer Mute	×		
Side Tone On/Off -D	×		
Call Indicator Lamp	×		
Serial Call -D	×		
Exclusive Hold -D	×		
Non-Exclusive Hold -D	×		
Call Park Group -D	×		
Speed Calling - Station	×		
Flash Button -D	×		
Do Not Disturb -D	×		
Software Multiple My Line Appearance	×		

×: Available N: Not Available -: Not Applicable C: Conditionally Available

Network Service Feature Availability

SERVICE FEATURE NAME	Provided	MAXIMUM	REMARKS
Tandem Switching of TIE Trunks - 2/4-Wire	×		
Inter-Office Off-Hook Queuing	×		
TIE Line Access	×		
Outgoing Trunk Busy Announcement	×		
Hotline	×		
Automatic Circuit Assurance	×		
Least Cost Routing-3/6-Digit	×		
Centrex Compatibility	×		
Echo Canceler Control	×		

×: Available N: Not Available -: Not Applicable C: Conditionally Available

CCIS Service Feature Availability

SERVICE FEATURE NAME	Provided	REMARKS
Voice Call - CCIS	C	A maximum of eight-digit Private Number is available.
Name Display - OAI - CCIS / Name Display - System - CCIS	×	
Dial Access To Attendant - CCIS	–	
Call Forwarding - All Calls - CCIS	×	
Call Forwarding - Busy Line - CCIS	×	
Call Forwarding - Don't Answer - CCIS	×	
Supervisory Call - CCIS	C	A maximum of eight-digit Private Number is available.
Call Back - CCIS	×	
Direct - In Termination - CCIS	×	
Step Call - CCIS	×	
Call Transfer - All Calls - CCIS	×	
Non - Delay Operation - CCIS	–	
Distinctive Ringing - CCIS	×	
Do Not Disturb - CCIS	×	
Hot Line - CCIS	×	
Busy Verification - CCIS	C	A maximum of eight-digit Private Number is available.
Brokerage Hot Line - CCIS	×	
Individual Attendant Access - CCIS	–	
Attendant Camp-On With Tone Indication - CCIS	C	A maximum of eight-digit Private Number is available.
Serial Call - CCIS	C	A maximum of eight-digit Private Number is available.
Step Call - CCIS	×	
Executive Right Of Way - CCIS	C	A maximum of eight-digit Private Number is available.

Centralized System Management - Report - CCIS	×	
Digital Display - Station - CCIS	×	Indicated by Private Number
Call Transfer - Attendant - CCIS	×	
Data Privacy On Demand - CCIS	×	
Centralized System Management - Report - CCIS	×	
Calling/Called Number Display - CCIS	×	Indicated by Private Number
Message Reminder - CCIS	C	A maximum of eight-digit Private Number is available.

×: Available N: Not Available -: Not Applicable C: Conditionally Available

ISDN Gateway Service Feature Availability

SERVICE FEATURE NAME	Provided	REMARKS
PRI/PRA Failsafe Routing	×	
Alternate Routing - PRI	×	
DID Addressing	×	
Calling Party Recognition Service	×	
H0 Switching	×	
Sub Address - Addressing	×	
SID to Terminating User - DTE	×	
Sub Address - Present	×	
SID To Network - Present	×	
SID To Network - Privacy	×	
Advice Of Charge (AOC) - Receipt And Display Of AOC From A Foreign Q.SIG Network	×	
Calling Party Recognition Service (C.F.-All Calls/Busy Line Don't Answer)	N	
Q.SIG/ISDN International Gateway Switching	×	
D Channel Packet Communication	–	

×: Available N: Not Available -: Not Applicable C: Conditionally Available

ISDN Supplementary Service Feature Availability

SERVICE FEATURE NAME	Provided	REMARKS
Q.SIG Look Ahead Routing (ETSI ETS 300-172/ISO-11572)	×	
Call Transfer Supplementary Service (SS-CT) With Rerouting	×	
Call Forwarding Supplementary Service ((SS-CF) With Rerouting	×	

Call Completion To Busy Subscriber (CCBS)	×	
Call Completion On Non Reply (CCNR)	×	
Call Transfer Supplementary Service (SS-CF) With Rerouting	×	
Event Based CCIS - ISDN Transport	N	
Event Based CCIS - Q.SIG Transport	N	
Virtual TIE Line	–	
Calling Party Information Transfer	×	
Transfer Message (TRM)	×	
ISDN over High-speed Digital Line Support	×	

×: Available N: Not Available -: Not Applicable C: Conditionally Available

PHS/PCS/Wireless Service Feature Availability

SERVICE FEATURE NAME	Provided	REMARKS
Data Communication	×	
Roaming	×	The virtual station assigned for Roaming PS is fixed to UGN=1. When a visitor PS calls to the station in visitor node, dial the Private Number (not only the Extension Number).
PS Location Information	–	
PS Location Information Display - PS/Dterm	×	
Group Call - Automatic Conference (20-Party)	×	
Automatic Announcement - Disconnected PS Out Of Zone	×	
Automatic Announcement - Called PS Out of Zone	×	
Automatic Announcement - Called PS Incoming Incomplete	×	Call Forwarding-All Calls/Busy cannot be assigned across the UGN on the forwarded destination station of incompleted call.
Name Display	×	
Two-Line Operation	×	
Call Park Group (PS)	×	
PS L2-Lines Termination	–	
PS Line Appearance on Dterm	×	
Short Text Message Service (STM Service)	×	
Wireless Adjunct Connection	×	

×: Available N: Not Available -: Not Applicable C: Conditionally Available

IP Service Feature Availability

SERVICE FEATURE NAME	Provided	REMARKS
CCIS Networking via IP	×	
FCCS Networking via IP	×	
Calling Number Display depend on Access Code?	×	
DtermIP/IP Enabled Dterm (Peer-to-peer)	×	
Logged Out IP Station -Call Destination	×	
Peer to Peer Bandwidth Control	×	
VLAN	×	
IP Control - FAX	×	Must be in the same server.
Survivable Remote MGC	×	
Least Cost Routing for MG-PRI	×	
LAN Redundancy	×	
IP Terminal Automatic Assignment	N	
Soft Phone Wireless Handset	–	Note that D-166 is available between Softphones.
Softphone Wired Phone Handset	–	

×: Available N: Not Available -: Not Applicable C: Conditionally Available

Programming

[Data for Main Node]

STEP 1: **ASYDN** - Assign SYS1, Index 801.
Index 801
Bit 2: AUGNN tenant data table development is Separate / Common = 0/1

STEP 2: **ASYDL/ASYDN** - Assign System Data 1.
Index 514/Index 957
Bit 0 to Bit 7: Network Data Memory (NDM) usage
Index 672 **Note 3**
Bit 0: Location of Telephone Number displayed on Desktop terminal
0/1 = 8-digit Extension Number Display on the first line of Desktop terminal/16-digit Extension Number Display on the second line of Desktop terminal
Index 800
Bit 0=0: ALGNL/ALGNN, ALGSL/ALGSN commands tenant UGN table development is Separate.
Bit 1: ASPAL/ASPAN command tenant data table development is Separate / Common = 0/1
Bit 2: ANPDL/ANPDN command tenant data table development is Separate / Common = 0/1
Index 801
Bit 1: AMNDL/AMNDN command tenant data table development is Separate / Common = 0/1
Index 818
Bit 5=1: Dial Plan to support IP Centrex Functionality is in service.
Bit 3: 16-digit Dial Plan Support for IP Centrex Functionality **Note 3**
0/1 = In Service/Out of Service

STEP 3: **ANPDN** - Reserve a Number Level for Private Number.

STEP 4: **ASPAN** - Assign an Access Code to the Private Number. SRV = "LCR: Least Cost Routing", LGRT = dummy route.

STEP 5: **AUGNN** - Assign a UGN to NPC.

TN: Tenant Number
LGRT: Logical Route Number (1-899) (Assign the dummy route specified in STEP 3.)
NPC: Number Pattern Code (Maximum 16 digits)
UGN: User Group Number (1-255)
PVNND: Private Number Necessary Digits (2-8) **Note 4**, **Note 5**

STEP 6: **AUPVN** - Assign Private Number to the assigned UGN.

UGN: User Group Number
PRVDC: Private Digit Code (Maximum 8 digits) For example, 8 + 95, Private Digit Code is 895. **Note 7**

STEP 7: **ARTDN** - Assign the number of digits for using as an Extension Number for the ISDN Bch route by using CDN 66. The number of digits is counted from the last digit of the received digits. This DC value corresponds to the DC parameter of ACDDN command in the next step.

CDN 66: Dialed Number Confirmation

STEP 8: **ACDDN** - Assign Change Digit Code to change the received digits to adequate number that conforms to the numbering plan of the main node.

D/N:	Day/Night
LGRT:	Logical Route Number (1-899)
FPC:	Fusion Point Code
RT:	Route Number
DC:	Digit Code (Maximum 8 digits)
CDC:	Change Digit Code (Maximum 16 digits) **Note 3**
AD:	Additional Dialing
XFR:	Transfer Service

STEP 9: **AMNDN** - Assign the maximum necessary digits for Private Number.

TN:	Tenant Number
DC:	Destination Code. Enter Access Code + Office Code. (Maximum 10 digits)
MA-ID:	Message Area ID. AMNDL only. (0-9999)
AREA NAME:	Area Name (Maximum 16 digits)
MND:	Maximum Necessary Digit (1-32)
TOLL:	Toll Code Identification (Local Code/Toll Code)
AN:	Optional Service Restriction Data (Out of service/Account code)
RATE:	Optional Service Data. Fixed to 0.
A/D:	Analog/Digital

Note 3: Set these data to either enable or disable the use and display of 9-16 digits for Private Number.

Note 4: Assign the number of digits for Access Code + Office Code.) For example, 8 + 95 = 3 digits, assign 3.

Note 5: The maximum eight digits can be assigned for PVNND and PRVDC. In the following cases, the maximum six digits can be assigned for PVNND and PRVDC.

- ASYDN, SYS1, Index 818, Bit 3=1 (16-digit Dial Plan Support for IP Centrex Functionality is out of service.)

[Data for Node to be accommodated to Main Node]

STEP 1: **ANPDN** - Reserve a Number Level for Private Number.

STEP 2: **ASPAN** - Assign an Access Code to the Private Number. SRV = "LCR: Least Cost Routing", LGRT = dummy route.

STEP 3: **AUGNN** - Assign a UGN to NPC.

TN:Tenant Number
LGRT:Logical Router (1-899) (Assign the dummy route specified in STEP 2.)
NPC:Number Pattern Code (Maximum 16 digits)
UGN:User Group Number (1-255)
PVNND:Private Number Necessary Digits (2-8) **Note 6**, **Note 7**

[Reference]
When Access Code = 8, Office Code = 95, Extension Number = 2×××
NPC = 8952, UGN = 20, PVNND = 3

STEP 4: **AUPVN** - Assign Private Number to the assigned UGN.

UGN: User Group Number (1-255)
PRVDC: Private Digit Code (Maximum 8 digits) **Note 7**

[Reference]
When Access Code = 8, Office Code = 95, Extension Number = 2×××
UGN = 20, PRVDC = 895

STEP 5: **AISTL** - Assign IP Station data.

TN: Tenant Number (1-63)
IP STN: IP Station Number (Maximum 5 digits)
KIND: Kine of IP Station
TEC: Telephone Class (0-31)
RSC: Route Restriction Class (0-15)
SFC: Service Feature Restriction Class (0-15)
LENS: Line Equipment Numbers

STEP 6: **ALGSN** - Assign Telephone Station data to LENS/PHIN or Station Number.

TYPE = 1 (Line Equipment Numbers/Phantom Index is used for specifying Telephone Number)
UGN: User Group Number (1-255)
TELN: Telephone Number (Maximum 16 digits)
FPC: Fusion Point Code (1-253)
LENS/PHIN: Line Equipment Numbers/Phantom Index

TYPE = 2 (Station Number is used for specifying Telephone Number)
UGN: User Group Number (1-255)
TELN: Telephone Number (Maximum 16 digits)
TN: Tenant Number (1-255)
STN: Station Number

STEP 7: **ARTDN** - Assign the number of digits that is used as an Extension Number for the ISDN Bch route by using CDN 66. The number of digits is counted from the last digit of the received digits. This DC value corresponds to the DC parameter of ACDDN command in the next step.

CDN 66: Dialed Number Confirmation

STEP 8: **ACDDN** - Assign Change Digit Code to change the received digits to adequate number that conforms to the numbering plan of the main node.

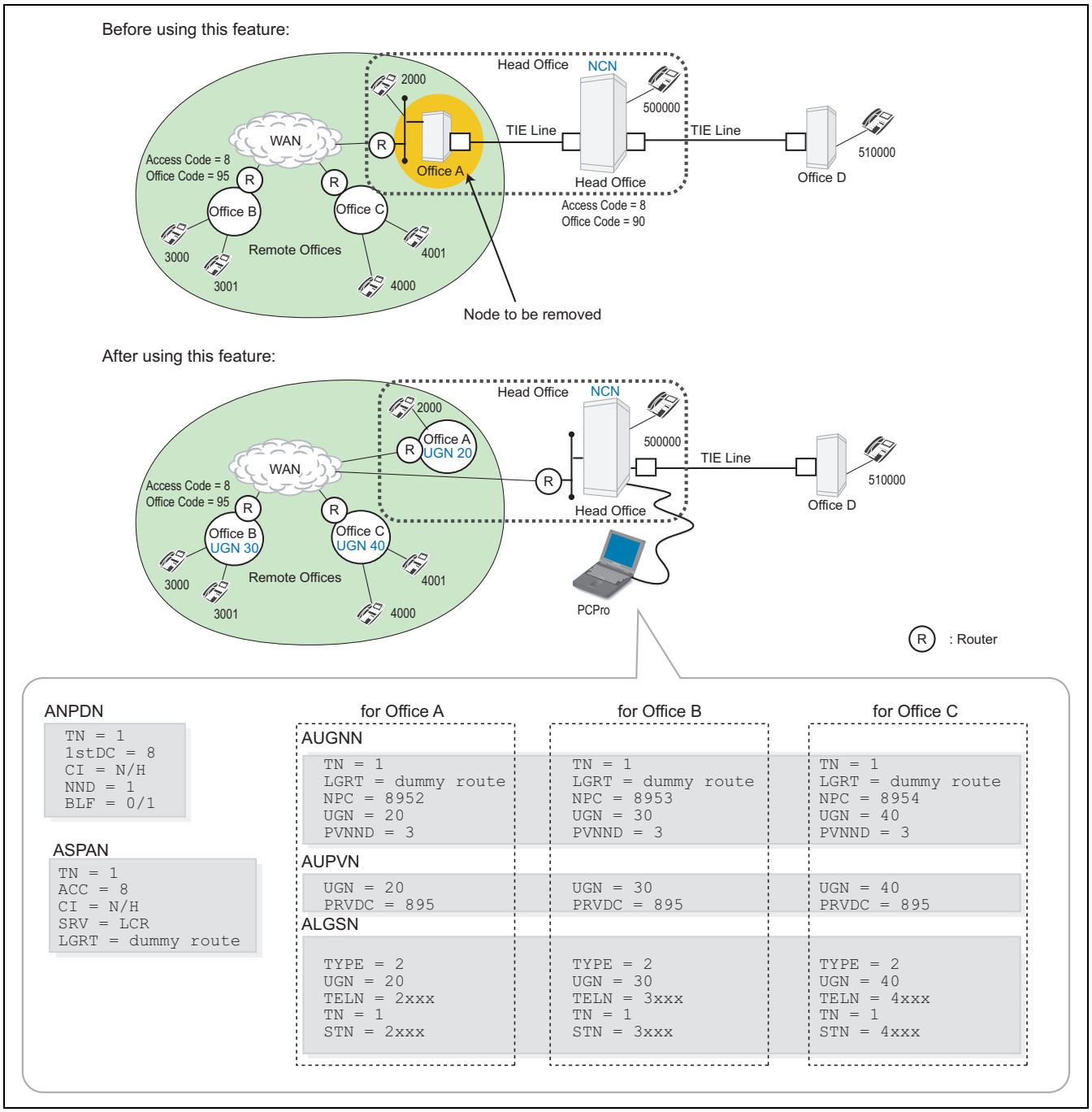
D/N: Day/Night
LGRT: Logical Route Number (1-899)
FPC: Fusion Point Code
RT: Route Number
DC: Digit Code (Maximum 8 digits)
CDC: Change Digit Code (Maximum 16 digits) **Note 6**
AD: Additional Dialing
XFR: Transfer Service

Note 6: Assign the number of digits for Access Code + Office Code.

Note 7: The maximum eight digits can be assigned for PVNND and PRVDC. In the following cases, the maximum six digits can be assigned for PVNND and PRVDC.
- ASYDN, SYS1, Index 818, Bit 3=1 (16-digit Dial Plan Support for IP Centrex Functionality is out of service.)

[Sample Office Data Assignment]

The following shows a sample data sheet for this particular network.



D-168 Distinctive Ringing - PBC

General Description

This feature allows the system to provide distinctive ringing patterns for EMEA Type system so that EMEA Type system users can distinguish between incoming trunk calls and station calls.

Note: This feature is only available for EMEA Type system.

Operating Procedure

No manual operation is required.

Service Conditions

1. The following Ringer Patterns are applied by ASYD, SYS1, Index 385, Bit 6 data setting.
   a.) Ringer Pattern 0 - Ringing for incoming trunk calls:
   0.25 seconds-ON, 0.25 seconds-OFF, 0.25 seconds-ON, 4.25 seconds-OFF
   b.) Ringer Pattern 1 - Ringing for station calls:
   1 second-ON, 4 seconds-OFF
   c.) Ringer Pattern 2 - Ringing for No Answer Timer for Call Hold:
   0.25 seconds-ON, 0.125 seconds-OFF, 0.25 seconds-ON, 0.125 seconds-OFF, 0.25 seconds-ON, 2 seconds-OFF
   d.) Ringer Pattern 3, 4 - Not used for EMEA Type system
   e.) Ringer Pattern 5 - Ringing for Call Back, Outgoing Trunk Queuing, Call Waiting, and Automatic/Dial Intercom:
   0.25 seconds-ON, 0.125 seconds-OFF, 0.25 seconds-ON, 0.125 seconds-OFF, 0.25 seconds-ON, 2 seconds-OFF
   f.) Ringer Pattern 6, 7 - Distinctive Ringing for Caller ID:
   1 second-ON, 2 seconds-OFF
2. ON/OFF timing of each Ringer Pattern is fixed (ON/OFF timing cannot be changed by ASYD, SYS3, Index 0, Index 1, Index 3, Index 7, Index 8 and Index 9 data setting).
3. To activate the system data (ASYD, SYS1, Index 385, Bit 6), initialize the corresponding ELC card and re-register the terminal accommodated on SP-PHD.
4. When multiple incoming calls are simultaneously terminated to the Multi Line on the Desktop terminal, the priority of Ringer Pattern is as follows:

Priority	1st	2nd	3rd	4th	5th	6th
Ringer Pattern	RG 7	RG 6	RG 5	RG 2	RG 0	RG 1

5. To activate this feature with PA-16ELCJ-B, the following firmware is required.

Equipment	Firmware
PA-16ELCJ-B	SP-3873 16ELCJB PROG-J Issue 1 or later

6. When Tone five, six, eight, nine or ten is specified on the Desktop terminal, Ringer Patterns provided by this feature become invalid. The tone can be changed by pressing the Feature + 3.

7. Only the following IP terminals are available for this feature.

   - DtermIP (Dterm Series *i*)
   - DtermIP INASET

   Note: Tones five to ten are only available with Dterm Series i. There will be only four tones for previous Dterm series and IP Enabled Dterms.

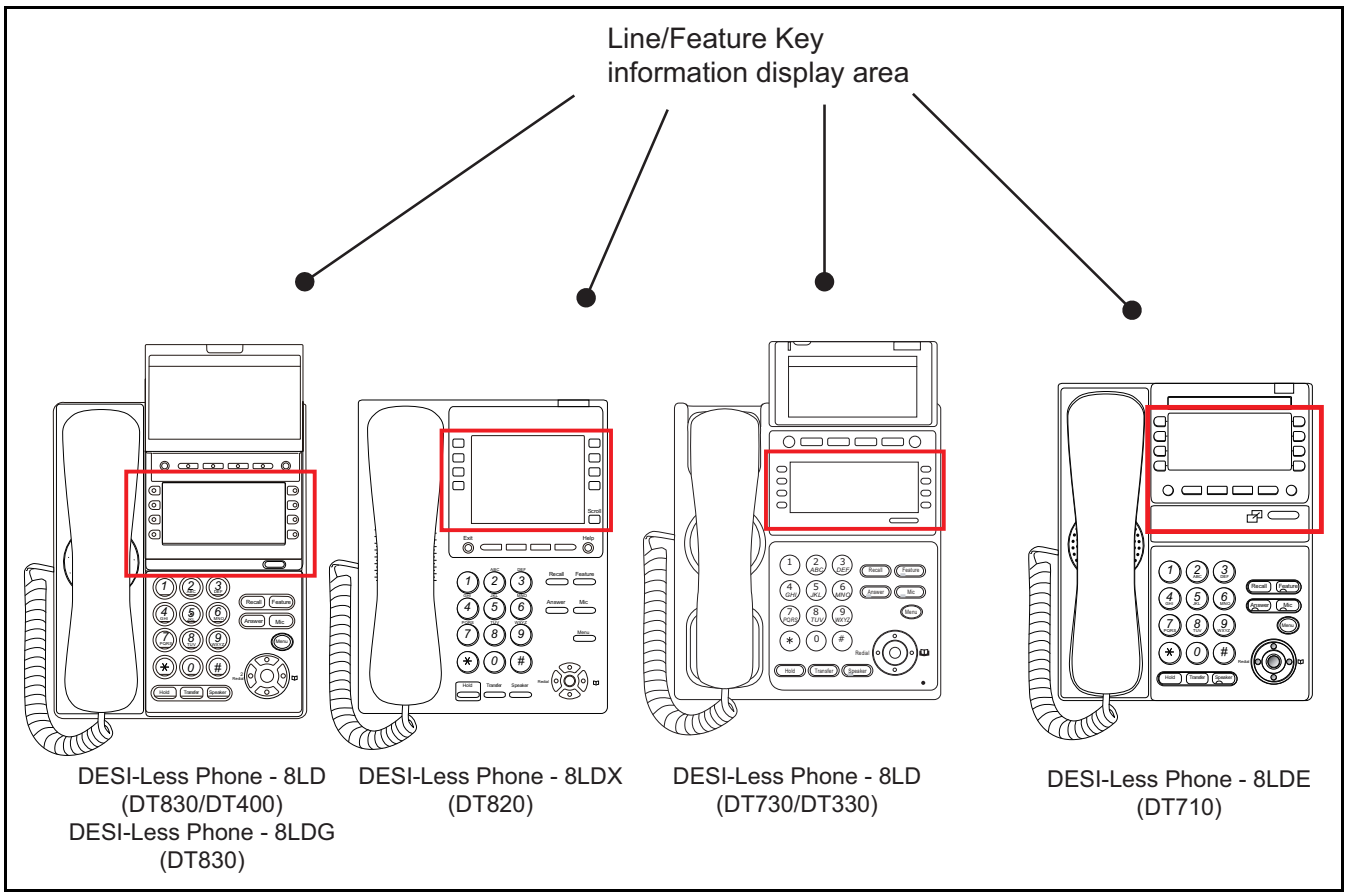
Programming

ASYD - SYS1, Index 385, Bit 6. Ringer Pattern for EMEA Type system. 0/1: Out of service/In service.

D-169 DESI-Less Terminal

General Description

DESI-Less Phone refers to Desktop terminals with Line/Feature key information display area. The information assigned on each Line/Feature key is displayed on the corresponding LCD.



Operating Procedure

Refer to the User's Guide or Operation Manual of each terminal.

Service Conditions

1. When Line/Feature key data is changed by PCPro, periodic refresh for Line/Feature key information display area is required. By assigning ASYDL, SYS1, Index 674, Bit 5=0, periodic refresh is executed every 90 seconds.

   Note: When periodic refresh is out of service (ASYDL, SYS1, Index 674, Bit 5=1), even when the display information is changed by ANDD or ALGSL/ALGSN commands, the displayed information on the LCD is not refreshed automatically. Execute one of the following to update the display:
   - Pull out and push in the modular/LAN cable of the terminal

- Re-login
- Initialize the ELC card on which the terminal is accommodated
- Initialize the system

2. System data (ASYD, SYS1, Index 78, Bit 4 and Index 389, Bit 0 to Bit 4) and ASLG command setting can change the language to be displayed on the LCD.

Note: Refer to [M-117] Multilingual Display for the data assignment of ASLG command.

3. The table below shows the number of characters to be displayed on the LCD of each key:

Line/Feature key information	Maximum number of characters to be displayed **Note 1**
Line key (Information about name and number)	8 characters
SPD key (Information about name and number)	8 characters
Feature key (Feature name of software key)	11 characters

Note 1: For the terminals other than DT800/DT700/DT400/DT300 Series, the maximum number of characters will be 6. Also, when the feature name is changed by Line key, SPD key, or AFDD command, the maximum number of characters will be 6.

4. When the number of characters exceeds the limit, system data (ASYD, SYS1, Index 241, Bit 6) can change the display option. **Note 2**
5. When a line is assigned on any Line/Feature key, the displayed information on Line/Feature key information display area can be chosen by the system data (ASYDL, SYS1, Index 673, Bit 5) to show name or number. However, when name information is not assigned, Telephone Number (if registered) or Station Number is displayed.
6. When a feature is assigned on any Line/Feature key, the same feature name with software key is displayed. When no feature is assigned, the FKY code is displayed. In both cases, you can modify the feature name by AFDD/AFDDN command. **Note 2**

Note 2: When periodic refresh is in service (ASYDL, SYS1, Index 674, Bit 5=0), the display is updated in about a minute after changing the office data.

7. Icons are displayed on the LCD corresponding to each Line/Feature key. They work according to the LED control of Line/Feature key.
8. Only for DESI-Less terminals in DT830/DT730/DT430/DT330, an icon for page scrolling is provided. Page icon moves you forward through pages to cover total 32 features/lines corresponding to 8LD/8LDE/8LDG per page.
9. Using ASFC/ASFCN command (SFI=264), Line/Feature key information display area can be fixed to a single page with scrolling disabled.

Note: Available since FP95-115 V5.

Interactions

Speed Calling - One Touch -D on DESI-Less Phone

The following are the conditions for the Name Display function of [S-26D] Speed Calling - One Touch -D on DESI-Less Phone.

1. When SPD key is assigned on any Line/Feature key, Line/Feature key information display area shows the name information of Speed Calling. When name information is not assigned for Speed Calling, the LCD displays the Speed Calling number or the feature name on a system base (ASYDL, SYS1, Index 674, Bit 6).
   However, when [D-165] Dialed Number Privacy Mode is in-service (ASYDL, SYS1, Index 674, Bit 7=1), "*" is displayed instead of Speed Calling number.
2. Name Display for Speed Calling is only available on DESI-Less Phone.
3. Name information can only be assigned, modified, or deleted on the stations using DESI-Less Phone (**Note 3**). Assignment from PCPro is not supported.

   Note 3: Refer to the User's Guide or Operation Manual of the terminal for the assignment procedure.

4. Alphanumeric characters and special characters are available to use for name information entry. Euro characters (uppercase, lowercase) are supported on DT800/DT700/DT400/DT300 Series.
5. Up to 4096 stations can assign the name information of Speed Calling per system.
6. Assigned name information can be saved by the maintenance command MEM_HDD. For Digital terminals accommodated on ELC card, use BOSD command to backup Speed Calling number and use MEM_HDD command to backup Speed Calling name information. See the table below.

Terminal Type	Speed Calling number	Speed Calling name information
IP station	MEM_HDD	MEM_HDD
Digital terminal	BOSD	MEM_HDD

7. When station data is deleted, name information of related station is also deleted.
8. Because Speed Calling name information is assigned toward a key, the name data is not cleared/changed by changing the assigned number.
9. Assigned name information is only used for the display on Line/Feature key information display area. The display when the terminal is dialing, hearing RBT, or being connected corresponds to the assigned number.
10. Name information entry is available only when the station is idle or ringing. Even when My Line is not used, name information entry does not work during other key operation.
11. Station user can continue assignment operation when a call terminates during name information entry. However, when digit code or any keys irrelevant to name information entry is pressed, the terminal exits the assignment and then acts as operated.
12. When 30 seconds of no key operation is detected, the station goes back to idle state discarding the name information entry.
13. SPD key data can be changed as many times as desired by pressing the SPD key again, unless any digit code or software key is pressed.
14. A blank space can be entered by pressing INS key or "→" key.
15. Name information will be deleted when SET key is pressed without entering any data.

Dialed Number Privacy Mode

By assigning ASYDL, SYS1, Index 674, Bit 7=1, [D-165] Dialed Number Privacy Mode can be activated with DESI-Less Phone under the following conditions.

1. The icon on DESI-Less Phone indicates as follows in each dial mode:
   - Normal Mode: No information
   - Privacy Mode 1: ■ (steady)
   - Privacy Mode 2: ■ (flashing)
2. When ASYDL, SYS1, Index 674, Bit 6=1 (Speed Calling number is displayed on the LCD of SPD key), Speed Calling number display can act together with the Dialed Number Privacy Mode depending on the system data setting (**Note 4**):

   <Example>
   When "12345" is registered to SPD key and ASYDL, SYS1, Index 674, Bit 7=1 (Dial mode works on Speed Calling number display):
   - Normal Mode: "12345" is displayed.
   - Privacy Mode 1, 2: "SPD" is displayed. **Note 5**

Note 4: Regardless of the ASYDL, SYS1, Index 674, Bit 7 setting, the feature name "SPD" is displayed when ASYDL, SYS1, Index 674, Bit 6=0.

Note 5: "Speed Call" is displayed when 11 characters is enabled (ASYDL, SYS1, Index 684, Bit 0 = 0).

Programming

STEP 1: **ASYD** - SYS1, Index 241, Bit 6: How to handle the characters/numbers when the number of digits exceeds the limit (8 digits) to display on the LCD of Line/Feature key information display area.

0 = The first 7 digits + an asterisk (*) are displayed (i.e. CALL-FO*)

1 = An asterisk (*) + the last 7 digits are displayed (i.e. *LLCALLS)

SYS1, Index 78, Bit 4 and Index 389 Bit 0 to Bit 4: Language displayed on the LCD

	Index 78	Index 389				
	Bit 4	Bit 4	Bit 3	Bit 2	Bit 1	Bit 0
English	0	0	0	0	0	0
Portuguese	0	0	0	0	0	1
Spanish	0	0	0	0	1	0
French	0	0	0	0	1	1

Note: Refer to [M-117] Multilingual Display for the data assignment of ASLG command.

STEP 2: **ASYDL** -
SYS1, Index 673, Bit 5: Choose the information to be displayed on DESI-Less Phone.
0/1 = Name/Number
SYS1, Index 674, Bit 5: Periodic refresh for display information on DESI-Less Phone.
0/1 = ON/OFF

STEP 3: **ASYDL/ASYDN** -
SYS1, Index 684, Bit 0: 11 characters indication for Feature key information
0/1 = ON/OFF

Note: Available for DT800/DT700/DT400/DT300 Series.

SYS1, Index 684, Bit 1: DESI-Less icon display
0/1 = ON/OFF

STEP 4: **ASFC/ASFCN** - Assign a Service Feature Restriction Class that allows SFI 259 to the station (DT710 DESI-Less)
DAY/NIGHT: Day/Night Mode
TN: Tenant Number
SFI: 259 (DT710 DESI-Less (The terminal display is switched to Home display by pressing Feature key)) **Note 6**, **Note 7**
SFI: 264 (DESI-Less Display - One Page Mode) **Note 7**, **Note 8**, **Note 9**
SFC: Service Feature Restriction Class
RES: 0/1 (Restricted/Allowed)

Note 6: Assign this data when using DT710 DESI-Less.

Note 7: Reboot the terminal after assigning this data.

Note 8: Assign this data when Line/Feature key information display area (DESI-Less Display) is fixed to a one page display.

Note 9: Available since FP95-115 V5.

<Data Programming for Line key information (My Line, Sub Line, Virtual Line)>

For the data programming of name information, refer to [N-28] Name Display - System.

<Data Programming for Feature key information>

Refer to AFDD/AFDDN command in Command Manual.

<Data Programming for SPD key information>

STEP 1: **ASYDL** - SYS1, Index 674
Bit 6: Display for SPD key on DESI-Less Phone. 0/1 = Feature name/Speed Calling number
Bit 7: Dialed Number Privacy Mode works with Speed Calling number display. 0/1 = OFF/ON

STEP 2: **AKYD** - Assign the Feature key data of Desktop terminal.
KYN: Key number
FKI: 1 (Function key)
FKY: 165 (SPD Name Assign)

STEP 3: **ADSL** - Assign the Software key data of Desktop terminal.
SN: 0 (Idle)
SKN: Software key number
FKY: 165 (SPD Name Assign)

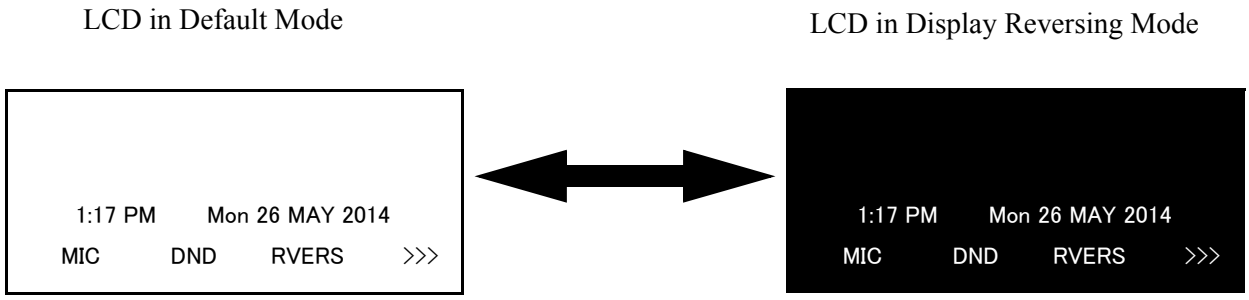
Note: Assign the name information of Speed Calling by pressing the SPD Name Assign key on DESI-Less terminal's Line/Feature key. For the assignment procedure, refer to User's Guide or Operation Guide of each terminal.

D-170 Display Reversing

General Description

This feature enables DT800/DT700 and DT400/DT300 Series terminal to reverse the black and white in its display: background, characters and icons. Perform one of the following to reverse the black and white in the display;

- Assign the feature to a Line/Feature key.
- Assign the feature to a Soft key.



Operating Procedure

Press Line/Feature key or Soft key assigned for Display Reversing. LCD will toggle between Display Reversing Mode and Default Mode each time the key is pressed.

Service Conditions

1. To maintain “Display Reversing” state,
   - When the terminal is initialized, the terminal maintains the state prior to initialization.
   - Following actions of terminals will erase data;
     a.) DT800 Series/DT700 Series: Turning the Telephony Server's power OFF-ON. **Note 1**
     b.) DT400 Series/DT300 Series: Mounting the ELC circuit card in a different mounting slot.
   - Saved data for “Display Reversing” returns to default when switching to SR-MGC, or when switching to Backup Node by Location Diversity [L-55].

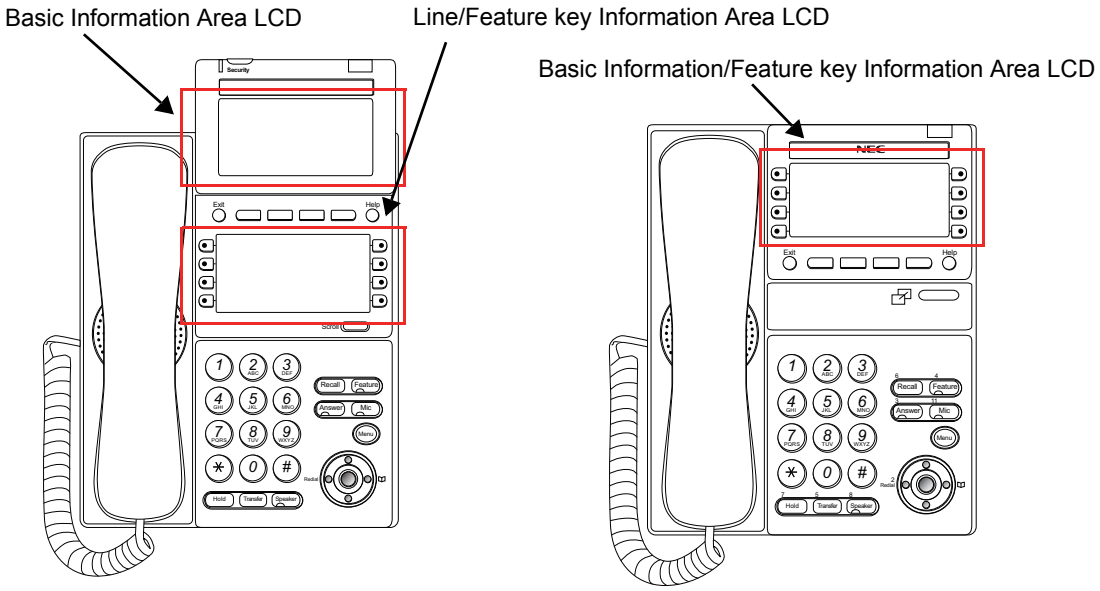
Note 1: To maintain data, save “One Touch Speed Calling for DtermIP” using the MEM_HDD command.

2. This feature is available for the following terminals;

   - DT800 Series/DT700 Series: DT830/DT730, DT830G/DT730G (12DG/24DG), DT710 **Note 2**
   - DT400 Series/DT300 Series: DT430/DT330

   Note 2: This feature is not available for the following DT800 Series/DT700 Series;
   - DT710 2-button/12-button type without LCD
   - DT830G/DT730G (12CG/24CG)
   - DT750

3. For DT830/DT730/DT710 (DESI-LESS)/DT430/DT330 (DESI-LESS) terminal: "Display Reversing" occurs in both Basic Information Area LCD and Line/Feature key information area LCD.



DT830/DT730/DT430/DT330 (DESI-LESS)

DT710 (DESI-LESS)

4. One of the ELC cards listed below is required for this feature.

ELC Card	Firmware Requirements
CH-16ELCA	SP-4035 16ELCA PROG-A Version 1 or later.
PA-16ELCN	SP-3977 16ELCN PROG-B Version 1 or later.
PA-16ELCN-A	SP-4013 16ELCN-A PROG-A Version 1 or later.

<Conditions Specific to This Feature>

1. This feature is not available in the terminal's Configuration menu.
2. Controlling the Line/Feature key lamp and Soft key display by configuring ASYDL, SYS1, Index 1256, Bit 2: **Note 3**

SYS1, Index 1256, Bit 2=	Mode	Line/Feature Key Lamp	Soft Key
0	Default	OFF	Remains the same.
	Display Reversing	OFF	Remains the same.
1	Default	OFF	Remains the same.
	Display Reversing	ON	RVERS

Note 3: When changing data for ASYDL, SYS1, Index 1256, Bit 2, user must execute the following:
- DT800 Series/DT700 Series: Initialize the Telephony Server
- DT400 Series/DT300 Series: Initialize the circuit card

Programming

<Assigning Display Reversing Feature to a Line/Feature key>

AKYD - Assign the following key data for DT Series terminals.
TN (Tenant Number): 1-63
UGN (User Group Number): 1-255
STN (Station Number): Maximum 5 digits for Business Services, 6 digits for Hotel services.
TEL (Telephone Number): Maximum 16 digits.
KYI (Service Index): 1 (Line/Feature Key)
FKY (Function Key Number): 190 (Display Reversing)

<Assigning Display Reversing Feature to a Soft key>

STEP 1: **ADSL** - Assign the Soft key pattern data.
SKP (Soft Key Pattern Data): 0-15
SN (Status Number): 0-15
SKN (Soft Key Number): 0-15
FKY (Function Key Number): 190 (Display Reversing)

STEP 2: **ADKS** - Assign the Soft key pattern data to DT Series terminals.
TN (Tenant Number): 1-63
STN (Station Number): Maximum 5 digits for Business Services, 6 digits for Hotel services.
SKP (Soft Key Pattern): SKP assigned by the ADSL command.

<Controlling the Line/Feature key lamp and Soft key Display>

ASYDL - Assignment of System Data for LDM.
SYS1
Index 1256
Bit 2=1
Behaviors of Line/Feature key lamp and Soft key display are;

- Line/Feature key: Lamp turns on.
- Soft Key: RVERS

Note: When changing data for ASYDL, SYS1, Index 1256, Bit 2, user must execute the following:

- DT800 Series/DT700 Series: Initialize the Telephony Server
- DT400 Series/DT300 Series: Initialize the circuit card

D-172 Display Enlarged Characters

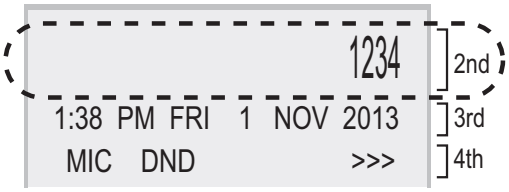
General Description

This feature enables a DT800 Series/DT700 Series terminal to switch the size of characters displayed in the 2nd line and the 3rd line on the LCD to a double height size. The following shows an example when using this feature:

1. No enlarge characters



2. Display enlarged characters (2nd line)



3. Display enlarged characters (3rd line)



Operating Procedure

To switch the characters displayed to double height size, press the Soft Key or the Line/Feature Key. By pressing the Soft Key or the Line/Feature Key, you can switch the characters displayed to double height size in the order of "Display enlarged characters (2nd line)"→"Display enlarged characters (3rd line)"→"No enlarge characters".

Service Conditions

1. The enlarged characters display will be cleared by turning off/on the power of the Telephony Server.

   Note: To maintain data, save “One Touch Speed Calling for DtermIP” using the MEM_HDD command.

2. When the terminal initialization is executed, the previous status of the terminal initialization will be maintained.

3. For using this feature, use the terminal with the following firmware versions.

Terminals	Firmware Version
DT830G Series	Version 3.0.0.0 or later
DT830 Series	Version 3.0.0.0 or later
DT730G Series	Version 1.0.3.0 or later
DT730 Series	Version 5.0.5.0 or later
DT750	Version 5.0.5.0 or later

Note: This feature is only available for the terminals with a 4-line LCD. For other terminals, this feature is not available.

4. If you delete the station data while enlarged characters are displayed, and assign another station number to the same location, the deleted information of enlarged characters is taken over.
5. The line displayed in enlarged characters can be specified per terminal.

Interactions

1. This feature and Name Display Service cannot be activated simultaneously. For using this feature enabling Name Display Service, assign System Data (ASYDL, SYS1, Index 673, Bit 0=1 and Bit 1=1), and disable the indication of Name Display information registered on Terminal Phonebook.
2. Even if call status or display contents are changed, the indication of enlarged characters in the line is maintained. However, while the terminal displays the screen for each of the following conditions, the indication of enlarged characters will be canceled. If you switch the line of enlarged characters while displaying the screen for each of the following features, the indication of enlarged characters will be enabled after the screen closed.

   - When Call History is displayed.
   - When Phonebook is displayed.
   - When help screen displayed pressing Soft Key.
   - When menu screen is displayed.

Programming

<Assigning Display Reversing Feature to a Line/Feature Key>

STEP 1: **AKYD/AKYD_T** - Assign the following key data for DT Series terminals.
TN (Tenant Number): 1-63
UGN (User Group Number): 1-255
STN (Station Number): Maximum 5 digits for Business Services, 6 digits for Hotel services.
TEL (Telephone Number): Maximum 16 digits.
KYI (Service Index): 1 (Line/Feature Key)
FKY (Function Key Number): 198 (Large Font)

<Assigning Display Reversing Feature to a Soft Key>

STEP 1: **ADSL** - Assign the soft key pattern data.
SKP (Soft Key Pattern Data): 0-15
SN (Status Number): 0-15
SKN (Soft Key Number): 0-15
FKY (Function Key Number): 198 (Large Font)

STEP 2: **ADKS** - Assign the soft key pattern data to DT Series terminals.
TN (Tenant Number): 1-63
STN (Station Number): Maximum 5 digits for Business Services, 6 digits for Hotel services.
SKP (Soft Key Pattern): SKP assigned by the ADSL command.

D-173 DNIS without CTI Server

General Description

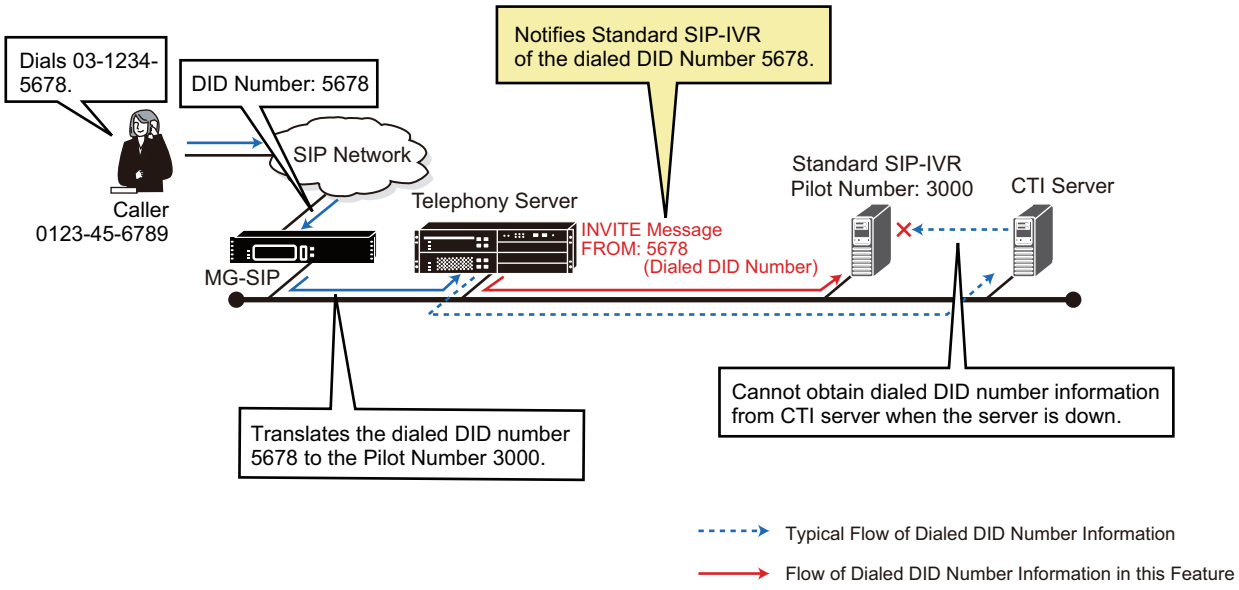
This feature provides reliable Dialed Number Identification Service (DNIS) that requires no CTI server. It allows the Telephony Server to directly notify Standard SIP-IVR server (**Note 1**) of Direct Inward Dialing (DID) number information received from public ISDN or SIP network.

A typical IVR system obtains dialed DID number information via a CTI server, which means it cannot access the information if the CTI server is down. With this feature, the Telephony Server sends dialed DID number information directly to your Standard SIP-IVR server using the FROM field of INVITE message to establish the connection to the IVR. This way your Standard SIP-IVR server can offer DNIS regardless of the state of the CTI server.

Note 1: The term “Standard SIP-IVR server” means an Interactive Voice Response (IVR) system that satisfies the service conditions specified in SIP HANDLER - SIP TERMINAL [S-167].

Note: This feature is available since FP95-112 V2.

Example: DID Call Received from SIP Network



Operating Procedure

None.

Service Conditions

<Important Notices>

This feature is only available for incoming calls from ISDN or SIP network; otherwise it does not work.

<Interactions with Other Features>

None.

<Hardware/Software>

None.

<Interactions with Networks>

This feature can be used in conjunction with the Peer-to-Peer FCCS routing. However, it does not support the DTI-FCCS routing.

<Conditions Specific to This Feature>

With the User Agent Class Data setting of the AUACL/AUACN command, you can specify whether to notify Standard SIP-IVR server of both caller's number (ANI) and dialed number (DNIS) or dialed number only.

<Other Conditions>

None.

Programming

The procedure differs, depending on whether the same User Agent information is applied to all Standard SIP-IVR servers in the system when this feature is activated.

- Using the Same User Agent Information on All Standard SIP-IVR Servers

  User Agent Class is identified by User Agent information commonly used among all Standard SIP-IVR servers in the system. For details, see <Using the Same User Agent Information on All Standard SIP-IVR Servers>.

- Using Fixed User Agent Class Specified on a Telephone Number Basis

  User Agent Class is identified by a login Telephone Number, regardless of the User Agent Information shared among IVR servers in the system. Using this procedure, you can configure User Agent Class on a Telephone Number basis for multiple Standard SIP-IVR servers. For details, see <Using Fixed User Agent Class Specified on a Telephone Number Basis>.

<Using the Same User Agent Information on All Standard SIP-IVR Servers>

STEP 1: **AUACL/AUACN** - Assignment of the User Agent Class Data for LDM/NDM

Configure User Agent Classes in accordance with the User Agent information available on the Standard SIP-IVR servers.

FUNC=User Agent Class Setting

TYPE=User Agent Class

UAN: User Agent Class Number (1-63)

UA: User Agent (Maximum 32 Characters)

Enter User Agent information used on the Standard SIP-IVR server.

INFORMATION: Information of SIP Phone (Maximum 32 Characters)

Enter additional information or comments on the User Agent Class, if necessary.

User Agent Class Data

DNIS Notification of INVITE: **Note 2**

Only dialed number information is provided in the FROM field of INVITE message for an incoming call.

0/1=Disabled/Enabled

DNIS and ANI Notification of INVITE: **Note 2**

Both dialed number and caller's number information is provided in the FROM field of INVITE message for an incoming call.

0/1=Disabled/Enabled

Note 2: Please note the following points:

- Parameters of **User Agent Class Data**, other than **DNIS Notification of INVITE** and **DNIS and ANI Notification of INVITE**, must be specified to suit the operating environment of the Standard SIP-IVR server. For further information, see the AUACL/AUACN section of Command Manual.
- Parameters **DNIS Notification of INVITE** and **DNIS and ANI Notification of INVITE** are mutually exclusive and cannot be enabled simultaneously.

<Using Fixed User Agent Class Specified on a Telephone Number Basis>

STEP 1: **AUACL/AUACN** - Assignment of the User Agent Class Data for LDM/NDM

Create particular User Agent Classes to be used on a Standard SIP-IVR server for this feature.

FUNC=User Agent Class Setting

TYPE=User Agent Class

UAN: User Agent Class Number (1-63)

UA: User Agent (Maximum 32 Characters) **Note 3**

Enter User Agent information used on the Standard SIP-IVR server.

Note 3: This parameter can be left blank, in which case User Agent data cannot be included in the list of User Agent Search Sequence.

INFORMATION: Information of SIP Phone (Maximum 32 Characters)

Enter additional information or comments on the User Agent Class, if necessary.

User Agent Class Data

DNIS Notification of INVITE: **Note 4**

Only dialed number information is provided in the FROM field of INVITE message for an incoming call.

0/1=Disabled/Enabled

DNIS and ANI Notification of INVITE: **Note 4**

Both dialed number and caller's number information is provided in the FROM field of INVITE message for an incoming call.

0/1=Disabled/Enabled

Note 4: Please note the following points:

- Parameters of **User Agent Class Data**, other than **DNIS Notification of INVITE** and **DNIS and ANI Notification of INVITE**, must be specified to suit the operating environment of the Standard SIP-IVR server. For further information, see the AUACL/AUACN section of Command Manual.
- Parameters **DNIS Notification of INVITE** and **DNIS and ANI Notification of INVITE** are mutually exclusive and cannot be enabled simultaneously.

STEP 2: **ADSDL/ADSDN** - Assignment of Detail SIP phone Data for LDM/NDM

Assign the User Agent Classes specified with the AUACL/AUACN command to Telephone Numbers available on the Standard SIP-IVR server.

UGN: User Group Number (1-255)

TELN: Telephone Number (Maximum 16 Digits (0-9, *, #))

Enter a Telephone Number available on the Standard SIP-IVR server.

KIND=User Agent of System

TYPE=User Agent Class

UAN: User Agent Class Number (1-63)

Enter an User Agent Class Number specified with the AUACL/AUACN command.

E-1 Executive Right-of-Way

General Description

This feature enables selected station users, upon encountering a busy condition at an Internal Station, to barge into the Busy Connection after transmitting a Warning Tone.

Operating Procedure

To enter a Busy Connection using Access Code

1. Dial the busy Station. Receive Busy Tone.
2. Switch hook flash. Receive Special Dial Tone.

   Note: The availability of special dial tone is changed depending on the combination of the Step Call - CCIS Restriction (ASYD, SYS2, Index 6, Bit 2) and SHF method (ASYD, SYS1, Index 68, Bit 0). Refer to Step Call - CCIS [S-59] in Data Programming Manual - CCIS for the detail.

3. Dial the EXECUTIVE RIGHT-OF-WAY Access Code. Warning Tone is sent.
4. The Calling Station has barged into a Three-way Call.

<The following procedures are applied for Desktop terminal only.>

To enter a Busy Connection using Line/Feature key

1. Dial the busy Station. Receive Busy Tone.
2. Press the EXECUTIVE RIGHT-OF-WAY OVER Feature key. The associated LED lights and Warning Tone is sent.
3. The Calling Station has barged into a THREE-WAY CALLING [T-2]. The LCD displays:

Interrupted Party Called Party

OVERRIDE	**XXXX**	**XXXX**
	(Time Display)	

The LCD then displays:

CONF
(Time Display)

Service Conditions

1. Two Burst Tones (.08 second) are transmitted upon activation of this feature to alert the connected parties that an EXECUTIVE RIGHT-OF-WAY Call has been barged into the connection.

   Note: This warning tone is not available for MG(BRI)-SIP.

2. The LCD on the interrupted party's Desktop terminal will give an indication only if INTERMEDIATE STATION NUMBER DISPLAY -D [I-10D] has been programmed in System Data.
3. EXECUTIVE RIGHT-OF-WAY is denied if the busy Station is:
   a.) dialing or in a LINE LOCKOUT [L-3] state
   b.) listening to a Tone
   c.) being rung
   d.) protected from override via CLASS OF SERVICE - INDIVIDUAL [C-15]
   e.) being connected to an ATTENDANT CONSOLE [A-3] Loop
   f.) activating a conflicting feature, such as:

ATTENDANT CAMP-ON WITH TONE INDICATION	[A-1] (Desktop terminal only)
ATTENDANT OVERRIDE	[A-7]
BOSS - SECRETARY OVERRIDE -D	[B-5D]
BUSY VERIFICATION	[B-3, B-9]
CALL HOLD	[C-6]
CALL TRANSFER	[C-10, C-11, C-50, C-54]
CALL WAITING - ORIGINATING	[C-31]
CALL WAITING - TERMINATING	[C-12]
CONSULTATION HOLD - ALL CALLS	[C-17, C-58]
DATA LINE SECURITY	[D-12, D-66]
DATA PRIVACY ON DEMAND	[D-17, D-68]
DIAL ACCESS TO ATTENDANT	[D-2, D-71]
EXCLUSIVE HOLD -D	[E-4D]
NON-EXCLUSIVE HOLD -D	[N-7D]
OFF-HOOK QUEUING	[O-7, O-11]
OUTGOING TRUNK QUEUING	[O-2]
PRIVACY -D	[P-14D]
PRIVACY RELEASE	[P-18]
THREE-WAY CALLING	[T-2]
VOICE CALL	[V-2, V-7]
Data Call through Data Adapter/Data Call through Data Module	(Desktop terminal only)

   Note: In any of the above instances, the Calling Station will receive Reorder Tone.

4. If one party disconnects after the Three-way Conference Barge is established, the remaining two parties will still be connected.
5. The EXECUTIVE RIGHT-OF-WAY Access Code may be assigned according to customer preference.
6. The maximum number of simultaneous EXECUTIVE RIGHT-OF-WAYs allowable is dependent on the number of Conference Function Trunks in the system. (Conference function is included in TSW and MUX circuit cards.)
7. This feature can be denied on DICTATION ACCESS [D-3] and PAGING ACCESS [P-1, 28] Connections through programming in System Data.

8. When Called Station activates CALL FORWARDING- BUSY LINE [C-2], and the call forwarding destination is busy:

   - When call forwarding destination is a station (including stations in FCCS network), ASYD, SYS1, Index 69, Bit 5 determines whether to set EXECUTIVE RIGHT-OF-WAY to the Forwarded-from Station, or to the Forwarded-to Station.

   - When the call forwarding destination is in outside of the self office, EXECUTIVE RIGHT-OF-WAY is restricted and does not work.

9. If the Called Station Line has a party on CALL HOLD [C-6], override is not allowed to the Called Station Line. Also override to a Station Line in CALL HOLD [C-6] is denied and Reorder Tone is provided. Terminating Override Calls are denied when the Called Party has another party in a CONSULTATION HOLD [C-17] condition.
10. If the Called Station is maintenance busy and EXECUTIVE RIGHT-OF-WAY is invoked, override is denied and Reorder Tone is returned to the Calling Station.
11. It is possible, via System Data, to disable the sending of a Warning Tone during an EXECUTIVE RIGHT-OF-WAY Connection. Normally the Two-burst Warning Tone is sent when the connection is initiated.
12. The Node accommodating the Called Station requires Conference Trunk (CFT) function in the FCCS Network.
13. When an EXECUTIVE RIGHT-OF-WAY Connection is established, all the parties (Calling Party, Called Party and Interrupted Party) hear Warning Tone for a certain time.

    Note: Interrupting calls in progress via PHC does not provide Warning Tone to the party connected with Called Party.

    Note: Interrupting calls in progress via CCIS network does not provide the lamp indication of CONF key on the party connected with Called Party.

14. When using this feature against a Station that is engaged in Tandem connection other than CCIS-CCIS, SFI 4 and SFI 11 require to be set to 1 for SFC 0 by using ASFC/ASFCN.
15. This feature can not be used when the Called Party is connected with the Attendant Console.
16. EXECUTIVE RIGHT-OF-WAY is denied if the called party connected via PHC trunk is hearing hold tone. EXECUTIVE RIGHT-OF-WAY is available, however, if the called party is in tandem connection via non-PHC trunk beyond the connection via PHC trunk.
17. To operate this feature from a SIP Handler Controlled SIP terminal, perform the one digit dial hooking operation while hearing Busy Tone (BT) and dial the access code for this feature. To hear Busy Tone (BT), enable IP-DTG with the AUACL/AUACN command.
18. This feature cannot be used when a SIP Handler Controlled SIP terminal is in Multi Line Mode, the SIP Handler Controlled SIP terminal cannot hear a Warning Tone whether it barges into or another station barges into the connection with it.

<The following conditions are applied for Desktop terminals only.>

1. EXECUTIVE RIGHT-OF-WAY can also be accessed via the dial pad (Access Code) or SPEED CALLING - ONE TOUCH -D [S-26D].

Programming

STEP 1: **ASYD** - SYS1, Index 70, Bit 7. Send Warning Tone to interrupted parties when Executive Right-of-Way is in operation 0/1: Required/Not Required.

SYS2, Index 0, Bit 1. Busy Station Service. Assign data "1".

SYS1, Index 69, Bit 5. EXCECUTIVE RIGHT-OF-WAY 0/1 = Set to Forwarded-from Station/Set to Forwarded-to Station

STEP 2: **ANPD** - Assign first digit of Access Code for EXECUTIVE RIGHT-OF-WAY.

STEP 3: **ASPA** - Assign an Access Code for EXECUTIVE RIGHT-OF-WAY, SID 4 for Connection Status Index (CI) of "B: Busy".

STEP 4: **ASFC** - Assign SFI 3 [EXECUTIVE RIGHT-OF-WAY (Calling Side)], SFI 4 [EXECUTIVE RIGHT-OF-WAY (Called Side)], and SFI 11 (Data Line Security).
When using this feature against a Station that is engaged in Tandem connection other than CCIS-CCIS, assign SFI 4 and SFI 11 to 1 for SFC 0.

STEP 5: **AKYD** - Assign EXECUTIVE RIGHT-OF-WAY to a Line/Feature key. Assign Feature key Executive Right-of-Way [3].

STEP 6: **ADSL** - Assign SN = 3 (Called Station is Busy) and Soft key Executive Right-of-Way [3].

FCCS Programming

This data setting describes only the additional or replacement commands used to activate FCCS Services. When used as an FCCS feature, these commands replace the non-network commands (for example, AAED is replaced by AAEDN). Note that the data setting for FCCS Service varies depending on the Node.

Note: When using ASYDL or ASYDN to program an FCCS feature, the basic programming of ASYD is required in addition to ASYDL or ASYDN.

STEP 1: **ANPDL** - Assign first digit of Access Code for EXECUTIVE RIGHT-OF-WAY.

STEP 2: **ASPAL** - Assign an Access Code for EXECUTIVE RIGHT-OF-WAY, SID 4 for Connection Status Index (CI) of "B: Busy".

E-3D Elapsed Time Display -D

General Description

This feature provides an LCD display of the time elapsed while a Desktop terminal is connected to any trunk.

Operating Procedure

No manual operation is required.

Service Conditions

1. The ELAPSED TIME is displayed as follows. (H: hour, M: minute, S: second)
   - For a legacy terminal
     M:SS ---> MM:SS ---> H:MM:SS
   - For an IP terminal
     H:MM:SS
2. The ELAPSED TIME is displayed in the eight left-side positions of the LCD.
3. The ELAPSED TIME can reach a maximum of 9 hours, 59 minutes and 59 seconds. After that time, the clock returns to zero.
4. After a conversation ends, the ELAPSED TIME is being displayed for approximately 5 seconds.
5. A continuous time display will occur during CONSULTATION HOLD - ALL CALLS [C-17] and THREE-WAY CALLING [T-2].
6. When a call is transferred, the time display of the party receiving the transfer begins from zero.
7. The ELAPSED TIME is not displayed when the Station is in a specific condition such as:
   a.) THREE-WAY CALLING [T-2]
   b.) connected to a Trunk with held Trunk, as in the case of CONSULTATION HOLD [C-17].
8. In DtermIP, when hooking-off status on the phone is kept after the speech has been finished, LCD displays Hang Up message. Then, if the phone is put in on-hook status, LCD displays “0:00:00”.

Programming

No programming is required.

E-4D Exclusive Hold -D

General Description

This feature allows a user to place a call on hold and to exclude all other users from retrieving the Held Call.

Operating Procedure

While a call is in progress

1. Press the HOLD key twice. The LCD displays:

Held Station or Trunk



2. To return to the Held Call, press the Held Line key. The conversation is re-established.

Service Conditions

1. EXCLUSIVE HOLD -D may be activated from any line appearing on a Desktop terminal.
2. After EXCLUSIVE HOLD has been set, the user can make or answer calls from any other line appearing on the Dterm.
3. Only the Dterm that sets EXCLUSIVE HOLD may retrieve the Held Call.
4. After a variable programmable period of time, the Held Call will enter AUTOMATIC RECALL [A-8] regardless of the status of the Dterm. Ringing, however, is disabled while DO NOT DISTURB -D [D-11D] is activated.
5. The LCD of other Dterm Stations on which the Held Line appears will be lit steadily.
6. Two-party Hold is not available when a Dterm is engaged in a Three-party Conference with a Station and/or Trunk.
7. The held Station is denied the following services:

ATTENDANT CAMP-ON WITH TONE INDICATION	[A-1]
ATTENDANT OVERRIDE	[A-7]
BOSS - SECRETARY OVERRIDE -D	[B-5D]
BUSY VERIFICATION	[B-3, B-9]
CALL WAITING - ORIGINATING	[C-31]
CALL WAITING - TERMINATING	[C-12]
EXECUTIVE RIGHT-OF-WAY	[E-1]

8. In the case of SIP Handler Controlled SIP terminals, use the IP-DTG tone assigned with the AUACL/AUACN command (SST is recommended to use) instead of tones assigned by ASYD, SYS1, Index 72, Bit 2 (Tone pattern to be sent out to stations on Exclusive Hold/Non-Exclusive Hold).
9. This feature cannot be used when a SIP Handler Controlled SIP terminal is in Multi Line Mode.
10. When the held party is an IP terminal and the connection with the party is disconnected due to the accidents such as network failure or unplugging of cables, the lamp of the Line key on the Desktop terminal that held the call keeps flashing. If a user of this Desktop terminal tries to retrieve the call in this condition, dial tone is heard and the lamp of the Line key will stop flashing and turn on.

Programming

STEP 1: **ASYD** - SYS1, Index 144. Automatic Recall Timer for Dterm hold. Assign as data 00H for default value of 30 seconds.

SYS2, Index 12, Bit 1: Automatic Recall Ringer for Exclusive Hold.

0/1 = Recall/No Recall

STEP 2: **ANPD** - Reserve a Number Level for Feature Access. Select Normal Service (N), Hooking Service (H), or Busy Service (B) from the combo box for Connection Indexes. Assign NND in accordance with a predetermined Numbering Plan.

STEP 3: **ASPA** - Assign an Access Code to Exclusive Hold -D, SRV = "SSCA: Service Code Appendix", SIDA = 32, 35, and assign Connection Status Index (CI) of "H: Hooking" and "N: Normal".

STEP 4: **ASFC** - Permit Exclusive Hold -D for each Service Feature Restriction Class.

SFC: Service Feature Restriction Class [0-15]

SFI: SFI 35=1 (Exclusive Hold -D, Non-Exclusive Hold -D)

FCCS Programming

This data setting describes only the additional or replacement commands used to activate FCCS Services. When used as an FCCS feature, these commands replace the non-network commands (for example, AAED is replaced by AAEDN). Note that the data setting for FCCS Service varies depending on the Node.

Note: When using ASYDL or ASYDN to program an FCCS feature, the basic programming of ASYD is required in addition to ASYDL or ASYDN.

STEP 1: **ANPDL** - Reserve a Number Level for Feature Access. Select Normal Service (N), Hooking Service (H), or Busy Service (B) from the combo box for Connection Indexes. Assign NND in accordance with a predetermined Numbering Plan.

STEP 2: **ASPAL** - Assign an Access Code to Exclusive Hold -D, SRV = "SSC: Service Code", SID = 5 for Connection Status Index (CI) of "B: Busy" and "N: Normal".

E-11 Emergency Call

General Description

This feature allows a user to make an EMERGENCY CALL by dialing a Special Telephone Number (Emergency Telephone Number). Even if the Calling Station goes on-hook, the Calling Station is recalled by either of the following methods and the speech path is maintained until the Emergency Telephone goes on-hook.

- Recall the Calling Station by pressing the switch hook or the TRANSFER key
- Recall the Calling Station automatically

When a Station dials the Emergency Telephone Number, a System Message is printed out to report the EMERGENCY CALL.

Operating Procedure

Recall the calling station by pressing the switch hook or the TRANSFER key

- Recalling the calling station by pressing the switch hook

1. Calling Station puts the handset down.
2. Emergency Telephone goes on-hook.
3. Calling Station is recalled.

- Recalling the calling station by pressing the TRANSFER key

Note: If the terminal is equipped with the TRANSFER key, operation by pressing the TRANSFER key is required. Operation by pressing the switch hook is not effective for the terminal equipped with the TRANSFER key.

1. Calling Station puts the handset down.
2. Emergency Telephone presses the TRANSFER key.
3. Calling Station is recalled.

Recall the calling station automatically

1. Single Line Telephone and Digital terminal can be used as the Emergency Telephone. IP terminal, SIP Multiple Line Terminal, and Soft Phone cannot be used as the Emergency Telephone.
2. This feature is available when an Emergency Telephone Number is dialed via STATION-TO-STATION CALLING [S-11].
3. This feature is also available when an Emergency Call is made via CALL FORWARDING - ALL CALLS [C-5] or CALL FORWARDING - BUSY LINE [C-2].
4. Switch hook flash services, such as CONSULTATION HOLD - ALL CALLS [C-17], may be restricted.
5. Only the Emergency Telephone is allowed to disconnect from an Emergency Call. Upon disconnection, the Calling Station will hear Reorder Tone.
6. This feature is not available between Emergency Stations. When either of the Emergency Stations goes on-hook, the other Emergency Station receives Reorder Tone, and CALL BACK [C-1] is not performed.

7. When a Station makes an EMERGENCY CALL by any method, other than STATION-TO-STATION CALLING [S-11] (STATION-TO-STATION CALLING - OPERATOR ASSISTANCE [S-12], OFF-HOOK ALARM [O-6], etc.), there is no System Message printout, but CALL BACK [C-1] to the Calling Station is performed.
8. When a Digital Terminal is used for an EMERGENCY CALL, VOICE CALL [V-2] can be used.
9. An Emergency Telephone can assign CALL FORWARDING - BUSY LINE [C-2] or CALL FORWARDING - ALL CALLS [C-5], but when transfer is made to other than an Emergency Station, this feature is not available.
10. An Emergency Station can assign CALL FORWARDING - DON'T ANSWER [C-3]. In this case, when a call terminates to the Emergency Station, a System Message printout is performed. When a call is transferred by CALL FORWARDING - DON'T ANSWER [C-3], the situation regarding when the Station goes on-hook is as follows:
    a.) If neither the Calling Station nor the Transfer Destination Station is an Emergency Station, there is no CALL BACK [C-1].
    b.) If either the Calling Station or the Transfer Destination is an Emergency Station, CALL BACK [C-1] is operative. (No printout is performed).
    c.) If both the Calling and Destination Stations are Emergency Stations, there is no CALL BACK [C-1]. (No printout is performed).
11. When the Emergency Station is busy, the caller receives Busy Tone, and this feature is not available.
12. The Emergency Station can be a member of a Hunting Group. If hunting results in transfer to a Station other than an Emergency Station, this service is not available.
13. The Emergency Station can be a member of a UCD [U-1] group. If a call is distributed to the Emergency Station after queuing, System Message printout is not performed.
14. When the Emergency Station is busy, CALL BACK [C-1] to the Emergency Station is effective (no printout is performed).
15. When the Emergency Station is busy, CALL WAITING - ORIGINATING [C-31] and CALL WAITING - TERMINATING [C-12] are effective, but this feature is not available. However, when a Two-way Call is made between an Emergency Station and another Station, CALL BACK [C-1] is performed when the Station releases.
16. When an Emergency Station is busy, ATTENDANT CAMP-ON WITH TONE INDICATION [A-1] to the Emergency Station is effective, but the feature is not available to the CAMP-ON Call.
17. When a Digital Terminal is used as an Emergency Station, VOICE CALL [V-2] to the Emergency Station is possible. If the Calling Station goes on-hook before the Emergency Station answers by pressing the SPEAKER key or going off-hook, CALL BACK [C-1] is not possible.
18. When Camp-On or Privacy is set to a Station, the Station cannot be recalled from the Emergency Station. Therefore, data security [SFI 11 = 0 (Data Line Privacy)] must be assigned to the Emergency Stations, so that Camp-On and Privacy cannot be set.

Interactions

1. STATION-TO-STATION CALLING [S-11] must first be provided.
2. SINGLE-DIGIT STATION CALLING [S-1] may be used with this feature.

Programming

STEP 1: **ASYD** - SYS1, Index 4, Bit 1. Recalling method of EMERGENCY CALL. 0/1: SHF or Pressing the TRANSFER key/Recalled Automatically. This allows for Called Party Release only on an Emergency Call. If the Calling Party releases, the Calling Party is rung by the Emergency Telephone.

SYS1, Index 4, Bit 7: Printout of Emergency Call System Messages.

0/1 = No printout/printout

STEP 2: **ASFC** - Assign to the Emergency Call Telephone a Service Feature Restriction Class that allows SFI 24 (Emergency Call). Stations that may call the Emergency Call Telephone are not allowed SFI 24.

Note: The SFC must be allowed for My Line of the telephone, even if the Sub Line is assigned as Emergency Call Telephone.

FCCS Programming

This data setting describes only the additional or replacement commands used to activate FCCS Services. When used as an FCCS feature, these commands replace the non-network commands (for example, AAED is replaced by AAEDN). Note that the data setting for FCCS Service varies depending on the Node.

STEP 1: **ANPDN** - Assign first digit of FCCS Numbering Plan.

STEP 2: **ASPAN** - Assign Numbering Plan Data for FCCS.

STEP 3: **ALGSN** - Assign FCCS Logical numbers to Stations used in the FCCS network.

E-18 E911-ANI Unified Number of Digits

General Description

This service unifies the number of ANI information digits sent out by E911-ANI Service.

Note: This feature is available in North America only.

Operating Procedure

The Operating Procedure is the same as for a normal Outgoing Connection.

Service Conditions

1. The ANI function for E911 Outgoing Connections is the same as in previous E911-ANI Service and is applicable to the following connections:
   a.) LCR Call from a Station by MF Signaling
   b.) LCR Call from an Attendant Console by MF Signaling.
2. ANI Sending Number by this Enhanced Service is NXX-XXXX.
3. The method of obtaining the ANI Sending Number is shown below:

   Caller's Telephone Number: 12345 (Closed Numbering)
   Outgoing Route Number (RT): **
   Station Outgoing: AT = 0
   Number of digits of Caller's Telephone Number: 16
   ANI Assigned Data (from the above data):
   Skip: 2 (Number of digits to be skipped from the top of the Telephone Number)
   Add: 4 (Number of digits to be added to ANI information)
   ANI information: 7890

   a.) The specified number of Skip digits is deleted from the Telephone Number: 12345
   b.) The ANI Sending Number is created by combining these digits with the ANI information: 7890345
   c.) Sending ANI information: KP-0-789-0345-ST
4. Sending digit patterns in E911-ANI:
   a.) Outgoing Call from Station.
   • If ANI Data is assigned:

   KP-0-NXX-XXXX-ST
   - NXX-XXXX: ANI sending number
   - 0: Information digit

   • If ANI Data is not assigned:

   KP-2-ST
   - 2: Information digit

b.) Outgoing Call from Attendant:

- If Individual Attendant Access Code is assigned, and also ANI Data is assigned:

KP-0-NXX-XXXX-ST
- ANI sending number
- Information digit

- If Individual Attendant Access Code is not assigned, and ANI Data is assigned:

KP-0-NXX-XXXX-ST
- ANI sending number
- Information digit

- If ANI Data is unassigned:

KP-1-ST
- Information digit

5. An independent route is required for E911 Interface.
6. After the Caller's Number is sent out to the Distant Office by E911-ANI, it takes some time until a speech path is established.
7. Sending Data is considered to be not assigned if:
   a.) Service version data specifies that Unified Number of Digits is provided.
   b.) No data is assigned for Unified Number of Digits Service.
   c.) Data for previous E911-ANI Service is assigned.
8. After digits have been sent out, DDD Seizure Tone is not sent.
9. Other interactions are the same as those of Outgoing Calls by MF Signaling.
10. An 8RST-K circuit card is used.

Programming

STEP 1: **ARTD** - Assign the Route Data as shown below.

RT: 4

1-OSGS	:7	2-ONSG	:4	3-ISGS	:7	4-INSG	:4	5-TF	:3
6-TCL	:5	7-L/T	:1	8-RLP	:2	10-SMDS	:1	13-AC	:1
15-LSG	:5	19-ANI	:1	30-PAD	:4	43-BT	:1	73-KPST	:7
74-KPPT	:13	75-STC	:15	77-MT	:1				

Note: To change the data for existing route (RT), initialize the circuit card.

STEP 2: **ATRK** - Assign RT 905 to LENS for TK 1, TK 2 (depending on Register Location). Assign RT 904 to LENS for TK 1, TK 2 (depending on Register Location).

STEP 3: **AAND** - Assign the following parameters:

RT: 0 (Outgoing RT Number)

AT: 0 (0/1: Station/Individual Attendant)

CC: 0 (0/1: Main Central/Satellite Central) Programming is only necessary at the Main Node.

COC: (Central Office Code, CCIS Office Code, or first digit of CCIS Remote Station Number). Use CCIS Office Code.

STNL: (Number of Digits in Caller's Number)

SKIP: (Number of Digits to be skipped from Caller's Number) This strips the Office Code from being sent out in front of the Calling Station Number.

ADD: 3 (Number of Digits to be added to the ANI information).

DC: 753 (ANI Sending Number)

Note: Using this programming, if the call was originated at Remote Node Station Number 2000, the number being sent to the E911 Central Office would be "753-2000"

STEP 4: **AANDE** - Assign the following parameters

TN: (1-255)

Skip: Call Originator digit

ADD: Office Code to be added (1-10)

DC: Office Code (1-10)

E-22 Emergency System Control

General Description

This feature provides the emergency terminating method to Attendant Console and the originating call restriction.
Operator calls which are originated from Local Nodes belonging to a Hub node during an emergency mode terminate to VMS, IVR or the specific ATTCON in the Main node, and the operator there will answer and operate.
All outgoing C.O. calls originated during an emergency mode are restricted from Local Nodes belonging to a Hub node.

Note: This feature is only available in Australia.

Operating Procedure

No manual operation is required.

Service Conditions

1. Assignment of the emergency mode
   a.) Designation of mode switching and the terminated Attendant Console may be made to the relevant Hub node from NMS/CMAT remotely.
   b.) Assignment of the emergency mode may be changed by the Attendant operator in the Main node or a maintenance administration staff.
   c.) When a Main node is set in the emergency mode, only the Main node is treated as the emergency mode.
   d.) When a Hub node is set in the emergency mode, not only the Hub node but also its Local Nodes are treated as the emergency mode.
   e.) Set/cancellation will be informed by the system message and MN lamp on the TOPU.
2. The terminating method to the Attendant Console during an emergency mode.
   a.) When the Main node is changed into the emergency mode during the Day mode:
      - all incoming C.O. calls (LDN) are not transferred to Another main node but terminate to VMS(UCD), IVR (monitor number) or the specific Attendant Console in the Main node.
      - ATND calls or transferred calls are indicated their termination on the specific ATTCON in the Main node.

   b.) When the Main node is changed into the emergency mode during the Night mode:
      - all incoming C.O. calls (LDN) terminate to VMS (UCD), IVR (monitor number) or the specific Attendant Console in Another main node.
      - ATND calls or transferred calls terminate to the specific ATTCON in Another main node.

   c.) When the Hub node is changed into the emergency mode during the Night mode:
      - incoming C.O. calls to the Hub or Local Nodes are not transferred to the Main node but terminate to VMS (UCD), IVR (monitor number) or the specific Attendant Console in the Hub node.
      - ATND calls or transferred calls terminate to the specific ATTCON in the Main node.

   d.) When the Hub node is changed into the emergency mode during the Day mode:
      - incoming C.O. calls to the Hub or Local Nodes are not transferred to the Main node but terminate to VMS (UCD), IVR (monitor number) or the specific Attendant Console in the Hub node.
      - ATND calls or transferred calls are indicated their termination on the specific ATTCON in the Hub node.

e.) If the specific Attendant Console is set as emergency termination, another ANSWER key may be selected instead of that for LDN, ATND calls or transferred calls on an office date basis.

3. Connection restrictions during the emergency mode.
   a.) When a Hub node is set in the emergency mode, the offices in the Hub node is forced to be changed into the "Priority Class of Service" mode. If the offices have already been in the "Priority Class of Service" mode, they will be in the same as the Night mode.
   b.) In the above case, originating calls from a station in the Hub or Local Node to the Central Office or CCIS are restricted.
   c.) In the above status, it is permitted to originate a call by dialing the AUTHORIZAION CODE or from the station in the special class.
4. The following table is showing service conditions 2,3.

Assigning node / Call	Main node		Hub node			Main node	Hub node	Local Node
	Day mode	Night mode	Day mode	Night mode	Station	Permitted	Permitted	Permitted
ATND	Main specific ATT	Another main specific ATT	Hub specific ATT	Main specific ATT	**OG(C.O.)**	Permitted	Restricted	Restricted
LDN	Main VMM	Another main VMM	Hub VMM	Hub VMM	**OG(Tie)**	Permitted	Restricted	Restricted
CF-ATT	Main specific ATT	Another main specific ATT	Hub specific ATT	Main specific ATT	**IC**	Permitted	Permitted	Permitted

5. If trouble is caused on the CCIS, the ISDN Central Office line is used as the alternate route.
6. An incoming C.O. call (LDN) through the MFC trunk can not be transferred in an emergency mode.
7. All calls originated to the specific Attendant Console are treated as the incoming calls in an emergency mode. (Incoming calls originated by dialing the specific Attendant Console number can not be differentiated from incoming calls in an emergency mode.)
8. If the Attendant Console in the Main (or Hub) node transfers the call, which was originated from a station in Another node (e.g. Local Node) during an emergency mode, to Another main node via CCIS, the transferring Attendant Console will be recalled when the transferred call is not answered. If the transferred call is which originated from a station within the office (Main or Hub node), even though it is not answered, recall to the transferring Attendant Console will not be made.

E-26D Expanded Multiple Line Operation -D

General Description

This feature allows operation of whichever lines from ELC Cards in the same IMG or in different IMGs in the same Node/CCN/RN, as the Dterm's programmable Multiple Lines (My Line + Prime Line/Sub Lines). While Multiple Line Operation -D [M-20] only allows such programming within the reach of ELC Cards in the same IMG, this feature enables Multiple Line Operation beyond the reach of a single IMG (though the ELCs must be in one Node/CCN/RN).

Operating Procedure

No manual operation is required.

Service Conditions

1. To use this feature (operation of Multiple Lines beyond different IMGs), a PA-16ELCJ-B/PA-16ELCN/PA-16ELCN-A/CH-16ELCA card is required.
2. (Conditions for PA-16ELCJ-B/PA-16ELCN/PA-16ELCN-A) Depending on the switch setting on the card, assignable Multiple Lines vary.
   a.) SW 10, Element 1 = ON:
   Dterm's My Line and the other Lines (other Analog Telephones/Dterm) accommodated on this card must be in the same IMG.

   Note: Even if Element 1 of SW10/11 is ON, data programming for this feature (Multiple Lines beyond different IMGs) is possible. However, the data does not become effective until the switch is manually switched to OFF.

   b.) SW 10, Element 1 = OFF:
   Dterm and other Lines (other Analog Telephones/Dterm) accommodated on this card can be in different IMGs, yet in the same Node/CCN/RN. For more details on PA-16ELCJ-B/PA-16ELCN/PA-16ELCN-A card, refer to the Circuit Card Description.
3. Three kinds of lines can appear on Line/Feature key of the Desktop terminal:
   - My Line (ML): This Line is associated with the Extension Number assigned to the port on the Interface Circuit.
   - Prime Line (PL): This Line is given priority for Outgoing and Incoming Calls. Prime Line is automatically seized when going off-hook.
   - Sub Line (SL): This Line designates any other Line appearing on a Dterm. My Line will serve as a Prime Line, unless differentiated in system data programming.
4. A maximum of 32 ELC cards (a total of 512 Ports) can be used for this feature (Multiple Lines beyond different IMGs), per Node/CCN/RN.

   Note: This is not the case if the Multiple Lines are programmed in a single IMG.

5. If the system is a 1-IMG, this feature is not available.
6. A Virtual Line (Software Line Appearance [S-33]), Single Line Telephone or Dterm can be assigned as a Sub Line.

7. To use this feature, flag "1" to ASYD, SYS1, Index 321, Bit 0.
8. When KD = 1 (Intercom) or KD = 2 (DSS key) is assigned on AKYD, this service is not available. Be sure to assign KD = 0 (Line).
9. When the Caller Number is displayed on the bottom line of the LCD, ASYD, SYS1, Index 390, Bit 2 must be "1."
10. Restriction of Call Termination on My Line:
    A Desktop terminal can deny any Incoming Call to its My Line, when a call is made/received via one of its Sub Lines.

    Conditions:

    a.) Restricts a call to My Line only. (A call to Sub Line is not restricted.)
    b.) Available when the caller is a Station, TIE Line, Central Office Line or Attendant Console.
    c.) Available for a call as the destination of Call Forwarding - All Calls.
    d.) When the Original Call on the Sub Line is placed on hold:
    • If the terminal is off–hook after placing the call to Sub Line on hold, a call to My Line is restricted.
    • If the terminal goes on–hook while the call to Sub Line is still on hold, a call to My Line is not restricted.

Interactions

1. This feature is available with the following services:

CALL HOLD	[C-6, 6D]
CALL FORWARDING - BUSY LINE	[C-2, 2D]
CALL FORWARDING - DON'T ANSWER	[C-3, 3D]
CALL FORWARDING - ALL CALL	[C-5, 5D]
VOICE CALL	[V-2]
CALL PICK UP - GROUP	[C-7, 7D]
BOSS - SECRETARY OVERRIDE -D	[B-5D]
BOSS - SECRETARY - MESSAGE LAMP CONTROL	[B-8]
BOSS - SECRETARY TRANSFER	[B-11]
STATION HUNTING - CIRCULAR	[S-7]
STATION HUNTING - SECRETARIAL	[S-8]
STATION HUNTING - TERMINAL	[S-9]
THREE WAY CALLING	[T-2, 2D]
CALL TRANSFER	[C-11]
CONSULTATION HOLD - ALL CALLS	[C-17]
BLIND TRANSFER TO ATTENDANT	[B-17]
BLIND TRANSFER TO STATION	[B-18]
SERIAL CALL	[S-15]
SERIAL CALL - LOOP RELEASE	[S-64]
SERIAL CALL - CCIS	[S-52]
SERIAL CALL -D	[S-127]
EXECUTIVE RIGHT-OF-WAY	[E-1]
CALL BACK	[C-1]
MESSAGE REMINDER	[M-19]
BROKERAGE - HOTLINE	[B-6]
SPLIT CALL FORWARDING	[S-99]
CALL WAITING - TERMINATING	[C-12]

CALL WAITING - ANSWER -D	[C-27D]
CALL WAITING - ORIGINATING	[C-31]
GUEST NAME DISPLAY -D	[G-6D]

2. This feature does not support OAI/ACD related services.
3. This feature is supported by Hotel related services.
4. If Multiple Lines are programmed beyond different IMGs, this feature does not provide PRIVACY RELEASE [P-18].
5. FOLLOW PHONE [F-31] is not available for this feature.
6. When Multiple Lines are programmed on a Dterm using SID TO TERMINATING USER - DISPLAY [S-93], CALLING PARTY NUMBER - NAME ASSIGNMENT AND DISPLAY [C-157] or NAME DISPLAY - SYSTEM [N-28], ASYD, SYS1, Index 390, Bit 2 must be "1".

Programming

STEP 1: **ASYD** - Flag "1" to SYS1, Index 321, Bit 0 (Extended Multiple Line Operation is allowed).
SYS1, Index 390, Bit 2 (Show bottom line of LCD used for caller number).

STEP 2: **ASDT** - LENS equipped with a Dterm Circuit Board must be assigned a Station Number and a Telephone Equipment Class (TEC) of 12.

STEP 3: **AKYD** - Assign a My Line, a Prime Line, and various Sub Lines to the Dterms.

→TN: 1
→STN: Station Number
TP: Type of Dterm (0-3)
PL TN STN: Prime Line Tenant and Station Number
KYN: Key Number (1-40)
KYI: Service Index
0: Key Not Used
1: Feature Key
2: Multi Line Key
KD: Not assigned for Feature Key
TN STN: Tenant Number and Station Number of Multi Line
RG: Ringing Assignment per Multi Line Appearance
0: No Ringing
1: Ringing in Day Mode only
2: Ringing in Night Mode only
3: Not Used
4: Ringing in Day and Night Mode - 30 second delay.
5: Ringing in Day and Night Mode - 20 second delay.
6: Ringing in Day and Night Mode - 10 second delay.
7: Ringing in Day and Night Mode.

E-27 Expanded Individual Speed Dial

(Series 8600 software feature)

General Description

This feature allows a user to dial certain frequently called exchange network numbers using fewer digits (abbreviated call codes) than normally required. Using this feature, a user can establish his own abbreviated codes.

Note: This feature is available in North America only.

Operating Procedure

1. Lift handset or press the SPEAKER key. Receive Dial Tone.
2. Dial Expanded Individual Speed Dial access code (one to three digits).
3. Dial the abbreviated call code (one to three digits). Call is completed.

To use via Feature key

1. Lift handset or press the SPEAKER key. Receive Dial Tone.
2. Press the EXPANDED INDIVIDUAL SPEED DIAL Feature key (FKY=146).
3. Dial the abbreviated call code (one to three digits). Call is completed.
4. Lift handset or press the SPEAKER key. Receive Dial Tone.

To set using the primary station only

1. Lift handset. Receive Dial Tone.
2. Dial Expanded Individual Speed Dial. Enter feature access code (one or two digits).
3. Dial the abbreviated call code (one to three digits). At this point no tone is returned.
4. Dial the outgoing number, including the route access code. A brief pause will occur before receiving Service Set Tone.

Service Conditions

1. Expanded Individual Speed Dial codes are programmed into RAM memory per local processor.
2. Each abbreviated call code can store a maximum of 24 digits, including access codes and pauses.
3. A maximum of 1400 stations per IMG can be equipped with Expanded Individual Speed Dial and Expanded Individual Speed Dial - Group.
4. 1400 memory blocks per IMG are assigned for Expanded Individual Speed Dial. A maximum of 10 numbers can be stored in each memory block. This allows for a total of 14,000 numbers per IMG. 14,000 includes Expanded Individual Speed Dial and Expanded Individual Speed Dial - Group.
5. Each station can be assigned ten Memory Blocks for Speed Calling. Therefore, it can be programmed for 100 numbers maximum.
6. Toll Denial/Toll Diversion [T-6] and Toll Restriction - 3/6-Digit [T-7] may or may not be applied to Expanded Individual Speed Dial, as desired.
7. Least Cost Routing - 3/6-Digit [L-5] may be provided via Expanded Individual Speed Dial access.
8. Expanded Individual Speed Dial data will be retained in memory (even if the system is initialized or in the event of a power failure) by means of a hard disk for a system.
9. A pause may be programmed, using the "*" key of a PB pad. This will take one of the 24 digits available. Rotary type phones cannot store a pause. A pause lasts about 3 seconds.

10. It is not necessary to erase a stored number in order to store another number. The second number replaces the first.
11. When a caller places a call via OGCA, and when the first outgoing route and the station setting Expanded Individual Speed Dial are restricted, this service cannot be set. In this case, the service can be set via the following procedure:
    a.) Assign a dummy route.
    b.) Clear the restriction between the dummy route and all the stations.
    c.) Assign the dummy route as the first route of OGCA.
12. This feature cannot be the target of a Call Forwarding Assignment - Attendant call.
13. When the attendant console transfers a call outside via Expanded Individual Speed Dial and the held party is a Dterm, the holding party will display Expanded Individual Speed Dial access code, plus the bin number when the call is released from the attendant console. The called number will not display.
14. The maximum number of digits that can be entered in Expanded Individual Speed Dial is 24. If the maximum length is exceeded, Reorder Tone will be returned.

Interactions

1. Blocks are not assigned sequentially (1-1400), but are assigned as a quantity of blocks in AIEA. These quantities will never exceed the value of 10.
2. If one block is assigned, the abbreviated call code will be digits 0-9. If two blocks are assigned, the abbreviated call code will be digits 00 - 19. If three blocks are assigned, the abbreviated call code will be digits 00 - 29, to a maximum of 10 blocks with Abbreviated Call Codes 00 - 99.
3. Only the primary station will be allowed to use the Expanded Individual Speed Dial access code. When assigning the abbreviated Expanded Individual Speed Dial access code and number, a delay of six to eight seconds will occur until Service Set Tone is returned.
4. Only those stations assigned as secondary stations are considered to be members of the Expanded Individual Speed Dial - Group.
5. Stations of an Expanded Individual Speed Dial must be in the same IMG (LP). Error 0x256, “IMG Number of My Line/Primary Station Mismatch” is displayed.
6. Expanded Individual Speed Dial requires the assignment of senders, Route 905. See Universal Sender [U-3] to use outgoing routes.
7. Outgoing Trunk Queuing [O-2] - This feature may be used with Expanded Individual Speed Dial - Group if all trunks are busy.
8. Route Advance [R-6] and Least Cost Routing - 3/6-Digit [L-5] may be used with this feature. The ASPA access code to either Route Advance [R-6] or Least Cost Routing - 3/6-Digit [L-5] will be stored in the memory of the Expanded Individual Speed Dial number.
9. If Expanded Individual Speed Dial is used with OAI make call facility, only LCR, LCR(S), or stations are supported destinations.

Programming

STEP 1: **ASYD** - SYS2, Index 1, Bit 3. Does Toll Restriction apply to Expanded Individual Speed Dial? 0/1: Yes/No. (If yes, refer to Toll Restriction - 3/6-Digit [T-10]).

STEP 2: **ANPD** - Reserve a number level for feature access. Select Normal Service (N), Hooking Service (H) or Busy Service (B) from the combo box for Connection Indexes. (applicable to any feature assigned to this level). Assign NND in accordance with the numbering plan.

STEP 3: **ASPA** - Two access codes must be assigned for this feature.

For the Expanded Individual Speed Dial number assignment, assign the tenant and Access Code (ACC). Assign for a Connection Index of Normal (N). For SRV = "SSC: Service Code Appendix", assign SID 134.

For individual station speed calling access, assign the tenant and Access Code (ACC). Assign for a Connection Index of "N: Normal". For SRV = "SSCA: Service Code Appendix", assign SID 135.

STEP 4: **ASFC** - Assign a Service Feature Restriction Class that allows SFI 197 to the stations to receive Expanded Individual Speed Dial.

STEP 5: **AIEA** - Program the tenant, station, type of station, and the number of Expanded Individual Speed Dial blocks to be assigned to the primary station.

TN:	Tenant Number
STN:	Station Number
STATION TYPE:	Primary/Secondary
	1: MST (Master Station)
	2: SLV (Slave Station)
BLOCK:	Number of Blocks
MST STN:	When assigning secondary station, assign the primary station number.

STEP 6: **AKYD** - Assign the EXPANDED INDIVIDUAL SPEED DIAL Feature key (FKY=146)

STEP 7: **ADSL** - Assign G_SPD key. Assign SN = 1 (Originating; DT Connection) and EXPANDED INDIVIDUAL SPEED DIAL Feature key (FKY=146).

This command is used for assigning a function to each Soft key.

STEP 8: **AIED** - This command is used to assign, delete, and display the destination number of Expanded Individual Speed Dial.

FCCS Programming

This data setting describes only the additional or replacement commands used to activate FCCS services. When used as an FCCS feature, these commands replace the non-network commands (for example, AAED is replaced by AAEDN). Note that the data setting for FCCS service varies depending on the node.

STEP 1: **ANPDL** - Reserve a number level for feature access. Select Normal Service (N), Hooking Service (H), or Busy Service (B) from the combo box for Connection Indexes. (applicable to any feature assigned to this level). Assign NND in accordance with the numbering plan.

STEP 2: **ASPAL** - Two access codes must be assigned for this feature.

For the individual station speed calling number assignment, assign the tenant and Access Code (ACC). Assign for a Connection Index of "N: Normal". For SRV = "SSC: Service Code", assign SID 14.

For individual station speed calling access, assign the tenant and Access Code (ACC). Assign for a Connection Index of "N: Normal". For SRV = "SSC: Service Code", assign SID 21.

When using ASYDL or ASYDN to program an FCCS feature, the basic programming of ASYD is required in addition to ASYDL or ASYDN.

E-28 Expanded Individual Speed Dial - Group

(Series 8600 software feature)

General Description

This feature allows a user to share a set of common Speed Dial numbers with other users in the group.

Note: This feature is only available in North America.

Operating Procedure

1. Lift handset or press the SPEAKER key. Receive Dial Tone.
2. Dial Expanded Individual Speed Dial access code (one to three digits).
3. Dial the abbreviated call code (one to three digits). Call is completed.

To use via Feature key

1. Lift handset or press the SPEAKER key. Receive Dial Tone.
2. Press Feature key Expanded Individual Speed Dial [146].
3. Dial the abbreviated call code (one to three digits). Call is completed.
4. Lift handset or press the SPEAKER key. Receive Dial Tone.

To set using the primary station only

1. Lift handset. Receive Dial Tone.
2. Dial Expanded Individual Speed Dial. Enter feature access code (one or two digits).
3. Dial the abbreviated call code (one to three digits). At this point no tone is returned.
4. Dial the outgoing number, including the route access code. A brief pause will occur before receiving Service Set Tone.

Service Conditions

1. Expanded Individual Speed Dial - Group codes are programmed into RAM memory per local processor.
2. Each abbreviated call code can store a maximum of 24 digits, including access codes and pauses.
3. A maximum of 1400 stations per IMG can be equipped with Expanded Individual Speed Dial and Expanded Individual Speed Dial - Group.
4. 1400 memory blocks per IMG are assigned for Expanded Individual Speed Dial. A maximum of 10 numbers can be stored in each memory block. This allows for a total of 14,000 numbers per IMG. 14,000 includes Expanded Individual Speed Dial and Expanded Individual Speed Dial - Group.
5. Expanded Individual Speed Dial - Group primary station will be assigned memory blocks, and associate stations in that group are assigned as secondary stations. The primary station has the capability to program ten memory blocks that are commonly used as Speed Dial numbers for that group. A group can consist of a maximum of 100 Speed Dial numbers.
6. Toll Denial/Toll Diversion [T-6] and Toll Restriction - 3/6-Digit [T-7] may or may not be applied to Expanded Individual Speed Dial, as desired.
7. Least Cost Routing - 3/6-Digit [L-5] may be provided via Expanded Individual Speed Dial access.
8. Expanded Individual Speed Dial - Group data will be retained in memory (even if the system is initialized or in the event of a power failure) by means of a hard disk for a system.
9. It is not necessary to erase a stored number in order to store another number. The second number replaces the first. If using the AIED command, the number must first be deleted.

10. When a caller places a call via OGCA, and when the first outgoing route and the station setting Expanded Individual Speed Dial are restricted, this service cannot be set. In this case, the service can be set via the following procedure:
    a.) Assign a dummy route.
    b.) Clear the restriction between the dummy route and all the stations.
    c.) Assign the dummy route as the first route of OGCA.
11. When the attendant console transfers a call outside via Expanded Individual Speed Dial and the held party is a Dterm, the holding party will display Expanded Individual Speed Dial access code, plus the bin number when the call is released from the attendant console. The called number will not display.
12. The maximum number of digits that can be entered in Expanded Individual Speed Dial is 24. If the maximum length is exceeded, Reorder Tone will be returned.

Interactions

1. Blocks are not assigned sequentially (1-1400), but are assigned as a quantity of blocks in AIEA. These quantities will never exceed the value of 10.
2. If one block is assigned, the abbreviated call code will be digits 0-9. If two blocks are assigned, the abbreviated call code will be digits 00 - 19. If three blocks are assigned, the abbreviated call code will be digits 00 - 29, to a maximum of 10 blocks with Abbreviated Call Codes 00 - 99.
3. Only the primary station will be allowed to use the Expanded Individual Speed Dial access code. When assigning the abbreviated Expanded Individual Speed Dial access code and number, a delay of six to eight seconds will occur until Service Set Tone is returned.
4. Only those stations assigned as secondary stations are considered to be members of the Expanded Individual Speed Dial - Group.
5. Stations of an Expanded Individual Speed Dial must be in the same IMG (LP). Error 0x256, “IMG Number of My Line/Primary Station Mismatch” is displayed.
6. Expanded Individual Speed Dial requires the assignment of senders, Route 905. See Universal Sender [U-3] to use outgoing routes.
7. Outgoing Trunk Queuing [O-2] - This feature may be used with Expanded Individual Speed Dial - Group if all trunks are busy.
8. Route Advance [R-6] and Least Cost Routing - 3/6-Digit [L-5] may be used with this feature. The ASPA access code to either Route Advance [R-6] or Least Cost Routing - 3/6-Digit [L-5] will be stored in the memory of the Expanded Individual Speed Dial number.
9. If Expanded Individual Speed Dial is used with OAI make call facility, only LCR, LCR(S), or stations are supported destinations.

Programming

STEP 1: **ASYD** - SYS2, Index 1, Bit 3. Does Toll Restriction apply to Expanded Individual Speed Dial? 0/1: Yes/No. (If yes, refer to Toll Restriction - 3/6-Digit [T-10]).

STEP 2: **ANPD** - Reserve a number level for feature access. Select Normal Service (N), Hooking Service (H) or Busy Service (B) from the combo box for Connection Indexes. (applicable to any feature assigned to this level). Assign NND in accordance with the numbering plan.

STEP 3: **ASPA** - Two access codes must be assigned for this feature.

For the Expanded Individual Speed Dial number assignment, assign the tenant and Access Code (ACC). Assign for a Connection Index of “N: Normal”. For SRV = “SSCA: Service Code Appendix”, assign SID 134. For individual station speed calling access, assign the tenant and Access Code (ACC). Assign for a Connection Index of “N: Normal”. For SRV = “SSCA: Service Code Appendix”, assign

SID 135.

STEP 4: **ASFC** - Assign a Service Feature Restriction Class that allows SFI 197 to the stations to receive Expanded Individual Speed Dial.

STEP 5: **AIEA** - Program the tenant, station, type of station, and the number of Expanded Individual Speed Dial blocks to be assigned to the primary station.

TN:	Tenant Number
STN:	Station Number
STATION TYPE:	Primary/Secondary
	1: MST (Master Station)
	2: SLV (Slave Station)
BLOCK:	Number of Blocks
MST STN:	When assigning secondary station, assign the primary station number.

STEP 6: **AKYD** - Assign EXPANDED INDIVIDUAL SPEED DIAL Feature key (FKY=146)

STEP 7: **ADSL** - Assign G_SPD key. Assign SN = 1 (Originating; DT Connection) and EXPANDED INDIVIDUAL SPEED DIAL Feature key (FKY=146).

This command is used for assigning a function to each Soft key.

STEP 8: **AIED** - This command is used to assign, delete, and display the destination number of Expanded Individual Speed Dial.

FCCS Programming

This data setting describes only the additional or replacement commands used to activate FCCS services. When used as an FCCS feature, these commands replace the non-network commands (for example, AAED is replaced by AAEDN). Note that the data setting for FCCS service varies depending on the node.

STEP 1: **ANPDL** - Reserve a number level for feature access. Select Normal Service (N), Hooking Service (H), or Busy Service (B) from the combo box for Connection Indexes. (applicable to any feature assigned to this level). Assign NND in accordance with the numbering plan.

STEP 2: **ASPAL** - Two access codes must be assigned for this feature.

For the individual station speed calling number assignment, assign the tenant and Access Code (ACC). Assign for a Connection Index of “N: Normal”. For SRV = “SSC: Service Code”, assign SID 14.

For individual station speed calling access, assign the tenant and Access Code (ACC). Assign for a Connection Index of “N: Normal”. For SRV = “SSC: Service Code”, assign SID 21.

When using ASYDL or ASYDN to program an FCCS feature, the basic programming of ASYD is required in addition to ASYDL or ASYDN.

E-29 Expanded Account Code

(Series 8700 software feature)

General Description

This feature is an adjunct to STATION MESSAGE DETAIL RECORDING [S-10], which provides a station user with the capability to enter a cost-accounting or client-billing code (up to 24 digits) into the system before dialing a long distance number.

Note: This feature is only available for North America.

Operating Procedure

To enter an Account Code:

1. Lift the handset. Receive Dial Tone.
2. Dial the Feature Access Code. Receive Service Set Tone.
3. Dial the ACCOUNT CODE (when the account code is less than 24 digits, a "#" must be dialed).
4. Receive dial tone and dial the called number.

Note: Feature Access Code - Access Code for ACCOUNT CODE entry will be a maximum of three digits.

To enter an ACCOUNT CODE after an AUTHORIZATION CODE [A-20]:

1. Lift the handset. Receive Dial Tone.
2. Dial the Feature Access Code for AUTHORIZATION CODE. Receive Service Set Tone.
3. Dial the AUTHORIZATION CODE (up to 10 digits). Receive second Service Set Tone.
4. Dial the ACCOUNT CODE (up to 24 digits). Receive Dial Tone.
5. Dial the number desired.

Service Conditions

1. The expansion of Account Code length is on a per system basis, controlled by ASYDL as 10 or 24 maximum digits in Account Code.
2. The Attendant Console, before extending a trunk to a station, can enter an account code up to 24 digits, but only 10 digits will record for SMDR service.
3. FCCS Networks must have the same Account Code length for nodes. See Programming.
4. The maximum number of digits for an ACCOUNT CODE is 24 digits. There is no limitation to the number of account codes used per system. There is no change in the maximum amount of digits in Forced Account or Authorization codes.

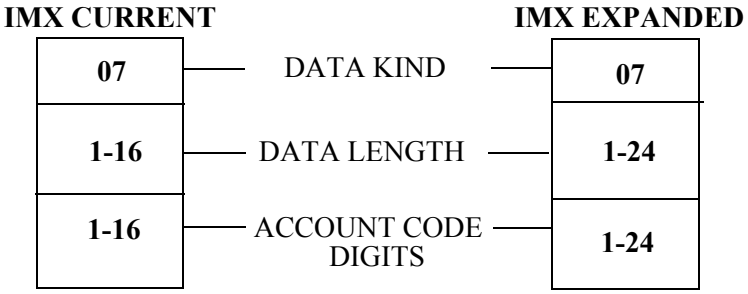
   Note: Service Set Tone after the Feature Access Code can be eliminated by System Data. System Data is common for ACCOUNT CODE, AUTHORIZATION CODE [A-20] and FORCED ACCOUNT CODE [F-7].

5. A station user can enter an ACCOUNT CODE of fewer digits than the maximum length defined and indicate the end of entry by pressing the # key.
6. The Expand Account Code feature is not compatible with Hotel systems (ASYD, SYS1, Index 160, Bit 0=1).

7. This feature is not available for station to station SMDR and account code.
8. Only IMX type SMDR format (ASYDL, SYS1, Index 578, Index 579, Index 580 and Index 581, Bit 0=1) can be used to record expanded account code to SMDR. All other formats record only the first 10 digits.
9. The user may decide if ACCOUNT CODE follows the AUTHORIZATION CODE (per tenant basis) however the ACCOUNT CODE maximum length is 24 per system.
10. ACCOUNT CODE dialing is also effective while in CONSULTATION HOLD [C-17]. Special dial tone is received after entering the ACCOUNT CODE.
11. Stations are assigned this the Account Code feature via Class of Service, account code length is assigned on a system wide basis.
12. ACCOUNT CODES can be output in the SMDR record for calls handled with OUTGOING TRUNK QUEUING [O-2] and OFF-HOOK QUEUING [O-7].
13. When the expanded account feature is selected, by system data, the maximum amount of digits is expanded to 24, if the amount of dialed digits exceed 24 reorder tone is returned.
14. When this feature is used in conjunction with STATION MESSAGE DETAIL RECORDING [S-10] the digit field for account code will expand from a maximum of 16 to a maximum of 24 digits.
    a.) Change in the IMX SMDR TEXT FORMAT on KIND-OF-CALL BASIS as outlined in the NEC SYSTEM MESSAGE DETAILED RECORD Interface Specification. **Note 1**
    b.) The expanded Account Code feature is System Data (ASYDL, SYS1, Index 805, Bit 5) controlled.

Note 1: Change in IMX SMDR format for data kind 07 (Account Code)

Current Method:
Data Length = 01 - 16
Account Code Data = 01 - 16
New Method:
Data Length = 01 - 24
Account Code Data = 01 - 24



Interactions

1. If FORCED ACCOUNT CODES [F-7] are used, ACCOUNT CODES can not be used.
2. ACCOUNT CODE is also effective while in CONSULTATION HOLD [C-17]. A special dial tone is received after the ACCOUNT CODE.
3. When using this feature on a single-line telephone or Dterm, the user can use the SPEED CALLING SYSTEM,GROUP, INDIVIDUAL or SPEED CALLING-ONE TOUCH (Dterm) to complete the call.
4. ACCOUNT CODES cannot be entered by a station upon answering an incoming trunk call.
5. The access code for ACCOUNT CODES is restricted to a maximum of three digits.

6. STATION MESSAGE DETAIL RECORDING [S-10] Split Billing (System Data Option): On transferred calls, or incoming calls via the Attendant Console, the ACCOUNT CODE will be recorded in all the call records.
7. This feature is not compatible with Centralized Billing - CCIS [C-55]. The maximum ACCOUNT CODE length for SMDR Centralized Billing - CCIS [C-55] is 10 digits.
8. When OUTGOING TRUNK QUEUING [O-2] is initiated, the ACCOUNT CODE will be recorded when the call is completed.
9. When a Dterm user initiates the LAST NUMBER DIALED [L-6D] feature, the ACCOUNT CODE will be recorded, if previously dialed.
10. Change in IMX SMDR format for data kind 07 (Account Code).
    Current 07 data is 01 - 16 account code length - account code data maximum of 16 digits.
    New 07 data is 01 - 24 account code length - account code data maximum of 24 digits. THERE IS NO CHANGE FOR ICS standard or expanded SMDR FORMAT.

Programming

STEP 1: **ASYD** - SYS1, Index 43, Bit 2. Is service set tone sent after the ASPA access code for ACCOUNT CODE? 0/1 = No/Yes. Assign data "1".

SYS 2, Index 3, Bit 4. Is ACCOUNT CODE used with Authorization Code? 0/1 = No/Yes.

STEP 2: **ASYDL** - SYS1, Index 805, Bit 5. Use 10 or 24 digits Account Code. 0/1 = 10 digits/24 digits.

SYS1, Index 578, Index 579, Index 580 and Index 581, Bit 0=1 for IMX type SMDR output.

STEP 3: **ANPD** - Reserve a number level for service feature access. Select Normal Service (N) or Hooking Service (H) from the combo box for Connection Indexes.(applicable to any service feature assigned to this level).

Assign NND in accordance with a predetermined numbering plan.

STEP 4: **ASPA** - For SRV = "SSC: Service Code", SID = 41, assign the access code (maximum of three digits). Assign the number of digits in the ASPA service feature access code plus the number of digits that in the ACCOUNT CODE for the NND parameter.

When ACCOUNT CODE operates with AUTHORIZATION CODE [A-20], the above assignment is omitted. In ASPA, assign for SRV = "SSC: Service Code", SID = 42.

The NND value will include the number of digits in the ASPA service feature access code plus the number of digits in the ACCOUNT CODE.

For example:

45 - 511234

2 + 6 = 8

ACC + ACCT CODE = NND

STEP 5: **ASFC** - For Stations to use ACCOUNT CODE, assign a Class of Service that allows SFI 27.

STEP 6: **AMND** - If Authorization Codes are used with ACCOUNT CODES, then for Tenant 0, assign the Destination Code (DC) as the first digit of all Authorization Codes and the Maximum Necessary Digits (MND) of the Authorization Codes. For the above example, a DC of six is assigned with an MND of six. For standard ACCOUNT CODE there is no entry required.

STEP 7: **AATC** - Assign when Authorization Code is used.

FCCS Programming

This data setting describes only the additional or replacement commands used to activate FCCS Services. When used as an FCCS feature, these commands replace the non-network commands (for example, AAED is replaced by AAEDN). Note that the data setting for FCCS Service varies depending on the Node.

STEP 1: **ASYDN** - SYS1, Index 805, Bit 5. Use 10 or 24 digits Account Code. 0/1 = 10 digits/24 digits.

STEP 2: **ANPDL** - Reserve a Number Level for Service Feature Access. Select Normal Service (N), Hooking Service (H), or Busy Service (B) from the combo box for Connection Indexes. (applicable to any feature assigned to this level). Assign NND in accordance with a predetermined Numbering Plan.

STEP 3: **ASPAL** - For SRV = "SSC: Service Code" SID = 41, assign the Access Code. Assign the number of digits in the ASPAL Service Feature Access Code plus the number of digits that will appear in the ACCOUNT CODE for the NND parameter. When ACCOUNT CODE operates with AUTHORIZATION CODE [A-20], the above assignment is omitted. In ASPAL, assign for SRV = "SSC: Service Code", SID = 42. The NND value will include the number of digits in the ASPAL service feature access code plus the number of digits that appear in the ACCOUNT CODE.

Note: When using ASYDL or ASYDN to program an FCCS feature, the basic programming of ASYD is required in addition to ASYDL or ASYDN.

F-1 Flexible Numbering of Stations

General Description

This feature provides the ability to assign Telephone Number of the Voice Station and Data Station to any corresponding instrument location, depending solely upon Numbering Plan limitations.

Operating Procedure

Refer to the Command Manual.

Service Conditions

1. Careful consideration should be given to any numbering assignment, especially in larger systems, to avoid the unnecessary loss of Access Codes or duplication of Telephone Numbers. There is no limit to the number of groups that can be assigned per system.
2. Where TENANT SERVICE [T-12] is provided, Telephone Numbers may not be duplicated between Tenants. A combination of one, two, three, four and five-digit numbering may be used.

   Example: a. 3 1
   b. 3 2 1
   c. 3 2 2 1

   Example a., b., c., can be assigned as three different Stations.
3. BUSY LAMP FIELD - FLEXIBLE [B-2] Service can be provided only when the defining Telephone Number digits exceed two.
4. The FCCS Network allows the user to use a Telephone Number in addition to the existing Station Numbers. FCCS features are activated when a Telephone Number is dialed. A Telephone Number, which can be assigned on a Station basis, is a Unique Number on the FCCS Network.

Note: A maximum of 16 digits can be used as a Telephone Number. However, the maximum of digits may vary depending on features or systems. For details, refer to the related manuals.

Interactions

Caution should be exercised when programming this feature. Large quantities of potential Telephone Number are consumed. As programmed below, the system will be equipped with just one two-digit Station and one three-digit Station.

Programming

STEP 1: **ASYD** - SYS1, Index 16. Enable one to five digits Station Numbering for the system. Assign 3FH. SYS1, Index 92, Bit 3. Must be assigned as data "1".

STEP 2: **ANPD** - Reserve an Access Code for Station Numbering. For the NND parameter, assign the NND equal to the smallest quantity of digits planned for Station Numbering that will use the Busy Lamp Field. Only the Stations with the NND assigned here will appear on the BLF.

STEP 3: **ASPA** - Assign the Station Numbering in such a way that the system will differentiate the Station Numbering via the Access Codes programmed here. The following data is programmed:

For STATION 31	For STATION 321	For STATION 3221
TN: 1	TN: 1	TN: 1
ACC: 31	ACC: 321	ACC: 322
CI = "N: Normal, "H: Hooking"	CI = "N: Normal, "H: Hooking"	CI = "N: Normal, "H: Hooking"
SRV = "STN: Station Number"	SRV = "STN: Station Number"	SRV = "STN: Station Number"
NND: 2	NND: 3	NND: 4

The following Access Codes should also be assigned: 323, 324, 325, 326, 327, 328, 329, 320. CI: "N: Normal", "H: Hooking"; SRV = "STN: Station Number"; and NND: 4 will be assigned with each of the above.

F-2 FX Access

General Description

This feature provides dial access to distant Central Offices using Foreign Exchange (FX) Trunks. All Incoming Calls to the System from the FX Central Office area are placed to the listed Foreign Exchange Directory Number and are answered by the Attendant in a manner similar to LDN Service. Outgoing Calls can be made on an Attendant-handled basis using direct or dial access and/or on a direct basis by Station.

Note: This feature is only available in North America.

Operating Procedure

To place an Outgoing FX Call from a Station

1. Lift the handset and receive Dial Tone.
2. Dial the FX Trunk Access Code and receive Second Dial Tone.
3. Dial the desired Telephone Number.

To answer an Incoming FX Call at the ATTENDANT CONSOLE [A-3]

1. The FX Lamp flashes.
2. An Audible Signal is received.
3. Press the FX or ANSWER key to answer.
4. Process the call as usual.

Service Conditions

1. One INCOMING CALL IDENTIFICATION [I-2] key may be used for all FX Groups.
2. Access to various types of Trunks may be denied to certain Stations according to CLASS OF SERVICE - INDIVIDUAL [C-15] programming.
3. TOLL DENIAL/TOLL DIVERSION [T-6] may be provided for FX ACCESS [F-2].
4. TOLL RESTRICTION - 3/6-DIGIT [T-7] may be provided for FX Trunks on a per Route basis.
5. Care should be exercised in System Data assignment when utilizing this feature in conjunction with LEAST COST ROUTING - 3/6-DIGIT [L-5], since Foreign Exchange Trunks may require that “1” be dialed as a prefix to the desired number.

Interactions

1. LEAST COST ROUTING - 3/6-DIGIT [L-5]: FX Trunk Groups may be assigned to a Least Cost Routing list.
2. ROUTE ADVANCE [R-6]: This feature routes Outgoing FX Calls over alternate facilities when the first-choice Trunk Group is busy.
3. UNIFORM CALL DISTRIBUTION [U-1]: This feature permits Incoming Calls, including FX Calls, to be terminated directly from the Central Office to the next Idle Station in a prearranged group of Stations without Attendant assistance.
4. DIRECT-IN TERMINATION [D-7]: This feature allows one FX Trunk to terminate directly to a designated Station. Incoming FX Calls to this Station are queued if the Station is busy.

5. OUTGOING TRUNK QUEUING [O-2]: This feature provides a means for users to dial a busy FX Trunk Group to be automatically placed in a queue, and to be called back when a Trunk in the Trunk Group becomes available.
6. When the system is in the Night mode of operation, Incoming FX Calls can be answered using NIGHT CONNECTION - FIXED [N-1], FLEXIBLE [N-2] or TRUNK ANSWER FROM ANY STATION [T-8].
7. DIRECT INWARD DIALING [D-8] is not provided for FX Trunks. All calls are normally routed to the Attendant, UCD Group, or a CAS Attendant.
8. Outgoing FX Trunk Calls can be restricted from preselected Stations by the assignment of one or more of the following restrictions:
   a.) Restriction from Outgoing Calls
   b.) Miscellaneous Trunk Restrictions
   c.) Toll Restriction
9. FX Central Office Access: This service provides access to a distant Central Office via FX Trunks. All Incoming Calls to the system (from the FX, Central Office Area) are placed to the listed FX Directory Number and are answered by the Attendant in a similar way to Listed Directory Number (LDN) Service. Outgoing Calls are made on an Attendant-handled basis or on a dial access basis by Stations.
10. A unique Attendant Console INCOMING CALL IDENTIFICATION (ICI) [I-2] may be displayed for each Local and FX LDN.
11. A unique Attendant Console Trunk Group Busy Lamp may be assigned to indicate the busy/idle status for an FX Trunk Group.
12. The COT Circuit Packages are used to provide the Trunk Interface Circuit.
13. An FX Route may be used with any other feature available to any other Route, including the following:

DIRECT OUTWARD DIALING	[D-9]
REMOTE ACCESS TO SYSTEM	[R-2]
TRUNK-TO-TRUNK CONNECTION	[T-10]
TRUNK ANSWER FROM ANY STATION (TAS)	[T-8]
TOLL RESTRICTION - 3/6 DIGIT	[T-7]

Programming

STEP 1: **ARTD** - Assign the FX ACCESS Route as shown below:

GROUND START:

RT: 1

1-OSGS	:2	2-ONSG	:3	3-ISGS	:1	4-INSG	:3	5-TF	:3
6-TCL	:2	7-L/T	:1	8-RLP	:2	15-LSG	:1		

LOOP START:

RT: 31

1-OSGS	:2	2-ONSG	:3	3-ISGS	:1	4-INSG	:3	5-TF	:3
6-TCL	:2	7-L/T	:1	8-RLP	:2				

Note: To change the data for existing route (RT), initialize or reboot the trunk device such as a circuit card or MG.

STEP 2: **ATRK** - Assign the LENS, FX ACCESS Trunk Route Number, Trunk Number, and Tenant Number.

STEP 3: **MBTK** - Assign the Make Idle Status to the FX Access Trunks.

STEP 4: **ANPD** - Reserve a Number Level for Trunk Access. Select Normal Service (N) or Hooking Service (H) from the combo box for Connection Indexes. Number of Necessary Digits (NND) is usually one or two. Busy Lamp Field is not activated.

STEP 5: **ASPA** - Assign the Access Code to the FX Route. The type of service is SRV = "OGC: Outgoing Call" for Outgoing Trunk. Assign the Route Number associated with this Access Code.

STEP 6: **ARSC** - The Route Restriction Class must be constructed in a manner to allow the Station access to the involved Routes.

F-3D Flash Button -D

General Description

This feature provides for a Dterm set to program one of Line/Feature keys to function as a FLASH BUTTON. The FLASH BUTTON will function in the manner a switch hook flash does for a Single Line Set.

Operating Procedure

When activating features common to both Dterm and Single Line sets

1. Press the FLASH BUTTON to receive Special Dial Tone.
2. Dial the Access Code of the desired feature.

Service Conditions

1. When using the FLASH BUTTON, the feature will not be activated again until digits are dialed. The FLASH BUTTON may function during a Two-way Connection.

   Note: If a transfer is attempted via the FLASH BUTTON, Three-way Conference will be established by pressing the FLASH BUTTON again. To retrieve the original connection after the connection with the Transferred Station, the TRANSFER key should be pressed.

2. The FLASH BUTTON is optional. Its use in various features will be explained throughout this document.

Interactions

1. This feature provides switch hook flash capabilities to a Dterm with a Connection Index (CI) of "B: Busy". See ASPA. The features in which a FLASH key are used include CALL BACK [C-1] and OUTGOING TRUNK QUEUING [O-2]. The FLASH BUTTON is usually required with the four-key Desktop terminal.
2. See ASYD, SYS3, Index 2 for programming the characteristics of a switch hook flash (Hooking).

Programming

STEP 1: **AKYD** - When assigning key data for the Desktop terminal, assign the following.

- TN: 1
- STN: Station Number
- TP: Type of Dterm (0-3)
- KYN: Key Number (1-40)
- KYI: Service Index
  - 0: Key Not Used
  - 1: Feature Key
  - 2: Multi Line Key
- KD: Not assigned for Feature key
- KYN: Feature Key Number
- FKY: 21 (Flash Button)

STEP 2: **ADSL** - SN = 5 (Consultation Hold) and Feature Key Flash Button [21].
This command is used for assigning the function to each Soft key.

F-4D Flash Entry -D

General Description

This feature allows a Desktop terminal user to insert a switch hook flash as the first digit in a stored SPEED CALLING - ONE TOUCH -D [S-26D] Number.

Operating Procedure

To add a flash to a stored SPEED CALLING - ONE TOUCH -D [S-26D] Number to a Desktop terminal

1. Stay on-hook.
2. Press the FEATURE key and SPEED CALLING - ONE TOUCH -D Feature key. The LCD display will either be blank, or display previously stored digits.
3. Press the RECALL key as the first digit. A switch hook flash is entered into the memory and the LCD displays:



4. If a Feature Access Code is to follow the Flash Entry, dial the desired digits.
5. Press the FEATURE key. The LCD displays:



To add a flash to a stored SPEED CALLING - ONE TOUCH [S-26D] Number to a Desktop terminal

1. Stay on-hook.
2. Press the FEATURE key and SPEED CALLING - ONE TOUCH Feature key. LCD will either be blank or display previously stored digits.
3. Press the RECALL key as the first digit. A switch hook flash is entered into the memory and the LCD displays:



4. If a Feature Access Code is to follow the FLASH ENTRY, dial the desired digits.
5. Press the FEATURE key. The LCD displays:



Service Conditions

1. A FLASH ENTRY may be entered only as a first digit. For example:
   a.) Call Transfer - ! + Telephone Number
   b.) Busy Service - + Calling Waiting - Originating Access Code, ! + Executive Right-of-Way Access Code
   c.) Call Hold - ! + Call Hold Access Code
2. A FLASH ENTRY, stored as a flash, may be used only once, unless digits are dialed between use.

Interactions

Refer to the Command Manual, ASYD, SYS3, Index 2 for programming the characteristics of a switch hook.

Programming

This is a Desktop terminal feature. It is inherent to the instrument and requires no programming via the PCPro.

F-5D Flexible Ringing Assignment -D

General Description

This feature enables Dterm Line Appearances to be individually programmed for ringing, not ringing, ringing (Day only), or ringing (Night only).

Operating Procedure

No manual operation is required once FLEXIBLE RINGING ASSIGNMENT has been programmed in System Data.

Service Conditions

1. Recall ringing supersedes all ringing possibilities and will ring regardless of other programming.
2. Incoming Central Office ringing supersedes Internal Calls and will interrupt Internal Call Ringing while it is in progress.
3. Recall Ringing, Central Office Ringing, and Internal Call Ringing all vary. Refer to DISTINCTIVE RINGING [D-10].

Interactions

The following ASYD assignments are for normal ringer patterns. For more information, refer to DISTINCTIVE RINGING [D-10].

Programming

STEP 1: **ASYD** - SYS3, Index 0 and Index 1. Assign Index 0 as 21 for one second ON, two seconds OFF (Trunk Calls). Assign Index 1 as 42 for two seconds ON, four seconds OFF (Station Calls).

STEP 2: **AKYD** - Assign the Station according to the following parameters:

TN: 1

STN: Station Number

TP: Type of Dterm (0-3)

KYN: Key Number (1-40)

KYI: Service Index

- 0: Key Not Used
- 1: Feature Key
- 2: Multi Line Key

KD: Not assigned for Feature Key

TN: Tenant Number

STN: Station Number

RG: Ringing Assignment per Multi Line Appearance

- 0: No Ringing
- 1: Ringing in Day mode only
- 2: Ringing in Night mode only
- 3: Not Used
- 4: Ringing in Day and Night Mode - 30 second delay.
- 5: Ringing in Day and Night Mode - 20 second delay.
- 6: Ringing in Day and Night Mode - 10 second delay.
- 7: Ringing in Day and Night Mode.

F-6 Faulty Trunk Report

General Description

This feature allows a Station to report a noisy or faulty Trunk Number by dialing a Special Access Code before hanging up. The FAULTY TRUNK REPORT consists of a Trunk Number, Telephone Number, Associated Time Division Switch and Reported Time. This information is displayed at the maintenance console (PCPro) and/or system printer.

Operating Procedure

To enter

1. Switch hook flash before disconnecting. Receive Special Dial Tone.
2. Dial the FAULTY TRUNK REPORT Access Code. Receive Service Set Tone.
3. The PCPro records the FAULTY TRUNK REPORT and returns the line to the original connection.

Service Conditions

1. This feature may be accessed during any Station-to-Station or Station-to-Trunk Connection.
2. The following information will appear on a Station-to-Station FAULTY TRUNK REPORT:
   a.) Station at which the operation was performed
   b.) Called Telephone Number
   c.) Time division switch
   d.) Time and date.
3. The following information will appear on a Station-to-Trunk FAULTY TRUNK REPORT:
   a.) Station at which the operation was performed
   b.) Route Number and Trunk Number
   c.) Time division switch
   d.) Time and date.
4. Stations are assigned this feature according to CLASS OF SERVICE - INDIVIDUAL [C-15] programming in System Data.
5. When this feature is activated, a Supervisory Alarm (SUP) will appear on the system.

Interactions

1. When this feature is activated, System Message [6-H] Faulty Trunk is generated. This message is sent to the PCPro or Fault Printer.
2. FAULTY TRUNK REPORT Data is displayed/printed in the Hexadecimal Numbering System.
3. The FAULTY TRUNK REPORT is displayed/printed periodically or can be obtained directly with the DFTD, Display of System Message command.

Programming

STEP 1: **ANPD** - Reserve a Number Level for Feature Access. Select Normal Service (N), Hooking Service (H) or Busy Service (B) from the combo box for Connection Indexes. Assign NND in accordance with a predetermined Numbering Plan.

STEP 2: **ASPA** - Assign an Access Code to FAULTY TRUNK REPORT, SRV = “SSC: Service Code”, SID = 46 and a Connection Status of hooking, CI = “H: Hooking”.

STEP 3: **ASFC** - Assign to the Stations allowed FAULTY TRUNK REPORT a Service Feature Restriction Class that allows SFI 46 (Bad Call Notify - Faulty Trunk Report).

F-7 Forced Account Code

General Description

This feature is a variation of the AUTHORIZATION CODE [A-20] feature, which makes it mandatory to enter an ACCOUNT CODE (up to ten digits) for all Outgoing Calls. The Account Code must be dialed before dialing the Outgoing Number. Calls are processed only when the dialed Account Code is valid.

The FORCED ACCOUNT CODE can be masked on the Desktop terminal display if necessary using the AUTHORIZATION CODE DISPLAY ELIMINATION [A-99].

Operating Procedure

When dialing an Outgoing Call

1. Lift the handset. Receive Dial Tone.
2. Enter the Access Code. Receive Service Set Tone.
3. Enter the FORCED ACCOUNT CODE (up to ten digits). Receive Dial Tone.
4. Dial the desired number.

Note: The Access Code for FORCED ACCOUNT CODE will be a maximum of three digits.

Service Conditions

1. The maximum number of digits per Forced Account Code is ten.
2. If both FORCED ACCOUNT CODE and AUTHORIZATION CODE [A-20] are provided in the system, the maximum number of codes are 20,000. **Note 1**

   Note 1: Capacity depends on system configuration.

   These specifications are based on the digits being assigned as follows:

1st digit:	A (when A equals a unique digit from 0 - 9)
2nd digit:	B - C (when b, c equal any two unique digits from 0 - 9)
3rd digit:	D-E (when d, e equal any two unique digits from 0 - 9)
4th digit:	F-G-H (when f, g, h equal any three unique digits from 0 - 9)
5th - 8th digits:	Any combination 0 - 9
9th and 10th digits:	Assigned using the checksum. These are for validation purposes only.

   Note: See ETI: 082 for details on AUTHORIZATION CODE/FORCED ACCOUNT CODE [F-7] assignment.

3. AUTHORIZATION CODE [A-20] and FORCED ACCOUNT CODE cannot be provided simultaneously for the same TENANT SERVICE [T-12] Group. System Data Programming determines whether AUTHORIZATION CODE [A-20] or FORCED ACCOUNT CODE is provided for each Tenant.
4. Stations are assigned this feature according to CLASS OF SERVICE - INDIVIDUAL [C-15] programming in System Data.
5. An assigned FORCED ACCOUNT CODE can be used for REMOTE ACCESS TO SYSTEM [R-2].

6. FORCED ACCOUNT CODE can be recorded on STATION MESSAGE DETAIL RECORDING (SMDR) [S-10] for INWATS and Central Office Incoming Calls, when used in conjunction with REMOTE ACCESS TO SYSTEM [R-2]. A maximum of ten digits can be registered as an account code for SMDR purposes.
7. When used in conjunction with REMOTE ACCESS TO SYSTEM [R-2], FORCED ACCOUNT CODE cannot be registered on SMDR if the Incoming Calls are terminated to Stations via Night Service or DID [D-8].
8. FORCED ACCOUNT CODE can be output in the STATION MESSAGE DETAIL RECORDING (SMDR) [S-10] record for Outgoing Calls from a Station handled with OUTGOING TRUNK QUEUING [O-2].

Note: There are two methods of FORCED ACCOUNT CODE assignment. These methods are Full Implementation and the Check Sum method. In the Full Implementation method, the entire FORCED ACCOUNT CODE is assigned in AATC and SYS1, Index 12, Index 13 and Index 14 are assigned 00H. In the Check Sum method, partial assignment for each FORCED ACCOUNT CODE is assigned in AATC. The last one or two digits (programmable option) are determined according to the data assignments in ASYD, SYS1, Index 12, Index 13 and Index 14.

Interactions

1. If FORCED ACCOUNT CODES are used on a per Tenant basis, AUTHORIZATION CODES [A-20] and ACCOUNT CODES [A-18] are not.
2. Station and Remote Access users share the same FORCED ACCOUNT CODES, assigned in AATC.
3. There are no restrictions regarding the first digit of FORCED ACCOUNT CODES except what is programmed in AMND. Tenant 0 does not conflict with the System Numbering Plan when separate Tenant Tables are provided. If Common Tenant Tables are used, a conflict between LCRS and OGC(A) used with SMDR will occur.

Note: The Access Code to the FORCED ACCOUNT CODES feature must be assigned within the System Numbering Plan.

4. When a Desktop terminal user initiates the LAST NUMBER CALL -D [L-6D] feature, the FORCED ACCOUNT CODE will be recorded if previously dialed.
5. AMND determines the length of FORCED ACCOUNT CODES based on the first digit of the code. For example, FORCED ACCOUNT CODES beginning with the digit “4” can be assigned as six digits long, and codes beginning with “5” can be assigned as three-digits long.
6. PEG COUNTS [P-7] of invalid FORCED ACCOUNT CODES are not available.
7. FORCED ACCOUNT CODES can be dialed by a user when establishing a CONSULTATION HOLD [C-17] Connection.
8. Access to the FORCED ACCOUNT CODE feature is determined by the Station’s CLASS OF SERVICE - INDIVIDUAL [C-15].
9. FORCED ACCOUNT CODES cannot be programmed for use via SPEED CALLING - SYSTEM/GROUP/INDIVIDUAL [S-7, 8, 9] features. As a feature to make Outgoing Calls, SPEED CALLING [S-7, 8, 9] does not first provide translation for feature access. However, FORCED ACCOUNT CODES can be programmed as part of the Desktop terminal SPEED CALLING - ONE TOUCH [S-26D] feature.

10. SIP Handler Controlled SIP terminals can place a call with preset dialing of Access Code, Forced Account Code and destination station number. To allow SIP Handler Controlled SIP terminals to receive a Service Set Tone (SST) or a Dial Tone (DT), enable the IP-DTG Tone Send Control Bit through the AUACL/ AUACN command.

Programming

STEP 1: **ASYD** - SYS1, Index 12. If Index 12=0, no Check Sum method is used and Index 13 and Index 14 are ignored regardless of their assignments.

Each Bit in Index 12 directly correlates to a digit in the FORCED ACCOUNT CODE assigned in AATC, and will determine the first digit of the Check Sum method. For this example, 26428 has been assigned as the FORCED ACCOUNT CODE in AATC. Arbitrarily pick any Bits, Bit 0 to Bit 4, and flag those Bits as data 1.

Note: Bit 0 through Bit 7 may be used when eight-digit FORCED ACCOUNT CODEs are assigned in the AATC command.

	Bit 4	Bit 3	Bit 2	Bit 1	Bit 0	Remarks
AATC	8	2	4	6	2	
SYS1, Index 12	0	1	0	0	1	
Designation	d4	d3	d2	d1	d0	

SYS1, Index 13. Each Bit in Index 13 directly correlates to a digit in the FORCED ACCOUNT CODE assigned in AATC, and will determine the second digit of the Check Sum method. Again pick any Bits, Bit 0 to Bit 4, but not the same combination of Bits as picked in Index 12.

	Bit 4	Bit 3	Bit 2	Bit 1	Bit 0	Remarks
AATC	8	2	4	6	2	
SYS1, Index 13	1	0	0	1	0	
Designation	d4	d3	d2	d1	d0	

SYS1, Index 14. Determines the algorithm used to calculate the last two digits of every FORCED ACCOUNT CODE. Only decimal numbers may be used in this index; hexadecimal values A to F may not be assigned.

SYS1, Index 14=XY. X and Y represent any arbitrarily chosen decimal digit 1-9; Y is used determine the first digit in the Check Sum method and X is used to determine the second digit in the Check Sum method. The following data, 37H; has been used for the example calculation:

	X				Y				
SYS1, Index 14	**Bit 7**	**6**	**5**	**4**	**3**	**2**	**1**	**0**	**Remarks**
DATA	0	0	1	1	0	1	1	1	

Note: If only one digit is to be added via the Check Sum method, Index 13 data will be 00H and Index 14, digit X (Bit 4 to Bit 7) = 0.

EXAMPLE CALCULATION

In AATC, 26428 has been assigned.

In SYS1, Index 12, Bit 0 and Bit 3 were chosen; 26428

2 + 2 = 4.

The decimal value of Bit 0 to Bit 3 of Index 14=7.

4 + N = 7. N = 3.

7 is obtained as any whole number greater than 4 that ends with a 7, the decimal value as determined in Index 14, Bit 0 to Bit 3.

In AATC, 26428 has been assigned.

In Index 13, Bit 4 and Bit 1 were chosen; 26428

8 + 6 = 14.

The decimal value of Bit 4 to Bit 7 of Index 14 is 3.

14 + N = 23. N = 9.

23 is obtained as any whole number greater than 14 that ends with a 3, the decimal value determined in Index 14, Bit 4 to Bit 7.

The FORCED ACCOUNT CODE will be 26428-39.

Note: Digits 39 determined by the Check Sum method will be unique to the FORCED ACCOUNT CODE 26428. In other words, 26428 will not be followed by any digits other than 39.

SYS1, Index 43, Bit 2. Is Service Set Tone sent after FORCED ACCOUNT CODE, ASPA Feature Access Code? 0/1: No/Yes.

SYS1, Index 93, Bit 6. Are the AMND Tables Separate or common for Tenants in the System? 0/1: Separate/Common. Assign data "0" for separate Tables.

SYS2, Index 3, Bit 5. Type of Code: 0/1 AUTHORIZATION CODE/FORCED ACCOUNT CODE. Assign data "1".

STEP 2: **ANPD** - Reserve a Number Level for Feature Access. Select Normal Service (N), Hooking Service (H) or Busy Service (B) from the combo box for Connection Indexes. (applicable to any feature assigned to this level). Assign NND in accordance with a predetermined Numbering Plan.

STEP 3: **ASPA** - For SRV = "SSC: Service Code" SID = 42, assign the Access Code. Assign for Connection Indexes of "N: Normal" and "H: Hooking".

STEP 4: **ASFC** - For Stations operating via FORCED ACCOUNT CODES, assign these Stations a Service Feature Restriction Class (SFC) that allows SFI 27 (Acct. Codes/Auth. Codes/Forced Acct. Codes). This SFC will be assigned to the Station in ASDT. Presently, the SFC parameter in AATC is inoperative.

NND = Access Code + Forced Account Code. For example, Access Code = 88 (two digits), Forced Account Code = 12345678 (eight digits). The NND is the sum of digits of the Access Code plus the Forced Account Code, therefore in this example the NND would be ten.

STEP 5: **ARSC** - For this feature to operate correctly, assign a Route Restriction Class (RSC) that Toll Restricts (see TOLL RESTRICTION - 3/6 DIGIT [T-10]) or denies access to Routes on a Direct Dial Access, RRI 3. This restricted RSC should then be assigned to the stations that use FORCED ACCOUNT CODE in ASDT. Additionally, construct an RSC that will be appropriate for the desired operation of the Station after the user has dialed the FORCED ACCOUNT CODE. This RSC can then be used when assigning an ACR of "1" AATC.

STEP 6: **AMND** - For Tenant 0, assign the first digit of all FORCED ACCOUNT CODES as the Destination Code (DC) and the Maximum Necessary Digits (MND) of the FORCED ACCOUNT CODE.

STEP 7: **AATC** - Assign the TN, Digit Code (DC) (the actual Forced Account Code itself), the FORCED ACCOUNT CODE Restriction (ACR) as data "1". Then assign the Route Restriction Class (RSC) that is allowing the outgoing call. ACR = 2, does not require RSC restriction.

STEP 8: **LATC** - Listing of all the FORCED ACCOUNT CODEs. This is the simplest way to determine what digits have been added to the system with the Check Sum method. This is not required if the Check Sum method is not used.

F-17 Flexible Assignment of Function Buttons

General Description

This feature allows Flex keys (REDIAL, FEATURE, SPEAKER, CONF, ANSWER, RECALL, HOLD, and TRANSFER) to be freely assigned on each individual Dterm. Flex keys are programmed through the maintenance console (PCPro) and are part of System Data.

Service Conditions

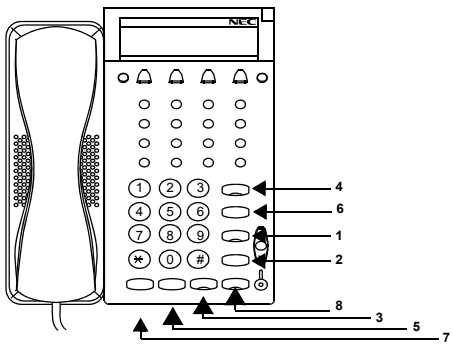
1. A Flex key cannot be used as a Line key.
2. SPEAKER, ANSWER, FEATURE, CONF, and REDIAL should be assigned to Flex keys that have lamps.
3. SPEAKER must always be assigned.
4. RECALL and TRANSFER must be assigned because they are used to indicate a Pause (-), Hooking (H) (!), and Voice Call (V) when programming a SPEED CALLING - ONE TOUCH -D [S-26D] Destination Number.

Programming

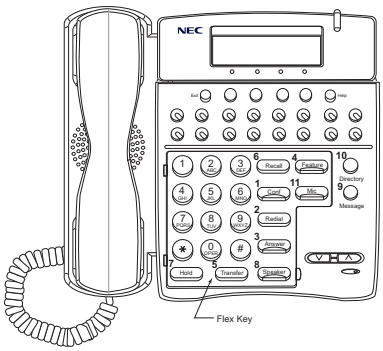
Flex key Assignment Example

AKYD -

TP:	2 or 3	KYN:	1~ 8	KYI2:	0 or 1		
CONF:	KYN 1	REDIAL:	KYN 2	ANS:	KYN 3	FEATURE:	KYN 4
TRANSFER:	KYN 5	RECALL:	KYN 6	HOLD:	KYN 7	SPEAKER:	KYN 8
Message:	KYN 9	Directory:	KYN 10				



KYN	FKY	STANDARD
3	28	Answer
2	8	Redial
1	13	Conference
4	16	Feature
5	18	Transfer
7	20	Hold
8	52	Speaker
6	57	Recall



KYN	FKY	STANDARD
3	28	Answer
2	8	Redial
1	13	Conference
4	16	Feature
5	18	Transfer
7	20	Hold
8	52	Speaker
6	57	Recall
9	147	Message
10	15	Directory

Note: For the examples of other terminals, refer to AKYD command in Command Manual.

F-24 Forced Release - Attendant

General Description

This service feature provides an attendant with the capability to override and release a busy trunk connection within the Telephony Server, via ATTENDANT CONSOLE [A-3]. Upon activation, a Warning Tone will be heard by the connected parties, after which they will be connected with the Attendant in a three-way bridge.

Note: This feature is not available in North America and Australia.

Operating Procedure

To activate FORCED RELEASE - ATTENDANT:

1. Depress TKSL key.
2. Dial the desired trunk route (2 or 3 digits) and identification number (3 digits).
3. Upon receiving Busy Tone, depress BV key.
4. A two-burst tone is sent to the connected parties.
5. The Attendant can now monitor or join the connection (ATTENDANT OVERRIDE [A-7]).
6. Depress CANCEL key to release the connection; the connected parties hear Reorder Tone and the Attendant becomes idle.

Service Conditions

1. Availability of this service feature is assigned by the system data.
2. ATTENDANT OVERRIDE [A-7] feature is a prerequisite for this service feature.
3. This service feature is effective when an Attendant enters a busy connection using ATTENDANT OVERRIDE [A-7].
4. For detailed service conditions, refer to ATTENDANT OVERRIDE [A-7].

F-31 Follow Phone

General Description

This feature allows individual station data of a terminal to be exchanged with another, without the assistance of the PCPro. The result of the exchange is printed out in the system message. The exchange of terminal data by this service is called SWAP service. The following information is included in the system message:

a.) Successful activation: Station data change notice.
b.) Unsuccessful activation: Station data change notice and error code to indicate the restricted condition.

Operating Procedure

To swap the station data of Station A (202) for Station B (209), operating from Station A

1. Go off-hook. Receive Dial Tone.
2. Dial the Swap service Access Code. Receive Special Dial Tone. The LCD displays.

SWAP **(Time Display)**

3. Dial the Station B number (209). Receive Service Set Tone. The LCD displays:

SWAP OK **209** **(Time Display)**

4. Replace the handset; the service will be activated.When the service is restricted, the user hears Reorder Tone after dialing the Target Telephone Number. The LCD displays:

Flashing

REST **209** **(Time Display)**

To confirm

5. Dial 202 from Station A; Station B rings, and Station A receives Ringback Tone, or vice versa.

Service Conditions

1. This service is available for the following Telephone Class:
   TEC= 1: Analog Telephone - DP (10 P)
   2: Analog Telephone - PB
   3: Analog Telephone - DP/PB
   4: Analog Telephone - DP (20 P)
   12: Digital Terminal (Dterm)
2. This service is available between the same types of Terminals. This service is not available for Dterm IP Telephones and Single Line Telephones connected to the MC.
3. The Target Station has to be idle in order to activate this service. In case the Station is busy or lock-out is set, the service will be restricted.
4. This service is available for the Terminals that belong to the same Tenant or Module Group.

5. This service is not available for the following Stations:
   a.) Night Station (ACSI)
   b.) HOTLINE [H-1] Station
   c.) HOUSE PHONE [H-2]
   d.) Data Line, or Dterm that has a Data Line assigned
   e.) Station setting/being set CALL BACK [C-1] or OUTGOING TRUNK QUEUING [O-2]
   f.) Stations where following services are set:

CALL FORWARDING - ALL CALLS	[C-5]
CALL FORWARDING - BUSY LINE	[C-2]
CALL FORWARDING - DON'T ANSWER	[C-3]
MESSAGE WAITING INDICATION -D	[M-85D]

   g.) DIT [D-7]
   h.) Dterm calling from a Sub Line
   i.) Station which has set or has been set for Busy Service.

   Note: The service is available after the message is canceled.

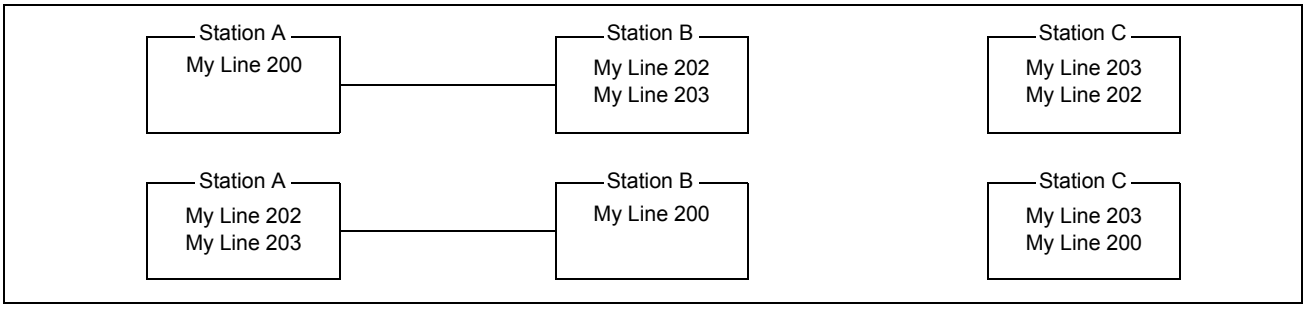
6. The following information is swapped when this service is activated:

ITEM	DATA SWAPPED IN FOLLOW PHONE SERVICE	STATION TYPE	
		SINGLE LINE	Dterm
1.	Feature Key information	Not Available	Available
2.	Add-On Module Key information **Note 1**	Not Available	Available
3.	Flexible Line/Feature Key information	Not Available	Available
4.	Speed Calling - One Touch Data **Note 2**	Not Available	Available
5.	Intercom information (Automatic, Manual, Dial)	Not Available	Available
6.	Telephone Number	Available	Available
7.	Route Restriction Class	Available	Available
8.	Service Class	Available	Available
9.	Speed Calling - Group/Station Data	Available	Available
10.	Phantom Number	Available	Available
11.	Name Display information	Available	Available

Note 1: The Add-On Module key information of Dterm does not swap.

Note 2: There are some hardware/firmware limitations in order to swap all the SPEED CALLING key information. SPEED CALLING key assigned on the Feature keys does not swap.

7. When this service is activated, the key information of Desktop terminal, which has the target station as a Multi Line will be changed automatically. For example:



8. If the Multi Line LED of Dterm is lit (busy) when the Station swaps, the line LED key may not light immediately after the service is activated. The LED lights normally after the user goes on-hook.
9. For the Stations that belong to the following Service Groups, this service is available only for those Stations belonging to the same group.

CALL PICKUP	[C-7, 30]
CALL PARK	[C-29]
STATION HUNTING - CIRCULAR	[S-7]
STATION HUNTING - TERMINAL	[S-9]
UCD	[U-1]

Note: In case of STATION HUNTING - SECRETARIAL [S-8], the function as the Secretarial Station does not swap even when the Station swaps. The Secretary Station is not considered as a member of the Hunt Group.

10. When the Following Station activates this service, its Destination Station is not swapped.
    a.) Stations where following services are set:
    - CALL FORWARDING - BUSY LINE set by the system
    - CALL FORWARDING - DON'T ANSWER set by the system
    - CALL FORWARDING - INTERCEPT / ANNOUNCEMENT [C-25]

    b.) Predetermined Station for HOTLINE [H-1]
    c.) Predetermined Station for OFF-HOOK ALARM [O-6]
    d.) DIRECT-IN TERMINATION (DIT) [D-7] Station
11. If Dterm has individual Terminal Data assigned, this service is available for Dterms with identical Terminal Data.
12. This service is not available for:
    a.) Desktop terminal with OAI service key
    b.) ACD Agent Console and Supervisory Console
    c.) Desktop terminal with CALL WAITING LAMP - UCD [C-76]
13. This feature cannot be activated between Dterms of different types. When different Dterms are used, there may be cases where the Incoming Call cannot be answered or some services cannot be activated, due to lack of Line/Feature keys. In this case, the data can be restored by activating the service again or by exchanging the Terminal sets.
14. When activating this service, the Terminals cannot be exchanged. (There may be some inconvenience if the number of SPEED CALLING keys, etc. differ between the Terminals.)

15. When activating this service, swap of the SPEED CALLING - ONE TOUCH -D key data start upon replacement of the handset. It takes approximately 42 seconds for the SPEED CALLING key data to swap, and during this time, FOLLOW PHONE Service and SPEED CALLING - ONE TOUCH operations are restricted.
16. After this service is activated, and if DM Load Reset is activated before the data is backed-up to the hard disk, the Stations return to the previous status. However, this is not the case with the SPEED CALLING - ONE TOUCH -D Feature keys.
17. The DSS Console data does not swap. The My Line of the connected Dterm may swap, but in this case, reassignment of DSS data and cabling will be necessary. The above is the same when using the DSS as an Add-On Module.
18. This feature is restricted at the office using Telephone Number.
19. When a Swap Restriction occurs, “System Message 16-X” will be generated.

Interactions

This service is restricted for a Dterm using Expanded Multiple Line Operation -D [E-26].

Programming

STEP 1: **ANPD** - Assign the minimum necessary number of digits for the first number of the SWAP Access Code.

STEP 2: **ASPA** - Assign the Access Code for SWAP. CI = “N: Normal”, SRV = “SSCA: Service Code Appendix”, SIDA = 96, SWAP.

F-35 Flexible Routing - FCCS

General Description

With this feature, an FCCS Trunk Call can automatically be routed via a Non-FCCS Trunk (Central Office or TIE Line), if all the FCCS Trunks are busy or a Data Link Failure occurs at the FCH Card.

Operating Procedure

No manual operation is required.

Service Conditions

Refer to the Data Programming Manual - FCCS.

Programming

Refer to the Data Programming Manual - FCCS.

F-36 FCCS Networking via IP

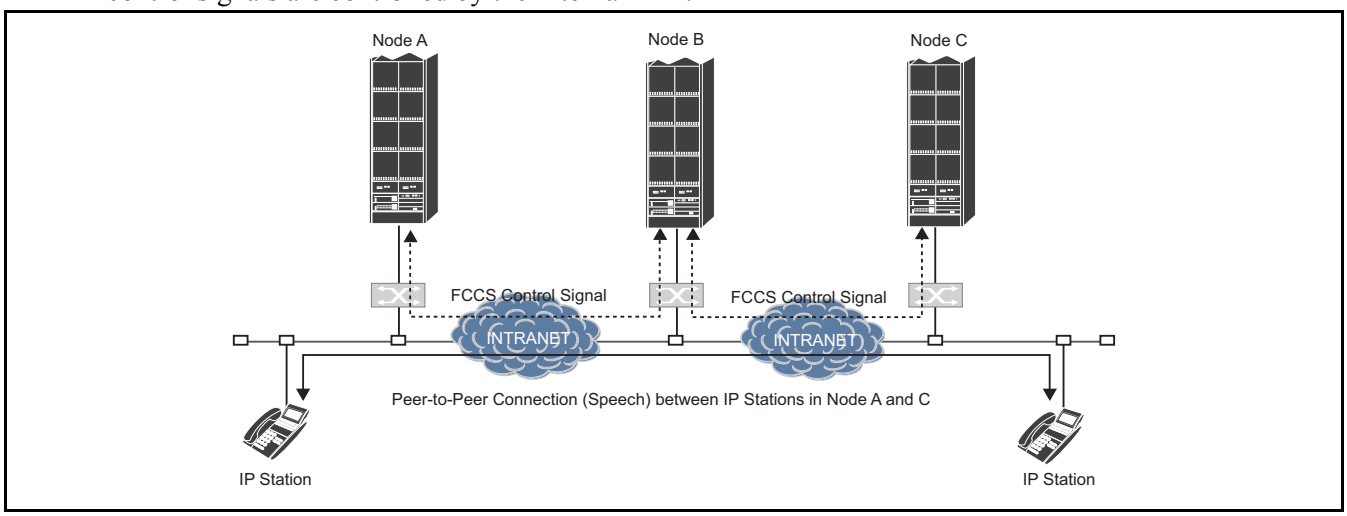
1. FCCS Networking via IP (Peer-to-Peer)

General Description

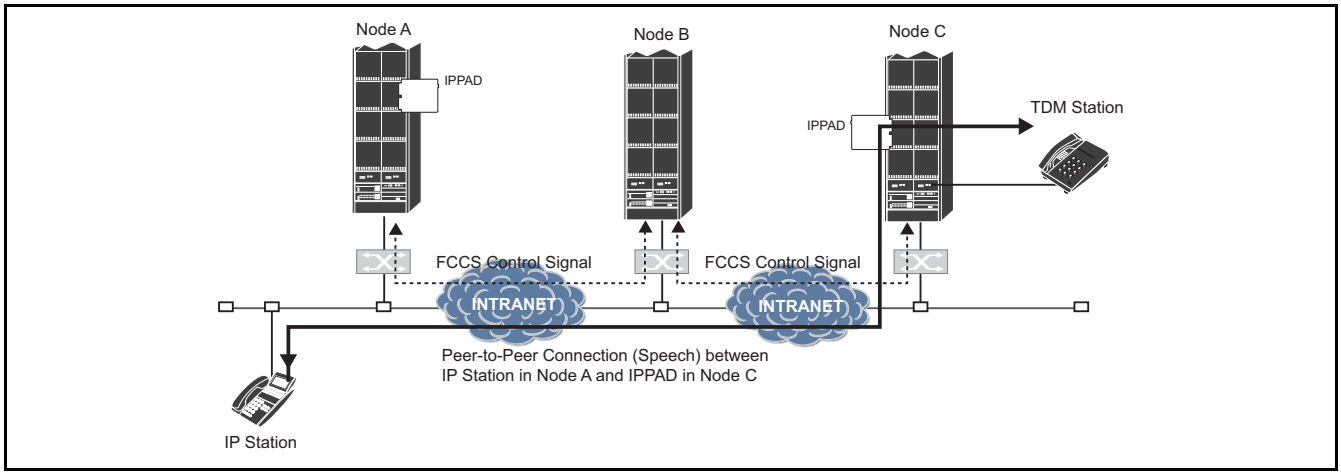
This feature allows the system to exchange both Speech and FCCS Signals over an Intranet. This feature can be activated by using Internal PHF. The Speech data between the IP Stations can be sent/received in a Peer-to-Peer connection, while the control signals are handled by the System. The following are typical connection patterns for this feature.

Note: The System has an interface function to send/receive the FCCS signals to/from the IP network. This interface function is called Internal PHF. Internal PHF converts the FCCS signals into the IP packets and vice versa.

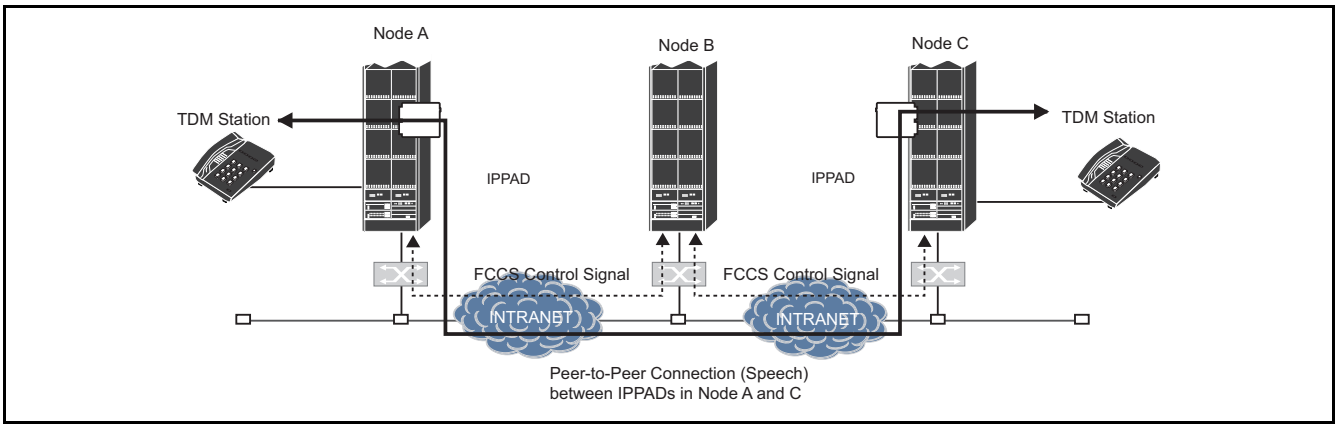
- Connection between the IP Stations: The Speech data is sent/received in a Peer-to-Peer connection. The control signals are controlled by the Internal PHF.



- Connection between the IP Station and Legacy Station: The Speech data is sent/received via IPPAD circuit card accommodated in the Legacy Station Node. At this time, Peer-to- Peer connection is established between the IPPAD and the other Node IP Station.



- Connection between the Legacy Stations: Each IP Station seizes the Self Node IPPAD, and Peer-to-Peer connection is established between the IPPADs.



Operating Procedure

No manual operation is required.

Service Conditions

1. This feature realizes the existing FCCS (hereafter called DTI-FCCS) services in a Peer-to-Peer network.
2. If the routing data for both Peer-to-Peer FCCS and DTI-FCCS is programmed for a single FPC, the Peer-to-Peer FCCS routing takes precedence. At this time, the DTI-FCCS routing is used as the backup, or used as an alternative if a link disconnection occurred in the Peer-to-Peer FCCS.
3. Tandem Path Through Control to Peer-to-Peer FCCS network is not available.
4. DP signal cannot be relayed via Peer-to-Peer FCCS. However, PB signals can be relayed on Peer-to-Peer FCCS when IPPAD with Sub Board is used.
5. A maximum of 63 nodes can be connected in a network.
6. The IP address of a destination Node is assigned by AFIPN. The port number of a destination Node (default is 60180) is assigned by ADTM. When the port number is changed by ADTM, Internal PHF must be initialized by either of the following methods.

PCPro	Command Menu	Descriptions
ADTM	Restart	All the IP-related port, including the Internal PHF communication port, are disconnected and then generated again. The destination Node information and Send Control information controlled by Internal PHF is cleared.
ADPM	Module Initialize	Internal PHF communication port is disconnected and then generated again. The destination Node information and Send Control information controlled by Internal PHF is cleared.
	Module Reset	All the IP-related port, including the Internal PHF communication port, are disconnected and then generated again. The destination Node information and Send Control information controlled by Internal PHF is maintained.

7. A maximum of four Nodes (three links) can be connected as tandem nodes.
8. This feature does not support Wide-Band services H0, H1 of ISDN/CCIS Line.

9. When using this feature, the tone such as RBT or Music is sent under the following conditions:

Tones	DTI-FCCS	Peer-to-Peer FCCS
Dial Tone (DT)	Self Node Tone	Self node
Special Dial Tone (SPDT)	Self Node Tone	Self node
Ringback Tone (RBT)	Connected Node Tone	IP terminal
Call Waiting Tone (CWT)	Self Node Tone	Self node
Busy Tone (BT)	Self Node Tone	Self node
Reorder Tone (ROT)	Self Node Tone	Self node
Service Set Tone (SST)	Self Node Tone	Self node
Second Dial Tone (SDT)	Connected Node Tone	IP terminal
Tone Ringer (TRG)	Self Node Tone	Self node
Sender Transmitting Tone (SDTT)	Connected Node Tone	IP terminal
Call Waiting RBT (CWRBT)	Connected Node Tone	IP terminal
Music on Hold (MUSIC)	Connected Node Tone	IP terminal
No Tone	Self Node Tone	Self node

Note: If a DTI-FCCS Node exists between the holding party and held party, the tone is sent from the DTI-FCCS Node. If no DTI-FCCS Node exists between the holding party and held party, the held IP terminal provides the tone itself.

10. If the Peer-to-Peer FCCS link is down due to the LAN failure, etc., the alternate routing to another outgoing trunk is available.
11. The restrictions for connection containing Peer-to-Peer FCCS and DTI-FCCS are as follows:
- SIP terminals are unavailable to use.
12. If Peer-to-Peer FCCS, DTI-FCCS and Peer-to-Peer CCIS coexist in the same network and each node is connected in tandem, the following must be considered:

Connection Patterns	Availability of Connection
Peer-to-Peer FCCS – Peer-to-Peer FCCS – Peer-to-Peer CCIS	Available
Peer-to-Peer FCCS – DTI-FCCS – Peer-to-Peer CCIS	Available
DTI-FCCS – Peer-to-Peer FCCS – Peer-to-Peer CCIS	Available **Note 1**
DTI-FCCS – DTI-FCCS – Peer-to-Peer CCIS	Available

Note 1: Peer-to-Peer CCIS ↔ Peer-to-Peer FCCS ↔ DTI-FCCS tandem connection is available. The restrictions for this connection are as follows:
- SP Controlled SIP terminals are unavailable to use.

13. The Number of PHF Packet Retransmissions appropriate for the network status can be selected from 1-5 (maximum) by assigning System Data [ASYDL, SYS1, Index 1078, Bit 0 to Bit 3 (Number of PHF Packet Retransmissions)]. **Note 2**, **Note 3**, **Note 4**

Note 2: Assign “1” for the number of PHF packet retransmissions. Increase the number only if a packet loss has occurred frequently because of the network.

Note 3: When the number of PHF packet retransmissions is increased, the entire system performance may go down depending on the network status.

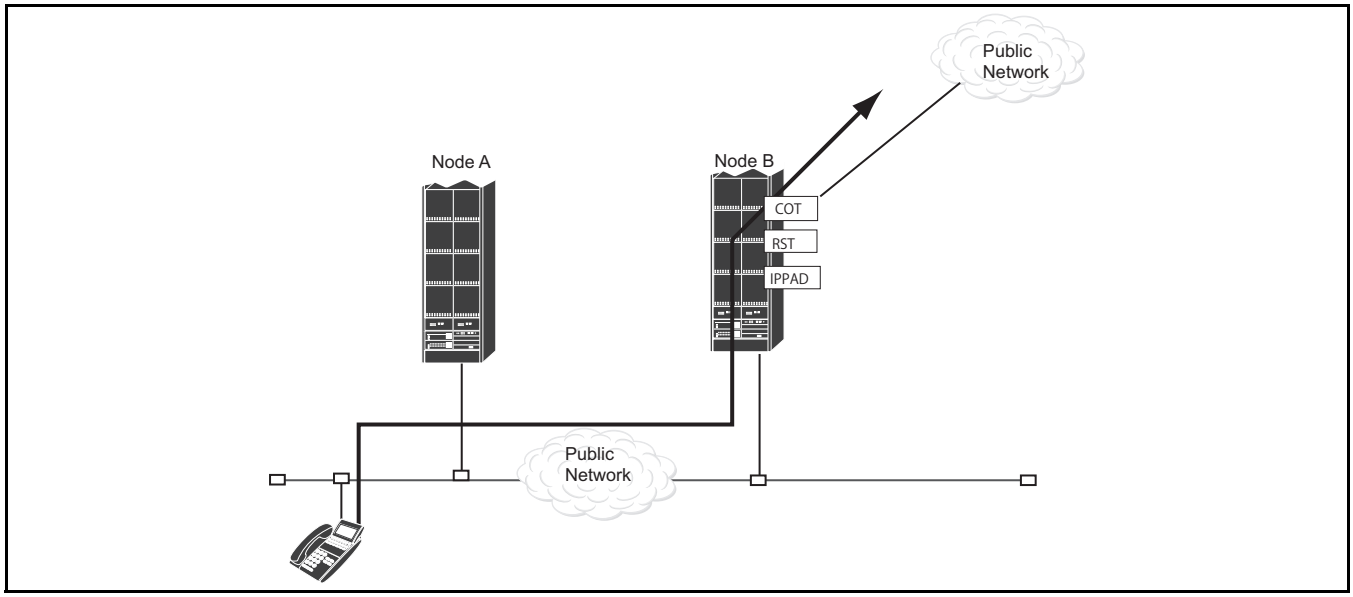
Note 4: When the number of PHF packet retransmissions is changed, Internal PHF Module Reset is required by using ADPM command.

14. When a PHF communication failure such as consecutive PHF packet retransmissions or packet loss occurred, the system message (IP PKG Warning [33-R]/IP PKG Failure [33-S]/IP PKG Restore [33-T]) can notify the maintenance personnel.

Note: PHF module must be restarted when the number of PHF packet retransmissions and PHF packet retransmission timer is changed by system data.

Note: When enabling the internal PHF packet retransmission control function, enable the internal PHF packet retransmission control function for all the nodes in the network and use the common value for the number of PHF packet retransmissions and PHF packet retransmission timer. However, in the case of [P2P-FCH-P2P] configuration, one of the Peer-to-Peer FCCS network only will be available for this function.

15. When originating a trunk call via Peer-to-Peer FCCS, if the same sender as that of the originating terminal in the node is not available, the same sender as that of the outgoing trunk in the node is selected.



a.) To use this function, accommodate an RST circuit card in the same node as the outgoing trunk.
b.) The relationship between the system data and the sender to select is as follows.

- In the case of the system data [ASYDN, SYS1, Index 641, Bit 6=0]

  The same sender as that of the originating terminal in the node is selected. If it cannot be selected, the same sender as that of the outgoing trunk is selected. If the both of them cannot be selected, outgoing call is restricted.

- In the case of the system data [ASYDN, SYS1, Index 641, Bit 6=1]

The same sender as that of the outgoing trunk in the node is selected. If it cannot be selected, the same sender as that of the originating terminal is selected. If the both of them cannot be selected, outgoing call is restricted.

c.) If the sender cannot be selected, “Busy” will be displayed on the LCD of Digital Terminal.
d.) The sender “RT905 (Sender Trunk)” is used in this function.
e.) The peg count for sender trunk seizure is processed in the node where the sender trunk is seized.
f.) When using the following features, this function is disabled.

- Individual Trunk Access - CCIS (MAT)
- Trunk Call Origination from Attendant Console (LCRS)
- Trunk Call Origination from ISDN Terminal
- OGC/OGCA Call Origination to ACSI line through Second Dial Tone.
- Hot Line Station through Trunk Line method (AHLS/AHLSN TYPE=3)
- Trunk Line Appearance

g.) This function is not available for Hotel system and OAI/ACD system.

16. To use this feature, the Speech Link Reconnection via FCCS Networking via IP function must be enabled [ASYDL, SYS1, Index 527, Bit 5=0].

Programming

For information about data programming for FCCS Networking via IP, refer to FCCS Networking via IP in Chapter 5 of Data Programming Manual - FCCS.

2. FCCS Networking via IP (by IPTRK)

Note: Initialize the appropriate IPTRK(s) after the office data assignment.

Note: FCCS Networking via IP (by IPTRK) is available only for a migrated system from IPX/SV7000/SV8500. For a newly installed the SV9500 system, use FCCS Networking via IP (Peer-to-Peer).

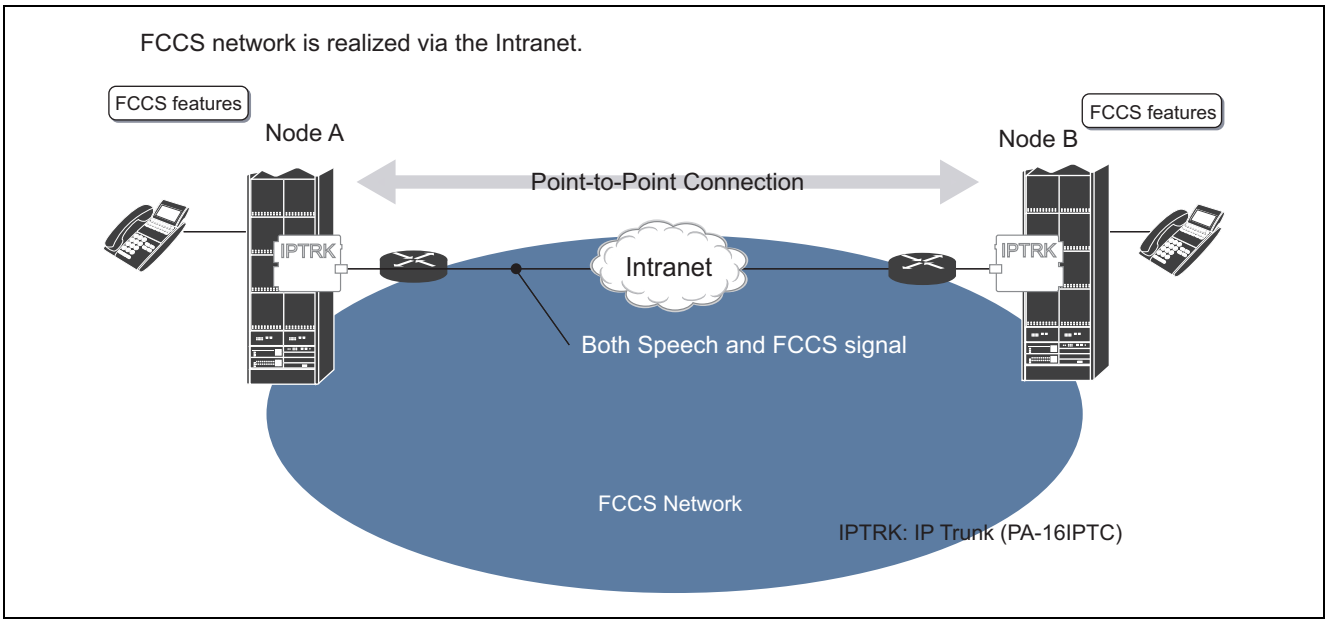
General Description

This feature allows the system to exchange both Speech and FCCS Signals over an Intranet. It is not necessary to use an FCH circuit card. To establish or release a call, the following Connection Pattern can be selected:

- Point-to-Point Connection: For both Speech and FCCS Signal Channels, an IPTRK circuit card is used. At this time, the IPTRK Cards in the Self Node and each of its connected node(s) must be in one-to-one ratio.
- Point-to-Multipoint Connection: For both Speech and FCCS Signal Channels, an IPTRK circuit card is used. At this time, the IPTRK Card, accommodated in the Self Node, can be connected to multiple destination (IPTRK Cards of multiple nodes).

Point-to-Point Connection is functionally a kind of form within Point-to-Multipoint Connection.

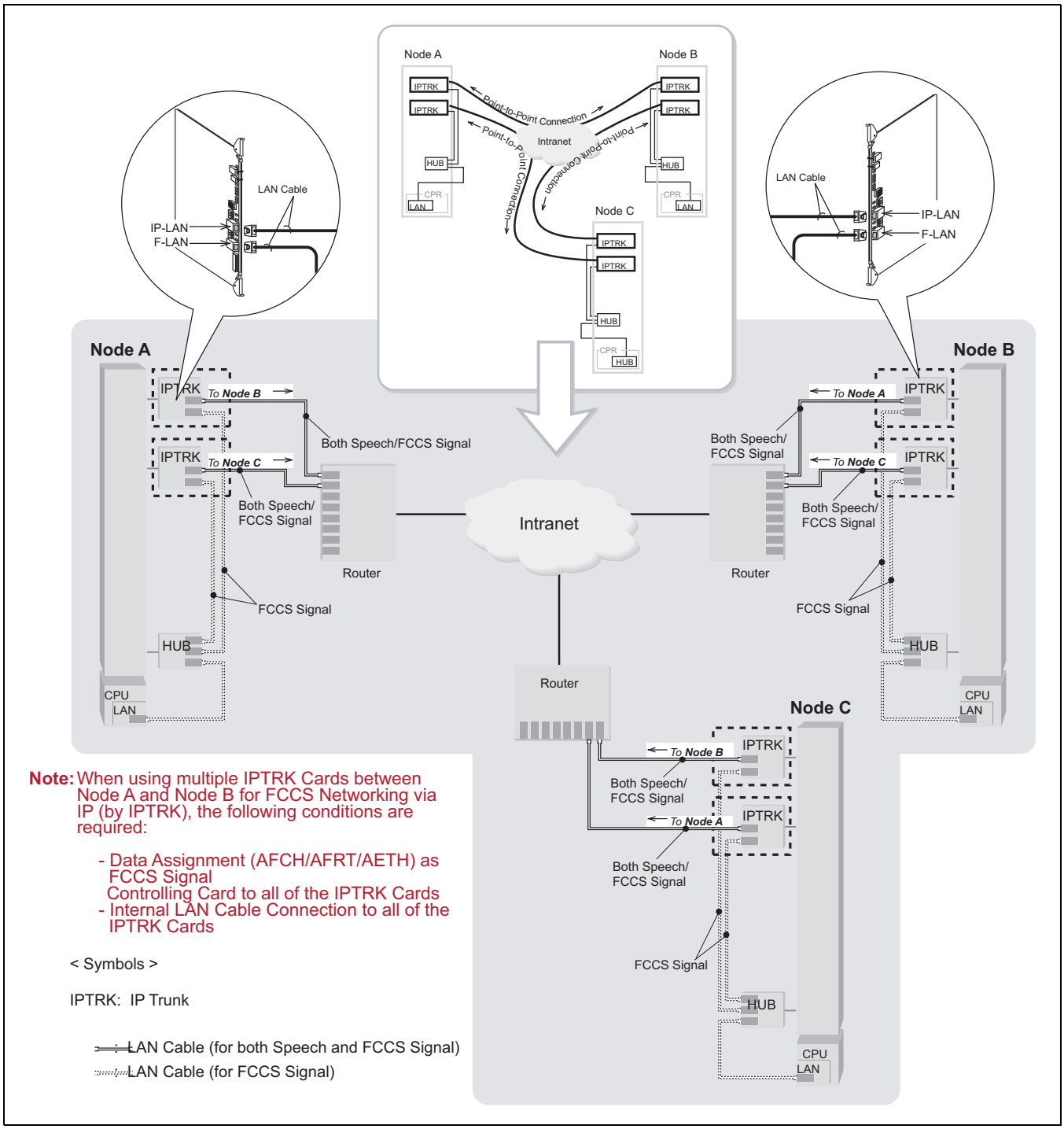
FCCS Networking via IP - Overview



- **Required Hardware (Point-to-Point Connection)**

  This feature is made available by using the following hardware equipment. As shown, both Speech and FCCS Signal Channels between the Nodes must be provided by Point-to-Point Connection of IPTRK Cards with LAN cables.

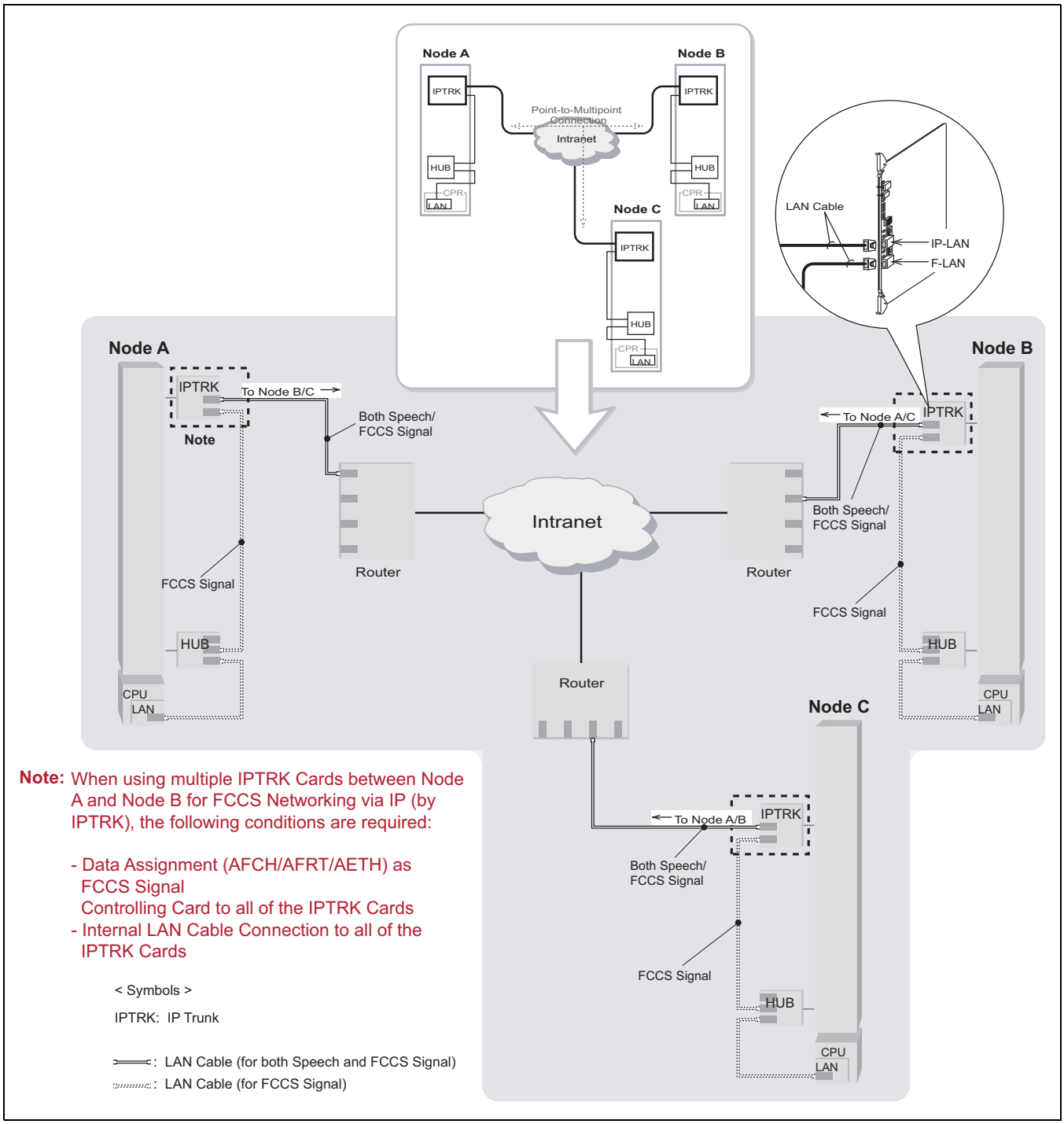
Required Hardware (Example) for FCCS Networking via IP [IPTRK: Point-to-Point]



- **Required Hardware (Point-to-Multipoint Connection)**

  This feature (allowing IPTRK Card for both Speech and FCCS Signal Channels) is made available by using the following hardware equipment. Once this feature is set, each IPTRK Card can make a Point-to-Multipoint Connection with IPTRKs of multiple nodes.

Required Hardware (Example) for FCCS Networking via IP [IPTRK: Point-to-Multipoint]

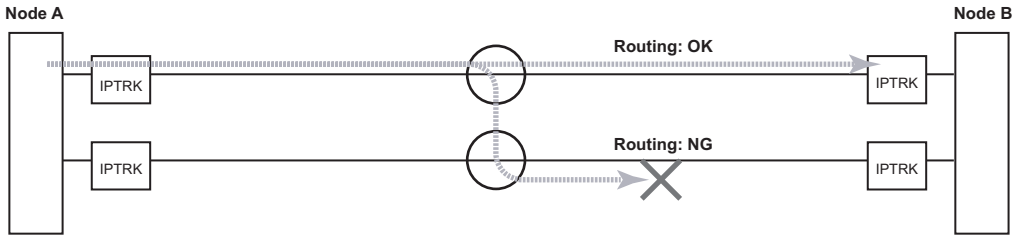


Operating Procedure

No manual operation is required.

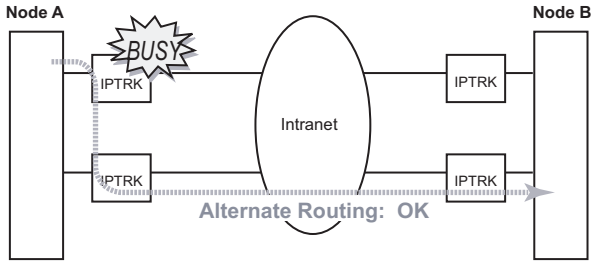
Service Conditions

1. FCCS Networking via IP (by IPTRK) is available only for a migrated system from IPX/SV7000/SV8500. For a newly installed the SV9500 system, use FCCS Networking via IP (Peer-to-Peer).
2. To send or receive voice information, each node will use IPTRK circuit cards. At this time, the IPTRK Cards in the Self Node and its connected nodes can be either in a one-to-one ratio (referred to as Point-to-Point Connection) or in a one-to-n ratio [(n=2 or more), referred to as Point-to-Multipoint Connection].
3. Use a Router for connections between the IP Network and an IPTRK Card.
4. The maximum number of IPTRK Cards allowed is two per UNIT. Accordingly, a total of 16 IPTRK Cards can be accommodated per IMG.
5. Only Voice and FAX (G3) information can be sent or received via an IPTRK Card.
6. A call cannot be established or routed across multiple IP Networks. Refer to the following figure:



7. When multiple IPTRK Cards are used for Node-to-Node Connections and all of the lines on the IPTRK are busy, Alternate Routing is available. However, if a Router is not used for some reason, the related IPTRK Routes are not available. Refer to the following figure.

As shown, when multiple IPTRKs exist, Alternate Routing is available:



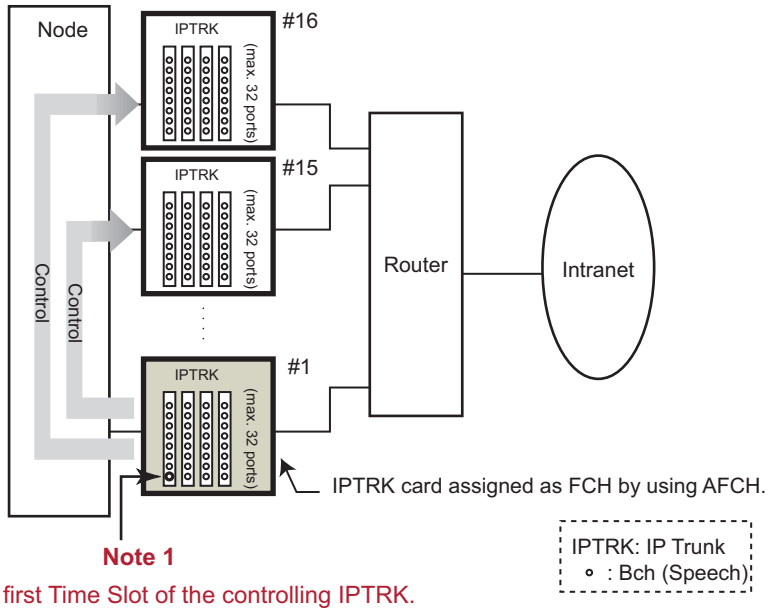
IPTRK: IP Trunk

8. When using multiple IPTRK Cards between Node A and Node B for FCCS Networking via IP (by IPTRK), the following conditions are required:

   - Data Assignment (AFCH/AFRT/AETH) as FCCS Signal Controlling Card to all of the IPTRK Cards
   - Internal LAN Cable Connection to all of the IPTRK Cards

   Refer to the following figure:

An IPTRK Card, assigned as an FCCS Signal Controlling Card by AFCH, can control the B-Channels of the card and a maximum of 15 other cards.



Note 1: Assign a B-Channel in the first Time Slot of the controlling IPTRK.

9. During an IPTRK-related Tandem Connection, speech quality can be diminished, or a dual connection failure may occur. DTMF and FAX Relay cannot be used.
10. This feature supports Quality of Service (QoS) function.
11. Tandem Path through function is not released.
12. A feature must be available for the FCCS Network to be available for this feature.
13. When a system message indicates “TCP Link Failure” due to data congestion on the Router or IP Network, a call cannot be made via an IPTRK circuit card. When a call is attempted despite this, the caller hears Busy Tone.
14. When a system message, related to “TCP Link Failure”, is created during Call Origination or Speech, that call’s Speech Channel is automatically released via Patrol Function.
15. When all lines of an IPTRK Card are busy, Alternate Routing to the Central Office Line or TIE Line is available. For more detailed information, refer to the table below.

Alternate Routing To Be Provided/Not Provided (FCCS Networking via IP)

×: Available NA: Not Available

	FROM FCCS Networking via IP ***(Point-to-Point/Point-to-Multipoint)***	**TO FCCS Networking via IP** ***(Point-to-Point/Point-to-Multipoint)***
Analog Central Office Line	NA	NA
Analog Tie Line	×	NA
CCIS	×	NA
ISDN	× (via Sub Address-Addressing only)	NA
Dp Channel	×	NA
FCCS Network over Intranet (via DTI)	×	×
FCCS Networking via IP (Point-to-Point/Point-to-Multipoint)	×	×
CCIS Networking via IP (Point-to-Point/Point-to-Multipoint)	×	NA

16. This feature does not support Broad Band Services H0, H1 of ISDN/CCIS Line. Supports 1B only.
17. Link Reconnection is available.
18. This feature does not support Hotel/Motel services.
19. When using IPTRK card, Dial Pulse (DP) relay broadcasting to FCCS Link (ASYDL, SYS1, Index 527, Bit 4=1) is not available.

Programming

Note: The data programming for both Point-to-Point Connection and Point-to-Multipoint Connection is identical.

Attention: *When any Office Data is assigned or modified, be sure to reset the IPTRK circuit card (by flipping the MB key OFF → ON → OFF).*

STEP 1: **ASYDL** - SYS1, Index 534, Bit 7=1 (NDM Data individual broadcasting from NCN to each LN)

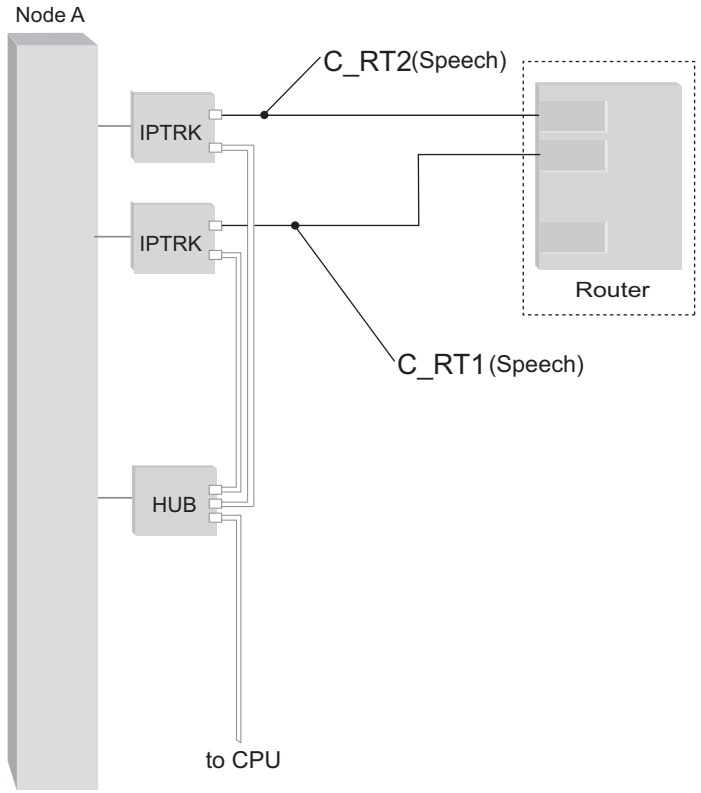
Note: Provided that IPTRK circuit card is equipped in the Node within FCCS Network, NDM data broadcasting is not to be executed. Therefore, when IPTRK Card is used in the FCCS Network, perform this data assignment at the Network Control Node (NCN).

SYS1, Index 527, Bit 5=0 (Speech Link Reconnection in FCCS via IP network is valid)

STEP 2: **ACRD** - Assign Route Class Data of Speech Routes. The IPTRK Card assigned with the AFCH command is to be used as the FCCS Signal Channel Controlling Card. It is not required to assign the Signal Route Data with the ACRD command. See the following sample data assignment.

Sample Data Assignment (ACRD)

ACRD: Speech Route (LSG=12, FGH=0, LKIND=3, IPTRK=1)

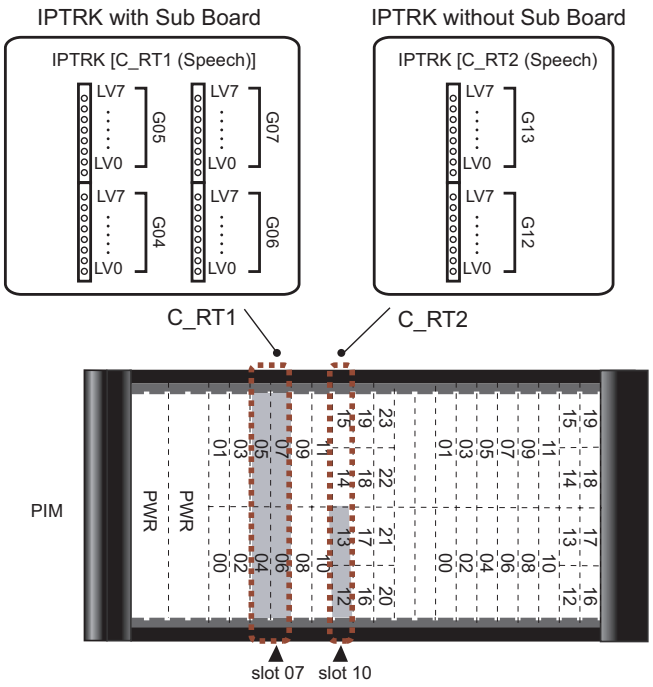


Type		Speech	Speech
CDN	C_RT	*1*	*2*
1	TF	3	3
2	TCL	4	4
3	RLP	2	2
4	SMDR	0	0
5	LSG	12	12
6	PAD		
7	TRKS	0	0
8	TC/EC		
9	FINT	0	0
10	FPEG	0	0
11	TC	0	0
12	MTC	0	0
13	STSEQ	0	0
14	FGH	0	0
15	MMN		
16	LKIND	3	3
17	IPLYR	0	0
18	IPTRK	1	1
19	IPPAD	0	0

Note: This data setting is an example.

STEP 3: **ACTK** - Assign the Connection Trunk Data of the IPTRK circuit card. It is not required to assign the Trunk Data for D-Channel. See the following sample data assignment.

Sample Data Assignment (ACTK)



Sample Data Assignment

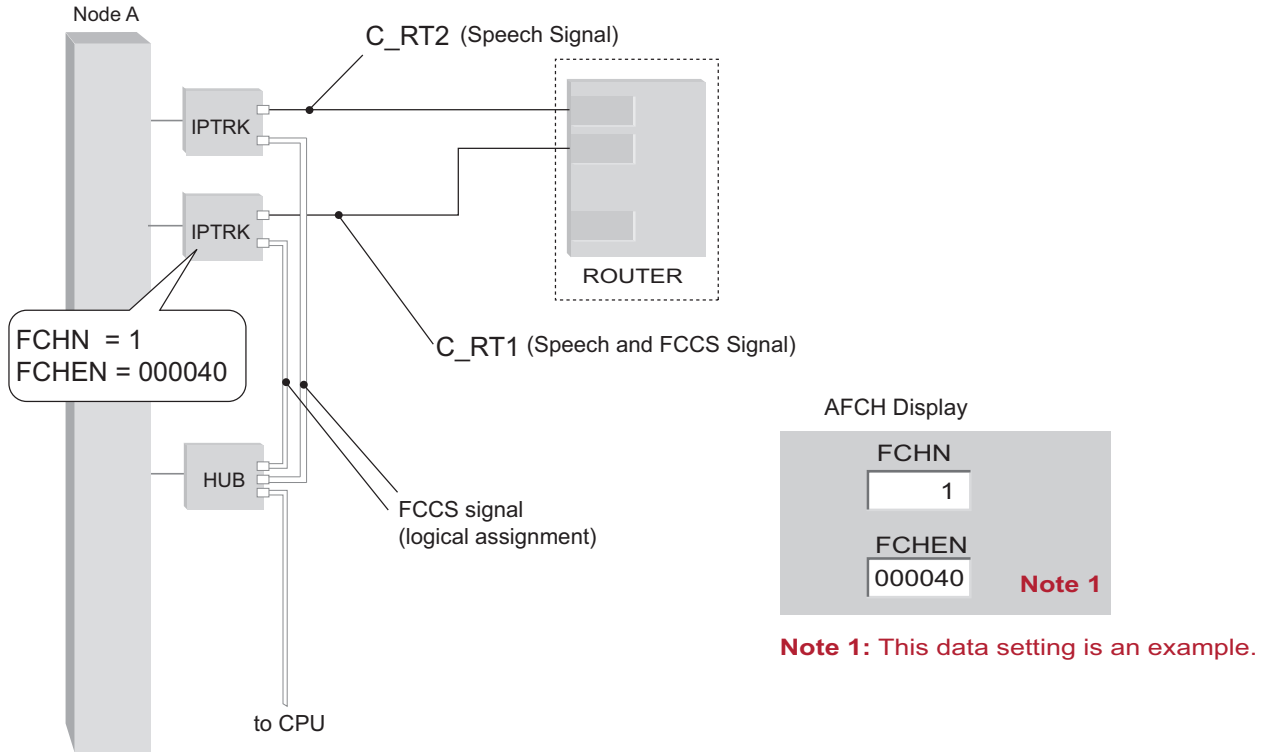
	C_RT (1-1023)	C_TK (1-4095)	C_LENS MG 00-07	U 0-3	G 00-23	LV 0-7
IPTRK(C_RT1)	1	1	0 0	0	0 4	0
		2	0 0	0	0 4	1
		3	0 0	0	0 4	2
		⋮ 32	⋮ 0 0	⋮ 0	⋮ 0 7	⋮ 7
IPTRK(C_RT2)	2	1	0 0	0	1 2	0
		2	0 0	0	1 2	1
		3	0 0	0	1 2	2
		⋮ 16	⋮ 0 0	⋮ 0	⋮ 1 3	⋮ 7

STEP 4: **MBCT** - Unbusy the Connection Trunk Data assigned in the previous step by using MBCT.

STEP 5: **AFCH** - Assign FCH Number (FCHN) to each IPTRK Card that is used as the FCCS Signal Controlling Card.

Assignment of FCHN (Example)

In this example, the FCHN = 1 is assigned to the IPTRK Card controlling the FCCS Signal.



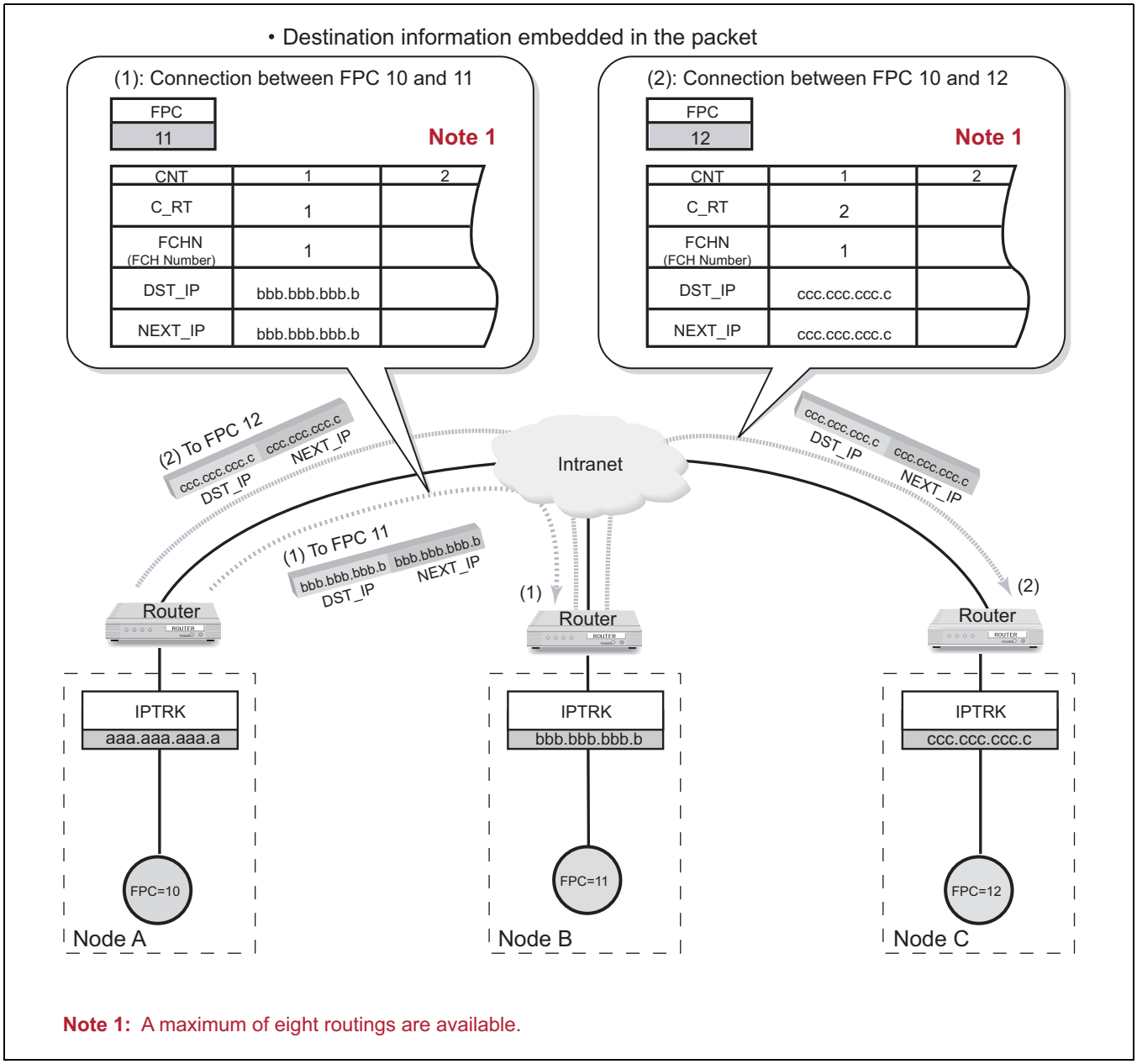
Note 1: This data setting is an example.

STEP 6: **AETH** - Assign the FCCS Routing Data as follows:

FPC (Fusion Point Code):	Assign FPC number (1 - 253) of the Destination Node.
FCHN (FCH Number):	Assign the FCH Number, specified in STEP 5.
C_RT (Connection Route):	Connection Route Number of Speech Channel (1-1023).
DST_IP (Destination IP Address):	Assign IPTRK IP address of the Destination Node.
NEXT_IP (Next IP Address):	Assign IPTRK IP address of the Destination Node.

A data assignment example is shown in the following figure. For additional information, refer to the FCCS Routing Sample Data Sheets at the end of this feature.

Sample Assignment of Destination IP and Next IP (AETH)



Note 1: A maximum of eight routings are available.

STEP 7: **ACAN** - Assign the Fusion Connection Index Code (FCIC) to the Self Node FPC:

FPC (Fusion Point Code): Assign FPC Number (1 - 253) of the Self Node.

FCIC (Fusion Connection Index Code): Assign unique Fusion CICs in the Self Node.

C_LENS (Connection_LENS): Assign IPTRK LENS Data (Speech Channel Data) specified in STEP 3.

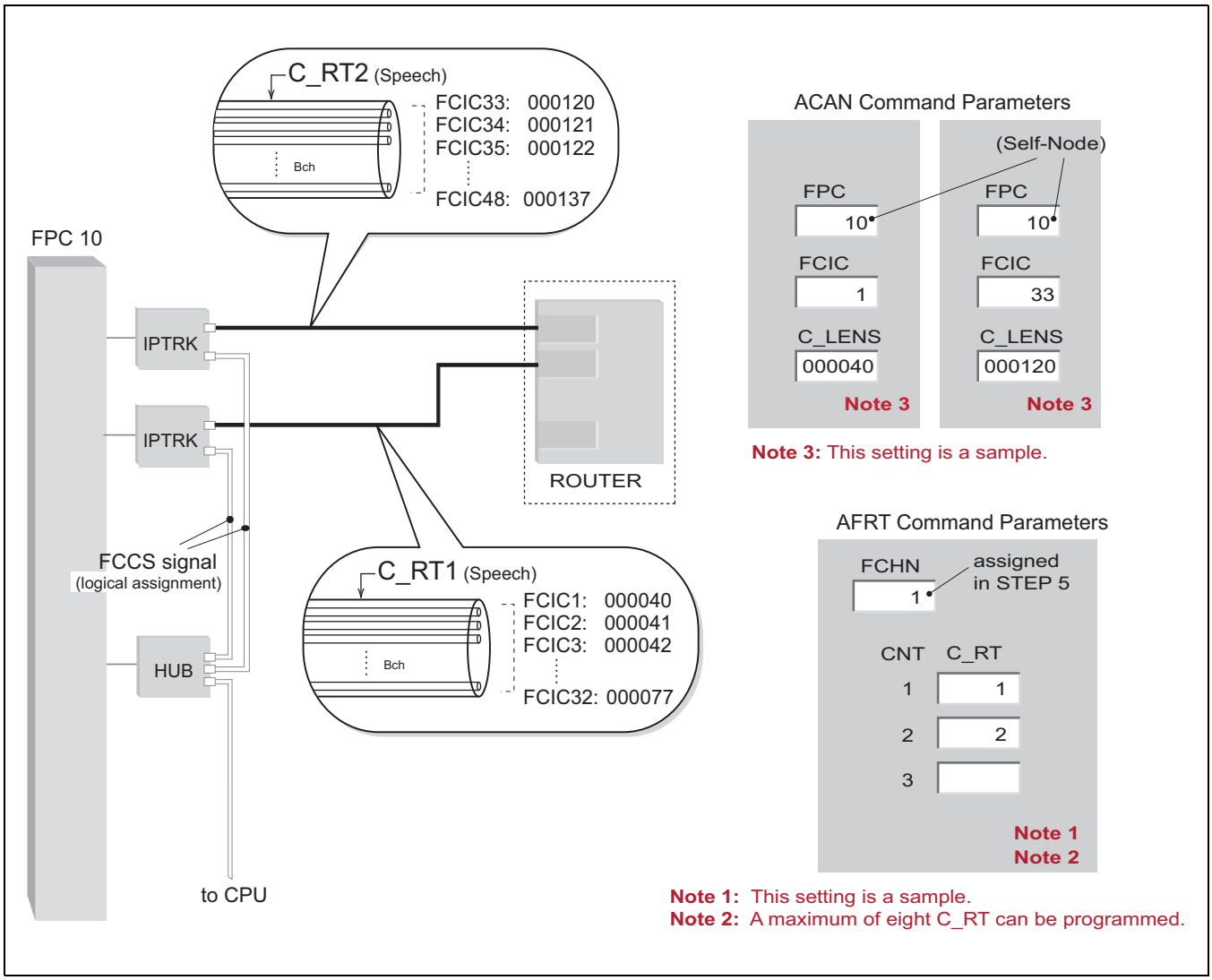
See the example data assignment shown in the figure following AFRT.

STEP 8: **AFRT** - Set Connection Route Numbers of the Speech Channels for the FCHN which was assigned in STEP 5.

FCHN (FCH Number): Assign FCHN, specified in STEP 5.

C_RT (Connection Route): Assign Connection Route of the Speech Channels, specified in STEP 2. See the following figure of example data assignment.

Sample Data Assignment (ACAN/AFRT)



STEP 9: **AGIP** - Assign the IP Addresses of IPTRK Card and its connected Router.

LENS (Line Equipment Numbers):	Assign the 1st LENS (MG + U + G: five digits) of IPTRK-accommodated HW block.
KIND (Kind of Selection):	Select “IPTRK (FCCS)”
IPTK_IP (IP Address of IPTRK):	Assign IP address of the IPTRK card. **Note 1**
DG_IP (IP Address of Default Gateway):	Assign IP address of the router.
NETMSK (Net Mask of IP Address):	Assign the Net Mask of IP address
ToS CONTROL:	Select the either, IP Precedence or DiffServ
QoS 1 (QoS Data for Signal Route):	When IP Precedence is selected, 0-7 When DiffServ is selected, 0x00-0x3F
QoS 2 (QoS Data for Speech Path):	When IP Precedence is selected, 0-7 When DiffServ is selected, 0x00-0x3F

NETID (Network ID):	Fixed to "0."
CSLINK_NUM (Maximum number of client/server connection):	Select the nearest number to the number of counter IPTRKs.
ARP (Frame Type of ARP):	Select "DIX" when the "Detail" box is checked.
RTP:	"RTP" checkbox is provided for detail assignment of the following parameters, if required. (Usually default data "0" is set for all the parameters.)
PKTLOSS:	Amount of Packet Loss [0%-100% (default is 0=8%)]
JIT_MAX:	Maximum Jitter Buffer [0-60 × 10 milliseconds (default is 0=120 milliseconds)]
JIT_MIN:	Minimum Jitter Buffer [0-60 × 10 milliseconds (default is 0= 40 milliseconds)]
MNGS:	Interval of Jitter Adjustment [0-255 times (default is 0=5 times)]
JIT_COUNT:	Interval of Jitter statistics [0-255 seconds (default is 0=1 seconds)]
BASE_ COUNT:	Interval of Time-base Correction [0-255 seconds (default is 0=10 seconds)]
JIT_FAST:	Judging rate of Fast Arrived Packet [0-100% (default is 0=100%)]
TIME_FAST:	Time-base Correction Judgment [0-100% (default is 0=80%)]

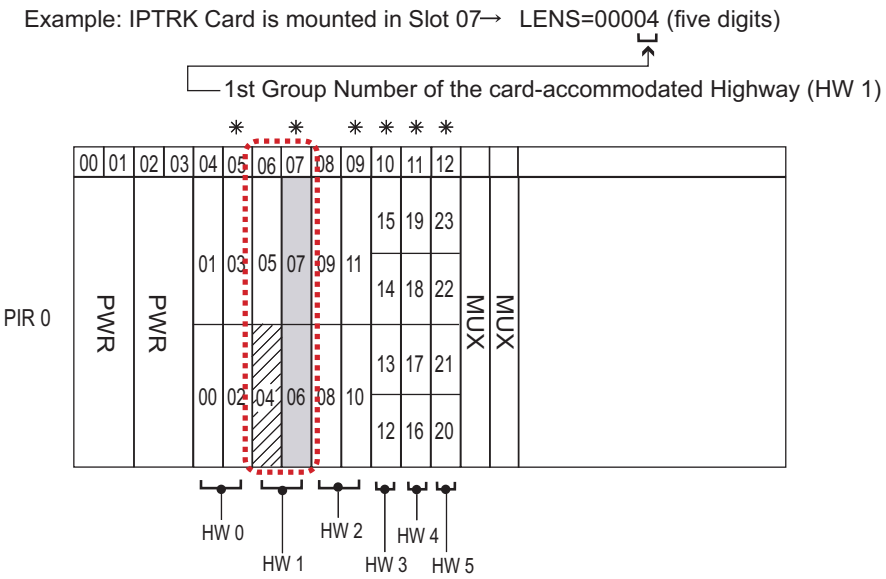
Note 1: Do not assign an IP address used in the internal LAN (172.16.254.xx/24).

STEP 10: **AVIPS** - Assign the detailed data of the IPTRK Card, such as Jitter Buffer, Payload Type/Size. When the data is not assigned, the default data is used.

Note: This command and AGIP cannot be activated simultaneously.

Note: IPTRK-related data specified by the LENS must be assigned by the AGIP command in advance.

LENS: Assign two-digit MG Number + one-digit Unit Number + 1st Group Number (two digits) of IPTRK-accommodated Highway.



Note: IPTRK circuit card can be mounted in slot marked with an asterisk (*).

KIND: Kind of Selection [IPTRK (FCCS)/IPTRK (CCIS)] **Note 2**

Note 2: The kind assigned by the AGIP command is displayed.

<Voice> [For KIND=IPTRK (FCCS)/IPTRK (CCIS)]

JIT_ACT: Select Jitter Buffer Mode
- 0: Operates according to the Switch Setting on the IPTRK Card
- 1: Static Jitter Buffer
- 2: Dynamic Jitter Buffer

PAY_TYP: Payload Type Selection Pattern
- 0: Follows to Switch Setting on the IPTRK
- 1: Standard Setting [G.729a -> G.723.1 (6.3 k) -> G.723.1 (5.3 k) -> G.711]
- 2: Voice Quality Precedence Mode [G.711 -> G.729a -> G.723.1 (6.3 k) -> G.723.1 (5.3 k)]
- 3: Band Precedence Mode 1 [G.723.1 (5.3 k) -> G.723.1 (6.3 k) -> G.729a -> G.711]
- 4: Band Precedence Mode 2 [G.723.1 (6.3 k) -> G.723.1 (5.3 k) -> G.729a -> G.711]

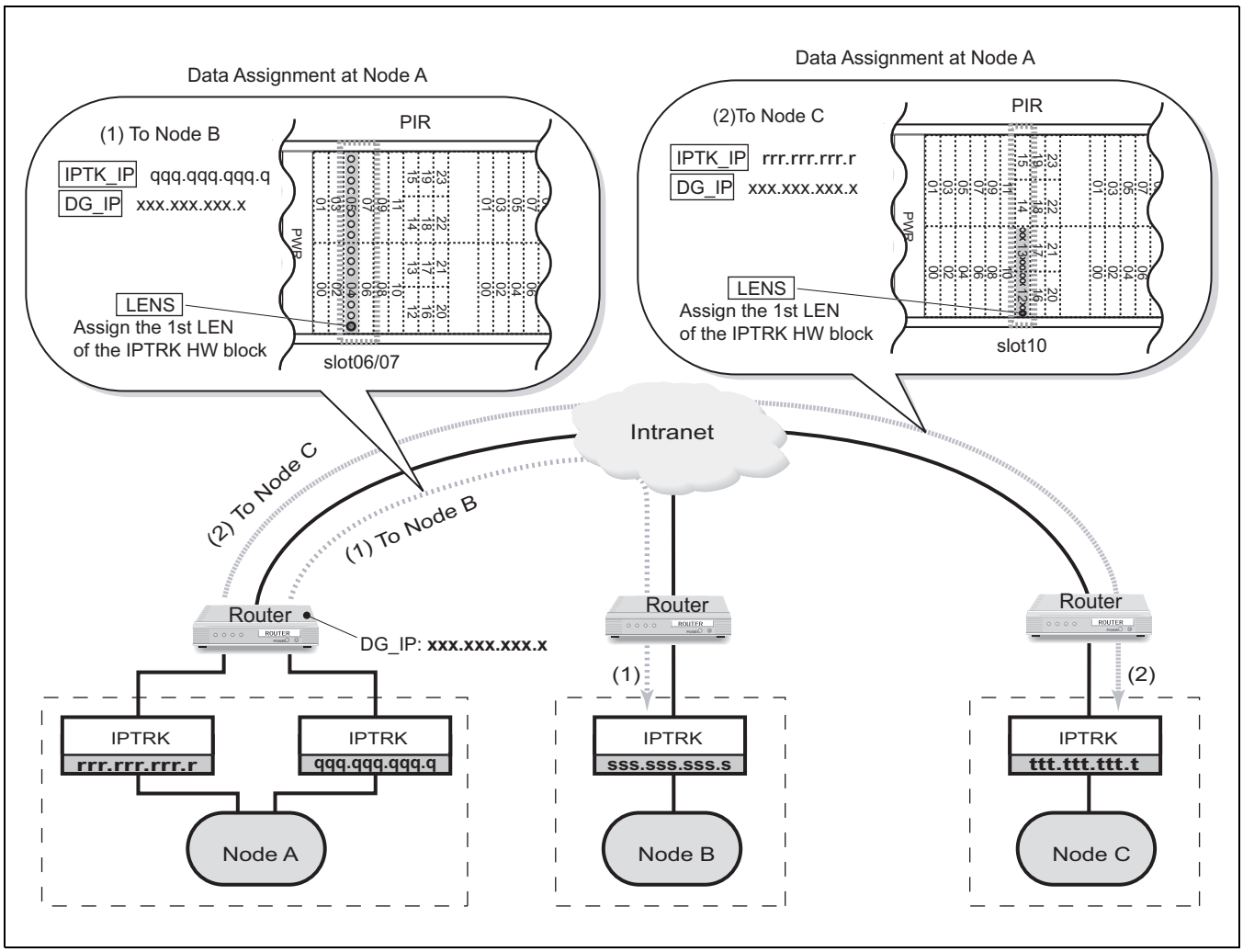
PAY_SIZ1: Specify the Payload Size 1 (G.711) data
- 0: Follows Switch Setting on the IPPAD Card
- 1: 10 (milliseconds)
- 2: 20 (milliseconds)
- 3: 30 (milliseconds)
- 4: 40 (milliseconds)

5-14: Not Used
- 15: G.711 is not used.

PAY_SIZ2:	Specify the Payload Size 2 (G.729a) data
0:	Follows Switch Setting on the IPPAD Card
1:	10 (milliseconds)
2:	20 (milliseconds)
3:	30 (milliseconds)
4:	40 (milliseconds)
	5-14: Not Used
15:	G.729a is not used
PAY_SIZ3:	Specify the Payload Size 3 (G.723.1) data
0:	Follows Switch Setting on the IPPAD Card
1:	10 (milliseconds)
2:	20 (milliseconds)
3:	30 (milliseconds)
4:	40 (milliseconds)
5-14:	Not Used
15:	G.723.1 is not used
FAX_DET:	Fax Error Detection Timer
0:	Follows to Switch Setting on the IPTRK
1:	Voice/FAX Automatic Distinction
2:	Fixed to Voice Mode
3:	FAX Detection Available Timer After Starting Speech => 1 minute
4:	FAX Detection Available Timer After Starting Speech => 2 minutes
5:	FAX Detection Available Timer After Starting Speech => 3 minutes
6:	FAX Detection Available Timer After Starting Speech => 4 minutes
7:	FAX Detection Available Timer After Starting Speech => 5 minutes
FAX_ECM:	Specify the FAX ECM Mode
0:	Follows to Switch Setting on the IPTRK
1:	ECM ON (Not Used)
2:	ECM OFF
FAX_TYP:	FAX Mode Control Precedence
0:	T.30 (Original Procedure) Not Used
1:	T.38 (UDP)
2:	T.38 (TCP) Not Used
FAX_RATE:	FAX Transmission Rate [bps]
0:	14400 bps
1:	2400 bps
2:	4800 bps
3:	7200 bps
4:	9600 bps
5:	12000 bps
RTCP_PTN:	RTCP Pattern
0:	Fixed Rate
1:	Random (Not Used)

RTCP_CYC:	RTCP Cycle
0:	5 seconds
1-5:	Not Used
6-120:	6-120 seconds

Assignment of IP Addresses of IPTRK Card, Its Connected Router (Example)

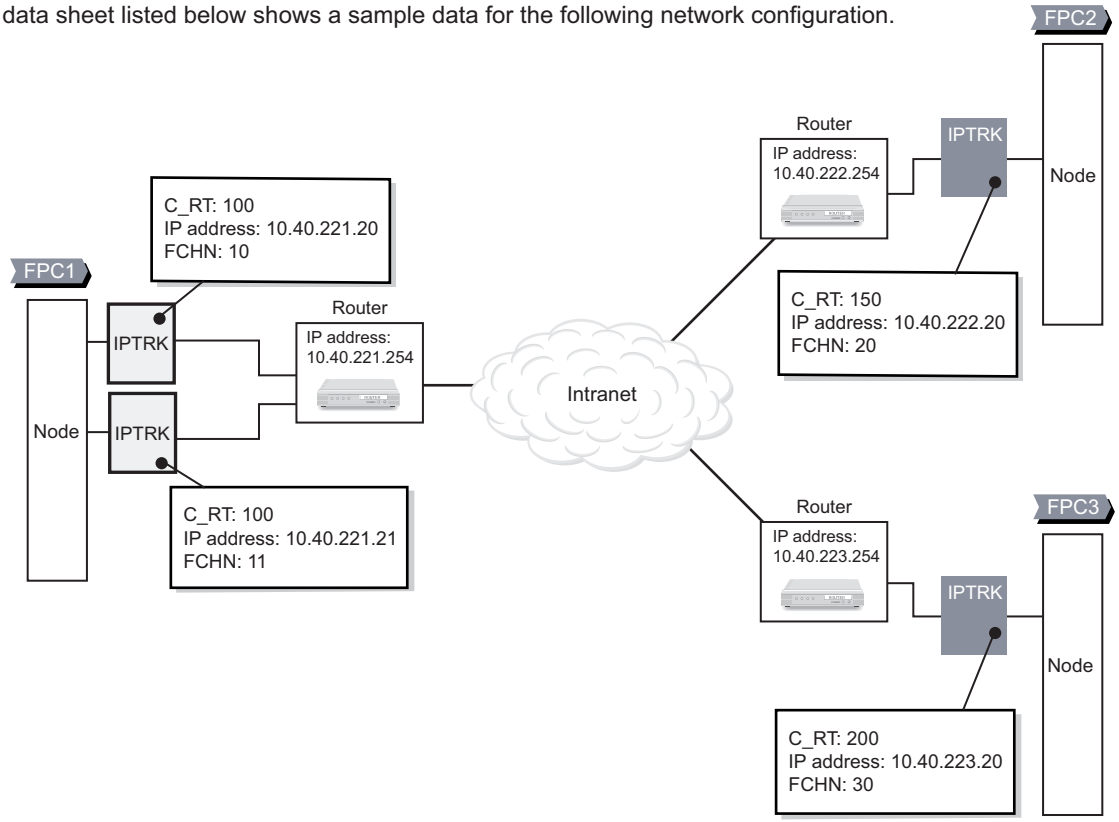


FCCS Routing Sample Data Sheets

1. FCCS Networking via IP (Point-to-Multipoint)

Sample Data Sheet (Example 1)

The data sheet listed below shows a sample data for the following network configuration.



Routing Information Assignment (by AETH) Example

FPC1
FPC=2, FCHN = 10, C_RT=100, DST_IP/NEXT_IP = 10.40.222.20
FPC=2, FCHN = 11, C_RT=100, DST_IP/NEXT_IP = 10.40.222.20
FPC=3, FCHN = 10, C_RT=100, DST_IP/NEXT_IP = 10.40.223.20
FPC=3, FCHN = 11, C_RT=100, DST_IP/NEXT_IP = 10.40.223.20

FPC2
FPC=1, FCHN = 20, C_RT=150, DST_IP/NEXT_IP = 10.40.221.20
FPC=3, FCHN = 20, C_RT=150, DST_IP/NEXT_IP = 10.40.223.20

FPC3
FPC=1, FCHN = 30, C_RT=200, DST_IP/NEXT_IP = 10.40.221.20
FPC=2, FCHN = 30, C_RT=200, DST_IP/NEXT_IP = 10.40.222.20

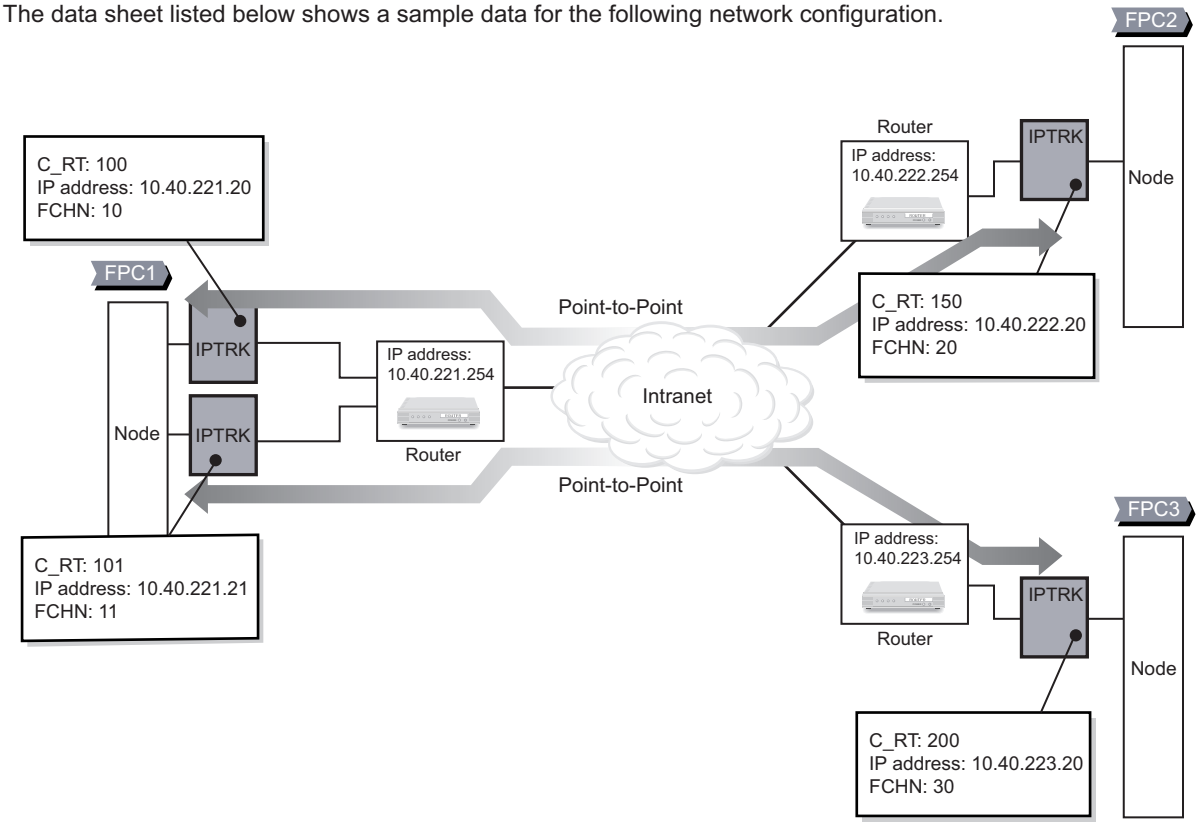
: IP trunk accommodating Speech and Control Channels.

: IP trunk accommodating Speech and Control Channels.
Speech Channels are shared between FCHN = 10 and FCHN = 11.

2. FCCS Networking via IP (Point-to-Point)

Sample Data Sheet (Example 2)

The data sheet listed below shows a sample data for the following network configuration.



Routing Information Assignment (by AETH) Example

FPC1 FPC=2, FCHN = 10, C_RT=100, DST_IP=10.40.222.20, NEXT_IP = 10.40.222.20
FPC=3, FCHN = 11, C_RT=101, DST_IP=10.40.223.20, NEXT_IP = 10.40.223.20

FPC2 FPC=1, FCHN = 20, C_RT=150, DST_IP=10.40.221.20, NEXT_IP = 10.40.221.20

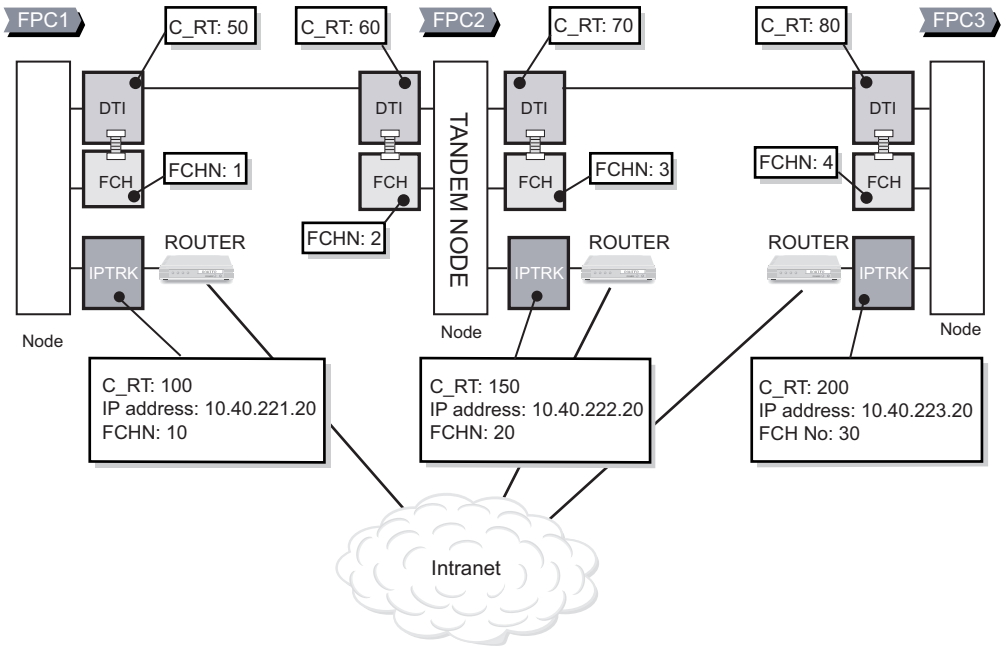
FPC3 FPC=1, FCHN = 30, C_RT=200, DST_IP=10.40.221.21, NEXT_IP = 10.40.221.21

: IP trunk accommodating Speech and Control Channels.

3. FCCS Networking via IP and FCH FCCS Networking (when assigning both AFPC and AETH)

Sample Data Sheet (Example 3)

The data sheet listed below shows a sample data for the following network configuration, where AFPC and AETH commands are required for an identical FPC at the tandem node.



Routing Information Assignment (by AFPC and AETH) Example

AFPC

FPC1

FPC=2, FCCH = 1, P_ROUTE=0/1, C_RT=50, FCHN=1
FPC=3, FCCH = 1, P_ROUTE=0/1, C_RT=50, FCHN=1

FPC2

FPC=1, FCCH = 1, P_ROUTE=0/1, C_RT=60, FCHN=2
FPC=3, FCCH = 1, P_ROUTE=0/1, C_RT=70, FCHN=3

FPC3

FPC=1, FCCH = 1, P_ROUTE=0/1, C_RT=80, FCHN=4
FPC=2, FCCH = 1, P_ROUTE=0/1, C_RT=80, FCHN=4

AETH

FPC1

FPC=2, FCHN = 10, C_RT=100, DST_IP=10.40.222.20, NEXT_IP = 10.40.222.20
FPC=3, FCHN = 10, C_RT=100, DST_IP=10.40.223.20, NEXT_IP = 10.40.223.20

FPC2

FPC=1, FCHN = 20, C_RT=150, DST_IP=10.40.221.20, NEXT_IP = 10.40.221.20
FPC=3, FCHN = 20, C_RT=150, DST_IP=10.40.223.20, NEXT_IP = 10.40.223.20

FPC3

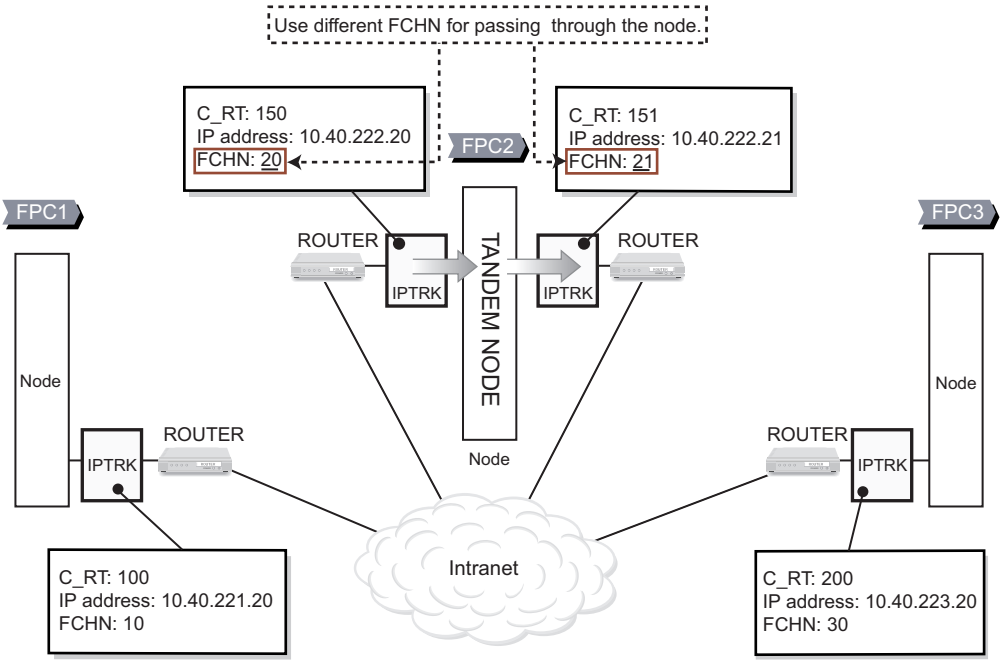
FPC=1, FCHN = 30, C_RT=200, DST_IP=10.40.221.20, NEXT_IP = 10.40.221.20
FPC=2, FCHN = 30, C_RT=200, DST_IP=10.40.222.20, NEXT_IP = 10.40.222.20

Note: Provided that both FCCS Networking via IP and FCH FCCS Networking are configured, it depends on the setting on “P_ROUTE” parameter whether the either routing take precedence. To take the priority for FCCS Networking via IP routing, assign data “1” to P_ROUTE parameter.

4. FCCS Networking via IP - FCCS Networking via IP Tandem Connection (when assigning AFPC dummy data)

Sample Data Sheet (Example 4)

The data sheet listed below shows a sample data for the following network configuration, where FCCS Routing Data is not assigned by using AFPC. In this case, be sure to assign the following Dummy Data.



Routing Information Assignment (by AFPC and AETH) Example

AFPC

FPC1

None

FPC2

FPC=1, FCCH = 1, P_ROUTE=1, C_RT=150, FCHN=20
FPC=3, FCCH = 1, P_ROUTE=1, C_RT=151, FCHN=21

FPC3

Be sure to assign this dummy data.

None

AETH

FPC1

FPC=2, FCHN = 10, C_RT=100, DST_IP=10.40.222.20, NEXT_IP = 10.40.222.20
FPC=3, FCHN = 10, C_RT=100, DST_IP=10.40.222.20, NEXT_IP = 10.40.222.20

FPC2

FPC=1, FCHN = 20, C_RT=150, DST_IP=10.40.221.20, NEXT_IP = 10.40.221.20
FPC=3, FCHN = 21, C_RT=151, DST_IP=10.40.223.20, NEXT_IP = 10.40.223.20

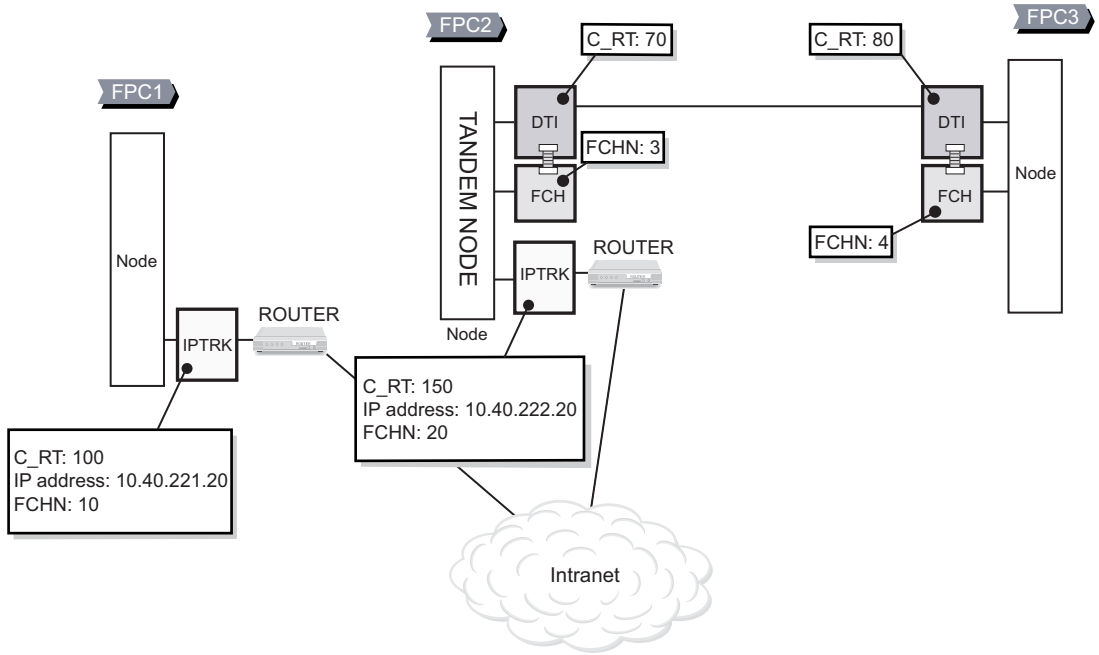
FPC3

FPC=1, FCHN = 30, C_RT=200, DST_IP=10.40.222.21, NEXT_IP = 10.40.222.21
FPC=2, FCHN = 30, C_RT=200, DST_IP=10.40.222.21, NEXT_IP = 10.40.222.21

5. FCCS Networking via IP and FCH FCCS Networking (when assigning AFPC or AETH)

Sample Data Sheet (Example 5)

The sheet listed below shows a sample data for the following network configuration, where AFPC or AETH command is required for an identical FPC at the tandem node.



Routing Information Assignment (by AFPC and AETH) Example

AFPC

FPC1

None

FPC2

Be sure to assign this dummy data.

FPC=1, FCCH = 1, P_ROUTE=1, C_RT=150, FCHN=20
FPC=3, FCCH = 1, P_ROUTE=0, C_RT=70, FCHN=3

FPC3

FPC=1, FCCH = 1, P_ROUTE=0, C_RT=80, FCHN=4
FPC=2, FCCH = 1, P_ROUTE=0, C_RT=80, FCHN=4

AETH

FPC1

FPC=2, FCHN = 10, C_RT=100, DST_IP=10.40.222.20, NEXT_IP = 10.40.222.20
FPC=3, FCHN = 10, C_RT=100, DST_IP=10.40.222.20, NEXT_IP = 10.40.222.20

FPC2

FPC=1, FCHN = 20, C_RT=150, DST_IP=10.40.221.20, NEXT_IP = 10.40.221.20
FPC=3, FCHN = 3, C_RT=70, DST_IP=0.0.0.0, NEXT_IP = 0.0.0.0

FPC3

Be sure to assign this dummy data.

None

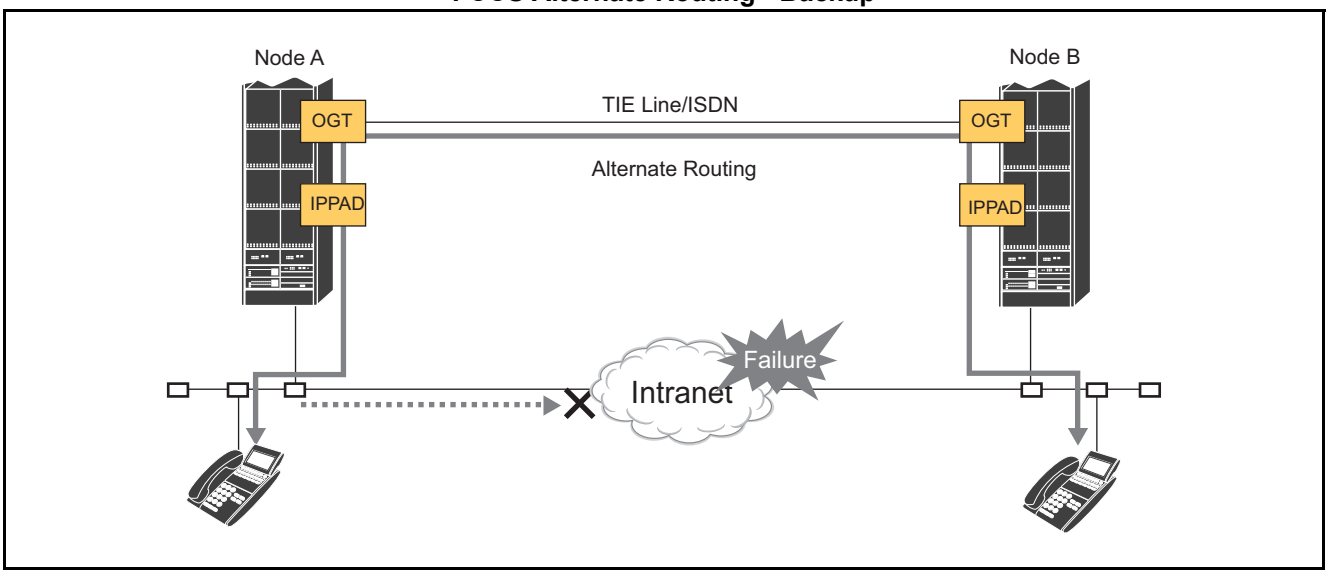
F-37 FCCS Alternate Routing - Backup

General Description

This feature automatically routes a FCCS over IP call over Alternate Routes when the first-choice Trunk Group is all busy or not available (such as the LAN cable is disconnected or LAN failure is occurred).

Note: This feature is available only for a migrated system from IPX and SV7000.

FCCS Alternate Routing - Backup



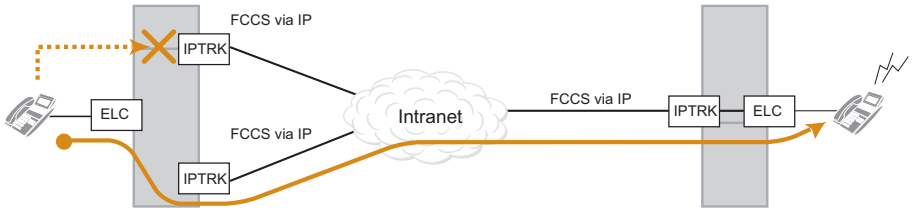
Service Conditions

Routing Patterns

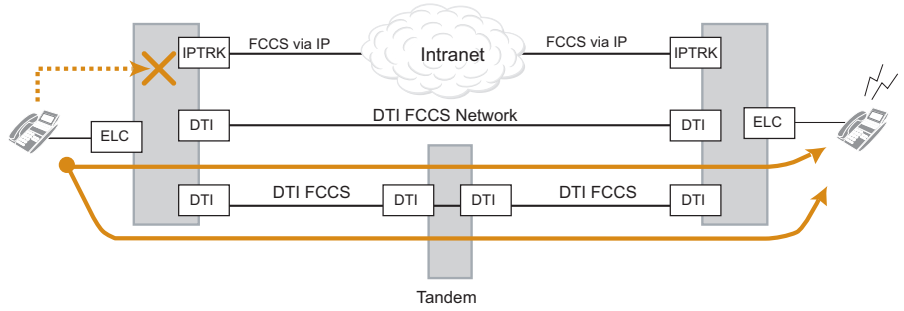
- Alternate Routing from IPTRK

Available Alternate Routing Patterns (1/2)

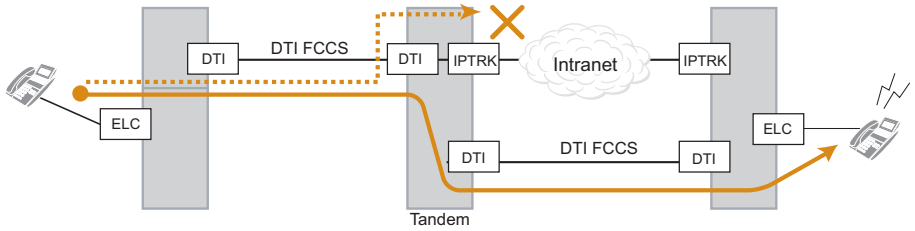
1. Routing to IPTRK (FCCS via IP)



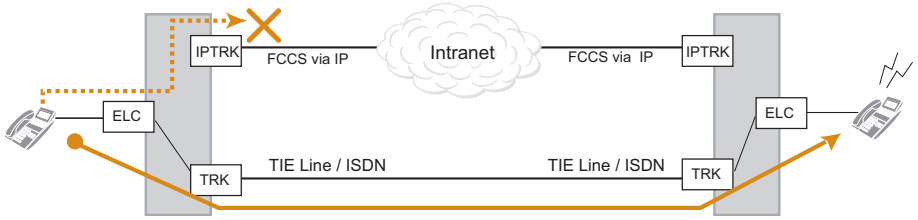
2. Routing to DTI (FCCS) at Self Node



3. Routing to DTI (FCCS) at Tandem Node



4. Routing to Trunk other than above (DTI, CCT, PRT etc.)

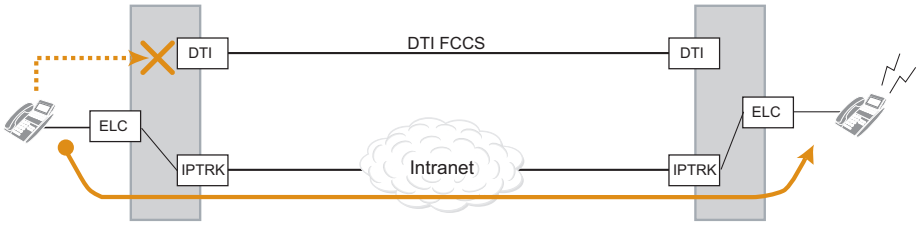




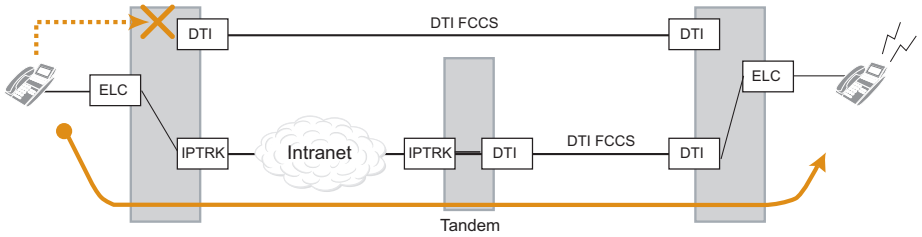
- Alternate Routing from DTI FCCS Trunk

Available Alternate Routing Patterns (2/2)

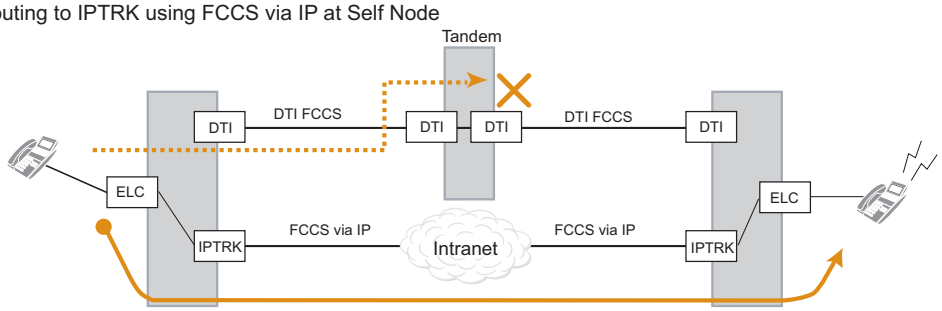
1. Routing to IPTRK



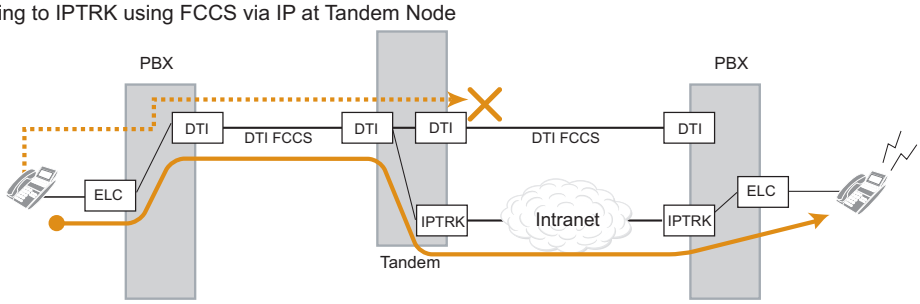
2. Routing to IPTRK via Tandem Node



3. Routing to IPTRK using FCCS via IP at Self Node



4. Routing to IPTRK using FCCS via IP at Tandem Node



: First Choice
: Alternate Routing

Alternate Routing IPTRK to TIE Line/ISDN

1. As a condition for this feature, Outgoing Calls via Alternate Routes (TIE Line/ISDN) must be made as usual.
2. The following Terminals listed in the figure below are available for this feature.
   - Calling Party: DtermIP, Dterm, Analog Terminal, ISDN Terminal, PS, COT, TIE Line, ATTOCON, DESKCON
   - Called Party: DtermIP, Dterm, Analog Terminal, ISDN Terminal
3. LCR or LCRS is required for this feature.
4. An Access Code for Alternate Routing is a maximum of 24 digits.
5. Alternate Routing to a Public Network is available for ISDN Lines only.
6. Use the Sub-address for Alternate Routes to a Public Network.
7. When Alternate Routing is performed, the DtermIP LCD shows an Outgoing/Incoming Trunk Call indication.
8. If all alternate routes assigned by AFRFL are busy, the system selects another route assigned by AOPR.

9. The following table shows available/unavailable features when performing FCCS Alternate Routing to TIE Line/ISDN.

FEATURE	PROVIDED/NOT PROVIDED
Station Hunting	Provided
Call Pickup	Not provided
Call Pickup - One Touch Key	Not provided
Direct Call Pickup	Not provided
Step Call	Provided
Call Back	Not provided
Call Waiting	Not provided
Executive Right-Of-Way	Not provided
Call Forwarding - Busy Line	Provided
Call Forwarding - Don't Answer	Provided
Split Call Forwarding	Provided
Call Transfer - All Calls	Provided
Blind Transfer to Station	Provided
Three-Way Calling	Not provided
8-Party Conference	Not provided
Hotline	Not provided
Priority Call	Provided when the destination is a Station
Station-to-Station Call	Provided
Emergency Call	Not provided
Uniform Call Distribution (UCD)	Provided
Virtual Line	Provided

Alternate Routing IPTRK to IPTRK, IPTRK to DTI FCCS, DTI FCCS to IPTRK

1. As a condition for this feature, Outgoing Calls via Alternate Routes (FCCS Routes) must be made as usual.

2. The following Terminals listed in the figure below are available for this feature.
   - Calling Party/Called Party:
     DtermIP, Dterm, Analog Terminal, ISDN Terminal, PS, COT, TIE Line, ATTOCON, DESK-CON

3. When an FCCS Trunk is all busy in the Alternating Route, the call can be advanced to another Route if any Flexible Route Data has been assigned by the AFRFL command.

Programming

When routing to Tie Line/ISDN Line Trunk

AFRFL -Assignment of Flexible Route Data for FCCS for LDM
Assign Access Code for the Alternative Route.

TN: Tenant Number (1-255)
FPC: Fusion Point Code of the Terminating Node (1-253)
ACC: Access Code for the Alternative Route

When routing to DTI FCCS

AFPC - Assignment of FCCH Routing Data for LDM

FPC: Fusion Point Code (1-253)
FCCH: 0/1= FCCH Not Use/FCCH Use
P_ROUTE: Priority Route (0/1= FCCH/FGH)
C_RT: Connection Route (1-1023)
FCHN/FPCN: 1-253

When routing to FCCS via IP from DTI FCCS

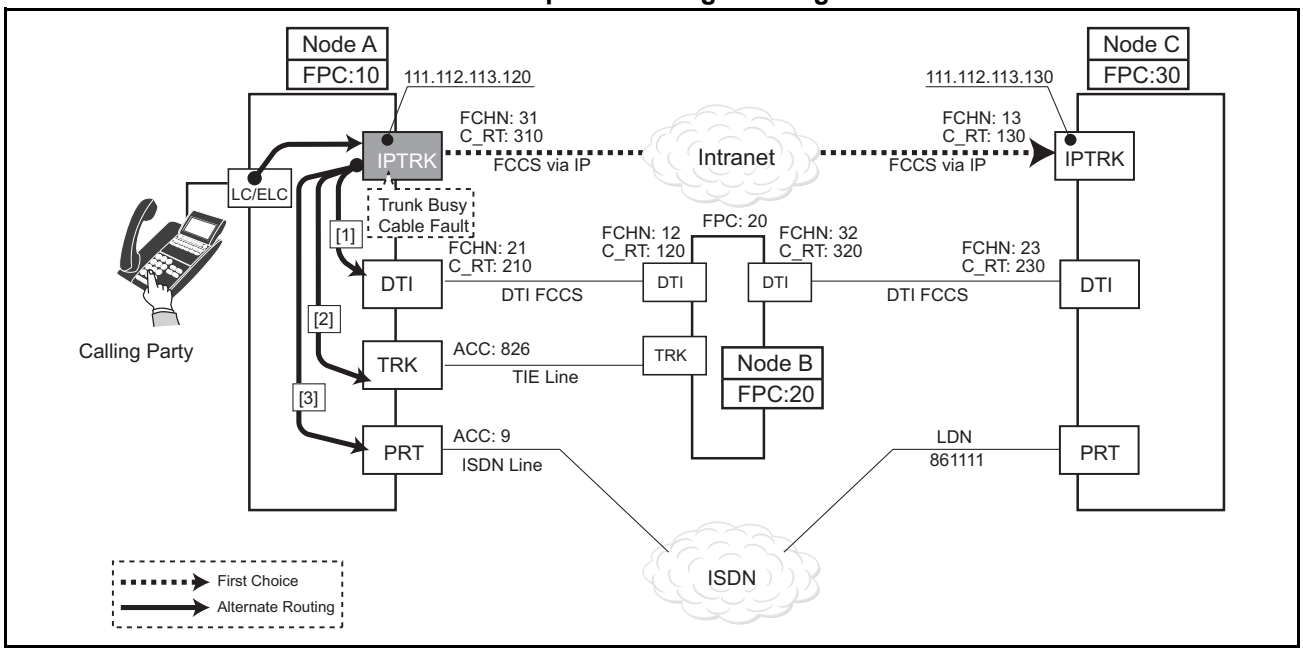
AETH - Assignment of External Router Connection Routing Data for LDM

FPC: Fusion Point Code (1-253)
FCHN: FCH Number (1-255)
C_RT: Connection Route (1-1023)
DST_IP: Destination IP Address
NEXT_IP: Next IP Address

Reference

This reference shows two examples of data programming for this feature

Example Data Programming



Data Programming at Node A

Routing to DTI FCCS

AFPC - Assignment of FCCH Routing Data for LDM
FPC=30 ⇐ Assign the FPC of the Destination Node
FCCH=1
P_ROUTE=1
C_RT=210
FCHN/FPCN=21
AETH - Assignment of External Router Connection Routing Data for LDM
FPC=30
FCHN=31
C_RT=310
DST_IP=111.112.113.130
NEXT_IP=111.112.113.130

Note: When both AFPC and AETH are assigned with P_ROUTE=1, FCCS via IPTRK assigned by AETH has priority. When all trunks are busy, the call routes to DTI FCCS.

Routing to Tie Line

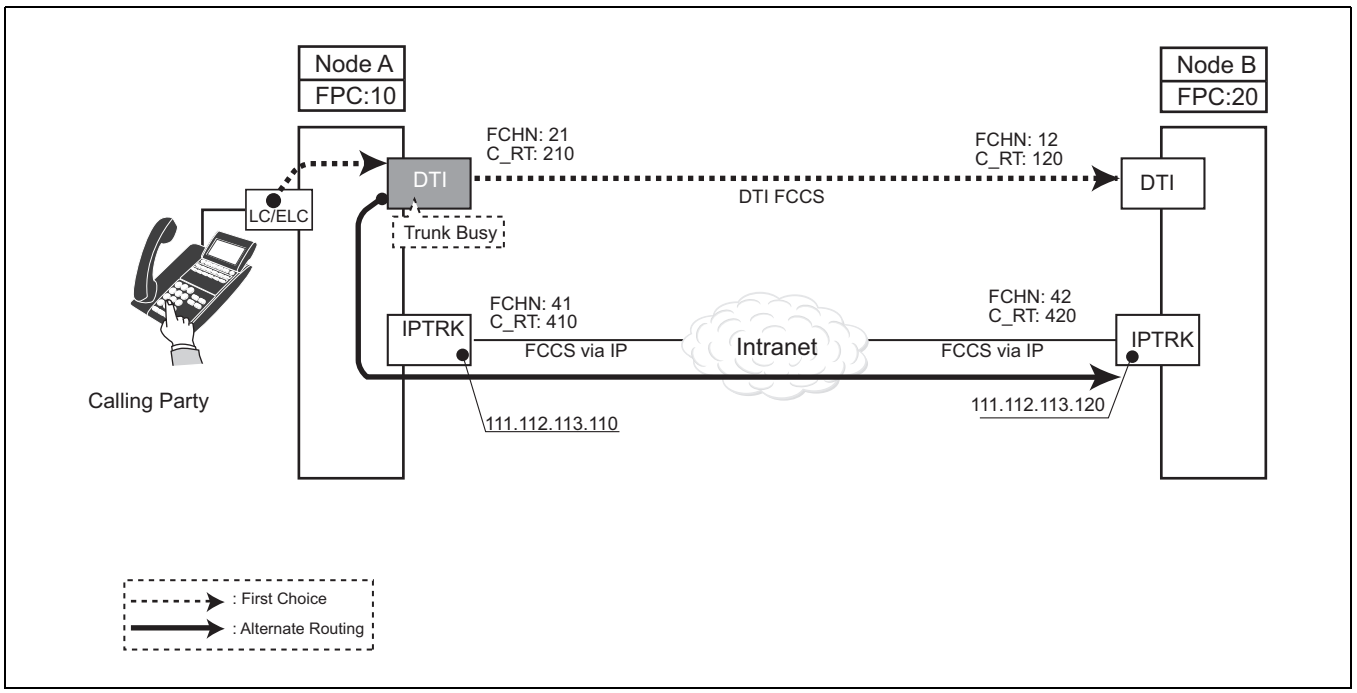
AFRFL - Assignment of Flexible Route Data for FCCS for LDM
TN=1
FPC=20 ⇐ Assign the FPC of the Destination Node
ACC=826 ⇐ Assign the Trunk Access Code for the Tie Line

Routing to ISDN Line

AFRFL - Assignment of Flexible Route Data for FCCS for LDM
TN=1
FPC=30 ⇐ Assign the FPC of the Destination Node.
ACC=9861111 * **Note 1**

Note 1: Be sure to add "*" at the end of the digits (sub-addressing).

Alternate Routing from DTI FCCS



Data Programming at Node A

AFPC - Assignment of FCCH Routing Data for LDM
FPC=20 ⇐ Assign the FPC of the Destination Node
FCCH=1
P_ROUTE=0
C_RT=210
FCHN/FPCN=21
AETH - Assignment of External Router Connection Routing Data for LDM
FPC=20
FCHN=41
C_RT=410
DST_IP=111.112.113.120
NEXT_IP=111.112.113.120

Note: When both AFPC and AETH are assigned with P_ROUTE =0, DTI FCCS assigned by AFPC has priority. When all trunks are busy, the call routes to FCCS via IPTRK.

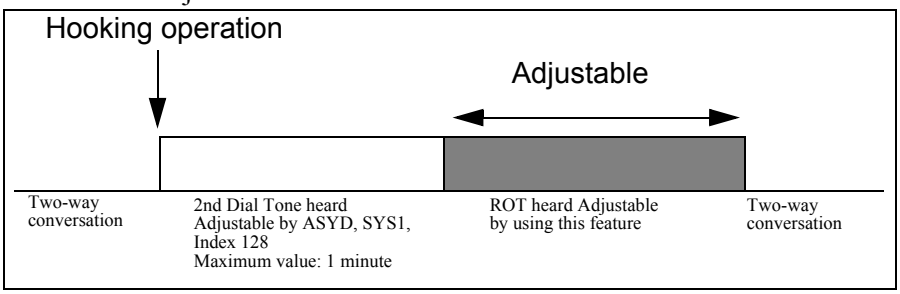
F-38 Flexible ROT Time

General Description

This feature enables the user to flexibly change the length of Reorder Tone (ROT) hearing time. At the recent system, hearing time of ROT is fixed to 30 seconds. Second dial tone is adjustable though the maximum time is only 1 minute and may be a cause of register block when system traffic is high because register is seized all time while second dial tone is being heard. Therefore, new timer is added to make ROT hearing time flexible.

Service Conditions

1. This feature is effective in all case which station hears the reorder tone.
   a.) Reorder tone when off-hook alarm.
   b.) Reorder tone when service feature restriction or route restriction.
   c.) Reorder tone when Dead level or unused number was dialed.
2. This feature is effective with the Non-IP terminal and IP terminal.
3. System data (ASYDL, SYS1, Index 663) need to be unified in all nodes when this feature is to be used in FCCS network environment.
4. The situation when Reorder tone has been adjusted is explained in the figure below.
5. When Reorder tone is adjusted.



6. This feature is not available with the PHS/PCS/VoWLAN system.

Interactions

Refer to the table below for further details on relation of this feature and Automatic Idle Return [A-69].

[F-38] Flexible ROT Time	[A-69] Automatic Idle Return	When Off-hook	When On-hook
Effective	Effective	Flexible ROT feature **Note 1**	Automatic Idle Return
Effective	Not effective	Flexible ROT feature	Flexible ROT feature
Not effective	Effective	30 seconds (default) **Note 1**	Automatic Idle Return
Not effective	Not effective	30 seconds (default)	30 seconds (default)

Note 1: Automatic Idle Return will be enabled by the System Data (ASYDL/ASYDN, SYS1, Index 686, Bit4=1) and the following conditions:
- Activating hands-free operation (on-hook)
- Using handset/headset (off-hook)

Programming

1. **Data assignment need be done in the procedure listed below.**

   ASYDL -Assignment of System Data for LDM
   SYS1, Index 663
   bit 0 - 3: 1 - =MTC
   bit 4 - 6: 0 - 7=TC
   bit 7: Not used

   TC: Timer Class, MTC: Miscellaneous Timer Counter
   TC = 0: -
   1: -
   2: -
   3: 2 seconds
   4: 30 seconds
   5: 5 minutes
   6: -
   7: -
   The value of TC x MTC will be the hearing time of Reorder tone.
   (Maximum: 5 minutes x 15 = 75 minutes)

   Note: Do not assign “0” for MTC value when TC = 3, 4 or 5.

   Note: There may be a gap between the assigned length of time and actual hearing time, the maximum range of gap will be same one assigned on “Interval range assigned for TC.”

   Note: Assign “00” for ASYDL, SYS1, Index 663 to disable this feature.

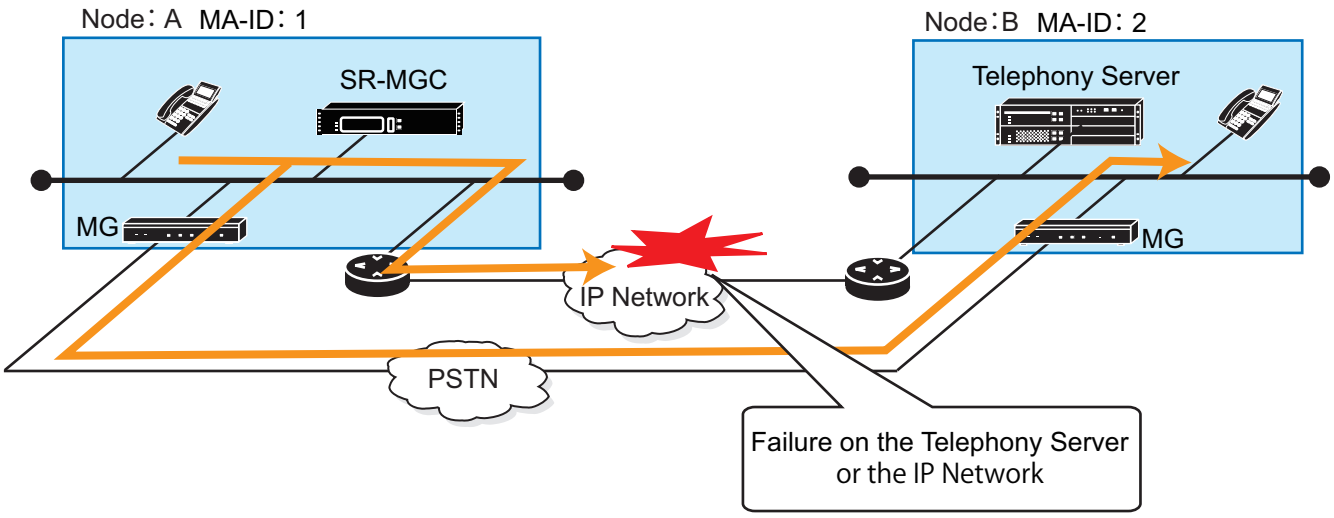
F-41 Flexible Network Re-routing

General Description

In the following cases, re-routing is available.

1. Automatic Re-routing when SR-MGC is in Operating Mode

This feature is useful when the Telephony Server or network failure occurs, and one of the nodes in a network is switched over to operation by SR-MGC, unable to connect to the opposite node via IP network. By assigning forwarding data to a trunk line in advance between the nodes, re-routing is automatically activated via a public network. Thus, this feature provides a connectivity equal to what could be attained by the Telephony Server via SR-MGC installed at each node.

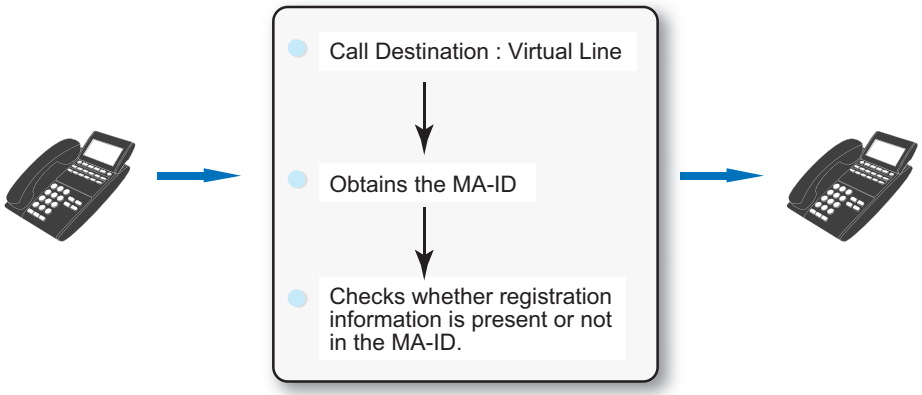


<Example>
In the above example, a network failure occurs at Node A. Therefore, Node A is operating by SR-MGC, Node B is operating by the Telephony Server.

- Call from Node A to Node B:
  The call is forwarded by forwarding control of Node A to PSTN, terminates to Node B via PSTN.
- Call from Node B to Node A:
  The call is forwarded by forwarding control of Node B to PSTN, terminates to Node A via PSTN.

2. Network Check Before Placing Call to Virtual Line

When placing a call to a virtual line between nodes, network check is performed at the Telephony Server side. At this time, if any network failure is detected, the call can be forwarded to a predetermined trunk line.



Note:

- The system specifies MA-ID assigned with the AMAA command for the virtual line. If the number of IP terminals/SIP Multiple Line terminals/SIP Handler Controlled SIP terminals* that are registered to the MA-ID becomes 0, network failure is detected.
- The terminal kinds that are checked for registration are IP terminals/SIP Multiple Line terminals/SIP Handler Controlled SIP terminals only. Other IP devices such as Analog MC, VS32, Media Gateway (MG) are not counted even if they are registered within the same MA-ID as the virtual line.
- IP terminals/SIP Multiple Line terminals whose configuration has been set as “Remote” are not counted even if they have been registered.
- Considerations should be taken for the following when attempting to load the MA-ID data with backup commands such as MEM_HDD and MEM_HDD_N while the system is in operation.

  If any difference is found between the MA-ID data on the MEM and that of CF card to which you are about to load the MA-ID data, discrepancy may be caused to the number of registration that is managed by a MA-ID basis when MA-ID data load is attempted. For this reason, network failure detection feature will not be able to work normally when calling a virtual line. In this case, after loading the MA-ID data, perform re-registration of all IP terminals/SIP Multiple Line terminals/SIP Handler Controlled SIP terminals in which MA-ID discrepancy is found.

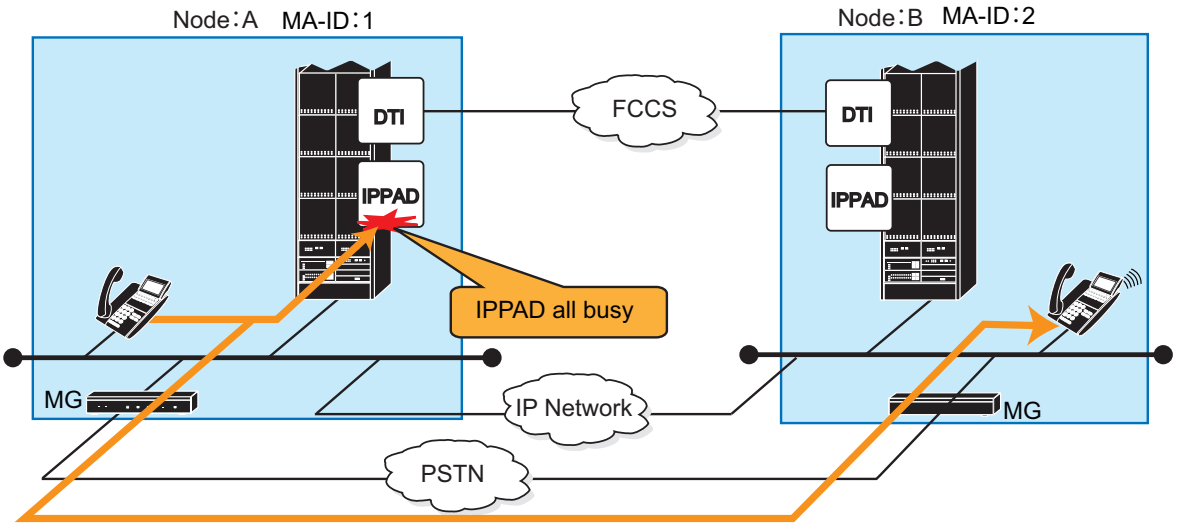
3. Re-routing between nodes in FCCS

In an FCCS system, if FCCS-link- all-busy or network failure is detected when a call is placed to another node, the call is forwarded to ISDN network specified in advance.

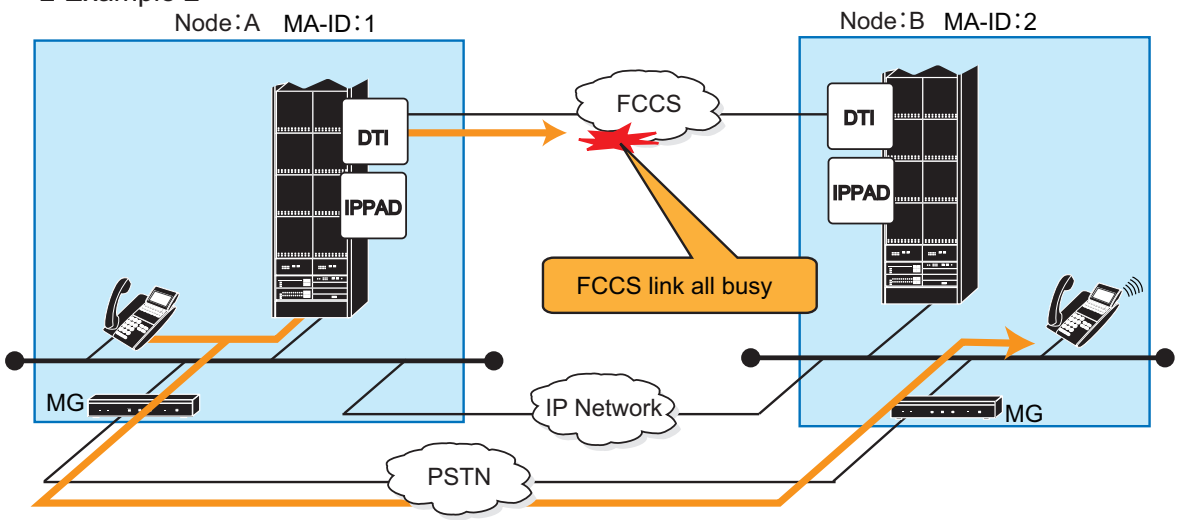
Note: FCCS link all busy means the following cases:

- IPPAD cards used for speech are all busy. (Only for IPPAD cards which are located in a call-originating node)
- Connection Trunks are all busy at the time of FCCS-DTI tandem connection.
- Bandwidth used for speech exceeds the limit which is configured with Peer to Peer Bandwidth Control [P-75].

■ Example 1



■ Example 2



Operating Procedure

Connection is automatically requested as follows depending on each terminal kind:

<Non-IP Terminal>

- Connection is requested by an IP device that is controlled by SR-MGC to a non-IP terminal that is controlled by the Telephony Server.

<IP Terminal/SIP Multiple Line terminals>

- Connection is requested to a logged-out Station (for which Logged Out IP Station - Call Destination has not been set).

<PS>

- Connection is requested to a PS that is out of range (for which Call Forwarding - PS Incoming Incomplete has not been set).

<SIP Handler Controlled SIP Terminal/SP Controlled SIP Terminal>

- Connection is requested to a SIP Handler Controlled SIP terminal/SP Controlled SIP terminal) which has no REGISTER information.

<Virtual Line>

- Connection is requested to a Virtual Line (no terminal registered to the same MA-ID **Note 1**).

<Attendant Console>

- Connection is requested by an IP device that is controlled by SR-MGC to an Attendant Console that is controlled by the Telephony Server.

Note 1: As to registration, care should be taken for the following:

- The terminal kinds that are checked for registration are IP terminals/SIP Multiple Line terminals/SIP Handler Controlled SIP terminals only. Other IP devices such as Analog MC, VS32, Media Gateway (MG) are not counted even if they are registered within the same MA-ID as the virtual line.
- IP terminals/SIP Multiple Line terminals whose configuration has been set as “Remote” are not counted even if they have been registered.
- Considerations should be taken for the following when attempting to load the MA-ID data with backup commands such as MEM_HDD and MEM_HDD_N while the system is in operation.
- If any difference is found between the MA-ID data on the MEM and that of CF card to which you are about to load the MA-ID data, discrepancy may be caused to the number of registration that is managed by a MA-ID basis when MA-ID data load is attempted. For this reason, network failure detection feature will not be able to work normally when calling a virtual line. In this case, after loading the MA-ID data, perform re-registration of all IP terminals/SIP Multiple Line terminals/SIP Handler Controlled SIP terminals in which MA-ID discrepancy is found.

Service Conditions

<Important Notices>

1. An ISDN line must be accommodated in node where this feature is used. Only an ISDN line can be specified as the destination trunk for re-routing. Assign the data so as to include ISDN route only in Route Selection Pattern for the re-routing. Available routing methods are LCR or LCRS. Note that when ISDN line is used for re-routing, the call is established by LCR, where the digits are sent to the network side after receiving all digits.
2. Unique Message Area ID (MA-ID) needs to be assigned to each node. MGs, IP Stations and SR-MGC in the same node must share the same MA-ID.
3. For re-routing, you have to assign the call forwarding destination number with the AFRRL/AFRRN command in advance. When assigning the data, set proper data in terms of numbering and restriction class so as not to forbid the call originator to reach the destination.
4. You can use Telephone Number only for the call forwarding destination (CFD) with AFRRL/AFRRN command. Station Number cannot be used.
5. Should this feature fail to re-route a call to the 1st RT due to all the ISDN trunks being busy or ISDN network going down, the existing re-routing feature will be activated according to the data setting of AOPR/AOPRL/AOPRN).
6. If N-ISDN2 is used for alternate re-routing, set CDN64 (NET) to 0 (user side, fix) in the ARTD/ARTDN command.

   Note: If CDN64 (NET) is set to 1 (network side), it will cause a ringback tone (RBT) or tone generated during re-routing to be unheard.

7. The following table shows the availability of re-routing to ISDN in the event of a network failure on the basis of the combination of communication lines and devices on the call source and destination.

X: Available P/A: Partially available N/A: Not available -: Not applicable

Call Source \ Call Destination		Registered to MGC			Registered to SR-MGC	
		• Digital Terminal • Analog Telephone • Virtual Line (Sub Line) • ISDN Terminal • CS/PS	• IP Terminal • SIP Multiple Line Terminal • Analog MC • Soft Phone • SP Controlled SIP Terminal/ SIP Handler Controlled SIP Terminal	• Attendant Console	• Sub Line (Virtual Line)	• IP Terminal • SIP Multiple Line Terminal • Analog MC • Soft Phone • SP Controlled SIP Terminal/ SIP Handler Controlled SIP Terminal
Registered to MGC	• Digital Terminal • Analog Telephone • Virtual Line (Sub Line) • ISDN Terminal • CS/PS	X	X	P/A	X	X
	• IP Terminal • SIP Multiple Line Terminal • Analog MC • Soft Phone • SP Controlled SIP Terminal/SIP Handler Controlled SIP Terminal	X	X	P/A	X	X
	• Attendant Console	N/A	N/A	N/A	N/A	N/A
Registered to SR-MGC	• Virtual Line (Sub Line)	X	X	P/A	X	X
	• IP Terminal • SIP Multiple Line Terminal • Analog MC • Soft Phone • SP Controlled SIP Terminal	X	X	P/A	X	X

Note: For the terminal types that can be accommodated in the SR-MGC, refer to the manual of SR-MGC.

8. The following are specific conditions for re-routing feature while SR-MGC is in Operating Mode.
   a.) ASYDL, SYS1, Index 1133, Bit 0 (Flexible Network Re-routing) must be enabled (Bit 0=1).
   b.) Re-routing can take place when SR-MGC which is in the same MA-ID group as the called trunk/station operates and the status of the called terminal is detected as a network failure (logged-out for IP Station, out-of-zone for PS). Note that this does not mean other terminals at the same node also fall into the network failure.
   c.) Re-routing is not available when the MA-ID of the called terminal is the same as that of the calling party.
   d.) MA-ID can be allocated for calling and called terminals (Station Number/Telephone Number (TELN)/Trunk/Logical Route Number (LGRT)/Line Equipment Numbers (LENS)/CS/Network) with the AMAA command.
   e.) When enabling this feature on Multiple Lines (Sub Lines), assign Multiple Lines to IP terminals in the same node.

      Note: If a Sub Line is assigned to IP terminals in a different node, network failure check will not work on the line.

   f.) When a call is placed to a destination in a different node via a Sub Line, the originator's MY Line is used for network failure check (of MA-ID).
   g.) To use this feature in FCCS connection, the following points must be considered:

   - When calls can be made between the nodes due to an FCCS link transmission line failure, MA-ID of the destination terminal cannot be read. In this case, automatic re-routing is available only when re-routing data are assigned to each TYPE=Telephone Number with the AFRRN command. (This feature is not activated with the rerouting data of TYPE=MA-ID.)

   - When a call is made from a Station operating under SR-MGC to a Station of another FPC node in the FCCS network, MA-ID of the destination terminal cannot be read. In this case, re-routing is available only when re-routing data are assigned to each TYPE=Telephone Number with the AFRRN command. (This feature is not activated with the rerouting data of TYPE=MA-ID.)

   h.) In the following cases, SR-MGC is placed in Operating Mode.

   - When SR-MGC starts operation because of a network failure or the Telephony Server failure

   - When "To Operating mode" is executed for Condition on Change Operating tab with the CMNT command.

      Note:

      - Even if SR-MGC has been assigned with the CMNT command, if the SR-MGC is not connected to network or no communications (Health Check) are made between MGC and the SR-MGC, when "To Operating mode" is performed for "Condition" on "Change Operating" tab, re-routing is activated.
      - Re-routing is not activated when the Telephony Server is initialized while the SR-MGC is in Operating Mode. This is because discrepancy is made between the two systems. To activate re-routing, you have to match the state of the two systems with the CMNT command (on Change Operating tab, Condition, To Operating mode).

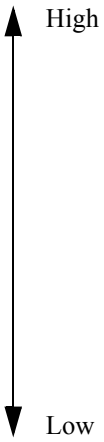
i.) Re-routing is also available when SR-MGC, which is in Operating Mode, requests connection to other devices (such as upstream Telephony Server or other FPC node in FCCS).
j.) When PIR becomes unavailable on the ground that the operation switched over to SR-MGC, a call request to a terminal accommodated in the PIR can also activate this feature.
k.) This feature is not available for calls originated over a trunk. When an outgoing trunk in a node outside of the source node cannot be seized because of a network failure, re-routing is not activated.

<Interactions with Other Features>

1. When multiple call forwarding features have been set, the priority order of them is shown below

Note: You can change the priority order with the following commands:
- ASYDL, SYS1, Index 1144, Bit 2=1: (2) Mobility Access < (4) Call Forwarding - All Calls
- ASFC/ASFCN SFI=209 RES=1: (3) Logged Out IP Station - Call Destination < (4) Call Forwarding - All Calls

Priority Order of Call Forwarding Features (Default)

Call Forwarding Features	Priority
(1) Number Sharing (Sub-station is set as Call Forwarding destination)	
(2) Mobility Access (MA) (EMEA Only)	
(3) Logged Out IP Station - Call Destination	
(4) Call Forwarding - All Calls	
(5) Flexible Network Re-routing (SR-MGC)	
(6) Multi-Boss/Multi-Secretary (EMEA Only)	
(7) Uniform Call Distribution (UCD)	
(8) Station Hunting - Terminal	
(9) Do Not Disturb	
(10) Call Forwarding - Don't Answer	

Changing the Priority Order of Call Forwarding Features

ASYDL, SYS1, Index 1144, Bit 2	ASFC/ASFCN SFI 209	Priority Order High <--> Low
0	0	(1) **(2) (3) (4)** (5) (6) (7) (8) (9) (10)
	1	(1) **(2) (4) (3)** (5) (6) (7) (8) (9) (10)
1	0	(1) **(3) (4) (2)** (5) (6) (7) (8) (9) (10)
	1	(1) **(4) (3) (2)** (5) (6) (7) (8) (9) (10)

2. Re-routing by this feature is counted as the number of Multiple Call Forwarding - All Calls/Busy Line. For the number of times of Multiple Call Forwarding, see “Programming on page 981”.
3. When using this feature, Billing Party is always the call originator regardless of the data setting of ASYD, SYS1, Index 20, Bit 7 (Billing Party for a call transferred by Call Forwarding - Outside Service).
4. Calls re-routed by this feature are counted as normal Route Peg Count.
5. Caller ID notification in this feature is as follows:

   - Station-to-Station calls: the Caller ID of the calling party is notified.
   - Trunk-to-Station calls: the Caller ID of the party who made the trunk call is notified.

6. Outgoing calls placed by using the following features can be re-routed by this feature:
   - Last Number Called - Single Line Station
   - Last Number Called - D
   - One touch Button
   - Speed Calling - System **Note 2**
   - Speed Calling - Station
   - Phone Book (Local Phone Book **Note 3**/System Phone Book), Terminal Phone Book
   - Message Reminder **Note 4**

   Note 2: Available only when the CD that is entered with the ASPD command is a Private Number in Dial Plan to Support IP Centrex Functionality.

   Note 3: Local Phone Book data is not copied from the Telephony Server to SR-MGC. For this reason, a station that is controlled by SR-MGC cannot make a phone call by using the Local Phone Book data.

   Note 4: A station that is controlled by SR-MGC cannot make a call to a station that is controlled by the Telephony Server by using Message Reminder.

7. This feature is not available for an outgoing call using Brokerage Hot Line.
8. This feature is not available for Preset Conference, where the conference leader dials the station number of conference leader port.
9. This feature is not available if a network failure occurs at a station that is specified as the following call forwarding destination.
   - Call Forwarding - All Calls - CCIS
   - Call Forwarding - Busy Line - CCIS
   - Call Forwarding - Don’t Answer - CCIS
10. If SR-MGC detects a network failure on a station that is specified as Call Forwarding - Don’t Answer destination when the forwarding is activated by Blind Transfer while Station/Trunk is on hold, this feature works as follows:

    - In case that Call Forwarding - Don’t Answer destination is PS/SP Controlled SIP terminal, this feature is available. However, if a network failure occurs on FCCS Network, this feature is not available (Call Forwarding - Don’t Answer also does not work).

    - In case that Call Forwarding - Don’t Answer destination is not PS/SP Controlled SIP terminal, this feature is not available (Call Forwarding - Don’t Answer also does not work).

11. When a call terminates on a station where a network failure is detected and the station detects a network failure, hunting does not take place for calls placed to the Hunting Group. In this case, the calls are forwarded according to the following forwarding information.

<Hunting Type>		<Re-routing Forwarding Information>
Pilot	=	to Pilot Station
Circular	=	to first called Station
Priority with Switchback	=	to first called Station
Priority	=	to first called Station

When Pilot Station (first called Station) is busy, and one of members other than the Pilot Station (first called Station) detects a network failure, hunting will take place. If there is no available Station in the group, the call is not forwarded and the caller receives BT.

Note: For inter node calls in FCCS system, considerations should be taken for the following. When a calling party dials the number of a Pilot Station (first called Station) and Bandwidth used for speech exceeds the limit which is configured with Peer to Peer Bandwidth Control [P-75] on the FCCS Link, hunting will take place (the call is not forwarded).

12. When the Pilot Station/first called Station of Hunting Group is a PS/SP Controlled SIP terminal, the following occurs when the PS/SP Controlled SIP terminal is called depending on the system data (ASYDL, SYS1, Index 688, Bit 0 (0/1=Hunting/PS Out of Zone Processing).
    a.) When “Hunting” is selected (Bit 0=0):

    When a network failure is found at the PS/SP Controlled SIP terminal which is working as a Pilot Station/first called Station of Hunting Group, hunting takes place to the next candidate Station in the same Hunting Group if available, and if not, it is not forwarded any more.

    b.) When “PS Out of Zone Processing” is selected (Bit 0=1):

    When a network failure is found at the PS/SP Controlled SIP terminal which is working as a Pilot Station/first called Station of Hunting Group, the call is forwarded according to the forwarding information specified for the Pilot Station/first called Station.

    Note: For inter node calls in FCCS system, considerations should be taken for the following.
    When a calling party dials the number of a pilot station/first called Station (PS/SP Controlled SIP terminal) and network failure is detected, the call is forwarded according to the call forwarding information set on the Pilot Station/first called Station regardless of the system data (ASYDL, SYS1, Index 688, Bit 0 (0/1=Hunting/PS Out of Zone Processing)).

13. When a call terminates on a station (UCD controlling station) where a network failure is detected, the call is forwarded according to the forwarding information of the controlling station.
    a.) When the controlling station is busy, and a member station detects a network failure, re-routing is not activated even if call forwarding information has been set for the member station. The incoming call terminates to available station in the UCD group.

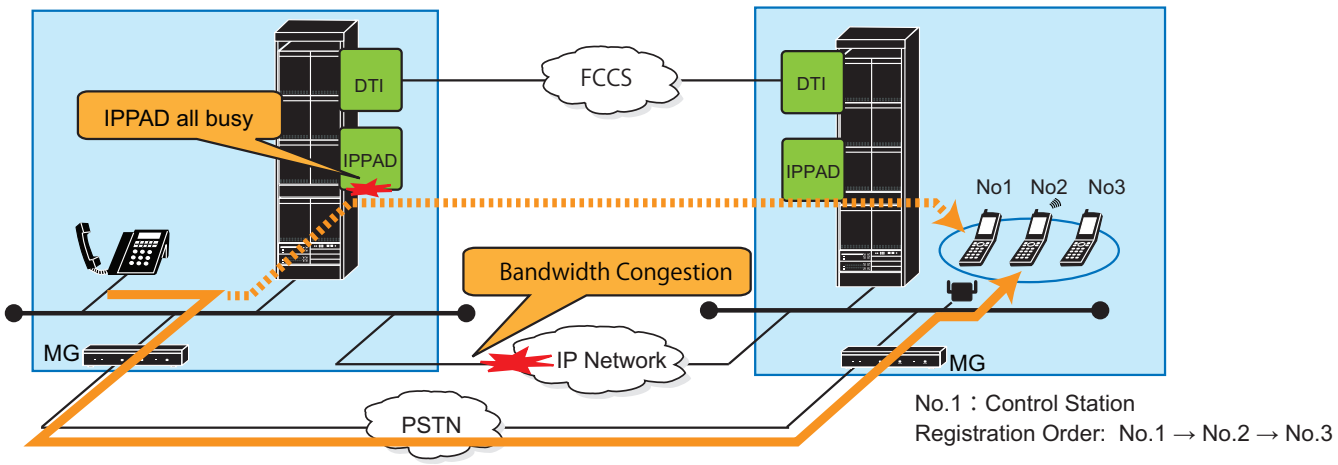
b.) If there is no available station in the UCD group since all stations are busy, the following occurs:

- When "UCD Queuing is Required (RBT)" with ASYD, SYS1, Index 60, Bit 3=0, re-routing is not activated. Places the call in queue and connects the call to a station if available.
- When "UCD Queuing is Not Required (BT)" with ASYD, SYS1, Index 60, Bit 3=1, the calling party receives BT.

Note: For inter node calls in FCCS system, considerations should be taken for the following:

- When a calling party dials a UCD number and IPPAD All busy is detected on Calling Node, the call is forwarded according to the forwarding information set on the controlling station.
- When a calling party dials a UCD number and Bandwidth used for speech exceeds the limit which is configured with Peer to Peer Bandwidth Control [P-75], re-routing is not activated and the incoming call terminates to available station in the UCD group.
- If IPPAD All busy is detected on Calling Node or Bandwidth used for speech exceeds the limit which is configured with Peer to Peer Bandwidth Control [P-75] when a calling party dials UCD number (PS/SP Controlled SIP terminal), the call is forwarded according to the forwarding information of the Controlling station. In this case, the forwarded call terminates not to UCD Control station but to available station in the UCD group.

[Example]
When a calling party dials a Control Station number (No.1), station hunting normally takes place according the registration order of terminals (No.1 - No.2 - No.3). However, when IPPAD All busy is detected on calling node or Bandwidth used for speech exceeds the limit, the call terminates not to No.1 station but to No.2 station.



Note: If BOSS ABSENT is set at BOSS' station, and CALL FORWARDING - ALL CALLS is set at secretary's station in "MULTI-BOSS/MULTI-SECRETARY Overriding from a Stand-In Secretary" feature, the call which terminates on BOSS' station number via FCCS link will be forwarded to the forwarding destination specified by the secretary. However, if FCCS-link-all-busy is detected at the forwarding destination, the call will be forwarded according to the forwarding information of the forwarding destination station.

14. When Least Cost Routing for MG-PRI is used, routing is determined by the MA-ID of the calling party at the time of re-routing.
15. When Location Diversity is in use, this feature is not supported.
16. ACD is not supported.

<Hardware/Software>

1. As to incoming calls to the following terminal/line, this feature can be disabled by system data. See "Programming " for more details.
   - IP terminal/SIP Handler Controlled SIP terminal
   - PS
   - SP Controlled SIP terminal
   - Virtual Line

   Note: If an FCCS link failure is detected when a call is placed to a PS/SP Controlled SIP terminal via FCCS network, terminal type cannot be read out. In this case, if ASYDL/ASYDN, SYS1, Index 1125, Bit 1 (Re-routing for legacy terminal/IP Station in the event of a network failure) is set to 1 (In service), re-routing will be activated regardless of ASYDL/ASYDN, SYS1, Index 1125, Bit 2 (Re-routing for PS/SP Controlled SIP terminal in the event of a network failure).

   Note: If an call is placed from a station which operates under SR-MGC (operating mode) to PS/SP Controlled SIP terminal in another FPC node via FCCS link, terminal type cannot be read out. In this case, if ASYDL/ASYDN, SYS1, Index 1125, Bit 1 (Re-routing for legacy terminal/IP Station in the event of a network failure) is set to 1 (In service), re-routing will be available regardless of ASYDL/ASYDN, SYS1, Index 1125, Bit 2 (Re-routing for PS/SP Controlled SIP terminal in the event of a network failure).

2. When a call is placed from a PS, the location is checked by using the MA-ID of CS that is used by the PS at the time of call origination. When PS is out of Zone or Power OFF, as MA-ID for PS, the MA-ID of CS that is used by the PS at the time of call origination/location registration/handover is applied (There is no need to assign MA-ID to PS using the AMAA command). If MA-ID has not been set for the CS, MA-ID=0000 is applied.
3. When a call is received by a PS while it is out of zone, the location check is made by MA-ID that has been allocated for the PS. When an IP terminal that is controlled by SR-MGC places a call to a PS that is controlled by other device in FCCS system, location check is not made for MA-ID, activating re-routing. When a central office incoming call is received by a PS which is in out of zone state, re-routing is not activated.

4. As to PS/SP Controlled SIP terminal, incoming incomplete timer value is fixed (about 8 seconds). The value is used for the system to recognize an out-of-zone terminal when a call is attempted to a PS/SP Controlled SIP terminal. For this reason, when CS/SP Controlled SIP terminal performs re-registration to the SR-MGC because of network failure, it will take about 8 seconds until the Telephony Server detects Health Check NG of the CS/SP Controlled SIP terminal, and the registration information held by the Telephony Server is deleted, and then re-routing is activated.
5. This feature can be enabled or disabled for the following calls by setting the system data ASYDL/ASYDN, SYS1, Index 1133, Bit 1 (Re-routing calls placed to an Attendant Console in the event of a network failure 0=Disabled/1=Enabled). The table below shows availability and conditions of re-routing when a call terminates to an attendant console for each type of attendant calling.

    - Call from IP equipment that is controlled by SR-MGC to an Attendant under the Telephony Server
    - Attendant call between nodes at the time of FCCS link transmission line failure in the FCCS system

X: Available P/A: Partially available N/A: Not available -: Not applicable

Features	Availability	Conditions
Operator Call	P/A	• Re-routing Call Forwarding information will be forwarded according to forwarding information against MA-ID that is set in ASYDL/ ASYDN, SYS1, Index 1134, Index 1135 (MA-ID of ATT). **Note:** MA-ID that is set for RT901(ATT) or RT915(Night ATT) with the AMAA command is not used. • In an FCCS System, when FCCS links to the Node which handles Ring Down (IC) Calls (ASYDL, SYS1, Index 640) are all busy, this feature is not available. • In an FCCS system, if FCCS links all busy is detected at a node accommodating an Attendant Console which is set as Call Forwarding - All Calls/Call Forwarding - Don't Answer destination by an Direct-In Termination destination, this feature is not available.
Individual Attendant Access	P/A	• Re-routing is available only for Individual Attendant Identification Number that is set with the ASATN command. Not available for Individual Attendant Identification Number that is set with the ASAT command. • Re-routing Call Forwarding information will be forwarded according to the Telephone Number of Individual Attendant Identification Number that is set with ASATN command, or forwarded according to the MA-ID that has been set for RT901 (ATT) or RT915 (Night ATT) with the AMAA command. • Although Individual Attendant Identification Number is set as forwarding destination for Call Forwarding-All Calls, Call Forwarding-Don't Answer, and Call Forwarding-Busy Line, the call forwardings are not activated.

Features	Availability	Conditions
LDN Call	P/A	• Re-routing is available only for LDN that is set with the ALDNN command. Not available for LDN Number that is set with the ALDN command. • Re-routing Call Forwarding information will be forwarded by the Telephone Number of LDN or MA-ID that is set with ASYDL/ ASYDN SYS1 Index 1134, 1135 (MA-ID of Attendant Console). **Note:** MA-ID that is set for RT901(ATT) or RT915(Night ATT) with the AMAA command is not used. • Availability of this feature is limited according to "TYPE: Night Transfer Type [0-3]" in the ALDNN command and day/night mode of node where an LDN call terminates. **Note:** The table below shows conditions for re-routing activation according to "TYPE: Night Transfer Type [0-3]" in the ALDNN command.

Note: Care should be taken for the following:
- Re-routing is not available for Attendant calls within the Telephony Server.
- This feature is not intended for an outgoing call from Attendant Console.

Note: When Flexible Network Re-routing - FCCS link all busy is enabled (ASYDL/ASYDN, SYS1, Index 1133, Bit 2=1), re-routing is available in the following cases.
- FCCS links are all busy.
- IPPAD cards used for speech are all busy. (Only for IPPAD cards which are located in a call originating node)
- Connection Trunks are all busy at the time of FCCS-DTI tandem connection.
- Bandwidth used for speech exceeds the limit which is configured with Peer to Peer Bandwidth Control [P-75].

X: Available N/A: Not available

Type (Night Transfer Type)	System	D/N mode	Re-routing	Conditions
0: No Night Transfer	SV9500	D	X	Re-routing is activated according to LDN number (pilot station number in an Attendant) that is set with the ALDNN command, or call forwarding information that is set for MA-ID in ASYDL/ ASYDN, SYS1, Index 1134, Index 1135 (MA-ID of Attendant).
		N3	X	Re-routing is activated according to LDN number (pilot station number in an Attendant) that is set with the ALDNN command, or call forwarding information that is set for MA-ID in ASYDL/ ASYDN, SYS1, Index 1134, Index 1135 (MA-ID of Attendant). **Note:** Although re-routing is activated, call is not terminated on an Attendant console and a calling party continues to hear a Ringback Tone (RBT).
	SR-MGC	D N3	X	Re-routing is activated according to LDN number (pilot station number in an Attendant) that is set with the ALDNN command, or call forwarding information that is set for MA-ID in ASYDL/ ASYDN, SYS1, Index 1134, Index 1135 (MA-ID of Attendant) regardless of D/N mode.
1: Telephone Number	SV9500	D	X	Re-routing is activated according to LDN number (pilot station number in an Attendant) that is set with the ALDNN command, or call forwarding information that is set for MA-ID in ASYDL/ ASYDN, SYS1, Index 1134, Index 1135 (MA-ID of Attendant).
		N3	X	When a network failure is detected at the telephone number of station in Night Connection destination, re-routing is activated according to call forwarding information that is set in LDN Night Connection destination.
	SR-MGC	D N3	X	Re-routing is activated according to LDN number (pilot station number in an Attendant) that is set with the ALDNN command, or call forwarding information that is set for MA-ID in ASYDL/ ASYDN, SYS1, Index 1134, Index 1135 (MA-ID of Attendant) regardless of D/N mode.

Type (Night Transfer Type)	System	D/N mode	Re-routing	Conditions
2: Trunk Line	SV9500	D	X	Re-routing is activated according to LDN number (pilot station number in an Attendant) that is set with the ALDNN command, or call forwarding information that is set for MA-ID in ASYDL/ ASYDN, SYS1, Index 1134, Index 1135 (MA-ID of Attendant).
		N3	N/A	Re-routing is not activated, and a call is forwarded to trunk-in number by Night Connection.
	SR-MGC	D	X	Re-routing is activated according to LDN number (pilot station number in an Attendant) that is set with the ALDNN command, or call forwarding information that is set for MA-ID in ASYDL/ ASYDN, SYS1, Index 1134, Index 1135 (MA-ID of Attendant) regardless of D/N mode.
		N3		
3: Abbreviated Digit Code	SV9500	D	X	Re-routing is activated according to LDN number (pilot station number in an Attendant) that is set with the ALDNN command, or call forwarding information that is set for MA-ID in ASYDL/ ASYDN, SYS1, Index 1134, Index 1135 (MA-ID of Attendant).
		N3	N/A	Re-routing is not activated, and a call is forwarded to Abbreviated Digit Code (ADC) that is set with the ASPDL/ ASPDN) command by Night Connection.
	SR-MGC	D	X	Re-routing is activated according to LDN number (pilot station number in an Attendant) that is set with the ALDNN command, or call forwarding information that is set for MA-ID in ASYDL/ ASYDN, SYS1, Index 1134, Index 1135 (MA-ID of Attendant) regardless of D/N mode.
		N3		

6. By setting system data, you can configure terminals to generate a beep when a station (including PS) activates this feature. See “Programming “for more details.

Note: Considerations should be taken for the following as to beep.

- The beep is the same as Overflow Tone that is provided for LCR/LCRS outgoing calls when route advance takes place. This tone is also the same as Service Set Tone (SST).
- No beep is provided for SP Controlled SIP terminals and SIP Handler Controlled SIP terminals.
- No beep is provided for VS32 conference, if re-routing is activated by network failure that is detected at conferee port and/or conference leader port.
- If ISDN trunks that are used for alternate routing are all busy, no beep will be provided when an idle trunk is seized as a result of Outgoing Trunk Queuing.
- If ISDN trunks that are used for alternate routing are MG(BRI) in Proprietary Protocol Mode, no beep will be provided.

7. The call cannot terminate to the PS/SP Controlled SIP terminal/SIP Handler Controlled SIP terminal while the terminal is moving to other node.

8. For a call originated from the SP Controlled SIP terminal, the location is checked by using MA-ID that has been set with AMAA command (KIND=Network) at the time of SP Controlled SIP terminal REGISTER. At the time of SP Controlled SIP terminal UN-REGISTER, the MA-ID that was applied at the time of REGISTER will be retained. If MA-ID of Kind=Network has not been assigned, MA-ID=0000 will be applied.
9. For a call originated from the SIP Handler Controlled SIP terminal, the location is checked by using MA-ID that has been set with AMAA command (KIND=LENS). At the time of SIP Handler Controlled SIP terminal UN-REGISTER, the MA-ID that was applied at the time of REGISTER will be retained. If MA-ID of Kind=LENS has not been assigned, MA-ID=0000 will be applied.
10. If a call is originated to an UN-REGISTER-state Standard SIP terminal which has not performed REGISTER after the system startup, MA-ID=0000 will be applied.
11. As for a call to the SIP Handler Controlled SIP terminal under the Telephony Server, when SR-MGC, which belongs to the same MA-ID group as the SIP Handler Controlled SIP terminal, goes into the operation mode and the status of the SIP Handler Controlled SIP terminal is detected as a network failure, the call does not arrive at the terminal although re-routing is activated.

<Interactions with Networks>

1. If connection between Telephony Servers is not available because of FCCS link transmission line failure, the priority order of call forwarding features is as follows when forwarding data has been assigned with the AFRFL command.

Call Forwarding Features	Priority
(1) Re-routing in case all the connection trunks are busy/FCCS link failure (AFRFL command)	High ↑
(2) Automatic Re-routing when SR-MGC is in Operating Mode **Note 5**	↓ Low

Note: When using “Automatic Re-routing when SR-MGC is in Operating Mode”, delete the forwarding data of the AFRFL command.

Note 5: In case that Flexible Network Re-routing - FCCS link all busy is enabled (ASYDL/ASYDN, SYS1, Index 1133, Bit 2=1), re-routing when FCCS links are all busy is available. FCCS link all busy means following status.

- IPPAD cards used for speech are all busy. (Only for IPPAD cards which are located in a call originating node)
- Connection Trunk is all busy at the time of FCCS-DTI tandem connection.
- Bandwidth used for speech exceeds the limit which is configured with Peer to Peer Bandwidth Control [P-75].

2. As to Tie Line/C.O. Line incoming calls, this feature can be disabled by system data. See “Programming ” for more details.

<Conditions Specific to This Feature>

1. You can assign the call forwarding destination number for each Telephone Number or MA-ID, where data for Telephone Number are preferentially read over data for MA-ID.
2. When a call is forwarded outside the network by this feature, the LCD of the originator’s Desktop terminal displays the current call status as “TRANSFER” without showing the intermediate station.

Note: You can specify whether to display call status on the LCD with the system data (ASYD, SYS1, Index 78, Bit 1), but information on the intermediate station is not provided regardless of the setting of the system data.

Note: In case that 1st route cannot be selected due to all the ISDN trunks being busy or ISDN network going down, and then 2nd route selected by the existing re-routing feature is activated according to the data setting of AOPR/AOPRL/AOPRN, “TRANSFER” is not displayed.

3. When a call is forwarded to public network by this feature, the LCD of the originator’s Desktop terminal displays the 8 digits of the Call Forwarding Information regardless of the setting of the system data (ASYDL, SYS1, Index 677, Bit 3), as follow:

T	R	A	N	S	F	E	R								1	2	3	4	5	6	7	8
1	0	:	1	4		A	M		M	O	N		1	4		N	O	V		2	0	1	1
	M	I	C																>	>	>	

4. Re-routing is also possible when making a call from a terminal that is registered in SR-MGC to a virtual line that is accommodated in other device’s Telephony Server regardless of the setting of ASYDL/ ASYDN, SYS1, Index 1125, Bit 4.

Note: If an FCCS link failure is detected when a call is originated to a virtual line between nodes, terminal type cannot be read out. In this case, if ASYDL/ASYDN, SYS1, Index 1125, Bit 1(Re-routing for legacy terminal/IP Station in the event of a network failure) is set to 1 (In service), re-routing will be available regardless of system data setting (ASYDL/ASYDN, SYS1, Index 1125, Bit 4).

<Other Conditions>

None

Programming

STEP 1: Assign the following system data as required with the ASYD/ASYDL/ASYDN command.

- ASYD, SYS1, Index 5:
  - Bit 1 to Bit 3: Maximum number of Multiple Call Forwarding-All Calls/Busy Line/Logout occurrence
    (1-5 times)

Bit 3	Bit 2	Bit 1	Number of Times	Bit 3	Bit 2	Bit 1	Number of Times
0	0	0	once	1	0	0	four times
0	0	1		1	0	1	five times
0	1	0	twice	1	1	0	
0	1	1	three times	1	1	1	

Note: Set twice or more for this feature.

- ASYD, SYS1, Index 69
  - Bit 7: Multiple Call Forwarding - Busy Line/All Calls/Logout
    0/1: Out of service/In service

Note: Set this bit only when you want to change the number of times for Multiple Call Forwarding.

- ASYDL, SYS1, Index 640
  - Bit 0 to Bit 7: Fusion Point Code (FPC) of the node providing Attendant Console. **Note 13**
- ASYDL, SYS1, Index 688:
  - Bit 0: Hunting/PS Out of Zone Processing when called PS is Out of Zone
    0/1: Hunting/PS Out of Zone Processing.

Note: Processing when a network failure is found at PS which has Station Hunting. If a network failure is found at a PS when selecting “PS Out of Zone Processing” the call is forwarded according to bandwidth congestion forwarding data that has been set to the PS. Note that this system data is shared with behavior pattern of general PS Out of Zone Processing.

- ASYDL/ASYDN, SYS1, Index 1125:
  - Bit 1: Re-routing calls placed to a legacy terminal/IP Station in the event of a network failure. **Note 6**, **Note 7**
    0/1 : Disabled/Enabled
  - Bit 2: Re-routing calls placed to a PS/SP Controlled SIP terminal in the event of a network failure. **Note 8**, **Note 9**
    0/1 : Disabled/Enabled

- Bit 3: Generate a beep when a legacy terminal/IP terminal call or PS call is automatically rerouted in the event of a network failure. **Note 6**, **Note 8**, **Note 10**
  0/1 : Disabled/Enabled
- Bit 4: Re-routing calls placed on a Virtual Line in the event of a network failure. **Note 7**
  0/1 : Disabled/Enabled
- Bit 6: Re-routing incoming TIE Line calls in the event of a network failure. **Note 11**
  0/1 : Disabled/Enabled
- Bit 7: Re-routing incoming C.O. Line calls in the event of a network failure. **Note 12**
  0/1 : Disabled/Enabled

Note 6: Set this bit to enable this feature for legacy/IP terminals/SIP Handler Controlled SIP terminal.

Note 7: Set this bit to enable this feature for virtual lines.

Note 8: Set this bit to enable this feature for PS.

Note 9: Set this bit to enable this feature for SP Controlled SIP terminal.

Note 10: Not available for SP Controlled SIP terminal/SIP Handler Controlled SIP terminal and VS32.

Note 11: Set this bit to enable this feature for TIE Line (CDN:6 (TCL) of ARTD/ARTDN is 4 - 31).

Note 12: Set this bit to enable this feature for CO Line (CDN:6 (TCL) of ARTD/ARTDN is 1 - 3).

- ASYDL/ASYDN, SYS1, Index 1133:
  - Bit 0: Flexible Network Re-routing
    0/1 : Disabled/Enabled
  - Bit 1: Flexible Network Re-routing - Attendant
    0/1 : Disabled/Enabled **Note 13**
  - Bit 2: Flexible Network Re-routing - FCCS link all busy
    0/1 : Disabled/Enabled

Note 13: Set this data to enable this feature for calls terminating to an attendant.

- ASYDL/ASYDN, SYS1, Index 1134, Index 1135:
  - MA-ID of the attendant
    [example: MA-ID = 16]
    Index 1134 = 10 (Hex)
    Index 1135 = 00 (Hex)

STEP 2: Assign the flexible rerouting data with the AFRRL/AFRRN command.

- When TYPE=TELN (Telephone Number)
  - **UGN**: User Group Number
  - **TELN**: Assign the Transfer Source Telephone Number (maximum 16 digits) for call forwarding source.

- **CFD (Call Forwarding Destination Number)**: Assign the call forwarding destination number (0-9, *, maximum 24 digits).**Note 18**, **Note 19**
  Enter a Public Network number (all digits) including the Trunk Access Code and Area Code.
- **Called Party Subaddress**: **Note 14**, **Note 15**
  Check the checkbox when adding the called party Subaddress, and then select either Automatic Addition Digit Code or Manual Input Digit Code.
- **Address Specification Method**:

  Automatic Addition Digit Code **Note 16**: Telephone Number/Private Number Code of the call forwarding destination is automatically added as the called party Subaddress.

  Manual Input Digit Code: Uses the data entered in IDC as the called party Subaddress.
- **IDC (Input Digit Code)**: Called Party Subaddress [0-9, *, maximum 24 digits ("#" is not used)]

- When TYPE=MA-ID
  - **MA-ID**: Charged Area ID (0-9999)
  - **CFD (Call Forwarding Destination Number)**: Assign the call forwarding destination number (0-9, *, maximum 24 digits). **Note 18**, **Note 19**
    Enter a Public Network number (all digits) including the Trunk Access Code and Area Code.
  - **Called Party Subaddress**: **Note 14**, **Note 15**
    Check the checkbox when adding the called party Subaddress, and then select either Automatic Addition Digit Code or Manual Input Digit Code.
  - **Address Specification Method**:
    Automatic Addition Digit Code **Note 16**: Telephone Number/Private Number Code of the forwarding destination is automatically added as the called party Subaddress.
    Manual Input Digit Code: Uses the data entered in IDC as the called party Subaddress.
  - **IDC (Input Digit Code)**: Called Party Subaddress [0-9, *, maximum 24 digits ("#" is not used)]

Note 14: Do not add the called party Subaddress when "#" is included in the Telephone Number/Private Number Code of the call forwarding destination.

Note 15: When you add the called party Subaddress, the number of digits for CFD assigned by AFRRL/AFRRN must be the same as MND assigned by AMND/AMNDL/AMNDN.
Set "Out of Service" for SUB (Sub Address Dialing) of ASPA/ASPAL/ASPN used for the trunk access code included in CFD assigned by AFRRL/AFRRN.
When the setting of the Subaddress is required for a normal call origination to an outside line, the trunk access code used for the normal call origination must be different from the one used for re-routing.

Note 16: Automatic Addition Digit Code
- When the AFRRL command is used, the Telephone Number (LDM) is added.
- When the AFRRN command is used and Dial Plan to support IP Centrex Functionality is disabled (ASYDN, SYS1, Index 818, Bit 5=0), the Telephone Number (NDM) is added.
- When the AFRRN command is used and Dial Plan to support IP Centrex Functionality is enabled (ASYDN, SYS1, Index 818, Bit 5=1), Private Number Code (**Note 17**) + Telephone Number (NDM) are added.

Note 17: ”Private Number Code” means a private number code which is assigned to UGN with the AUPVN command.

Note 18: You can assign the called number (remote access number) and called party subaddress (ID code plus called party number) which are used in “Automated Attendant [A-82]”/“Remote Access to System [R-2]” as a destination number.

Note 19: If Called Party Subaddress (ID code) is assigned as a destination number, note the following.

- Assign outside line access codes used in re-routing for destination number with and without ID code respectively.
- Assign the total digits of ID code and called subaddress according to the maximum digits indicated in “Maximum Digits for Called Party Subaddress Information” table.

[Example]

CFD: Call Forwarding Destination Number				Called Party Subaddress	
0	04XX-11-1111	*	3333	*	12345678901234
Access Code	Called Number 1111: Remote access code		ID code	(Automatic addition)	Called Party Subaddress

Maximum digits for called party subaddress

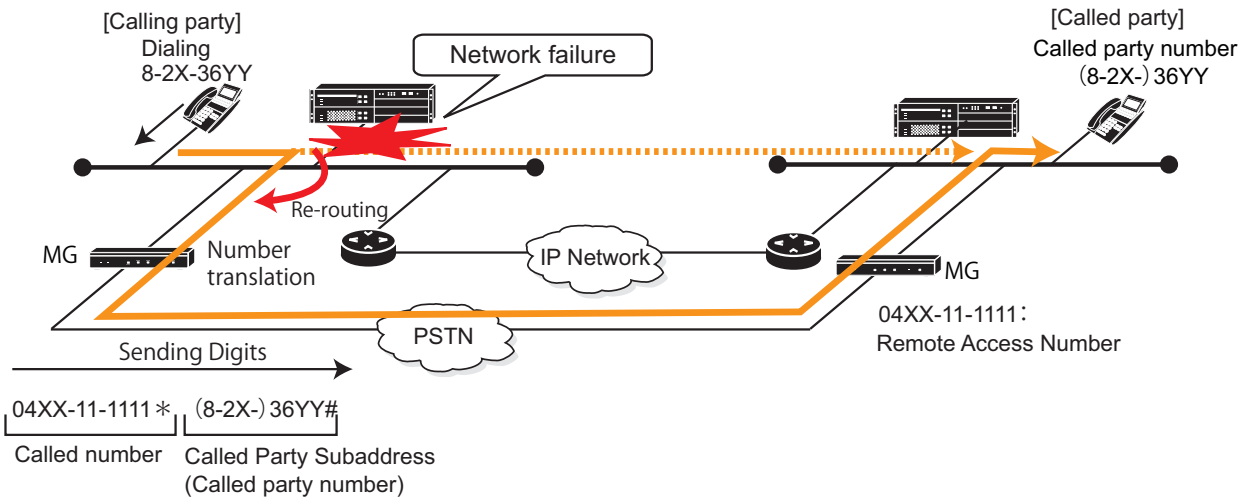
Maximum Digits for Called Party Subaddress Information

ARTD/ ARTDN CDN65 (INT)	Interface Specification for ISDN (0-15)	Maximum Digits
1	N-ISDN2	20
2	Australia	20
3	INS 1500(NTT)	19
4	ITU(CCITT), ETSI(PRI)	20
5	AT&T(#4/#5 ESS)	20
6	INS 64(NTT), ETSI(BRI)	19
7	NTT DMS100/DMS250	Not available
9	TTC Q931 a protocol Tie Line (JAPAN)	19
10	QSIG(ETS 300 172)/IS-11572	19

<Without ID code>
Called number (Remote access number) plus called subaddress (Called party number).

(Example)

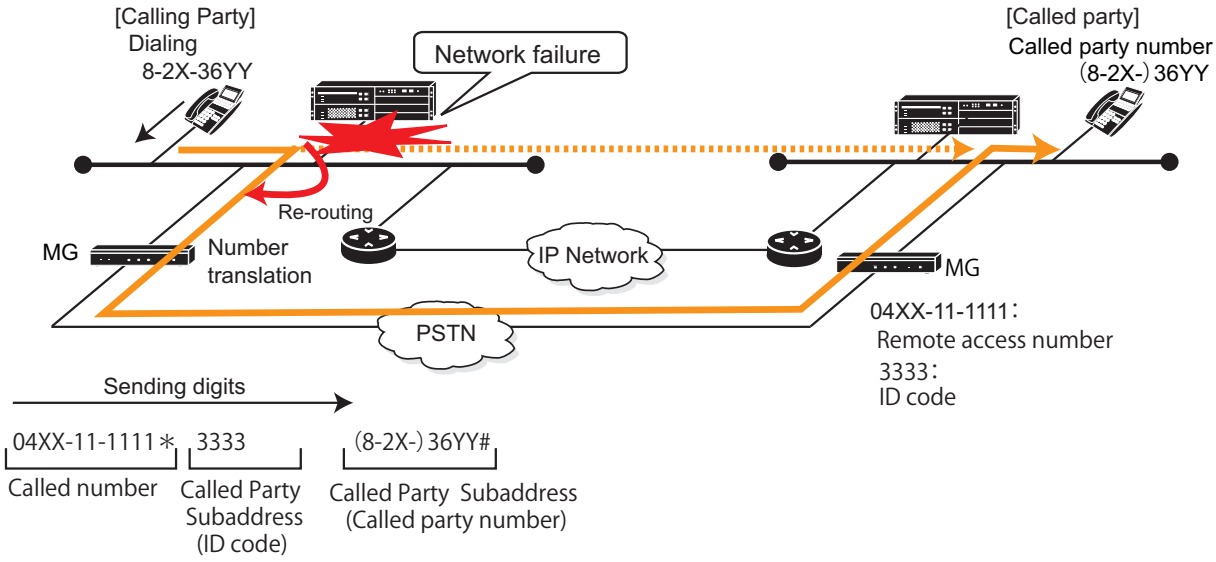
- **UGN**: 1
- **TELN**: 3683
- **CFD**: 0 (Access code) 04XX 11 1111 (Called number (1111=Remote access code))
- **Called Party Subaddress**: Address Specification Method = Automatic Addition Digit Code



<With ID code>
Called number (Remote access number) plus called subaddress (ID code plus called party number).

(Example)

- **UGN**: 1
- **TELN**: 3683
- **CFD**: 0 (Access code) 04XX 11 1111 (Called number (1111=Remote access code))
  *3333 (ID code)
- **Called Party Subaddress**: Address Specification Method = Automatic Addition Digit Code



STEP 3: Assign the MA-ID number for SR-MGC number with the CMNT command.

<Setting Tab>

MGC No.: Assign a number for the SR-MGC. (0-255)
Office Name: Set the office name for the SR-MGC. (Maximum 20 characters)
IP Address: Enter the IP address of the SR-MGC.

Note: Assign the actual IP address, not the default one.

Mode: Assign the type of changeover and recovery.

1 = Automatically switched to SR-MGC operation when communication failure has been detected between MGC and SR-MGC.

Automatically switched back to MGC operation when communication recovery has been detected between MGC and SR-MGC.

2 = Automatically switched to SR-MGC operation when communication failure has been detected between MGC and SR-MGC.

Automatically switched back to MGC operation when communication recovery between MGC and SR-MGC, and normality of communication between MGC and a selected terminal have been detected.

C_MGC: Assign the operating mode for SR-MGC.**Note 20**

0: Normal Operation (used to set the SR-MGC at the Self Node)

1: Backup node (used to set the SR-MGC at the backup node)

Note 20: This is required when using FCCS Cluster. For node where this data = 1, office data copy and initialization from the Telephony Server are not available.

Assigning MA-ID: Check this box to assign MA-ID for the SR-MGC.

MA-ID: Message Area ID (0 - 9999)

Note: Only one SR-MGC No. can be assigned for one MA-ID. Attempting to assign an already allocated MA-ID will result in error message.

STEP 4: Assign MA-ID and the Area Name with the AMAIL/AMAIN command.

MA-ID: Message Area ID (0-9999) Max. 4digits (fixed)

AREA NAME: Area Name (8-digit indication, 16-digit indication)

Note: For SP Controlled SIP terminal, set Message Area also for MA-ID.

STEP 5: Assign MA-ID allocation data for each terminal and trunk with the AMAA command.

KIND: Select a kind.

LENS: Assigned to the originating.

Network: Assigned to a network.

Note: Considerations should be taken for the following when setting this parameter.

[For SP Controlled SIP terminals]
- Be sure to select KIND = Network and assign MA-ID.
- MA-ID cannot be assigned to each SP Controlled SIP terminal.
- The MA-ID of KIND=Network is applied when the SP Controlled SIP terminal registers to the Telephony Server. If MA-ID is not assigned, MA-ID=0 is applied.
- The MA-ID registered at the time of REGISTER will be retained when the terminal performs UNREGISTER.
- If a call terminates on UN-REGISTER-state SP Controlled SIP terminal, which has not performed REGISTER after the system startup, MA-ID=0 will be applied.
- This feature cannot be used if KIND = Network has been set for equipment other than SP Controlled SIP terminal.
- If MA-ID is assigned to the station number of SP Controlled SIP terminal, the registration result will not be displayed in the setting data of the AMAA command.

[For SIP Handler Controlled SIP terminals]
- Be sure to select KIND = LENS and assign MA-ID.

[For PS]
- Assign MA-ID to CS-IP.
- Be sure to select KIND = LENS and FUNCTION = CS when assigning MA-ID.
- PS acquire MA-ID of CS at the time of location registration and the MA-ID is automatically applied to PS (There is no need to assign MA-ID to PS using the AMAA command). When a call is placed from PS or handed over to another base station, PS acquire MA-ID for CS newly, and the MA-ID is applied to PS. if MA-ID is not assigned to CS, MA-ID=0 will be applied to PS.

- When PS is in out of zone, the MA-ID applied to PS is retained.
- If a call is placed to out-of-zone-state PS, which has not performed REGISTER after the system startup, MA-ID=0 will be applied.
- If MA-ID is registered to station number of PS, the registration result is not displayed in the setting data of the AMAA command.

Note: Only one SR-MGC No. can be assigned for one MA-ID. Attempting to assign an already allocated MA-ID will result in error message.

- KIND=LENS, FUNCTION=Station: Used when specifying terminal type of terminal you want to set data by Station.

  DETAIL: Select a type where MA-ID is to be set (STN/TELN (NDM)/TELN (LDM))

  TYPE: Select whether STN is set individually or collectively (Individual/Collective)

  FPC: Fusion Point Code (when DETAIL=STN or DETAIL=TELN(LDM))

  TN: Tenant Number (when DETAIL=STN)

  UGNN/L: User Group Number (when DETAIL=TELN (NDM)/TELN (LDM))

  STN: Station Number (when DETAIL=STN, TYPE=Individual)

  TELNN/L: Telephone Number (when DETAI=TELN (NDM) or TELN (LDM) and TYPE=Individual)

  STN START-END: Specify the range of Station Number (when DETAIL=STN, TYPE=Collective)

  TELN START-END: Specify the range of TELN (when DETAIL=TELN (NDM) or TELN (LDM) and TYPE=Collective)

  MA-ID: Message Area ID (0 - 9999)

- KIND=LENS, FUNCTION=Trunk: Used when specifying terminal type of terminal you want to set data by Trunk.

  DETAIL: Select a type where MA-ID is to be set (RT/LGRT)

  TYPE: Select whether RT/LGRT is set individually or collectively (Individual/Collective)

  FPC: Fusion Point Code (when DETAIL=RT)

  RT: Route Number (when DETAIL=RT, TYPE=Individual)

  LGRT: Logical Route Number (when DETAIL=LGRT, TYPE=Individual)

  RT START-END: Specify the range of Route Number (when DETAIL=STN, TYPE=Collective)

  LGRT START-END: Specify the range of Logical Route Number (when DETAIL=LGRT and TYPE=Collective)

  MA-ID: Message Area ID (0 - 9999)

- KIND=LENS, FUNCTION=CS: Used when specifying terminal type of terminal you want to set data by CS.

  When selecting TYPE=Individual

  PCN: PHS Community Number

ERN: Calling Area Number

GRN: Group Number

CSN/ZTN: CS/ZT Number

MA-ID: Message Area ID (0 - 9999)

When selecting TYPE=Collective

PCN: PHS Community Number

ERN START-END: Specify the range of Calling Area Number

GRN START-END: Specify the range of Group Number

CSN/ZTN START-END: Specify the range of CS Number

MA-ID: Message Area ID (0 - 9999)

- KIND=LENS, FUNCTION=LENS: Used when specifying terminal type of terminal you want to set data by LENS.

  TYPE: Select whether LENS is set individually or collectively (Individual/Collective)

  FPC: Fusion Point Code

  LENS: Line Equipment Numbers (when TYPE=Individual)

  LENS START-END: Specify the range of LENS (when TYPE=Collective)

  MA-ID: Message Area ID (0 - 9999)

- KIND=Network: Used when allocating MA-ID to SP Controlled SIP terminal.

  FPC: Fusion Point Code

  IP Address: IP address of the node

  SubNet Mask: Subnet mask of the node

  MA-ID: Message Area ID (1 - 9999) **Note 21**

  Note 21: Assign the Message Area ID within the range from 1 to 9999. When 0 is assigned, the MA-ID is cleared.

G-2 Group Calling

General Description

This feature allows a user to dial certain frequently Called Stations within a System Group using an easily remembered number. The same number can be assigned to different user groups. The Group Call can initiate CALL WAITING - ORIGINATING [C-31] to a busy Station.

Operating Procedure

To initiate a Group Call

1. Lift the handset. Receive Dial Tone.
2. Dial the GROUP CALLING Access Code (maximum of three digits) plus the GROUP CALLING Station Code (one or two digits).
3. The call is completed.

To initiate a Group Call from a Dterm

1. Press the REDIAL key, plus GROUP CALLING Station Code (one or two digits).
2. Call is completed.

Service Conditions

1. Each Primary Station equipped in a Calling Group will program the GROUP CALLING Numbers through their telephone.
2. GROUP CALLING Data is retained in memory, even in the event of power failure, by means of Flash Card Backup via MEM_HDD.
3. The memory for GROUP CALLING is shared with SPEED CALLING - STATION or GROUP [S-21, 23].
4. A group for GROUP CALLING is identical to that of a SPEED CALLING - GROUP [S-23].
5. The number of groups per Module Group for GROUP CALLING and SPEED CALLING - STATION/ GROUP [S-21, 23] is limited to the following:

   GROUP CALLING Station Code: One digit (X) Maximum 100 Groups
   Two digits (XX)Maximum 10 Groups
6. The number of Stations that can be called per group can be set within the following limit:
   a.) Number of Groups × Number of Station Codes = Less than 1,000.
   b.) The number of Stations per group is assigned in multiples of 10.
7. For more information, refer to SPEED - CALLING GROUP [S-23].
8. A Station using GROUP CALLING must be allowed individual SPEED CALLING - STATION [S-21] and CALL WAITING - ORIGINATING [C-31] Service.
9. CALL WAITING - ORIGINATING [C-31] Service can be set when the Called Station is busy. CALL WAITING - ORIGINATING [C-31] will not be activated under the following conditions:
   a.) DO NOT DISTURB (DND) is set to the Called Station.
   b.) DATA LINE PRIVACY is set to the Called Station.
   c.) The Called Station is connected to the Attendant Console.
   d.) The Called Station is in a THREE-WAY CALLING [T-2] condition.
   e.) CALL BACK, CAMP-ON or CALL WAITING - ORIGINATING Service is set to the Called Station.

10. To initiate CALL WAITING - ORIGINATING with GROUP CALLING, a Dterm must:
    a.) Dial the CALL WAITING - ORIGINATING Access Code.
    b.) Dial the GROUP CALLING Access Code (maximum of three digits).
    c.) Dial the GROUP CALLING Station Code (one or two digits).
    d.) CALL WAITING - ORIGINATING Access Code is not needed if automatic is set.
11. This feature can be activated via FCCS.

Interactions

1. As an example of a GROUP CALLING application, the following conditions exist:
   System Station Numbering: 5XXXX and 6XXX
   Speed Calling Access: 3 (Assigned in ASPA)
   100 Groups of ten Stations
   GROUP CALLING Station Numbering consists of the following: 30-39
2. A Station in GROUP CALLING dials only two digits to reach other members of the group.
3. All normal STATION-TO-STATION CALLING [S-11] capabilities and restrictions apply, such as VOICE CALL [V-2], MESSAGE REMINDER [M-19], CALL BACK [C-1], etc.
4. GROUP CALLING applies to the My Line of the Dterm.

Programming

STEP 1: **ANPD** - Reserve a Number Level for Feature Access. Select Normal Service (N), Hooking Service (H) or Busy Service (B) from the combo box for Connection Indexes. (applicable to any feature assigned to this level). Assign NND in accordance with a predetermined Numbering Plan.

STEP 2: **ASPA** - Two Access Codes must be assigned for this feature. Assign a Tenant and an Access Code (ACC) for assigning Individual Speed Calling. For INDIVIDUAL SPEED CALLING - ENTRY, assign a Connection Index of "N: Normal". For SRV = "SSC: Service Code", assign SID 14. For INDIVIDUAL SPEED CALLING - ACCESS, assign a Tenant and an Access Code (ACC). Assign Connection Index of "N: Normal". For SRV = "SSC: Service Code", assign SID 21.

STEP 3: **ASFC** - Assign a Service Feature Restriction Class that allows SFI 19 to the Stations allowed SPEED CALLING - STATION [S-21].

STEP 4: **AISA** - Program the Tenant, Station, type of Station, and the number of SPEED CALLING blocks to be assigned to the Source Station. One Station of each GROUP CALLING Group is assigned as the Source. The Source then assigns all the Stations and Virtual Numbers as Abbreviated Codes.

TN: Tenant Number
STN: Station Number
MST/SLV: Primary/Secondary
1: MST (Master Station)
2: SLV (Slave Station)
BLOCK: Number of Blocks
MST STN: When assigning the secondary Station, assign the Primary Station Number.

STEP 5: **AKYD** - Assign the virtual circuit to a Desktop terminal key. Assign KYI = 2, Multi Line key, and assign the Station Number of the virtual circuit. This key will be the GROUP CALLING key.

STEP 6: **AISD -** can also be used to assign INDIVIDUAL SPEED CALLING Data.

G-3 Ground Button Access

General Description

This feature designates the GROUND BUTTON provided on a Station Telephone as either the only means or an optional means of performing any available Hooking Services.

Note: This feature is only available in Australia.

Operating Procedure

No manual operation is required.

Service Conditions

1. Hooking can be performed by pressing the GROUND BUTTON when the status of the Station permits the Hooking Service.
2. System Data can be set to designate only the GROUND BUTTON or both the GROUND BUTTON and switch hook as effective means for performing hooking operations.
3. If only the GROUND BUTTON is designated as effective by System Data, any Hooking Service attempt by means of a switch hook flash will result in the release of the existing connection.
4. In the case of CAS Satellite Offices, both the GROUND BUTTON and SHF must be designated as effective for hooking operations.

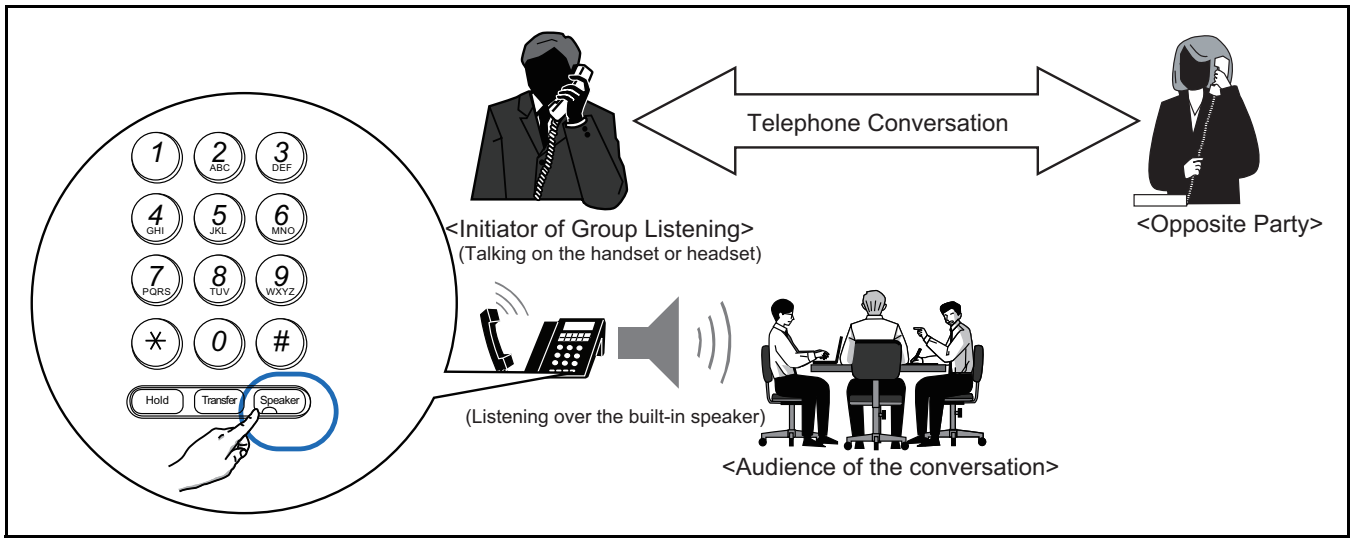
Programming

ASYD - SYS1, Index 6, Bit 3 (Tone for O/G Connection): 0/1=NEC Standard/Australian (Second Dial Tone on LCR). Assign "1".
SYS1, Index 68, Bit 6 (Switch Hook Flash): 0/1= Available/Not Available. Assign "1" when the GROUND BUTTON is used for hooking operations. Assign "0" when the GROUND BUTTON and SHF are used.

G-29 Group Listening

General Description

Group Listening enables you to hear the opposite party through the built-in speaker on the Desktop terminal by pressing the SPEAKER key while you are talking on the handset or headset.
To activate the built-in speaker during a call, you can simply press the SPEAKER key without replacing the handset (on-hook), thus allowing the opposite party's voice to come from both the handset or headset and the built-in speaker simultaneously.
This feature is useful when you need to share a telephone conversation with others nearby without them being overheard by the opposite party.



The following Desktop terminals are supported for this feature:

- Digital terminals: Dterm Series *i* and DT400 Series/DT300 Series **Note 1**
- SIP Multiple Line terminals: DT800 Series/DT700 Series

In this section, descriptions and specifications apply to both Digital and SIP Multiple Line terminals unless otherwise noted.

Note 1: Cordless terminal (DT430/DT330 (12BT)) and the terminal where a Bluetooth adapter (BHA-LD UNIT) is installed are not supported for this feature.

Operating Procedure

To activate Group Listening

1. Press the SPEAKER key on the Desktop terminal while talking on the handset or headset.
2. The Group Listening mode is activated and the LED on the SPEAKER key is turned on. (You can hear the opposite party through both the handset/headset and the built-in speaker.)

To deactivate Group Listening by using SPEAKER key

1. Press the SPEAKER key on the Desktop terminal in the Group Listening mode (then, the LED on the SPEAKER key is on).
2. The Group Listening mode is deactivated and the LED on the SPEAKER key is turned off. (You can hear the opposite party only through the handset or headset.)

To deactivate Group Listening by replacing the handset (on-hook)

1. Replace the handset (on-hook) in the Group Listening mode (then, the LED on the SPEAKER key is on).
2. The Group Listening mode is deactivated (then, the LED on the SPEAKER key remains on).

Service Conditions

<Important Notices>

1. This feature only operates on a Station where a Service Feature Restriction Class allowing SFI=258 is assigned with the ASFC/ASFCN command.
2. If any change is made in the above data (SFI=258) while the system is in operation, perform one of the following:

   [For Digital terminals]

   - Desktop terminal initialization by the MBST command
   - ELC card initialization

   Note: Changes in the data setting of this feature will not be made effective by inserting and removing the cable.

   [For SIP Multiple Line terminals]

   - System initialization or SP-PHD initialization **Note 2**
   - Terminal re-registration

   Note 2: To initialize SP-PHD, do one of the following:
   - System initialization
   - Internal PHE module initialization with the ADPM command

<Interactions with Other Features>

1. Group Listening is deactivated by:

   (a) switching lines with Line/Feature key or ORG key

   (b) reconnecting to the same line with RECALL key, or

   (c) activating the following features related to Call Hold:

- Non-Exclusive Hold -D [N-7D]
- Exclusive Hold -D [E-4D]
- Call Park Group -D [C-176D]
- Automatic Hold [A-68]

2. In the Group Listening mode, turning on the microphone (MIC) of the Desktop terminal does not enable the Hands-free features. To initiate hands-free conversation in this mode, turn on the microphone (MIC), and replace the handset (on-hook.)
3. While Group Listening is active, receiver volume control is available on the built-in speaker but not on the handset. This may need prior volume tune up of the handset or temporary deactivation of Group Listening to make adjustment.
4. If the function key data (the AKYD command, FKY=52) is assigned to a Line/Feature key to act like the SPEAKER key, Group Listening can also be activated by the key.
5. While Group Listening is active, Call Waiting Tone or Override Tone can be heard from both the handset or headset and the built-in speaker of the Desktop terminal.
6. Group Listening is available while using the consultation hold or 3-way conference. The Group Listening mode is maintained after the call is switched back to the basic two-way connection from the consultation hold/3-way conference.
7. **[For Digital terminals]**
   The following is the availability of Group Listening when a Desktop terminal is connected to an analog telephone using the Analog Port Adaptor (APR):

   (a) B1-Channel mode

   - Group Listening is not activated by pressing the SPEAKER key during a telephone conversation with the handset of the analog telephone.
   - Group Listening is activated by pressing the SPEAKER key during a telephone conversation with the handset of the Desktop terminal.

   (b) B2-Channel mode

   - Group Listening cannot be used on the analog telephone.
   - Group Listening can be used on the Desktop terminal even when a conversation is in progress on the analog telephone.]

8. This feature is not available for Hotel System.
9. This feature is not available for ACD/OAI system.

<Hardware/Software>

1. This feature is available for the following terminals:

   - Dterm Series *i*
   - DT400 Series/DT300 Series
   - DT800 Series/DT700 Series

Note: Cordless terminal (DT430/DT330 (12BT)) and the terminal where a Bluetooth adapter (BHA-LD UNIT) is installed are not supported for this feature.

2. Headsets can also be used.
3. **[For Digital terminals]**
   Requirement for circuit card to use this feature is shown in the table below:

Devices	Firmware
CH-16ELCA	SP-4035 16ELCA PROG-A Issue 1 or later
PA-16ELCN-A **Note 3**	SP-4013 16ELCN PROG-A Issue 1 or later

Note 3: This feature only works when the Terminal Mode Setting of Switch 11 (SW11-5/6) on ELC circuit card is set to ON/ON (Standard Setting). For details on the switch settings of the circuit cards, see sections related to ELC cards in Chapter 3 of the Circuit Card Description.

<Interactions with Networks>

None

<Conditions Specific to This Feature>

1. Acoustically degraded conditions such as background noise and residual echo can adversely affect communication with Group Listening.
2. In the Group Listening mode, clarity of the voice through the built-in speaker can be reduced.

<Other Conditions>

None

Programming

STEP 1: **ASYDL** - Assign "1" (the memory block is used) to the corresponding bit.
SYS1, Index 513 (Local Data Memory (LDM) usage)
SYS1, Index 514 (Network Data Memory (NDM) usage)

STEP 2: **ASFC/ASFCN** - Assign the following data to the Service Feature Restriction Class (SFC) for each station.
D/N: D/N (Day/Night Mode)
TN: 1 - 63 (Tenant Number)
SFC: 0 - 15 (Service Feature Restriction Class)
SFI: 258 (Group Listening) **Note 4**
RES: 1 (Allowed)

Note 4: Assign the same settings to both Day and Night modes if ASYD, SYS1, Index 75, Bit 1=1.

H-1 Hotline

General Description

This feature permits a pair of Station sets to be associated with one another on an automatic ringdown basis.

Operating Procedure

a.) Station A lifts the handset.
b.) Station B is rung.
c.) Station B lifts the handset. The conversation proceeds.

Service Conditions

1. If both phones in a Hotline Pair are equipped with dials, either one may transfer a Hotline Call to another Station.
2. Only one call destination is available per Hotline station.
3. There is no limit to the number of Hotline Stations.
4. Hotline assignments are programmed into System Data via the PCPro.
5. The Hotline Station will hear Reorder Tone when the Called Station is in busy, lockout or make busy condition.
6. When the Station called from the Hotline has activated CALL FORWARDING - BUSY LINE [C-2], the call will be forwarded when the Called Station is busy.
7. When the Station called from the Hotline has activated CALL FORWARDING - ALL CALLS [C-5], the call will be forwarded.
8. When the Station called from the Hotline has activated CALL FORWARDING - DON'T ANSWER [C-3], the call will be forwarded when the Called Station doesn't answer after a predetermined time interval.
9. STATION HUNTING [S -7, 8, 9] and UCD [U-1] will be activated for a call from the Hotline, but the Incoming Hotline Call cannot be placed in the UCD [U-1] Queue.
10. Calling party hears Reorder Tone (ROT) in the following cases.
    a.) When the Called Station has set CALL FORWARDING - ALL CALLS - OUTSIDE [C-28]
    b.) When the Called Station has set CALL FORWARDING - BUSY LINE - OUTSIDE [C-60] and line is busy
    c.) When the Called Station has set LOGGED OUT IP STATION-CALL DESTINATION [L-53] (for Trunk) and in logout status
11. An ISDN Terminal cannot be assigned as a destination.
12. Only the Station assigned Telephone Number is available for activating this feature via FCCS.
13. The Station, the Attendant Console/Desk Console (Operator Call), and the Trunk can be a destination for Hotline.
14. When Call Forwarding - All Calls/Busy Line/Don't Answer Service has been set to the Called Station, these features are available for the call to the Station.
15. When a Calling Party encounters a Busy Station, the call may bridge into the connection depending on the data assigned by ASYD.

Interactions

1. The connecting Station does not have to be a HOTLINE Station. If the connecting Station is busy, the HOTLINE Station will hear Reorder Tone.
2. CONSULTATION HOLD [C-17]: The HOTLINE Station may put the Destination Station on CONSULTATION HOLD [C-17]. In this state, the HOTLINE Station may make an Outgoing Call, and is therefore affected by CLASS OF SERVICE - INDIVIDUAL [C-15].
3. CALL WAITING - ORIGINATING [C-31]: A HOTLINE will not be allowed to camp-on to a busy Destination Station.
4. If the connecting Station is also a HOTLINE Station, then the reverse connection must also be programmed.

Programming

STEP 1: **ASDT** - Assign the Station as a TEC = 14 for HOTLINE.

STEP 2: **AHLS** - Assign the Station Number programmed above for the following parameters:

HOT TN: HOTLINE Tenant Number
HOT STN: HOTLINE Station Number
CON TN: Destination Tenant Number
CON STN: Destination Station Number

FCCS Programming

This data setting describes only the additional or replacement commands used to activate FCCS Services. When used as an FCCS feature, these commands replace the non-network commands (for example, AAED is replaced by AAEDN). Note that the data setting for FCCS Service varies depending on the Node.

AHLSN - This data must be assigned to the NCN.

HOT UGN: HOTLINE User Group Number
HOT TELN: HOTLINE Telephone Number of the Calling Party
TYPE: 1 (The Called Party is a Station)
CON UGN: User Group Number of the Called Party
CON TELN: Telephone Number of the Called Party

Note: The following data should also be set in addition to the above-mentioned commands. ASDT must be set to the Node accommodating the Calling Party.

H-2 House Phone

General Description

This feature allows selected Stations to reach the Attendant Console by going off-hook.

Operating Procedure

To reach the Attendant Console

1. The HOUSE PHONE user lifts the handset. The Attendant is called automatically.
2. The Attendant answers.
3. The Attendant connects the user to the desired Station or Trunk.

Service Conditions

1. Attendant Console indications will appear as follows:
   a.) ATND Lamps flash.
   b.) Console buzzer sounds.
2. HOUSE PHONES may be equipped with or without dials.
3. HOUSE PHONE assignments are programmed into System Data via the PCPro.
4. There is no limit to the number of HOUSE PHONES permitted in the system.

Interactions

1. AHLS is not needed. When a Station is programmed as TEC: 14, HOTLINE, it will automatically be connected to the Attendant Console.
2. If the HOUSE PHONE is to be used for Outgoing Calls via the Attendant Console (ARSC command RRI 2), assign a Route Restriction Class that will allow these connections. The same consideration should be made for Incoming Calls via the Attendant Console (RRI 0).

Programming

ASDT - Assign the Station as a TEC = 14 for HOTLINE.

H-4D Hands-free Answer Back -D

General Description

This feature allows the user to respond to a VOICE CALL [V-2] without lifting the handset.

Operating Procedure

To answer an Incoming Call

1. Lift the handset and press a Line key to answer the call.
2. Press the FEATURE key and “1”. The MIC LED lights.
3. Press the FEATURE key and “5” to select HANDS-FREE operation.

   Note: This step is not necessary for SIP Multiple Line terminals.

4. Press the SPEAKER key and replace the handset. Respond to the call HANDS-FREE.

To answer a VOICE CALL

1. Press the FEATURE key and “1”. The MIC LED lights.
2. Press the FEATURE key and “5” to select HANDS-FREE operation.
3. Respond to the call HANDS-FREE.

Service Conditions

1. The MIC may be left ON at all times.
2. HANDS-FREE operation may be selected at all times.
3. HANDS-FREE ANSWER BACK may be used when responding to a VOICE CALL [V-2] on My Line.

Programming

No programming is required. See VOICE CALL [V-2].

H-5D Hands-free Dialing/Monitoring -D

General Description

This feature allows the Dterm user to dial or monitor and call without lifting the handset.

Operating Procedure

To dial a call HANDS-FREE

1. Press the SPEAKER key. Receive Dial Tone.
   a.) Dial the desired number.
   b.) When the Called Party answers, lift the handset.

To monitor a call HANDS-FREE

1. When placed on hold, press the SPEAKER key.
2. Replace the handset. When the call resumes, pick up the handset.

Service Conditions

None.

Programming

No programming is required.

H-9 Hotline - Outside

General Description

This feature allows a user to access an outside destination by going off-hook or selecting the HOTLINE - OUTSIDE [H-1] Feature key and automatically dialing a SPEED CALLING - SYSTEM [S-3] Number.

Operating Procedure

To place a HOTLINE - OUTSIDE Call using a Single Line Telephone

1. User lifts the handset.
2. The System automatically selects a Trunk and dials the Outside Number.
3. The Called Number is answered and conversation proceeds.

To place a HOTLINE - OUTSIDE Call from a Dterm

1. The Dterm user presses the HOTLINE - OUTSIDE Feature key and lifts handset or presses the SPEAKER key.
2. The System automatically selects a Trunk and dials the Outside Number.
3. The Called Number is answered and the conversation proceeds.

Service Conditions

1. There is no limit to the number of HOTLINE Stations.
2. The Destination Parties for this service use Abbreviated Call Codes prepared in SPEED CALLING - SYSTEM [S-3]. Therefore, the number of Destination Stations is limited by the capacity of SPEED CALLING - SYSTEM [S-3].
3. When a user is confronted with a situation whereby all Trunks are busy, or all Senders are busy at the same time calling in this service, Reorder Tone is heard at the HOTLINE Stations.
4. Outside connection in BROKERAGE HOTLINE - OUTSIDE [B-12] Service can be used when the LCD indicates:
   a.) Trunk Classification such as DDD, TIE, and CCSA and Trunk Number for an Outgoing Call.
   b.) Time duration, Trunk Classification, and Trunk Number for an Incoming Call.
5. When a user of BROKERAGE - HOTLINE - OUTSIDE [B-12] is confronted with a situation in which all Trunks are busy or all Senders are busy, the user hears Reorder Tone. No indication is shown on the LCD of the user's Dterm.
6. STATION MESSAGE DETAIL RECORDING (SMDR) [S-10] is available for Outgoing Calls with the HOTLINE - OUTSIDE Service.
7. OUTGOING TRUNK QUEUING [O-2] is not provided for a HOTLINE - OUTSIDE Call.
8. The Prime Line of a Dterm cannot be assigned as a HOTLINE - OUTSIDE.
9. When a Dterm activates HOTLINE - OUTSIDE, the display will be for an Outgoing Trunk Connection.
10. This feature can be activated via FCCS.
11. Only the Station assigned Telephone Number is available for the Station assignment when activating this feature via FCCS.
12. The Station, the Attendant Console/Desk Console (Operator Call), and the Trunk can be a destination for Hotline.

Interactions

SPEED CALLING - SYSTEM [S-3] must first be provided.

Programming

STEP 1: **ASYD** - SYS2, Index 1, Bit 7. Does TOLL RESTRICTION apply to SPEED CALLING - SYSTEM [S-3]? 0/1: Yes/No. If TOLL RESTRICTION applies, construct an RSC that will allow the Station access to the Outgoing Number.

STEP 2: **ASFC** - Construct a Service Feature Restriction Class that allows SFI 12 (Speed Calling - System). Assign this SFC to the Hotline Station.

If using a Virtual Circuit (TEC = 18) for the Hotline, SFI 36 (Special Common Battery/Hotline) must be set to "1" for the Virtual Circuit's SFC.

STEP 3: **ASDT** - Assign the HOTLINE - OUTSIDE Station as a TEC = 18, Virtual Circuit or TEC = 14, Hotline.

TN: Tenant Number
STN: Station Number
LENS: Line Equipment Numbers (six digits)
TEC: Telephone Class (1-31)

- 1: DP (10 PPS)
- 2: PB
- 3: DP/PB
- 12: Dterm
- 13: Data Terminal via Dterm
- → 14: HOTLINE
- 15: CAS Line
- 18: Virtual Circuit
- 23: ISDN Terminal
- 27: Eight Conference Equipment

RSC: Route Restriction Class (0-15)

For assignment of RSC, ARSC.

SFC: Service Feature Restriction Class (0-15)

For assignment of SFC, ASFC.

STEP 4: **ASPD** - Enter Tenant Number (TN), Abbreviated Digit Code (ADC), and the Called Number (CD) to be sent. Include in the CD the Access Code of the Route.

STEP 5: **AHLS** - Assign the HOTLINE - OUTSIDE Station with the following parameters:

HOT TN: HOTLINE - OUTSIDE Tenant Number
HOT STN: HOTLINE - OUTSIDE Station Number
TYPE: 2 (Calling Party is an Outside Party using Abbreviated Digit Code (ADC))
CON TN: Connecting Stations Tenant. Assign same TN Number as in HOT TN:
CON ADC: Speed Calling Code of corresponding Outside Number.

FCCS Programming

This data setting describes only the additional or replacement commands used to activate FCCS Services. When used as an FCCS feature, these commands replace the non-network commands (for example, AAED is replaced by AAEDN). Note that the data setting for FCCS Service varies depending on the Node.

STEP 1: **AHLSN** - This data must be assigned to the NCN. Assign the HOTLINE - OUTSIDE Station with the following parameters:

HOT UGN: HOTLINE User Group Number of the Calling Party
HOT TELN: HOTLINE Telephone Number of the Calling Party
TYPE: 2 (Calling Party is an Outside Party using Abbreviated Digit Code (ADC))
3 (Called Party is a Trunk)
CON UGN: User Group Number of the Called Party (for TYPE = 2)
CON ADC: The Dialed Number for the Called Party (for TYPE = 2)
CON LRN: Logical Route Number of the Called Party (for TYPE = 3)
CON TK: Trunk Number of the Called Party (for TYPE = 3)

Note: When using ASYDL or ASYDN to program an FCCS feature, the basic programming of ASYD is required in addition to ASYDL or ASYDN.

Note: The following data should also be set in addition to the above-mentioned commands. The commands to be set to the Node accommodating the Calling Party: ASYD/ ASFC/ASDT/ASPD

I-1 Immediate Ringing

General Description

This feature causes Called Stations to ring immediately upon establishment of connections.

Operating Procedure

No manual operation is required.

Service Conditions

1. Ringing is applied to a Called Station as soon as the switch has determined that the Station is idle. There is no delay caused by waiting for a ring cycle.
2. This feature applies to STATION-TO-STATION [S-11], Attendant (originated or completed), Direct Inward Dialing (DID), and TIE Line Calls.
3. An idle Station rings immediately after the last digit of the Called Station is dialed.

Programming

ASYD - SYS1, Index 27, Bit 7. Assign data “1”.

I-2 Incoming Call Identification

General Description

This feature allows an ATTENDANT CONSOLE [A-3] to visually identify the type of service and/or Trunk Group that is arriving or waiting to be answered.

Operating Procedure

No manual operation is required.

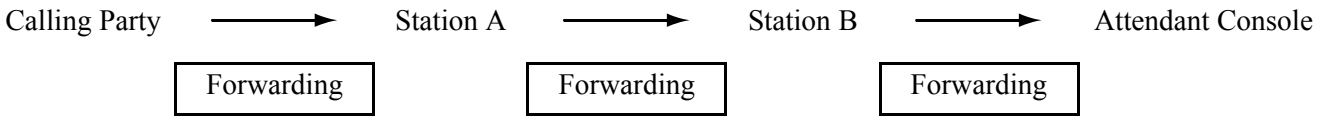
Service Conditions

1. A flashing lamp indicates a Call Waiting to be answered.
2. A steady lamp indicates an answered call.
3. A limit of 12 lamp and key groups can be associated with INCOMING CALL IDENTIFICATION.
4. Lamps and keys may be assigned as follows:

LDN:	Listed Directory Number (Central Office)
WATS:	Wide Area Telephone Service
FX:	Foreign Exchange
ATND:	Attendant Call
RECALL:	Attendant Recall Calls
ICPT:	Call Forwarding - Intercept
NANS:	Call Forwarding - Don’t Answer
BUSY:	Call Forwarding - Busy Line
TIE:	TIE Line
TF:	Inter-Position Transfer
CAS:	Centralized Attendant Access
PR1:	Priority Call 1
PR2:	Priority Call 2
PR3:	Priority Call 3
SC/RC:	Serial Call Recall key
EMG:	Off-Hook Alarm

5. More than one INCOMING CALL IDENTIFICATION Lamp can be duplicated at the ATTENDANT CONSOLE [A-3]. The Attendant can select any Incoming Call by pressing the associated key.
6. Calls can be answered via the ANSWER key for first-in, first-out priority. (CALL QUEUING [C-9]).
7. ATTENDANT CONSOLEs [A-3] are limited to 12 unique INCOMING CALL IDENTIFICATION indications. However, more than one Incoming Trunk Group can be programmed for the same INCOMING CALL IDENTIFICATION.

8. When a call is terminated to the Attendant Console in consequence of Multi-hops for Call Forwarding - All Calls/Busy Line/Don't Answer, the lamp indication on the Attendant Console will be as follows:



		STATION A		
		CALL FORWARDING - ALL CALLS	CALL FORWARDING - BUSY LINE	CALL FORWARDING - DON'T ANSWER
STATION B	Call Forwarding - All Calls	Note 1	Note 1	NANS
	Call Forwarding - Busy Line	BUSY	BUSY	NANS
	Call Forwarding - Don't Answer	Note 1	BUSY	NANS

Note 1: The lamp indication varies depending on the Calling Party.

Interactions

1. Step 1 is not required to fulfill the requirements of the feature description. However, the CALL WAITING LAMP [C-13] should be programmed in relationship to the INCOMING CALL IDENTIFICATION.
2. SYS2, Index 8, Bit 6 and Bit 7, and Index 9, Bit 0 to Bit 3 can be flagged to represent an Incoming Call, but the meaning of the Incoming Call can be reassigned via AAKP.
3. The first six keys (LDN to CCSA) are fixed and cannot be changed. AAKP is used to program the remaining six optional keys.

Programming

STEP 1: **ASYD** - SYS2, Index 8 and Index 9, flag the type of Incoming Calls the Attendant Console should expect. **Note 2**

Index 8
Bit 0: Listed Directory Number
Bit 1: Incoming Station Call
Bit 2: Attendant Recall
Bit 3: FX
Bit 4: WATS
Bit 5: CCSA (DID)
Bit 6: TIE Line
Bit 7: Call Forwarding - Busy Line

Index 9
Bit 0: Call Forwarding - Don't Answer
Bit 1: Call Forwarding - Intercept

Bit 2: Special Common Battery Station
Bit 3: Interposition Transfer

Note 2: SYS2, Index 8, Bit 6 and Bit 7, and SYS2, Index 9, Bit 0 to Bit 3 can be assigned different meanings via AAKP.

STEP 2: **ARTD** - Flag the type of Route in CDN 6:TCL, Trunk Class.

0: Not used
1: DDD Line (LDN)
2: FX Line (FX)
3: WATS Line (WATS)
4: TIE Line (TIE)
5: CCSA Line (CCSA)
6: Toll Line
7: CAS Line
8: Paging
9: Not used
10: Not used
11: General Page
12: Radio Page

Note: To change the data for existing route (RT), initialize the circuit card.

STEP 3: **AAKP** - This command may be used to change the meaning of the top six keys of the Call Identification keys of the Attendant Console. (See SYS2, Index 8 and Index 9.)

The following functions may be assigned to Key Numbers (KYN) 1-6.

Function:

a: CAS
b: Off-Hook Alarm
c: Priority Call 1
d: Priority Call 2
e: Priority Call 3

I-3 Incoming Central Office Call to TIE-Line Connection

General Description

This feature permits an Attendant to connect an Incoming Public Network Exchange Call, via a TIE Line, to a Station at a distant office.

Operating Procedure

To connect an Incoming Call

1. The Attendant presses the LDN key and answers the Incoming Trunk Call.
2. Dial the TIE Line Access Code. Receive Dial Tone from distant office.
3. Dial the distant office Station Number.
4. Press the RELEASE key.
5. The TIE Line and Central Office Call are connected.

Service Conditions

1. The Attendant can connect a Trunk and TIE Line Call either before or after the distant Station answers.
2. There is no limitation on the number of Incoming Trunks that may be connected to TIE Lines using this feature.
3. TRUNK-TO-TRUNK CONNECTION [T-10] may significantly degrade transmission quality.
4. TRUNK-TO-TRUNK CONNECTION [T-10] can be restricted through alternative Route Restriction Assignments.
5. AUTOMATIC RECALL [A-8] is denied on Incoming Central Office Call to TIE Line Connection.
6. Since AUTOMATIC RECALL [A-8] cannot be initiated by a Trunk, the ATTENDANT LOCKOUT [A-5] feature is automatically disabled in order to allow the Attendant to reenter TRUNK-TO-TRUNK CONNECTION [T-10] Calls held on the ATTENDANT CONSOLE [A-3].
7. This feature is applicable to the following types of Incoming Central Office Calls:
    - FX
    - WATS
    - CCSA
    - DID (Direct Inward Dialing)
    - COT (Ringdown/Loop Start)
8. Loop Start Central Office Trunks can be connected to TIE Trunks since Release Supervision is provided by the TIE Trunks.
9. When Release Signal is not provided by TIE Line Trunk, the Attendant must monitor the connection and manually disconnect after use.

Interactions

1. The Attendant Console must first be programmed.
2. For TIE Line Route to Central Office Route Connection, the reverse must be programmed. However, program for ARI-D-RES:, Direct Connection. This connection may also be Toll Restricted by using ATDP.

3. A Station can transfer a call to a Station or to the Attendant Console at a distant office via a TIE Line Connection. This Station must be assigned a correct Class of service.
4. Stations must be allowed access to the Routes they will need via the appropriate Connection Status, either via the Attendant Console (RRI 2) or Direct Connection (RRI 3).
5. Various combinations of TRUNK-TO-TRUNK [T-10] Connections can be restricted by Alternate Route Restriction (ARRC).

Programming

STEP 1: **ARTD** - Assign the Central Office Route as shown below:

GROUND START:

RT: 1

1-OSGS	:2	2-ONSG	:3	3-ISGS	:1	4-INSG	:3	5-TF	:3
6-TCL	:1	7-L/T	:1	8-RLP	:2	15-LSG	:1		

LOOP START:

RT: 1

1-OSGS	:2	2-ONSG	:3	3-ISGS	:1	4-INSG	:3	5-TF	:3
6-TCL	:1	7-L/T	:1	8-RLP	:2				

Assign the TIE Line Route as shown below:

2 OR 4 WIRE E & M:

RT: 1

1-OSGS	:2	2-ONSG	:3	3-ISGS	:2	4-INSG	:3	5-TF	:3
6-TCL	:4	7-L/T	:1	8-RLP	:2	15-LSG	:5		

Note: To change the data for existing route (RT), initialize or reboot the trunk device such as a circuit card or MG.

STEP 2: **ATRK** - Assign the LENS, Central Office Route Number, Trunk Number, and Tenant Number.

Assign the LENS, TIE Line Route Number, Trunk Number, and Tenant Number.

STEP 3: **MBTK** - Assign the Make Idle status to the Trunks.

For TIE Line:

STEP 4: **ANPD** - Reserve a Number Level for Trunk Access. Select Normal Service (N) or Hooking Service (H) from the combo box for Connection Indexes. Number of Necessary Digits is usually one or two. Busy Lamp Field is not activated.

STEP 5: **ASPA** - Assign the Access Code as assigned in ANPD above. Type of Service, SRV = "OGC: Outgoing Call" for Outgoing Trunk. Assign the Route Number associated with this Access Code.

STEP 6: **ARSC** - For the Incoming Central Office Route, allow for RRI: 0, Incoming Call via the Attendant Console (with Call Transfer). This Route Restriction Class (RSC) is to be assigned for the Attendant Console in ATRK. (RSC 0).

For Outgoing TIE Line Route and for RSC 0, allow for RRI: 2 Outgoing Call via the ATTCON (with Call Transfer). RSC 0 is to be assigned to the ATTCON in ATRK.

STEP 7: **ARRC** - Assign 1 to the parameter ARI-D-RES for allowing Tandem Connection with an Operator assistance. Allow the Incoming Route Number (ICRT) of the Incoming Central Office Route to be connected to the Outgoing Route Number (OGRT) of the Outgoing TIE Line Route.

FCCS Programming

This data setting describes only the additional or replacement commands used to activate FCCS Services. When used as an FCCS feature, these commands replace the non-network commands (for example, AAED is replaced by AAEDN). Note that the data setting for FCCS Service varies depending on the Node.

STEP 1: **ALRTN** - Assign this command to connect the Logical Route to the Physical Route.

STEP 2: **ARTDN** - Assign the Central Office Route as shown below:

GROUND START:

LGRT: 1

1-OSGS	:2	2-ONSG	:3	3-ISGS	:1	4-INSG	:3	5-TF	:3
6-TCL	:1	7-L/T	:1	8-RLP	:2	15-LSG	:1		

LOOP START:

LGRT: 1

1-OSGS	:2	2-ONSG	:3	3-ISGS	:1	4-INSG	:3	5-TF	:3
6-TCL	:1	7-L/T	:1	8-RLP	:2				

Assign the TIE Line Route as shown below:

2 OR 4 WIRE E & M:

LGRT: 1

1-OSGS	:2	2-ONSG	:3	3-ISGS	:2	4-INSG	:3	5-TF	:3
6-TCL	:4	7-L/T	:1	8-RLP	:2	15-LSG	:5		

Note: To change the data for existing route (RT), initialize or reboot the trunk device such as a circuit card or MG.

STEP 3: **ANPDL** - Reserve a Number Level for Trunk Access. Select Normal Service (N) or Hooking Service (H) from the combo box for Connection Indexes. Number of Necessary Digits is usually one or two. Busy Lamp Field is not activated.

STEP 4: **ASPAL** - Assign the Access Code as assigned in ANPD above. Type of Service, SRV = "OGC: Outgoing Call" for Outgoing Trunk. Assign the Logical Route Number associated with this Access Code.

STEP 5: **ARSCN** - For the Incoming Central Office Route, allow for RRI: 0, Incoming Call via the Attendant Console (with Call Transfer). This Route Restriction Class (RSC) is to be assigned for the Attendant Console in ATRK. (RSC 0).

For Outgoing TIE Line route and for RSC 0, allow for RRI: 2 Outgoing Call via the ATTCON (with Call Transfer). RSC 0 is to be assigned to the ATTCON in ATRK.

STEP 6: **ARRCN** - Assign 1 to the parameter ARI-D-RES for allowing Tandem Connection with an Operator assistance. Allow the Incoming Route Number (ICRT) of the Incoming Central Office Route to be connected to the Outgoing Route Number (OGRT) of the Outgoing TIE Line Route.

I-4 Individual Trunk Access

General Description

Individual Trunk Access allows Station/Attendant Console users to seize specific Trunk by dialing. By using the feature, the user can perform the Connection Tests. For details, refer to table below:

KIND OF TEST	ORIGINATING TERMINAL (Allowed: × Not allowed: —)			
	STATION	PS TERMINAL	ATT CON	ISDN TERMINAL
Resister Test	Allowed	Not Allowed	Not Allowed	Not Allowed
Sender Test	Allowed	Not Allowed	Not Allowed	Not Allowed
Three-party Conference Trunk Test	Allowed	Not Allowed	Not Allowed	Not Allowed
Tone Test	Allowed	Not Allowed	Not Allowed	Not Allowed
Interrupted Ringing (IR) Test	Allowed	Not Allowed	Not Allowed	Not Allowed
External Trunk (Central Office/TIE/MFC/ CCIS) Test	Allowed **Note 1**	Not Allowed	Allowed **Note 1**	Not Allowed

Note 1: A Specific ISDN Trunk can be seized from the Station and Attendant Console.

Individual Trunk Access (PCPro)

By using TOGC, the user can perform the tests even involving the Tandem Connection. For more details refer to the Operations and Maintenance Manual.

Operating Procedure

Operating Procedure for Individual Access from Attendant Console

By using a feature key (TKSL)

1. Press the TKSL key.
2. Dial the route number + trunk number (for example, 002+005), or dial the C.O line number (for example, 3001).
3. Dial the called party number and receive Ring Back Tone (RBT).
4. Press the CANCEL key after the call is finished.

By using the access code

1. Press a Loop key.
2. Dial the access code.
3. Dial the route number + trunk number (for example, 002+005), or dial the C.O line number (for example, 3001).
4. Dial the called party number and receive Ring Back Tone (RBT).
5. Press CANCEL key after the call is finished.

Operating Procedure for Individual Access from Station

By using the access code

1. Pick up the handset (or perform an equivalent operation for the specific type of station you are using such as pressing the Speaker button, etc.)
2. Dial the access code.
3. Dial the route number + trunk number (for example, 002+005), or dial the C.O line number (for example, 3001).
4. Dial the called party number and receive Ring Back Tone (RBT).
5. Press CANCEL key after the call is finished.

Service Conditions

1. A designated connection to a Trunk is limited to CCIS, TIE, Central Office, MFC and ISDN Line of either second DT System or Sender System. Connection Tests are not available for a Q-SIG Line.
2. This feature is provided when combination of Telephone Equipment Class (TEC) for Outgoing Terminal and ONSG (assigned by ARTD) for Outgoing Trunk is suitable. Details on the combinations are as follows:

OUTGOING TRUNK SIDE / OUTGOING TERMINAL SIDE	CENTRAL OFFICE/TIE			MFC	CCIS/ISDN
	PB (ONSG = 2, 8)	DP (ONSG = 1, 5, 7, 9)	DP/PB (ONSG = 3)	MFC (ONSG = 10)	PB (ONSG = 2)
PB (TEC = 2, 12)	Provided	Not Provided	Provided	Provided	Provided
DP (TEC = 1, 4)	Not Provided	Provided	Provided	Provided	Provided
DP/PB (TEC = 3)	Not Provided	Not Provided	Provided	Provided	Provided

3. In this feature, confirmation of an Answer Back Signal from other Office is not performed.
4. A digital display will appear on the ATTENDANT CONSOLE [A-3], which will consist of either:
   a.) TENANT SERVICE [T-12] Number, Route Number, and Trunk Number.
   b.) Central Office Code Number. This data is programmed on a per Route basis.
5. If the Trunk is busy, the Attendant will receive a Busy Tone.
6. The Attendant can use ATTENDANT OVERRIDE [A-7], if it is included in System Programming, to gain entry into a particular Trunk when it is in use.
7. If the Trunk is made busy, either by hardware or software control, the Attendant can still access that Trunk.
8. Central Office Code (COC) is not available to access an individual Trunk through FCCS.
9. This feature is not available for the following connections:
   - CCIS over IP (Point-to-Multipoint) connections by IPTRK card
   - CCIS over IP connections by PHC
10. This feature is not applicable to the billing.

11. Trunk Access to IPPAD is available. The conditions are below:
    - Both of Single Line Telephone/Digital terminals, and IP Station can access to Station/Trunk.
    - Access to IPPAD is available in released and soft make-busy status.
    - For IPPAD channel assigned while executing TOGC command, MB should not be canceled by MBCT command.
    - The combination with other Trunk Access features (CCIS, TIE, Central Office, MFC and ISDN Line) are not available.
    - Echo Canceler training is also valid toward IPPAD which is executed Trunk Access.
    - Trunk Access to IPPAD has priority over the function for IP terminals to seize the IPPAD within the same network preferentially (ASYDL, SYS1, Index 810, Bit 4).

Interactions

If the Trunk is busy, ATTENDANT OVERRIDE [A-7] may be used. THREE-WAY CALLING [T-2] is established following a Warning Tone. If the Station should disconnect, the Attendant Console can either disconnect or extend the Trunk to another Station via the keypad by pressing the START key and entering the desired Telephone Number.

Programming

Data assignment for individual access from Station

STEP 1: **ANPD** -Assign first digit of an Access Code for Individual Trunk Access and Number of Digit Number of the Access Code.

TN:	Tenant Number
1stDC:	1st Digit Code (0-9,*, #)
CI:	Select Normal Service (N) from the combo box for Connection Indexes.
NND:	Number of Necessary Digits

STEP 2: **ASPA** -Assign the Access Code for Individual Trunk Access

ACC:	Access Code
CI:	Connection Status Index = “N: Normal”
SRV:	Kind of Service = “SSC: Service Code”
SID:	Service Index = 17 (Individual Trunk Access)

STEP 3: **ARNP** -Assign the reverse number to the Route when Answer Back Signal method is Second Dial Tone (mark-out).

RT:	Route Number
ACC:	Access Code

STEP 4: **ASFC** -Allow the Service Feature Restriction Class for Individual Trunk Access.

SFC:	Service Feature Restriction Class (1-15)
SFI:	39 (Individual Trunk Access) = 1 (Allowed)

Data assignment for individual access from Attendant Console

STEP 1: **ASYD** -SYS1, Index 68

Index 68, Bit 1=Designate the method that the Attendant Console must use to operate Individual Trunk Access.
0/1 = External Route and Trunk Number/Central Office Code (assigned by ACOC)

Index 68, Bit 4=Is Trunk Select key in service? 0/1 = No/Yes Assign "1".

STEP 2: **ANPD** -Assign first digit of an Access Code for Individual Trunk Access and Number of Digit Number of the Access Code.

TN:	Tenant Number
1stDC:	1st Digit Code (0-9,*, #)
CI:	Select Normal Service (N) from the combo box for Connection Indexes.
NND:	Number of Necessary Digits

STEP 3: **ASPA** -Assign the Access Code for Individual Trunk Access

ACC:	Access Code
CI:	Connection Status Index = "N: Normal"
SRV:	Kind of Service = "SSC: Service Code"
SID:	Service Index = 17 (Individual Trunk Access)

STEP 4: **ARTD** -When the Central Office Code is used, assign CDN 25 = 1.

Note: To change the data for existing route (RT), initialize the circuit card.

STEP 5: **ACOC** -When the Central Office Code is used, Trunks in the system must be programmed to have an associated Central Office Code. This code is dialed after pressing the Attendant Console's Trunk Select key.

Note: Central Office Code does not apply to Stations. Stations permitted this feature must dial an Access Code, the Trunk Route Number and the Trunk Number.

I-5 Inter Position Transfer

General Description

This feature allows Attendants to transfer calls at their ATTENDANT CONSOLE [A-3] to another Attendant's Console in systems where MULTIPLE CONSOLE OPERATION [M-4] has been provided.

Operating Procedure

To transfer a call from Attendant Console A to Attendant Console B

1. Press the START key.
2. Attendant A dials Attendant Console B Identification Number, or "0" for all Attendants.
3. The call is indicated at Attendant Console B or all Attendant Consoles.
4. Attendant B presses the ANSWER key or TF ICI key.
5. Attendant A presses the RELEASE key to transfer, or may consult first, then release.

Service Conditions

1. A designated Identification Number is assigned to each ATTENDANT CONSOLE [A-3] for the purpose of INDIVIDUAL ATTENDANT ACCESS [I-6].
2. This feature can be provided on a TENANT SERVICE [T-12] basis.
3. An Attendant can receive one INTER-POSITION TRANSFER Call at a time. An INCOMING CALL IDENTIFICATION (TF ICI) [I-2] key can be designated for receipt of these calls.
4. An Attendant cannot transfer a Conference Connection to another Attendant.
5. An Attendant can extend an Inter Position Transferred Call to any other destination.
6. NIGHT CONNECTION - FIXED/FLEXIBLE [N-1, 2]: When Night Service is active, an Attendant Position Call to another Attendant cannot be answered by the Night Service arrangements. The Calling Attendant hears Reorder Tone.
7. When the Operator Call Code is dialed, the call is routed to the predetermined Attendant Console designated in System Data for the FCCS Network.
8. When assigning Function Code (FUNC) to KYN 1 of an Attendant Console by AAKP, this feature cannot be used.

Interactions

The ASAT Number may not be programmed in place of a Telephone Number in ASID and ASGD. See OFF-HOOK ALARM [O-6], ATTENDANT NIGHT TRANSFER [A-19], and PRIORITY CALL [P-9].

Programming

STEP 1: **ASYD** - SYS2, Index 6, Bit 5. Enable INTER-POSITION TRANSFER? 0/1: No/Yes. Assign data 1. Assigned on a per Tenant basis.

SYS2, Index 9, Bit 3. Is INTERPOSITION TRANSFER considered an Incoming Call to the Attendant Console and to be counted as a Call Waiting? 0/1: No/Yes. See CALL WAITING LAMP [C-13]. Assigned on a per Tenant basis.

STEP 2: **ASAT** - Assign a specific Attendant Number to each Attendant Console.

STEP 3: **ATNR** - Allow Station-to-Station Connection on an Intra and Inter-Tenant basis.

FCCS Programming

This data setting describes only the additional or replacement commands used to activate FCCS Services. When used as an FCCS feature, these commands replace the non-network commands (for example, AAED is replaced by AAEDN). Note that the data setting for FCCS Service varies depending on the Node.

Note: When using ASYDL or ASYDN to program an FCCS feature, the basic programming of ASYD is required in addition to ASYDL or ASYDN.

ASPAL - This data must be set to the Node calling Attendant, Assign SRV = “SSC: Service Code”, SID = 2 (Operator Call), and FPC of the Node that Operator Calls are to be terminated to.

I-6 Individual Attendant Access

General Description

This feature permits a user to call a particular ATTENDANT CONSOLE [A-3] via an Individual Attendant Identification Number.

Operating Procedure

1. The user dials the Individual Attendant Identification Number.
2. The directed Attendant gets INCOMING CALL IDENTIFICATION (ICI) [I-2] indication ("ATND") and can answer using the normal operating procedure.

Service Conditions

1. Attendant Identification Numbers (maximum of five digits) must be programmed and published for each Attendant Console.
2. The ASAT Number should be the same number of digits as any other Station Number. ASAT assigns a Station Number to each Attendant Console.
3. ATTENDANT NIGHT TRANSFER [A-19] feature cannot be activated for calls in this feature.
4. The Individual Attendant Identification Number cannot be used to assign the destination of Call Forwarding Service.
5. Only a Telephone Number can be assigned as the Individual Attendant Identification Number when this feature is activated via FCCS.

Programming

STEP 1: **ASAT** - Assign a Station Number to each Attendant Console. The Attendant Console Number is the Trunk Number assigned in ATRK, Route 901.

STEP 2: **ATNR** - Allow Tenant Stations either access or denial to the Tenant in which the Attendant Console is located.

FCCS Programming

This data setting describes only the additional or replacement commands used to activate FCCS Services. When used as an FCCS feature, these commands replace the non-network commands (for example, AAED is replaced by AAEDN). Note that the data setting for FCCS Service varies depending on the Node.

STEP 1: **ALGNN** - This command must be assigned to the NCN. Assign the User Group Number and Telephone Number for the Individual Attendant Identification Number.

STEP 2: **ASATN** - This command must be assigned to the NCN. Assign User Group Number (UGN), Telephone Number assigned by ALGNN for specifying the Attendant Console (ACC), Fusion Point Code accommodating the designated Attendant Console, Tenant Number to which the Attendant Console belongs (TN), and the Trunk Number assigned by ATRK for the Attendant Console (ATN).

Note: Assign the ATNR command data to both the Nodes accommodating the Station and the Attendant Console.

I-7D I-Hold Indication -D

General Description

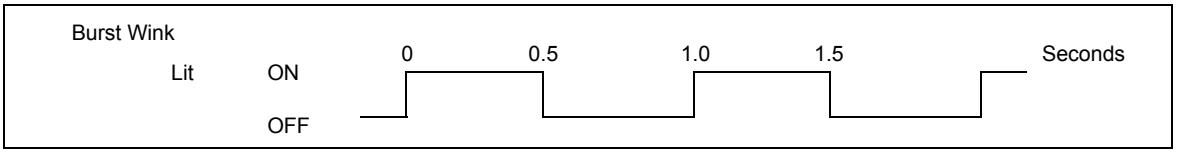
This feature provides the Desktop terminal with a distinctive flash to differentiate between a call the user placed on hold from other calls.

Operating Procedure

No manual operation is required.

Service Conditions

Condition of the LED is as follows:



Programming

No programming is required.

I-8D I-Use Indication -D

General Description

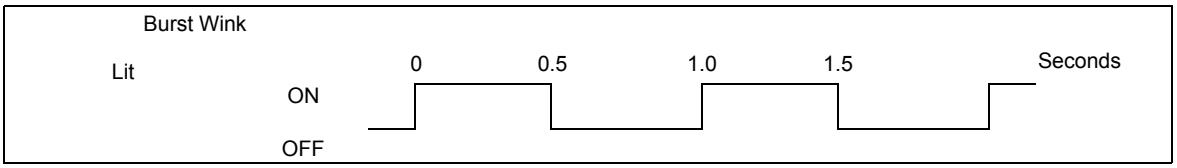
This feature provides the Desktop terminal with a unique LED indication to display the particular line the user is on.

Operating Procedure

Manual operation is not required.

Service Conditions

Condition of the LED is as follows:



Programming

No programming is required.

I-9 Intercom Calling

General Description

This feature allows a Sub Line Appearance, assigned to a Desktop terminal, to be used for INTERCOM CALLING.

Note: This feature is only available in North America.

Operating Procedure

To originate

1. Press the Intercom Sub Line Appearance.
2. Lift the handset or press the SPEAKER key. Dial Tone is heard.
3. Dial an Intercom Station, receive Ringback Tone. The display will show:

ICM	XXXX
(Time Display)	

4. The Called Party answers and the conversation begins.

To answer

1. Ringing is heard from the Intercom Sub Line.
2. Press the Intercom Sub Line Appearance. The display will show:

ICM	XXXX
(Time Display)	

3. Lift the handset or press the SPEAKER key.
4. The conversation can then begin.

Service Conditions

1. An Intercom Sub Line Appearance cannot be the Prime Line of a Desktop terminal.
2. An Intercom Sub Line Appearance cannot call any Station or Trunk in a system other than another Intercom Sub Line Appearance.
3. An Intercom Sub Line Appearance cannot activate any features.
4. An Intercom Sub Line Appearance must be assigned as a Software Line Appearance.
5. This feature is only available for Desktop terminals.
6. Intercom Sub Lines must be allowed Service Feature Restriction of SFI 70 (Intercom Group).
7. FCCS is not available for this feature.

Interactions

1. As an example for Group Calling application, the following conditions exist:
   System Station Numbering
   5XXXX and 6XXX
   Speed Calling Access:
   3 (Assigned in ASPA)
   100 Groups of Ten Stations

Group Calling Station
Numbering Consists of the following:
30-39
Stations in Group Calling would dial only two digits to reach other members of the group.

2. All My Line STATION-TO-STATION CALLING [S-11] capabilities such as VOICE CALL [V-2], MESSAGE REMINDER [M-19], CALL BACK [C-1], etc. are denied because INTERCOM CALLING uses a Virtual Circuit Sub Line Appearance.
3. An Incoming Intercom Call will be provided with distinctive three-burst ringing.
4. An Intercom Station is only allowed to call other Intercom Stations. If a Non-Intercom Station attempts a call, Reorder Tone is heard, and the LCD displays:

REST **(Time Display)**

5. An Intercom Station cannot call the Attendant Console.

Programming

STEP 1: **ASFC** - Construct a Service Feature Restriction Class that allows SFI 19 [Individual Speed Calling (Station/Group)] and SFI 70 (Intercom Group). Assign this SFC to the Stations that act as INTERCOM CALLING Extension Numbers. SFI 70 (Intercom Group) is only used on Virtual Lines in their own SFC.

STEP 2: **ASDT** - Program the Stations within a group as a virtual line (TEC = 18). Determine LENS from available time slots in the system.

TN: Tenant Number
STN: Station Number
LENS: Line Equipment Numbers (six digits)
TEC: Telephone Class (1-31)
- 1: DP (10 PPS)
- 2: PB
- 3: DP/PB
- 12: Desktop terminal
- 13: Data Terminal Via Desktop terminal
- 14: Hotline
- 15: CAS Line
- 16: Data Terminal via Data Module
- → 18: Virtual Circuit
- 23: ISDN Terminal
- 27: Eight Conference Equipment

RSC: Route Restriction Class (0-15)
For assignment of RSC, ARSC.
SFC: Service Feature Restriction Class (0-15)
For assignment of SFC, ASFC.

STEP 3: **AKYD** - Assign the Virtual Circuit to Desktop terminal key. Assign FYI = 2, Multi Line key, and assign the Station Number of the Virtual Circuit. This key will be the GROUP CALLING key.

I-10D Intermediate Station Number Display -D

General Description

This feature displays on the LCD screens of calling and called parties the number of an Intermediate station, if any.

Note: "Intermediate station" mentioned here refers to a station through which a call is routed.

Operating Procedure

No manual operation is required.

Service Conditions

1. For a Transferred Call, the LCD displays:

Call Status (**Note 1**)	Intermediate Station	Calling/Called Station
TRANSFER	**XXXX (Time Display)**	**XXXX**

Note: When the Telephone Number of the Intermediate or Calling/Called Station exceeds eight digits, last eight digits are displayed. However, a maximum of 16-digit Calling/Called Telephone Numbers can be displayed on the middle line of the LCD depending on the System Data.

Note 1: Call status indication is provided by SERVICE DISPLAY -D [S-25D].

2. The LCD displays one of the following features being used:

Call Status	Feature
TRANSFER	Blind Transfer To Attendant [B-17]
TRANSFER	Blind Transfer To Station [B-18]
TRANSFER	Call Transfer - All Calls [C-11]
PICKUP	Call Pickup - Group [C-7]
CF ALL	Call Forwarding - All Calls [C-5]
CF BUSY	Call Forwarding - Busy Line [C-2]
CF NANS	Call Forwarding - Don't Answer [C-3]
CF-OUT	Call Forwarding - PS Incoming Incomplete [C-155]
CF LOUT	Call Forwarding - Logout
HUNTING	Station Hunting - Circular [S-7]

Call Status	Feature
HUNTING	Station Hunting - Secretarial [S-8]
HUNTING	Station Hunting - Terminal [S-9]

3. The INTERMEDIATE STATION NUMBER DISPLAY will not be displayed if a call is announced in CALL TRANSFER - ALL CALLS [C-11].
4. For the FCCS Network, when a Telephone Number has been assigned for a Station, it is displayed on the Dterm LCD. However, if a Telephone Number is seven digits or more, the Physical Station Number is displayed.

Programming

ASYD - SYS1, Index 78, Bit 0 and Bit 1. Assign both as data “1” to enable all displays on the 16-button Dterm instrument.

I-11 Inter-Office Off-Hook Queuing

General Description

This feature can be employed in a Main-Satellite configuration to allow a Satellite user to queue for LEAST COST ROUTING - 3/6-DIGIT [L-5] at the Main Location. This feature can also be used when all Outgoing Facilities are concentrated at the Main Location.

Operating Procedure

Operation at the Main Location

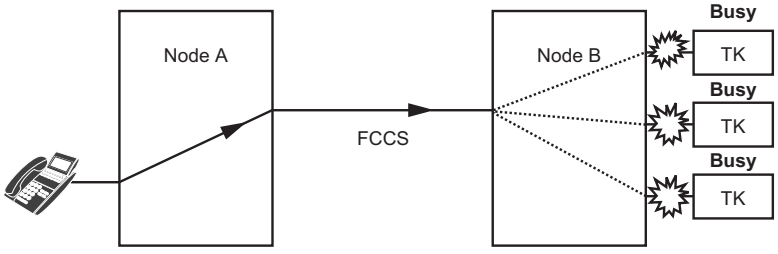
1. The Main Location receives dialed digits from the Satellite Location via a TANDEM TIE TRUNK [T-1].
2. The Main Location determines that no Trunks are available and places the call in an available hold on queue slot.
3. The Main Location returns Service Set Tone to the Satellite Caller.
4. The Satellite Caller remains off-hook, and waits for the call to be completed.
5. If there are no hold on queue slots available, the Main Location will return Reorder Tone to the Satellite Caller.
6. If the Satellite Caller goes on-hook (TANDEM TIE TRUNK [T-1] releases) while in queue, the assigned hold on queue slot is cleared.

Service Conditions

1. This feature is usually used in conjunction with LEAST COST ROUTING - 3/6-DIGIT [L-5].
2. The maximum queuing slots per system are: 64 calls for 1-IMG system
   256 calls for 4-IMG system
   The queue slots are commonly used by OFF-HOOK QUEUING [O-7], INTER-OFFICE OFF-HOOK QUEUING [I-11], and OUTGOING TRUNK QUEUING [O-2, 2D, 16].
3. If the Incoming TIE Trunk is a Second Dial Tone Signal Trunk, the caller will hear Service Set Tone while waiting in queue for a maximum of three minutes. After this period, the Trunk will automatically be released.
4. When a Trunk becomes available, the Dialed Number is automatically sent to the connecting Central Office.
5. If all OFF-HOOK QUEUING [O-7] slots are in use, Reorder Tone (fast Busy Tone) will be sent to the Satellite Office.
6. This feature can also be activated when using a Trunk in the Remote Node via FCCS.

Example: Node A and Node B are in an FCCS Network.

- Although a Station belonging to Node A tried to call outside using a Trunk belonging to Node B, all the Trunks are busy. The call can be in the queue using this feature.



7. OGQ Data is transmitted to and stored in the Network Control Node (NCN). Therefore, all the OGQ Data will be cleared by the System Initialization of the NCN.
8. When the access to the NCN is impossible (system down, cable disconnection or trouble has occurred), the user cannot set or cancel OGQ Data. After dialing the Access Code or pressing the Feature keys for Set/Cancel OGQ, the user hears the Reorder Tone.
9. When the Trunk becomes idle status after the NCN is recovered, and the call in the queue will be called back, if data has not been cleared. If data has been cleared, the call is canceled automatically.
10. A maximum of 1024 Stations per system may use OFF-HOOK QUEUING [O-7] and OUTGOING TRUNK QUEUING [O-2] simultaneously in an FCCS Network.
11. The Class Of Service for OFF-HOOK QUEUING [O-7] is different from the Class Of Service for OUTGOING TRUNK QUEUING [O-2]. Both services are mutually exclusive.
12. This feature does not work for the tandem connection when the incoming call terminates by DIRECT-IN TERMINATION (DIT) [D-7].

Interactions

1. This feature is a combination of other features. See TRUNK-TO-TRUNK CONNECTION [T-10], TANDEM SWITCHING OF TIE TRUNKS-2/4 WIRE [T-1], OFF-HOOK QUEUING [O-7], OUTGOING TRUNK QUEUING [O-2] and LEAST - COST ROUTING - 3/6 - DIGIT [L-5].
2. Calls in queue are served on a first in/first out basis. All Routes share a common queue.
3. This feature is used with LEAST COST ROUTING - 3/6 - DIGIT [L-5]. It is not used with ALTERNATE ROUTING [A-16], or MISCELLANEOUS TRUNK ACCESS [M-2].
4. Do not disable continuous Service Set Tone by changing SYS1, Index 42, Bit 7 to data “1”. This changes “continuous” Service Set Tone to “burst” Special Dial Tone, and Special Dial Tone to Burst Special Dial Tone followed by continuous Dial Tone. This will also disable TANDEM OFF-HOOK QUEUING.
5. When testing OGQ, at least one Trunk, of the Trunk Group being tested, must be engaged in a Station-to-Trunk or Trunk-to-Trunk Connection. Use of the MB key on the Trunk circuit card will not allow OGQ to test and operate properly.

Programming

STEP 1: **ASYD** - SYS1, Index 42, Bit 7. This Bit must be assigned as data "0", continuous Service Set Tone.

STEP 2: **ARTD** - Assign the Incoming TIE Line Route as follows:

2 and 4 WIRE E & M:

RT: 2

1-OSGS	:2	2-ONSG	:3	3-ISGS	:2	4-INSG	:3	5-TF	:3
6-TCL	:4	7-L/T	:1	8-RLP	:2	15-LSG	:5	41-TDMQ	:1

LCR(S) FLAG ROUTE:

RT: 31

6-TCL	:1	7-L/T	:1	8-RLP	:2	10-SMDR	:1	13-AC	:1
41-TDMQ	:1								

Note: To change the data for existing route (RT), initialize the circuit card.

STEP 3: **ARRC** - Allow Trunk-to-Trunk Connection between the Incoming TIE Route and the Outgoing Route in which INTER-OFFICE OFF-HOOK QUEUING will apply. Also allow Trunk-to-Trunk Connection between the Incoming TIE Route and the LCR (S) Flag Route. Assign both of these for ARI-D-RES. The Outgoing Route and the LCR(S) Flag Route must be programmed to allow OUTGOING TRUNK QUEUING [O-2].

**When incoming trunk is CCIS, assign the following data.*

STEP 4: **ASFC** - Assign SFI21=1 (Off-Hook OG Queuing is to be allowed) for the Service Feature Restriction Class of stations.

FCCS Programming

This data setting describes only the additional or replacement commands used to activate FCCS Services. When used as an FCCS feature, these commands replace the non-network commands (for example, AAED is replaced by AAEDN). Note that the data setting for FCCS Service varies depending on the Node. **Note 1**

Note 1: When using ASYDL or ASYDN to program an FCCS feature, the basic programming of ASYD is required in addition to ASYDL or ASYDN.

STEP 1: **ALRTN** - Assign this command to connect the Logical Route to the Physical Route.

STEP 2: **ASYDN** - This data must be assigned to the NCN.

SYS1, Index 41: Designate the Trunk Queuing Override Timer

Index 68, Bit 3: Enables automatic Cancel Timer

0/1 = -/Automatic Cancel Timer is available

Index 159: Designate Automatic Cancel Timer

STEP 3: **ARTDN** - Assign the Incoming TIE Line Route as follows:

2 and 4 WIRE E & M:

LGRT: 2

1-OSGS	:2	2-ONSG	:3	3-ISGS	:2	4-INSG	:3	5-TF	:3
6-TCL	:4	7-L/T	:1	8-RLP	:2	15-LSG	:5	41-TDMQ	:1

LCR(S) FLAG ROUTE:

LGRT: 31

6-TCL	:1	7-L/T	:1	8-RLP	:2	10-SMDR	:1	13-AC	:1
41-TDMQ	:1								

STEP 4: **ARRCN** - Allow Trunk-to-Trunk Connection between the Incoming TIE Route and the Outgoing Route in which INTER-OFFICE OFF-HOOK QUEUING will apply. Also allow Trunk-to-Trunk Connection between the Incoming TIE Route and the LCRS Flag Route. Assign both of these for ARI-D-RES. The Outgoing Route and the LCR(S) Flag Route must be programmed to allow OUTGOING TRUNK QUEUING [O-2].

STEP 5: **ASPAN** - These data must be assigned to the NCN.

TN: Tenant Number
ACC: Access Number
CI: "B: Busy" (for Entry), "N: Normal" (for Cancel)
SRV: "SSC: Service Code"
SID: 19 = OG Queuing-Entry
CI: "N: Normal"
SID: 20 = OG Queuing-Cancel

**When incoming trunk is CCIS, assign the following data.*

STEP 6: **ASFCN** - Assign SFI21=1 (Off-Hook OG Queuing is to be allowed) for the Service Feature Restriction Class of stations.

Note: The following data should also be set in addition to the above-mentioned commands:
The command to be set to the Node that accommodates the Trunk: ARTD
The commands to be set to all the Nodes: ASFC, AKYD

I-20 Immediate Ringback Tone

General Description

IMMEDIATE RINGBACK TONE will be heard by the caller immediately upon determination by the system that the Called Station is idle. IMMEDIATE RINGBACK TONE is connected, even if the call occurs during the OFF portion of the ringing cycle.

Operating Procedure

When a Station calls another Station or Attendant

1. The Station goes off-hook, hears Dial Tone, and dials the desired Station or Attendant.
2. IMMEDIATE RINGBACK TONE is returned.

When an Attendant calls another Station or Attendant

1. The Attendant presses an Idle Loop key and dials the desired Station or Attendant.
2. IMMEDIATE RINGBACK TONE is returned.

When a Trunk is connected to a Station or Attendant

1. The Calling Party will hear IMMEDIATE RINGBACK TONE when the source of the Ringback Tone is the Telephony Server, as is the case on DID and TIE Line Calls.

Service Conditions

The initial Ringback Tone is a brief burst, after which the system re-synchronizes the Ringback Tone with the ringing cycle.

Programming

ASYD - SYS1, Index 27, Bit 7. Assign data “1”.

I-21 Internal Zone Paging

General Description

This feature allows an individual to make a voice call through the Desktop terminal speakers of a designated group by dialing the INTERNAL ZONE PAGING Access Code or pressing the INTERNAL ZONE PAGING key on the Desktop terminal.
When an IP station is the calling party, INTERNAL ZONE PAGING can be used with several extra restrictions (for a detailed explanation, see [Service Conditions Specific to IP Station Calling Party]).

Operating Procedure

To originate

1. Lift the handset. Receive Dial Tone.
2. Dial the Access Code or press the Feature key for INTERNAL ZONE PAGING. Receive Second Dial Tone.
3. Dial the 2 or 3 digit Group Number of INTERNAL ZONE PAGING.
4. Announce the call over the built-in speakers on the Desktop terminals in the INTERNAL ZONE PAGING Group.

To answer

1. Voice Call by Internal Zone Paging [I-21] (Single)

IZP: Internal Zone Paging

a.) To answer by using Prime Line: **Note 1**

OPERATION		TONE CONTROL		DESKTOP TERMINAL DISPLAY	
CALLING PARTY	CALLED PARTY	CALLING PARTY	CALLED PARTY	CALLING PARTY	CALLED PARTY
(1) Lift the Handset (Speaker key)		(1) Dial Tone can be heard			
(2) Dial the Access Code for IZP [I-21] (IZP key)		(2) Second Dial Tone can be heard		PG	
(3) Dial the IZP Group Number **Note 3**			(1) Chimes can be heard **Note 2**	PG GPNO	PG XXXXX P: flash *P = Prime Line
Voice Call is made from Desktop terminal speakers in the IZP Group.					
	• Lift the Handset (after/before) • Press Prime Line key for the IZP Call			XXXXX	XXXXX P: light
Two-way conversation is made.					

Note 1: Using Prime Line or Off-Hook to answer is selected by ASYD, SYS1, Index 388, Bit 4=0/1: Press Prime Line key after Off-Hook/Press Off-Hook.

Note 2: The number of chimes can be set by ASYD, SYS1, Index 321, Bit 7=0/1: 4/1 times.

Note 3: The number of digits of NIZPG Number is determined by ASYDL/ASYDN, SYS1, Index 804, Bit 3.
Bit 0=0:Two-digit (Group 01~99) Ex. When NIZPG = 1, dial 01.
Bit 0=1:Three-digit (Group 001~255) Ex. When NIZPG = 1, dial 001.

b.) To Answer by using Off-Hook: **Note 4**

OPERATION		TONE CONTROL		DESKTOP TERMINAL DISPLAY	
CALLING PARTY	CALLED PARTY	CALLING PARTY	CALLED PARTY	CALLING PARTY	CALLED PARTY
(1) Lift the Handset (Speaker key)		(1) Dial Tone can be heard			
(2) Dial the Access Code for IZP [I-21] (IZP key)		(2) Second Dial Tone can be heard		PG	
(3) Dial the IZP Group Number **Note 6**			(1) Chimes can be heard **Note 5**	PG GPNO	PG XXXXX P: flash *P = Prime Line
Voice Call is made from Desktop terminal speakers in the IZP Group.					
	(1) Lift the Handset			XXXXX	XXXXX P: light
Two-way conversation is made.					

Note 4: Using Prime Line or Off-Hook to answer is selected by the ASYD, SYS1, Index 388, Bit 4=0/1: Press Prime Line key after Off-Hook/Press Off-Hook.

Note 5: The number of chimes can be set by ASYD, SYS1, Index 321, Bit 7=0/1: 4/1 times.

Note 6: The number of digits of NIZPG Number is determined by ASYDL/ASYDN, SYS1, Index 804, Bit 3.
Bit 0=0:Two-digit (Group 01~99) Ex. When NIZPG = 1, dial 01.
Bit 0=1:Three-digit (Group 001~255) Ex. When NIZPG = 1, dial 001.

c.) To answer by using CALL PICKUP - DIRECT [C-30]:

OPERATION		TONE CONTROL		DESKTOP TERMINAL DISPLAY	
CALLING PARTY	CALLED PARTY	CALLING PARTY	CALLED PARTY	CALLING PARTY	CALLED PARTY
(1) Lift the Handset (Speaker key)		(1) Dial Tone can be heard			
(2) Dial the Access Code for IZP [I-21] (IZP key)		(2) Second Dial Tone can be heard		PG	
(3) Dial the IZP Group Number **Note 8**			(1) Chimes can be heard **Note 7**	PG GPNO	PG XXXXX P: flash *P = Prime Line
Voice Call is made from Desktop terminal speakers in the IZP Group.					
	(1) Lift the Handset		(2) Dial Tone can be heard		
	(2) Dial the Access Code for Call Pickup - Direct [C-30]		(3) Second Dial Tone can be heard		
	(3) Dial the desired Station Number in the IZP [I-21] Group			XXXX	XXXXX P: light
Two-way conversation is made.					

Note 7: The number of chimes can be set by ASYD, SYS1, Index 321, Bit 7=0/1: 4/1 times.

Note 8: The number of digits of NIZPG Number is determined by ASYDL/ASYDN, SYS1, Index 804, Bit 3.
Bit 0=0:Two-digit (Group 01~99) Ex. When NIZPG = 1, dial 01.
Bit 0=1:Three-digit (Group 001~255) Ex. When NIZPG = 1, dial 001.

d.) To answer by using CALL PICKUP - GROUP [C-7]:

OPERATION		TONE CONTROL		DESKTOP TERMINAL DISPLAY	
CALLING PARTY	CALLED PARTY	CALLING PARTY	CALLED PARTY	CALLING PARTY	CALLED PARTY
(1) Lift the Handset (Speaker key)		(1) Dial Tone can be heard			
(2) Dial the Access Code for IZP [I-21] (IZP key)		(2) Second Dial Tone can be heard		PG	
(3) Dial the IZP Group Number **Note 10**			(1) Chimes can be heard **Note 9**	PG GPNO	PG XXXXX P: flash *P = Prime Line
Voice Call is made from Desktop terminal speakers in the IZP Group.					
	(1) Lift the Handset		(2) Dial Tone can be heard		P: light
	(2) Dial the Access Code for Call Pickup - Group [C-7] (Call Pickup key)			XXXXX	XXXXX
Two-way conversation is made.					

Note 9: The number of chimes can be set by ASYD, SYS1, Index 321, Bit 7=0/1: 4/1 times.

Note 10: The number of digits of NIZPG Number is determined by ASYDL/ASYDN, SYS1, Index 804, Bit 3.
Bit 0=0:Two-digit (Group 01~99) Ex. When NIZPG = 1, dial 01.
Bit 0=1:Three-digit (Group 001~255) Ex. When NIZPG = 1, dial 001.

2. Voice Call by Internal Zone Paging [I-21] (Station Hold)

IZP: Internal Zone Paging

a.) To answer using Prime Line: **Note 11**

OPERATION		TONE CONTROL			DESKTOP TERMINAL DISPLAY		
OPERATING PARTY	CALLED PARTY	OPERATING PARTY	CALLED PARTY	CALLING PARTY	OPERATING PARTY	CALLED PARTY	CALLING PARTY
(1) Receive a call					XXXXX		XXXXX
(2) Perform the SHF		(1) Special Dial Tone can be heard		(1) MSC can be heard	XFR		HOLD
(3) Dial the Access Code for IZP [I-21] (IZP key)		(2) Second Dial Tone can be heard			PG		HOLD
(4) Dial the IZP Group Number **Note 13**			(1) Chimes can be heard **Note 12**		PG GPNO	PG XXXXX P: flash	HOLD
Voice Call is made from Desktop terminal speakers in the IZP Group							
	• Lift the Handset (after/ before) • Press Prime Line key for the IZP Call				XXXXX	XXXXX P: light	HOLD
Two-way conversation is made.							
(5) Replace the handset				(2) MSC is stopped	(time)	XXXXX	XXXXX
	Two-way conversation between the Calling and Called Party is made.						

Note 11: Using Prime Line or Off-Hook to answer is selected by the ASYD, SYS1, Index 388, Bit 4=0/1: Press Prime Line key after Off-Hook/Press Off-Hook.

Note 12: The number of chimes can be set by ASYD, SYS1, Index 321, Bit 7=0/1: 4/1 times.

Note 13: The number of digits of NIZPG Number is determined by ASYDL/ASYDN, SYS1, Index 804, Bit 3.
Bit 0=0:Two-digit (Group 01~99) Ex. When NIZPG = 1, dial 01.
Bit 0=1:Three-digit (Group 001~255) Ex. When NIZPG = 1, dial 001.

b.) To answer by using Off-Hook: **Note 14**

OPERATION		TONE CONTROL			DESKTOP TERMINAL DISPLAY		
OPERATING PARTY	CALLED PARTY	OPERATING PARTY	CALLED PARTY	CALLING PARTY	OPERATING PARTY	CALLED PARTY	CALLING PARTY
(1) Receive a call					XXXXX		XXXXX
(2) Perform the SHF		(1) Special Dial Tone can be heard		(1) MSC can be heard	XFR		HOLD
(3) Dial the Access Code for IZP [I-21] (IZP key)		(2) Second Dial Tone can be heard			PG		HOLD
(4) Dial the IZP Group Number **Note 16**			(1) Chimes can be heard **Note 15**		PG GPNO	PG XXXXX P: flash *P = Prime Line	HOLD
Voice Call is made from Desktop terminal speakers in the IZP Group							
	(1) Lift the Handset				XXXXX	XXXXX P: light	HOLD
Two-way conversation is made.							
(5) Replace the handset				(2) MSC is stopped	(time)	XXXXX	XXXXX
	Two-way conversation between the Calling and Called Party is made.						

Note 14: Using Prime Line or Off-Hook to answer is selected by the ASYD, SYS1, Index 388, Bit 4=0/1: Press Prime Line key after Off-Hook/Press Off-Hook.

Note 15: The number of chimes can be set by ASYD, SYS1, Index 321, Bit 7=0/1: 4/1 times.

Note 16: The number of digits of NIZPG Number is determined by ASYDL/ASYDN, SYS1, Index 804, Bit 3.
Bit 0=0:Two-digit (Group 01~99) Ex. When NIZPG = 1, dial 01.
Bit 0=1:Three-digit (Group 001~255) Ex. When NIZPG = 1, dial 001.

c.) To answer by using Call Pickup - Direct [C-30]:

OPERATION		TONE CONTROL			DESKTOP TERMINAL DISPLAY		
OPERATING PARTY	CALLED PARTY	OPERATING PARTY	CALLED PARTY	CALLING PARTY	OPERATING PARTY	CALLED PARTY	CALLING PARTY
(1) Receive a call					XXXXX		XXXXX
(2) Perform the SHF		(1) Special Dial Tone can be heard		(1) MSC can be heard	XFR		HOLD
(3) Dial the Access Code for IZP Number [I-21] (IZP key)		(2) Second Dial Tone can be heard			PG		HOLD
(4) Dial the IZP Group Number **Note 18**			(1) Chimes can be heard **Note 17**		PG GPNO	PG XXXXX P: flash *P = Prime Line	HOLD
Voice Call is made from Desktop terminal speakers in the IZP Group							
	(1) Lift the Handset (Speaker key)		(2) Dial Tone can be heard				
	(2) Dial the Access Code for Call Pickup - Direct [C-30]		(3) Second Dial Tone can be heard				
	(3) Dial the desired Station Number in the IZP [I-21] Group				XXXXX	XXXXX P: light	HOLD
Two-way conversation is made.							
(5) Replace the handset				(2) MSC is stopped	(time)	XXXXX	XXXXX
	Two-way conversation between the Calling and Called Party is made.						

Note 17: The number of chimes can be set by ASYD, SYS1, Index 321, Bit 7=0/1: 4/1 times.

Note 18: The number of digits of NIZPG Number is determined by ASYDL/ASYDN, SYS1, Index 804, Bit 3.
Bit 0=0:Two-digit (Group 01~99) Ex. When NIZPG = 1, dial 01.
Bit 0=1:Three-digit (Group 001~255) Ex. When NIZPG = 1, dial 001.

d.) To answer by using Call Pickup - Group [C-7]:

OPERATION		TONE CONTROL			DESKTOP TERMINAL DISPLAY		
OPERATING PARTY	CALLED PARTY	OPERATING PARTY	CALLED PARTY	CALLING PARTY	OPERATING PARTY	CALLED PARTY	CALLING PARTY
(1) Receive a call					XXXXX		XXXXX
(2) Perform the SHF		(1) Special Dial Tone can be heard		(1) MSC can be heard	XFR		HOLD
(3) Dial the Access Code for IZP Number [I-21] (IZP key)		(2) Second Dial Tone can be heard			PG		HOLD
(4) Dial the IZP Group Number **Note 20**			(1) Chimes can be heard **Note 19**		PG GPNO	PG XXXXX P: flash *P = Prime Line	HOLD
Voice Call is made from Desktop terminal speakers in the IZP Group							
	(1) Lift the Handset		(2) Dial Tone can be heard				
	(2) Dial the Access Code for Call Pickup - Group [C-7] (Call Pickup - Group key)				XXXXX	XXXXX P: light	HOLD
Two-way conversation is made.							
(5) Replace the handset				(2) MSC is stopped	(time)	XXXXX	XXXXX
	Two-way conversation between the Calling and Called Party is made.						

Note 19: The number of chimes can be set by ASYD, SYS1, Index 321, Bit 7=0/1: 4/1 times.

Note 20: The number of digits of NIZPG Number is determined by ASYDL/ASYDN, SYS1, Index 804, Bit 3.
Bit 0=0:Two-digit (Group 01~99) Ex. When NIZPG = 1, dial 01.
Bit 0=1:Three-digit (Group 001~255) Ex. When NIZPG = 1, dial 001.

3. Voice Call by Internal Zone Paging (Trunk Hold)

a.) To answer by using Prime Line: **Note 21**

OPERATION		TONE CONTROL			DESKTOP TERMINAL DISPLAY		
OPERATING PARTY	CALLED PARTY	OPERATING PARTY	CALLED PARTY	CALLING PARTY	OPERATING PARTY	CALLED PARTY	CALLING PARTY
(1) Receive a call					XX:XX DDD2		
(2) Perform the SHF		(1) Special Dial Tone can be heard		(1) MSC can be heard	XFR		
(3) Dial the Access Code for IZP [I-21] (IZP key)		(2) Second Dial Tone can be heard			PG		
(4) Dial the IZP Group Number **Note 23**			(1) Chimes can be heard **Note 22**		PG GPNO	PG XXXXX P: flash *P = Prime Line	
Voice Call is made from Desktop terminal speakers in the IZP Group							
	• Lift the Handset (after/ before) • Press Prime Line key for the IZP Call				XXXXX	XXXXX P: light	
Two-way conversation is made.							
(5) Replace the handset				(2) MSC is stopped	(time)	XX:XX DDD2	
	Two-way conversation between the Calling and Called Party is made.						

Note 21: Using Prime Line or Off-Hook to answer is selected by the ASYD, SYS1, Index 388, Bit 4=0/1: Press Prime Line key after Off-Hook/Press Off-Hook.

Note 22: The number of chimes can be set by ASYD, SYS1, Index 321, Bit 7=0/1: 4/1 times.

Note 23: The number of digits of NIZPG Number is determined by ASYDL/ASYDN, SYS1, Index 804, Bit 3.
Bit 0=0:Two-digit (Group 01~99) Ex. When NIZPG = 1, dial 01.
Bit 0=1:Three-digit (Group 001~255) Ex. When NIZPG = 1, dial 001.

b.) To answer by using Off-Hook: **Note 24**

OPERATION		TONE CONTROL			DTERM DISPLAY		
OPERATING PARTY	CALLED PARTY	OPERATING PARTY	CALLED PARTY	CALLING PARTY	OPERATING PARTY	CALLED PARTY	CALLING PARTY
(1) Receive a call					XX:XX DDD2		
(2) Perform the SHF		(1) Special Dial Tone can be heard		(1) MSC can be heard	XFR		
(3) Dial the Access Code for IZP [I-21] (IZP key)		(2) Second Dial Tone can be heard			PG		
(4) Dial the IZP Group Number **Note 26**			(1) Chimes can be heard **Note 25**		PG GPNO	PG XXXXX P: flash *P = Prime Line	
Voice Call is made from Desktop terminal speakers in the IZP Group.							
	(1) Lift the Handset				XXXXX	XXXXX P: light	
Two-way conversation is made.							
(5) Replace the handset				(2) MSC is stopped	(time)	XX:XX DDD2	
	Two-way conversation between the Calling and Called Party is made.						

Note 24: Using Prime Line or Off-Hook to answer is selected by the ASYD, SYS1, Index 388, Bit 4=0/1: Press Prime Line key after Off-Hook/Press Off-Hook.

Note 25: The number of chimes can be set by ASYD, SYS1, Index 321, Bit 7=0/1: 4/1 times.

Note 26: The number of digits of NIZPG Number is determined by ASYDL/ASYDN, SYS1, Index 804, Bit 3.
Bit 0=0:Two-digit (Group 01~99) Ex. When NIZPG = 1, dial 01.
Bit 0=1:Three-digit (Group 001~255) Ex. When NIZPG = 1, dial 001.

c.) To answer by using Call Pickup - Direct [C-30]:

OPERATION		TONE CONTROL			DESKTOP TERMINAL DISPLAY		
OPERATING PARTY	CALLED PARTY	OPERATING PARTY	CALLED PARTY	CALLING PARTY	OPERATING PARTY	CALLED PARTY	CALLING PARTY
(1) Receive a call					XX:XX DDD2		
(2) Perform the SHF		(1) Special Dial Tone can be heard		(1) MSC can be heard	XFR		
(3) Dial the Access Code for IZP [I-21] (IZP key)		(2) Second Dial Tone can be heard			PG		
(4) Dial the IZP Group Number **Note 28**			(1) Chimes can be heard **Note 27**		PG GPNO	PG XXXXX P: flash *P = Prime Line	
Voice Call is made from Desktop terminal speakers in the IZP Group.							
	(1) Lift the Handset (Speaker key)		(2) Dial Tone can be heard				
	(2) Dial the Access Code for Call Pickup - Direct [C-30]		(3) Second Dial Tone can be heard				
	(3) Dial the desired Station Number in the IZP [I-21] Group				XXXXX	XXXXX P: light	
Two-way conversation is made.							
(5) Replace the handset				(2) MSC is stopped	(time)	XX:XX DDD2	
	Two-way conversation between the Calling and Called Party is made.						

Note 27: The number of chimes can be set by ASYD, SYS1, Index 321, Bit 7=0/1: 4/1 times.

Note 28: The number of digits of NIZPG Number is determined by ASYDL/ASYDN, SYS1, Index 804, Bit 3.
Bit 0=0:Two-digit (Group 01~99) Ex. When NIZPG = 1, dial 01.
Bit 0=1:Three-digit (Group 001~255) Ex. When NIZPG = 1, dial 001.

d.) To answer by using Call Pickup - Group [C-7]

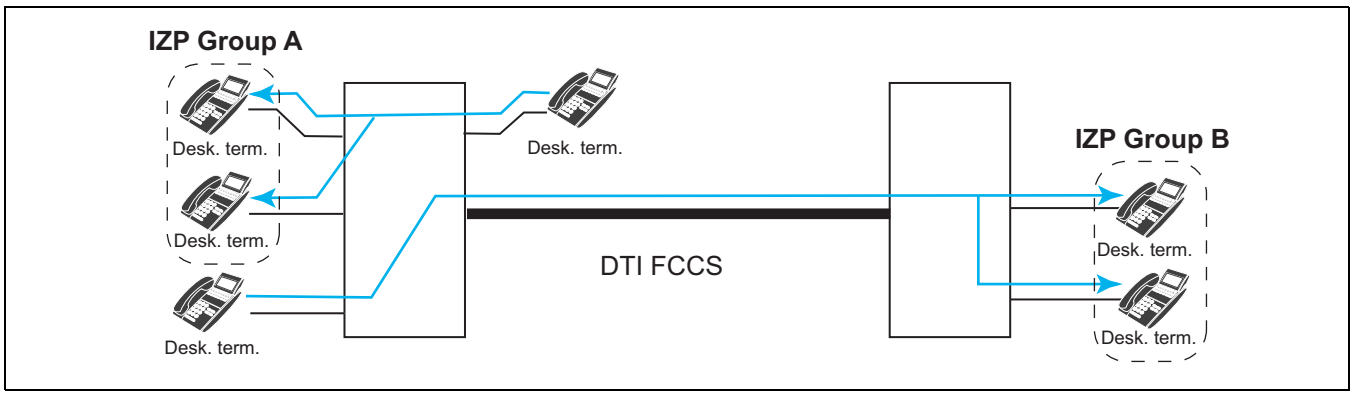
OPERATION		TONE CONTROL			DESKTOP TERMINAL DISPLAY		
OPERATING PARTY	CALLED PARTY	OPERATING PARTY	CALLED PARTY	CALLING PARTY	OPERATING PARTY	CALLED PARTY	CALLING PARTY
(1) Receive a call					XX:XX DDD2		
(2) Perform the SHF		(1) Special Dial Tone can be heard		(1) MSC can be heard	XFR		
(3) Dial the Access Code for IZP [I-21] (IZP key)		(2) Second Dial Tone can be heard			PG		
(4) Dial the IZP Group Number **Note 30**			(1) Chimes can be heard **Note 29**		PG GPNO	PG XXXXX P: flash *P = Prime Line	
Voice Call is made from Desktop terminal speakers in the IZP Group							
	(1) Lift the Handset (2) Dial the Access Code for Call Pickup - Group [C-7] (Call Pickup - Group key)		(2) Dial Tone can be heard		XXXXX	XXXXX P: light	
Two-way conversation is made.							
(5) Replace the handset				(2) MSC is stopped	(time)	XX:XX DDD2	
	Two-way conversation between the Calling and Called Party is made.						

Note 29: The number of chimes can be set by ASYD, SYS1, Index 321, Bit 7=0/1: 4/1 times.

Note 30: The number of digits of NIZPG Number is determined by ASYDL/ASYDN, SYS1, Index 804, Bit 3.
Bit 0=0:Two-digit (Group 01~99) Ex. When NIZPG = 1, dial 01.
Bit 0=1:Three-digit (Group 001~255) Ex. When NIZPG = 1, dial 001.

Service Conditions

1. The number of Internal Zone Paging Groups is a maximum of 99 Groups for each Tenant.
2. Each Desktop terminal can be a member of Multiple Internal Zone Paging Groups.
3. A maximum of 255 Groups per system can be simultaneously Paged using this feature.
4. This feature is activated to idle Desktop terminals in an Internal Zone Paging Group.
5. When all INTERNAL ZONE PAGING Groups are busy, Busy Tone is sent.
6. This service is not available from the Attendant Console, Central Office and TIE Line.
7. If all Desktop terminals in the designated group are unavailable, a Busy Tone will be sent.
8. When originating a call (or answering an Incoming Call) from a Multi (Sub) Line on the Desktop terminal called by INTERNAL ZONE PAGING, the Voice Call to the Prime Line stops and becomes idle, and the path on the Multi Line is established. Even if the call is completed, the Voice Call is not established again.
9. When a user cancels this feature with a Held Call, the user is recalled.
10. If a Desktop terminal in the Paged Group is busy when the INTERNAL ZONE PAGING Call is originated, the terminal is not called/recalled even if it becomes idle later.
11. After a Desktop terminal in the Paged Group answers the INTERNAL ZONE PAGING Call, ringing at the other member Desktop terminals stop and two-way conversation starts.
12. While member Desktop terminals are Paged by INTERNAL ZONE PAGING Call, the Prime Lines of Desktop terminals in the group are in busy status. Thus, the Desktop terminals cannot receive another call from Desktop terminal, TIE Line, Central Office Line or Attendant Consoles during Paging.
13. When a Prime Line of a Desktop terminal in the Paged Group is a Sub Line of another Desktop terminal, the Sub Line assigned to another Desktop terminal cannot be affected by the INTERNAL ZONE PAGING Call (the Sub Line key does not flash or light).
14. The user can Page any INTERNAL ZONE PAGING Groups in the DTI FCCS using this feature.

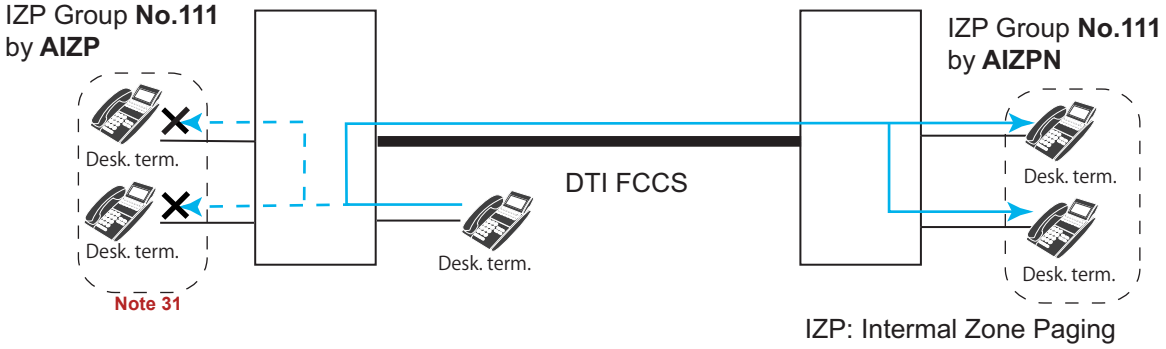


15. When activating this feature via the FCCS Network, an Internal Zone Paging Group can consist of the Desktop terminals accommodated in the same Node. (The group cannot be comprised of the Desktop terminals in different Nodes.)
16. The maximum number of INTERNAL ZONE PAGING Group per FCCS Network is 255 Groups.
17. Using Prime Line or Off-Hook to answer can be selected. If this selection is not unified between each Node within the FCCS Network, execute the method of answering assigned to each Node when answering a telephone.
18. When the Connection Trunk for the Node that accommodates the designated group is in one of the following status, Busy Tone is sent or “Busy” is displayed on the LCD.

- the designated Connection Trunk is busy
- FCCS link failure (Layer 2 down) occurs at the Connection Trunk
- the Connection Trunks in an idle state are fewer than the Idle Stations in the designated group

19. A Desktop terminal that has a call on Hold cannot answer the voice call via FCCS using CALL PICKUP-DIRECT [C-30] and CALL PICKUP-GROUP [C-7].
20. When the IZP Group Number assigned by AIZP and IZP Group Number assigned by AIZPN are mixed in the FCCS Network, the IZP Group Number assigned by AIZPN takes priority. Refer to the following figure.

- When identical IZP Group Numbers assigned by AIZP and AIZPN are used in the FCCS network.

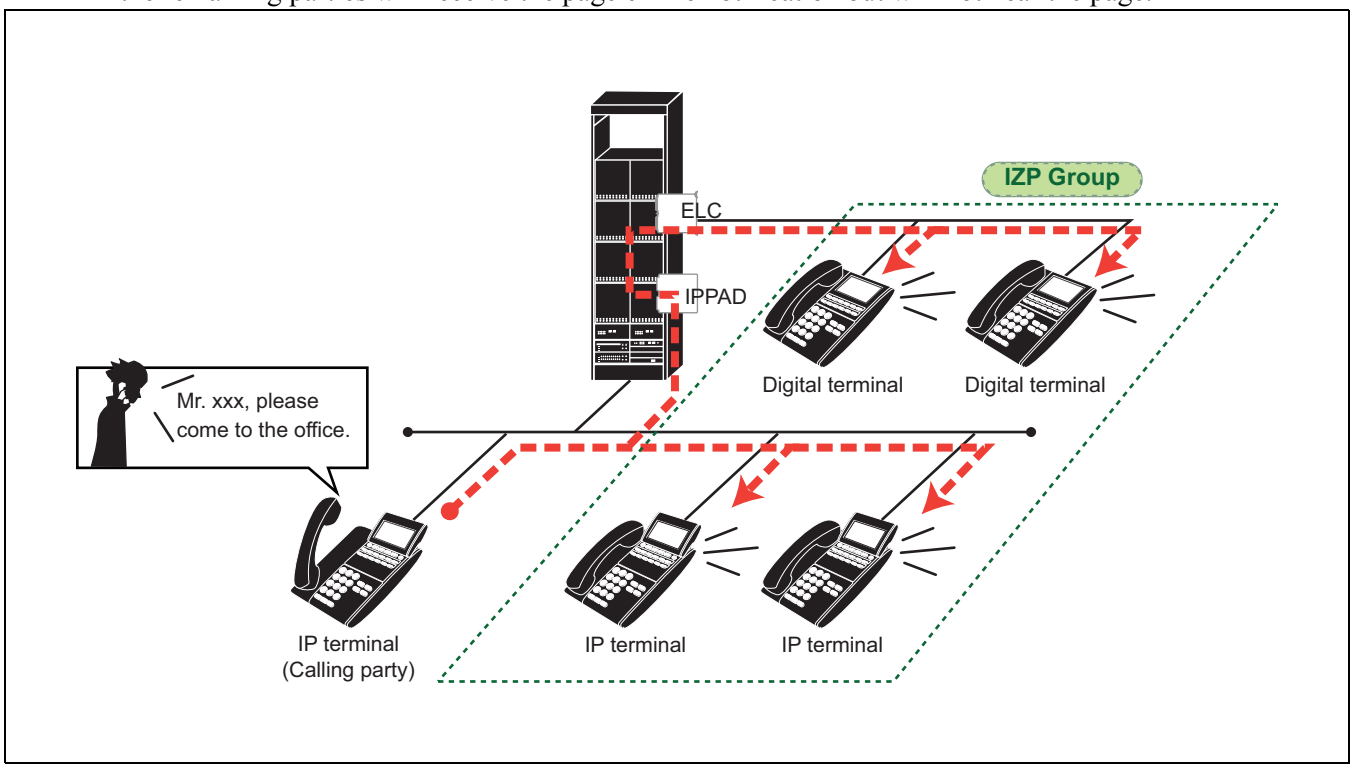


Note 31: In this case, this IZP Group cannot be paged.

21. Multiple voice calls cannot be made at a time.
22. This service is not available for FCCS over IP Network.
23. A maximum of four Desktop terminals for INTERNAL ZONE PAGING can be assigned to a Group (Level 0 - Level 7) for physical ports connected to the system.
24. A maximum number of 24 Desktop terminals can be set in a group with this feature

[Service Conditions Specific to IP Station Calling Party]

1. When the calling party is DT830G/DT730G, the IZP group cannot be bigger than four members. If more than four stations are programmed in the same group, only four stations in the group will hear the page; the remaining parties will receive the page chime notification but will not hear the page.



2. IP stations cannot be the calling party except for DT830G/DT730G.

   Note: If the calling party is DT730G, it must be 1.0.3.0 release or higher.

3. The PCPro AIZP command will not show an alert if more than four stations are programmed.

Interactions

When a Station answers a Voice Call using this feature, the call becomes Station-to-Station Connection Status. Accordingly, various hooking services are available.

Programming

Refer to CALL PICKUP - DIRECT [C-30] and CALL PICKUP - GROUP [C-7] when using those features with INTERNAL ZONE PAGING.

STEP 1: **ASYD** - SYS1, Index 31 (Mounting capacity of Common Memory) must be assigned as data "04". This data is required when using the Virtual PIR.
SYS1, Index 321, Bit 7:0/1 = Voice Call Chime 4 chime/1 chime
SYS1, Index 388, Bit 4:0/1 = Answer the INTERNAL ZONE PAGING call by Prime Line key after off-hook/by going off hook.

STEP 2: **ASYDL** - SYS1, Index 679, Bit 1: 1 (Multiple Session for SIP Multiple Line Terminals: In Service). **Note 32**
SYS1, Index 804, Bit 3 (Designation of the maximum digit number of INTERNAL ZONE PAGING Group Number) 0/1 = two/three digits. **Note 33**

STEP 3: **ANPD** - Reserve a Number Level for Feature Access. Select Normal Service (N) or Hooking Service (H) from the combo box for Connection Indexes. Assign NND according to the predetermined Numbering Plan.

STEP 4: **ASPA** - Assign the Access Code for INTERNAL ZONE PAGING, SRV = "SSCA: Service Code Appendix", SIDA = 98. Connection Status Indexes for "N: Normal" and "H: Hooking" services.

STEP 5: **AIZP** - Assign INTERNAL ZONE PAGING Group Data.

TN: Tenant Number of Calling Party
IZPG: INTERNAL ZONE PAGING Group Number (1~99)
IZPTN: Tenant Number of IZPSTN
IZPSTN: Station Number assigned to the INTERNAL ZONE PAGING Group

STEP 6: **AKYD** - For Desktop terminal sets, INTERNAL ZONE PAGING can be activated using a programmable Line/Feature key. Assign KYI = 1, Feature key Answer [28] for answering INTERNAL ZONE PAGING and KYI = 1, Feature key Internal Zone Paging [85] for calling INTERNAL ZONE PAGING Group.

STEP 7: **ASFC** - Assign a Service Feature Restriction Class that allows SFI 134 (Internal Zone Page Calling) to the Stations that originate INTERNAL ZONE PAGING.

Assign a Service Feature Restriction Class that allows SFI 145 (Internal Zone Paging Answer) to the Stations that answer to INTERNAL ZONE PAGING.

Note 32: Only when IP stations are used. After the setting, a re-registration must be performed.

Note 33: When selecting 3 digits, ASYDL setting is performed regardless of the LDM programming.

FCCS Programming

The following command assignments are required when your system is configured as an FCCS Network.

STEP 1: **ASYD** - SYS1, Index 31 (Mounting capacity of Common Memory) must be assigned as data "04".
SYS1, Index 321, Bit 7:0/1 = Voice Call Chime 4 chime/1 chime
SYS1, Index 388, Bit 4 0/1= Answer the INTERNAL ZONE PAGING call by Prime Line key after off-hook/by going off-hook.

STEP 2: **ASYDL/ASYDN** - SYS1, Index 679, Bit 1: 1 (Multiple Session for SIP Multiple Line Terminals: In Service). **Note 34**
SYS1, Index 804, Bit 3 (Designation of the maximum digit number of INTERNAL ZONE PAGING Group Number) 0/1 = two or three digits.

STEP 3: **ANPD/ANPDL/ANPDN** - Reserve a Number Level for Feature Access. Select Normal Service (N) or Hooking Service (H) from the combo box for Connection Indexes. Assign NND according to the predetermined Numbering Plan.

STEP 4: **ASPA/ASPAL/ASPAN** - Assign the Access Code for INTERNAL ZONE PAGING, SRV = "SSCA: Service Code Appendix", SIDA = 98. Connection Status Indexes for "N: Normal", and "H: Hooking" services.

STEP 5: **AIZPN** - Assign INTERNAL ZONE PAGING Group Data.

NIZPG: Network Internal Zone Paging Group Number (1~255) **Note 35**
UGN: User Group Number
TELN: Telephone Number
CNT: Number of Telephone Number

STEP 6: **AKYD** - For Dterm sets, INTERNAL ZONE PAGING can be activated using a programmable Line/Feature key. Assign KYI (Service Condition) = 1 (Feature Key) and Feature key Internal Zone Paging [85]. Assign KYI (Service Condition) = 1 (Feature Key) and Feature key Answer [28] for answering INTERNAL ZONE PAGING.

STEP 7: **ASFC** - Assign a Service Feature Restriction Class that allows SFI 145 (Internal Zone Paging Answer) to the Stations that answer to INTERNAL ZONE PAGING. Assign a Service Feature Restriction Class that allows SFI 134 (Internal Zone Page Calling) to the Stations that originate INTERNAL ZONE PAGING.

Note 34: Only when IP stations are used. After the setting, a re-registration must be performed.

Note 35: The number of digits of Internal Zone Paging Group Number is determined by SYS1, Index 804, Bit 3.

I-43 IP Enabled Dterm

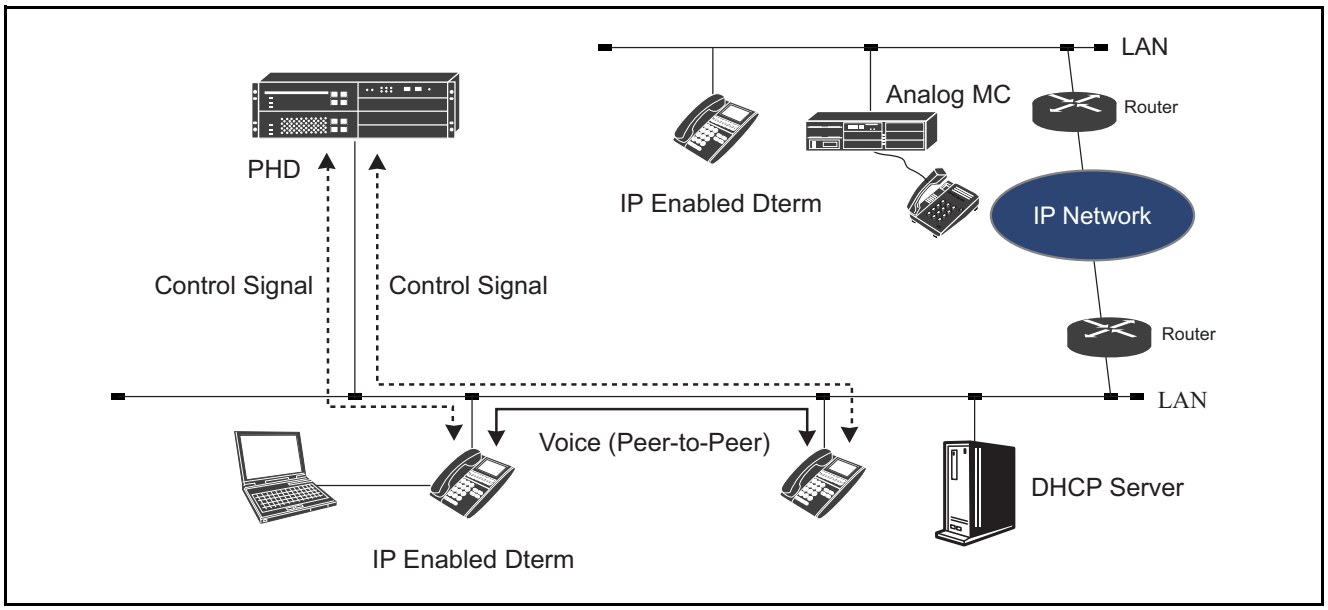
General Description

This feature allows the system to use the Dterm Series *i* telephone sets on the LAN. A Dterm Series *i* telephone set that is equipped with an adapter (hereafter referred to as IP Enabled Dterm) can send or receive Voice Signals to or from other IP terminals (such as an Analog MC, other IP Enabled Dterm, or IPPAD circuit card). An IP Enabled Dterm can be relocated by simple plug-in operation, without changing the office data.
At this time, the control signals are processed by the System, while the speech path is established directly between the IP terminals (voice data are sent/received in a Peer-to-Peer connection).

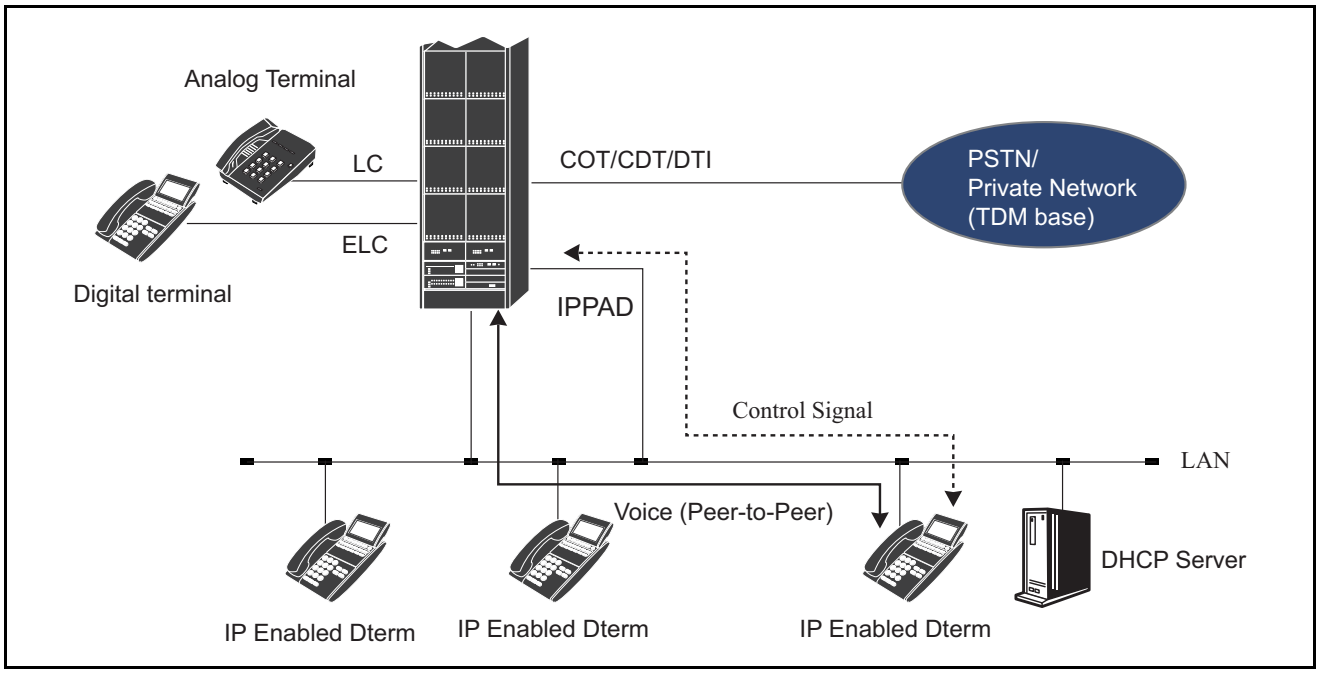
Note: An IPPAD circuit card is used to help establish a call between IP Enabled Dterm (IP Station) and another TDM-based Conventional Telephone/Network (such as Analog Telephone, Dterm Series i without an adapter, Public Network, TIE Line.) IPPAD provides a gateway function between IP (Internet Protocol on the LAN side) and TDM (Time Division Multiplexer). For more details, refer to the **IPPAD Data** section in Chapter 2 of this manual.

Note: IP Enabled Dterm does not support Follow Phone [F-31].

IP Enabled Dterm



IP Enabled Dterm with IPPAD



Hardware Requirements

The following hardware is required for this feature:

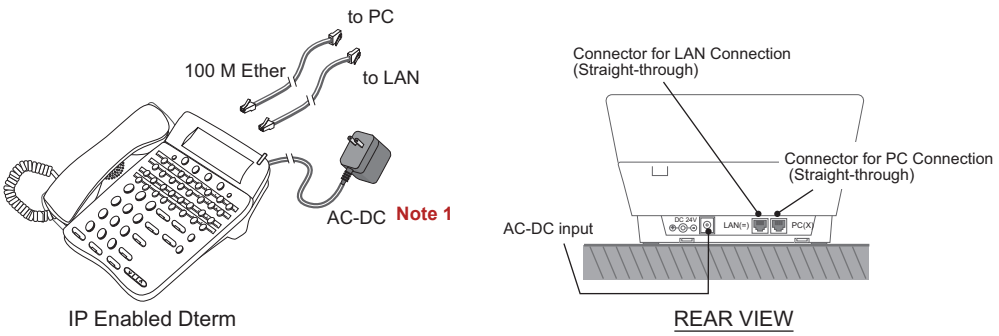
- ☐ IP Enabled Dterm

  Consists of the following:

  - Dterm Series *i*
  - IP Interface Adapter
  - AC Power Adapter

Note: An IP Enabled Dterm is composed of a Desktop terminal and an IP Interface Adapter.

- ☐ Switching HUB (10/100M)



Note 1: Center power feeding by using a Power Patch Panel is also available.

- DHCP Server - Option
- FTP (TFTP) Server - Option
- IPPAD - Option

  This circuit card is necessary when a call is to be established between the TDM-based Conventional Telephones/ Network and IP Enabled Dterms. At this time, IPPAD provides a gateway function: Voice Signal (IP Packets) sent from the LAN is disassembled as the TDM-based Signal, and the opposite directions assembled into the IP Packets. Thus, a maximum of 32 calls can be dealt with per IPPAD Card. For more detailed information, refer to Section 10 of CHAPTER 2 or the Circuit Card Description.

Operating Procedure

See [D-159] for the operating procedure in MAC Authentication mode and Login mode.

Service Conditions

1. SP-PHD is required for this feature.

   Note: SP-PHD handles the Control Signals sent to and from IP Enabled Dterm. SP-PHD can handle a maximum of 4000 IP Enabled Dterms. For more detailed information on PH, refer to the Protocol Handler Reference Note.

2. IP Enabled Dterm operates for Dterm Series *i* only.
3. Desktop terminal without IP Interface Adapter cannot be used.
4. IP Enabled Dterm cannot use features supported by other adapters like Analog Port adapter, Ancillary Device adapter, and so on.
5. A DSS console can accommodate IP Enabled Dterms as Multiple Lines, but cannot be connected to them.

Programming

See [D-159] DtermIP for the data programming.

I-44 IP Control-FAX

General Description

This feature allows the user to specify jitter-buffer size, payload type (G.711, G.726, T.30, and T.38) and payload size of RTP packet from the PCPro for the Peer to Peer FAX connection.

Operating Procedure

None

Service Conditions

1. When no FAX data is assigned, or connecting to IPPAD without VCT, data transmission as voice data (jitter-buffer size=150ms, payload type=G.711, payload size=40ms) is used.
2. Available FAX communication methods are T.30, T.38 and data transmission as voice data (G3 FAX (V.17, V.21, V.29, V.27ter, V33)).

   Note: Payload Type T.38 is available from FP95-114 V4. It is only available as explained in the connection pattern (f) of the service condition number 4.

3. For FAX communication between Legacy Stations/External Trunks accommodated in different nodes in FCCS over IP network, communication is established between IPPADs in those nodes. FAX control data can be changed by ASYDL/ASYDN, Index 652 to Index 654.
4. The following table shows the FAX connection patterns. Priority of FAX data to be used is different in each Connection Pattern.

   Note: Data settings and priorities for connection patterns (a) to (d) are the same even if the connections are established in FCCS over IP network.

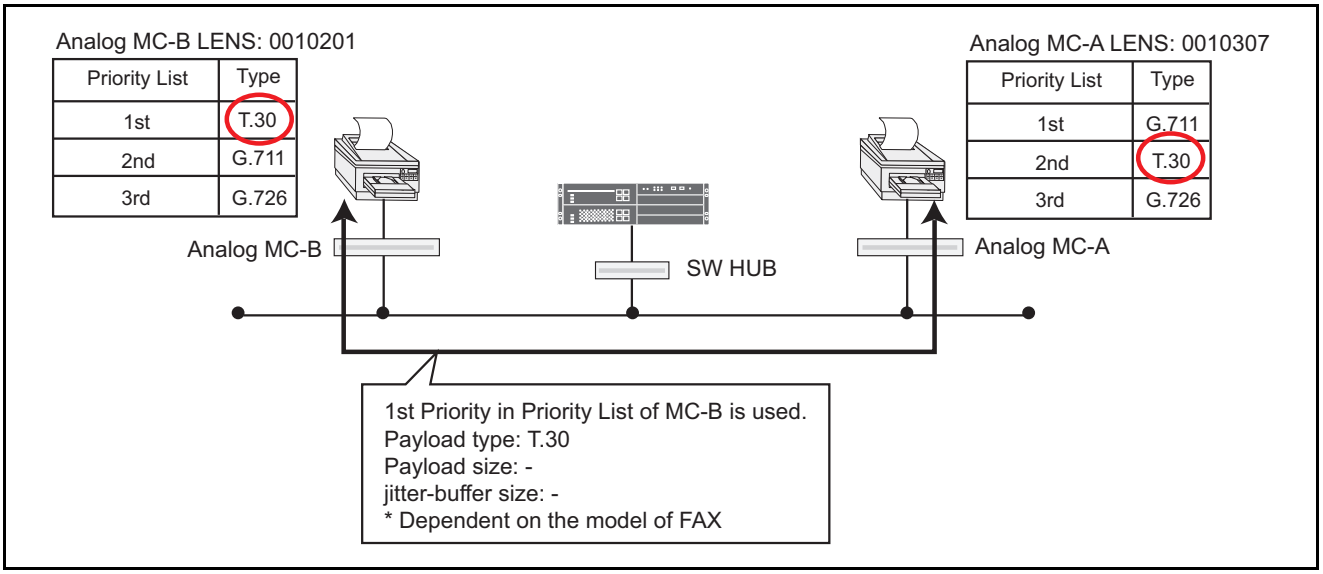
   Note: “MC” mentioned here is MC side equipped with Analog MC and MC&MG-COT (hereafter MC).

Connection Pattern	Data Setting	PCPro Command	Priority
(a) IP Station (MC) ⇔ IP Station (MC)	Set the data per MC port	AISTL	-
(b) IP Station (MC) ⇔ IPPAD (IP Station [MC] ⇔ Legacy Station/External Trunk)	When MC is to be connected to the external trunk, the FAX data for each route is used.	ARTI/ARTIN	High ↕ Low
	When MC is to be connected to the external trunk, and no FAX data is set to the external trunk, the data set to the MC is used.	AISTL	
	When MC is to be connected to Station, the data set to the MC is used.	AISTL	
	When no FAX data is set, the data transmission as voice data (payload type=G.711, payload size=40ms, jitter-buffer size=150ms) is used.	-	

Connection Pattern		Data Setting	PCPro Command	Priority
(c) IP Station (MC) ⇔ MG		When MC is to be connected to the MG, FAX data for each external trunk route set to the MG is used.	ARTI/ARTIN	High ↕ Low
		When no FAX data (data set by ARTI/ARTIN command) is set to the external trunk route on MG, the data transmission as voice data (payload type=G.711, payload size=40ms, jitter-buffer size=150ms) is used.	-	
(d) CCIS over IP by PHC		FAX data set to the virtual speech route for CCIS over IP by PHC is used.	ARTI/ARTIN	-
		When no FAX data is set, the data transmission as voice data (payload type=G.711, payload size=40ms, jitter-buffer size=150ms) is used. **Note:** FAX data set to the destination through CCIS over IP by PHC is not valid.	-	
(e) Legacy Station/External Trunk ⇔ Legacy Station/External Trunk via FCCS over IP network	Legacy Station ⇔ Legacy Station	System data set to each node is used.	ASYDL/ ASYDN	-
	Legacy Station ⇔ Legacy External Trunk	FAX data set to each external route is used.	ARTI/ARTIN	
		When no FAX data is set, the data transmission as voice data (payload type=G.711, payload size=40ms, jitter-buffer size=150ms) is used.	-	
(f) SIP Handler Controlled Standard SIP Terminal ⇔ SIP Handler Controlled Standard SIP Terminal/ MG(SIP)	SIP Handler Controlled Standard SIP Terminal ⇔ SIP Handler Controlled Standard SIP Terminal	FAX data is set by enabling multimedia communication. **Note:** T.38 is the only possible payload type. **Note:** SIP Handler Controlled Standard SIP Terminals must have software installed for T.38 FAX use.	AUACL/ AUACN	-
	SIP Handler Controlled Standard SIP Terminal ⇔ MG(SIP)		ARTI/ARTIN AUACL/ AUACN	

(a) IP Station (MC) ⇔ IP Station (MC)

Note: The figures in (a) to (c) show the connections in case of via Analog MC.

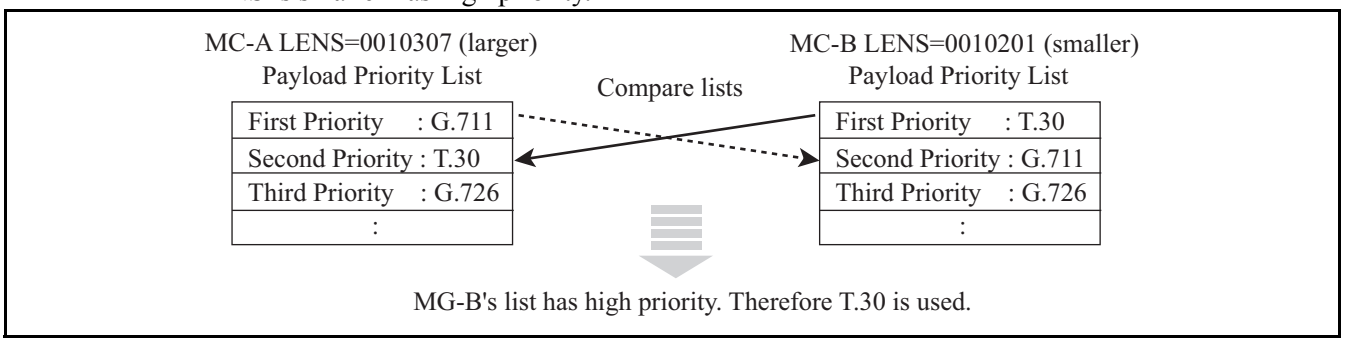


Command: AISTL

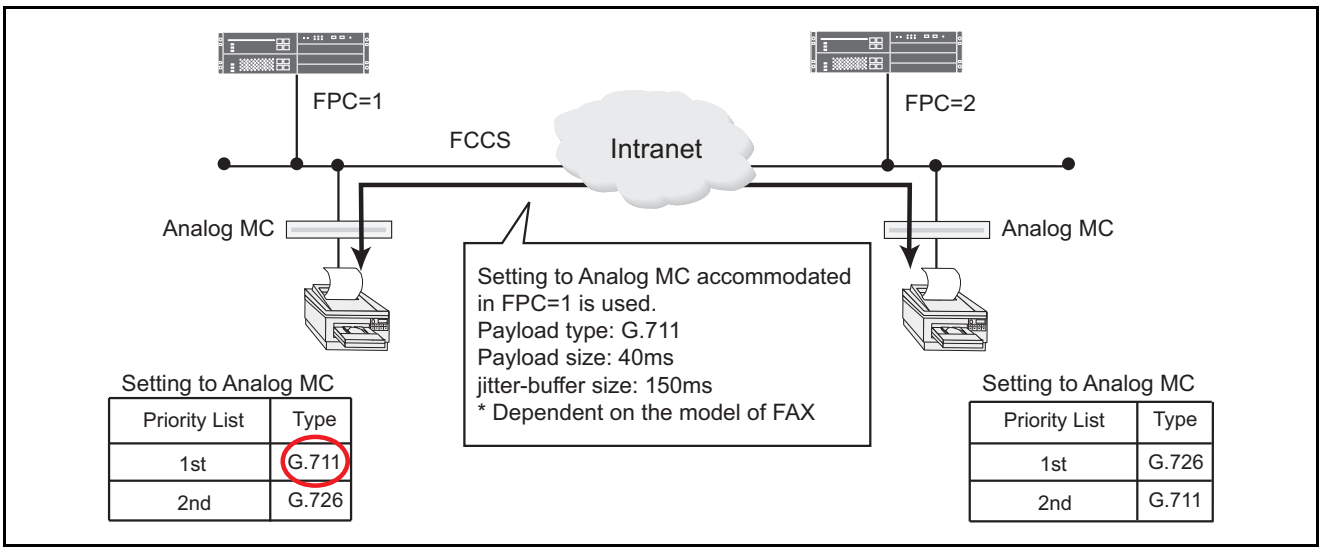
Setting Pattern	Setting Intent	Payload Type	Payload Size	Jitter-Buffer Size	Remarks
Recommended	Speed, Bandwidth Priority	T.30	-	-	Dependent on the model of FAX
Available	Connection Priority	G.711	40ms	150ms	-
Available	Bandwidth Priority	G.726	40ms	150ms	-

- Up to four priorities can be set by combining payload types and payload sizes (AISTL).
- When establishing FAX connection between MCs, specify the payload priority list and jitter-buffer size per MC port by AISTL.
- The payload type and payload size set in the both MCs' priority lists are used for the FAX connection between MCs.
- When the payload type and payload size set in the both terminals' priority lists are not found, G.711 is automatically selected and the data transmission as voice data (payload size=40ms, jitter-buffer size=150ms) is used.

- When there are differences in both terminals' payload priority list as shown below, MC whose LENS is smaller has high priority.



Example: FAX connection between IP Stations via FCCS over IP network



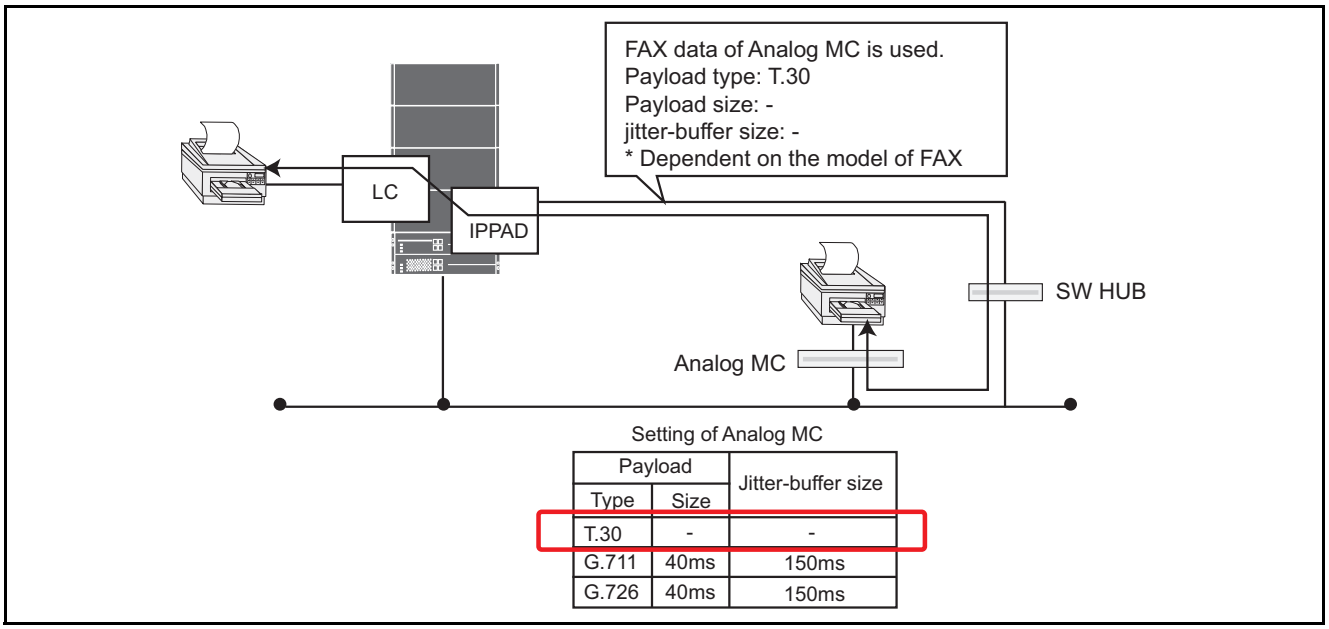
Command: AISTL

Setting Pattern	Setting Intent	Payload Type	Payload Size	Jitter-Buffer Size	Remarks
Recommended	Connection Priority	G.711	40ms	150ms	-
Available	Bandwidth Priority	G.726	40ms	150ms	-
Not available	Speed, Bandwidth Priority	T.30	-	-	-

Note: AISTL data set to MC accommodated in the node whose FPC is smaller has high priority.

(b) IP Station (MC) ⇔ IPPAD

[MC - IPPAD - TSW - LC (Legacy Station)]



Setting of Analog MC

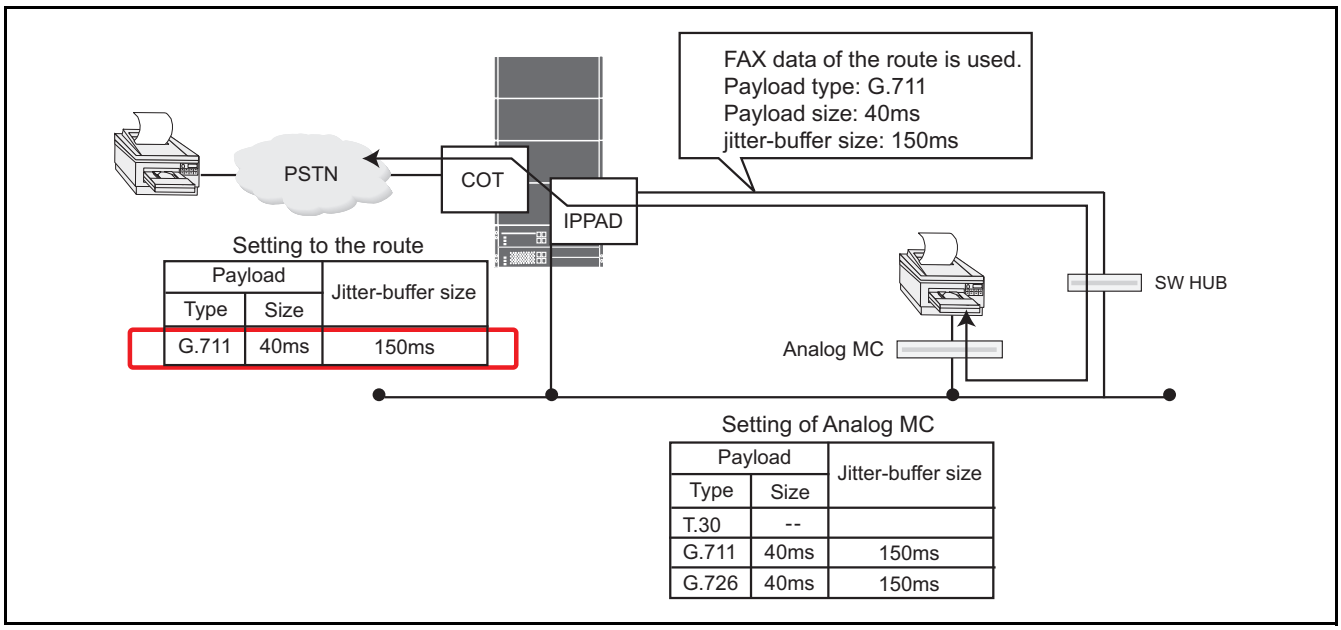
Payload		Jitter-buffer size
Type	Size	
T.30	-	-
G.711	40ms	150ms
G.726	40ms	150ms

Command: AISTL

Setting Pattern	Setting Intent	Payload Type	Payload Size	Jitter-Buffer Size	Remarks
Recommended	Speed, Bandwidth Priority	T.30	-	-	Dependent on the model of FAX
Available	Connection Priority	G.711	40ms	150ms	-
Available	Bandwidth Priority	G.726	40ms	150ms	-

[MC - IPPAD - TSW - External Trunk (Legacy Trunk)]

Example: FAX connection when COT circuit card is used as the External Trunk

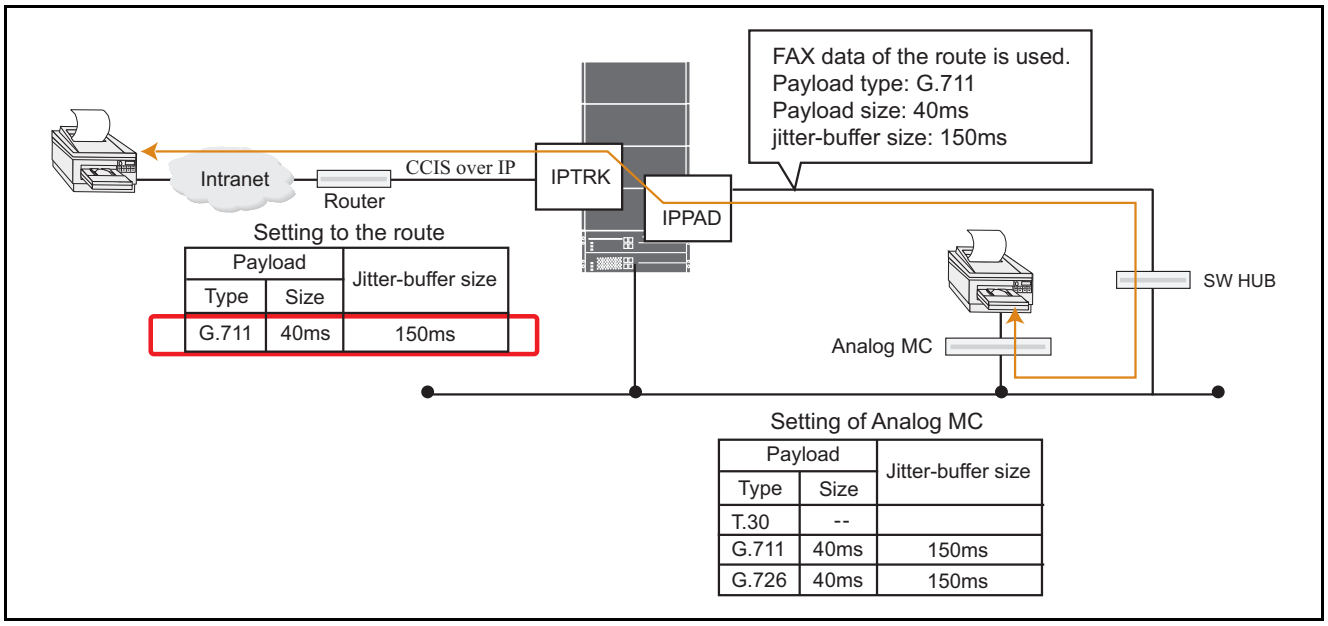


Command: ARTI/ARTIN, AISTL

Setting Pattern	Setting Intent	Payload Type	Payload Size	Jitter-Buffer Size	Remarks
Recommended	Connection Priority	G.711	40ms	150ms	-
Available	Bandwidth Priority	G.726	40ms	150ms	-
Available with restrictions	Speed, Bandwidth Priority	T.30	-	-	Dependent on the model of FAX (Available only when there is no FAX Relay on the external route.)

- The FAX data set to the external trunk route is used for the connection between MC and IPPAD.
- Even if the FAX data is set to the MCs, the FAX data set to the external trunk route is used.
- When no FAX data is set to the external trunk route, the FAX data set to the MC is used.
- Legacy external trunks are all the external trunks whose circuit cards are accommodated in PIR. IP-TRK (CCIS over IP) is also included.

Example: FAX connection when IPTRK (CCIS over IP) is used as External Trunk.

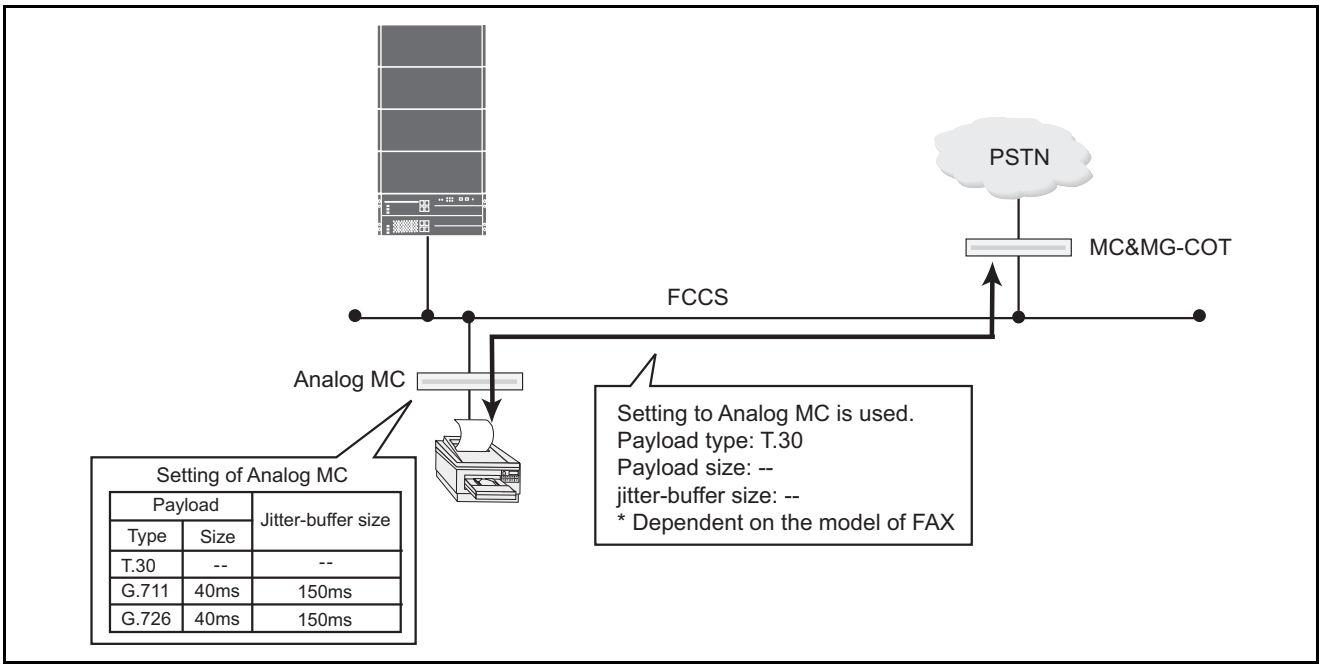


Command: ARTI/ARTIN

Setting Pattern	Setting Intent	Payload Type	Payload Size	Jitter-Buffer Size	Remarks
Recommended	Connection Priority	G.711	40ms	150ms	-
Available	Bandwidth Priority	G.726	40ms	150ms	-
Available with restrictions	Speed, Bandwidth Priority	T.30	-	-	Dependent on the model of FAX (Not available when there is FAX Relay on the IPTRK route.)

- The FAX data set per IPTRK route is used for the connection between MC and IPPAD.
- Even if the FAX data is set to MCs, the FAX data set to the IPTRK route is used.
- When IPPAD is connected to the external trunk having route number, the FAX data set per the route (by ARTI/ARTIN) is used.
- When IPPAD is connected to the Station, the FAX data set per MC is used.
- When there is no VCT on the IPPAD, or no FAX data is set to the IPTRK route, the data transmission as voice data (payload type = G.711, payload size=40ms, jitter-buffer size=150ms) is used.

(c) IP Station (MC) ⇔ MG



Command: ARTI/ARTIN, AISTL

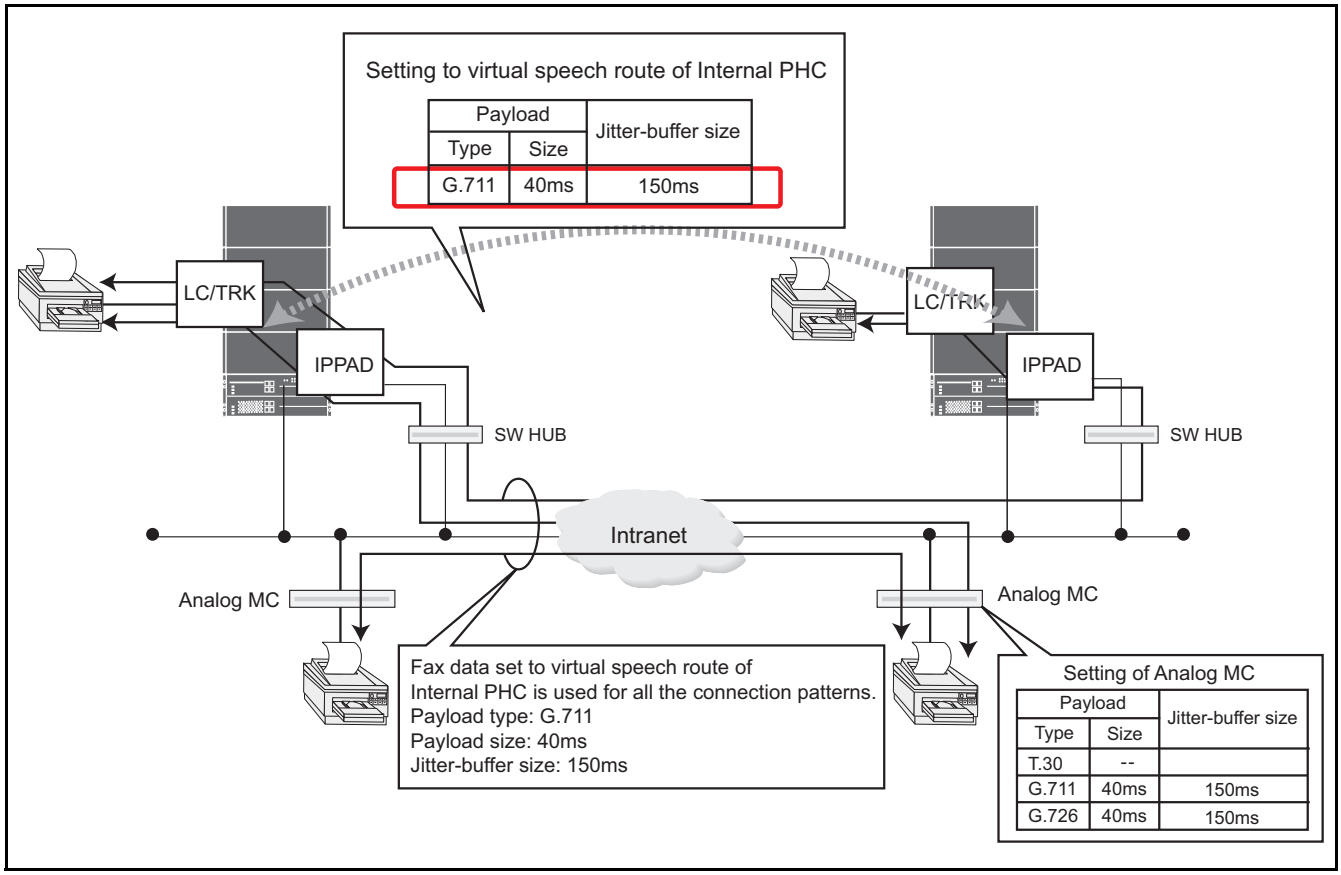
Setting Pattern	Setting Intent	Payload Type	Payload Size	Jitter-Buffer Size	Remarks
Recommended	Connection Priority	G.711	40ms	150ms	-
Available	Bandwidth Priority	G.726	40ms	150ms	-
Available with restrictions	Speed, Bandwidth Priority	T.30	-	-	Dependent on the model of FAX (Available only when there is no FAX Relay on the external route.)

- The above FAX data set to the external trunk route is used for the connection between MC and MG.
- Even if the FAX data is set to the MC side, the FAX data set to the external trunk route is used.
- When no FAX data (data set by ARTI/ARTIN command) is set to the external trunk route, the data transmission as voice data (payload type=G.711, payload size=40ms, jitter-buffer size=150ms) is used.
- Legacy external trunks are all the external trunks whose circuit cards are accommodated in PIR. IP-TRK (CCIS over IP) is also included.

(d) CCIS over IP by PHC

Example: FAX connection to Internal PHC in Counter-office

FAX data set to virtual speech route is used as payload type, payload size, and jitter-buffer size in CCIS over IP by PHC connection. When no data is set to the virtual speech route, the FAX data set to MC that communicates with the destination via CCIS over IP by PHC is not used. Instead of that, the data transmission as voice data is used.



Command: ARTI/ARTIN

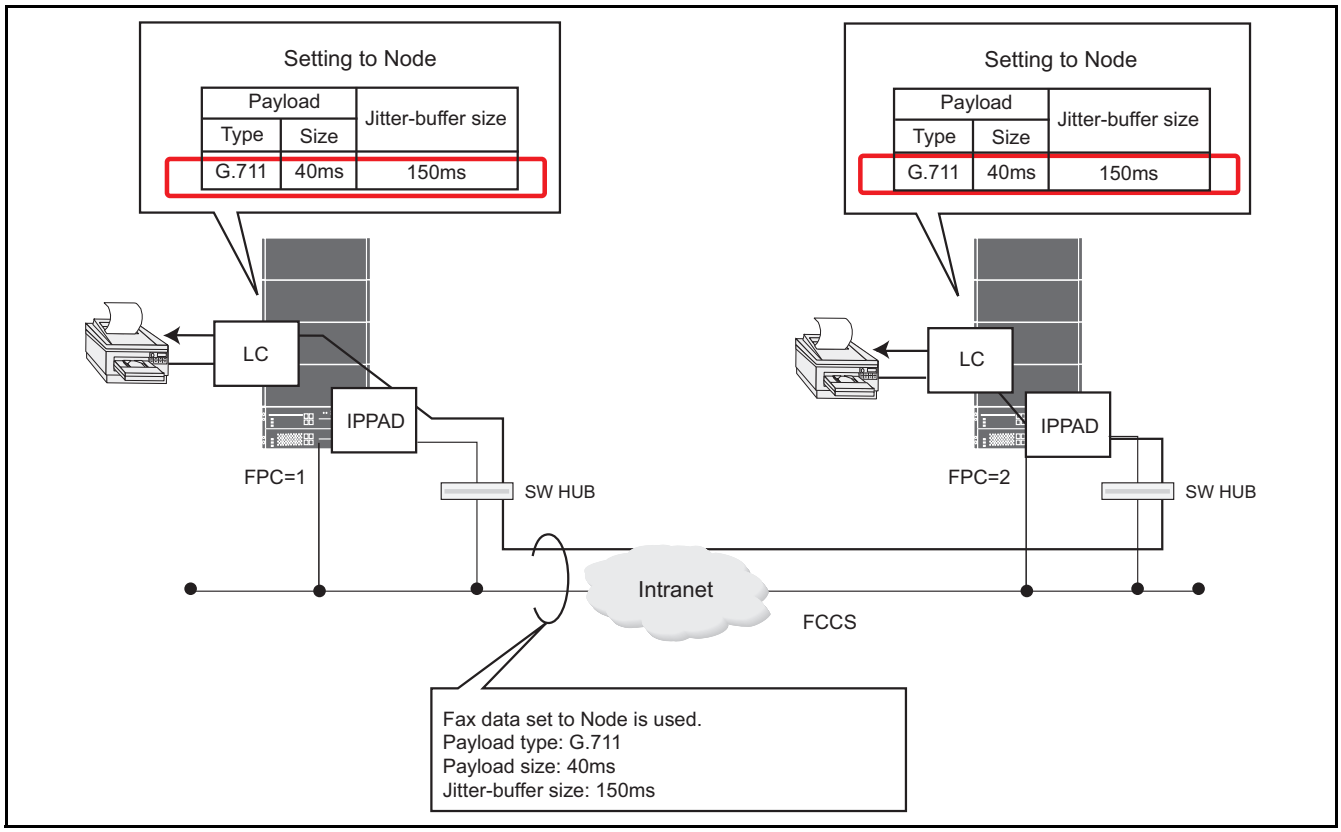
Setting Pattern	Setting Intent	Payload Type	Payload Size	Jitter-Buffer Size	Remarks
Recommended	Connection Priority	G.711	40ms	150ms	-
Available	Bandwidth Priority	G.726	40ms	150ms	-
Available with restrictions	Speed, Bandwidth Priority	T.30	-	-	Available only when there is no FAX Relay except PHC on the route for the FAX connection.

- The above FAX data set to each CCIS over IP by PHC virtual speech route is used for the FAX connection via CCIS over IP by PHC.

- Even if the FAX data is set to the sending and receiving MCs, the FAX data set to CCIS over IP by PHC route is used.
- When no FAX data is set to the CCIS over IP by PHC virtual speech route, the data transmission as voice data (payload type = G.711, payload size=40ms, jitter-buffer size=150ms) is used even if the FAX data is set to the sending and receiving MCs.

(e) Legacy Station/External Trunk ⇔ Legacy Station/External Trunk via FCCS over IP network

Example: FAX connection between Legacy Stations

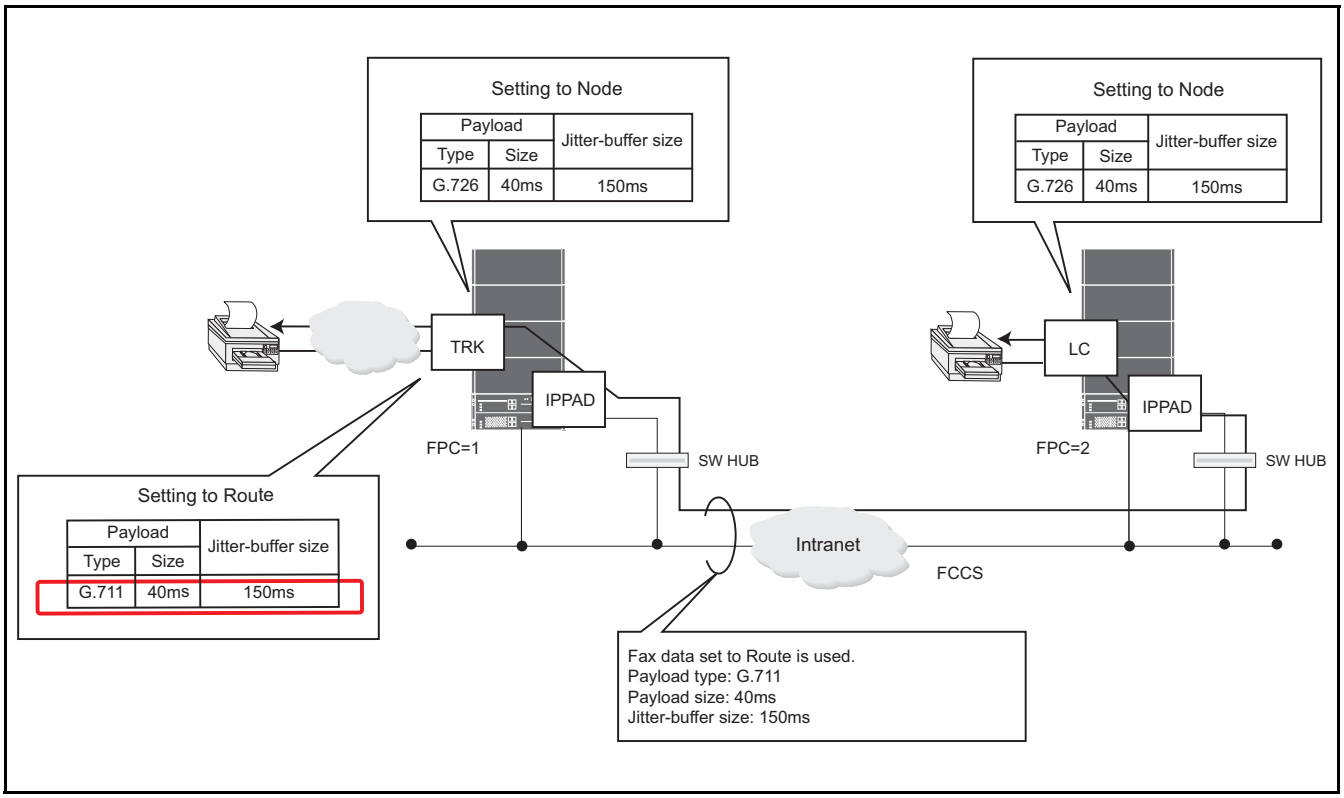


Command: ASYDL/ASYDN

Setting Pattern	Setting Intent	Payload Type	Payload Size	Jitter-Buffer Size	Remarks
Recommended	Connection Priority	G.711	40ms	150ms	-
Available	Bandwidth Priority	G.726	40ms	150ms	-
Available with restrictions	Speed, Bandwidth Priority	T.30	-	-	Dependent on the model of FAX (Available only when there is no FAX Relay on the external route.)

- When the FAX connection is established between Legacy Stations, system data set to each node is used. Set the same payload type, payload size and jitter-buffer size to the all nodes by ASYDL command.
- When no system data is set, the data transmission as voice data (payload type = G.711, payload size=40ms, jitter-buffer size=150ms) is used.

Example: FAX connection between Legacy Station and External Legacy Trunk



Command: ARTI/ARTIN

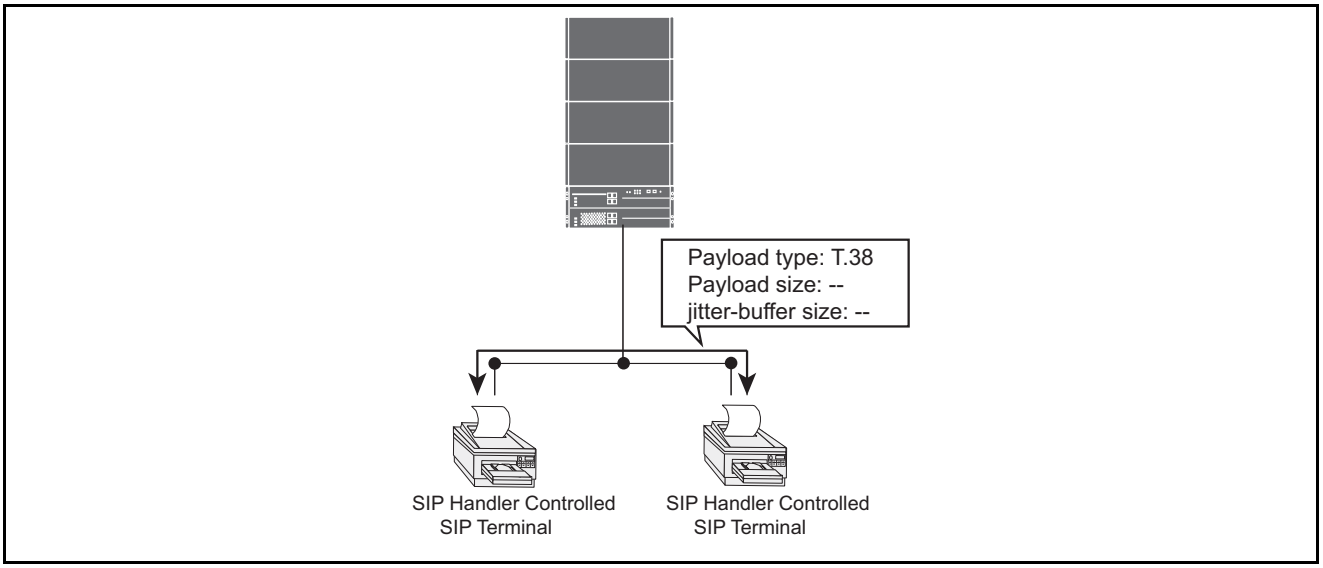
Setting Pattern	Setting Intent	Payload Type	Payload Size	Jitter-Buffer Size	Remarks
Recommended	Connection Priority	G.711	40ms	150ms	-
Available	Bandwidth Priority	G.726	40ms	150ms	-
Available with restrictions	Speed, Bandwidth Priority	T.30	-	-	Dependent on the model of FAX (Available only when there is no FAX Relay on the external route.)

- When no system data is set, the data transmission as voice data (payload type = G.711, payload size=40ms, jitter-buffer size=150ms) is used.
- When IPPAD without VCT is used, data transmission as voice data (jitter-buffer size=150ms, payload type=G.711, payload size=40ms) is used.

(f) SIP Handler Controlled Standard SIP Terminal ⇔ SIP Handler Controlled Standard SIP Terminal/ MG(SIP)

Note: Operations are not guaranteed for other features of SIP Handler Controlled SIP terminal than T.38 FAX data.

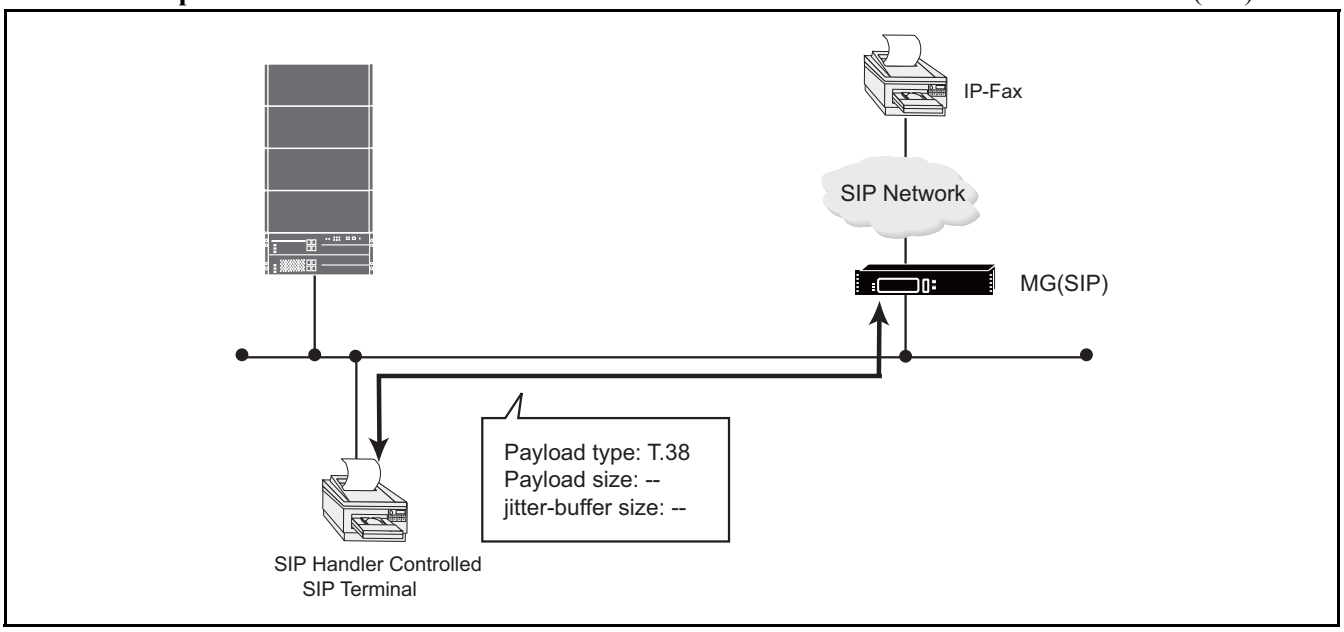
Example: FAX connection between SIP Handler Controlled Standard SIP Terminals



Command: AUACL/AUACN

- FAX data is set by enabling multimedia communication for SIP Handler Controlled Standard SIP Terminals.
- T.38 is the only possible payload type.
- SIP Handler Controlled Standard SIP Terminals must have software installed for T.38 FAX use.

Example: FAX connection between SIP Handler Controlled Standard SIP Terminal and MG(SIP)



Command: ARTI/ARTIN and AUACL/AUACN

- FAX data is set by enabling multimedia communication for SIP Handler Controlled Standard SIP Terminals and MG(SIP).
- T.38 is the only possible payload type.
- SIP Handler Controlled Standard SIP Terminals must have software installed for T.38 FAX use.

5. Conditions for FAX using T.38 payload type are as follows:
   a.) FAX data can be transmitted using only the following connection patterns:
      - SIP Handler Controlled SIP terminal and SIP Handler Controlled SIP terminal
      - SIP Handler Controlled SIP terminals and MG(SIP)

   b.) Telephony Server does not assign or control the payload type for FAX data. FAX data can be sent as long as both SIP Handler Controlled Standard SIP terminals use T.38 payload type. Terminals operations when the SIP Handler Controlled Standard SIP terminal does not have T.38 payload type are defined by each actual SIP Handler Controlled Standard SIP terminal's model specifications.

   c.) If UDP is used for FAX data, packets may be fragmented and congestion control is not available. For this reason, if there is the risk of UDP packets fragmentation, TCP setting (**Note 7**) is recommended.

   Note 7: When using TCP, make sure that all the terminals employed support TCP. Refer to the manual of each terminal for the TCP settings. See the <Interactions with Other Features> section of SIP HANDLER - SIP TERMINAL [S-167] for the conditions of TCP for Internal SIP Handler.

   d.) When using PEER TO PEER BANDWIDTH CONTROL [P-75], bandwidth of FAX data is not counted for the bandwidth calculation. Even if the FAX data is used as voice only and is using a codec supported by the Telephony Server, bandwidth will not be counted.

e.) SIP Handler Controlled Standard SIP terminals and devices used for FAX data must be compliant with the following RFCs:
- RFC 2327
- RFC 3261
- RFC 3262
- RFC 3264
- RFC 3311
- RFC 3323
- RFC 3515
- RFC 3550
- RFC 3891
- RFC 4028

f.) The SIP Handler Controlled Standard SIP terminal must have transmission capabilities for FAX data. A connection test performed by NEC is required before using a Standard SIP terminal (3rd party product).

g.) In a FCCS system FAX data can be sent only with FCCS NETWORKING VIA IP [F-36].

h.) CCIS NETWORKING VIA IP [C-163] of Data Programming Manual - CCIS cannot be used for FAX data. Station-To-Station connection after the Link Reconnection of CCIS networking via IP is also not supported for FAX data.

i.) Connection between two SIP Handler Controlled Standard SIP terminals or devices must be Peer-to-Peer.

j.) In case there is a failure in the session information inquiry, FAX data is possible based on the session information (**Note 8**) exchanged during the establishing of the call. If such session information exchanged during the establishing of the call also fails, FAX transmission is not possible.

Note 8: This session information exchange (proprietary protocol) is used for the media information (SDP information) exchange of the SIP Handler Controlled Standard SIP terminal (trough the Internal SIP Handler of the Telephony Server) and the MG(SIP). Session information inquiry is also included in the Session information exchange protocol.

k.) SIP Handler Controlled SIP terminal in Multiline Mode does not perform session information inquiry for a call received to a subline. However, FAX data is possible based on the session information exchanged during the establishing of the call.

l.) FAX data sending is not possible using the subline of a desktop terminal that has assigned the my line of a SIP Handler Controlled SIP terminal. In this case the desktop terminal can only maintain a voice communication.

m.) According to the session information inquiry for FAX data, the call sender notifies the receiver of the desired codec for FAX data with an SDP included in the INVITE. However, in the following cases the INVITE message cannot include an SDP:
- When receiving a blind transfer of REFER of a SIP Handler Controlled SIP terminal
- In the recalling after CALL HOLD [C-6] for a long time
- In the recalling of CALL BACK [C-1]
- A call received to the subline when being in Multiline Mode

Note: The reINVITE message send by the Telephony Server when switching the connection path may or may not include an SDP depending on the specific environment and other conditions.

However, the FAX data is possible in the cases above with the response of the receiving side based on the session information exchanged during the establishing of the call.

n.) The DTMF transparency conditions of FAX data sending are the same as the conditions of a voice call. For voice call conditions see <Hardware/Software> section of SIP HANDLER - SIP TERMINAL [S-167].
RFC2833, SIP-INFO and In-band Signaling can be used but the method is decided depending on the specifications of the actual SIP Handler Controlled Standard SIP terminal.

o.) In some cases, if a change in the media information using a reINVITE request is performed during FAX data sending, the SIP Handler Controlled Standard SIP terminal may reply with an error message regardless of the settings of CDN9 (Reject unacceptable reINVITE) in the AUACL/AUACN command.

Note: A change in the media information means a change in the codec, etc without changing the call state.

p.) Set 2nd DT: 0 (Second DT is not sent) in the ASPA/ASPAL/ASPAN command for the connection of SIP Handler Controlled Standard SIP terminals and MG(SIP) when using FAX data.

q.) MG(SIP) conditions are as follows:

- The following MG(SIP)s are supported:
  - MG-SIP128 [MG-SIP128 [MG-128SIPMGL-A]
  - Software-based MG-SIP with firmware version SP-4085 MGSIPVM PROG-E Issue 1 or later
- When MG-SIP SHARED ROUTE [M-123] is used, make sure that all the used MG(SIP) are usable for FAX data and have the same version. If different versions are used, even if the same access code is used, FAX data may become not possible depending on the seized trunk.
- When a SIP Handler Controlled Standard SIP terminal connected with MG(SIP) is put on hold, MG(SIP) does not send a hold notification to the IP network, but sends a hold tone.
- T.38 payload type only supports UDPTL protocol for packets sending.
- If the MG(SIP) does not have the suitable codec when receiving the reINVITE/UPDATE request from SIP network side, Error response assigned by the configuration command “set no_media_code” (**Note 9**) is sent back to the SIP network side.
- If the MG(SIP) does not have the suitable codec when receiving the reINVITE/UPDATE request from Telephony Server network side, 488 (Reject) response is sent back to the Telephony Server side.

  Note: The “set no_media_code” command is not used in the Telephony Server network side because the command is used to answer a request from the SIP network side.
- FAX data sending is available only when “enable” is assigned by the configuration command “set sip_tel_service”. (**Note 9**)

Note 9: For details of configuration commands, see the MG-SIP128 [MG-128SIP-MGG-B/MG-128SIPMGG] section of CHAPTER 2 in Peripheral Equipment Description (IP Devices).

6. Conditions of MG(SIP) usage for FAX communication, except in case of using T.38, are as follows:

a.) FAX communication can be established with G.711 assigned to payload for Voice and FAX.

However, regarding G.711, when the codec change feature and G.711 FAX Renegotiation feature are enabled, FAX communication can be established while a call is in progress with G.729a. For more details, refer to the command section (set codec_change) of each MG(SIP) in Peripheral Equipment Description (IP Devices). **Note 10**

b.) FAX communication can be established only when Payload Sizes for Voice and FAX are the same.

c.) To establish FAX communication with SIP network via MG-SIP, be sure to assign G.711 as Payload Type for all the FAX accommodated in the system.

Note 10: When using G.711, the following firmware is required for MG(SIP):

- MG-SIP16 [SCA-16SIPMG (US)] with SP-3988 MGSIP (16) PROG-G Issue 3 or later
- MG-SIP96 with SP-3987 MGSIP PROG-B Issue 3 or later
- MG-SIP128 [MG-128SIPMGG]/[MG-128SIPMGG-B]/ MG-SIP128 [MG-128SIPMGL-A]: any version
- Software-based MG-SIP: SP-4085 MGSIPVM PROG-E

Programming

- For MC Port

STEP 1: **ASYD** - SYS1, Index 64, Bit 0. G.711 Voice Compression CODEC Conversion μ–Law/A-Law Selection = 0/1

STEP 2: **AISTL** (Assignment of IP Station Data)

TN: Tenant Number [1-63]

KIND: Select Analog MC/MC-MG (COT)/8MC-SIP from the combo box.

UG50 Multi-slot mode: Check the checkbox to register UG50 in Multi-slot mode.

Check the FAX Setting Data box to assign the following FAX control data. If the check box is not checked, the default values will be applied.

JITSIZ: Specify Jitter Buffer size for FAX control [1-60] (The values from 1 to 60 indicate that 10 milliseconds to 600 milliseconds.)

PAYTYP1-4: Specify Payload Type for FAX control [G711, G726, or T30]

PAYSIZ1-4: Specify Payload Size for FAX control [10-40 milliseconds] (This data is required only when G.711 or G.726 is specified in the FXPT parameter)

Note: When deleting FAX IP data, follow the procedure below.
AISTL (Assignment of IP Station Data)
KIND: Select Analog MC from the combo box.
Uncheck the FAX Setting Data box to delete the existing FAX IP data. (FAX connection is established with the default values.)

To Delete: FAX data is deleted by unchecking the check box of “FAX setting data”. After the data is deleted, default FAX data is applied.

- For External Trunk (Legacy Trunk, MG, CCIS over IP by PHC virtual speech channel) Route

STEP 1: **ASYD** - SYS1, Index 64, Bit 0. G.711 Voice Compression CODEC Conversion μ–Law/A-Law Selection = 0/1

STEP 2: **ARTI/ARTIN** (Assignment of Trunk Application Data)

RT: Assign Route Number used for FAX connection [1-255]

CDN71: FXD: Enable/Disable IP FAX Data (FXJS, FXPT, FXPS) =0/1

CDN72: FXJS: Specify Jitter Buffer size for FAX control [1-60] (The values from 1 to 60 indicate that 10 milliseconds to 600 milliseconds.)

CDN73: FXPT: Specify Payload Type for FAX control [G.711, G.726, or T.30]

CDN74: FXPS: Specify Payload Size for FAX control [10-40 milliseconds] (This data is required only when G.711 or G.726 is specified in the FXPT parameter)

Note: To change the data for existing route (RT), initialize or reboot the trunk device such as circuit card or MG.

To Delete: FAX data is deleted by specifying "0" for the following data. After the data is deleted, default FAX data is applied.

ARTI/ARTIN command:
CDN71: FXD
CDN72: FXJS
CDN73: FXPT
CDN74: FXPS

- For connection between Legacy Stations/External Trunks via FCCS over IP network

STEP 1: **ASYD** - SYS1, Index 64, Bit 0. G.711 Voice Compression CODEC Conversion μ–Law/A-Law Selection = 0/1

STEP 2: **ASYDL/ASYDN** - SYS1, Index 652, Bit 0 to Bit 3. Payload Type for FAX control (0: -, 1: G.711, 2: G.726, 3: T.30, 4-15: Not used). *Assign Payload Type for FAX in hexadecimal format. (Ex: G.726.1 → 02 [hex]) **Note 11**

SYS1, Index 653, Bit 0 to Bit 4. Payload Size for FAX control (0: -, 1: 10ms, 2: 20ms, 3: 30ms, 4: 40ms, 5-31: Not used) *Assign Payload Type for FAX in hexadecimal format. (Ex: 30ms → 03 [hex]) **Note 12**

SYS1, Index 654, Bit 0 to Bit 7. Jitter Buffer size for FAX control (1-30 **Note 13, Note 14**) *Assign the Jitter Buffer size in hexadecimal format. (Ex: 10ms → 01, 100ms → 0A, 200ms → 14, 300ms → 1E)

Note 11: When ASYDL/ASYDN, SYS1, Index 652=0 is set, "payload type: G.711" and "payload size: 40ms" are applied as default data. (Even if payload size is set by Index 653, payload size is fixed to 40ms.)

Note 12: Assign this data when ASYDL/ASYDN, SYS1, Index 652=1 (G.711) or 2 (G.726) is set.
When Index 652=0 is set, default data "payload size: 40ms" is applied even if payload size is set by this parameter.
When Index 652=1 (G.711) or 2 (G.726) is set and Index 653=0 is set, default data "payload size: 40ms" is applied.

Note 13: The values from 1 to 30 indicate that 10 milliseconds to 300 milliseconds.

Note 14: When the value 0 is set,"150 milliseconds" is set as default data.

To Delete: FAX data is deleted by specifying "0" for the following data. After the data is deleted, default FAX data is applied.

ASYDL/ASYDN command:
Index 652, Bit 0 to Bit 7=0
Index 653, Bit 0 to Bit 7=0
Index 654, Bit 0 to Bit 7=0

- For connection between SIP Handler Controlled Standard SIP Terminals or with MG(SIP)

Note: FAX data assignment is done after finishing the basic programming for the used terminals and devices.
For SIP Handler Controlled SIP Terminal, see data programming in SIP HANDLER - SIP TERMINAL [S-167].
For MG(SIP), see data programming in MEDIA GATEWAY SIP [M-111].

<For SIP Handler Controlled SIP Terminal>

STEP 1: **ASYDL**- Assign the following.
Index 1394, Bit 0 and 1: Select transport protocol for session information exchange of SIP devices.**Note 15 Note 16**

Transport Protocol	Bit 0	Bit 1
AUTO **Note 17**	0	0
TCP **Note 18**	1	0
UDP	0	1

Note 15: If FCCS system is used, set these bits in the ASYDN to ensure that all nodes have the same settings.

Note 16: in the FCCS system and UMGi system, this setting is valid for session information exchanges of Internal SIP Handlers between nodes. In other combinations, UDP is used.

Note 17: With this setting the packets are sent using UDP, but if the size exceeds 1400 bytes TCP is automatically used.

Note 18: If there is the risk of packet fragmentation when using UDP, set to TCP.

Index 1408, Bit 0: Use TCP for SIP Messages.
0= Disabled
1= Enabled

Note: If this setting is changed, Internal SIP Handler needs to be initialized with the ADPM, ADTM or SINZ command.

Note: Since TCP for big messages is recommended, set this bit to Enabled is also recommended.

STEP 2: **AUACL/AUACN** - Assign the following.
CDN 20 (Multimedia Communication): 1 (Enable)
CDN 21 (Hold Detection at Multimedia): **Note 19**
0: Audio
1: Audio and Video

2: Non-detection
CDN 22 (Transport Layer) **Note 20**

0: AUTO
1: UDP
2: TCP

Note: Verify that ASYDL, SYS1, Index 1382, Bit 2 =0.

Note 19: Please note the following points:

- This setting requires CDN 20 to be set to 1.
- For SIP Handler Controlled Standard SIP video terminals that notify media changes as the voice being muted while the video communication is kept, select pattern 1 or 2.
  For SIP Handler Controlled Standard SIP video terminals that do not notify media changes as the voice being muted, select pattern 0 (default).For a terminal without video capacity, even if 1 is set, operations will be the same as if 0 is set.
- A change in the parameter is applied when there is call sending/receiving.

Note 20: Please note the following points:

- This setting does not decide the transport layer for the terminals. What is set is the protocols used by the SIP Handler of the Telephony Server when assisting the communication between the terminals.
- This setting requires the TCP setting to be enabled. If TCP is not enabled (because system data settings or other reason), TCP for the sending and receiving of SIP messages is not possible.
- This setting is used only if the transport protocol is not set in the transport parameter of the Contact header in the REGISTER request. If the transport protocol is set in the REGISTER, that protocol is used regardless of the setting of CDN22.
- The default setting uses TCP for sending SIP messages of more than 1300 bytes or answering to TCP request messages (regardless of the size of the answering message).
- If UDP is selected, the SIP Handler sends the request messages using UDP (be aware that long messages may become fragmented). However, the SIP Handler is also able to process messages received with TCP and any message received with TCP is also answered with TCP regardless of this parameter setting.
- If TCP is selected, the SIP Handler uses TCP for sending messages but is able to process received messages with UDP.
- A change in the parameter is applied when there is call sending/receiving.

<For MG(SIP)>

STEP 1: **ARTI/ARTIN** - Assign the following.
CDN 110 (Multimedia Communication): 1 (In Service)

Note: Regarding the number plan (LCR/LCRS) of the MG(SIP) sending connection, set 2nd DT: 0 (Second DT is not sent) with the ASPA/ASPAL/ASPAN command.

I-45 Individual Call Hold Recall Timer

(Series 8400 software feature)

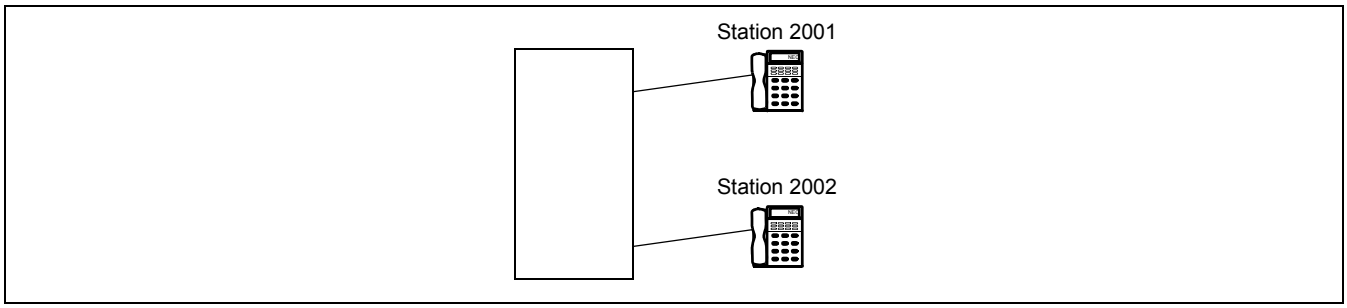
General Description

This feature provides the Desktop terminal with the ability to individually set the options for the Call Hold Recall Timer setting.

Note: This feature is only available in North America.

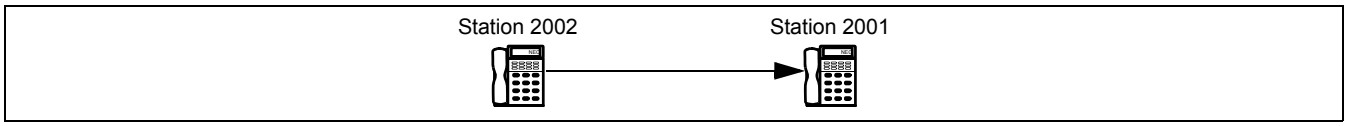
Operating Procedure

1. Station 2001 places or receives a call and presses the HOLD key. After Call Hold Recall Timer Individual expires, Station 2001 is recalled.

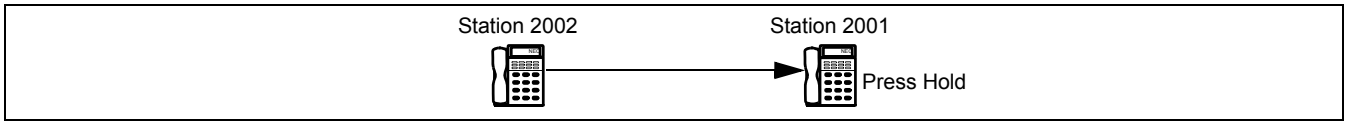


Desktop terminal Procedure for 30-second Timer

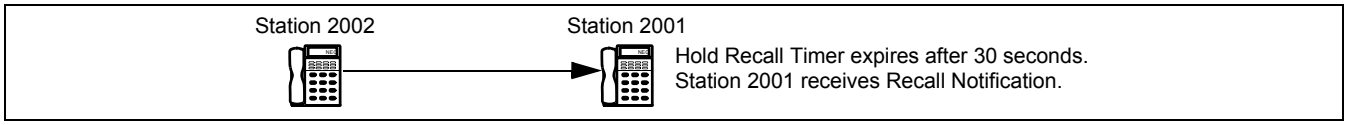
1. The ASYD, SYS1, Index 144 is set to 10 seconds.
2. The ADAI command has 144 MTC set to "F", 144 TC set to "3" (30 seconds), and 144 RA set to "0".
3. Station 2001 receives and answers a call from Station 2002.



4. The User at Station 2001 presses the HOLD key and places the call on Hold.

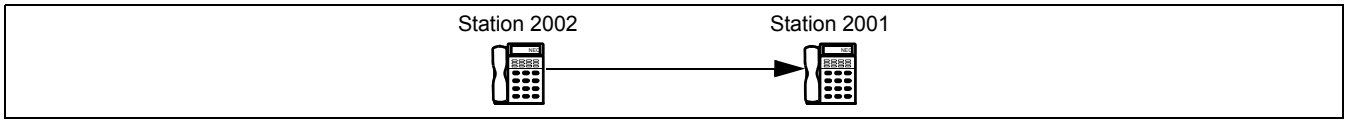


5. The ADAI setting of 30 seconds takes priority over System Data (ASYD, SYS1, Index 144) setting of ten seconds. Station 2001 receives Recall Notification in 30 seconds.

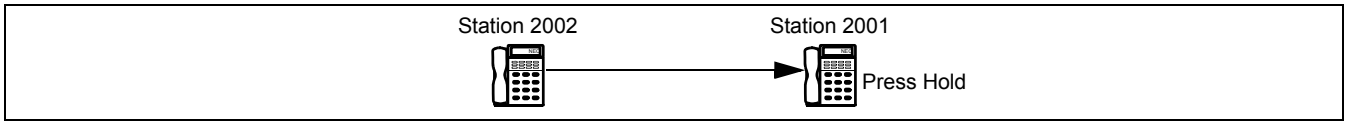


Desktop terminal Procedure for 60-second Timer

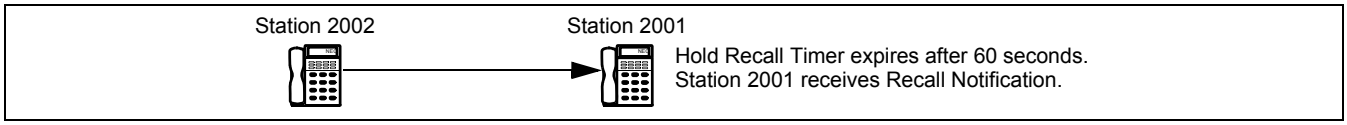
1. The ASYD, SYS1, Index 144 is set to 10 seconds.
2. The ADAI command has 144 MTC set to "8", 144 TC set to "7" (60 seconds), and 144 RA set to "0".
3. Station 2001 receives and answers a call from Station 2002.



4. User at Station 2001 presses the HOLD key and places call on Hold.

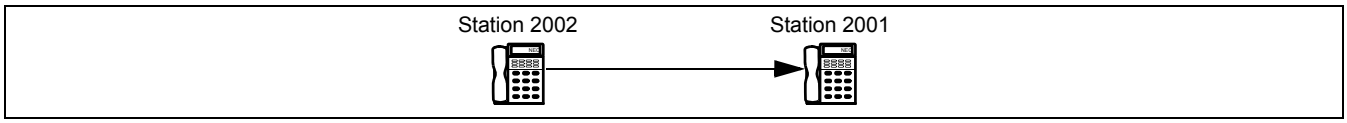


5. The ADAI setting of 60 seconds takes priority over System Data (ASYD, SYS1, Index 144) of ten seconds. Station 2001 receives Recall Notification in 60 seconds.

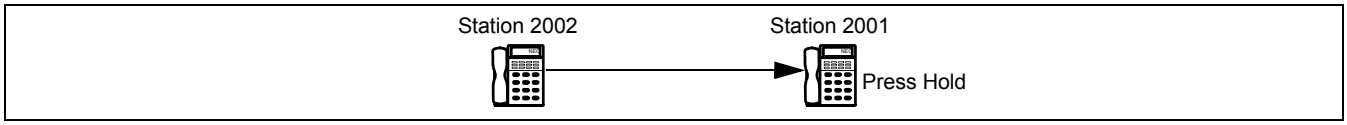


Desktop terminal Procedure for no Timer

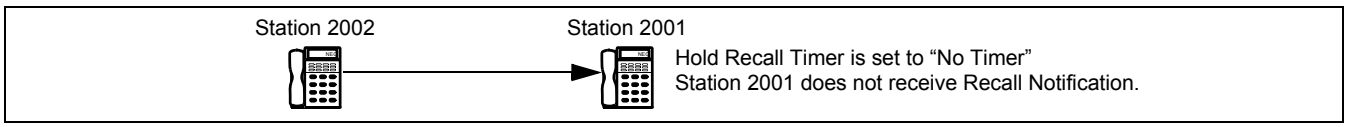
1. The ASYD, SYS1, Index 144 is set to 10 seconds.
2. The ADAI command has 144 MTC set to "0", 144 TC set to "0", and 144 RA set to "1" (no Timer).
3. Station 2001 receives and answers a call from Station 2002.



4. The User at Station 2001 presses the HOLD key and places the call on Hold.



5. The ADAI setting of no Timer takes priority over System Data (ASYD, SYS1, Index 144) of ten seconds. Station 2001 does not receive Recall Notification.



Service Conditions

1. This feature is only be available for a Desktop terminal.
2. Pressing the HOLD key on the Desktop terminal accesses the feature.
3. Without the ADAI command, the feature uses the System Data (ASYD, SYS1, Index 144).
4. This feature is not available for an Analog Station.
5. The use of an Access Code is available on both an Analog Station and a Desktop terminal.
6. When using an Access Code, both an Analog Station and a Desktop terminal use System Data Index 139.
7. When ADAI Recall Timer is set, it takes priority of the setting in ASYD.
8. The 144 RA parameter in the ADAI command is for a “No Recall” option.
9. The 144 RA parameter in ADAI will only override the System Data (ASYD, SYS1, Index 144).

Programming

To assign

Station 2000 calls Station 2001. Hold Recall Timer is set to 3F, which overrides System Data (ASYD, SYS1, Index 144).

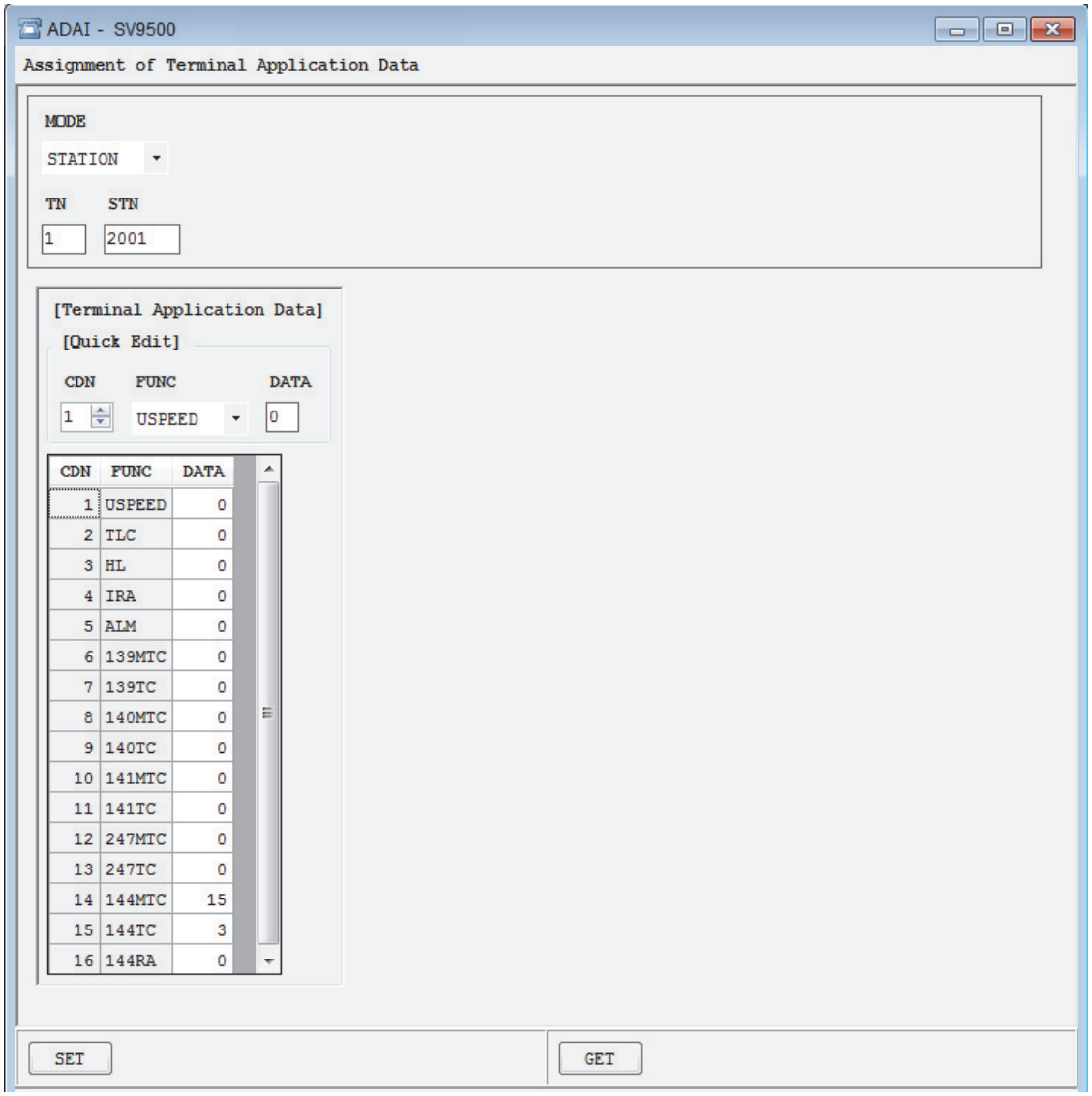
ADAI - Assign the following data.

Mode: Station

TN: 1

STN: 2001

See the following example:



To delete

ADAI - Assign the following data.

Delete:
Mode: Station
TN: 1
STN: 2000
DEL?: Y

I-46 Individual Call Block

(Series 8800 software feature)

General Description

This feature allows up to 1023 LENS per IMG the ability to set 16 individual Call Block bins (up to 16 digits each). Individual Call Block can be turned off or on for all Call Block Numbers by means of an Access Code, without the need to delete the Call Block bins.
Individual Call Block allows a Trunk Caller to be routed to a common announcement instead of a Busy Tone.
In addition, the last number received feature allows the last calling number received to be saved without dialing the Block Number or Individual Calling Number (only one number can be saved in this manner).

Note: This feature is only available in North America.

Operating Procedure

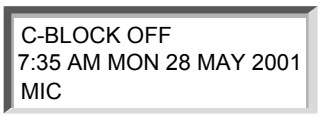
To turn on Individual Call Block

1. Lift the handset and dial the Call Block Active On Access Code "SSCA SIDA 126).
2. Wait for the Service Set Tone.
3. Replace the handset.
4. The Desktop terminal LCD displays:



To turn off Individual Call Block

1. Lift the handset and dial the Call Block Active Off Code Access Code (SSCA SIDA 127).
2. Wait for Service Set Tone.
3. Replace the handset.
4. The Desktop terminal LCD displays:



To set Call Block Restriction DC to a bin

1. Lift the handset and dial the Call Block Set Access Code (SSCA SIDA 129).
2. Receive Stutter Dial Tone.
3. The Desktop terminal displays:

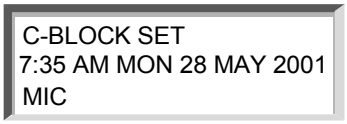


4. Dial the individual Bin Number 00-15, receive Special Dial Tone.

   Note: Call Block last number received is automatically saved to Block Bin 00.

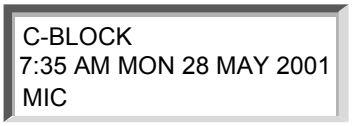
5. Dial the exact digits to be blocked (a maximum of 16 digits).

6. Wait for Service Set Tone.
7. Replace the handset.
8. The Desktop terminal LCD displays:

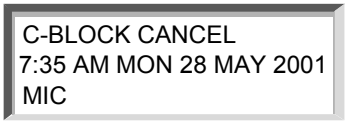


To delete Call Block Restriction DC from a bin

1. Lift the handset and dial the Call Block Set Access Code (SSCA SIDA 129).
2. Receive Special Dial Tone.
3. The Desktop terminal LCD displays:



4. Dial the individual Bin Number 00-15, receive Dial Tone.
5. Wait for Service Set Tone.
6. Replace the handset.
7. The Desktop terminal LCD displays:

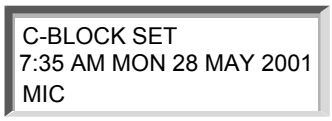


To set Call Block DC for last number received

1. Lift the handset and dial the Call Block Last Number Set Access Code (SSCA SIDA 128).
2. Receive Service Set Tone.
3. Replace the handset.

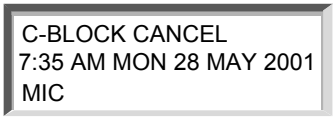
Note: Block Bin 00 is used for setting the last number block received feature. Block Bin 00 will be overwritten each time the set Call Block DC for last number received is dialed.

4. The Desktop terminal LCD displays:



TO remove Call Block DC for last number received

1. Lift the handset and dial the Call Block DEL Access Code plus “00” (SSCA SIDA 129).
2. Wait for Service Set Tone.
3. Replace the handset.
4. The Desktop terminal LCD displays:

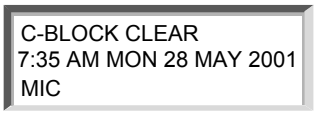


To delete all Call Block Numbers

1. Lift the handset and dial the Call Block Delete All Access Code (SSCA SIDA 130).
2. Wait for Service Set Tone
3. Replace the handset.

Note: This procedure deletes all Block Bins 00-15.

4. The Desktop terminal LCD displays:



Service Conditions

1. Individual Call Block is mutually exclusive to Call Block [C-150] by Service Feature Restriction Class. Service Feature Restriction Class 165 must be restricted (SFI 165 = 0) in order to use the Individual Call Block feature. Stations using the Individual Call Block enhancements must delete all Call Block [C-150] entries. If the blocked numbers entered by Call Block [C-150] are not deleted, unexpected results will occur. See feature Call Block [C-150] for complete deletion instructions.
2. Individual Call Block is programmable to be automatically active or inactive by means of ASYDL (See the Programming section).
3. Individual Call Block can only be set/canceled/activated/de-activated by an individual station using the appropriate Access Code. A PCPro command is not available for the entry or deletion of Block Numbers for Individual Call Block.
4. Individual Call Block bin contents can be displayed by the PCPro using the DICBD command.
5. This feature is only available for stations with Service Feature Restriction Released (SFI 165 = 1).
6. The Station User is allowed to overwrite any Individual Call Block List Number.
7. Individual Call Block can support up to 1023 Individual Call Block Lists per IMG.
8. Only one Individual Call Block List is allowed per LENS with a maximum of 1023 lists/LENS per LP.
9. Individual Call Block enhancements can be used with any trunk supplying a valid Calling Party Number Identification or ANI to the System (Caller ID/ISDN or MF).
10. Individual Call Block will only be effective when the Incoming Trunk supplies a Calling Party Identification Number. If a Calling Party Number is not received, the Anonymous Call Block feature must be used to block the Incoming Call.
11. When the System is initialized with data memory load, the Call Block feature will be in the last state saved to the System hard drive.
12. Call Forwarding - All Calls [C-5] and Call Forwarding - Busy Line [C-2] take precedence over this feature.

13. If Call Forwarding - Busy Line is set and the Incoming Calling Number is restricted by Individual Call Block, the Incoming Call will follow Call Forwarding - Busy Line regardless of the station being idle or busy.
14. Dterm Stations with addition Line Appearances must program each Line Appearance with the number to be blocked (for example, Dterm Station 1000 has Line Appearances of 1001 and 1002 appearing on its phone. To restrict a specific Calling Party Number the Station User 1000 must program Lines 1000, 1001 and 1002 with the number to be restricted).
15. Station-to-Station Calls in the same Local Node or in the same FCCS Network as the Blocking Station can be restricted using Individual Call Block. However, any override feature (such as (but not limited to) Executive Right-of-Way [E-1], Privacy Release [P-18] and Call Waiting - Terminating [C-12]) will take precedence over this restriction.
16. Dterm Stations blocked via Individual Call Block will receive Reorder Tone and display either DND or RST depending on Service Feature Restriction Class programming.
17. Single-line Stations restricted by Individual Call Block will receive Reorder Tone.
18. When no announcement is provided: ASYD, SYS1, Index 59, Bit 2=1 must be set for a MF DID to receive Reorder Tone. If this bit is not set, no indication is sent to the trunk that the call has been blocked.
19. Dterms with addition Line Appearances must program each Line Appearance on its set to be restricted. However, the Call Block Number entered will apply for all Line Appearances of that number.
20. A Virtual Station Appearance can have Individual Call Block set. However, the Call Block Number entered will apply for all LINE APPEARANCEs of that Virtual Station.
21. When a Station or Virtual Line is deleted, the Call Block List will be automatically removed.
22. Individual Call Block Announcement Service is not available for Station-to-Station Blocked Calls.
23. Individual Call Block restrictions are not available for calls transferred by the Attendant Console or a station.
24. Individual Call Block enhancements are not available for the Attendant Console (RT 901), Night Attendant (RT 915), BRI or Wireless Terminals, even if Individual Call Block Service Feature Restriction Class 165 is allowed.
25. Internal Telephony Server Stations using Individual Call Block cannot restrict Incoming Calls from the System Operator (RT 901 or 915).
26. Direct Incoming Trunks (DID, ISDN, DIT) that are set for Call Forwarding - All Calls or Call Forwarding - Busy Line to a station that has restricted the Calling Party Number will receive Busy Tone (or if available, the Individual Call Block Enhancement Announcement).
27. The Call Block last number received function will always enter in to Bin 00 the last valid Calling Party Number received. In the case of Call Waiting - Terminating [C-12], the Call Waiting Call must be answered and have a valid Calling Party Number in order to be restricted using the Call Block last number received function.
28. Individual Call Block last number received function is not available for Attendant Camp-on With Tone Indication [A-1] or Attendant/Station Transferred Calls, regardless of a valid Calling Party Number.
29. After a System initialization (with or without data memory load) the Call Block last number received feature cannot be used by a station, until an Incoming Call with a valid Calling Party Number is received.
30. Individual Call Block Bin Numbers are from 00 to 15. If an invalid Block Number is entered, Reorder Tone is returned.
31. Call Pickup - Group [C-7], Call Pickup - Direct [C-30], Call Hold Pickup [C-154] and Call Pickup - One Touch key [C-207] take precedence over this feature.

32. Individual Call Block will not function if all stations programmed in a UCD Group have blocked the Incoming Calling Party Number. The call will remain in the UCD queue. Busy Tone or Call Block announcement is not available. UCD Overflow is available providing the Overflow Stations have not set individual Call Block to the Calling Party Number.
33. Pilot and Circular Hunt take precedence over this feature. If all members of the Hunt Group have restricted the Calling Party Number, Busy Tone or individual Call Block Enhancement Announcement is returned.
34. A DIT Trunk that has been restricted from Incoming Calls by means of Individual Call Block can reroute to the Attendant Console by system programming.
35. A DIT Trunk Call that is blocked and is not otherwise rerouted will continue to receive Ringback Tone unless an announcement (AAED/N Equipment Number 63) is available. Tandem Trunk Connection must be allowed between the DIT Trunk Route and the Announcement Trunk Route.

    Note: ASYD, SYS2, Index 11, Bit 7 setting to the Attendant Console will take precedence over the announcement.

36. Individual Call Block takes precedence over Slumber Time - Do Not Disturb [S-106].
37. Individual Call Block enhancements cannot be set in a Consultation Hold condition.
38. Call Block announcement must be set for Multiple Connection to announcement by means of switch setting options on the DAT card. Callers cannot be guaranteed to start from the beginning of the message.
39. A Dterm cannot display a Call Block Bin List.
40. When a SIP Handler Controlled SIP terminal is in Multi Line Mode, Individual Call Block is set to the seized line. Normally, Individual Call Block is set to My Line if it is not used by another station.

Programming

To assign Individual Call Block

STEP 1: **ANPD/ANPDL/ANPDN** - Assign a number level for Individual Call Block.

STEP 2: **ASPA/ASPAL/ASPAN** - Assign the following Access Codes for CI = "N: Normal"
Call Block List On:SRV = "SSCA: Service Code Appendix", SIDA = 126
Call Block List Off:SRV = "SSCA: Service Code Appendix", SIDA = 127
Call Block Set: SRV = "SSCA: Service Code Appendix", SIDA = 129
Call Block Last Number Set:SRV = "SSCA: Service Code Appendix", SIDA = 128
Call Block - Delete All:SRV = "SSCA: Service Code Appendix", SIDA = 130

STEP 3: **ASYDL** - Index 804, Bit 5 - Individual Call Block automatic active 0/1 = On/Off.

STEP 4: **ASFC** - Assign a Service Feature Restriction Class to allow Call Block [C-150] for stations (SFI 192, RES = 1 to release restriction).

STEP 5: **AAED/N** - Assign Announcement Equipment Number 63 to route Call Block Calls to an announcement.
"R" must be set to "0" to send ring back. If "R" is set to "1", ISDN will not operate with the announcement.

STEP 6: **ARRC/N** - Allow Incoming Routes to Announcement Routes.

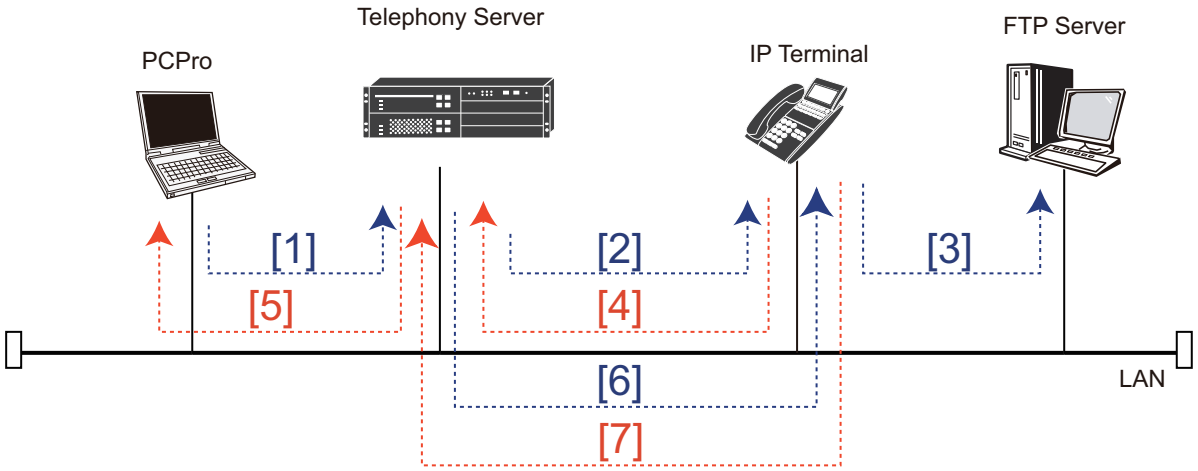
To delete Individual Call Block

The deletion procedure is the reverse order of assigning the feature.

I-47 IP Device Firmware Remote Download

General Description

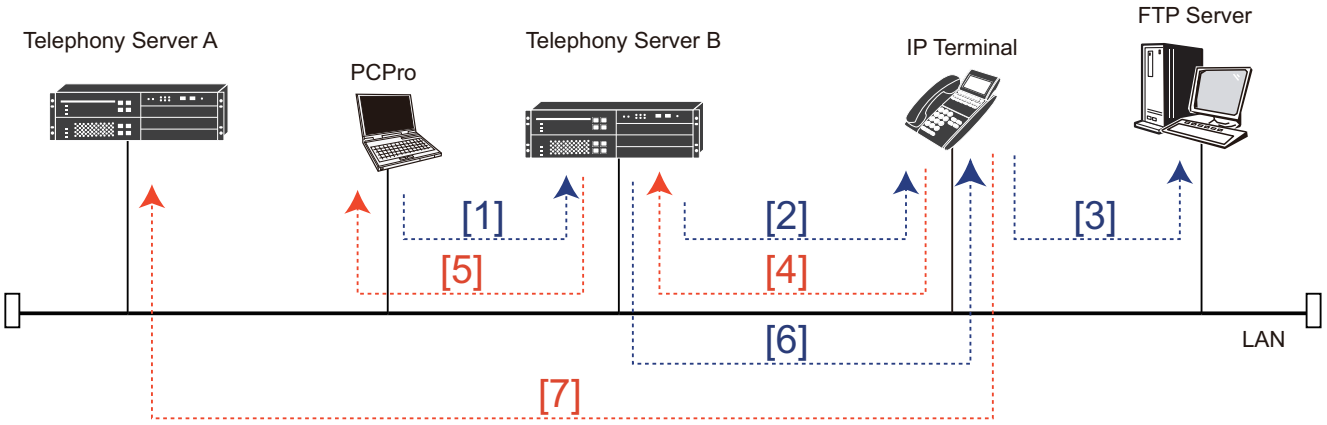
This feature allows IP terminals to update their firmware by downloading new firmware file. At this time, the downloading can be performed manually at any time or automatically at a predetermined time.
After downloading a new firmware file, IP terminals will perform reregistration to the Telephony Server with the new firmware.



This feature is activated through the following steps.

1. The PCPro specifies the terminal and sends the downloading order to the Telephony Server.
2. The Telephony Server orders the terminal to download a new firmware file.
3. The new firmware file on the FTP server (**Note 1**) is downloaded to the assigned terminal dependent on the parameters defined via the PCPro.
4. The terminal informs the Telephony Server of the download state.
5. The Telephony Server informs the PCPro of the download state and the PCPro displays it.
6. The Telephony Server requires the terminal to restart.
7. The terminal is reset and registered to Telephony Server with its new firmware.

It is also possible to download a config file (DRS setup file). After downloading the file, the IP terminal that has been set with the new config file is registered to the Telephony Server.



1. The PCPro specifies the terminal and sends the downloading order to the Telephony Server.
2. The Telephony Server orders the terminal to download a new config file.
3. The new config file on the FTP server (**Note 1**) is downloaded to the assigned terminal dependent on the parameters defined via the PCPro.
4. The terminal informs the Telephony Server of the download state.
5. The Telephony Server informs the PCPro of the download state and the PCPro displays it.
6. The Telephony Server requires the terminal to restart.
7. The terminal is reset and registered to the Telephony Server with its new config file.

For convenience, firmware and config files for IP terminals are hereinafter called “download files”.

Note 1: In case of using an external PC as the FTP server, set the proper file permissions.

Operating Procedure

Be sure to register the following two settings for the FTP server:

1. Store the download file of the IP terminal on the FTP server.

   Store the download file on the FTP server. NEC recommends using the default directory that the users have access to when they are logged in to the FTP server. When using the Telephony Server as the FTP server, log in from a terminal on which FTP client software is installed and upload the download files with the “put” command.

2. Register the FTP user account of the IP terminal to the FTP server.

   Register the FTP user account of the IP terminal to the FTP server.
   a.) When using the Telephony Server as the FTP server, register the FTP user account on “Maintenance User Account Settings” of the ADTM command.
   b.) When using External FTP server, register the FTP user account on the external server.

   (Example: For Microsoft® Windows® XP/Vista)

1. Select “Start” - “Control Panel” - “Administrative Tools” - “Computer Management.”
2. Right-click on the “Users” under the “Local Users and Groups”.
3. Select “New User...” on the menu to display the “New User” screen.
4. Assign the necessary data for the FTP User Account.

Note: For other External FTP Server programs, refer to the documentation for that software when setting up the Login Account.

Service Conditions

1. The following are necessary in FTP server:
   - Downloading via TFTP is not supported.
   - Download file must be put in an appropriate location.
   - The FTP user account for IP terminals must be registered.

2. When using Automatic Download by specifying a predefined time.
   - The terminal must be Peer-to-Peer type. (Dterm with IP adapter for IPELC are not supported.)
   - The terminal has been registered in the System in advance.
   - Firmware update information is specified correctly in ATDL command.
   - The terminal coincides with the terminal information list in the ATDL command.
   - Download time is set in ATDL command.
   - The terminal is connected via LAN from the download start time until it has completed. During this time, the LAN connection and the power to the terminal cannot be interrupted.

3. When using Manual Download by individual setting
   - The terminal must be Peer-to-Peer type. (Dterm with IP adapter for IPELC are not supported.)
   - The terminal has been registered in System in advance.
   - Firmware update information is specified correctly in ATDL command.
   - The terminal coincides with the terminal information list in the ATDL command.

4. When downloading different types of files, create a separate directory for each download file that is placed on the Internal FTP Server. The name of the file, that is placed on the Internal FTP server, must match exactly the “File Name” field on the “Registered File Name List” in the ATDL command. If the name of the file does not appear in the list, an entry can be created.

5. The number of IP terminals that can download the download file simultaneously varies between External and Internal FTP Server (embedded in the Telephony Server).
   (The Internal FTP server uses the CPU in the System. When using the internal server, log in to the System from a PC with FTP. As default, the data type for Internal FTP is binary. Therefore, use the FTP “bin” command, prior to the FTP “put” command, when transferring files to the Internal FTP Server.)
   a.) When using Internal FTP Server, the file cannot be downloaded to more than one terminal at the same time. Download method is automatically set to “Sequential.”
   b.) When using External FTP Server, the Download Method can be assigned as “parallel.” At this time, a maximum of 8 terminals can download the new firmware simultaneously.

6. When the file downloading ends in failure, the terminal starts up with the previous version. The cause of fault is described below:

a.) Assigned FTP server is out of service.
b.) The network from the terminal to the FTP server doesn't work properly.
c.) Login account for FTP server is not registered correctly.
d.) Download file is not located in an appropriate directory or does not exist in the designated FTP server.
e.) The maximum number of concurrent connections to FTP server is exceeded.

7. When using Internal FTP Server:
   a.) Be sure to place the download files in the default FTP directory (home directory), which is created with the ADTM command so that logged in users can access it.
   b.) After downloading is completed, the unnecessary downloaded file must be deleted from the directory.

8. While IP terminal has an ongoing call or is originating a call, the file cannot be downloaded.

9. If a file is downloaded while IP terminal is receiving a call, the file will be downloaded as the terminal rings. The ringing will stop when the download is complete.

10. With IP terminals, do not disconnect a power cord that is plugged into them until a clock screen appears.

11. DT800 Series/DT700 Series cannot use this service during its registration to SR-MGC.

12. File access for an IP terminal is restricted when the FTP commands (such as "*put*" and "*get*") are being used by users other than the FTP user account for IP terminals to access a download file on the Internal FTP server (Telephony Server), and vice versa.

13. Stop Routine Diagnosis while downloading.

14. The information on terminals that execute downloading can be added to the [Terminal List] by either of the following manner.
   a.) Adding from Office Data to [Terminal List]
   The information on each IP terminal, such as a station number, LENS and Terminal Type is read out and is added to the [Terminal List].
   b.) Adding from File to [Terminal List]. **Note 2**
   The information on terminals is added to the [Terminal List] by importing the file that is created beforehand to specify terminals that execute downloading.
   There are two ways to output the file to specify terminals to execute downloading.
   • Exporting
   • Outputting a template firm for specifying terminals

Note 2: When adding the information by using a file, note the following:
- Only one file can be imported to [Terminal List]. If the information on multiple terminals are to be input, all information must be listed in one file.
- The exported file (*.text) can be edited directly with a text editor or Excel. When editing with Excel, see the section of <Editing the Data File with Excel>.

Programming

STEP 1: **ADTM (Assignment of Data for TCP/IP Module)**

Register the FTP user account of IP terminals with the ADTM command when using the Telephony Server as the FTP server.

[Select LAN Interface]
Select LAN1

[Options Tab]

(1) Select [Maintenance User Account Setting] under [LAN1] from [A list of Functions], and click the EXECUTE button.

(2) Click the ADD button on the [Maintenance Account Setting] screen.

(3) Configure the following settings on the [Maintenance Account Setting] sub screen, then click OK.

- Login Name: 4~12 characters
- Password: 4~12 characters
- Check [FTP]

Note: For details on the availability of characters used for Login Name and Password, see [ADTM: Assignment of Data for TCP/IP Module] in the Command manual.

Note: The level is fixed to low.

(4) Click the write button on the [Maintenance Account Setting] screen.

STEP 2: **ADTL (Assignment of Terminal for Download)**

Follow the below procedure to configure settings for IP terminals that execute downloading.

(1) Setting the download file name

(2) Adding information of the terminals that execute downloading

(3) Selecting the terminals and executing downloading

1. Setting the download file name

☐ Display the List of Download File Name:
Click this checkbox when displaying [List of Download File Name].

Terminal Type: Type of IP terminal (maximum of 64 characters)
The types of IP terminals corresponding to each download file are displayed.

HWTP: Hardware Type(00~FF)
The hardware types corresponding to each type of IP terminals are displayed.

HWVER: Hardware Version (00~FF)
The versions of hardware corresponding to each type of IP terminals are displayed.

File Name: File Name (maximum of 16 characters)
The names of files to be downloaded for each terminal are displayed.

Note: The file of List of Download File Name can be imported by pressing the Import button, and it can be output to a desired folder by pressing the Export button. When pressing the Reset to the Default Setting button after importing the file, the information listed in the [List of Download File Name] returns to the default values. Also, the imported information returns to the default values if the command processing is aborted. If you wish to change the default values, import the file at the time of the desired command booted.

2. Adding information of the terminals that execute downloading

Up to 4000 items of download file information can be registered by either of the following manner.

(a) [Adding from Office Data to [Terminal List]]

(b) [Adding from File to [Terminal List]]

[Adding from Office Data to [Terminal List]]

Process Type:

- Execute downloading
  Specify a terminal and execute downloading.

Terminal Specifying Method:

- Office Data
  Information of the downloading terminal is read out from Office Data.

Terminal Specifying Type:

- Station Number
  Select this when specifying a station number of the IP terminal.

  TN: Tenant Number
  STN: Station Number

- Line Equipment Numbers
  Select this when specifying Line Equipment Numbers of the IP terminal.

  LENS: Line Equipment Numbers (six digits)

- Terminal Type
  Select this when specifying a type of the IP terminal.

  Terminal Type:
  Select a type of the IP terminal.

  HWTP-HWVER: Hardware Type-Hardware Version
  Select the hardware type and the hardware version corresponding to the selected 'Terminal Type'.

Specify Conditions for Filtering:

- All

- Station Number

  Select this from the drop-down list box to search for terminals by specific Tenant Number (TN) and Station Number (STN) range.

  Firmware Version:
  The selected Terminal Type can be narrowed down by choosing Firmware Version.

  Version:
  Enter the Firmware Version(00.00-99.99)
  Conditions:
  Select the conditions for filtering (!=, <, <=, ==, >=, >).

  TN: Tenant Number

  Start: First Tenant Number to start searching.
  End: Last Tenant Number to stop searching.

  STN: Station Number

  Start: First Station Number to start searching.
  End: last Station Number to stop searching.

- Line Equipment Numbers

  Select this from the drop-down list box to search for terminals by specific Line Equipment Numbers (LENS) range.

  Firmware Version:
  The selected Terminal Type can be narrowed down by choosing Firmware Version.

  Version:
  Enter the Firmware Version(00.00-99.99
  Conditions:
  Select the conditions for filtering (!=, <, <=, ==, >=, >).

  LENS: Line Equipment Numbers

Start: First LENS to start searching (six digits).
End: Last LENS to stop searching (six digits).

- IP Address
  Select this from the drop-down list box to search for terminals by specific IP address range.

  Firmware Version:
  The selected Terminal Type can be narrowed down by choosing Firmware Version.

  Version:
  Enter the Firmware Version(00.00-99.99)
  Conditions:
  Select the conditions for filtering (!=, <, <=, ==, >=, >).

  IP Address:

  Start: (0.0.0.1 - 255.255.255.254)
  End: (0.0.0.1 - 255.255.255.254)

[Adding from File to [Terminal List]]

Process Type:

- Execute downloading
  Specify a terminal and execute downloading.

Terminal Specifying Method:

- File
  Import the file that is created beforehand for specifying terminals.

File Path: (Display only)
File path for the file for specifying terminals will display.

Note: LENS has first priority to be used when both LENS and TN/STN are included in the specified file. For information on creating a file, refer to 'How to create a file for specifying terminals'.

3. Selecting the terminals and executing downloading

Terminal List:
Downloading is executed for the checked items of information. The file for specifying terminals can be output to any folder by clicking the Export button.

Note: When information is added to [Terminal List] by selecting 'Execute downloading' for Process Type, the terminals which cannot execute downloading will appear in gray in [Terminal List]. The checkbox displayed on the head of the rows cannot be checked in this case.

Terminal Type
HWTP: Hardware Type
HWVER: Hardware Version
Version
LENS: Line Equipment Numbers
SW/SP: SW/SP Number
TN: Tenant Number
STN: Station Number
RT: Route Number
TK: Trunk Number
PCN: PHS Community Number
ERN: Calling Area Number
GRN: Group Number
CSN/ZTN: CS/ZTN Number
IP Address
File Name
Status/Result
Error Detail

Settings for Downloading:

FTP Server Type:
Select one of the following to specify the FTP server for storing download files.

- ACT CPU (to specify the ACT side of the Telephony Server as FTP server)
- STBY CPU (to specify the STBY side of the Telephony Server as FTP server, or in dual configuration)
- External Server (to specify the FTP server connected to LAN)

IP Address:
This parameter is displayed only when “External Server” is selected for **FTP Server Type**. Assign the IP Address of FTP server connected to LAN (0.0.0.1 - 255.255.255.254).

Login Name/Password [Maximum 15 digits]:
Enter the Login Name and Password required to log in to the FTP server.

☐ Specify File Storage Directory:
Click this check box to specify the directory where the download files are stored. When no directory is specified, the home directory (default directory) that the users are allowed to access after logging in to the FTP server is automatically used.

Directory Name:
Enter a directory name within 63 characters. NEC recommends using the default directory that the users have access to when they are logged in to the FTP server.

Download Method:
Select either of the following to specify the download method. Automatically set to Sequential when "ACT CPU" or "STBY CPU" is selected in **FTP Server Type**.

- Sequential
  After downloading to one terminal has been completed, download to the next sequentially numbered terminal starts immediately.
- Parallel
  Used when "External Server" is selected in **FTP Server Type**. The maximum number of terminals to which download is concurrently executed depends on the number singed to **Concurrent Number**.

  Concurrent Number (2-8)

☐ Specify the download start time.

Click the checkbox of 'Specify the download start time' to enter the desired date and time.

Start Date (YY/MM/DD)

Start Time (HH/MM/SS)

How to create a file for specifying terminals

Follow the procedure below to create a file for specifying terminals.

(1) Output the file for specifying terminals.

(2) Edit the output file for specifying terminals.

1. Output the file for specifying terminals.

   The file for specifying terminals is output by either of the following manner.

   (a) [Exporting]

   (b) [Outputting a template firm for specifying terminals]

[Exporting]
Terminal information, retrieved from Office Data, is exported.

Process Type:

- Creating a file for specifying terminals.

Terminal Specifying Method:

- Office Data
  Information of the downloading terminal is read out from Office Data.

Terminal Specifying Type:
Add terminal information referring to ‘Adding from Office Data to [Terminal List]’ of ‘4. Adding information of the terminals that execute downloading.’
Save the file for specifying terminals in any folder by clicking the Export button.

[Outputting a template firm for specifying terminals]

Output the template firm for specifying terminals.

Process Type:

- Creating a file for specifying terminals.

Terminal Specifying Method:

- Edit directly.
  A template file for specifying terminals is output.

2. Editing the output file for specifying terminals.

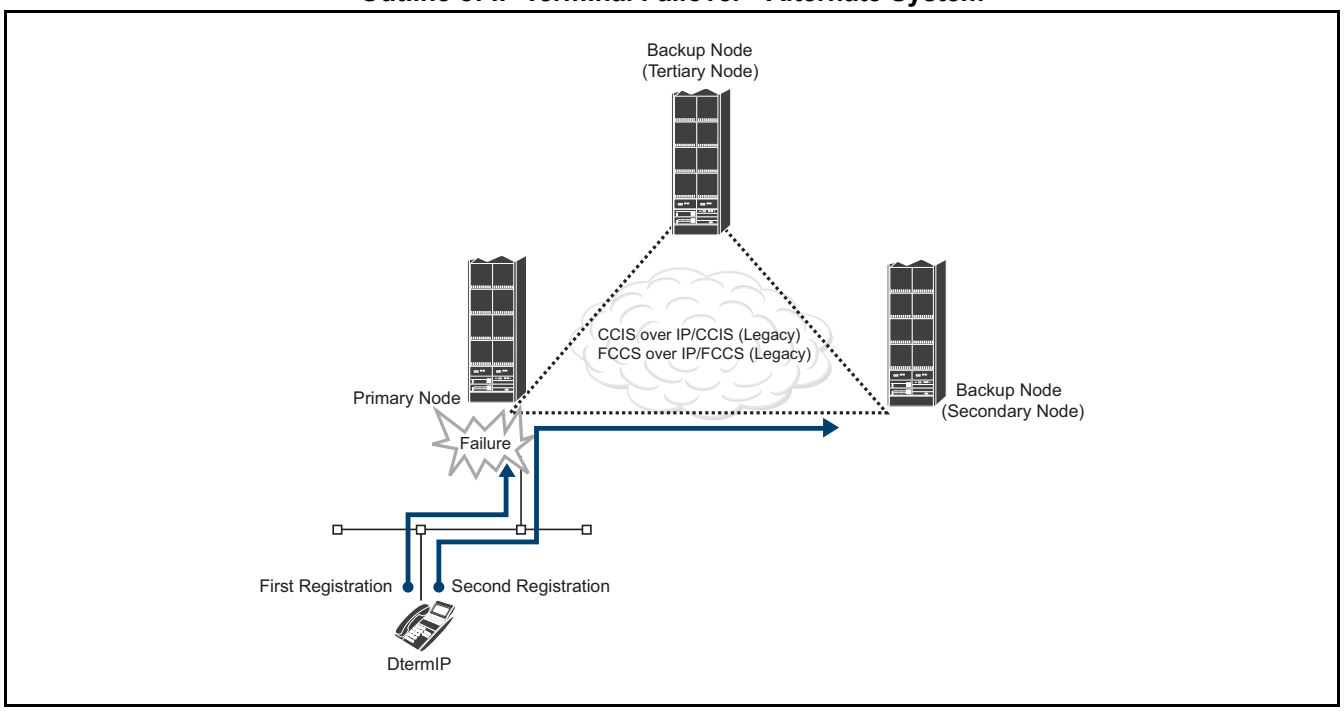
   The exported file can be edited directly with a text editor or Excel. When editing with Excel, see the section of <Editing the Data File with Excel>.

I-48 IP Terminal Failover - Alternate System

General Description

In CCIS over IP/CCIS (Legacy)/FCCS over IP/FCCS (Legacy) network, if a failure occurs in the Primary Telephony Server, DtermIPs can automatically reregister to a predetermined Backup Telephony Server. While the DtermIPs are placed under the control of Backup Node (Telephony Server), they can operate as they ware in Primary Node (Telephony Server).

Outline of IP Terminal Failover - Alternate System



This feature is comprised of the following four processes:

- Polling and Creating of Visitor Station User Profile Database
- Creating of Physical Station Pool (by ABKS)
- Providing Physical Station Resource for Visitor Station
- Updating Process of Free Location Table

- **Polling and Creating of Visitor Station User Profile Database**

  This process determines which the Telephony Server is used as Secondary Telephony Server for each Station. Also, Station Numbers used for Second-registration are registered on Secondary Telephony Server by this process. Secondary Telephony Server reads the Station Profile information (key data, Speed Calling data, etc.) from the Primary Telephony Server accommodating the appropriate Stations, at the specified time.

- **Creating of Physical Station Pool (assigned by ABKS)**

  For Visitor Stations, it is necessary to assign Physical Station Numbers on the Secondary Telephony Server in advance. Resources of the pre-assigned Station Numbers are provided for Registration process of Visitor Stations as necessary.

- **Providing Physical Station Resource for Visitor Station**

  If the MAC address of the Visitor Station is not registered on the Secondary Telephony Server, the Secondary Telephony Server queries MAC address data table in the User Profile database read from Primary Telephony Server. At this time, when the Secondary Telephony Server can find the appropriate MAC address in the User Profile database, the Secondary Telephony Server will use one of the pre-assigned Physical Station Numbers in Physical Station Pool for Registration of the Visitor Station. When the Secondary Telephony Server cannot find the appropriate MAC address, the Secondary Telephony Server displays login prompt on the LCD of the Visitor Station. Then, if the Secondary Telephony Server can find the Station Number and the password entered by Visitor Stations in the User Profile database, the Secondary Telephony Server will accept the Registration request from the Visitor Station.

- **Updating Process of Free Location Table (Station Number Conversation Table)**

  To permit the call termination to the Visitor Station that has completed Second Registration process, it is necessary to update all the Free Location Tables in the network. The Secondary Telephony Server sends FLF Update request packets to Master Telephony Server after the Second Registration Process of the Visitor Station. Then, to share the Physical Station Number allocated to the Visitor Station among the network, the Master Telephony Server sends the updated FLF information to all the Telephony Server in the network. When this updating has completed, the VISITOR key on the Visitor Station flashes green to inform the user that it operates in Visitor mode. At this time, when this key is assigned to a Soft key, "VISIT" on the LCD will flash.

- **Master/Sub-Master Telephony Server**

  It is necessary to place the Telephony Server that controls the Free Location tables created by each Telephony Server in the network (including a maximum of 64 Telephony Servers). This Telephony Server is called "Master Telephony Server." Master Telephony Server receives the update information of the Free Location tables from each Telephony Server in the network. For this reason, TCP links must be established between Master Telephony Server and all the other Telephony Servers, at all times.

  A maximum of two Master Telephony Servers can be placed in a network (at this time, another one is called "Sub-Master Telephony Server). By placing two Master Telephony Servers (Master Telephony Server and Sub-Master Telephony Server), redundancy for this Free Location table update mechanism can be accomplished.

Operating Procedure

- **Registration to Secondary Telephony Server from IP Station in MAC Authentication mode**

  It is necessary to assign the IP address of Secondary Telephony Server at each IP terminal in advance. When using DHCP Server, assign IP addresses of both Primary Telephony Server and Secondary Telephony Server on the DHCP Server (Index 161). By assigning MAC addresses of IP terminals on Primary Telephony

Server in advance, the IP terminals can be automatically backed up by Secondary Telephony Server when Primary Telephony Server breaks down. At this time, the VISITOR key on the Visitor Station flashes green to inform the user that it operates in Visitor mode.

- **Registration to Secondary Telephony Server from IP Station in Login mode**

  When MAC address of Visitor Station is not assigned on Primary Telephony Server, the Visitor Station must perform Login operation for the Registration to Secondary Telephony Server. At this time, the VISITOR key on the Visitor Station flashes green to inform the user that it operates in Visitor mode.

  Note: When Visitor Stations perform Registration to Secondary Telephony Server, they cannot use their Station Numbers as the default password (for the security reason).

- **Recovery from Visitor Mode**

  Visitor Station operated under the control of Secondary Telephony Server monitors Primary Telephony Server periodically. When the Primary Telephony Server recovers from a network fault, the VISITOR key on the Visitor Station lights green. At this time, if MAC address of Visitor Station has been assigned on Primary Telephony Server, it performs Registration to the Primary Telephony Server automatically when it become idle. Conversely, if MAC address of Visitor Station has not been assigned on Primary Telephony Server, the user must perform Login operation again for the Registration to Primary Telephony Server. After the Registration to Primary Telephony Server has completed successfully, the VISITOR key will be extinguished.

- **Example Application for Login Operation from Visitor Station**

  Even if Primary Telephony Server operates normally, a Visitor Station whose Station Number is assigned on Secondary Telephony Server can log into the Secondary Telephony Server by using its Station Number. For example, when a branch officer uses an IP terminal accommodated in the main office Telephony Server, the officer can log into the main office Telephony Server by using one's Station Number. At this time, the officer logs into the main office Telephony Server as a Visitor Station. While the Station Number is used in Visitor mode, VISITOR key on the terminal at the branch office flashes red.

Service Conditions

1. This feature is available only for the network with Closed Numbering.
2. This feature is available for IP Enabled Dterm and DtermIP (NEC Proprietary) such as ITR series but is not available for DtermIP (SIP) such as ITN series and DT800 series/DT700 series.
3. A Secondary Telephony Server can accept Registration requests from a maximum of 1000 Visitor Stations.
4. Secondary Telephony Server can accept Registration requests from Visitor Stations without consuming its licenses.
5. Secondary Telephony Server provides the following data to Visitor Stations.
   - Station Forwarding data (Call Forwarding-All Calls/Busy/Don't Answer)
   - Speed Calling - One Touch data
   - Function key data (except the data related to Multi Line and Intercom)
   - Login Password data
   - Name Display data
6. The following data assigned on a System-wide basis can be backed up by Secondary Telephony Server.
   - MAC Address and IP terminal Mapping Information

7. When Secondary Telephony Server backs up the Stations accommodated in Primary Telephony Server, Secondary Telephony Server does not inherit their SFC/RSC data from Primary Telephony Server. They are operated with the SFC/RSC determined on Secondary Telephony Server. For this reason, there may be cases where the some features which are allowed at Primary Telephony Server are restricted in Secondary Telephony Server and vice versa.
8. Station Number of a Visitor Station (not Physical Station Number) can be output to SMDR as a Calling Telephone Number.
9. Station Number of Visitor Station (not Physical Station Number) can be output to MCI as Calling/ Transferring Telephone Number.
10. Station Number of Visitor Station (not Physical Station Number) can be displayed on the Dterm's LCD via Off-Hook Line Number Display -D [O-37D].
11. It is necessary to place a Master Telephony Server that controls the Free Location tables created by each Telephony Server in the network. A maximum of two Master Telephony Servers can be placed in a network If there are only two Telephony Servers (Primary Telephony Server and Secondary Telephony Server) in the network, the Secondary Telephony Server can work as Master Telephony Server.

    Note: When this feature is activated in the FCCS Network, Master Telephony Server must be placed in VNDM Center Node. At this time, only one Master Telephony Server can be placed in a network.

12. When Stations recover from Visitor mode, Primary Telephony Server does not inherit their setting data (forwarding data, etc.) from Secondary Telephony Server.
13. The Physical Station Numbers on Secondary Telephony Server are automatically allocated to Visitor Stations for each Registration to Secondary Telephony Server. Therefore, the Visitor Stations cannot use the previous setting data.
14. The Physical Station Numbers on Secondary Telephony Server are automatically allocated to Visitor Stations on a per Tenant Number basis. If a Visitor Station cannot detect the Physical Station Number belonging to the same Tenant Number as assigned on Primary Telephony Server, the default Physical Station Number Pool (Pool Number = 0) is allocated to the Visitor Station.
15. For Visitor Station, the number as My Line must be assigned for Prime Line because Secondary Telephony Server inherits only the My Line data from Primary Telephony Server. If the Station whose Primary Line is assigned as Sub Line is backed up by Secondary Telephony Server, My Line Number is automatically assigned for Prime Line.
16. When some features (such as Camp-On and Call Forwarding - Don't Answer features) are activated between a Visitor Station and the other Station via CCIS, Physical Station Number automatically allocated to Visitor Station is displayed on the LCD of the called party.
17. To activate this feature, the firmware version 02.00 or later must be equipped on the Dterm.
18. The physically non-existent ports of Group No. 24 - 31 cannot use as Physical Station Number Pool.
19. The necessary number of Physical Station Number Pools for Visitor Stations must be assigned on Secondary Telephony Server in advance.
20. Physical Station Number Pools can be assigned on the Virtual PIR.
21. Do not assign the duplicate number for Physical Station Number, Visitor Station Number/Telephone Number and Feature Access Code within the same network.
22. A maximum of 96 Physical Station Numbers can be assigned per Pool.

Interactions

1. Visitor Stations under the control of Secondary Telephony Server cannot use ACD feature provided by ACDP built in Primary Telephony Server. To solve this problem, manipulate Agent Positions accommodated in Primary Telephony Server via ACDP built in Secondary Telephony Server by using AGENT ANYWHERE - ACD [A-133A].
2. Visitor Stations under the control of Secondary Telephony Server cannot use OAI feature provided by UAP connected with Primary Telephony Server. To solve this problem, manipulate OAI Terminals accommodated in Primary Telephony Server via UAP connected with Secondary Telephony Server.
3. The CCIS Cluster port assignment, which is set via ADTM command ("Configure Application Port" window on option tab), has to be the same side with the registration port setting, which is shown as "REG" in the window. The recommended side is LAN1 (Default).
4. The following tables show the availability of each Telephony Server feature.

List of OG Call Features for IP Terminal Failover - Alternate System

x: Available N: Not Available C: Conditionally available –: Not Applicable

SERVICE FEATURES	AVAIL-ABILITY	CARRY OVER	REMARKS
Direct Outward Dialing (DOD)	x	N	
Miscellaneous Trunk Restriction	x	N	
Toll Denial/Toll Diversion	x	N	
Toll Restriction - 3/6-Digit	x	N	
Speed Calling-Station	N	N	
Speed Calling-System	N	N	
Outgoing Trunk Queuing	x	N	
Off-Hook Queuing	x	N	
Alternate Routing	x	N	
Pushbutton-To-Rotary Conversion	–	–	

Note: The above Outgoing Call features can be provided by using Outgoing Trunks, Outgoing Restriction Tables, Outgoing Access Codes Authorization Codes assigned on Secondary Telephony Server.

List of IC Call Features for IP Terminal Failover - Alternate System

x: Available N: Not Available C: Conditionally available –: Not Applicable

SERVICE FEATURES	AVAIL-ABILITY	CARRY OVER	REMARKS
Attendant Console	x	N	
Direct-In Termination (DIT)	N	N	
Direct Inward Dialing (DID)	x	N	
Automated Attendant	–	N	
Remote Access To System	N	N	
Master Console	x	N	
Power Failure Transfer	–	–	
Internal Music On Hold	x	N	
External Music On Hold	x	N	
Distinctive Ringing	x	N	
Tenant Service	x	N	
Slumber Time-Do Not Disturb	N	N	
Multiple Music On Hold	x	N	

Note: The above Incoming Call features can be provided by using Incoming Trunks assigned on Secondary Telephony Server.

List of Station Features for IP Terminal Failover - Alternate System (1/2)

x: Available N: Not Available C: Conditionally available –: Not Applicable

SERVICE FEATURES	AVAIL-ABILITY	CARRY OVER	REMARKS
Station Hunting	N	N	
Call Pickup	N	N	
Call Pickup-Direct	N	N	
Step Call	C	N	Available except when Visitor Station is the destination of STEP CALL.
Call Back	x	N	
Automatic Call Back Cancel	x	N	
Call Waiting	x	N	
Executive Right-of Way	x	N	
Privacy-D	x	N	
Call Forwarding-Busy Line	C	x	Available except when Visitor Station is assigned as the forwarding party of the other Station.
Call Forwarding-Busy Line-Outside	C	x	Available except when Visitor Station is assigned as the forwarding party of the other Station.
Call Forwarding-Don't Answer	C	x	Available except when Visitor Station is assigned as the forwarding party of the other Station.
Call Forwarding-Don't Answer-Outside	C	x	Available except when Visitor Station is assigned as the forwarding party of the other Station.
Call Forwarding-All Calls	C	x	Available except when Visitor Station is assigned as the forwarding party of the other Station.
Call Forward All/Busy/No Answer Clear	x	N	
Call Forwarding-All Calls-Outside	C	x	Available except when Visitor Station is assigned as the forwarding party of the other Station.
Split Call Forwarding	x	x	
Tone to be sent out when the handset is off-hook at the station on which C.F. - All Calls service is set.	x	N	
Call Transfer-All Calls	x	N	
Blind Transfer	x	N	
Boss Secretary	N	N	
Three-Way Calling	x	N	
Station-Controlled Conference	x	N	
Call Hold	x	N	
Automatic Recall	x	N	
Exclusive Hold	x	N	
Hot Line	N	N	

List of Station Features for IP Terminal Failover - Alternate System (2/2)

x: Available N: Not Available C: Conditionally available –: Not Applicable

SERVICE FEATURES	AVAIL-ABILITY	CARRY OVER	REMARKS
Priority Call	C	N	Available except when this feature call is directed to the predetermined Station.
Station-To-Station Calling	x	N	
Class of Service-Individual	x	N	
Day/Night Class of Service	x	N	
Timed Reminder	N	N	
Call Forwarding-Intercept	N	N	
Call Forwarding-All Calls-Announcement	x	N	
Howler Tone Sending	–	–	
Line Lockout	x	N	
Off-Hook Alarm	C	N	Available except when this feature call is directed to the predetermined Station.
Periodic Time Indication Tone	x	N	
Immediate Ringing	–	–	
Pushbutton Calling	–	–	
Flexible Numbering of Station	x	N	
Line Circuit Reverse Relay Control	–	–	
Emergency Call	x	N	
Station 5dB Pad	x	N	
Call Forwarding-I'm Here	C	x	Available except when Visitor Station is assigned as the forwarding party of the other Station.
Dual Station Call	N	N	
Camp-on Busy	x	N	
Remote Call Forwarding Control	N	N	

List of Other Features for IP Terminal Failover - Alternate System

x: Available N: Not Available C: Conditionally available –: Not Applicable

SERVICE FEATURES	AVAIL-ABILITY	CARRY OVER	REMARKS
Last Number Call	x	N	
Call Park	N	N	
Call Park Group	N	N	
Uniform Call Distribution (UCD)	N	N	
Delay Announcement-UCD	N	N	
Long Line Trunk	–	–	
Announcement Services	x	N	
Paging Access	N	N	
Paging Transfer	N	N	
Radio Paging	x	N	
Call Forwarding-Intercept/Announcement	x	N	
Route Restriction-Announcement	x	N	.
DSS Console	N	N	
Authorization Code	N	N	
Account Code	N	N	
Number Sharing	N	N	
Multiple Music on Hold Source	x	N	
Message Center Interface	x	N	

List of Dterm Features for IP Terminal Failover - Alternate System (1/2)

x: Available N: Not Available C: Conditionally available –: Not Applicable

SERVICE FEATURES	AVAIL-ABILITY	CARRY OVER	REMARKS
Voice Call	x	N	
Hands-Free Dialing/Monitoring-D	x	N	
Intermediate Station Number Display-D	x	N	
Outgoing Trunk Queuing-D	x	N	
Speed Calling-One Touch-D	x	x	
Speed Calling-Station	N	N	Visitor Station cannot use Speed Calling Numbers assigned on Primary Telephony Server.
Elapsed Time Display-D	x	N	
Call Pickup-Group-D	N	N	This feature data cannot be backed up by Secondary Telephony Server.
Call Hold -D	x	N	
Last Number Call-D	x	N	
Service Display-D	x	N	
Three-Way Calling-D	x	N	
Time Display-D	x	N	
EXCLUSIVE HOLD-D	x	N	
Call Transfer-All Calls-D	x	N	
Save and Repeat-D	x	N	
Distinctive Ringing	x	N	
Do Not Disturb-D	x	N	
Flexible Ringing Assignment -D	x	N	
Volume Control-D	x	N	
Exclusive Right of Way-D	x	N	
Call Back-D	x	N	
Name Display-System	x	x	
Hands-Free Answer Back-D	x	N	
Called Station Status Display-D	x	N	
Call Forwarding-All Calls-D	C	N	Available except when Visitor Station is assigned as the forwarding party of the other Station.
Flash Button-D	x	N	
Non-Square Line Assignment-D	-	-	
Prime Line Pickup-D	N	N	

List of Dterm Features for IP Terminal Failover - Alternate System (2/2)

x: Available N: Not Available C: Conditionally available –: Not Applicable

SERVICE FEATURES	AVAIL-ABILITY	CARRY OVER	REMARKS
Splitting-D	x	N	
Pause Entry	x	x	
Non-Exclusive Hold-D	x	N	
Multiple Line Operation-D	N	N	Available only for My Line.
Message Remainder-D	x	N	
Call Forwarding-Busy Line-D	C	x	Available except when Visitor Station is assigned as the forwarding party of the other Station.
Call Forwarding-Don't Answer-D	C	x	Available except when Visitor Station is assigned as the forwarding party of the other Station.
Privacy-D	x	x	
Speed Calling-One Touch-D	x	x	
Page Changeover	x	x	
Manual Signaling	N	N	
Automatic/Manual Intercom	N	N	
Brokerage Hot Line	N	N	
Last Number Called-D	x	N	
Call Park Group-D	N	N	
Internal Zone Paging	N	N	Due to the specifications of IP terminal.
Serial call-D	x	N	
Off-Hook Line number Display-D	x	N	
Ringer Mute	x	N	
Soft Key	x	N	

Programming

STEP 1: **ASYD** - SYS1, Index 31 (Mounting capacity of Common Memory) must be assigned as data "06." SYS1, Index 79, Bit 6=0 (OAI service is available).

STEP 2: **ADTM** - Assign the IP address of Master Telephony Server (and Sub-Master Telephony Server) to all Telephony Servers in the network. The identical data must be assigned to all Telephony Servers in the network.

Options - Registration Service

Telephony Server Search Address 1: Enter the IP address of Master Telephony Server

Telephony Server Search Address 2: Enter the IP address of Sub-Master Telephony Server if necessary.

STEP 3: **ASYDL** - SYS1, Index 848, Bit 6=1 (Internal FLF Server / Client is available).

STEP 4: **ASFC** - SFI 205 must be assigned as 1 to the stations used as Physical Station Pool in the Backup Telephony Server.

STEP 5: **ARTD** - Assign the following Route Class Data for CCIS Trunk Route.

IND (CDN: 112) Inter-office Name Display: 1 (In service)

Note: To change the data for existing route (RT), initialize the circuit card.

STEP 6: **ABKS** - Assign the following data on each tab.

Pool Station tab - Assign the following Physical Station Number Pool data on Secondary Telephony Server.

POOL NUMBER:	Specify the Pool Number (0-63)
PHONE TYPE:	Select "Digital" from the combo box.
TN:	Tenant Number
RSC:	Route Restriction Class (0-15)
SFC:	Service Feature Restriction Class (0-15)
START STN:	First Station Number (a maximum of six digits)
START LENS:	First Line Equipment Numbers (five digits)
NUMBER OF REGISTRATION:	Assign the number of Stations belonging to the Pool assigned in POOL NUMBER parameter. (the number from 8 to 96, in multiples of 8)

Note: "*" and "#" shall not be included in the last two digits of Station Number.

Note: The following explains the relationship among "STN START", "LENS START" and "NUMBER OF REGISTRATION" parameters.
[Example of Data Assignment]
START STN = 3000
START LENS = 00006
NUMBER OF REGISTRATION = 8

Station Number	*LENS*
3000	**00006**0
3001	000061
3002	000062
3003	000063
3004	000064

Station Number	*LENS*
3005	000065
3006	000066
3007	000067

Backup Station tab - Assign Station Number or Telephone (Logical) Number of Visitor Station on Secondary Telephony Server.

STN: Assign Station Number (a maximum of six digits) or Telephone Number (a maximum of 12 digits) of Visitor Station

STN/TELN: Select either STN or TELN.

PC: Assign Point Code of Primary Telephony Server. (1-4095) This parameter is assigned only when

STN/TELN = STN is selected.

Specify the Polling time on Secondary Telephony Server.

Option tab

TYPE: Select "Polling" by clicking the appropriate radio button.

FUNCTION:

Manual Start: Polling process is executed immediately.

Auto Start: Polling process is executed at the specified time.

Polling start time: Specify the Polling time in hours and minutes. (00:00-23:59) This parameter appears only when FUNCTION = Auto Start is selected.

Note: When FUNCTION = Manual Start is selected, click "Start" button to start Polling of User Profile Database.

STEP 7: **AKYD** - Assign a programmable Line/Feature key as a Visitor key. KYI = 1 (Feature key). FKY = 156 (Visitor key).

Note: A Soft key can also be used as Visitor key. At this time, "VISIT" is displayed on the Dterm's LCD.

I-49 Internal Zone Paging - Multi-Zone

General Description

This feature allows an individual to make a Voice Call through the Dterm speakers of the first 4 (1-4) designated IZP groups by dialing the MULTI INTERNAL ZONE PAGING Access Code or pressing the MULTI INTERNAL ZONE PAGING key on the Desktop terminal. Refer to the Internal Zone Paging [I-21].

Note: This feature is only available in North America.

Operating Procedure

To originate

1. Lift the handset. Receive Dial Tone.
2. Dial the Access Code or press the Feature key for MULTI INTERNAL ZONE PAGING.
3. Announce the call over the built-in speakers on the Dterms in the INTERNAL ZONE PAGING [I-21] Groups 1-4.

To originate with Station or Trunk Hold

1. Receive a call or make a call to a Station or Trunk.
2. Perform a Switch hook flash.
3. Dial the Access Code or press the Feature key for MULTI INTERNAL ZONE PAGING.
4. Announce the call over the built-in speakers on the Dterms in the INTERNAL ZONE PAGING [I-21] Groups 1-4.

To answer

Same as INTERNAL ZONE PAGING [I-21]. Refer to the INTERNAL ZONE PAGING [I-21].

Service Conditions

1. See Service Conditions of INTERNAL ZONE PAGING [I-21].
2. This feature is only supported on the CPRRI-A (600mhz) and newer processors.
   a.) This feature uses a large amount of processing. If the System is already under a load of greater than 50%, expect short delays of service when MIZP is used.
   b.) This would be the same condition as if 4 24 member IZP groups were paged at the same time.
3. For MIZP the IZP groups 1-4 must be in the same node (FPC) of the FCCS network.
   a.) If any of groups 1-4 are in different FPC's the caller will receive RST and be sent Reorder tone.
   b.) This does not prevent the Calling party from being in a different FPC than IZP groups 1-4.
4. If all the Stations in all 4 groups are busy or in a state that will not accept the IZP the user is returned Busy Tone.
5. A maximum of 96 Stations can be paged using MIZP.
6. A maximum of 255 IZP Groups per system can be simultaneously Paged using MIZP and IZP together.
   a.) For all systems with MIZP the maximum number of IZP groups that can be simultaneously paged is 252.
   b.) Memory blocks 252-255 can be used for MIZP only.
7. A maximum of 1 MIZP can be made simultaneously.
8. First party release must be set by SYS1, Index 4, Bit 3=1 and SYS1, Index 4, Bit 2=0.
   a.) Improper operation will result if this is not set.

9. Directed Call Pick-Up may be used to answer a MIZP only if the destination of the MIZP is in the same FPC as the Station Picking-Up station.
10. To be used in an FCCS network the Access code for MIZP must be programmed in ASPAN or common numbering for this access code must be used in the network.
11. When using from IP Stations the connections are limited by the number of IPPAD trunks in the switch. You must have 1 trunk per paged Station.

Programming

See Programming for INTERNAL ZONE PAGING [I-21].
Extra Programming for MIZP follows:

STEP 1: **ANPD** - Reserve a Number Level for Feature Access. Select Normal Service (N) or Hooking Service (H) from the combo box for Connection Indexes. Assign NND according to the predetermined Numbering Plan.

STEP 2: **ASPA** - Assign the Access Code for MULTI INTERNAL ZONE PAGING, SRV = "SSCA: Service Code Appendix", SIDA = 143. Connection Status Indexes for "N: Normal and "H: Hooking" services.

STEP 3: **AIZP** - Assign INTERNAL ZONE PAGING Group Data.

TN: Tenant Number of Calling Party

IZPG: INTERNAL ZONE PAGING Group Numbers 1-4 are used by MIZP

IZPTN: Tenant Number of IZPSTN

IZPSTN: Station Number assigned to the INTERNAL ZONE PAGING Group.

STEP 4: **AKYD** - For Dterm sets, MULTI-INTERNAL ZONE PAGING can be activated using a programmable Line/Feature key. Assign KYI = 1, Feature key Answer [28] for answering MULTI-INTERNAL ZONE PAGING and KYI = 1, Feature key Internal Zone Paging [154] for calling MULTI-INTERNAL ZONE PAGING.

STEP 5: **ASFC** - Assign a Service Feature Restriction Class that allows SFI 215 (Internal Zone Page Calling) to the Stations that originate MULTI-INTERNAL ZONE PAGING.

Assign a Service Feature Restriction Class that allows SFI 145 (Internal Zone Paging Answer) to the Stations that answer to MULTI-INTERNAL ZONE PAGING.

STEP 6: **ASYD** - Assign SYS1, Index 4, Bit 3=1 and SYS1, Index 4, Bit 2=0 This is first party release and must be set for this feature to operate correctly.

FCCS Programming

This data setting describes only the additional or replacement commands used to activate FCCS Services. When used as an FCCS feature, these commands replace the non-network commands (for example, AAED is replaced by AAEDN). Note that the data setting for FCCS Service varies depending on the Node.

STEP 1: **ANPD/ANPDL/ANPDN** - Reserve a Number Level for Feature Access. Select Normal Service (N) or Hooking Service (H) from the combo box for Connection Indexes. Assign NND according to the predetermined Numbering Plan.

STEP 2: **ASPA/ASPAL/ASPAN** - Assign the Access Code for INTERNAL ZONE PAGING, SRV = "SSCA: Service Code Appendix", SIDA = 143. Connection Status Indexes for "N: Normal", and "H: Hooking" services.

STEP 3: **AIZPN** - Assign INTERNAL ZONE PAGING Group Data.

NIZPG: Network Internal Zone Paging Group Number 1-4 are used for MIZP All 4 must be in the same node for MIZP to function.

UGN: User Group Number

TELN: Telephone Number

CNT: Number of Telephone Number

L-1 Lamp Check

General Description

This feature allows the Attendant, by pressing the LCHK key, to light all lamps and sound the ATTENDANT CONSOLE [A-3] buzzer.

Operating Procedure

1. Attendant presses the Position Busy key. The LED lights.
2. Attendant simultaneously presses the L5, L6, and SRC key in Position Busy.

   Note: For Desk Console.

3. All lamps will light.

Service Conditions

1. Every Lamp Field, including the BUSY LAMP FIELD - FLEXIBLE [B-2], will be lit upon pressing of the three keys located on the ATTENDANT CONSOLE [A-3].
2. The Console Lamp Test displays the digit “8” in each field of the digital display.
3. When the keys are released, all lamps and Digit Display Fields are immediately restored to normal.

Programming

No programming is required.

L-3 Line Lockout

General Description

This feature provides for the automatic release of a station from the common equipment when it has remained off-hook for longer than the usual interval before dialing. The system may be programmed to return a Howler Tone to the analog station in the line lock-out mode.

Operating Procedure

For Line Lock-out

1. User goes off-hook and receives Dial Tone.
2. After approximately 12 seconds, if the user has not initiated dialing, he receives Reorder Tone.
3. After 30 seconds of Reorder Tone, the station is automatically placed into the line lock-out state.
4. Upon replacing the handset, the station is released from the line lock-out state and operation returns to normal.

For Line Lock-out with Howler Tone

1. User goes off-hook and receives Dial Tone.
2. After approximately 12 seconds, if the user has not initiated dialing, he receives Reorder Tone.
3. After 30 seconds of Reorder Tone, a Howler Tone is sent to the station.
4. Upon replacing the handset, the station is released from the Howler Tone State and operation returns to normal.

Service Conditions

1. A station in the line lock-out state can neither receive nor originate calls.
2. This feature is provided on a system-wide basis.
3. The Attendant cannot activate any feature to a station in the line lock-out state.
4. After 30 seconds of Reorder Tone, the station is automatically placed into line lock-out, if the system is not provided with Howler Tone Service.
5. Howler Tone should not be used with OPX lines.
6. Howler Tone can be provided or denied on a system-wide basis.
7. Howler Tone is only sent to analog stations.
8. When using OFF-HOOK ALARM [O-6] (ASYD, SYS1, Index 168, Bit 4=1), Howler Tone cannot be sent to the Guest Room of the Hotel System.

Interactions

1. When the BUSY LAMP FIELD - FLEXIBLE (BLF) [B-2] feature is invoked, the associated station status lamp on the ATTENDANT CONSOLE [A-3] will flash, indicating the Telephone Number is in line lock-out.
2. CALL FORWARDING - DON'T ANSWER [C-2, 3]: A call cannot extend a lock-out station via CALL FORWARDING - DON'T ANSWER [C-3].
3. BUSY VERIFICATION [B-3]: Busy verification of a locked-out station results in Reorder Tone being heard by the Attendant.

4. NIGHT CONNECTION - FLEXIBLE/FIXED [N-1, 2] with TAS [T-8] feature not active: When the Night station is in a lock-out state, calls directed to the Night station will receive Ringback Tone. If the TAS [T-8] feature is active, calls are directed to the Night Bell.
5. CALL WAITING [C-12, 31]: Call Waiting cannot be activated on a station in lock-out. The calling party will receive Reorder Tone.
6. CALL HOLD [C-6]: From a Special Dial Tone State, line lock-out will occur if a user attempting to place a station on hold dials the CALL HOLD [C-6] access code and partially dials any number. The sequence prior to lock-out is: interdigit time-out for six seconds, Reorder Tone for 30 seconds, optional Howler Tone for 30 seconds. When the station in lock-out goes on-hook, it will be rung by the station on hold. The Busy Lamp Field will not flash
7. EXECUTIVE RIGHT-OF-WAY [E-1]: Reorder Tone will be returned to a calling party that is attempting Executive Override toward a station in lock-out.
8. THREE-WAY CALLING [T-2]: An attempt to establish a Three-Way Conference with a station in lock-out will result in the Reorder Tone being returned to the user making the attempt. The user can return to the station on hold by switch hook flashing.
9. With the following calls, when a user fails to hang up after the other party has disconnected, the station will be placed in a lock-out condition after a 30-second Reorder Tone (time-out), optional 30-second Howler Tone Warning (time-out):

    DIRECT INWARD DIALING (DID) call [D-8]
    STATION-TO-STATION CALLING call [S-11]
    Trunk-to-Station call
    Attendant-to-Station call
10. CALL BACK [C-1]: To activate Call Back, the user dials the appropriate access code after reaching a busy station. If the user fails to go on-hook within 30 seconds after receiving Service Set Tone, line lock-out will occur. The sequence prior to lock-out is: Service Set Tone for 30 seconds, Reorder Tone for 30 seconds, optional Howler Tone Warning for 30 seconds. Call Back cannot be activated toward a station in lock-out.
11. CALL WAITING - ORIGINATING [C-31]: If a user fails to complete dialing of the access code while attempting to activate CALL WAITING - ORIGINATING [C-31], line lock-out will occur.
12. A station in line lock-out will be restored to service when the user goes on-hook for 280 milliseconds or more.
13. MESSAGE REMINDER [M-19]: A Message Reminder cannot be sent to a station in line lock-out.
14. ATTENDANT NIGHT TRANSFER [A-19] is not available if the Night Transfer station is in Line lock-out. The caller will hear Ringback Tone then Reorder Tone.

Programming

ASYD - SYS1, Index 64, Bit 3 and Bit 4. When is Howler Tone sent? 00: Not sent. 01: Only once. 10: Every thirty seconds.

SYS1, Index 70, Bit 1. Does station in line lock-out flash at BLF? 0/1: Yes/No.

SYS1, Index 128. This timer determines how long a Register will be held up before the first digit is dialed. Assign 00H for default value of 12 seconds (RAM Data is 06H).

SYS1, Index 146. First Howler Tone is sent. Assign 00H for standard default setting of 30 seconds (RAM Data is 3FH), on analog ports.

L-5 Least Cost Routing - 3/6-Digit

General Description

This feature allows the system to be programmed to route outgoing calls over the most economical facility (WATS, FX, DDD). Based on the Area Code and Office Code dialed (six-digit analyzing), the system examines the programming Tables and chooses the facilities in the order specified. Least Cost Routing (LCR) may also be performed on a Sender basis (LCRS).

Operating Procedure

1. Lift the handset. Receive Dial Tone.
2. Dial the trunk Access Code (usually "9"). Receive Second Dial Tone.
3. Dial the Area Code, Office Code and number.
4. The system automatically completes the call via the most economical route based on the programming tables.

Service Conditions

The following is a step-by-step explanation, which when reviewed in conjunction with the figure on the following page, will provide better insight as to how the system switch accomplishes LCR:

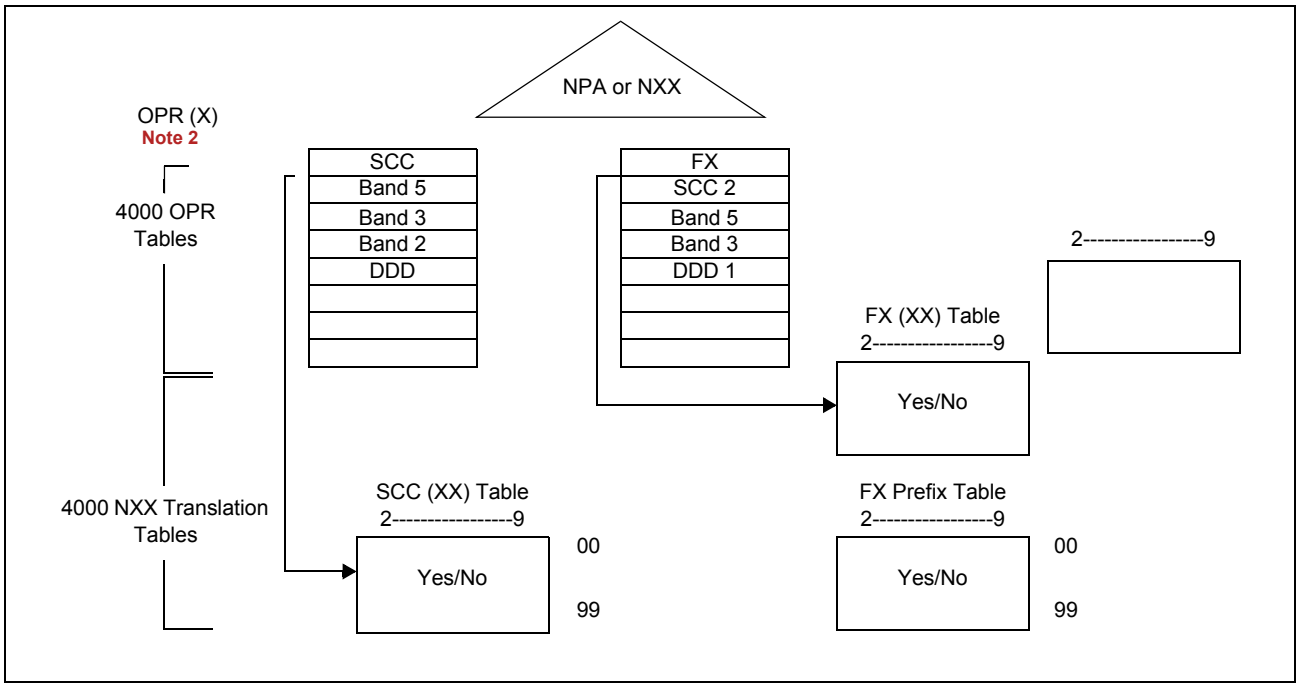
1. If dialing a Toll or Local Call (NPA-NXX-XXXX or NXX-XXXX), routing begins with translation of three-digit (NPA or NXX) area or Office Code. For specific numbers, this translation may result in the immediate selection of the three-digit routing patterns. If the number dialed is an NPA and six-digit translation is required, the NXX Code is verified in the SCC (XX) or the FX (XX) Table **Note 1**. If the NXX is not located in either Table, refer to Service Condition "3". There is a maximum of 4000 Outgoing Pattern Routing (OPR) Tables with a maximum of eight routes for selection.

   Note 1: a. SCC (XX) is the Special Common Carrier NXX Translation Table.
   b. FX (XX) is the FX NXX Translation Table.

2. First-choice Route option in an NPA Pattern will usually be a Specialized Common Carrier. If a trunk is available in this group, further digit translation will be required in order to determine whether the called number is served by the carrier. NXX digit translation will also be required in instances where a call is routed over a Tie or FX trunk. For these purposes, 4000 NXX Patterns are available.
3. In the case of Specialized Common Carriers, associated NXX Patterns will return either a Yes or a No answer to the switch, indicating whether or not the dialed number is within the designated service area of the selected SCC. If service is available, the switch will seize the trunk and complete the connection. If the no answer is returned, the switch searches for the next available trunk group.
4. Calls routed over FX trunk groups may require out-pulsing of prefix digit "1", depending on the NXX number dialed. For this purpose, another group of associated NXX Patterns may be used, operating on the same Yes/No principle. A Yes answer returned to the switch in this instance will result in prefix "1" being added automatically before sending digits to the Called Central Office.
5. Where the selected route is an FX trunk group, NXX translation may be used to either allow or disallow the connection. This is accomplished via Yes/No answers from a designated group of NXX Patterns, and enables calls to selected Central Office areas to be effectively restricted to only the most economical routes.

The routing method described provides some of the most flexible options available for call routing, and functions as a highly effective method of cost control. For details, see the following figure.

3/6-Digit Least Cost Routing Translation Table, Dialed Number 9-NPA-NXX-XXXX



Note 2: X = 1 to 4000

6. In certain cases, three and six-digit translation will provide sufficient information for identification of the service area. 4000 separate Outgoing Routing Patterns are available for this circumstance.
7. Up to eight trunk routes can be in each Outgoing Routing Pattern.
8. STATION MESSAGE DETAIL RECORDING (SMDR) [S-10] may be used in conjunction with LEAST COST ROUTING - 3/6-DIGIT. The route selected by the switch will be recorded by the SMDR unit.
9. Tie lines can be included in Route Patterns.
10. Application of NXX Patterns provides for inclusion of multiple common carriers serving different areas.
11. When a Tie trunk requiring an Off-net Connection to the Local Central Office is included as a Route option, the System can provide a 2.8-second prepause before sending out the Selecting Signal.
12. Senders (DP or DTMF) are provided with this feature for outpulsing.
13. Pre-translation function minimizes waiting time after dialing. After enough digits are collected to determine a route, the system starts outpulsing before the station completes dialing the entire number (LCR - SENDER).
14. If OUTGOING TRUNK QUEUING [O-2] is activated during LCR, all routes listed in the Outgoing Routing Pattern will be available to the queue, if one of these trunks becomes idle.
15. When LEAST COST ROUTING - 3/6-DIGIT is provided, the Attendant cannot pass Dial Tone to a station. The desired number must be dialed by the Attendant.
16. Stations having different Route Restriction Classes, but using the same LEAST COST ROUTING - 3/6-DIGIT Pattern, can be denied access to the last choice trunk routes in that pattern. Route Restriction Classes may be set up so that a station dialing an Area Code can use WATS, FX, and DDD, or only the WATS line, if such an arrangement is desired. This is a form of forced LCR.

17. NXX Patterns may be programmed to restrict all stations in the system from completing calls to specific NXX numbers over FX trunks.
18. The system will include the ability to detect a dialed DDD number and compare it with a DDD List Table to determine if the call is directed to a facility served by the trunk network. If the dialed number is found in the DDD List Table, a translation is provided to the seven-digit on-net number (ABC-XXXX) served by the Tie trunk. This allows an expensive Public Network Dialed Number to be trapped and automatically diverted to an available leased facility. If a Tie line is not available, the call will be routed as dialed.
19. The system is capable of converting a maximum of 24 receiving digits into a required 24-digit number entirely. Interdigit pauses can be inserted whenever necessary. This provides for an extremely flexible Numbering Plan for a complex network.
20. When deleting AOPR, this also deletes the ASDC Table using the OPR Pattern.
21. Advancing to Central Office Line is not available in the FCCS network.
22. The outgoing call using a trunk in the remote node via FCCS is available. Therefore, Route Restriction Class data must be identical in the FCCS network. For more detailed information, refer to the Data Programming Manual - FCCS.
23. ASDC is not applicable in the FCCS network.
24. When using this feature within an FCCS network, deleting AOPRL/AOPRN also deletes the ASDCL/ ASDCN Table using the OPR Pattern.
25. In case of LCRS feature, the system starts connecting and communication between the calling party and the called party without completing the maximum digits (MND) of destination number which is set in AMND from the calling party.
26. LCRS Call without completing the MND of AMND:
    a.) ARTI - Assign the parameter DCANS (CDN:54) as “1” to the objected Route.
    b.) ARTD - Assign the parameter OSGS (CDN:1) as “2” (Second Dial Tone) and LSG (CDN:15) as other than “12” (CCIS/ISDN) to the objected Route Class of outgoing trunk.
    c.) ASPA - For LCRS Flag Route, parameter Second Dial Tone = 1.
27. This feature is available:
    a.) when a call is directly originated from a station via LCRS.
    b.) when a station/trunk call, once placed on hold, is originated via LCRS.
    c.) when a trunk call is originated via LCRS (tandem connection).

Note: This feature is not available when the LCRS call from a Attendant Console or call made by Individual Trunk Access.

Interactions

1. AFRS and AMND Tables must correlate to each other. If dialing patterns listed in AFRS are not programmed in AMND, the calls will not go through. Also, if dialing patterns are not set in AFRS, including simple Local Calls, the calls will not go through.
2. LCR(S) provides only Route selection based on digits dialed and Route availability. Once a Route has been determined to be available because of dialed digits, Route Restriction Class will determine if this Route will be made available to the station. If denied and other routes are available, the same process will occur. If no other route is available, the user will hear a Reorder Tone.
3. In ARTD, CDN 9:TQ must be assigned for the LCR(S) Flag Route 31 if Trunk Queuing is to operate in conjunction with LCR(S).

Programming

STEP 1: **ASYD** - SYS1, Index 6, Bit 0. Should Special Sender Tone be sent to station when connected to a Sender? 0/1: No/Yes. (Optional)

SYS1, Index 6, Bit 2. Should Special Sender Tone be sent to the Attendant Console when connected to a Sender? 0/1: No/Yes. (Optional)

SYS1, Index 65. Assign the number of routes in the system.

SYS 2, Index 2, Bit 0. What is the sender type for an OG trunk connection to a station? 0/1: Dial Pulse/ Pushbutton. This data is valid only when CDN 2:ONSG of route data is assigned as data "3", PB/DP. 0

SYS2, Index 2, Bit 1. What is the sender type for an OG trunk to the Attendant Console? 0/1: Dial Pulse/ Pushbutton. This data is valid only when CDN 2:ONSG of route data is assigned as data "3", PB/DP.

STEP 2: **ARTD** - Assign the LCR(S) route as shown below. For uniformity, always use route 31 as the LCR(S) Flag route. In CDN 13:AC, include the LCR(S) Access Code in LCR(S) development.

Assign the LCR(S) Flag Route as shown below:

RT: 31

6-TCL	:1	7-L/T	:1	8-RLP	:2	9-TQ	:1	13-AC	:1

Note: To change the data for existing route (RT), initialize the circuit card.

STEP 3: **ATRK** - Assign Originating Register Trunks (ORT) to the system in a quantity calculated from expected traffic. Registers are located on Levels 0-3 of any 8RST circuit card.

Senders must be assigned to the system in a quantity calculated from expected traffic. Senders are located on Levels 4-7 of any 8RST circuit card.

RT: Route Number

Trunk Route Number

Intra-Office Route Number (901-931)

901-ATT 913-TCFT for ATT

902-ORT 915-Night ATT

903-IRT 916-MFCR

905-Sender 917-MFCS

909-DCFT 919 to 926-Modem

TK: Trunk Number

TN: Tenant Number

RSC: Route Restriction Class

SFC: Service Feature Restriction Class

For the outgoing trunks used in LCR(S), refer to those specific route access features.

STEP 4: **MBTK** - Assign the make idle status to all Register Sender trunks.

STEP 5: **ANPD** - Reserve a level, usually "9", for LCR or LCR (S) Access. Select Normal Service (N) or Hooking Service (H) from the combo box for Connection Indexes. Number of Necessary Digits is one. Busy Lamp Field is not activated.

STEP 6: **ASPA** - Assign the LCR (S) Access Code, to the ACC field (usually "9"). For SRV (Kind of Service), assign SRV = "LCR: Least Cost Routing" or SRV = "LCRS: Register Sender LCR". Assign the LCR(S) Flag Route Number, usually "31", associated with this Access Code to the RT field. Assign Connection Indexes, CI = "N: Normal" and CI = "H: Hooking". For Second Dial Tone, assign the data as "1". If Procedure II is used for AUTHORIZATION CODE [A-20], assign the data as "1" to the AH parameter.

STEP 7: **ARNP** - Assign the Physical Route Number as defined in ARTD. Relate these routes to the LCR(S) Access Code by assigning the value, previously defined in ASPA [for SRV = "LCR: Least Cost Routing" / "LCRS: Register Sender LCR"] to the ACC field of ARNP. The LCR(S) Access Code is usually "9". The routes to be assigned will be those routes listed in the various OPRs of AOPR.

Note: These same routes may also be listed in ASPA for SRV = "OGC: Outgoing Call". In that case, these routes may also be available via Direct Dial Access Codes. In ARNP, these Access Codes are not assigned to implement the LCRS service. This digit code is the first digit used when assigning the ATDP Table.

STEP 8: **ARSC** - LCR(S) uses only Bothway or Outgoing Only routes. For these types of routes assign appropriate Restriction Data to Route Restriction Indexes 2 (PRI-2) and 3 (PRI-3). To allow, assign a Restriction Data of "1" (RES-1). To restrict, assign a Restriction Data of "0" (RES-0), or for Toll Restriction, assign a Restriction Data of "2" (RES-2). Toll Restriction will be applied to the available route as determined by the selection in LCR(S). Therefore, if a user dials a long distance number and is restricted from DDD Access (RRI-3 = 0) and the only available route is a DDD, the call will be denied. Toll Restriction acts upon the LCR(S) selection. Also, remember to allow Route 31, the LCR(S) Flag Route to all RSCs. Do not Toll Restrict the Flag Route.

STEP 9: **AMND** - Assign the Maximum Necessary Digit (MND) the system must translate for a correct route selection. List the following type of dialing sequences.

1 + DIALING (N = 2-9 P = 0 OR 1, X = 0-9)				
CALL TYPE	**DIGITS DIALED Note 3**	**DIGITS NEEDED**	**MND**	**TOLL**
Local Calls	9-NNX-XXXX	9-N	8	Local Code
Near Distant Calls (1+)	9-1-NNX-XXXX	9-1-NN	9	Toll Code
Long Distant Calls	9-1-NPA-NXX-XXXX	9-1-NP	12	Toll Code

Note 3: The digits underlined are to be listed in the DC parameter. Overseas or Operator-assisted Calls should be listed with an MND that equals the calculated number of necessary digits. Special dialing sequences such as 411 with an MND = 4 should also be listed.

WITHOUT 1 + DIALING (N = 2-9 P = 0 OR 1, X = 0-9)				
CALL TYPE	**DIGITS DIALED Note 4**	**DIGITS NEEDED**	**MND**	**TOLL**
Local Calls	9-NNX-XXXX	9-NN	8	Local Code
Long Distant Calls	9-NPA-NXX-XXXX	9-1-NP	11	Toll Code

Note 4: The digits underlined are to be listed in the DC parameter. Overseas or Operator-assisted Calls should be listed with an MND that equals the calculated number of necessary digits. Special dialing sequences such as 411 with an MND = 4 should also be listed.

STEP 10: **AFRS** - List the dialing patterns in groups that will be served by a specific sequence of routes. For example, consider the following dialing patterns: 9-1-617, 9-1-413, 9-1-207, 9-1-203, 9-1-603.

Each of these Area Codes exist in the Northeastern United States. From New York City, it may be determined that the least expensive way to place any of these calls is by a Band 1 WATS, then BAND 5 WATS, then DDD. This determination of routes is labeled as an Outgoing Route Selection Pattern Number or OPR. In AFRS, program the Tenant, Route Number [LCR(S) Flag Route, usually "31"], the Number Pattern Code (NPC) or dialing patterns, and the OPR to be chosen when these digits are dialed.

STEP 11: **AOPR** - AFRS designates the dialing sequence and the OPR to be selected. The AOPR defines the routes and in what order these routes will be selected. Assign the following:

TDPTN Number (TDPTN) - Time of Day Change Pattern Data can be applied to an OPR. Assign a value, from 0 to 7, that correlates to a predefined Time of Day Change Pattern Number assigned in the ATCP command (and which is to be applied to this OPR).

Note: If the Time of Day feature is not applicable, assign a value of "0" in the TDPTN field for this OPR.

OPR Number (OPR) - Outgoing Route Selection Pattern Number. Enter the OPR Number referenced in AFRS. OPR Numbers range from 0-4000.

Route Advance Number (RA) - Route Advancing Order. The order in which a Route will be selected (0-7). For the above example - RA 0 = Band 1, RA 1 = Band 5, RA 2 = DDD.

Note: Always build your OPR beginning with an RA of "0". Assign any subsequent Route Advances numerically.

Route Advance End Indication (E) - The last route in the Route Advance Group is indicated with data "0". All others are indicated with data "1" (more).

Route Number (RT) - External Route Number. The Physical Route Number to be selected as programmed in ARTD.

Skip Digits (SKIP) - The amount of digits to be skipped when translating the dialed digits as defined in the NPC parameter of the AFRS command. The SKIP value should, at a minimum, include the LCR(S) Access Code. For example, if ARTD is assigned as shown (CDN 13-AC = 1) and ASPA is assigned as shown [ACC = 9, SRV = "LCR: Least Cost Routing" / "LCRS: Register Sender LCR"], then assign "1" to skip the LCR(S) Access Code.

Note: The SKIP parameter removes the amount of digits defined from the beginning of the dial number.

Pattern Number Location (PNL) - If digits are to be added after translating the dialed digits in AFRS, assign a PNL (1-999). The digits to be added are programmed to the respective PNLs in AADC. If no digits are to be added, PNL = 0 should be assigned.

Note: Take into consideration the value entered in the SKIP parameter when assigning a PNL greater than "0".

Overflow Tone (OVFT) - If tone is required to be heard when an outgoing call is placed over the last route (RA) of an OPR, assign the data as "1". If tone is not required, assign the data as "0".

Note: When activating Call Forwarding - Don't Answer - Outside [C-74]/Call Forwarding - Don't Answer - CCIS [C-47]/Call Forwarding - PS Incoming Incomplete, the Overflow Tone is not provided.

Priority Restriction Class (PRSC) - If the Restriction Class, referred to in the AEFR command is required, assign PRSC using Classes 1-15. If it is not required, enter data "0".

STEP 12: **AADC** - If additional digits are required, assign the digits to Pattern Number Location (PNL) here. In the PNL field, assign a Pattern Number Location value from 1-999.

Note: This should be the same PNL Number that was assigned in AOPR. For the DC field assign the additional Digit Information. A maximum of 24 digits (0-9, #, *, P) can be assigned to each PNL. The digits entered in the DC field will be added in front of the dialed digits (with the exception of this field being used for the PFX parameter of the ASDC command).

STEP 13: **ASDC** - If six-digit translation is required, whereby both Area and Office Codes must be examined, this command must be used. If no assignment is made in this command, all NXXs are allowed. After an NXX is assigned as "allowed", all other NXXs will subsequently be denied. The Table becomes an Allow Table. Program each Office Code that is to be allowed with EFFECT = 1.

The number of NXXs to be either allowed or denied will determine the method of programming used. For example, if it is determined that only 555 is to be restricted from a Special Common Carrier, it is more desirable to allow all NXXs and restrict only this NXX. The method would be as follows:

To create a Deny Table assign the following:

Step a): For KIND (Kind of Selection), select either Prefix, FX-Foreign Exchange (PFX) or Special Common Carrier (SCC).

For Time of Day Change Pattern Number (TDPTN) enter a value, from 0-7 (this value should correlate to a predefined pattern already assigned in ATCP) that should also assigned to the applicable OPR in AOPR.

For Outgoing Route Selection Pattern Number (OPR), assign the number of the OPR that this table will serve.

For Route Advance Order (RA), enter a value (from 0-7) that specifies the route in AOPR, that this table will serve.

For OFFICE (Office Code), enter 555 for this example.

Next click 'GET'.

Continue by selecting 'ALL' for Office Code Flag (OFLG) and enter "1" for EFFECT.

Step a, allows all NXXs for the specified OPR and RA, thus making a Deny Table.

Step b): For KIND, TDPTN, OPR, RA, and OFFICE, enter the same data from "Step a".

Next click 'GET'.

Continue by selecting 'INPUT' for OFLG and enter "0" for EFFECT.

Note: Step b restricts 555 only.

When assigning for PFX, each NXX is assigned as the following:**Note 5**
EFFECT:0 = Does not require a prefix.
EFFECT:1 = Allow, with 1 + dialing **Note 6**
EFFECT:2 = Only for display.
When assigning for SCC, each NXX is assigned as the following:
EFFECT:0 = Restricted. The Office Code is not allowed (for the Area Code that is dialed) for the specified OPR and RA.
EFFECT:1 = Allowed. The Office Code is allowed (for the Area Code that is dialed) for the specified OPR and RA.
EFFECT:2 = Only for display.

Note 5: When PFX, EFFECT = 1 is assigned. Add the required digit in AADC.

Note 6: For this command to work, it is very important that the NPC field of the AFRS command be programmed correctly. The "OFFICE" parameter, in ASDC will be the next three digits after those assigned in AFRS.
See the example, below.
Example:AFRS RT 31 = 91214 OPR 1
ASDC checks next three digits: 91214 XXX XXXX

STEP 14: **ATCP** - A maximum of eight Time-of-Day Pattern changes (0-7) are available, whereby the system will refer to different OPR assignments as a function of time. As a result, different Routes will be listed per OPR and therefore different assignments in ASDC. Assign the following:
Times Setting
FROM: In Military Time, and in 30 minute increments, on the hour and half hour.
TO:
PATTERN NUMBER: 0-7

STEP 15: **ARTI** - Assignment of Trunk Application Data - Assign DCAN (CDN: 54) = 1 for Tie line routes that use LCRS, and require that call processing proceed, when digits received are less than AMND.

Note: This requires that ARTD, CDN-1 is not programmed as a "1" or "12" and CDN-15 is a "2". Step 15 is optional.

Note: To change the data for existing route (RT), initialize the circuit card.

FCCS Programming

This data setting describes only the additional or replacement commands used to activate FCCS Services. When used as an FCCS feature, these commands replace the non-network commands (for example, AAED is replaced by AAEDN). Note that the data setting for FCCS Service varies depending on the Node.

Note: When using ASYDL or ASYDN to program an FCCS feature, the basic programming of ASYD is required in addition to ASYDL or ASYDN.

STEP 1: **ALRTN** - Assign this command to connect the Logical Route to the Physical Route.

STEP 2: **ARTDN** - Assign the Route Class Data to the Logical Route that is used for the LCR/LCRS Flag Route.

6-TCL: 1 7-L/T: 1 8-RLP: 2 9-TQ: 1 13-AC:1

Note: To change the data for existing route (RT), initialize the circuit card.

STEP 3: **ANPDL/ANPDN** - Reserve a level, usually "9", for LCR/LCRS access. Select Normal Service (N) or Hooking Service (H) from the combo box for Connection Indexes. Number of Necessary Digits (NND) is "1" and Busy Lamp Field (BLF) is not activated.

STEP 4: **ASPAL/ASPAN** - Assign the LCR/LCRS Access Code. Access Code to the ACC field (usually "9"). For SRV (Kind of Service), assign SRV = "LCR: Least Cost Routing" or SRV = "LCRS: Register Sender LCR". Assign the Logical Route Number for the LCR(S) Flag Route, usually "31", that is associated with this Access Code to the RT field. Assign Connection Indexes, CI = "N: Normal" and CI = "H: Hooking". For Second Dial Tone, assign data as "1". If Procedure 2 is used for AUTHORIZATION CODE [A-20], assign the data as "1" to the AH parameter.

STEP 5: **ARNPL** - Assign the Logical Route Numbers as defined in ARTDN. Relate these routes to the LCR(S) Access Code by assigning the value, previously defined in ASPAL/ASPAN (for SRV = "LCR: Least Cost Routing") to the ACC field of ARNPL. The LCR/LCRS Access Code is usually "9". The routes to be assigned will be those routes listed in the various OPRs of AOPRL/AOPRN.

Note: These same routes may also be listed in ASPAL/ASPAN for SRV = "OGC: Outgoing Call". In that case, these routes may also be available via Direct Dial Access Code. In ARNPL, these Access Codes are not assigned to implement the LCRS service in this command.

STEP 6: **ARSCN** - Assign the Route Restriction Class for all the Logical Route Number.

STEP 7: **AFRSL/AFRSN** - Allocate the dial number necessary to determine the OPR used to select the routes.

STEP 8: **AOPRL/AOPRN** - AFRSL/AFRSN designates the dialing sequence and the OPR to be selected. The AOPRL/AOPRN defines the routes and in what order these routes will be selected. Assign the following:

TDPTN Number (TDPTN) - Time of Day Change Pattern Number. Time of Day Change Pattern data can be applied to an OPR. Assign a value, from 0 to 7, that correlates to a predefined Time of Day Change Pattern Number assigned in the ATCP command (and which is to be applied to this OPR).

Note: If the Time of Day feature is not applicable, assign a value of "0" in the TDPTN field for this OPR.

OPR Number (OPR) - Outgoing Route Selection Pattern Number. Enter the number referenced in AFRSL/AFRSN. OPR Numbers range from 0-4000.

Route Advance Number (RA - Route Advancing Order) - The order in which a route will be selected (0-7).

For example - RA 0 = Band 1, RA 1 = Band 5, RA 2 = DDD.

Note: Always build your OPR beginning with an RA of "0". Continue assigning subsequent Route Advances numerically.

Route Advance End Indication (E) - The last Route in the Route Advance Group is indicated with data "0" (Ended). All others are indicated with data "1" (more).

Logical Route Number (LGRT) - The Logical Route Number to be selected as programmed in ARTDN.

Skip Digits (SKIP) - The number of digits to be skipped from Dialed Numbers.

Pattern Number Location (PNL) - If digits are to be added after translating the dialed digits in AFRSL/AFRSN, assign a PNL (1-999). The digits to be added are programmed to the respective PNLs in the AADCL/AADCN commands. If no digits are to be added, "0" should be entered.

Overflow Tone (OVFT) - If tone is required to be heard when an outgoing call is placed over the last route (RA) of an OPR, assign the data as "1". If tone is not required, assign the data as "0".

Note: When activating Call Forwarding - Don't Answer - Outside [C-74]/Call Forwarding - Don't Answer - CCIS [C-47]/Call Forwarding - PS Incoming Incomplete, the Overflow Tone is not provided.

Priority Restriction Class (PRSC) - If the Restriction Class defined in AEFR is required, assign PRSC using Classes 1-15. If it is not required, enter data "0".

STEP 9: **AADCL/AADCN** - If the additional digits are required, assign the digits to Pattern Number Location (PNL) here. In the PNL field, assign a Pattern Number Location value from 1-999.

Note: This should be the same PNL Number that was assigned in AOPRL/AOPRN. For the DC field, assign the additional Digit Information. A maximum of 24 digits (0-9, #, *, P) can be assigned to each PNL. The digits entered in the DC field will be added in front of the dialed digits (with the exception of this field being used for the PFX parameter of the ASDC command).

STEP 10: **ASDCL/ASDCN** - If six-digit translation is required, whereby both Area and Office Codes must be examined, this command must be used. If no assignment is made in this command, all NXXs are allowed. After an NXX is assigned as "allowed", all other NXXs will subsequently be denied. The Table becomes an Allow Table. Program each Office Code that is to be allowed with EFFECT = 1.

The number of NXXs to be either allowed or denied will determine the method of programming used. For example, if it is determined that only 555 is to be restricted from a Special Common Carrier, it is more desirable to allow all NXXs and restrict only this NXX. The method would be as follows:

To create a Deny Table assign the following:

Step a: For KIND (Kind of Selection), select either Prefix, FX-Foreign Exchange (PFX) or Special Common Carrier (SCC).

For Time of Day Change Pattern Number (TDPTN) enter a value, from 0-7 (this value should correlate to a predefined pattern already assigned in ATCP) that is also assigned to the applicable OPR in

AOPRL/AOPRN.

For Outgoing Route Selection Pattern Number (OPR), assign the number of the OPR that this table will serve.

For Route Advance Order (RA), enter a value from 0-7 that specifies the route, in AOPRL/AOPRN, that this table will serve. For OFFICE (Office Code), enter 555 for this example.

Next click 'GET'.

Continue by selecting 'ALL' for Office Code Flag (OFLG) and enter "1" for EFFECT. Step a, allows all NXXs for the specified OPR and RA, thus making a Deny Table.

Step b: For KIND, TDPTN, OPR, RA, and OFFICE, enter the same data from Step a.

Next click 'GET'.

Continue by selecting 'INPUT' for OFLG and enter "0" for EFFECT.

Note: Step b restricts 555 only.

When assigning for PFX, each NXX is assigned as the following:

EFFECT:0 = Not require a prefix

EFFECT:1 = Allow, with 1 + dialing (system adds the 1) **Note 7**

EFFECT:2 = Not allowed.

When assigning for SCC, each NXX is assigned as the following

EFFECT:0 = Restricted. The Office Code is not allowed (for the Area Code that is dialed) for the specified OPR and RA.

EFFECT:1 = Allowed. The Office Code is allowed (for the Area Code that is dialed) for the specified OPR and RA.

EFFECT:2 = Only for display.

Note: When PFX, EFFECT = 1 is assigned. Add the required digit in AADCL/AADCN.

Note 7: For this command to work, it is very important that the NPC field of the AFRSL/AFRSN command be programmed correctly. The "OFFICE" parameter in ASDCL/ASDCN, will be the next three digits after those assigned in AFRS. See the example below.

Example: AFRSL/AFRSN RT 31 = 91214 OPR1

ASDCL/ASDCN checks next three digits:91214 XXX XXXX

Office

L-6 Last Number Called - Single-Line Station

General Description

When a station user originates a station-to-station call or an outgoing trunk call, but the call is not established, this feature allows the calling station to recall the same destination by dialing only a Special Code instead of dialing all the digits of the number.

Operating Procedure

1. Lift the handset. Receive Dial Tone.
2. Dial the desired Party's Number.
3. The call has not been established.
4. Busy Tone or Reorder Tone is heard.
5. Hang up.
6. Lift the handset. Receive Dial Tone.
7. Dial the LAST NUMBER CALL Access Code (a maximum of three digits).
8. The system will automatically redial the last number dialed from that station.

Service Conditions

1. The number of dial digits to which this service is applicable is 18 maximum.
2. Dialing any digits after going off-hook causes previously stored digits to be erased from memory and the new digits to be programmed into memory.
3. In a STEP CALL, the final combination of digits is stored in memory. For example, dial 201, 201 is busy. Dial 2, Station 202 rings. Station 202 is stored in the LAST NUMBER CALLED memory.
4. In CALL FORWARDING [C-2, 3, 5], CALL PICKUP - GROUP [C-7], and STATION HUNTING [S-7, 8, 9], the number stored is the number dialed (the number displayed in the INTERMEDIATE STATION NUMBER DISPLAY -D [I-10D]).
5. This feature can be activated when the station accesses STATION-TO-STATION CALLING [S-11] or station-to-trunk calling.
6. SPEED CALLING - SYSTEM [S-3], SPEED CALLING - STATION [S-21] (when the originating party is other than station), or SPEED CALLING - GROUP [S-23] numbers (abbreviated Digit Codes) can be stored in the LAST NUMBER CALLED memory.
7. An AUTHORIZATION CODE [A-20] will not be stored, and must be entered each time the LAST NUMBER CALLED is accessed.
8. When the station that has set the SPLIT CALL FORWARDING - ALL CALLS for an incoming call from an outside party is restricted from call origination according to data assignment in ASYD, SYS2, Index 6, Bit 4, both of the call originations to an internal station and an outside party by using this feature are restricted.

   For the station that has been set SPLIT CALL FORWARDING - ALL CALLS for station-to-station calls only:

   - Call origination to an internal station by using this feature is not available.
   - Call origination to an outside party by using this feature is available.

   For the station that has been set SPLIT CALL FORWARDING - ALL CALLS for outside calls only:

   - Call origination to an internal station by using this feature is not available.

- Call origination to an outside party by using this feature is not available.

9. Only when a station is vacant, last number redial can work.

Interactions

LAST NUMBER CALLED will operate for both internal and outgoing calls.

Programming

STEP 1: **ANPD** - Reserve a Number Level for Feature Access and Cancel. Select Normal Service (N), Hooking Service (H) or Busy Service (B) from the combo box for Connection Indexes. (applicable to any feature assigned to this level). Assign NND in accordance with a predetermined Numbering Plan.

STEP 2: **ASPA** - Assign an Access Code to LAST NUMBER CALLED, SID = 44. Assign Connection Status Index (CI) for "N: Normal" service.

STEP 3: **ASFC** - Assign the stations that will receive LAST NUMBER CALLED a Service Feature Restriction Class that allows SFI = 44 (Last Number Call Redial).

FCCS Programming

This data setting describes only the additional or replacement commands used to activate FCCS services. When used as an FCCS feature, these commands replace the non-network commands (for example, AAED is replaced by AAEDN). Note that the data setting for FCCS service varies depending on the node.

STEP 1: **ANPDL** - Reserve a Number Level for Feature Access and Cancel. Select Normal Service (N), Hooking Service (H), or Busy Service (B) from the combo box for Connection Indexes. (applicable to any feature assigned to this level). Assign NND in accordance with a predetermined Numbering Plan.

STEP 2: **ASPAL** - Assign an Access Code to LAST NUMBER CALLED, SID = 44. Assign Connection Status Index (CI) for "N: Normal" service.

L-6D Last Number Called -D

General Description

This feature allows the user of a Desktop terminal to store the last five numbers dialed, and to redial the numbers by pressing the Redial call. The Desktop terminal user can choose any destination out of the last five calls that have been stored in memory, thus enabling the user to place a call without redialing the full number.

Operating Procedure

How to check the Dialed Numbers stored in the memory (up to 5 destinations)

1. Press the REDIAL key on a Desktop terminal.
   a.) The Prime Line lights green.
   b.) The Speaker Lamp lights red.
   c.) Receive Dial Tone from the speaker with the following display on the Desktop terminal **Note 1**.

LNR [*] / SPD [_ _] - 1			
			811625643
MIC	FDB	FDN	>>>

Note 1: Redial Access Code (LNR [*]) can be changed by ASYD, SYS1, Index 384, Bit 3. For details, see Programming.

2. Press the REDIAL key on the Desktop terminal again.
   a.) Receive Dial Tone from the speaker with the following display on the Desktop terminal.

LNR [*] / SPD [_ _] - 2			
			25191
MIC	FDB	FDN	>>>

Note: The indication is changed in sequence by pressing the Redial key. Once the stored Telephone Number display reaches LNR[*] or LNR[#]/SPEED[_ _]-5, LNR[*] or LNR[#]/SPEED[_ _]-1 appears again by pressing Redial.

How to originate a call

1. Press the Redial key on a Desktop terminal.
   a.) The Prime Line lights green.
   b.) The Speaker Lamp lights red.
   c.) Receive Dial Tone from the speaker with the following display on the Desktop terminal.

LNR [*] / SPD [_ _] - 1			
			811625643
MIC	FDB	FDN	>>>

2. Repeat the above procedure until you reach your desired destination.

LNR [*] / SPD [_ _] - 4			
			82625191
MIC	FDB	FDN	>>>

3. Press "*" to originate your call while the desired destination is being displayed on the Dterm **Note 2**.
   a.) Receive Ringback Tone from the speaker with the following display on the Desktop terminal.

```
                    82625191
   10:24  AM  FRI  15
```

Note 2: "*" button can be changed by ASYD, SYS1, Index 384, Bit 3. For details, see Programming.

4. Lift the handset when the Called Party answers the call.
   a.) The Prime Line lights green.
   b.) The Speaker Lamp is OFF.

```
                    82625191
   10:24  AM  FRI  15
```

Service Conditions

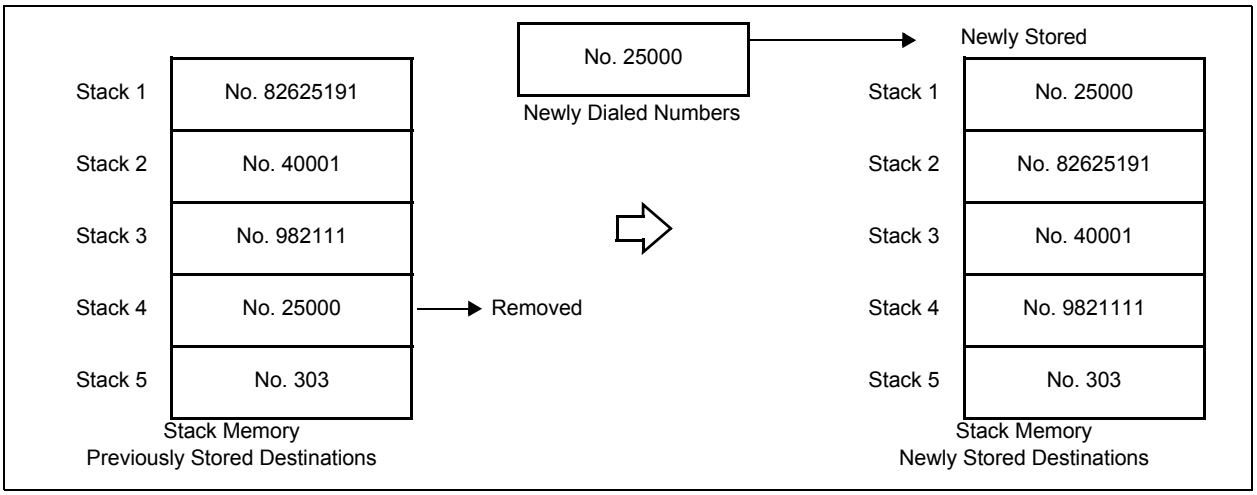
1. A maximum of five destinations can be stored.
2. The maximum digits of the destination is 32-digits.
3. When a Feature key has been assigned as Last Number Called, the Feature key also works as REDIAL key.
4. This feature is only effective when the Calling Party presses the REDIAL key while the Desktop terminal is in the off-hook or idle status.
5. When the REDIAL key is pressed, * or the REDIAL key must be pressed within 12 seconds. Otherwise, the Calling Party hears Reorder Tone.
6. The Calling Party can enter Abbreviated Codes after pressing the REDIAL key.
7. Any number entry (other than #) from a Desktop terminal after pressing the REDIAL key is recognized as Abbreviated Code (one or two digits) of SPEED CALLING - STATION.
8. A Desktop terminal can display the upper 16-digits among the stored 32 digits.
9. When any destination has not been stored in the system, which occurs after initialization of the circuit card, the Dterm displays no destination as shown below.

```
        LNR [*] / SPD [_ _] - 1

MIC          FDB          FDN          >>>
```

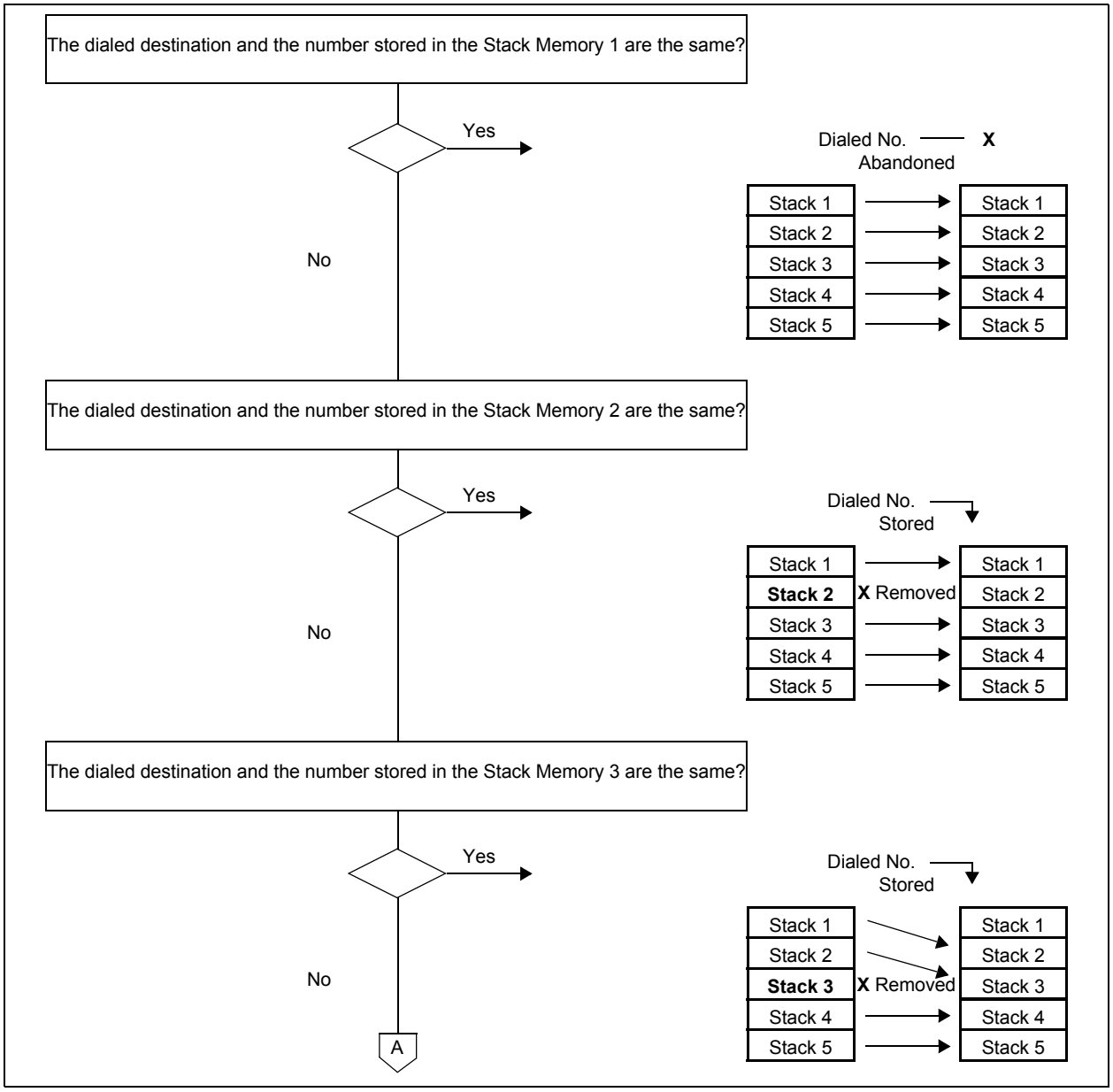
10. Only when a station is vacant, last number redial can work.

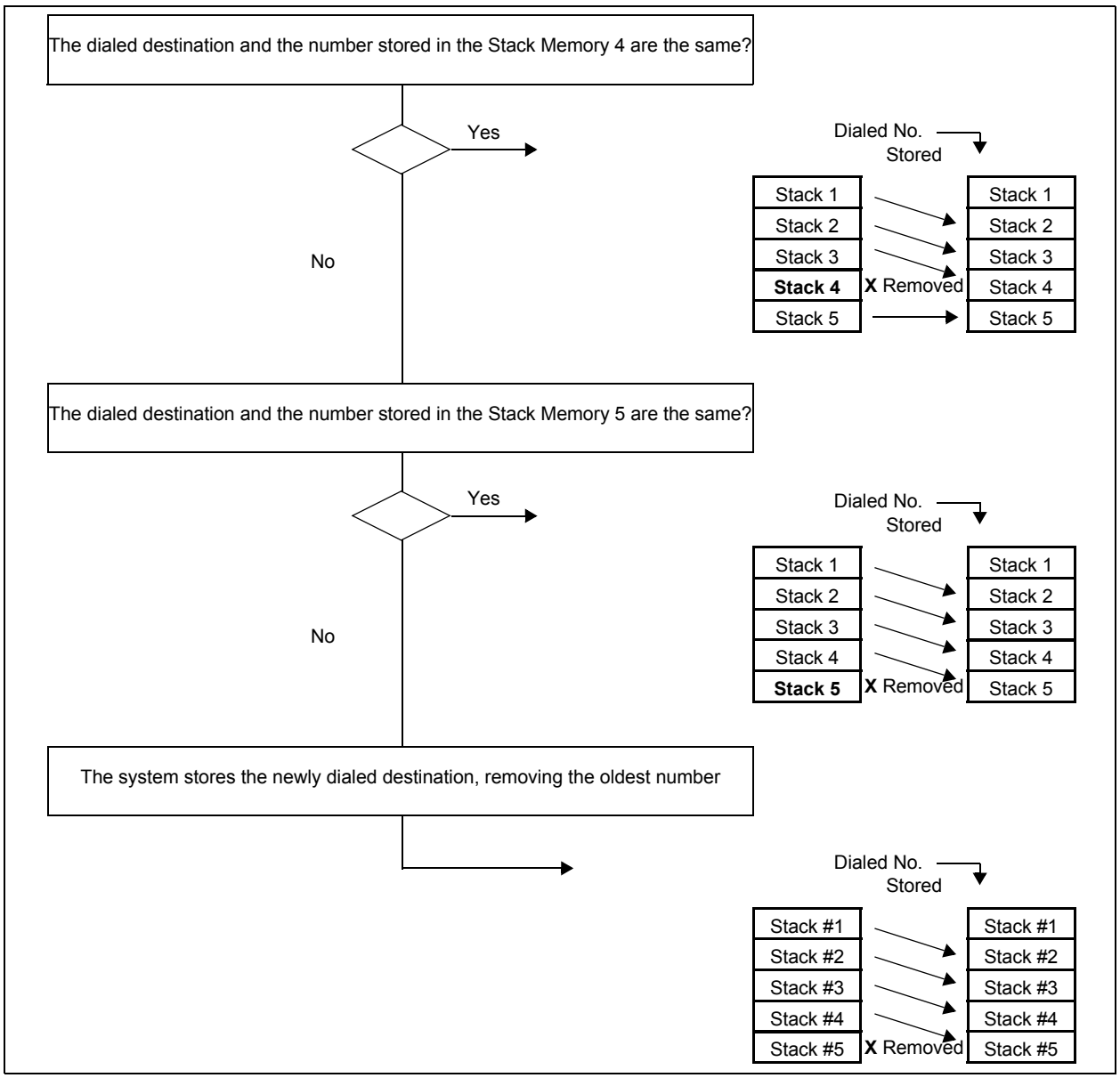
11. When a Dterm user dials the same destination that has already been stored in the system, the newly dialed destination is stored in the following manner:



12. The number dialed during the call can be stored as the LAST NUMBER CALLED - stock redial with a maximum of 32 digits.

13. If making an abbreviated call using the REDIAL key, the number dialed is not stored. Therefore, redialing through LAST NUMBER CALLED - stock redial is not available. A newly dialed destination is stored in accordance with the following flowchart:





14. When the following System Data (ASYDL/ASYDN, SYS1, Index 1257, Bit 2=1) is assigned, the Redial feature is performed for DT800/DT700/DT400/DT300 Series instead of this feature. For details of Redial feature, refer to Redial [R-60] in this manual.
15. While the Cordless Terminal (DT330 (12BT)) is used, when "*" button on the Bluetooth Handset is pressed after pressing the REDIAL key, the call is originated from the speakerphone on the Bluetooth Cradle.
16. To use this feature on the Bluetooth Handset of the Cordless Terminal (DT330 (12BT)), assign the LAST NUMBER CALLED Feature key (FKY=8) to the Line/Feature key between1-8 by using the AKYD command.

Programming

STEP 1: **ASYD** - SYS1, Index 384, Bit 3. Redial Access Code (LNR)
[When you assigned SYS1, Index 64, Bit 3=0 (for μ law countries)]
0/1 = "*" key / "#" key
[When you assigned SYS1, Index 64, Bit 3=1 (for A law countries)]
0/1 = "#" key / "*" key

STEP 2: **AKYD** - Assign a programmable Line/Feature key as the LAST NUMBER CALLED key, if necessary.
KYI: Service Index 1 (Feature Key)
FKY: 8 (Last Number Dialed)

STEP 3: **ASDT** - Assign a Desktop terminal with Telephone Equipment Class (TEC) 12.

L-7D Line Reconnect - Same-Line -D

General Description

This feature allows a user to disconnect a call and receive Dial Tone by pressing the RECALL key.

Operating Procedure

While a call is in progress

1. Press the RECALL key. Disconnect and receive Dial Tone on the same line.
2. User is ready to dial again.

Service Conditions

1. Conflicting features are: CALL TRANSFER - ALL CALLS [C-11], CONSULTATION HOLD - ALL CALLS [C-17] and CALL HOLD [C-6]. Pressing the RECALL key will result in reconnection of the held line.
2. System should be programmed to provide for first party release.
3. If the system is not programmed for first party release and programmed as calling, called, or both party release, a Release Guard is activated. If a party is not allowed to terminate a call and presses the RECALL key, that station remains connected to the original call.

Programming

ASYD - SYS1, Index 4, Bit 2 and Bit 3.
Releasing method for STATION-TO-STATION CALLING service

Bit 3	Bit 2 = Calling party release
0	0
Bit 3	Bit 2 = Called party release
0	1
Bit 3	Bit 2 = First party release
1	0
Bit 3	Bit 2 = Both party release
1	1

L-8D Line Reconnect - Other-Line -D

General Description

This feature allows a user to select another Line key without going on-hook.

Operating Procedure

While off-hook on the original line, press a Line key that is either idle or ringing. The user is disconnected from the original line and connected to the new line.

Service Conditions

1. LINE RECONNECT - OTHER-LINE -D cannot be used with a line currently in use, unless PRIVACY RELEASE [P-18] is in service (LED lit steady). LINE RECONNECT - OTHER-LINE -D can be used when the other line is on NON-EXCLUSIVE HOLD -D [N-7D].
2. System data programming should provide for first party release.

Programming

ASYD - SYS1, Index 4, Bit 2 and Bit 3.
Releasing method for STATION-TO-STATION CALLING service

Bit 3	Bit 2 = Calling party release
0	0
Bit 3	Bit 2 = Called party release
0	1
Bit 3	Bit 2 = First party release
1	0
Bit 3	Bit 2 = Both party release
1	1

L-9D Line Pre-selection -D

General Description

This feature provides the station user with the ability to select an idle or ringing line before going off-hook.

<Line Pre-selection for Terminal Using Terminal Phonebook>
A Desktop terminal, if employing terminal phonebook, can preselect a ringing or held line as well as display the party's data registered on terminal phonebook. The display lasts while Line Pre-selection works, or 10 seconds.

Operating Procedure

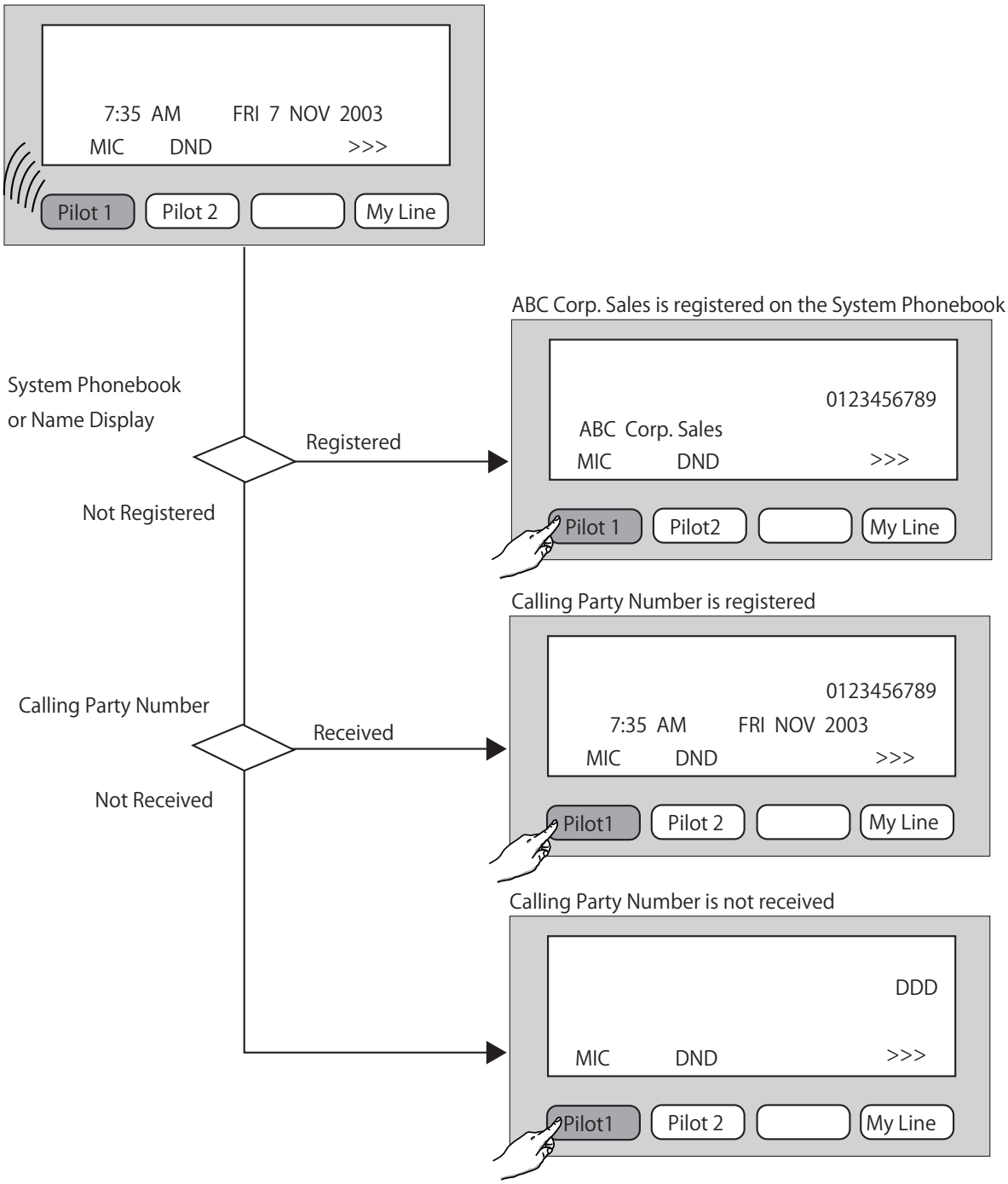
To originate

1. Press the desired Line key. Receive Dial Tone.
2. Dial the desired destination.

To answer

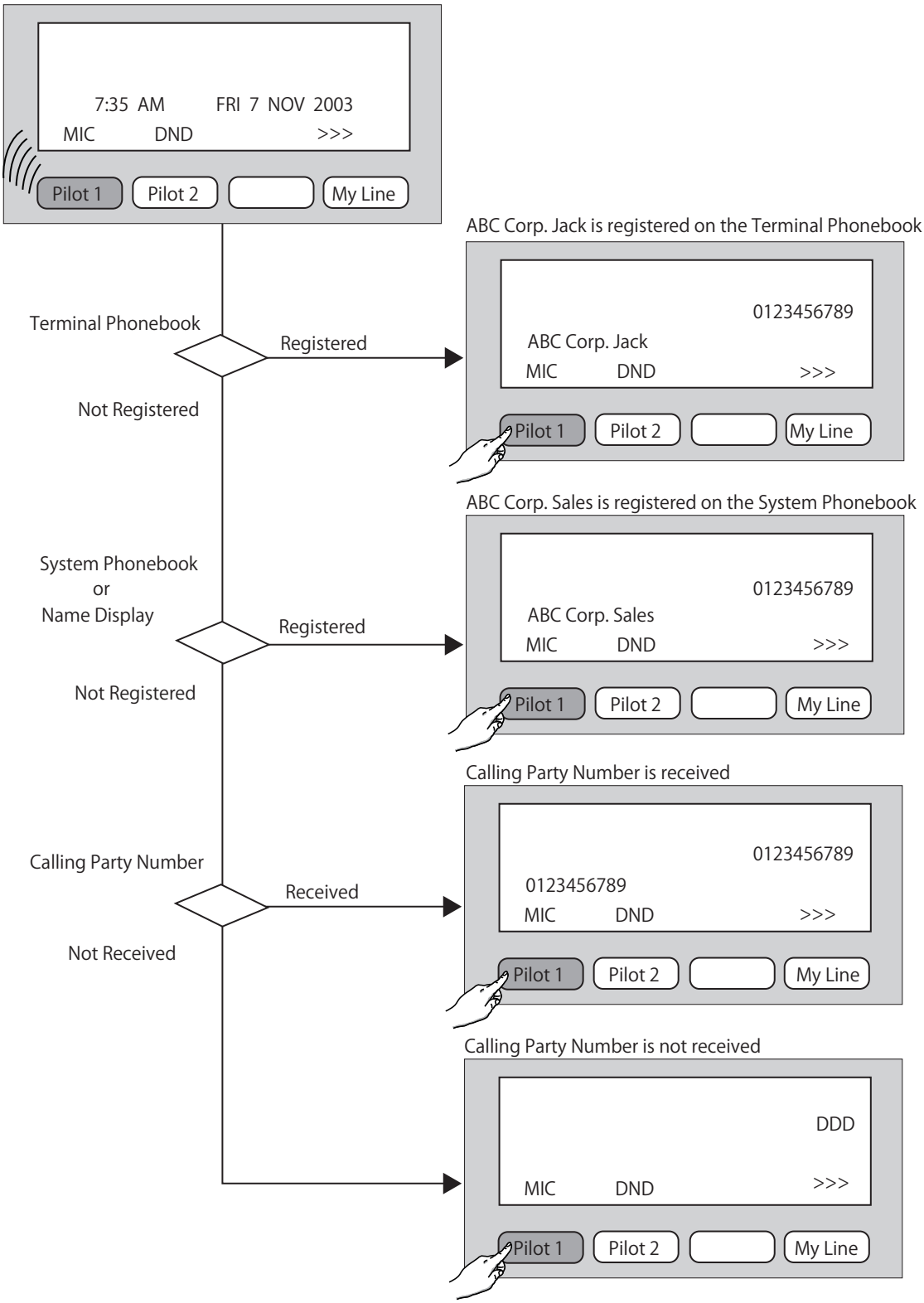
1. Press the Line key (ringing and flashing LED alerts incoming call).
2. Lift the handset or press the SPEAKER key. Answer the incoming call.

Example: Pre-selecting an incoming sub-line



Note: When DT800/DT700 Series is used for this feature, refer to the next page (Example: Pre-selecting an incoming sub-line (Terminal Phonebook is used)).

Example: Pre-selecting an incoming sub-line (Terminal Phonebook is used)



Note: Desktop terminals which have Terminal Phonebook built-in are only available for this feature.

Service Conditions

1. A Line key whose associate LED is lit steady cannot be interrupted by pressing its key, unless PRIVACY RELEASE is in service.
2. After the desired Line key is pressed, LINE PRESELECTION -D will remain in effect for 10 seconds. After 10 seconds, Line selection returns to the primary Line.

<Line Pre-selection for Terminal Using Terminal Phonebook>

1. After Line Pre-selection expires with displayed information (10 seconds), Prime Line information appears if Prime Line receives an incoming call.
2. A Desktop terminal with terminal phonebook displays a party's information for 10 seconds based on the following priority order: terminal phonebook; system phonebook; name display; caller ID.
3. Displayed information lasts for 10 seconds even if the party abandons the call or another terminal answers the call before Line Pre-selection expires.
4. A Desktop terminal does not display a party's information when pre-selecting a call from C.O. line which does not include the caller ID.
5. If you pre-select an idle line, displayed information of a party is cleared with the following exception: if Prime Line receives an incoming call, its information is displayed (only when the following data is set: ASYD, SYS1, Index 390, Bit 2=1: Caller ID display on the bottom line of LCD is in service). In this case, the corresponding data of system Phonebook or user name retrieved from the system is displayed, but not the data of the terminal phonebook.
6. After you pre-select an idle line on a Desktop terminal with terminal phonebook, if an incoming call arrives in the line within 10 seconds, the party's information is displayed (only when the following data is set: ASYD, SYS1, Index 390, Bit 2=1: Caller ID display on the bottom line of LCD is in service). In this case, the corresponding data of system Phonebook or user name retrieved from the system is displayed. Pre-select the line again to switch the display to data of terminal phonebook.

Programming

No programming is required. The following is only for displaying a party's data on a Desktop terminal with terminal phonebook.

Note: See [P-83] Phonebook to check the service conditions before assigning the following.

STEP 1: **ASYDL** - SYS1, Index 679, Bit 0=1 (Terminal phonebook display on Line Pre-selection is in service)
STEP 2: **ASFC** - SFC = 1-15, SFI = 247=1

L-10 LCR - Time Of Day Routing

General Description

This feature provides automatic routing of outgoing calls over alternative customer facilities, based on the Destination Code. The system will select the most economical route available at the time of connection. The pattern of alternate routing can be changed up to 8 times per day, based on a prearranged time schedule.

Operating Procedure

To operate

a.) Lift the handset. Receive Dial Tone.
b.) Dial the Trunk Access Code (usually "9"). Receive Second Dial Tone.
c.) Dial the Area Code, Office Code, and Subscriber Number.
d.) The system automatically completes the call via the most economical route available.

Service Conditions

1. A maximum of 8 patterns can be provided during a 24-hour interval for LCR - TIME OF DAY ROUTING.
2. Each pattern can be programmed at a maximum increment of 30 minutes.

Interactions

1. For each Time-of-Day Change, the entire AOPR and ASDC tables must be created. Caution should be exercised to use data memory economically.
2. LEAST-COST ROUTING-3/6 DIGIT [L-5] must first be provided.

Programming

STEP 1: **AOPR** - The AFRS command designates the dialing sequence and the OPR to be selected. The AOPR defines the routes and in what order these routes will be selected. Assign the following:

TDPTN - Time-of-Day Pattern Change. If for an AFRS assignment, different OPRs will be required as a function of time; designate 1-7 to which Time-of-Day Pattern Change Number (TDC NO) this OPR is assigned.

OPR No. (OPR) - Outgoing Pattern Route. The number referenced in AFRS, consisting of a series of routes to be chosen based on number dialed.

Route Advance Number (RA) - The order in which a route is selected. (0-7).

Route Advance End Display (E) - The last route in the Route Advance Group. Indicate last route with data "0".

Route Number (RT) - The Physical Route Number to be selected.

Skip Digits (SKIP) - The digits to be skipped when translating the dialed digits in AFRS. If ARTD is assigned as shown, at a minimum, always assign "1" to skip the LCR (S) Access Code.

Pattern Number Location (PNL) - If digits are to be added after translating the dialed digits in AFRS, go to the AADC command.

Overflow Tone (OVFT) - Assign if a tone is wanted for the Last Choice Route.

Priority Restriction Class (PRSC) - If this is required, assign PRSC here; classes 1-15.

STEP 2: **ASDC** - If 6-digit translation is required where both the Area and Office Codes must be examined, this command must be used. The system is automatically set as an Allow table. To create a Deny table, assign the following:

OFFICE: 0
EFFECT: 1
Program the type of route as PFX (Prefix, Foreign Exchange, FX) or as SCC (Special Common Carrier).
Choose TDC PTN, Time-of-Day Pattern. Go to ATCP.
Choose the OPR number this six-digit table will serve.
Choose the RA (Route Advance) Number to specify the route in the OPR.
When assigning for PFX, each NXX is assigned as the following:*
EFFECT: 0 = Allow, No 1 + dialing.
EFFECT: 1 = Allow with 1 + dialing (system adds the 1).
EFFECT: 2 = Not Allowed.
When assigning for SCC, each NXX is assigned as the following:
EFFECT: 0 = Restricted
EFFECT: 1 = Allowed
EFFECT: 2 = Not Used

Note: PFX, EFFECT: 1 is presently not available. Use the AADC command to add the required digits.

STEP 3: **ATCP** - A maximum of eight Time-of-Day Pattern changes (0-7) are available, whereby the system will refer to different OPR assignments as a function of time. As a result, different routes will be listed per OPR and will require different assignments in ASDC.
Assign the following:
Time Setting
TO:
FROM: (In military time, and only on the half hour and hour)
PATTERN NUMBER: (0-7).

FCCS Programming

This data setting describes only the additional or replacement commands used to activate FCCS services. When used as an FCCS feature, these commands replace the non-network commands (for example, AAED is replaced by AAEDN). Note that the data setting for FCCS service varies depending on the node.

STEP 1: **AOPRL** - The AFRS command designates the dialing sequence and the OPR to be selected. The AOPR defines the logical routes and in what order these logical routes will be selected. Assign the following:
TDPTN - Time-of-Day Pattern Change. If for an AFRS assignment, different OPRs will be required as a function of time; designate 1-7 to which Time-of-Day Pattern Change Number (TDC NO) this OPR is assigned.
OPR No. (OPR) - Outgoing Pattern Route. The number referenced in AFRS, consisting of a series of routes to be chosen based on number dialed.
Route Advance Number (RA) - The order in which a route is selected. (0-7).
Route Advance End Display (E) - The last route in the Route Advance Group. Indicate last route with data “0”.
Route Number (RT) - The Physical Route Number to be selected.

Skip Digits (SKIP) - The digits to be skipped when translating the dialed digits in AFRS. If ARTD is assigned as shown, at a minimum, always assign "1" to skip the LCR (S) Access Code.

Pattern Number Location (PNL) - If digits are to be added after translating the dialed digits in AFRS, go to the AADC command.

Overflow Tone (OVFT) - Assign if a tone is wanted for the Last Choice Route.

Priority Restriction Class (PRSC) - If this is required, assign PRSC here; classes 1-15.

STEP 2: **ATCPL** - A maximum of eight Time-of-Day Pattern changes (0-7) are available, whereby the system will refer to different OPR assignments as a function of time. As a result, different routes will be listed per OPR and will require different assignments in ASDC.

Assign the following:

Time Setting

TO:

FROM: (In military time, and only on the half hour and hour)

PATTERN NUMBER: (0-7).

L-11 Line Load Control - Attendant

General Description

This feature allows the Attendant to deny a preselected group of stations the ability to originate calls by activating a key. The sole intention of this feature is to temporarily restrict the least important internal stations from outward calling during an excessive traffic load or emergency conditions.

Operating Procedure

To activate from the ATTENDANT CONSOLE [A-3]

a.) Press a Loop key.
b.) Dial the LINE LOAD CONTROL Access Code. Receive Service Set Tone.
c.) Press the RELEASE key.

To cancel from the ATTENDANT CONSOLE [A-3]

a.) Press the Loop key.
b.) Dial the LINE LOAD CONTROL Cancel Code. Receive Service Set Tone.
c.) Press the RELEASE key.

Service Conditions

1. This feature is controlled entirely from the ATTENDANT CONSOLE [A-3].
2. Individual stations are programmed for this feature via CLASS OF SERVICE - INDIVIDUAL [C-15].
3. When LINE LOAD CONTROL [L-11, 20, 21] is set in the system, the message “LLC” will be output on the display of Attendant Console.
4. Incoming calls from a trunk can be restricted in system data (ASYD, SYS1, Index 59, Bit 0=1), regardless of Service Feature Restriction Class setting.
5. When LINE LOAD CONTROL is activated or canceled, a system message will be output via the PCPro.
6. When LINE LOAD CONTROL is activated, a single line station will receive Busy Tone [or “BUSY” is displayed on LCD for Dterm (no tone)] when going off-hook.
7. A station in LINE LOAD CONTROL [L-11, 20, 21] can receive incoming calls.
8. Incoming calls terminated via CCIS/ISDN trunks or MG-COT cannot be restricted. Incoming calls terminated via FCCS cannot be restricted as well.
9. The incoming calls from a PS Line or an SP Controlled SIP terminal accommodated on programmable Line/Feature key of Desktop terminal can not be restricted.
10. The visitor PS cannot be allowed to originate a call during Line Load Control because SFI = 16 cannot be assigned to the Service Feature Restriction Class of the visitor PS.
11. When SOFT PHONE - PS [S-158] or SOFT PHONE WLAN HANDSET [S-159] is activated, outgoing calls cannot be restricted by LINE LOAD CONTROL.
12. When SOFT PHONE WIRED-PHONE HANDSET [S-147] is activated, the restriction for outgoing calls must be assigned in CLASS OF SERVICE - INDIVIDUAL [C-15] of the wired telephone.
13. Restriction by LINE LOAD CONTROL will be executed within each LP unit.

Interactions

1. Calls in progress are not affected when this feature is activated.
2. This feature is implemented on a per-system basis.
3. Only the Attendant Console can activate this feature by seizing a loop and dialing the Feature Access Code. A designated key cannot be assigned at the Attendant Console to control this feature.
4. Access to the system data via the PCPro is allowed while LINE LOAD CONTROL is active.
5. STATION-TO-STATION CALLING [S-11] to affected stations is not allowed during LINE LOAD CONTROL.
6. HOTLINE [H-1] and HOUSE PHONE [H-2] connections become inoperative during LINE LOAD CONTROL.

Programming

STEP 1: **ASYD** - SYS1, Index 59, Bit 0. Should the system restrict incoming calls during LINE LOAD CONTROL? 0/1: No/Yes.

Note: Incoming Calls terminated to CCIS/ISDN Trunks or MG-COT cannot be restricted.

STEP 2: **ANPD** - Reserve a Number Level for Feature Access and Cancel. Select Normal Service (N), Hooking Service (H) or Busy Service (B) from the combo box for Connection Indexes. (applicable to any feature assigned to this Level). Assign NND in accordance with a predetermined Numbering Plan.

STEP 3: **ASPA** - Assign an Access Code to LINE LOAD CONTROL. Assign SRV = "SSCA: Service Code Appendix", SIDA = 46 for Line Load Control Start. Assign SRV = "SSCA: Service Code Appendix", SIDA = 47 for Line Load Control Cancel. Assign a Connection Index, CI = "N: Normal".

STEP 4: **ASFC** - Assign a Service Feature Restriction Class that allows SFI 16 to the stations that will be unaffected by LINE LOAD CONTROL.

L-18 Long Line Trunk

General Description

This service allows accommodation of a station that has high loop resistance (long-line station). Long-line stations that have a maximum 1,500 ohms of loop resistance, including resistance of the telephone set, may be equipped by this feature.

Note: This feature is not available in North America and Australia.

Operating Procedure

No manual operation is required.

Service Conditions

1. Any number of long line stations may be used up to the total number of stations accommodated in the system.
2. PA-16LC BW or the equivalent is required for long line stations.
3. The following services are not available for long line stations:
   a.) MESSAGE WAITING LAMP
   b.) GROUND BUTTON ACCESS
   c.) Parallel (Branch) Connection
   d.) HOWLER TONE SENDING.
4. No compensation of speech level is provided for long line stations.

L-20 Line Load Control - Automatic

General Description

This feature allows the system to automatically deny a preselected group of stations the ability to originate calls. The sole intention of this feature is to temporarily restrict the least important internal stations from outward calling during an excessive traffic load or emergency conditions.

Operating Procedure

No manual operation is required.

Service Conditions

1. Individual stations are programmed for this feature via CLASS OF SERVICE - INDIVIDUAL [C-15].
2. When LINE LOAD CONTROL [L-11, 20, 21] is set in the system, the message "LLC" will be output on the display of Attendant Console.
3. Incoming calls from a trunk can be restricted in system data (ASYD, SYS1, Index 59, Bit 0=1), regardless of Service Feature Restriction Class setting.
4. When LINE LOAD CONTROL - AUTOMATIC [L-20] is activated or canceled, a system message will be output via the PCPro.
5. When LINE LOAD CONTROL is activated, a single line station will receive Busy Tone [or "BUSY" is displayed on LCD for Dterm (no tone)] when going off-hook.
6. A station in LINE LOAD CONTROL [L-11, 20, 21] can receive incoming calls.
7. Incoming calls terminated via CCIS/ISDN trunks or MG-COT cannot be restricted. Incoming calls terminated via FCCS cannot be restricted as well.
8. The incoming calls from a PS Line or an SP Controlled SIP terminal accommodated on programmable Line/Feature key of Desktop terminal can not be restricted.
9. The visitor PS cannot be allowed to originate a call during Line Load Control because SFI = 16 cannot be assigned to the Service Feature Restriction Class of the visitor PS.
10. When SOFT PHONE - PS [S-158] or SOFT PHONE WLAN HANDSET [S-159] is activated, outgoing calls cannot be restricted by LINE LOAD CONTROL.
11. When SOFT PHONE WIRED-PHONE HANDSET [S-147] is activated, the restriction for outgoing calls must be assigned in CLASS OF SERVICE - INDIVIDUAL [C-15] of the wired telephone.
12. Restriction by LINE LOAD CONTROL will be executed within each LP unit.

Programming

STEP 1: **ASYD** - SYS1, Index 56 and Index 57. Set LINE LOAD CONTROL-AUTOMATIC Execution (Index 56)/Execution cancel (Index 57) CPU Occupancy Rate. If no data is assigned, RAM Data 97% (Index 56) and 90% (Index 57) are used.

SYS1, Index 59, Bit 0. Incoming Trunk Call Restriction during LINE LOAD CONTROL 0/1: Not Required/Required

Note: Incoming Calls terminated to CCIS/ISDN Trunks or MG-COT cannot be restricted.

STEP 2: **ASFC** - Assign SFI = 16 to the Service Feature Restriction Class of the Stations that are to be able to originate calls during Line Load Control.

L-21 Line Load Control - Manual

General Description

This feature allows the PCPro to deny a preselected group of stations the ability to originate calls by activating a command key. The sole intention of this feature is to temporarily restrict the least important internal stations from outward calling during an excessive traffic load or emergency conditions.

Operating Procedure

The ALLC command is used to designate start and stop of LINE LOAD CONTROL - MANUAL [L-21].

Service Conditions

1. Individual stations are programmed for this feature via CLASS OF SERVICE - INDIVIDUAL [C-15].
2. When LINE LOAD CONTROL [L-11, 20, 21] is set in the system, the message “LLC” will be output on the display of Attendant Console.
3. Incoming calls from a trunk can be restricted in system data (ASYD, SYS1, Index 59, Bit 0=1), regardless of Service Feature Restriction Class setting.
4. When LINE LOAD CONTROL is activated, a single line station will receive Busy Tone [or “BUSY” is displayed on LCD for Dterm (no tone)] when going off-hook.
5. A station in LINE LOAD CONTROL [L-11, 20, 21] can receive incoming calls.
6. When LINE LOAD CONTROL - MANUAL [L-21] is activated or canceled, a system message will be output on the PCPro.
7. Incoming calls terminated via CCIS/ISDN trunks or MG-COT cannot be restricted. Incoming calls terminated via FCCS cannot be restricted as well.
8. Incoming calls from a PS Line or an SP Controlled SIP terminal accommodated on a programmable Line/ Feature key of a Desktop terminal cannot be restricted.
9. The visitor PS cannot be allowed to originate a call during Line Load Control because SFI = 16 cannot be assigned to the Service Feature Restriction Class of the visitor PS.
10. When SOFT PHONE - PS [S-158] or SOFT PHONE WLAN HANDSET [S-159] is activated, outgoing calls cannot be restricted by LINE LOAD CONTROL.
11. When SOFT PHONE WIRED-PHONE HANDSET [S-147] is activated, the restriction for outgoing calls must be assigned in CLASS OF SERVICE - INDIVIDUAL [C-15] of the wired telephone.
12. Restriction by LINE LOAD CONTROL will be executed within each LP unit.

Programming

STEP 1: **ASYD** - SYS1, Index 59, Bit 0. Incoming Trunk Call Restriction during LINE LOAD CONTROL 0/1: Not Required/Required.

STEP 2: **ASFC** - Assign SFI = 16 (Line Load Control) to the Service Feature Restriction Class of the stations that are to be able to originate calls during Line Load Control.

L-22 Line Fault Detection

General Description

This feature detects faulty conditions on exchange lines (Line Disconnection etc.), and indicates them as "Line Faults", restricting further seizure of the faulty line by other connections.

Note: This feature is only available in Australia.

Operating Procedure

No manual operation is required.

Service Conditions

1. A line fault may be detected on any exchange lines in any of the following cases:
   a.) Both Line A and Line B are short-circuited.
   b.) Line A is open.
   c.) Line B is open.
   d.) Connection of Line A and Line B is reversed.
2. If a line fault is detected on an exchange line in the idle state, the line fault lamp (yellow) on the package and the SUP lamp on the frame light up, and an appropriate message is output to the PCPro.
3. When the exchange line recovers from the line fault state, the line fault lamp goes off and the fault recovery message is output to the PCPro.
4. The SUP lamp on the frame should be turned off by means of either the RALM command or "ALM RST" key operation.
5. No exchange lines in the line fault state may be seized by new connections.

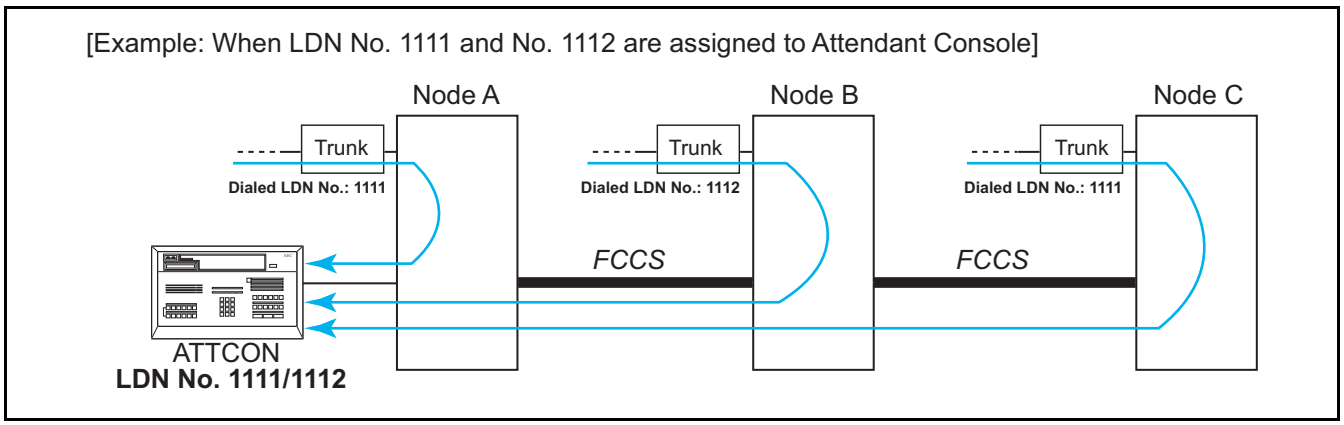
Programming

No programming is required for this feature.

L-24 Listed Directory Number

General Description

This feature allows Listed Directory Number display on the ATTENDANT CONSOLE [A-3] when the Attendant has answered a Listed Directory Number call.
In addition, this feature enables ATTENDANT CONSOLE to receive a Listed Directory Number (LDN) call, terminated at a remote node, via FCCS link.



Operating Procedure

1. Listed Directory Number (LDN) is dialed.
2. The Attendant at an ATTENDANT CONSOLE [A-3] answers the LDN call (DID, TIE).
   a.) The DEST lamp lights up.
   b.) The Listed Directory Number is displayed.
3. The Attendant presses the SRC key.
   a.) The DEST lamp goes out and the SRC lamp lights up.
   b.) The route/trunk number is displayed.

Service Conditions

1. This service is effective when the Attendant at an ATTENDANT CONSOLE [A-3] has answered a Listed Directory Number call.
2. If the Attendant presses the SRC key while the Listed Directory Number is displayed, the display changes to the calling party's route/trunk number. In this case, the Listed Directory Number of the call being handled at present cannot be displayed again.
3. While the Listed Directory Number is displayed, the Attendant can transfer the call to a desired station by entering the destination number on the keypad. In this case, the Listed Directory Number of the call being handled at present cannot be displayed again.
4. While the Listed Directory Number is displayed, the Attendant can place the present call on hold by pressing the HOLD key. In this case, the Listed Directory Number will not be displayed again when the Attendant returns to the call on hold.
5. While the Listed Directory Number is displayed, the Attendant can set CALL PARK [C-29] service. In this case, the Listed Directory Number of the call placed on CALL PARK [C-29] cannot be displayed again.
6. The LISTED DIRECTORY NUMBER should be assigned in system data.

<FCCS Conditions>

1. The user can program FPC of node with ATTCON, which receives LDN Calls, to each node by using ALDNN command.
2. When "LDN assigned by ALDN command" and "LDN assigned by ALDNN command" are mixed in a system, "LDN assigned by ALDNN command" takes precedence over "LDN assigned by ALDN command".
3. FPC of a Node which transmits Day/Night changeover information is determined by SYS1, Index 640, and the information is transmitted to a node of FPC designated by SYS1, Index 704.
4. When FCCS link between a node received a LDN call and a node with ATTCON is all busy, locked out, make busy or etc., the calling party receives Busy Tone.

Programming

STEP 1: **ASYD** - SYS1, Index 70, Bit 0. Called Number Display when Call Forwarding to ATTCON 0/1: Out/ In Service.

STEP 2: **ALDN** - Assign the LISTED DIRECTORY NUMBER.

TN:	Tenant Number (1-255)
LDN:	Enter the DID Number that is to terminate the call to the Attendant Console in the Day Mode (six digit maximum)
TYPE (Night Transfer Type):	Select a value from 0-3, using the following:
TYPE 0:	No Night Transfer is selected
TYPE 1:	"To Station" is selected. Enter a valid Station Number (six digit maximum). Specific Attendant Numbers assigned using the ASAT command cannot be used.
TYPE 2:	"To Outside" is selected. Enter the Outside Party Number (eight digit maximum).
TYPE 3:	"To ADC" is selected. Enter an Abbreviated Digit Code that has been predefined in the ASPD command (four digit maximum).

FCCS Programming

The following command assignments are required to use this feature via FCCS link.

STEP 1: **ASYDL** - SYS1, Index 640, Bit 0 to Bit 7. Assign FPC of a node which transmits Day/Night changeover information.

SYS1, Index 704 to Index 735. Assign FPC of a node which receives the Day/Night changeover information transmitted from the node assigned by SYS1, Index 640, Bit 0 to Bit 7 above.

STEP 2: **ANPDN** - Assign the first digit for Listed Directly Number (LDN).

TN: Tenant Number (1-255)

1stDC: 1st Digit Code (0-9,*,#)

CI: Select Normal Service (N), Hooking Service (H), or Busy Service (B) from the combo box for Connection Indexes.

NND: Number of Necessary Digits (max. 6 digits)

BLF: Busy Lamp Field Display Activated on ATT (0 = Out of Service/1 = In Service)

STEP 3: **ASPAN** - Assign the number of digit of Access Code for LDN.

TN: Tenant Number (1-255)

ACC: Access Code (maximum of 6 digits)

CI: Connection Status Index ("N: Normal"/"H: Hooking"/"B: Busy")

SRV: Kind of Service (Designate “TELN: Telephone Number”)

NND: Necessary Digits (1-16)

STEP 4: **ARTDN** - CDN 25 R/L (Attendant-Identification of Incoming Trunk Display) 0/1. Data “0”: TN, RT, TK display. Data “1”: Central Office Line Number Display (using ACOC).

STEP 5: **ALDNN** - Assign FPC of a node with Attendant Console which receives the LDN call by dialing the LDN.

TN: Tenant Number (1-255)

LDN: Listed Directory Number (maximum of 16 digits)

FPC: Fusion Point Code of the Node with Attendant Console which receives LDN calls

TYPE: Number Type of Night Mode destination number of LDN. (Designate “0” = No night transfer)

L-28 LDN Night Connection

General Description

This feature routes LISTED DIRECTORY NUMBER (LDN) calls to a preselected station when the system is in Night Mode.
In addition, this feature enables Attendant Console to forward terminated LDN calls to preprogrammed station via FCCS link in the FCCS network, also.

Operating Procedure

To answer LDN NIGHT CONNECTION calls (DID, TIE)

a.) An incoming LDN call in Night Mode is automatically routed to a preselected LDN NIGHT CONNECTION station within the system.
b.) The calling party receives Ringback Tone.
c.) The ringing signal is sent to the LDN NIGHT CONNECTION station.
d.) The LDN NIGHT CONNECTION station goes off-hook to answer the incoming call. If necessary, this station can transfer the incoming call to another station within the system using the CALL TRANSFER - ALL CALLS [C-11] feature.

Service Conditions

1. NIGHT CONNECTION - FIXED [N-1] and NIGHT CONNECTION - FLEXIBLE [N-2] take priority over this feature.
2. The destination station can be assigned on a LDN basis.
3. This feature is available for a maximum of 16 LDNs per tenant.
4. A data terminal cannot be used as a destination station for this feature.
5. The following features are not available for the destination station:

CALL FORWARDING - ALL CALLS - OUTSIDE	[C-28]
CALL FORWARDING - BUSY LINE - OUTSIDE	[C-60]
CALL FORWARDING - DON'T ANSWER - OUTSIDE	[C-74]

6. A UCD station can be assigned as the destination of LDN Night Connection or a transfer destination station of the CALL FORWARDING - ALL CALLS [C-5]/BUSY LINE [C-2], but queuing functions of UCD such as DELAY ANNOUNCEMENT - UCD [D-31] and CALL WAITING LAMP - UCD [D-76] are not available when all stations within the UCD Group are busy or CALL FORWARDING-ALL CALLS [C-2]/BUS LINE [C-11] to UCD Group station is set to the LDN Night Connection station.
7. When the destination station is busy, an incoming call is handled as follows: The call waits in queue. The status of the destination station is periodically checked. NIGHT CONNECTION OUTSIDE - SYSTEM [N-18] or TRUNK ANSWER FROM ANY STATION [T-8] is available when CALL FORWARDING - BUSY LINE [C-2] to the Attendant or CALL FORWARDING - BUSY LINE - OUTSIDE [C-60] has been assigned.
8. MISCELLANEOUS TRUNK RESTRICTION [M-3] is not available for this service.
9. NIGHT CONNECTION - FIXED [N-1] or TRUNK ANSWER FROM ANY STATION [T-8] is available while the destination station is in the following states:
   a.) LOCK OUT [L-3] make busy.
   b.) CALL FORWARDING - ALL CALLS [C-5] set to the Attendant.
   c.) CALL FORWARDING - ALL CALLS - OUTSIDE [C-28] set.

10. The LDN NIGHT CONNECTION station and the Attendant Console must be accommodated in the same node for the FCCS network.

<FCCS Conditions>

1. FPC of a node which transmits Day/Night changeover information is determined by SYS1, Index 640, and the information is transmitted to a node of FPC designated by SYS1, Index 704.
2. A station within the FCCS Network Console signed as forwarding destination for Night Mode (LDN NIGHT CONNECTION [L-28]).

Interactions

1. A maximum of 16 LDNs can be assigned per tenant, each with its own separate destination station. The same destination may also be used for all LDNs.
2. STATION HUNTING-CIRCULAR [S-7] and STATION HUNTING-TERMINAL [S-9] are valid from the Night Transfer destination station.
3. MULTIPLE CALL FORWARDING-ALL, BUSY, DON'T ANSWER [M-44, M-24, M-25] and MULTIPLE CALL FORWARDING-ALL, BUSY, DON'T ANSWER-CCIS [M-40, M-41, M-42] are valid from Night Transfer destination station.
4. An incoming DID call will get Busy Tone if the Night Transfer destination station is CALL FORWARDING-ALL [C-5] or CALL FORWARDING-BUSY LINE [C-2] to the Attendant, and TRUNK ANSWER FROM ANY STATION [T-8] is not service.
5. An incoming DID call will ring at the Night Transfer destination station if CALL FORWARDING-DON'T ANSWER [C-3] is set to the Attendant.
6. There is no restriction check for LDN NIGHT CONNECTION destination stations. The station can be totally restricted from receiving incoming calls and still receive LDN NIGHT CONNECTION calls.
7. If the Night Transfer destination station is busy, LDN calls will go into queue.
8. If the system is switched to Day Mode while LDN calls are in queue, only those calls that are in queue will be redirected to the Attendant. LDN calls that are ringing the Night Transfer destination station will continue to ring that station and will not be redirected to the Attendant.
9. If the Night Transfer destination station is in the Make Busy state (MBST Command = 1 for that station) or is in LINE LOCKOUT [L-3], the LDN call will go to TRUNK ANSWER FROM ANY STATION [T-8].
10. If the Night Transfer destination station (Station A) is in CALL FORWARDING-BUSY LINE [C-2] to Station B and both stations (Station A and Station B) are busy, the incoming LDN call will queue to both Station A and Station B, and will ring the first idle station.
11. If the Night Transfer destination (Station A) is in CALL FORWARDING-BUSY LINE [C-2] over CCIS to Station C in the remote office, and both Station A and Station C are busy, Busy Tone is returned to the incoming LDN calling party.
12. Incoming LDN calls to a Night Transfer destination station that has set CALL FORWARDING-ALL CALLS, BUSY LINE, DON'T ANSWER-CCIS [C-45, C-46, C-47] over CCIS to a busy UCD Group in the remote office will queue to that group, and will activate the CALL WAITING LAMP-UCD [C-76], if programmed.
13. The order of priority of forwarding features used when the system is Night Mode are as follows.

Higher Priority



Lower Priority

- NIGHT CONNECTION FLEXIBLE [N-2]
- NIGHT CONNECTION FIXED [N-1]
- LDN NIGHT CONNECTION [L-28]
- NIGHT CONNECTION-OUTSIDE-SYSTEM [N-17]
- TRUNK ANSWER FROM ANY STATION (TAS) [T-8]

Note: When forwarding destination for this feature is not assigned, Busy Tone is sent.

14. When assigning a forwarding destination number for Night Mode using ALDNN command, the maximum 16 digits can be programmed by using TYPE = 1 (Station Number) / 2 (DC).
    If the user requires digit number greater than 16 or more, assign an Abbreviated Digit Number (ADC) of the forwarding destination, preprogrammed by ASPD command, by using TYPE = 3 (Abbreviated Digit Number).
15. The user can assign different forwarding destination numbers (Telephone Number) to each LDN.

Note: The numbers of Telephone Number which can be programmed in a system is 65535 maximum.

16. Whether Special Transmission Tone is sent or not while number data is transmitted depends on SYS1, Index 6, Bit 0.
17. Forwarding destination terminal, available for this feature, is station or Voice Prompt System (VPS).
18. When using VPS as forwarding destination of this feature, the user programs voice information guidance number, recording service number or etc. In regard Numbering Plan for these service, refer to service condition of VPS.
19. Forwarding features except for the following are available for forwarding destination.
    - CALL FORWARDING-ALL CALLS [C-28] to Attendant Console
    - CALL FORWARDING-BUSY LINE [C-2] to Attendant Console
    - CALL FORWARDING-DON'T ANSWER [C-3] to Attendant Console
    - CALL FORWARDING-ALL CALLS-OUTSIDE [C-28]
    - CALL FORWARDING-BUSY LINE-OUTSIDE [C-60]
    - CALL FORWARDING-DON'T ANSWER-OUTSIDE [C-74]
    - CALL TRANSFER-ATTENDANT [C-10]
20. A route between Incoming Trunk received a LDN call and forwarding station for the LDN call is not restricted.
21. When STATION HUNTING-CIRCULAR [S-7], STATION HUNTING-TERMINAL [S-9] or UNIFORM CALL DISTRIBUTION (UCD) [U-1] is set to the forwarding destination, the hunting feature is available.
22. NIGHT CONNECTION FIXED [N-1] or TRUNK ANSWER FROM ANY STATION [T-8] is available while the destination is in the following states:
    - Forwarding destination is not assigned by ALDNN command.
    - CALL FORWARDING-BUSY LINE [C-2] to Attendant Console is set.
    - CALL FORWARDING-ALL CALLS [C-5] to Attendant Console is set.
    - CALL FORWARDING-INTERCEPT/ANNOUNCEMENT [C-25] is set.
    - CALL FORWARDING-BUSY LINE-OUTSIDE [C-28] is set.
    - CALL FORWARDING-ALL CALLS-OUTSIDE [C-60] is set.
    - FCCS link is all busy/make busy/rocked out.
    - Trunk/Route is all busy.
    - Toll is restricted.
    - Line is locked out.

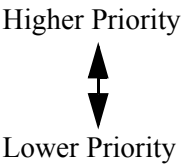
Note: NIGHT CONNECTION FIXED [N-1] takes priority over TRUNK ANSWER FROM ANY STATION [T-8].

23. When forwarding destination (including CCIS line) is not set, busy or restricted, Busy Tone or Reorder Tone is sent.
24. LDN NIGHT CONNECTION [N-1] is available for connections shown in table below:

× : Possible – : Impossible		Outgoing		
		CCIS Trunk	DOD Trunk	MFC Trunk
Incoming	CCIS Trunk	×	×	×
	DID Trunk	×	×	×
	MFC Trunk	×	×	–

25. When forwarding destination is busy, the call is queued, and Ringback Tone is sent, however, if the call is terminated from MFC line, Reorder Tone is sent.
26. When a LDN call is placed in a queue because forwarding destination station is busy, system checks station state (idle/busy) at intervals of 12 seconds. After that if the destination station becomes idle state, the LDN call is connected.
27. When plural calls are queued in a forwarding destination station, the queued call may not connect the destination stations in termination order of call.
28. When system is changed from Night Mode to Day Mode, queued calls terminate to Attendant Console to make up for forwarding to destination.
29. When system is changed from Day Mode to Night Mode while a LDN call is terminating to Attendant Console, the LDN call is transferred to preprogrammed forwarding destination. If the LDN call cannot be transferred according to normal priority by restriction for the forwarding destination, one of the features below (**Note 1**) is used to process the LDN call. (The features in the list are in priority order with the highest priority at the top.)

- TRUNK ANSWER FROM ANY STATION [T-8]
- DELAY ANNOUNCEMENT-ATTENDANT [D-144]
- NIGHT CONNECTION-FIXED [N-1]



Note 1: When these features are not set, the LDN call is queued.

Programming

STEP 1: **ALDN** - Assign LDN number and its transfer destination.

TN: Tenant Number

LDN: Listed Directory Number (must match Station Numbering Plan)

Night Transfer:Y/N

If no, then LDN will only go to Attendant.

If yes, then the following parameters will be enabled:

TYPE: Night Transfer Type

1: Station Number

2: Outside (up to 8 digits)

3: ADC (when more than 8 digits are required)

STN: For Types 1 and 2.

For Type 1, assign Station Number here.

For Type 2, assign Access Code and Outside Number here.

(A CCIS remote office Station Number can be input here also).

ADC: For Type 3.

Input ADC number desired as assigned in the ASPD command.

STEP 2: **ACFR** - Assign TSFI: 2 (Night Connection-Fixed) to be allowed CCI: 1 (LDN). This allows LDN to go to night transfer destination. Assign TSFI: 5 (TAS) to be allowed CCI: 1 (LDN). This allows LDN to go to TRUNK ANSWER FROM ANY STATION [T-8]. Restrict if not using TAS.

FCCS Programming

The following command assignments are required to use this feature via FCCS link.

● When destination number is programmed by using TYPE = 1 (TELN) of ALDNN command.

STEP 1: **ASYDL** - SYS1, Index 640, Bit 0 to Bit 7. Assign FPC of a node which transmits Day/Night changeover information.

SYS1, Index 704 to Index 735. Assign FPC of a node which receives the Day/Night changeover information transmitted from the node assigned by SYS1, Index 640, Bit 0 to Bit 7 above.

STEP 2: **ANPDN** - Assign the first digit for Listed Directly Number (LDN) and forwarding destination.

TN: Tenant Number (1-255)

1stDC: 1st Digit Code (0-9,*,#)

CI: Select Normal Service (N), Hooking Service (H), or Busy Service (B) from the combo box for Connection Indexes.

NND: Number of Necessary Digits (max. 6 digits)

BLF: Busy Lamp Field Display Activated on ATT (0 = Out of Service/1 = In Service)

STEP 3: **ASPAN** - Assign the number of digit of Access Code for LDN and forwarding destination.

TN: Tenant Number (1-255)

ACC: Access Code (max. 6 digits)

CI: Connection Status Index (“N: Normal”/”H: Hooking”/”B: Busy”)

SRV: Kind of Service (Designate “TELN: Telephone Number”)

NND: Necessary Digits (1-16)

STEP 4: **ALGSN** - Assign Telephone Number to Night Mode destination number (Station Number).

a.) TYPE = 1 (When Line Equipment Numbers/Phantom Index is used to specify the Telephone

Number)
UGN: User Group Number (1-255)
TELN: Telephone Number (max. 16 digits)
FPC: Fusion Point Code of the designated LENS/PHIN
LENS/PHIN:Line Equipment Numbers of physical station

b.) TYPE = 2 (When Station Number is used to specify the Telephone Number)
UGN: User Group Number (1-255)
LDN: Listed Directory Number (max. 16 digits)
TN: Tenant Number (1-255)
STN: Station Number

STEP 5: **ALDNN** - Assign FPC of a node with Attendant Console which receives the LDN Call by dialing the LDN.
TN: Tenant Number (1-255)
LDN: Listed Directory Number (max. 16 digits)
FPC: Fusion Point Code of the node with Attendant Console which receives LDN Calls
TYPE: Number Type of Night Mode destination number of LDN. (Designate 1 = TELN.)
STN: Station Number of Forwarding Destination (max. 16 digits) or specific Attendant Console number assigned by ASAT command.

STEP 6: **ACFR** - Allow the Call Forwarding Restriction Class.
TN: Tenant Number (1-255)
TSFI: Transfer Service Feature Index (Assign 2 = Direct in Termination (Night only))
CCI: Call Category Index (Assign 1 = LDN (Central Office Line))
RES: Restriction Data (Assign 1 = Allowed)

● When destination number is programmed by using TYPE = 3 (ADC) of ALDNN command.

STEP 1: **ASYDL** - SYS1, Index 640, Bit 0 to Bit 7. Assign FPC of a node which transmits Day/Night changeover information.
SYS1, Index 704 to Index 735. Assign FPC of a node which receives the Day/Night changeover information transmitted from the node assigned by SYS1, Index 640, Bit 0 to Bit 7 above.

STEP 2: **ANPDN** - Assign the first digit for Listed Directly Number (LDN) and forwarding destination.
TN: Tenant Number (1-255)
1stDC: 1st Digit Code (0-9,*,#)
CI: Select Normal Service (N), Hooking Service (H), or Busy Service (B) from the combo box for Connection Indexes.
NND: Number of Necessary Digits (max. 6 digits)
BLF: Busy Lamp Field (0 = Out of Service/1 = In Service)

STEP 3: **ASPAN** - Assign the number of digit of Access Code for LDN and forwarding destination.
TN: Tenant Number (1-255)
ACC: Access Code (max. 6 digits)
CI: Connection Status Index (“N: Normal”/”H: Hooking”/”B: Busy”)
SRV: Kind of Service (Designate “TELN: Telephone Number”)
NND: Necessary Digits (1-16)

STEP 4: **ALGSN** - Assign Telephone Number to Night Mode destination number (Station Number).

a.) TYPE = 1 (When Line Equipment Numbers/Phantom Index is used to specify the Telephone Number.)
UGN: User Group Number (1-255)

TELN: Telephone Number (max. 16 digits)
FPC: Fusion Point Code of the designated LENS
LENS/PHIN: Line Equipment Numbers/Phantom Index of physical station

b.) TYPE = 2 (When Station Number is used to specify the Telephone Number.)
UGN: User Group Number (1-255)
LDN: Listed Directory Number (max. 16 digits)
TN: Tenant Number (1-255)
STN: Station Number

STEP 5: **ASPD** - Assign speed Calling Number to Abbreviated Digit Code (ADC).
TN: Tenant Number (1-255)
ADC: Abbreviated Digit Code (maximum of four digits)
CD: Desired Speed Calling Number (maximum of 24 digits)

Note: The user must be assigned this command at node which receives the LDN Call.

STEP 6: **ALDNN** - Assign FPC of a node with Attendant Console which receives the LDN Call by dialing the LDN.
TN: Tenant Number (1-255)
LDN: Listed Directory Number (max. 16 digits)
FPC: Fusion Point Code of the node with Attendant Console which receives LDN Calls
TYPE: Number Type of Night Mode destination number of LDN. Assign 3 (ADC).
ADC: Abbreviated Digit Code assigned by ASPF command.

STEP 7: **ACFR** - Allow the Call Forwarding Restriction Class.
TN: Tenant Number (1-255)
TSFI: Transfer Service Feature Index (Assign 2 = Direct in Termination (Night only))
CCI: Call Category Index (Assign 1 = LDN (Central Office Line))
RES: Restriction Data (Assign 1 = Allowed)

L-30 LDN Night Connection - Outside

General Description

This feature routes LISTED DIRECTORY NUMBER (LDN) calls to a preselected station outside the system when the system is in Night Mode.
In addition, this feature enables Attendant Console to transmit terminated LDN call to outside using outgoing trunk in the FCCS network.

Operating Procedure

To answer LDN NIGHT CONNECTION calls (DID, TIE)

1. An incoming LDN call in Night Mode is automatically routed to a preselected LDN NIGHT CONNECTION station outside the system.
2. The calling party receives Ringback Tone.
3. The ringing signal is sent out to the LDN NIGHT CONNECTION-OUTSIDE station.
4. The LDN NIGHT CONNECTION station answers the incoming call. If necessary, this station can transfer the incoming call to another station outside the system using the CALL TRANSFER - ALL CALLS [C-11] feature.

Service Conditions

1. NIGHT CONNECTION - FLEXIBLE [N-2] and NIGHT CONNECTION - FIXED [N-1] take priority over this feature.
2. The maximum number of digits in the destination number is 8. SPEED CALLING - SYSTEM [S-3] can be used when more than 8 digits are needed.
3. The destination station outside the system can be assigned on a LDN basis.
4. This feature is available for a maximum of 16 LDNs per tenant.
5. LDN NIGHT CONNECTION - OUTSIDE [L-30] will not route an incoming call on an MFC signaling trunk through another MFC signaling trunk.
6. NIGHT CONNECTION - FIXED [N-1] or TRUNK ANSWER FROM ANY STATION [T-8] is available while the destination station is in the following states:

MISCELLANEOUS TRUNK RESTRICTION	[M-3]
TOLL RESTRICTION - 3/6-DIGIT	[T-7]
Outgoing Trunk Busy	

FCCS Conditions

1. FPC of a node which transmits Day/Night changeover information is determined by SYS1, Index 640, and the information is transmitted to a node of FPC designated by SYS1, Index 704.

   Note: The numbers of Telephone Number which can be programmed in a system is 65535 maximum.

2. Outgoing trunk within the FCCS network can be assigned as forwarding destination for Night Mode (LDN NIGHT CONNECTION-OUTSIDE [L-30]).
3. The order of priority of forwarding features used when the system is Night Mode are as follows.

   Higher Priority
   ↕
   Lower Priority

   - NIGHT CONNECTION FLEXIBLE [N-2]
   - NIGHT CONNECTION FIXED [N-1]
   - LDN NIGHT CONNECTION [L-28]
   - NIGHT CONNECTION-OUTSIDE-SYSTEM [N-17]
   - TRUNK ANSWER FROM ANY STATION (TAS) [T-8]

   Note: When forwarding destination for this feature is not assigned, Busy Tone is sent.

4. When assigning a forwarding destination number for Night Mode using ALDNN command, the maximum 16 digits can be programmed by using TYPE = 1 (Station Number) / 2 (DC).
   If the user requires digit number greater than 16 or more, assign an Abbreviated Digit Number (ADC) of the forwarding destination, preprogrammed by ASPD command, by using TYPE = 3 (Abbreviated Digit Number).
5. The user can assign different forwarding destination numbers (Telephone Number) to each LDN.

   Note: The number of Telephone Numbers which can be programmed in a system is 65535 maximum.

6. The user must restrict tandem connection that release-guard signal cannot be detected by tandem-office.
7. Whether Special Transmission Tone is sent or not while number data is transmitted depends on SYS1, Index 6, Bit 0.
8. LDN call terminated from MFC Line cannot be forwarded to MFC line.
9. Forwarding features except for the following are available for forwarding destination.
   - CALL FORWARDING-ALL CALLS [C-28] to Attendant Console
   - CALL FORWARDING-BUSY LINE [C-2] to Attendant Console
   - CALL FORWARDING-DON'T ANSWER [C-3] to Attendant Console
   - CALL FORWARDING-ALL CALLS-OUTSIDE [C-28]
   - CALL FORWARDING-BUSY LINE-OUTSIDE [C-60]
   - CALL FORWARDING-DON'T ANSWER-OUTSIDE [C-74]
   - CALL TRANSFER-ATTENDANT [C-10]
10. NIGHT CONNECTION FIXED [N-1] or TRUNK ANSWER FROM ANY STATION [T-8] is available while the destination is in the following states:
    - Forwarding destination is not assigned by ALDNN command.
    - CALL FORWARDING-BUSY LINE [C-2] to Attendant Console is set.
    - CALL FORWARDING-ALL CALLS [C-5] to Attendant Console is set.
    - CALL FORWARDING-INTERCEPT/ANNOUNCEMENT [C-25] is set.
    - CALL FORWARDING-ALL CALLS-OUTSIDE [C-28] is set.

- CALL FORWARDING-BUSY LINE-OUTSIDE [C-60] is set.
- LDN call originated from MFC line is forwarded to MFC line.
- FCCS link is all busy/make-busy/rocked out.
- Trunk/Route is all busy.
- Toll is restricted.
- Line is locked out.

Note: NIGHT CONNECTION FIXED [N-1] takes priority over TRUNK ANSWER FROM ANY STATION [T-8].

11. When forwarding destination (including CCIS line) is not set, busy or restricted, Busy Tone or Reorder Tone is sent.
12. LDN NIGHT CONNECTION [N-1] is available for connections shown in table below:

× : Possible – : Impossible		Outgoing		
		CCIS Trunk	DOD Trunk	MFC Trunk
Incoming	CCIS Trunk	×	×	×
	DID Trunk	×	×	×
	MFC Trunk	×	×	–

13. When system is changed from Day Mode to Night Mode while a LDN call is terminating to Attendant Console, the LDN call is transferred to preprogrammed forwarding destination. If the LDN call cannot be transferred according to normal priority by restriction for the forwarding destination, one of the features below (**Note 1**) is used to process the LDN call. (The features in the list are in priority order with the highest priority at the top.)

- TRUNK ANSWER FROM ANY STATION [T-8] is provided.
- DELAY ANNOUNCEMENT-ATTENDANT [D-144] is provided.
- NIGHT CONNECTION-FIXED [N-1] is provided.

Higher Priority



Lower Priority

Note 1: When above feature is not set, the LDN call is queued.

Interactions

1. A maximum of 16 LDNs can be assigned per tenant; each with its own separate destination number.
2. If the outgoing route is busy or if the incoming LDN route is restricted from the outgoing destination route, the LDN calling party will receive Busy Tone or will go to TAS, if programmed.
3. If using TYPE: 3 ADC parameter in the ALDN command, the ADC must first be assigned in the ASPD command. TYPE: 3 must be used for Outside Numbers greater than eight digits.

Programming

STEP 1: **ALDN** - Assign LDN number and its transfer destination.

TN: Tenant Number

LDN: Listed Directory Number (must match Station Numbering Plan.)

Night Transfer: Y/N

If no, then LDN will only go to Attendant.

If yes, then the following parameters will be enabled:

TYPE: Night Transfer Type

1: Station Number

2: Outside (up to 8 digits)

3: ADC (when more than 8 digits are required)

STN: For Types 1 and 2.

For Type 1, assign Station Number here.

For Type 2, assign Access Code and Outside Number here.

(A CCIS remote office Station Number can be input here also.)

ADC: For Type 3.

Input ADC number desired as assigned in ASPD command.

STEP 2: **ASPD** - Assign an Access Code and Outside Number to an ADC (000-999). This step is required only if Access Code and Outside Number are greater than eight digits.

STEP 3: **ACFR** - Assign TSFI: 2 (Night Connection-Fixed) to be allowed CCI: 1 (LDN). This allows LDN to go to Night transfer destination. Assign TSFI: 5 (TAS) to be allowed CCI: 1 (LDN). This allows LDN to go to TRUNK ANSWER ANY STATION (TAS). Restrict if not using TAS.

STEP 4: **ARRC** - The Incoming Route of the LDN must allow Trunk-to-Trunk Connection to the outgoing destination route and to the LCR flag route; ARI-D-RES, Direct Connection.

FCCS Programming

The following command assignments are required to use this feature via FCCS link.

● When destination number is programmed by using TYPE = 2 (DC) of ALDNN command.

STEP 1: **ASYDL** - SYS1, Index 640, Bit 0 to Bit 7. Assign FPC of a Node which transmits Day/Night changeover information.

SYS1, Index 704 to Index 735. Assign FPC of a Node which receives the Day/Night changeover information transmitted from the Node assigned by SYS1, Index 640, Bit 0 to Bit 7 above.

STEP 2: **ANPDN** - Assign the first digit for Listed Directly Number (LDN) and forwarding destination.

TN: Tenant Number (1-255)

1stDC: 1st Digit Code (0-9,*,#)

CI: Select Normal Service (N), Hooking Service (H), or Busy Service (B) from the combo box for Connection Indexes.

NND: Number of Necessary Digits (maximum of 6 digits)

BLF: Busy Lamp Field Display Activated on ATT (0 = Out of Service/1 = In Service)

STEP 3: **ASPAN** - Assign the number of digit of Access Code for LDN and forwarding destination.

TN: Tenant Number (1-255)

ACC: Access Code (maximum of 6 digits)

CI: Connection Status Index (“N: Normal”/”H: Hooking”/”B: Busy”)

SRV: Kind of Service (Designate “TELN: Telephone Number” for LDN, “OGC: Outgoing Call”/ ”OGCA: Outgoing Call with route advance”/”LCRS: Register Sender LCR” for forwarding destination.)

NND: Necessary Digits (1-16)

STEP 4: **ALRTN** - Assign Logical Route Number to Physical Route Number.

LGRT: Logical Route Number (1-89)

FPC: Fusion Point Code (1-253)
RT: External Route Number (1-255)

STEP 5: **ALDNN** - Assign FPC of a Node with Attendant Console which receives the LDN call by dialing the LDN.
TN: Tenant Number (1-255)
LDN: Listed Directory Number (maximum of 16 digits)
FPC: Fusion Point Code of the node with Attendant Console which receives LDN calls
TYPE: Number Type of Night Mode destination number of LDN
(Designate DC (destination number including Access Code))
DC: Assign the Outside Destination Number including Access Code.

STEP 6: **ACFR** - Allow the Call Forwarding Restriction Class.
TN: Tenant Number (1-255)
TSFI: Transfer Service Feature Index (Assign 2 = Direct in Termination (Night only))
CCI: Call Category Index (Assign 1 = LDN (Central Office Line))
RES: Restriction Data (Assign 1 = Allowed)

● When destination number is programmed by using TYPE = 3 (ADC) of ALDNN command.

STEP 1: **ASYDL** - SYS1, Index 640, Bit 0 to Bit 7. Assign FPC of a Node which transmits Day/Night changeover information.
SYS1, Index 704 to Index 735. Assign FPC of a Node which receive the Day/Night changeover information transmitted from the Node assigned by SYS1, Index 640, Bit 0 to Bit 7 above.

STEP 2: **ANPDN** - Assign the first digit for Listed Directly Number (LDN) and forwarding destination.
TN: Tenant Number (1-255)
1stDC: 1st Digit Code (0-9,*,#)
CI: Select Normal Service (N), Hooking Service (H), or Busy Service (B) from the combo box for Connection Indexes.
NND: Number of Necessary Digits (max. 6 digits)
BLF: Busy Lamp Field Display Activated on ATT (0 = Out of Service/1 = In Service)

STEP 3: **ASPAN** - Assign the number of digit of Access Code for LDN and forwarding destination.
TN: Tenant Number (1-255)
ACC: Access Code (maximum of 6 digits)
CI: Connection Status Index (“N: Normal”/”H: Hooking”/”B: Busy”)
SRV: Kind of Service (Designate “TELN: Telephone Number” for LDN, “OGC: Outgoing Call”/ ”OGCA: Outgoing Call with route advance”/”LCRS: Register Sender LCR” for forwarding destination.)
NND: Necessary Digits (1-16)

STEP 4: **ALRTN** - Assign Logical Route Number to Physical Route Number.
LGRT: Logical Route Number (1-89)
FPC: Fusion Point Code (1-253)
RT: External Route Number (1-255)

STEP 5: **ASPD** - Assign Speed Calling Number to Abbreviated Digit Code (ADC).
TN: Tenant Number (1-255)
ADC: Abbreviated Digit Code (maximum of four digits)
CD: Desired Speed Calling Number (maximum of 24 digits)

Note: The user must be assigned this command at Node which receives the LDN call.

STEP 6: **ALDNN** - Assign FPC of a node with Attendant Console which receives the LDN call by dialing the LDN.

TN: Tenant Number (1-255)
LDN: Listed Directory Number (max. 16 digits)
FPC: Fusion Point Code of the node with Attendant Console which receives LDN calls
TYPE: Number Type of Night Mode destination number of LDN. Assign 3 (ADC).
ADC: Abbreviated Digit Code assigned by ASPF command

STEP 7: **ACFR** - Allow the Call Forwarding Restriction Class.

TN: Tenant Number (1-255)
TSFI: Transfer Service Feature Index (Assign 2 = Direct in Termination (Night only))
CCI: Call Category Index (Assign 1 = LDN (Central Office Line))
RES: Restriction Data (Assign 1 = Allowed)

L-52 Last Number Called - Attendant

General Description

This feature allows an operator to store the LAST NUMBER DIALED, and to place a station call or a trunk call from Attendant Console by dialing an access code of LAST NUMBER CALLED.

Note: This feature is not available in North America and Australia.

Operating Procedure

1. Press the LOOP key on the Attendant Console.
2. Dial the desired number and receive Ring Back Tone (RBT) or Busy Tone (BT)/Reorder Tone (ROT).
3. Press the CANCEL key to cancel the call.
4. Press the LOOP key and dial the access code for LAST NUMBER CALLED.

Service Conditions

1. This feature is available when the Attendant places a call via SPEED CALLING-SYSTEM [S-3]/ STATION [S-21]/LEAST COST ROUTING-3/6 DIGIT [L-5].
2. This feature is not available in following cases:
   a.) when the Attendant picks up a station/trunk call that was placed on hold.
   b.) when the Attendant places a call via Camp-On service.
   c.) when the called party has set CALL FORWARDING-ALL CALLS [C-5].
3. A maximum of 18 digits can be stored by this feature.
4. By this feature, the Attendant is connected to the party whose number was dialed last, regardless of the used Loop key. The dialed numbers cannot be stored on each Loop key.

Programming

STEP 1: **ASPA** - For SRV = "SSC: Service Code" SID = 44, assign the access code for LAST NUMBER CALLED.

STEP 2: **ASFC**- For the Attendant allowed LAST NUMBER CALLED, assign a Class of Service (SFC 0) that allows SFI 44.

L-53 Logged Out IP Station - Call Destination

General Description

This feature permits a call to an IP Station in logout state to be immediately forwarded to a predetermined Station, Attendant Console or Announcement Equipment. The forwarding destination can be set at each terminal in advance. The forwarding destination for this feature can also be assigned from the PCPro, on a tenant basis or on a route basis (**Note 1**).

Note 1: Data assignment on a route basis is available since FP95-114 V4 Issue 2.0.

To set LOGGED OUT IP STATION-CALL DESTINATION by Access Code

1. Lift the handset or press the SPEAKER key. Receive Dial Tone.
2. Dial the specific LOGGED OUT IP STATION-CALL DESTINATION Access Code; receive Special Dial Tone.
3. Dial the desired target number; receive Service Set Tone.
4. Replace the handset or press the SPEAKER key.

To cancel LOGGED OUT IP STATION-CALL DESTINATION by Access Code

1. Lift the handset or press the SPEAKER key. Receive Dial Tone.
2. Dial the specific LOGGED OUT IP STATION-CALL DESTINATION Cancel Code; receive Service Set Tone.
3. Replace the handset or press the SPEAKER key.

To set LOGGED OUT IP STATION-CALL DESTINATION by Feature (Soft) key

1. Lift the handset or press the SPEAKER key. Receive Dial Tone.
2. Press the specific LOGOUT IP STATION-CALL DESTINATION Feature key (Soft key); receive Special Dial Tone.
3. Dial the desired target Station Number; receive Service Set Tone. The LOGGED OUT IP STATION-CALL DESTINATION Feature key lights red. (Soft key; FDL is displayed on the LCD).
4. Replace the handset or press the SPEAKER key.

To verify LOGGED OUT IP STATION-CALL DESTINATION by Feature (Soft) key

1. Press LOGGED OUT IP STATION-CALL DESTINATION Feature key lighting red or press the Soft key.
2. Forwarding destination will display.

To cancel LOGGED OUT IP STATION-CALL DESTINATION by Feature (Soft) key

1.Lift the handset or press the SPEAKER key. Receive Dial Tone.
2.Press the specific LOGGED OUT IP STATION-CALL DESTINATION feature (Soft) key; receive Service Set Tone.
3.LOGGED OUT IP STATION-CALL DESTINATION lamp goes off (Soft key; FDL indication disappears on the LCD).
4. Replace the handset or press the SPEAKER key.

Service Conditions

1. "Logout Status" refers to the following status:
   1. Login prompt is displayed on the LCD.
   2. An error message is displayed on the LCD.
   3. Terminal is reset due to the LAN failure such as LAN cable disconnection.
   4. Terminal power fails.

2. This feature is available for the following terminals:
   - DtermIP
   - IP Enabled Dterm
   - Soft Phone
   - DtermIP INASET
   - Terminal connected to Analog MC/MC&MG-COT (only for the LAN failure)
3. There is no limitation to the number of Stations that could set LOGGED OUT IP STATION-CALL DESTINATION simultaneously.
4. This feature data can be assigned on a terminal basis, according to Service Feature Restriction Class.
5. This feature can be set from Sub Line. At this time, the Sub Line key on the operating terminal does not light (My Line key on the terminal associated with the Sub Line lights.)

   Note: The Sub Line cannot be a Virtual (TEC-18) station number.

6. Announcement Access Code can be specified as the forwarding destination for this feature. At this time, the call can be routed to the Announcement Trunk.
7. When a call is directed to IP Station in Logout status, there are the following conditions:
   a.) When the following features are set at a terminal in addition to this feature, the priority order is shown below.
   Number Sharing (Sub-station is set as Forwarding Destination) > Logged Out IP Station > Call Destination > Call forwarding > All Calls > Uniform Call Distribution > Call Forwarding > Busy Line > Station Hunting > Do Not Disturb > Call Forwarding > Don't Answer
   b.) When both CALL FORWARDING - ALL CALLS [C-5] and LOGGED OUT IP STATION - CALL DESTINATION [L-53] are set at a terminal, the priority order of these two features can be determined according to Service Feature Restriction Class (SFC) of the Station.
   SFI 209 = 0: Logged Out IP Station-Call Destination > Call Forwarding - All Calls
   = 1: Call Forwarding - All Calls > Logged Out IP Station-Call Destination
   When this feature is not set:
   If CALL FORWARDING-ALL CALLS [C-5] is set, CALL FORWARDING-ALL CALLS [C-5] is activated.
   If CALL FORWARDING-ALL CALLS [C-5] is not set, the calling party hears RBT. (If the called line is assigned as a Multi Line of other Station, the call can be answered by using the appropriate Multi Line.
8. The forwarding destination for this feature can be assigned with the ACFO/ACFOL/ACFON command, on a tenant basis. At this time, the following conditions must be considered.
   a.) When this feature is set from both the terminal and PCPro, the setting from the terminal has precedence over the setting from the PCPro.
   b.) ACFO/ACFOL/ACFON command setting is effective only when FORWARDING-ALL CALLS [C-5] and this feature are not set at the terminal.
   c.) The feature data via DM, LDM and NDM can coexist in a System. (Priority: NDM > LDM > DM)

9. The forwarding destination for this feature can be assigned with ACFO/ACFOL/ACFON command on a route basis. For details, see CALL FORWARDING DESTINATION PER TRUNK ROUTE - INCOMING CALL [C-209].

   Note: Data assignment on a route basis is available since FP95-114 V4 Issue 2.0 (earlier than FP95-114 V4 Issue 2.0, the destination can be assigned with the ACFO/ACFOL/ACFON command on a tenant basis only).

10. If the IP terminal is placed to logout status due to the power failure or LAN cable disconnection while a call is terminated to the terminal, the calling party hears RBT. However, Call Forwarding-Don't Answer is set at the terminal, the call is transferred to another party.
11. While incoming to IP terminal, if power failure or LAN cable disconnection caused logout status, Calling Party continues to hear Ringback Tone (Note that CALL FORWARDING - DON'T ANSWER [C-3] has been activated, LOGGED OUT IP STATION-CALL DESTINATION service would be available).
12. LOGGED OUT IP STATION-CALL DESTINATION at routine diagnosis is not performed.
13. LOGGED OUT IP STATION-CALL DESTINATION data is backed up to the Flash Card at the routine diagnosis. For details, refer to the System Maintenance and Operation Guide.
14. When this feature is set from Sub Line, the terminal associated with the Sub Line cannot go into logout status while hearing Service Set Tone at the operating terminal.
15. When LOGGED OUT IP STATION-CALL DESTINATION is performed via CCIS, forwarding destination is accepted as CALL FORWARDING ALL CALLS [C-5]. For this reason, LCD displays CALL FORWARDING ALL CALLS message.
16. Assign the System Data (ASYDL, SYS1, Index 677, Bit 3=1), and press the LOGGED OUT IP STATION-CALL DESTINATION Feature key, a maximum of 16 digits Call Forwarding Information can be displayed.

    Note: This feature is only available for the terminal that can display 24 digits on LCD.

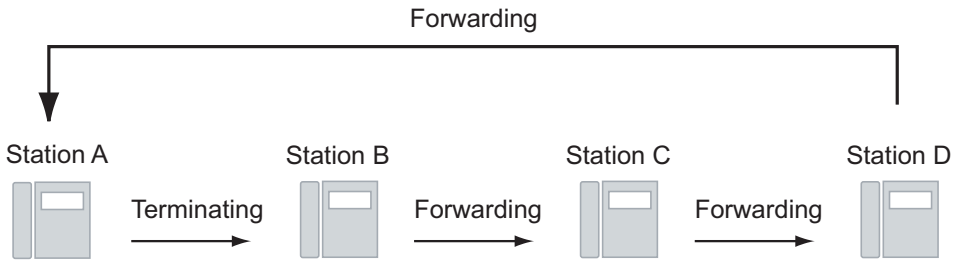
17. Multiple Call Forwarding - Busy Line/All Calls/Logout (Logged Out IP Station - Call Destination [L-53]) can be selected to use or not by ASYD command [SYS1, Index 69, Bit 7 (0/1 = Out of service/In service)].

    - When using Multiple Call Forwarding - Busy Line/All Calls/Logout.

      A maximum number of Multiple Call Forwarding - All Calls/Busy Line/Logout occurrences can be selected by ASYD command [SYS1, Index 5, Bit 1 to Bit 3].

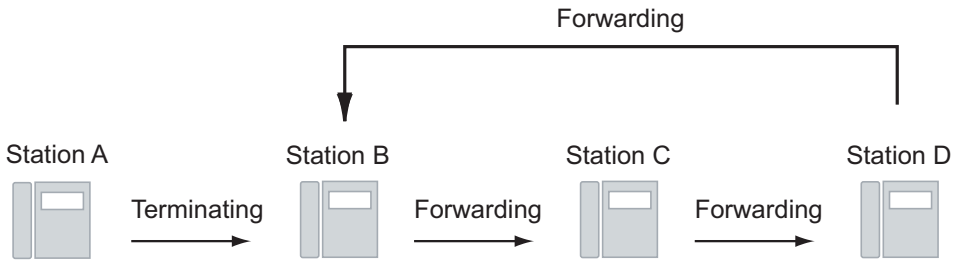
      The following shows the call forwarding operation that is looped within the assigned number of Multiple Call Forwarding - Busy Line/All Calls/Logout occurrences.

< If target station is Station A. >



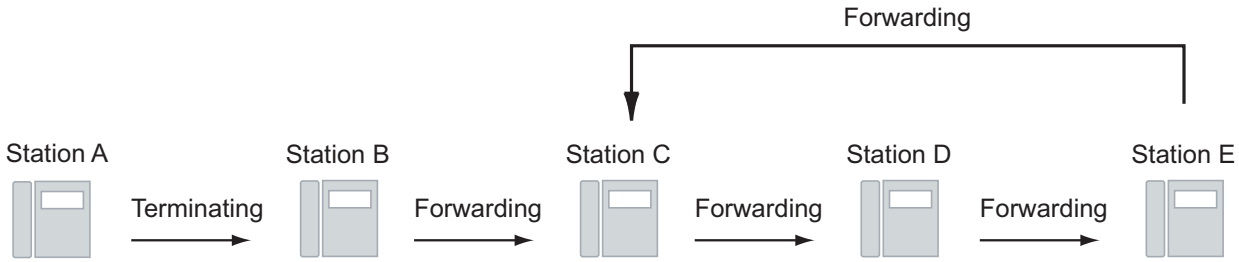
The call of Station A calls Station D. If Station D is busy, Station A hears a busy tone.

< If call forwarding is looped among Station B-D >



The call of Station A calls Station C. If Station C is busy, Station A hears a busy tone.

< If call forwarding is looped among Station C-E >



The call of Station A calls Station C. If Station C is busy, Station A hears a busy tone.

- When not using Multiple Call Forwarding - Busy Line/All Calls/Logout.

Depending on the call forwarding setting for destination terminal, the operation will change as follows.

Example: Call Forwarding Operation among Station A-D When Station A calls Station B

Station B sets	Station C sets	Operation
"Call Forwarding - All Calls [C-2]/Logged Out IP Station - Call Destination [L-53]" for Station C	"Call Forwarding - Busy Line [C-2]" for Station D	Station C is called.
"Call Forwarding - All Calls [C-2]/Logged Out IP Station - Call Destination [L-53]" for Station C	"Call Forwarding - All Calls [C-2]/Logged Out IP Station - Call Destination [L-53]" for Station D	Station D is called.
"Call Forwarding - Busy Line [C-2]" for Station C	"Call Forwarding - All Calls [C-2]/Logged Out IP Station - Call Destination [L-53]" for Station C	Station C is called.
"Call Forwarding - Busy Line [C-2]" for Station C	"Call Forwarding - Busy Line [C-2]" for Station D	Station A hears a busy tone.

18. Speed Calling number cannot be used for a destination number of a call forwarded by this feature.

Interactions

1. When a call is routed to IP Station via the following features, this feature can be activated.

Services (for Stations)	Services (for Attendant Console)
OFF - HOOK ALARM [O-6]	BLIND TRANSFER TO ATTENDANT [B-17]
BLIND TRANSFER TO STATION [B-18]	SERIAL CALL [S-15]
TIMED REMINDER [T-62]	Incoming Call via ATT
DIRECT IN TERMINATION [D-7]	STATION - TO - STATION CALLING - OPERATOR ASSISTANCE [S-12]
HOTLINE [H-1]	STEP CALL - ATT [S-13]
PRIORITY CALL [P-9]	ATTENDANT NIGHT TRANSFER [A-19]
STEP CALL [S-13]	
SERIAL CALL -D [S-127]	
DIRECT INWARD DIALING - REMOTE ACCESS TO SYSTEM / AUTOMATED ATTENDANT	
DSS CONSOLE [D-149]	

2. When UCD Member Station is placed in Logout status, the following conditions must be considered.
   a.) When this feature is individually set from Terminal:
   UCD Controlling Station: This feature can be activated.

Non-Controlling Station: When Calling Party dials Controlling Station Number, that Station is skipped in any UCD hunt.

b.) When this feature is not individually set from Terminal:
Both UCD Controlling Station and Non-Controlling Station are skipped in any UCD hunt.

3. When Station belonging to Station Hunting Group is placed in Logout status, the following conditions must be considered.

a.) When this feature is individually set from Terminal:
Pilot Station: This feature can be activated.
Non-Pilot Station: When Calling Party dials Pilot Station Number, that Station is bypassed. If the Station is the Lowest Priority Station, Calling Party hears Busy Tone.

b.) When this feature is not individually set from Terminal:
Both Pilot Station and Non-Pilot Station are bypassed.

4. SIP Handler Controlled SIP terminals are available. In the following situations, a SIP Handler Controlled SIP terminal becomes in logout state which is terminal state cannot be displayed in the IPAN or DISD command window. In the logout state, the SIP Handler Controlled SIP terminal cannot place or receive calls until it perform registration to the system.

- The Telephony Server does not receive REGISTER for updating Expire within the time of register expires. **Note 1**
- The Telephony Server receives Expire=0 as Expiration of REGISTER from the SIP Handler Controlled SIP terminal.
- Terminal is reset with the DISD command **Note 2**
- SIP Handler is initialized with the ADPM command **Note 2**
- System Initialization is executed with the SINZ command. **Note 2**

Note: System Message 33-S: IP PKG Failure is not output in either situation.

Note 1: 3600 seconds is set for the default. This data can be changed with the Register Expires parameter in the AUACL/AUACN command.

Note 2: To make the Standard SIP terminal in login state, the Standard SIP terminal needs to perform registration. The Standard SIP terminal performs registration at the timing of REGISTER update of the Standard SIP terminal so Standard SIP terminal cannot be used until REGISTER update after reset or initialization is executed by using the command.

Programming

STEP 1: **ASYD** - SYS1, Index 5, Bit 1 to Bit 3.
Maximum number of Multiple Call Forwarding - Busy Line/All Calls/Logout (Logged Out IP Station - Call Destination [L-53]).

Note: This data is valid only when SYS1, Index 69, Bit 7 is assigned as"1".

SYS1, Index 69, Bit 1; Send Short Tone when a Call Forward via CALL FORWARDING - ALL CALLS [C-5] service is answered. 0/1 = Not Required/Required
Bit 7; Multiple call service in MULTIPLE CALL FORWARDING - BUSY LINE [M-24] / ALL CALLS [M-44], LOGGED OUT IP STATION-CALL DESTINATION. 0/1 = Out/In Service
SYS1, Index 79, Bit 2; SPLIT CALL FORWARDING [S-99] service. 0/1 = Out/In Service

Note: Use the specific Access Code to assign and cancel LOGGED OUT IP STATION-CALL DESTINATION service (Refer to STEP 3:**ASPA**).

STEP 2: **ASYDL** - SYS1, Index 677, Bit 3.
Display a maximum of 16 digits Call Forwarding Information on LCD? 0/1 = No (default)/Yes.
SYS1, Index 959. Health Check cycle
00 Hex. = 60 seconds (default)
01-FE Hex. = 1-254 seconds
FF Hex. = Health Check stop
-between PHD and IP terminal: system data set value
-between MGC and PHD: system data set value×1.5

STEP 3: **ANPD** - Assign the minimum number of digits for each Access Code for LOGGED OUT IP STATION-CALL DESTINATION when you want to assign or cancel by Access Code.

STEP 4: **ASPA -** Assign the Access Code for LOGGED OUT IP STATION-CALL DESTINATION when you want to assign or cancel by Access Code.

(a) When SPLIT CALL FORWARDING [S-99] is not used (ASYD, SYS1, Index 79, Bit 2=0)
CI = "N: Normal" SRV = "SSCA: Service Code Appendix" SIDA = 136 (Assignment of LOGGED OUT IP STATION-CALL DESTINATION)
CI ="N: Normal" SRV = "SSCA: Service Code Appendix" SIDA = 137 (Cancel of LOGGED OUT IP STATION-CALL DESTINATION)

(b) When SPLIT CALL FORWARDING is used (ASYD, SYS1, Index 79, Bit 2=1)
CI = "N: Normal" SRV = "SSCA: Service Code Appendix" SIDA = 136 [Assignment of LOGGED OUT IP STATION-CALL DESTINATION (for Trunk)]
CI = "N: Normal" SRV = "SSCA: Service Code Appendix" SIDA = 137 [Cancel of LOGGED OUT IP STATION-CALL DESTINATION (for Trunk)]
CI = "N: Normal" SRV = "SSCA: Service Code Appendix" SIDA = 138 [Assignment of LOGGED OUT IP STATION-CALL DESTINATION (for Station)]
CI = "N: Normal" SRV = "SSCA: Service Code Appendix" SIDA = 139 [Cancel of LOGGED OUT IP STATION-CALL DESTINATION (for Station)]

STEP 5: **ASFC**
SFI:198 = 1 (LOGGED OUT IP STATION-CALL DESTINATION)
SFI 209: Assign as data "1" when CALL FORWARDING - ALL CALLS [C-5] has precedence over LOGGED OUT IP STATION-CALL DESTINATION [L-53].

STEP 6: **AKYD**

Assign this feature to a Function key on the Desktop terminal.

KYN: Key number(1-40)

KYN = 1 (Function Key)

FKY = 148 (LOGGED OUT IP STATION-CALL DESTINATION)

To use Soft Key, assign the ADSL command.

SN = 1

FKY = 148 (LOGGED OUT IP STATION-CALL DESTINATION)

STEP 7: **ACFO/ACFOL/ACFON** - Assign Forwarding Destination Number for this feature per Tenant basis.

TN: Tenant Number

CF: Select either of the following Call Forwarding features.

1 = Call Forwarding - Busy Line

2 = Call Forwarding - Don't Answer

3 = Call Forwarding - Intercept

4 = Night Connection Outside

5 = Logged Out IP Station - Call Destination

CFI: Specify the Call Forwarding Destination

1 = Attendant Console

2 = Station (for ACFO), TELN (for ACFOL/ACFON)

STN: Enter Station Number when assign "2" to "CFI" parameter. (for ACFO)

UGN: User Group Number (for ACFOL/ACFON)

TELN: Enter Telephone Number of Destination Station when assign "2" to "CFI" parameter. (for ACFOL/ACFON)

ADC: Enter Abbreviated Digit Code to the forwarded destination when "CF = Night Connection Outside" is selected.

STEP 8: **ATNR** - Allow Tenant-to-Tenant Connection for this feature.

TRI = 1

STEP 9: **ASYD** - Assign the periodical backup data for this feature.

SYS1, Index 305, Bit 1=1 (Logged Out IP Station - Call Destination Data Save)

L-54 Least Cost Routing for MG-PRI

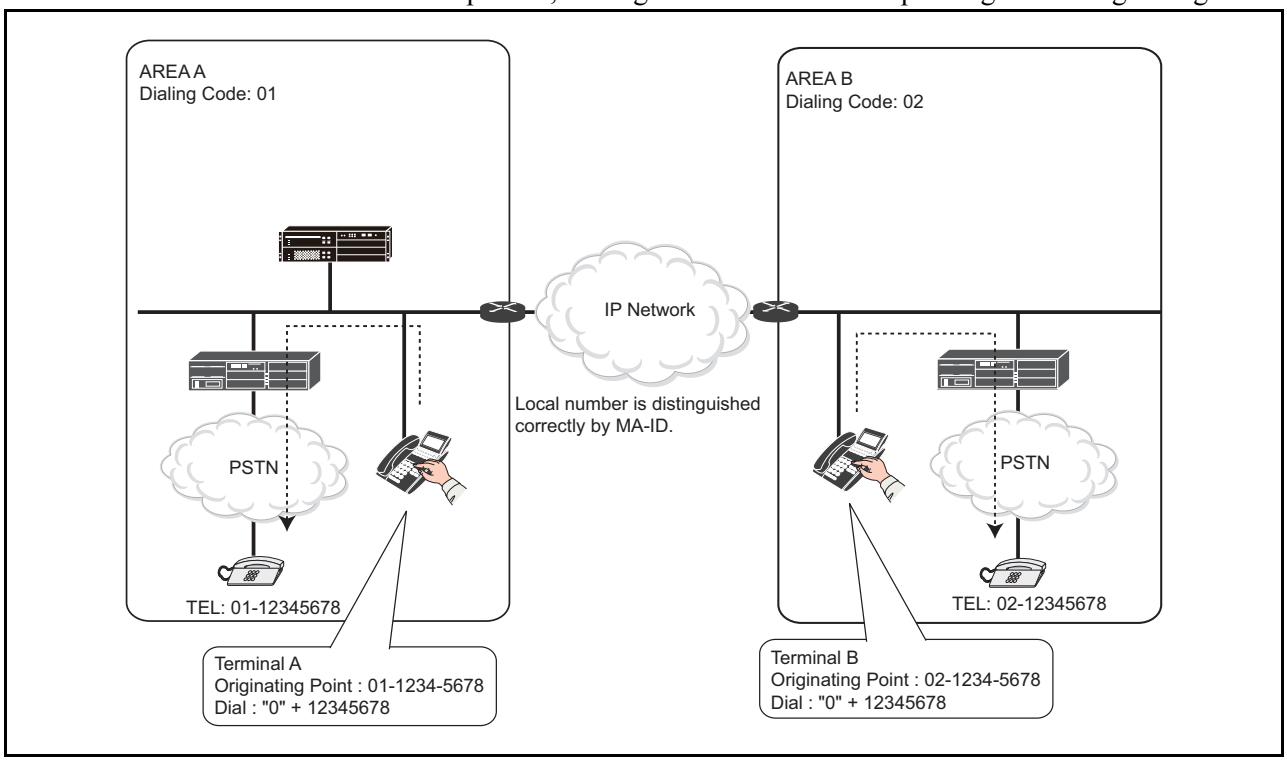
General Description

When multiple MG-PRIs are installed, this feature provides routing suitable for the network by using Message Area IDs added to the MG-PRIs. Once the network administrator registers MA-IDs, the routing can be managed.

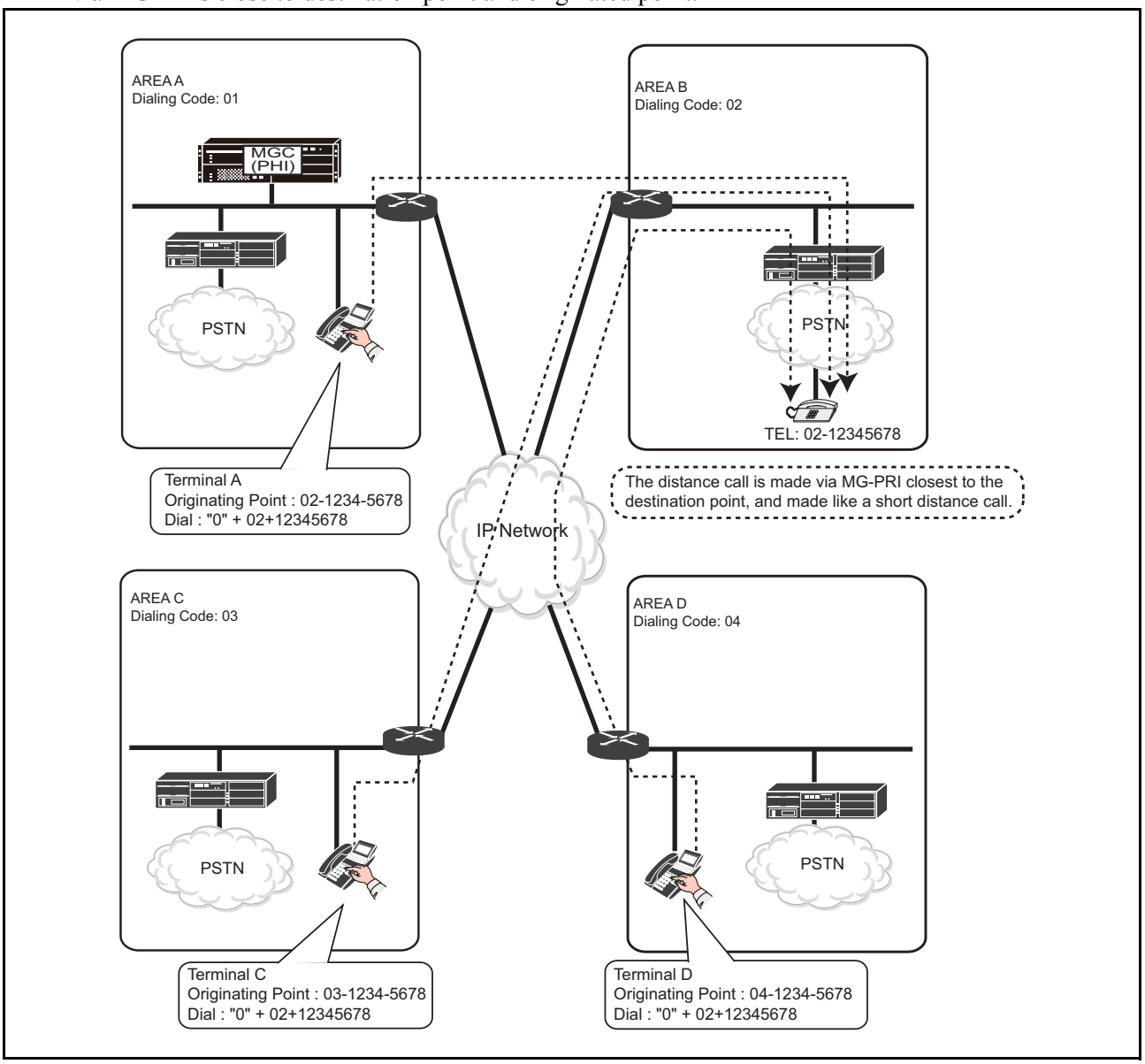
MA-IDs can be assigned to Virtual Line. By assigning MA-ID to Virtual Line, a telephone number assigned to Virtual Line can be notified to the called party. For example, when the pilot number of a company is assigned to Virtual Line, this number can be notified to the called party by placing a call after seizing Virtual Line. This enables the called party to call back to the calling party through the pilot number.

The following figures show examples of using this feature:

- When local call is made from telephones, routing can be determined depending on the originating location.



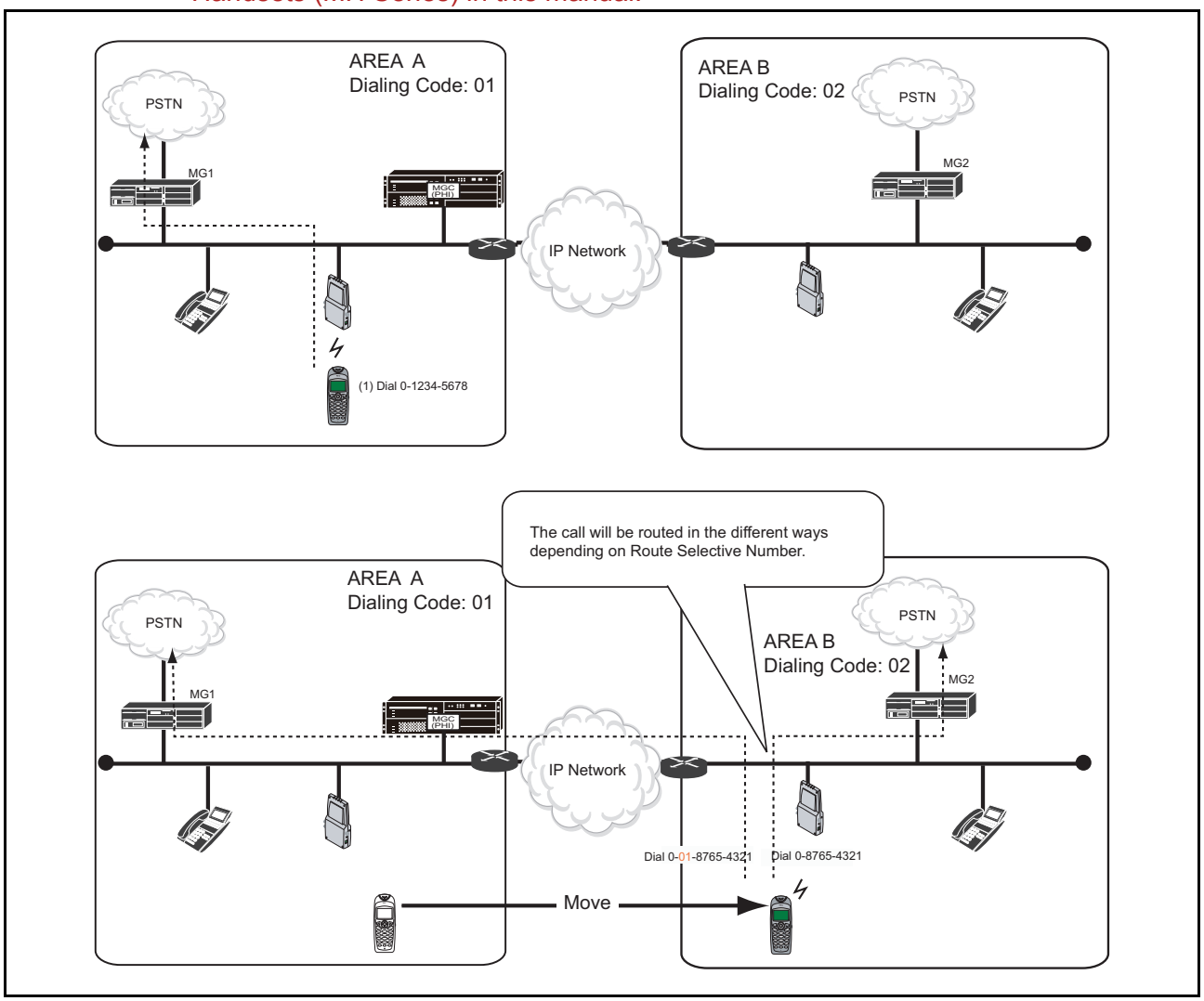
- When outgoing call is made from telephones, cost-effective phone calls can be made through IP network via MG-PRIs close to destination point and originated point.



- When call is originated from SP Controlled SIP terminal (**Note 1**) even though the telephone is moved to another area, cost-effective phone calls can be made through IP network and PSTN via MG-PRIs close to destination point and originated point.
  When call is originated from the SIP Handler Controlled SIP terminal, if the telephone is moved to another area, the MA-ID needs to be changed with the following operations.

  a.) Set the MA-ID by the KIND=LENS parameter in the AMAA command.
  b.) Set the MA-ID by the Access Code.

Note 1: SP Controlled SIP terminals refers to SP Controlled Standard SIP terminals and WLAN Handsets (MH Series) in this manual.



Operating Procedure

Operating Procedure for My Line

<Setting MA-ID by Access Code>

STEP 1. Lift the handset (push monitor/speaker button).

STEP 2. Hear Dial Tone (DT).

STEP 3. Dial Access Code for MA-ID setting.

STEP 4. Hear Special Dial Tone (SPDT).

STEP 5. Dial MA-ID.

STEP 6. Hear Service Set Tone (SST).

STEP 7. Replace the handset (push monitor/speaker button).

<Setting/confirming MA-ID by the Line/Feature key or Soft key of Desktop terminal>

(a) Setting MA-ID

STEP 1. Lift the handset (push monitor/speaker button).

STEP 2. Hear Dial Tone (DT).

STEP 3. Press the Line/Feature key or Soft key for MA-ID setting.

STEP 4. Hear Special Dial Tone (SPDT).

STEP 5. Dial MA-ID.

STEP 6. Hear Service Set Tone (SST).

STEP 7. Replace the handset (push monitor/speaker button).

STEP 8. To make sure the settings, push assigned MA-ID button in Line/Feature keys or Soft keys. If the MA-ID setting has been completed, the LCD displays “MAID” (default) or characters that you registered with AFDD.

(b) Confirming MA-ID

STEP 1. Press Programmable Line/Feature key or Soft key for MA-ID confirmation.

STEP 2. MA-ID is displayed on the upper line of LCD like the following example.

	Setting MA-ID	Confirming MA-ID
Terminal with 16-character LCD	MAID SET MAIN OF	MAID MAIN OFFICE
Terminal with 24-character LCD	MAID SET MAIN OFFICE	MAID MAIN OFFICE

	Setting MA-ID		Confirming MA-ID	
When AMAIL/AMAIN command is not assigned	MAID SET	0001	MAID	0001

Operating Procedure for Virtual Line

<Setting MA-ID by Access Code>

STEP 1. Press Programmable Line/Feature key or Soft key to which Virtual Line is assigned.

STEP 2. Lift the handset (push monitor/speaker button).

STEP 3. Hear Dial Tone (DT).

STEP 4. Dial Access Code for MA-ID setting.

STEP 5. Hear Special Dial Tone (SPDT).

STEP 6. Dial MA-ID.

STEP 7. Hear Service Set Tone (SST).

STEP 8. Replace the handset (push monitor/speaker button).

<Setting/confirming MA-ID by the Line/Feature key or Soft key of Desktop terminal>

(a) Setting MA-ID

STEP 1. Press the Line/Feature key or Soft key to which Virtual Line is assigned.

STEP 2. Lift the handset (push monitor/speaker button).

STEP 3. Hear Dial Tone (DT).

STEP 4. Press the Line/Feature key or Soft key for MA-ID setting.

STEP 5. Hear Special Dial Tone (SPDT).

STEP 6. Dial MA-ID.

STEP 7. Hear Service Set Tone (SST).

STEP 8. Replace the handset (push monitor/speaker button).

(b) Confirming MA-ID

STEP 1. Press the Line/Feature key or Soft key to which Virtual Line is assigned.

STEP 2. Press the Line/Feature key or Soft key for MA-ID confirmation.

STEP 3. MA-ID is displayed on the upper line of LCD like the following example.

	Setting MA-ID	Confirming MA-ID
Terminal with 16-character LCD	MAID SET MAIN OF	MAID MAIN OFFICE
Terminal with 24-character LCD	MAID SET MAIN OFFICE	MAID MAIN OFFICE
When AMAIL/AMAIN command is not assigned	MAID SET 0001	MAID 0001

Service Conditions

1. This feature is available via FCCS network, under the following conditions:
   - Each MA-ID in the FCCS network must be unique.
   - The NDM data has precedence over the LDM data.
   - Each node controls the appropriate MA-ID independently. Thus, the backup of MA-ID must be performed on a node-by-node basis.
2. This feature is available when the originating system is LCR or LCRS.
3. Set each MA-ID to the trunk. When outgoing transfer has been established to a station, the call arriving in the station will be redirected by using route selection patterns according to the MA-ID.
4. A station/trunk that is not assigned MA-ID will perform according to the setting of default MA-ID (0000). Therefore, be sure to assign the Maximum Necessary Digits and route selection pattern for the default MA-ID (0000) in advance. Otherwise, a call origination from the station/trunk without MA-ID will be restricted.
5. MA-ID must be assigned to each station/trunk. Local call originating is performed by selecting the Maximum Necessary Digits and route selection patterns according to MA-ID assigned to the station/trunk.
6. Regardless of the location, station can perform local originating in the specific area.
7. When a station is moved to another area, the station can select previous route selection pattern or new route selection pattern.
8. MA-ID is determined from 0000 to 9999.
9. MA-ID can be selected by using the AMAA command, an Access Code, Line/Feature key or Soft key.
10. If MA-ID is changed while the following features are enabled, Maximum Necessary Digits and the route selection pattern is used according to the newly changed MA-ID:
    - Speed Calling-Station
    - Speed Calling-System
    - Hotline - Outside

- Hotline - CCIS
- Outgoing trunk call from Hotline Station
- Call Transfer
- Call Forwarding-Busy Line
- Call Forwarding-Don't Answer
- Call Forwarding-Logout

11. MA-ID setting is available for both My Line and Virtual Line. Refer to Operation Procedure for more information on MA-ID setting. **Note 2**

Note 2: When setting MA-ID, note the following:
- To assign MA-ID to My Line, assign a Service Feature Restriction Class (SFC) that allows SFI 213 = 1 (MA-ID setting) to the station by the ASFC/ASFCN command.
- Regarding Sub Lines, MA-ID can be assigned only to Virtual Line. MA-ID cannot be assigned to the other Sub Lines (My Line of other Desktop terminal, a Station Number of Single Line Telephone and a PS/SIP line).
- To assign MA-ID to Virtual Line, assign a Service Feature Restriction Class (SFC) that allows SFI 213 = 1 (MA-ID setting) to both My Line and Virtual Line by the ASFC/ASFCN command.

12. When Virtual Line is seized and MA-ID confirmation is performed by pushing a Line/Feature key or Soft key, MA-ID of Virtual Line appears on the LCD. Refer to Operation Procedure for more information on MA-ID confirming. **Note 3**

Note 3: If MA-ID is confirmed after seizing Sub Lines other than Virtual Line, MA-ID of My Line appears on the LCD.

13. Note the following when using the Multi Line services of Desktop terminals:

(a) When an outside call is made after seizing My Line or Sub Lines other than Virtual Line (My Line of another station, a Station Number of Single Line Telephone and a PS/SIP line) that are assigned to the Line/Feature key on the Desktop terminal and place an outside call, MA-ID of My Line is used to route the call.

(b) If an outside call is made after seizing Virtual Line that is assigned to the Line/Feature key on the Desktop terminal, MA-ID used to route the call differs depending on the combination of the data settings of SFI261 (Maximum Necessary Digits / Route Selection for using Virtual Line) for My Line and Virtual Line. Refer to the following table for more details.

		Virtual Line (SFI261)	
		0	1
My Line (SFI261)	0	MA-ID of My Line	MA-ID of My Line
	1	MA-ID of My Line	MA-ID of Virtual Line

14. When making a call by using Sub Lines including Virtual Line, a station number of My Line is assigned is output to SMDR as the calling party of the outgoing call.

    Note: The MA-ID of the network the My Line is assigned is also output to the SMDR when SMDR Multi Area ID via IP [S-144] is in service.

15. Note the following when using HOT LINE - OUTSIDE [H-9], HOT LINE - CCIS [H-12] or BROKERAGE HOT LINE - CCIS [B-10]:

    (a) If an outside call is made after seizing Virtual Line that is assigned to the Line/Feature key on the Desktop terminal (Hotline station), MA-ID used to route the call differs depending on the combination of the data settings of SFI261 (Maximum Necessary Digits / Route Selection for using Virtual Line) for My Line and Virtual Line. Refer to the following table for more details.

		Virtual Line (SFI261)	
		0	1
My Line (SFI261)	0	MA-ID of My Line	MA-ID of My Line
	1	MA-ID of My Line	MA-ID of Virtual Line

16. MA-ID can be backed up and verified by using the related commands (MEM_HDD/MEM_HDD_N) and Telephony Server Maintenance Menu. The MA-ID can be easily modified by terminal and the Telephony Server sides, which might lead to verification error. Therefore, the backup data should be synchronized with the existing one. For detailed information of Office Data Backup, refer to Operations and Maintenance Manual.

17. Using System Data diagnoses and backs up the MA-ID data periodically. The result is indicated in the system message (7-O/7-P).

18. If the MA-ID data is stored in the flash card, the data will be loaded in the system initialization.

19. In the case of using emergency call and special service call, an MG closest to the originating terminal will be used.

20. If MA-ID assigned to each network is changed by the AMAA command in operation, SP Controlled SIP terminals belonging to the network will be available after the Station registers re-location.

21. When an SP Controlled SIP terminal is unregistered, the MA-ID that it uses is reset to “0.”

22. MA-ID registration by the AMAA command is not necessary for SP Controlled SIP terminals.

23. When SP Controlled SIP terminal originates a call, the call will take a route by using a MA-ID, which the Station registered as a location.

24. SP Controlled SIP terminals cannot manage MA-IDs by Special Access Codes.

25. SIP Handler Controlled SIP terminals can manage MA-IDs by Special Access Codes but MA-ID cannot be confirmed.

26. To perform Call Forwarding feature of SP Controlled SIP terminal, which is assigned by the ACFS command, MA-ID (0000) is used as a default.

27. When SIP terminal is assigned as a destination point by other Stations, the MA-ID to be taken are as follows, depending on a type of the terminals:

Terminal Assigned Call Transfer	MA-ID
Telephone	MA-ID of a terminal that has transferred a call.
PS	MA-ID of CS with which PS transfers a call.
SP Controlled SIP terminal	MA-ID of a place in which SP Controlled SIP terminal used to register as its location.

28. When SP Controlled SIP terminal originates a call by abbreviated digit code, the call takes a route with predetermined MA-IDs.

29. When Call Forwarding Data has been saved/loaded by the backup commands (MEM_HDD, HDD_FDD) and Telephony Server Maintenance Menu, make sure the MA-IDs are saved/loaded.

30. When SP Controlled SIP terminal has been assigned the out line used for a destination point of Call Forwarding features, the SP Controlled SIP terminal transfers a call with MA-ID of a network where the station has registered at that time.

Interactions

1. When a Desktop terminal accommodating a line of SP Controlled SIP terminal makes an external call via the SIP line, routing is performed by using MA-ID of Desktop terminal My Line.

2. When SP Controlled SIP terminal accommodating a line of SP Controlled SIP terminal makes a call via the SIP line, routing is performed with MA-ID assigned to the location where SP Controlled SIP terminal has registered.

3. When a Desktop terminal accommodating a line of the SIP Handler Controlled SIP terminal makes a call, routing is performed with MA-ID of SIP Handler Controlled SIP terminal My Line.

4. When the SIP Handler Controlled SIP terminal is in Multi Line Mode and its My Line is used by a different terminal, MA-ID cannot be assigned to the SIP Handler Controlled SIP terminal by using an Access Code because this operation attempts to assign the MA-ID from the Sub Line on the SIP Handler Controlled SIP terminal.

5. When the SIP Handler Controlled SIP terminal is used, this feature does not work if KIND=Network is set through the AMAA command. SIP Handler Controlled SIP terminals need to have MA-ID assigned in the AMAA command by the KIND=LENS parameter.

Programming

STEP 1: **ASYDL** - Specify the applied object type to be modified tenant data
SYS1
Index 801, Bit 1=0/1 (per tenant/common)

STEP 2: **AMAIL/AMAIN** - Specify MA-ID and the area information

STEP 3: **AMAA** - Assign MA-ID allocation data.

KIND: Select LENS or Network.
LENS: Assigned to the originating.
Network: Assigned to a network.

In case of SP Controlled SIP terminal:
Select KIND=Network and assign MA-ID.

Note:

- MA-ID cannot be assigned to each SP Controlled SIP terminal.
- The MA-ID of KIND=Network is applied when the SP Controlled SIP terminal registers to the Telephony Server. If MA-ID is not assigned, MA-ID=0 is applied.
- If the MA-ID is modified with AMAA command for each network in operation, MA-ID of the SP Controlled SIP terminal in the network does not become effective until the SP Controlled SIP terminal registers to the Telephony Server.
- Even if KIND=Network is set for any equipment other than the SP Controlled SIP terminal, this feature cannot be used.
- Even if MA-ID is assigned to the station number of SP Controlled SIP terminal, the registration information is not displayed in the setting data of the AMAA command.

In case of PS:
Assign MA-ID to CS by selecting KIND=LENS and FUNCTION=CS.

In case of others:
Select KIND=LENS when assigning MA-ID.

STEP 4: **ANPD/ANPDN** - Specify a required minimum of the first digit of MA-ID Access Code.
Specify the appropriate items, which are used for the minimum number of digits needed to determine the service required to the first digit received (pre-translation).

STEP 5: **ASPA/ASPAN** - Specify the MA-ID access code (ACC)
CI="N: Normal"
SRV="SSCA: Service Code Appendix"
SIDA=142 (MA-ID Setting)

STEP 6: **ASFC/ASFCN** -Assign a Service Feature Restriction Class of the station.
Enable appropriate Service Feature Restriction Class (SFC) to take MA-ID Setting as follows:

SFC: Specify the Service Feature Restriction Class (1-15)
SFI 213=1 (MA-ID Setting)
RES=1 (MA-ID Setting is allowed)
SFI 261 (Maximum Necessary Digits / Route Selection for using Virtual Line) **Note 1**
RES=0/1 (MA-ID of My Line / MA-ID of Virtual Line)

Note 1: When Virtual Line is used, this data setting is required for both My Line and Virtual Line.

STEP 7: **AKYD** - Assign the function to the Programmable Line/Feature of IP terminals
KYI=1 (Line/Feature key)
FKY=151 (MA-ID Setting)

STEP 8: **ADSL** - Add the function to the Soft keys of IP terminals (Only when using a Soft key)
SKN: Specify the Soft Key Number (0~15)
SN=1 (DT connection when calling)
FKY=151 (MA-ID Setting)

STEP 9: **AFDD** - Specify the Soft key name displayed on LCD (Only when using a Soft key)
FKY=151 (MA-ID Setting)
DISP: Specify characters that will display on LCD (a maximum of 6 digits)
If you do not enter the MA-ID in DISP, the default value ("MAID") is set automatically.

STEP 10: **AMNDL** - Specify a required maximum of the digits for MA-ID and DC.
When activating LEAST COST ROUTING FOR MG-PRI [L-54] via FCCS network, this data must be assigned on a node-by-node basis.

Note: When the necessary data such as Destination Code and Maximum Necessary Digits Code has been already assigned, these data will be read out.

STEP 11: **ARNPL** - Specify the Reverse Numbering Plan Data for LDM

STEP 12: **AFRSL/AFRSN** - Specify an Outgoing Route Selection Number (OPR) **Note 2**
Specify the MA-ID and the appropriate items, which is used to select a route in trunk routes (LCR/LCRS) registered with flexible routes (Dummy Route) when the dialed number and MA-ID is used. The setting examples are described in **Examples of Data Settings**.

STEP 13: **AOPRL/AOPRN** - Specify routes and the condition for using the outgoing route selection patterns
Specify the appropriate items, which are used to determine the actual route priority and the conditions. The setting examples are described in **Examples of Data Settings**.

Note 2: By changing the system data, you can reduce the amount of registered data for the AFRSN command, and the office data consumption. For details, see 2.3 Reduction of Data Consumption of the AFRSN command, CHAPTER 5 DATA PROGRAMMING of Data Programming Manual - FCCS.

[Examples of Data Settings]

Setting routes varies depending on your network. The network administrator should consider the routes carefully and create the effective routes for the network by setting Tenant Number (TN), Logical Route Number (LGRT), Number Pattern Code (NPC), MA-ID, and Outgoing Route Selection Number (OPR). For more information, refer to the Command Manual.
The following shows two examples indicating how to work by setting these items.

1. In the case of adding MA-ID

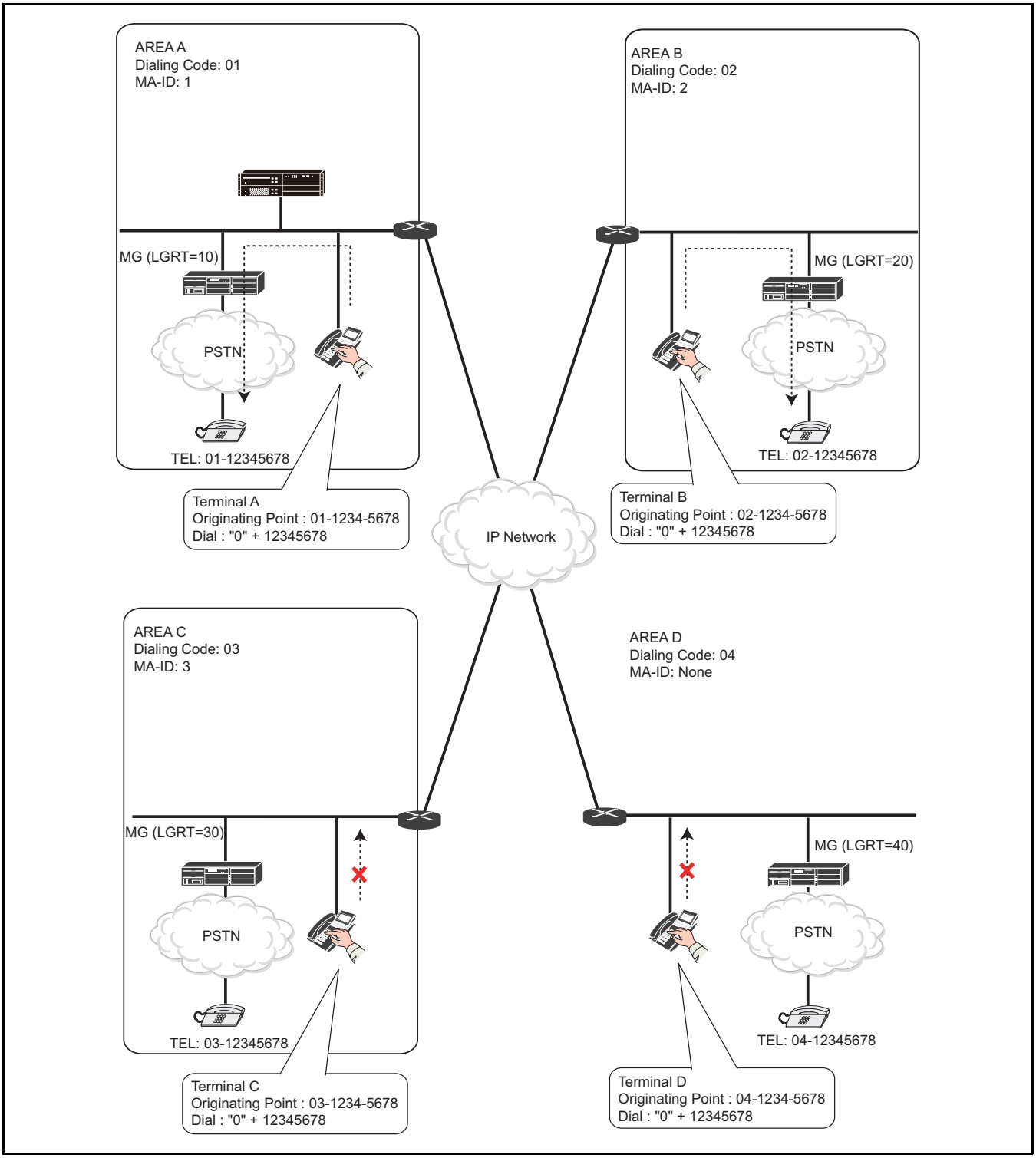
STEP 1 The following are specified by AFRSL.

TN	LGRT	NPC	MA-ID	OPR
1	300	012	1	101
1	300	012	2	102

STEP 2 The following are specified by AOPRL.

OPR101→LGRT=10
OPR102→LGRT=20

RESULT When MA-IDs have been added to MGs, the call will be routed via the appropriate MG. In this case, terminals belonging to unknown MA-IDs or not belonging to any MA-ID will not be able to send any call.



2. In the case of default registration

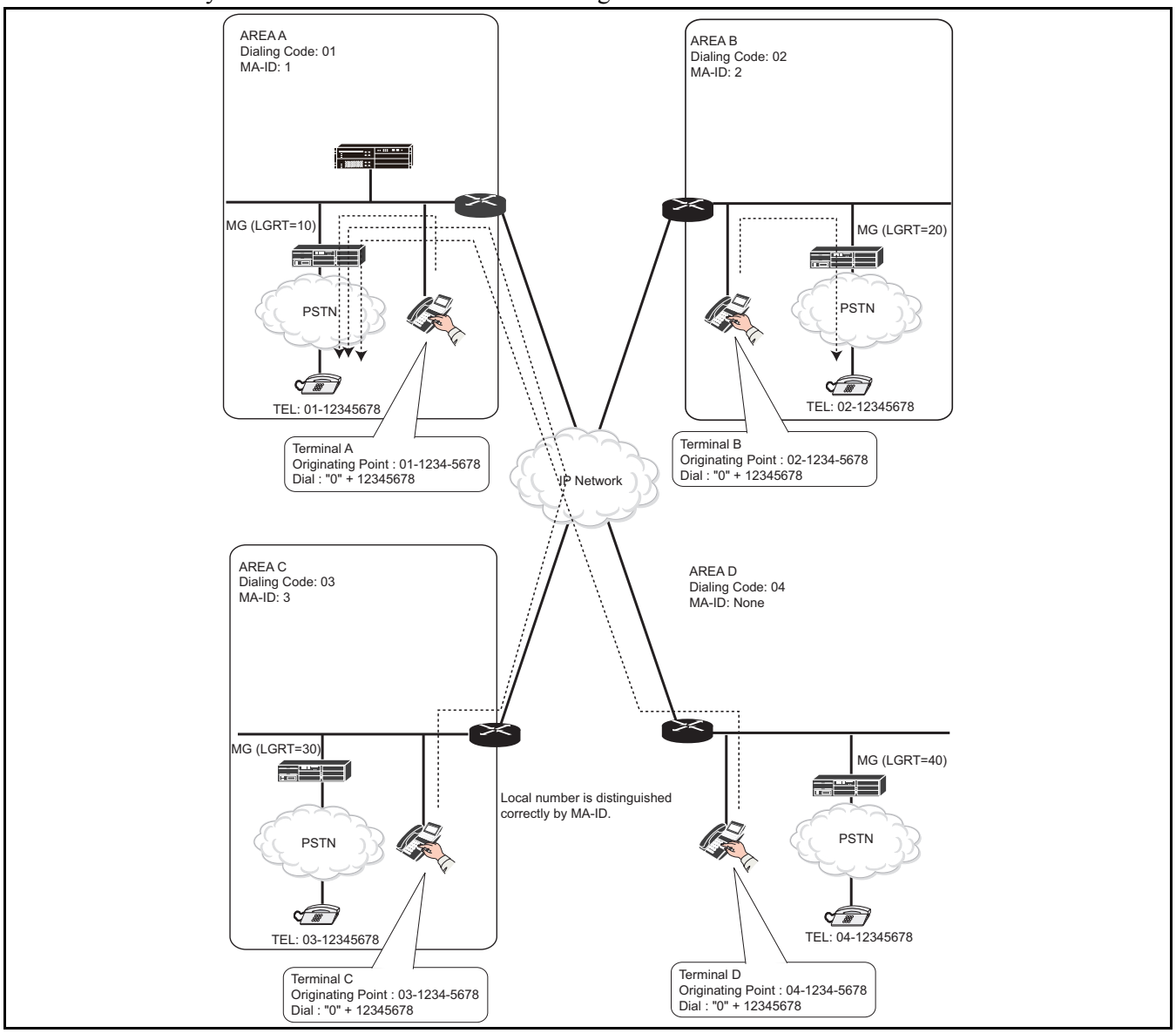
STEP 1 The following are specified by AFRSL.

TN	LGRT	NPC	MA-ID	OPR
1	300	012	0 (Default)	100
1	300	012	2	102

STEP 2 The following are specified by AOPRL.

OPR100→LGRT=10
OPR102→LGRT=20

RESULT When default MA-ID (0000) has been added, the calls belonging to unknown MA-ID or not belonging to any MA-ID will be routed via MG having its default MA-ID.



L-55 Location Diversity

Location Diversity [L-55] is comprised of the following two services: FCCS Cluster and SR-MGC with FCCS Cluster. These services are explained under the following headings:

FCCS Cluster

General Description (FCCS Cluster)

Operating Procedure (FCCS Cluster)

Service Conditions (FCCS Cluster)

Programming (FCCS Cluster)

SR-MGC with FCCS Cluster

General Description (SR-MGC with FCCS Cluster)

Operating Procedure (SR-MGC with FCCS Cluster)

Service Conditions (SR-MGC with FCCS Cluster)

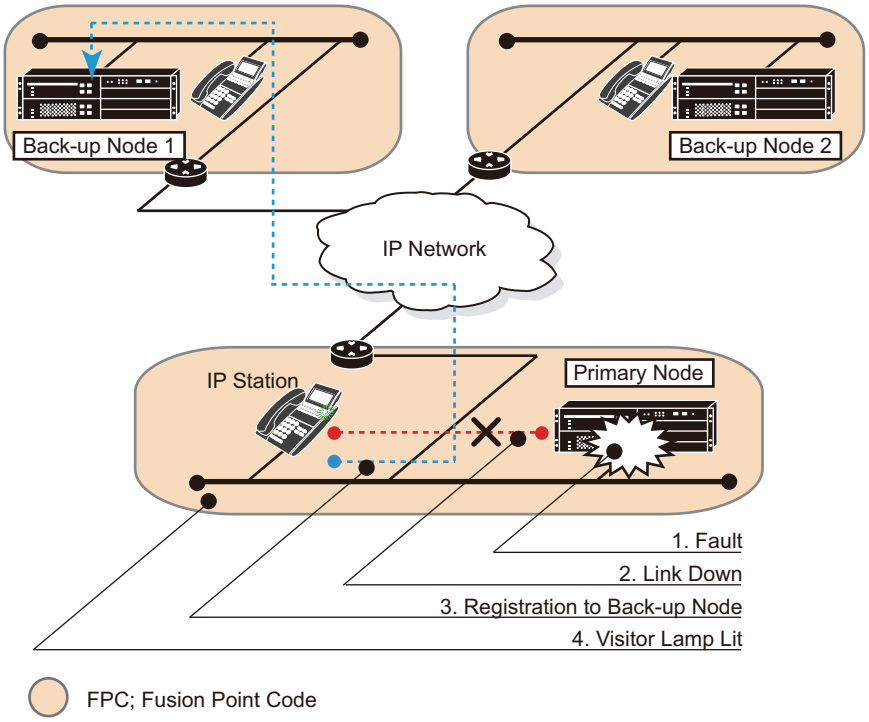
Programming (SR-MGC with FCCS Cluster)

FCCS Cluster

General Description (FCCS Cluster)

In the FCCS network, when a communication failure occurs between terminals (IP stations/equipment) and Primary Node, the terminals can automatically Re-register to a predetermined Backup Node. While the terminals are placed under the control of Backup Node, the terminals can operate as they were in Primary Node. By using MAC Authentication feature, the terminals automatically become operative after the Registration to Backup Node.

The following figure describes the outline of this feature.



After the terminals start to operate in Backup Node, the Backup Node monitors the connection status between Primary Node and the terminals periodically. When Primary Node recovers from a failure, terminals can go back to under the control of Primary Node according to the instruction from the Backup Node. The office data setting (AFCMN command) can decide on the timing of the recovery of the terminals (to Primary Node).

- When the connections between terminals and Primary Node are reestablished and the terminals become operative (Immediately)
- At designated times (By the office data setting)
- When the VISITOR key on each terminal is pressed or the terminal is reset by the operation (CBKS command, etc.) from PCPro (Manual operation)

Operating Procedure (FCCS Cluster)

<Connection Failure Between Terminals (IP stations/equipment) and Primary Node>

No manual operation is required.

- Terminals automatically re-register to Backup Node.
- The VISITOR key on the IP terminal (including Soft Phone) lights within eight seconds after the station completes the Registration process. **Note 1**, **Note 2**

  Note 1: For terminals operated in MAC Authentication mode only. For terminals operated in Login mode, Login screen is displayed.

  Note 2: If the VISITOR key is pressed while it lights green, the terminal will be reset. Note that Call Forwarding data, Login Password, Speed Dialing data and the status Message Waiting Lamp may be cleared after the reset of the terminal.

- When connection failure occurs between terminals and Primary Node, each terminal operates as follows:

Status of Terminal	Operation
Idle	After the terminal is reset, the terminal registers to Backup Node.
Busy	Call in progress can be maintained. After the call is completed (on-hook), the terminal is reset and registers to Backup Node. Trunk (MG) is restarted after a few minutes (from busy status), and then registers to Backup Node.
During Ringback Tone	Ringback Tone can be maintained. After a few minutes, the terminal is reset and registers to Backup Node. **Note 3**
During Ringing Tone	Incoming Call can be kept ringing. After a few minutes, the terminal is reset and registers to Backup Node.

Note 3: When Soft Phone (SIP) is used, after going on-hook, the terminal is reset and registers to Backup Node.

- The stations/equipment that can be rescued are as follows.

Terminal to be rescued by this feature	
Protocol mode	**Terminal**
Proprietary Protocol	DtermIP/IP Enabled Dterm
	Soft Phone [Proprietary Protocol]
	Analog 2MC
	MCMG [SCA-4LC2COTA/MG-4LC2COTA]
	MG(SIP)
	MG-COT Card [SCA-6COTA]
	MG-COT Card [SCA-6COTB/SCA-6COTC]
	UG50 (DLC) **Note 4**
	UG50 (LC) [Proprietary Protocol] **Note 4**
	UG50 (PRT 1.5M) [Proprietary Protocol] **Note 4**
	UG50 (PRT 2M) [Proprietary Protocol] **Note 4**
	UG50 (COT-TYPE1) **Note 4**
	UG50 (COT-TYPE2) [Proprietary Protocol] **Note 4**
	UG50-2MC
	UG50-24PRIA [Proprietary Protocol]
	UG50-30PRIA [Proprietary Protocol]
	UG50-4LC2COTA (COT)
	UG50-6COT [Proprietary Protocol]
SIP	DT800 Series
	DT700 Series
	DtermIP (SIP)
	8LC Card [SCA-8LCA/SCA-8LCA-EMEA/SCA-8LCC/SCA-8LCC-EMEA]
	Soft Phone (SIP)
	MG(BRI)
	MG-T1 (SIP) [SCA-24DTIA]
	MG-COT Card [SCA-6COTB/SCA-6COTC]
	UG50 (LC) [SIP] **Note 4**
	UG50 (PRT 1.5M) [SIP] **Note 4**
	UG50 (PRT 2M) [SIP] **Note 4**
	UG50 (COT-TYPE2) [SIP] **Note 4**
	UG50-8LC
	UG50-24PRIA [SIP]
	UG50-30PRIA [SIP]
	UG50-6COT [SIP]

Note 4: Available from FP95 V4-114 Issue 2.0.

<Detection of Communication Fault>

If Health Check NG occurs between the terminal and the System, the terminal is reset under the following conditions. **Note 5**

- If the terminal sends Health Check packets to the System:
  The terminal is reset when the Health Check time-out occurs.
- If the System sends Health Check packets to the terminal:
  The terminal is reset when the terminal does not receive Health Check packets for a certain period of time.

Note 5: Health Check is performed using LAN1 port of the Telephony Server as the interface.

After the terminal is reset, it registers to Primary Node. If the terminal cannot register to Primary Node, it registers to Backup Node.

The following table shows the time required for terminals to fail over from Primary Node to Backup Node.

Terminal		Maximum time required for terminals to fail over from Primary Node to Backup Node (Note 6)
DT800/DT700 Series	SIP	7 min.
DtermIP	Proprietary Protocol	3 min. 45 sec
	SIP	7 min.
Soft Phone	Proprietary Protocol	3 min. 15 sec
	SIP	3 min. 45 sec
Analog 2MC		5 min.
8LC Card [SCA-8LCA/SCA-8LCA-EMEA/ SCA-8LCC/SCA-8LCC-EMEA]		4 min. 30 sec
MCMG [SCA-4LC2COTA/MG-4LC2COTA]		5 min. 45 sec
MG-COT [SCA-6COTA]		4 min. 45 sec
MG-COT [SCA-6COTB]	Proprietary Protocol	4 min. 45 sec
	SIP	4 min. 30 sec
MG-COT [SCA-6COTC]	Proprietary Protocol	4 min. 45 sec
	SIP	4 min. 30 sec
MG(BRI) [SCA-2BRIA/MG-2BRIA]		5 min.
MG(PRI) [SCA-24PRIA/MG-24PRIA]	Proprietary Protocol	4 min. 45 sec
	SIP	5 min.
MG(PRI) [SCA-30PRIA/MG-30PRIA]	Proprietary Protocol	4 min. 45 sec
	SIP	5 min.
MG-T1(SIP) [SCA-24DTIA]		5 min.
MG(SIP)		5 min.
UG50-2MC		3 min. 45 sec

Terminal		Maximum time required for terminals to fail over from Primary Node to Backup Node (Note 6)
UG50-8LC		4 min. 15 sec
UG50-24PRIA	Proprietary Protocol	3 min.
	SIP	4 min.
UG50-30PRIA	Proprietary Protocol	3 min.
	SIP	4 min.
UG50-4LC2COTA (COT)		3 min. 45 sec
UG50-6COT	Proprietary Protocol	3 min. 45 sec
	SIP	4 min. 15 sec
UG50 (DLC) Note 7		3 min. 45 sec
UG50 (LC) Note 7	Proprietary Protocol	3 min. 45 sec
	SIP	4 min. 15 sec
UG50 (PRT 1.5M) Note 7	Proprietary Protocol	3 min
	SIP	4 min
UG50 (PRT 2M) Note 7	Proprietary Protocol	3 min
	SIP	4 min
UG50 (COT-TYPE1) Note 7		3 min. 45 sec
UG50 (COT-TYPE2) Note 7	Proprietary Protocol	3 min. 45 sec
	SIP	4 min. 15 sec

Note 6: Time interval between the detection of a communication failure and the moment when the terminal sends Initial Setting Request to Backup Node.

Note 7: Available from FP95 V4-114 Issue 2.0.

<Connection Recovery Between terminals (IP stations/equipment) and Primary Node>

Note: When using IP terminal (including Soft Phone), assign the VISITOR key by the AUPKDN command in advance.

(a) When "After confirmation of connection establishment with the Home Node" is specified on AFCMN command:

No manual operation is required.

- The terminal is reset and automatically registers to Primary Node after the connection establishment is verified (as a result of connectivity check).

Note: The terminals operated in Login mode will not automatically go back to Primary Node. By pressing the VISITOR key while it flashes green, the terminal can register (Login screen is displayed) to Primary Node safely after the terminal is reset automatically.

- When connection failure between the terminal and Primary Node is recovered, the terminal operates as follows:

Status of Terminal	Operation
Idle	When the connection establishment is verified (as a result of connectivity check), the terminal is reset and registers to Primary Node.
Busy	Call in progress can be maintained. When the connection establishment is verified (as a result of connectivity check), the terminal is reset and registers to Primary Node after the call is completed (on-hook).
During Ringback Tone	Ringback Tone can be maintained. When the connection establishment is verified (as a result of connectivity check), the terminal is reset and registers to Primary Node after the terminal goes on-hook.
During Ringing Tone	Incoming Call can be kept ringing. When the connection establishment is verified (as a result of connectivity check), the terminal is reset and registers to Primary Node after the calling party goes on-hook.

Note: Trunks (such as MGs) will not be reset and register to Primary Node until all their channels become idle.

(b) When "After station data synchronization is completed" is specified on AFCMN command:

No manual operation is required.

- According to the time assigned to "Station data synchronization processing time" parameter on the AFCMN command, the terminal is reset and automatically registers to Primary Node after the connection establishment is verified and station data synchronization is completed (as a result of connectivity check).

Note: After the connection establishment is verified, the green LED of the VISITOR key on IP terminal (including Softphone) changes from "steady on" to "flashing". The terminals operated in Login mode will not automatically go back to Primary Node. By pressing the VISITOR key while it flashes green, the terminal can register (Login screen is displayed) to Primary Node safely after the terminal is reset. LAN1 is used for verifying the connection establishment between Primary Node and Backup Node or between Primary Node and the terminals.

- When a connection failure between the terminal and Primary Node is recovered, the terminal operates as follows:

Status of Terminal	Operation
Idle	When the connection establishment is verified (as a result of connectivity check), the terminal is reset and registers to Primary Node.

Status of Terminal	Operation
Busy	Call in progress can be maintained. When the connection establishment is verified (as a result of connectivity check), the terminal is reset and registers to Primary Node after the call is completed (on-hook).
During Ringback Tone	Ringback Tone can be maintained. When the connection establishment is verified (as a result of connectivity check), the terminal is reset and registers to Primary Node after the terminal goes on-hook.
During Ringing Tone	Incoming Call can be kept ringing. When the connection establishment is verified (as a result of connectivity check), the terminal is reset and registers to Primary Node after the calling party goes on-hook.

Note: Trunks (such as MGs) will not be reset and register to Primary Node until all their channels become idle.

(c) When "Only when [Visitor Indication] button is pressed" is specified on AFCMN command:

- For IP terminal (including Soft Phone):

  When the connection establishment between Primary Node and Backup Node can be verified (as a result of connectivity check), the status of the VISITOR key on IP terminal changes from "lighting green" to "flashing green". By pressing the VISITOR key while it flashes green, the terminal is reset and registers to Primary Node.

  Note: Be sure not to reset the terminal by [Visitor Indication] button or CBKS command until the [Visitor Indication] button flashes green. Otherwise, Terminal Data (Transfer Data, Login Password, One-touch Speed Dialing, and Message Waiting Lamp Status) may be lost.

- For terminals except for IP terminal:

  Terminals except for IP terminal can register to Primary Node after they are reset by using CBKS command, etc.

  Note: In the case of MC&MG-COT:
  If MC stations are specified on CBKS command, only Media Converter (MC) side will be reset. At this time, Media Gateway (MG) side will not be reset. To reset the entire MC&MG-COT, specify the appropriate route(s) on CBKS command.

- When a connection failure between the terminal and Primary Node is recovered, the terminal operates as follows:

Status of Terminal	Recovery Method	Operation
Idle	Pressing the VISITOR key	Terminals register to Primary Node after they are reset.
	Using CBKS command	
Busy	Pressing the VISITOR key	Not Available.
	Using CBKS command	Call in progress will be disconnected. Terminals register to Primary Node after they are reset.
During Ringback Tone	Pressing the VISITOR key	Not Available.
	Using CBKS command	Outgoing Call will be disconnected. Terminals register to Primary Node after they are reset.
During Ringing Tone	Pressing the VISITOR key	Incoming Call will be disconnected. Terminals register to Primary Node after they are reset.
	Using CBKS command	

<Connectivity Check>

a.) When any terminal is registered in Backup Node, the node checks the connectivity with the terminals via Primary Node.

b.) When Primary Node recovers and the connectivity check succeeds continuously three times, the reset instruction will be sent to the terminals after the station data synchronization. **Note 8**

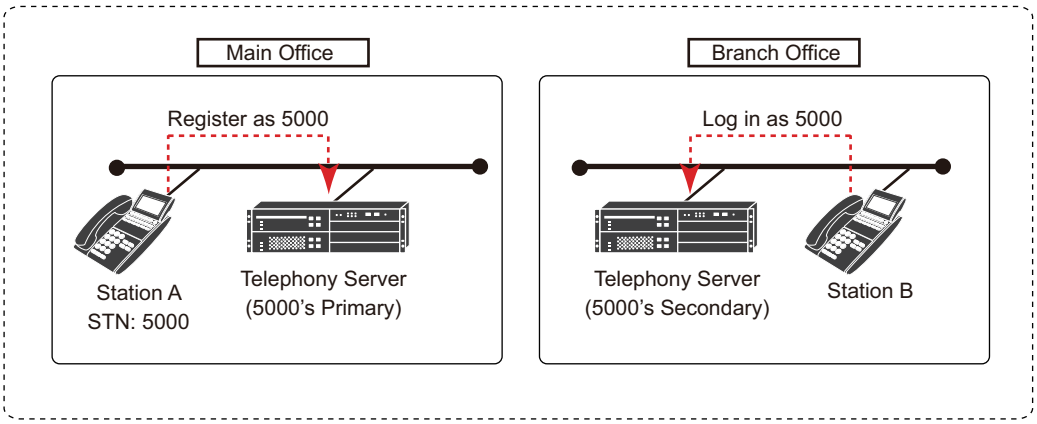
c.) LAN1 is used for the connectivity check.

d.) To complete connectivity check, about 10-15 minutes are required. Depends on the network condition, the number of accommodated devices, and the time setting of connectivity check, it may take longer to complete.

Note 8: Even if the station data synchronization after a connectivity check failed, the reset instruction will be sent to the terminals when the subsequent connectivity check succeeds continuously three times. In this case, the station data that has been set in Backup Node may be disabled after the terminals register to Primary Node.

<Example When Visitor Station Log On To The Secondary Node>

Even if Primary Telephony Server operates normally, a Visitor Station whose Station Number is assigned on Secondary Telephony Server can log into the Secondary Telephony Server by using its Station Number. For example, when a main office user has an IP station in the branch office Telephony Server, the user can log on to the branch office Telephony Server by using the own Station Number (5000) of the main office Station. At this time, user logs into the branch office Telephony Server as a Visitor Station. While the Station Number (5000) is used in Visitor mode, VISITOR key on the terminal at the main office flashes red (The Station Number (5000) cannot be used in main office).



Service Conditions (FCCS Cluster)

1. This feature is performed between systems that are set up with FCCS networks; therefore, the numbering plans (including station numbers) have to be made with NDM. Also, each feature performed between systems is restricted by conditions for FCCS network.
2. This feature is not available for OAI or ACD system.
3. To enable FCCS Cluster in Hotel system, see the Data Assignment for FCCS Cluster section in Data Programing Manual - Hotel.
4. FCCS Cluster is available for the following IP stations/equipment.

IP Station/Equipment		Firmware Version
DT800/DT700 Series	SIP	Available since the first version
DtermIP/IP Enabled Dterm	Proprietary Protocol	Version 2.06 or later
DtermIP (SIP)	SIP	Available since the first version (Version 1.00 or later)
DtermSP30	Proprietary Protocol	Version 5 or later
	SIP	Version 7 or later
Analog 2MC		SP-3753 2AMC PROG-B Issue 15 or later
8LC Card [SCA-8LCA/SCA-8LCA-EMEA]		SP-3893 8MC PROG-A Issue 1 or later
8LC Card [SCA-8LCC/SCA-8LCC-EMEA]		SP-3983 8MC PROG-B Issue 1 or later
MCMG Card [SCA-4LC2COTA]		SP-3927 MCMG PROG-B Issue 1 or later
MCMG Box [MG-4LC2COTA]		
MG-COT [SCA-6COTA]		SP-3875 6MGCOT PROG-A Issue 1 or later
MG-COT [SCA-6COTB]	Proprietary Protocol	SP-3948 6MGCOT PROG-A Issue 1 or later
	SIP	SP-3948 6MGCOT PROG-A Issue 1 or later
MG-COT [SCA-6COTC]	Proprietary Protocol	SP-3948 6MGCOT PROG-A Issue 1 or later
	SIP	SP-3948 6MGCOT PROG-A Issue 1 or later
MG(BRI) Card [SCA-2BRIA]	SIP	SP-3826 BRI MG SIP BSC PROG-A Issue 1 later
MG(BRI) Box [MG-2BRIA]		

IP Station/Equipment		Firmware Version
MG(PRI) Card [SCA-24PRIA]	Proprietary Protocol	Available since the first version (SP-3884 MG PRI PROG-A Issue 1 or later)
	SIP	SP-3884 MG PRI PROG-A Issue 5 or later (except for Issue 12)
MG(PRI) Box [MG-24PRIA]	Proprietary Protocol	Available since the first version (SP-3884 MG PRI PROG-A Issue 1 or later)
	SIP	SP-3884 MG PRI PROG-A Issue 5 or later (except for Issue 12)
MG(PRI) Card [SCA-30PRIA]	Proprietary Protocol	Available since the first version (SP-3884 MG PRI PROG-A Issue 1 or later)
	SIP	SP-3884 MG PRI PROG-A Issue 5 or later (except for Issue 12)
MG(PRI) Box [MG-30PRIA]	Proprietary Protocol	Available since the first version (SP-3884 MG PRI PROG-A Issue 1 or later)
	SIP	SP-3884 MG PRI PROG-A Issue 5 or later (except for Issue 12)
MG-T1(SIP) [SCA-24DTIA]		SP-3926 MG DTI PROG-A Issue 1A or later
MG (SIP) 16	Proprietary Protocol	SP-3843 SIPMG PROG-B Issue 5 or later
MG (SIP) 16 [SCA-16SIPMGA]	Proprietary Protocol	SP-3905 SIPMG (16) PROG-A Issue 2 or later
MG (SIP) 16 [MG-16SIPMGA]		
MG (SIP) 16 [SCA-16SIPMG(US)]	Proprietary Protocol	SP-3988 MGSIP (16) PROG-G Issue 1 or later
MG-SIP96	Proprietary Protocol	SP-3987 MGSIP PROG-B Issue 1 or later
MG-SIP128 [MG-128SIPMGL-A]	Proprietary Protocol	SP-4089 MGSIP(MM) PROG-A Issue 1 or later
MG-SIP128 [MG-128SIPMGG-B/MG-128SIPMGG]	Proprietary Protocol	SP-4058 MGSIP PROG-J Issue 1 or later
		SP-4051 MGSIP PROG-G Issue 1 or later

IP Station/Equipment		Firmware Version
MG-SIP128 [MG-128SIPMGJ-B/MG-128SIPMGJ]	Proprietary Protocol	SP-4032 MGSIP PROG-E Issue 1 or later
		SP-4060 MGSIP PROG-L Issue 1 or later
Software-based MG-SIP	Proprietary Protocol	SP-4080 MGSIPVM PROG-D Issue 1 or later
		SP-4078 MGSIPVM PROG-B Issue 1 or later
		SP-4085 MGSIPVM PROG-E Issue 1 or later
VS32		Not supported
IPG Digital		Not supported
IPG Analog		Not supported
UG50 (DLC)		UG50 software version 2.00 or later
UG50 (LC)	Proprietary Protocol	UG50 software version 2.00 or later
	SIP	UG50 software version 2.00 or later
UG50 (PRT 1.5M)	Proprietary Protocol	UG50 software version 2.00 or later
	SIP	UG50 software version 2.00 or later
UG50 (PRT 2M)	Proprietary Protocol	UG50 software version 2.00 or later
	SIP	UG50 software version 2.00 or later
UG50 (COT-TYPE1)		UG50 software version 2.00 or later
UG50 (COT-TYPE2)	Proprietary Protocol	UG50 software version 2.00 or later
	SIP	UG50 software version 2.00 or later
UG50 (PGT-TYPE1)		Not supported
UG50 (PGT-TYPE2)	Proprietary Protocol	Not supported
	SIP	Not supported
UG50-IPG (Digital)		Not supported

IP Station/Equipment		Firmware Version
UG50-IPG (Analog)		Not supported
UG50-2MC		Available since the first version
UG50-8LC		Available since the first version
UG50-24PRIA	Proprietary Protocol	Available since the first version
	SIP	Available since the first version
UG50-30PRIA	Proprietary Protocol	Available since the first version
	SIP	Available since the first version
UG50-4LC2COTA (COT)		Available since the first version
UG50-4LC2COTA (PGT)		Not supported
UG50-6COT	Proprietary Protocol	Available since the first version
	SIP	Available since the first version

<When MAC Authentication Mode is used>

All the Nodes that are assigned by AUPDN (for Stations)/ALRTN (for Routes) must be set as destination IP addresses from each terminal (including obtaining IP address from DHCP server).

<When User Authentication (Login Mode) is used>

Login can be possible if any node assigned by office data has been registered to terminal. Terminals perform registration in the order of the setting on each terminal because they cannot obtain the destination IP address order information from the destination node. However, if a Node that is assigned as “Backup (Secondary or Tertiary)” in AUPDN is set as “Primary” in DRS setting of terminals, Visitor lamps on the terminals will light green because the terminals register to the Backup Node according to the assignment of AUPDN. Conversely, if a Node that is assigned as “Primary” in AUPDN is set as “Backup” in DRS setting of the terminals, the Visitor lamps on the terminals will not light because the terminals register to the Primary Node according to the assignment of AUPDN.

5. The following table shows FCCS Cluster operation when a failure occurs at the Primary Node where the following application ports are set to LAN1; DRS(REG), Internal PHE, Internal PHI-BRI, SR-MGC/NMS, Internal PHF, Internal PHI-PRI, and FCCS Cluster.

FCCS Cluster Operation - LAN1

Equipment supporting FCCS Cluster	Failure/Recovery Pattern	Operation at Failure/Recovery
<For Internal PHE> • Analog 2MC • MC&MG-COT (MC part) • UG50 (LC) [Proprietary Protocol] (**Note 10**) • UG50-IPG (Analog) • UG50-2MC • UG50-4LC2COTA (COT) (MC part)	LAN1 Failed	To be registered to backup node.
	LAN1 Recovered	To be registered to Primary Node.
	LAN2 Failed	To be registered to Primary Node. (No status change)
	LAN2 Recovered	To be registered to Primary Node. (No status change)
<For SP-PHD (SIP mode) > • SIP Multiple Line Terminal • Soft Phone [SIP]	LAN1 Failed	To be registered to backup node.
	LAN1 Recovered	To be registered to Primary Node.
	LAN2 Failed	To be registered to Primary Node. (No status change)
	LAN2 Recovered	To be registered to Primary Node. (No status change)

FCCS Cluster Operation - LAN1

Equipment supporting FCCS Cluster	Failure/Recovery Pattern	Operation at Failure/Recovery
<For SP-PHI (SIP mode) > • MG(BRI) [SIP] • MG(PRI) [SIP] • MG(6COT) [SIP] • 8LC • UG50 (PRT 1.5M)[SIP](**Note 10**) • UG50 (PRT 2M) [SIP](**Note 10**) • UG50-24PRIA [SIP] • UG50-30PRIA [SIP] • UG50-8LC • UG50-6COT [SIP] • UG50 (LC)[SIP](**Note 10**) • UG50(COT-TYPE2)[SIP](**Note 10**)	LAN1 Failed	To be registered to backup node.
	LAN1 Recovered	To be registered to Primary Node.
	LAN2 Failed	To be registered to Primary Node. (No status change)
	LAN2 Recovered	To be registered to Primary Node. (No status change)
<For SP-PHD (Proprietary Protocol mode) > • IP Terminal • Soft Phone [Proprietary Protocol] • UG50-IPG (Digital) • UG50 (DLC) (**Note 10**)	LAN1 Failed	To be registered to backup node.
	LAN1 Recovered	To be registered to Primary Node.
	LAN2 Failed	To be registered to Primary Node. (No status change)
	LAN2 Recovered	To be registered to Primary Node. (No status change)
<For SP-PHI (Proprietary Protocol mode) > • MC&MG(COT) [MG part] • MG(BRI) [Proprietary Protocol] • MG(6COT) [Proprietary Protocol] • UG50(COT-TYPE1/TYPE2)[Proprietary Protocol]) (**Note 10**) • UG50-4LC2COTA (COT) (MG part)	LAN1 Failed	To be registered to backup node.
	LAN1 Recovered	To be registered to Primary Node.
	LAN2 Failed	To be registered to Primary Node. (No status change)
	LAN2 Recovered	To be registered to Primary Node. (No status change)

FCCS Cluster Operation - LAN1

Equipment supporting FCCS Cluster	Failure/Recovery Pattern	Operation at Failure/Recovery
<For Internal PHI-BRI (Proprietary Protocol mode) > • MC&MG(COT) [MG part] • MG(BRI) [Proprietary Protocol] • MG(6COT) [Proprietary Protocol] • MG(SIP) • UG50(COT-TYPE1/TYPE2) [Proprietary Protocol] (**Note 10**) • UG50-4LC2COTA (COT) (MG part) • UG50-6COT [Proprietary Protocol]	LAN1 Failed	To be registered to backup node.
	LAN1 Recovered	To be registered to Primary Node.
	LAN2 Failed	To be registered to Primary Node. (No status change)
	LAN2 Recovered	To be registered to Primary Node. (No status change)
<For Internal PHI -PRI (Proprietary Protocol mode) > • MG(PRI) [Proprietary Protocol] • UG50 (PRT 1.5M)[Proprietary Protocol] (**Note 10**) • UG50 (PRT 2M)[Proprietary Protocol] (**Note 10**) • UG50-24PRIA[Proprietary Protocol] • UG50-30PRIA [Proprietary Protocol](**Note 11**)	LAN1 Failed	To be registered to backup node.
	LAN1 Recovered	To be registered to Primary Node.
	LAN2 Failed	To be registered to Primary Node. (No status change)
	LAN2 Recovered	To be registered to Primary Node. (No status change)

6. The following table shows FCCS Cluster operation when a failure occurs at the Primary Node where the following application ports are set to LAN2; DRS(REG), Internal PHE, Internal PHI-BRI, SR-MGC/NMS, Internal PHF, Internal PHI-PRI, and FCCS Cluster.

FCCS Cluster Operation - LAN2

Equipment supporting FCCS Cluster	Failure/Recovery Pattern	Operation at Failure/Recovery
<For Internal PHE> • Analog 2MC/MC&MG-COT (MC part) • UG50 (LC) [Proprietary Protocol] (**Note 10**) • UG50-IPG (Analog) • UG50-2MC • UG50-4LC2COTA (COT) (MC part)	LAN1 Failed	To be registered to Primary Node. (No status change) **Note 9**
	LAN1 Recovered	To be registered to Primary Node.
	LAN2 Failed	To be registered to backup node.
	LAN2 Recovered	To be registered to Primary Node.
<For SP-PHD (SIP mode) > • SIP Multiple Line Terminal • Soft Phone [SIP]	LAN1 Failed	To be registered to backup node.
	LAN1 Recovered	To be registered to Primary Node.
	LAN2 Failed	To be registered to backup node.
	LAN2 Recovered	To be registered to Primary Node.
<For SP-PHI (SIP mode) > • MG(BRI) [SIP] • MG(PRI) [SIP] • MG(6COT) [SIP] • 8LC • UG50 (PRT 1.5M)[SIP] (**Note 10**) • UG50 (PRT 2M)[SIP] (**Note 10**) • UG50-24PRIA [SIP]/UG50-30PRIA [SIP] • UG50-8LC • UG50-6COT [SIP] • UG50 (LC) [SIP] (**Note 10**)	LAN1 Failed	To be registered to backup node.
	LAN1 Recovered	To be registered to Primary Node.
	LAN2 Failed	To be registered to backup node.
	LAN2 Recovered	To be registered to Primary Node.

FCCS Cluster Operation - LAN2

Equipment supporting FCCS Cluster	Failure/Recovery Pattern	Operation at Failure/Recovery
<For SP-PHD (Proprietary Protocol mode) > • IP Terminal • Soft Phone [Proprietary Protocol] • UG50-IPG (Digital) • UG50 (DLC) (**Note 10**)	LAN1 Failed	To be registered to backup node.
	LAN1 Recovered	To be registered to Primary Node.
	LAN2 Failed	To be registered to backup node.
	LAN2 Recovered	To be registered to Primary Node.
<For SP-PHI (Proprietary Protocol mode) > • MC&MG(COT) [MG part] • MG(BRI) [Proprietary Protocol] • MG(6COT) [Proprietary Protocol] • UG50(COT-TYPE1/TYPE2) [Proprietary Protocol]) (**Note 10**) • UG50-4LC2COTA (COT) (MG part)	LAN1 Failed	To be registered to backup node.
	LAN1 Recovered	To be registered to Primary Node.
	LAN2 Failed	To be registered to backup node.
	LAN2 Recovered	To be registered to Primary Node.
<For Internal PHI-BRI (Proprietary Protocol mode) > • MC&MG(COT) [MG part] • MG(BRI) [Proprietary Protocol] • MG(6COT) [Proprietary Protocol] • MG(SIP) • UG50(COT-TYPE1/TYPE2)[Proprietary Protocol] (**Note 10**) • UG50-4LC2COTA (COT) (MG part) • UG50-6COT [Proprietary Protocol]	LAN1 Failed	To be registered to Primary Node. (No status change) **Note 9**
	LAN1 Recovered	To be registered to Primary Node.
	LAN2 Failed	To be registered to backup node.
	LAN2 Recovered	To be registered to Primary Node.

FCCS Cluster Operation - LAN2

Equipment supporting FCCS Cluster	Failure/Recovery Pattern	Operation at Failure/Recovery
<For Internal PHI -PRI (Proprietary Protocol mode) > • MG(PRI) [Proprietary Protocol] • UG50 (PRT 1.5M)[Proprietary Protocol] (**Note 10**) • UG50 (PRT 2M)[Proprietary Protocol] (**Note 10**) • UG50-24PRIA[Proprietary Protocol] • UG50-30PRIA[Proprietary Protocol] (**Note 11**)	LAN1 Failed	To be registered to Primary Node. (No status change) **Note 9**
	LAN1 Recovered	To be registered to Primary Node. (No status change)
	LAN2 Failed	To be registered to backup node.
	LAN2 Recovered	To be registered to Primary Node.

Note 9: When FCCS Cluster and SR-MGC coexist, which means that the office data for SR-MGC (the CMNT command) and SP Failure Detection valid (ASYDL, SYS1, Index 843, Bit 0=0, Bit 1=0) are assigned in Primary Node/backup node, the operation will change; to be registered to backup node.

Note 10: Available from FP95 V4-114 Issue 2.0.

7. For using this feature with SIP Multiple Line terminal that requires SIP terminal authentication, be sure to register to Primary Node at first. Otherwise, registration to backup node will not be available.
8. Ping signal needs to go through the Primary Node and the terminal when performing this feature. When using the Softphone on Windows XP SP2 with Windows Firewall enabled, the Ping return must be allowed by adding a check into the [Allow incoming echo request] checkbox by accessing [Control Panel] → [Windows Firewall] → [Advanced] tab → [ICMP] field.
9. For Location ID assignment, the same data has to be assigned to all nodes.
10. Master Node IP Address or Sub Master Node IP Address has to be assigned to each node by using the ACFMN command (Node Date Setting screen) before using this feature. Refer to the Command Manual for further details.

Note 11: UG50-24PRIA [Proprietary Protocol], UG50-30PRIA [Proprietary Protocol], UG50(PRT 1.5M) [Proprietary Protocol] and UG50(PRT 2M) [Proprietary Protocol] set the IP address of LAN1 as a registration destination. However, LAN2 is used for transmitting a call control signal after registration.

Note: To change Master Node IP address or Sub Master Node IP address, follow the procedure below. If you do not follow the procedure below, FCCS Cluster information of Backup Station/Route may be deleted and it may become impossible to make/receive a call.

(1) Make sure that no terminal registers to Backup Node.
(2) Assign ASYDN, SYS1, Index 963, Bit 1=0.
(3) Change Master Node IP address and Sub Master Node IP address.
(4) Assign ASYDN, SYS1, Index 963, Bit 1=1.

11. Route Sharing is not supported.
12. A total number of Primary Stations and Backup Stations cannot exceeds the amount of the system capacity license within each node. The maximum number of Stations and MGs (in total) using this feature is 32,000 within FCCS Network.
13. When using this feature, it is recommended to operate terminals in MAC Authentication mode. MAC data can be assigned to NDM from AMACN (directly) or each terminal (the password used for assigning MAC data to NDM from each terminal needs to be assigned on the DtermIP MAC Address Registration Password (NDM) screen of AFCMN).

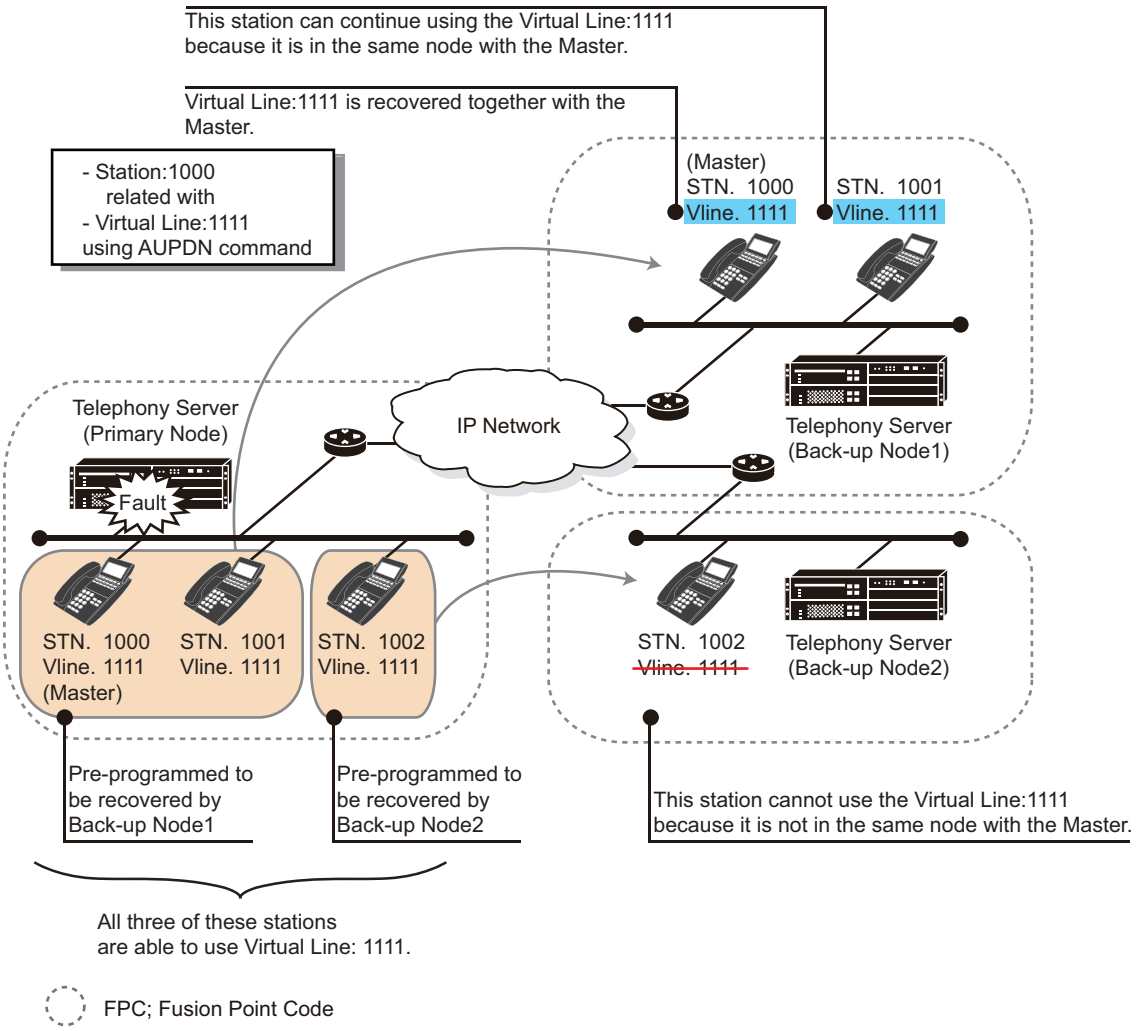
    Note: MAC Authentication mode is not available for Soft Phone. Thus, Soft Phone does not support the encryption when using this feature.

14. If specifying "After confirmation of connection establishment with the Home Node" or "After station data synchronization is completed" on the Home Node monitoring interval screen of AFCMN command, Soft Phone does not register to Primary Node automatically because Soft Phone always operates in Login Mode. By pressing the VISITOR key, Soft Phone is reset and registers to Primary Node.

    - Virtual line moves together with the Master Station preassigned by AUPDN command. The call origination/termination from/to the virtual line accommodated within the IP station, is available only if Master station is accommodated under the same node. When the Master station and the virtual line are accommodated in different nodes, virtual line key of the terminal within other node will turn red to indicate that the call origination/termination from/to that line is restricted.

      Note: Be sure to assign the specified virtual line to a Feature key of the Master Station.

      Note: When the node previously registered is isolated (before the Master Station and virtual line move), in a situation which the Master node and Sub master node can not communicate, virtual line is only available within the isolated node (because the exchange of FCCS Cluster information is not available between the nodes). Button will be lit in red and call origination/termination will be restricted until Master node and Sub master node can communicate.



15. When the terminal within the station hunting group was recovered by the Backup Node, call hunting is still available if the call hunting group data was assigned toward the NDM. Call hunting data will be skipped if the data was assigned toward the Local Node (DM or LDM).
16. Boss-Secretary features are available only between terminals accommodated within the same node.
17. When the system message includes the information of physical number or the LENS, physical number and LENS information will be output with the information of Backup Node (The node which terminal has performed the registration).
18. As for Name Display feature, the existing data (ANDD) is displayed. Therefore, if you want to display the name display newly assigned with the user profile (AUPDN), it is essential to delete the existing data for name display first.
19. The billing information of terminal operating under the Backup Node is generated and output from the Backup Node.
20. MA-ID must be assigned individually towards the physical station and physical route of each node.
21. If Login is to be performed with MAC Authentication password assigned by the AFCMN command, the MAC address of the registering Station will be programmed into NDM. Assigned data can be confirmed by using the AMACN command.

22. When assigning the key data by using the AUPKDN command or key template of AKTDN, Multiple Line can be assigned over FCCS, but call origination/termination is available only when the Multiple Line is accommodated within the same node to which the terminal registered. If the terminal registered to other node, the Multiple Line key will be lit in red.

    The following explains the terminal status depending on the timing of Registration in "**Primary Node <-> Backup Node**".

    - Terminal Actions when terminals are registered to Backup Node from Primary Node

Idle	Speeching	Calling (Originating Call)	Called (Arrived Call)
Registered in Backup Node	Speeching Status continues. a calling is terminated by On-Hook, the terminal reset itself and register to Backup Node.	Calling Status (Originating Call) continues. A few minutes later, the terminal resets itself and register to Backup Node.	Called Status (Arrived Call) continues. A few minutes later, the terminal resets itself and register to Backup Node.

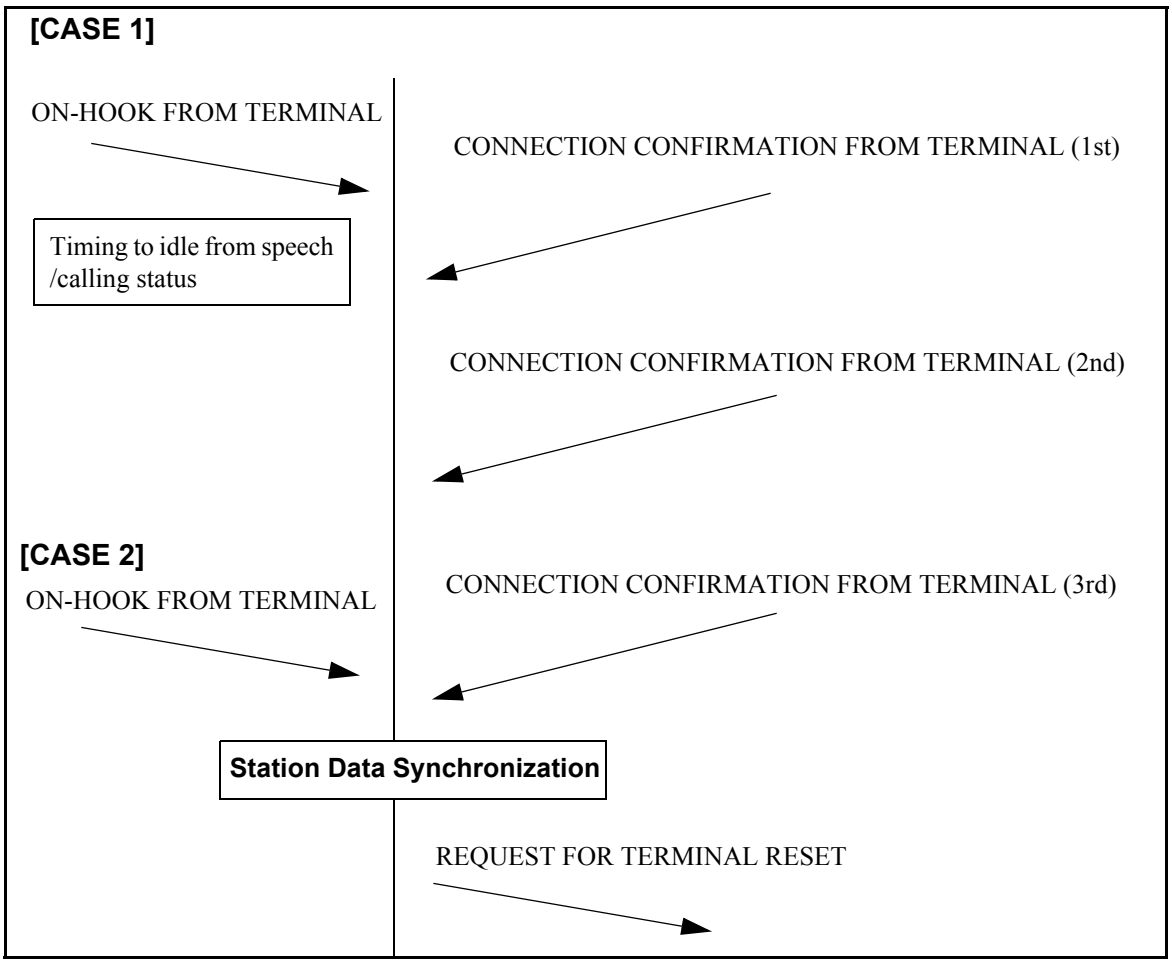
23. Settings for LOGGED OUT IP STATION - CALL DESTINATION [L-53] will be regulated while in Backup Node. In addition, if Primary Node fails, LOGGED OUT IP STATION - CALL DESTINATION [L-53] will not be available. LOGGED OUT IP STATION - CALL DESTINATION [L-53] will be available after Primary Node recovers from failure.
24. CALL FORWARDING-ALL CALLS [C-5] is not available if Primary Node fails and the terminal is not registered to a Backup Node. CALL FORWARDING-ALL CALLS [C-5] will be available after Primary Node recovers from failure or if the terminal is in registered status to a Backup Node.

- Terminal Actions when terminals are registered to Primary Node from Backup Node

Timing	Idle	Speeching	Calling (Originating Call)	Called (Arrived Call)	Remarks
Immediate	After connection confirmation (**Note 12**) is successfully completed, the terminal is registered to Primary Node by Terminal Reset.	Speeching Status continues. After calling is terminated (On-Hook) and connection confirmation (**Note 12**) is successfully completed, the terminal is registered to Primary Node by Terminal Reset.	Calling Status (Originating Call) continues. After calling is terminated (On-Hook) and connection confirmation (**Note 12**) is successfully completed, the terminal is registered to Primary Node by Terminal Reset.	Called Status (Arrived Call) continues. The status is changed into idle. After connection confirmation (**Note 12**) is successfully completed, the terminal is registered to Primary Node by Terminal Reset.	In the case of Media Gateways (Trunks), when all the channels are idle, the terminal is registered to Primary Node.
Specified-Time	Same as above				
Press VISITOR key on Terminal	Terminal is registered to Primary Node by Terminal Reset.	VISITOR key is not available during the speech. After the speech is terminated by On-Hook, it becomes available.	VISITOR key is not available. After the calling is terminated by On-Hook, it becomes available.	Terminal is registered to Primary Node by Terminal Reset. The calling status is changed to idle. (Calling process in Called Party side continues. A few minutes later, the line will be terminated.)	
Send Terminal Reset by using the CBKS/DISD Command	Terminal is registered to Primary Node by Terminal Reset.	Terminal is registered to Primary Node by Terminal Reset. The line will be disconnected. (Called Party is on line. A few minutes later, the line will be disconnected.)	Terminal is registered to Primary Node by Terminal Reset. The calling status is changed to idle. (Calling process in Called Party side continues. A few minutes later, the line will be disconnected.)	Terminal is registered to Primary Node by Terminal Reset. The calling status is changed to idle. (Calling process in Called Party side continues. A few minutes later, the line will be disconnected.)	

Note 12: When terminals registered in Backup Nodes exist in networks, the Backup Nodes perform "**Connection Confirmation**" to the appropriate terminals via their Primary Nodes. When the Primary Nodes are recovered, if connection is confirmed as alive in three consecutive times, data synchronization among Stations will be performed and then Terminal Reset will be transmitted to each terminal.

Note: An interval to perform Connection Confirmation
When you assign "60 minutes" for Home Node monitoring interval by the AFCMN command, a period between line disconnection to Terminal Reset sending will be approximately five minutes.
The following diagram shows the Connection Confirmation processing.



- When Station Data Synchronization ends in failure
Even though Station Data Synchronization ends in failure, if it later succeeds for three consecutive times, Terminal Reset signal will be sent to reset a terminal. A terminal receiving its signal will be reset and registered to the Primary Node. In this case, Station Data in Backup Node side will be discarded.

25. When this feature is performed with Analog MC, all the MC ports require User Profile Data assignment by the AUPDN command. If only a part of the MC channels is assigned, the rest (unregistered MC channels) will become unavailable. When this feature is not performed with Analog MC, data assignment by the AUPDN command is not required.
When IP Station Data have been assigned by the AISTL command, registration towards the nodes and use of registered MC channels become available. When this feature is performed with Analog MC, refer to "Sample of Data Assignment" in the FCCS Location Diversity part of the Data Programming Manual - FCCS.

    Note: When you assign the data for Analog MC by the AUPDN command and the AISTL command data has been assigned at the Home node, delete the AISTL data before assigning the AUPDN data.

26. When this feature is performed with MC&MG-COT, all MG trunks need to be registered by AMGIL command even though the trunks will not be used.
27. When you use the CBKS command to reset an MG(PRI), consider the following:

    When a check box for a B-Channel route is selected and the RESET button is clicked, the CBKS command automatically resets the selected B-Channel and the corresponding D-Channel routes, and vice versa.
    For this reason, if you select the check boxes for both B-Channel and D-Channel routes simultaneously and click the RESET button, "SP H/I Error" is returned for the channel that is processed later than the other. In that case, run the command again to see if the reset of both channels has been completed.

28. While IP station A is setting [C-1] Call Back on IP station B, if a changeover or switchback is executed for nodes, the operation will be as follows.

    <In the case of Changeover to Backup Node>

    The Call Back both on the IP station A and B will be canceled, and the node will be changed over to backup node.

    <In the case of Switchback to Primary Node>

    After the Primary Node is confirmed released, if the IP station B is being talking, the Call Back of IP station A will be canceled, and the node will be changed over to the Primary Node.

    When the call of IP station B is released after the Primary Node is confirmed released, if the Call Back is still being set on the IP station B, the node will be switchbacked to the Primary Node after executing the Call Back.

29. To know more details about the relationship between this feature and other features, refer to Interactions in this feature.
30. When IP Station Data have been assigned by the AISTL command, registration towards the nodes and use of registered MC channels become available. When this feature is performed with Analog MC, refer to “Sample of Data Assignment” in the FCCS Location Diversity part of the Data Programming Manual - FCCS. To know more details about the relationship between this feature and other features, refer to Interactions in this feature.
31. In system configuration that includes non-FCCS Cluster support terminals, SR-MGC is unable to replace the Backup Node in rescuing non-FCCS Cluster support terminals.
32. Even if the controlling station of Uniform Call Distribution [U-1] has been registered in backup node, the station number is available as the UCD pilot number. However, Call Waiting Lamp - UCD [C-76] and Delay Announcement -UCD [D-31] is unavailable at this time.
33. This feature is not supported in the UMGi/RNIP system.

Programming (FCCS Cluster)

For data programming of this feature, refer to the programming part of FCCS Cluster in the Data Programming Manual - FCCS.

SR-MGC with FCCS Cluster

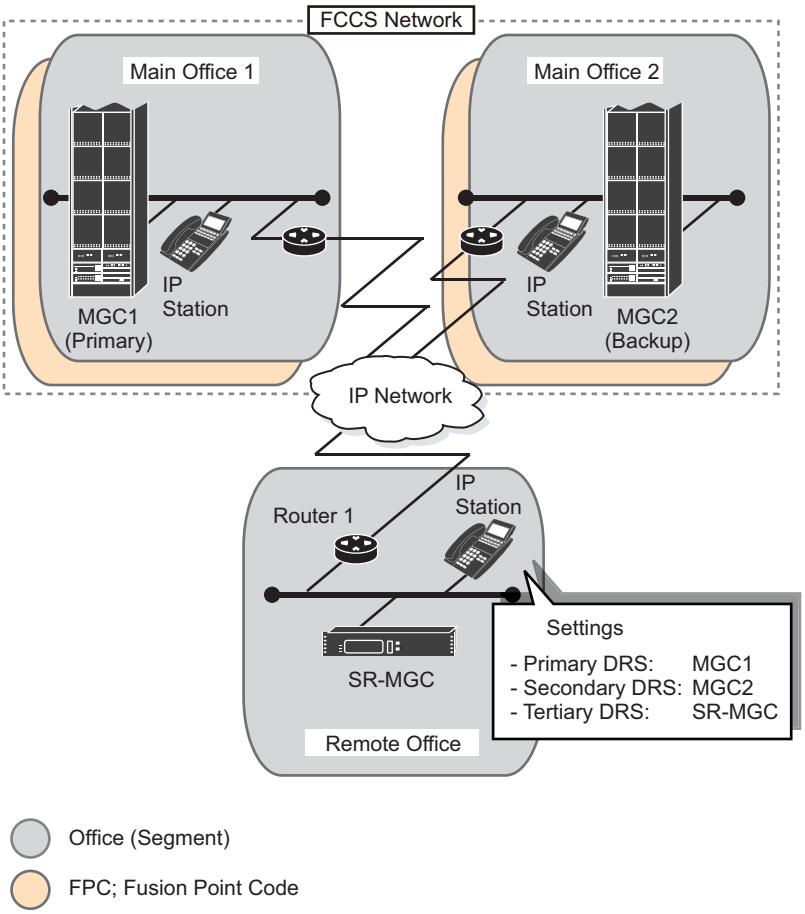
General Description (SR-MGC with FCCS Cluster)

This feature is available in cooperation with SR-MGC, which is called “SR-MGC with FCCS Cluster”. When network failure occurs and causes connection failure between terminals and MGC (**Note 13**), in addition, if keeping terminals' connection with the FCCS Cluster feature is not available, SR-MGC will be activated.

For example, in the next figure, if Router 1 in Remote Office 1 is down, SR-MGC will have trouble communicating with MGC1 (Primary) and MGC2 (Backup). And then it soon realizes the managed-terminals' connections are not able to be kept by other MGCs, and the SR-MGC activates itself to keeping those connections alive.

The hardware installation and data assignment for the SR-MGC are described in the Peripheral Equipment Description. This section describes only Service Conditions specific to SR-MGC with FCCS Cluster.

Note 13: “MGC” used in this section indicates the Telephony Server.



When the SR-MGC is used in the network and a failure occurs, the registration of servers and terminals is changed as follows:

CAUSE	TERMINAL STATUS
MGC1 has network failure	Terminals in Remote Office 1 are registered to MGC2.
MGC1 and MGC2 have network failure	Terminals in Remote Office 1 are registered to SR-MGC.

Operating Procedure (SR-MGC with FCCS Cluster)

No manual operation is required.

Service Conditions (SR-MGC with FCCS Cluster)

The following are specific to SR-MGC with FCCS Cluster feature. For more information about the SR-MGC's Service Conditions including a list of IP devices/stations available for this feature, refer to Service Conditions of Survivable Remote MGC [S-141].

1. SR-MGC control (such as initializing, switching operating modes, office data copy, and automatic changeover) can be performed by using the CMNT command only from Primary MGC.
2. Terminal can be switchbacked from SR-MGC by assigning ASYDL, SYS1, Index 852, Bit 5: 0/1 = Manual/Automatic switchback. **Note 14**

Note 14: The SWITCHBACK means returning the terminal registration destination to the Telephony Server when the communication failure between the Telephony Server and SR-MGC is released.

- The switchback process in the case of automatic switchback [ASYDL, SYS1, Index 852, Bit 5=1].

  As soon as communication recovery between the Telephony Server and SR-MGC is detected on SR-MGC side, the terminal registered on SR-MGC will automatically be switchbacked to the Telephony Server at primary or backup node.

  [Conditions for Automatic Switchback]

  a.) When "1" is selected in Mode of the CMNT command:
  The switchback will be executed for the Telephony Server at primary or backup node that are released first.
  b.) When "2" is selected in Mode of the CMNT command:
  When SR-MGC detects communication recovery with the Telephony Server at Primary Node, and also communication status between the Telephony Server at Primary Node and pilot station is confirmed normal, the terminal registered on SR-MGC will automatically be switchbacked to the Telephony Server at Primary Node. If the Telephony Server at backup node is released first, the switchback will not be executed.

[Ongoing call check when the automatic switchback is executed]

a.) SR-MGC checks the ongoing call status and switchbacks the terminal not in busy status to the Telephony Server in turn.
b.) If the terminal has several channels, the terminal will forcibly be switchbacked to the Telephony Server when all the channels are released.

The switchback will be executed for the Telephony Server at primary or backup node that are released first.

- The switchback process in the case of manual switchback [ASYDL, SYS1, Index 852, Bit 5=0].

When the communication failure between the Telephony Server at Primary Node and SR-MGC is released, the terminal registered on SR-MGC can be switchbacked to the Telephony Server at Primary Node by executing CONDITION (To Standby mode with terminal reset) in [Change Operating] tab of CMNT command. If the Telephony Server at backup node is released first, the manual switchback cannot be executed.

3. Network Time information can be obtained by Primary MGC only.
4. The state of SR-MGC changes to Operating Mode when:

- Primary Node and all of the Backup nodes have trouble communicating with its SR-MGC.
- A fault occurs in only one direction (To Operating mode). **Note 15**

Note 15: To enable the connectivity check (To Operating mode), the following system data setting is required.
- Identical value must be assigned in all the node. Assign 30 seconds or more as a value of Health Check cycle (ASYDL, SYS1, Index 850) and 5 seconds or more as a value of Health Check waiting time (ASYDL, SYS1, Index 851).
  Health Check cycle > Health Check waiting time × 2
- When changing the data for Health Check cycle (ASYDL, SYS1, Index 850), it should be changed according to the procedure explained in the note on ASYDL, SYS1, Index 850 in Command Manual.

5. Firmware Version of each terminal that can be used in an environment where FCCS Cluster and SR-MGC coexist is as follows.

IP Station/Equipment		Firmware Version
DT800/DT700 Series	SIP	All versions
DtermIP/IP Enabled Dterm	Proprietary Protocol	Version 2.00 or later
DtermIP (SIP)	SIP	Version 2.0.4.0 or later
Soft Client SP350	SIP	All versions
DtermSP30	Proprietary Protocol	All versions
	SIP	Version 7 or later
Analog 2MC		SP-3753 2AMC PROG-B Issue 8 or later
8LC Card [SCA-8LCA/SCA-8LCA-EMEA]		SP-3893 8MC PROG-A Issue 1
8LC Card [SCA-8LCC/SCA-8LCC-EMEA]		SP-3983 8MC PROG-B Issue 1
MCMG Card [SCA-4LC2COTA]		SP-3927 MCMG PROG-B Issue 1 or later
MCMG Box [MG-4LC2COTA]		
MG-COT [SCA-6COTA]		SP-3875 6MGCOT PROG-A Issue 1 or later
MG-COT [SCA-6COTB]	Proprietary Protocol	SP-3948 6MGCOT PROG-A Issue 1 or later
	SIP	SP-3948 6MGCOT PROG-A Issue 1 or later
MG-COT [SCA-6COTC]	Proprietary Protocol	SP-3948 6MGCOT PROG-A Issue 1 or later
	SIP	SP-3948 6MGCOT PROG-A Issue 1 or later
MG(BRI) Card [SCA-2BRIA]	SIP	SP-3826 BRI MG SIP BSC PROG-A Issue 1 or later
MG(BRI) Box [MG-2BRIA]		
MG-PRI/MG-PRA	Proprietary Protocol	SW-423 PRI MG PROGRAM-A Issue 1 or later
MG(PRI) Card [SCA-24PRIA]	Proprietary Protocol	All versions (SP-3884 MG PRI PROG-A Issue 1 or later)
	SIP	SP-3884 MG PRI PROG-A Issue 1 or later
MG(PRI) Box [MG-24PRIA]	Proprietary Protocol	All versions (SP-3884 MG PRI PROG-A Issue 1 or later)
	SIP	SP-3884 MG PRI PROG-A Issue 1 or later
MG(PRI) Card [SCA-30PRIA]	Proprietary Protocol	All versions (SP-3884 MG PRI PROG-A Issue 1 or later)
	SIP	SP-3884 MG PRI PROG-A Issue 1 or later
MG(PRI) Box [MG-30PRIA]	Proprietary Protocol	All versions (SP-3884 MG PRI PROG-A Issue 1 or later)
	SIP	SP-3884 MG PRI PROG-A Issue 1 or later
MG (SIP) 16	Proprietary Protocol	SP-3843 SIPMG PROG-B Issue 5 or later
MG (SIP) 16 [SCA-16SIP-MGA]	Proprietary Protocol	SP-3905 SIPMG (16) PROG-A Issue 2 or later
MG (SIP) 16 [MG-16SIP-MGA]		

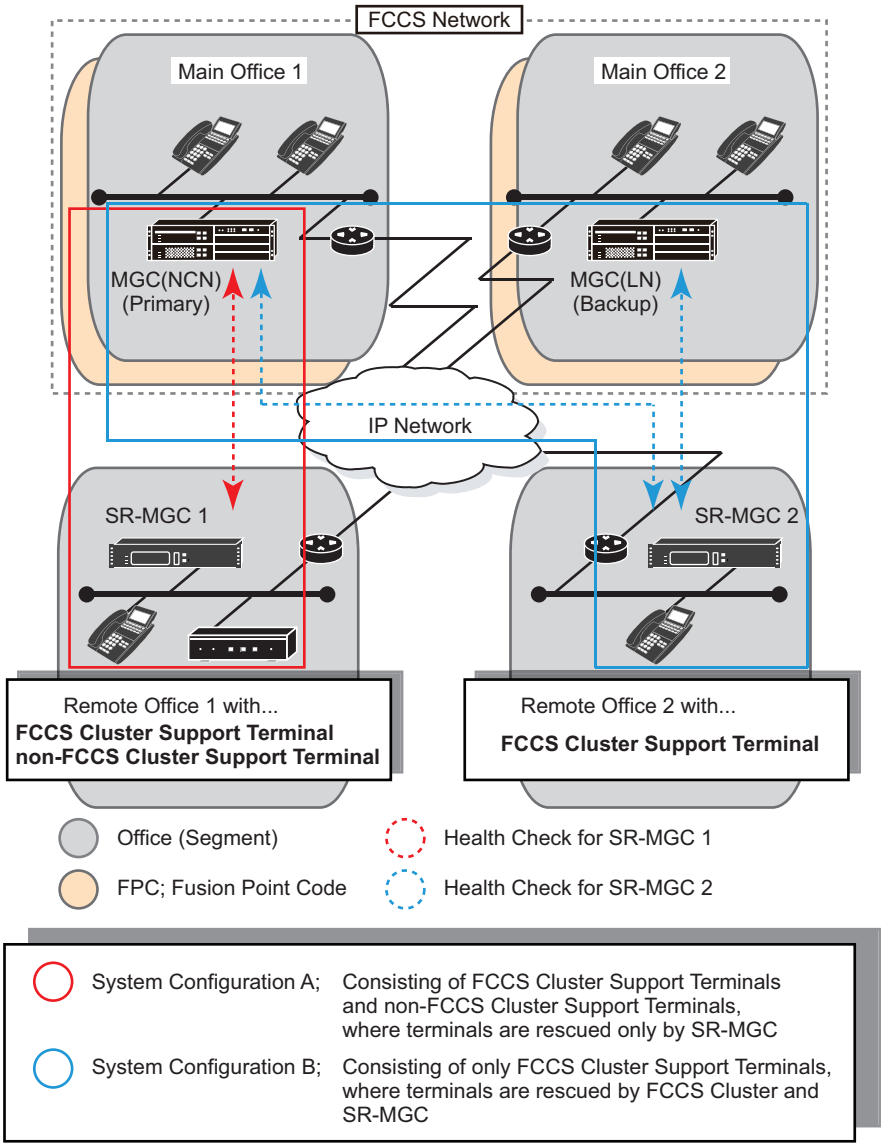
IP Station/Equipment		Firmware Version
MG (SIP) 16 [SCA-16SIPMG(US)]	Proprietary Protocol	SP-3988 MGSIP (16) PROG-G Issue 1 or later
MG-SIP96	Proprietary Protocol	SP-3987 MGSIP PROG-B Issue 1 or later
MG-SIP128 [MG-128SIP-MGL-A]	Proprietary Protocol	SP-4089 MGSIP(MM) PROG-A Issue 1 or later
MG-SIP128 [MG-128SIPMGG-B/MG-128SIPMGG]	Proprietary Protocol	SP-4058 MGSIP PROG-J Issue 1 or later
		SP-4051 MGSIP PROG-G Issue 1 or later
MG-SIP128 [MG-128SIPMGJ-B/MG-128SIPMGJ]	Proprietary Protocol	SP-4032 MGSIP PROG-E Issue 1 or later
		SP-4060 MGSIP PROG-L Issue 1 or later
Software-based MG-SIP	Proprietary Protocol	SP-4080 MGSIPVM PROG-D Issue 1 or later
		SP-4078 MGSIPVM PROG-B Issue 1 or later
		SP-4085 MGSIPVM PROG-E Issue 1 or later
VS32	Proprietary Protocol	Not supported
	SIP	Not supported
UG50 (DLC)		UG50 software version 2.00 or later
UG50 (LC) [Proprietary Protocol]		UG50 software version 2.00 or later
UG50 (LC) [SIP]		UG50 software version 2.00 or later
UG50 (PRT 1.5M) [Proprietary Protocol]		UG50 software version 2.00 or later
UG50 (PRT 1.5M) [SIP]		UG50 software version 2.00 or later
UG50 (PRT 2M) [Proprietary Protocol]		UG50 software version 2.00 or later
UG50 (PRT 2M) [SIP]		UG50 software version 2.00 or later
UG50 (COT-TYPE1)		UG50 software version 2.00 or later
UG50 (COT-TYPE2) [Proprietary Protocol]		UG50 software version 2.00 or later
UG50 (COT-TYPE2) [SIP]		UG50 software version 2.00 or later
UG50 (PGT-TYPE1)		Not supported
UG50 (PGT-TYPE2) [Proprietary Protocol]		Not supported
UG50 (PGT-TYPE2) [SIP]		Not supported
UG50-IPG (Digital)		Not supported
UG50-IPG (Analog)		Not supported
UG50-2MC		Available since the first version
UG50-8LC		Available since the first version
UG50-24PRIA [Proprietary Protocol]		Available since the first version
UG50-24PRIA [SIP]		Available since the first version

IP Station/Equipment	Firmware Version
UG50-30PRIA [Proprietary Protocol]	Available since the first version
UG50-30PRIA [SIP]	Available since the first version
UG50-6COT [Proprietary Protocol]	Available since the first version
UG50-6COT [SIP]	Available since the first version
UG50-4LC2COTA (COT)	Available since the first version
UG50-4LC2COTA (PGT)	Not supported

6. When using SR-MGC with FCCS Cluster, the Primary Node and backup node are used as one system. Maximum of 256 SR-MGC can be accommodated to one system.
7. In environment where FCCS Cluster and SR-MGC coexist, SR-MGC is capable of rescuing equipment that supports FCCS Cluster.
8. When using FCCS Cluster with SR-MGC, if FCCS Cluster support terminals and non-FCCS Cluster support terminals coexist, a single SR-MGC is not capable of rescuing both types of terminals.
   If both types of terminals coexist, when building a FCCS Cluster/SR-MGC system, precautions are needed for operation. Depending on operation conditions, a non-FCCS Cluster support terminal may not be rescued by either FCCS Cluster or SR-MGC.
   See the following for the operation conditions in this environment.

[When Non-FCCS Cluster Support Terminals Coexist With FCCS Cluster support Terminals]

When using FCCS Cluster with SR-MGC, 2 types of system configurations must coexist, in regard to the different types of terminals (FCCS Cluster support terminal/non-FCCS Cluster support terminal, SR-MGC is capable of rescue /SR-MGC is incapable of rescue).



- System Configuration A: System rescue is performed only by SR-MGC
  When failure occurs in Primary Node, SR-MGC 1 rescues FCCS Cluster support terminals and non-FCCS Cluster support terminals.

  a.) System is normal: Primary Node (MGC(NCN)) manages the terminals.

  b.) Failure in Primary Node: SR-MGC 1 rescues FCCS Cluster support terminals and non-FCCS Cluster support terminals.

  c.) Failure in Primary & Backup Node: Terminals are already rescued by SR-MGC 1.

  *Health Check is provided only between SR-MGC 1 and MGC(NCN).

- System Configuration B: System rescue is performed by FCCS Cluster and SR-MGC
  When failure occurs in Primary Node and Backup Node, SR-MGC 2 rescues FCCS Cluster support terminals.

  a.) System is normal: Primary Node (MGC(NCN)) manages the terminals.

  b.) Failure in Primary Node: Backup Node (MGC(LN)) rescues FCCS Cluster support terminals.

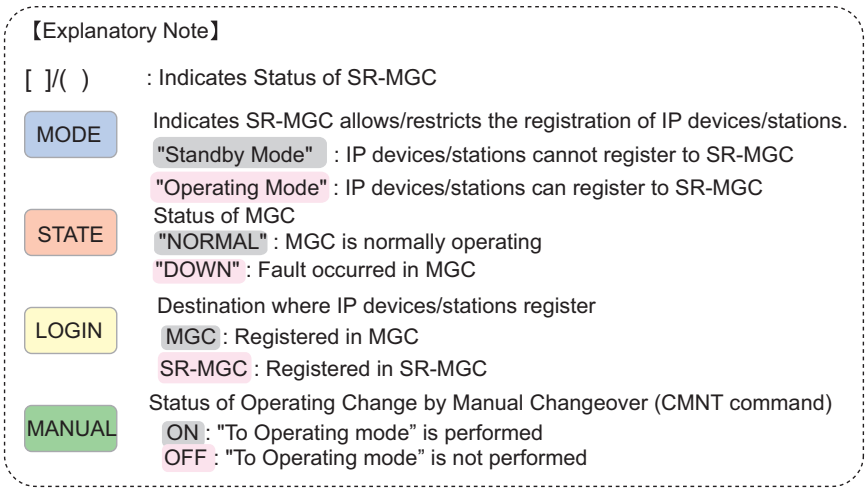
  c.) Failure in Primary & Backup Node: SR-MGC 2 rescues FCCS Cluster support terminals.

  *Health Check is provided between SR-MGC 2 and each FCCS Cluster Node.

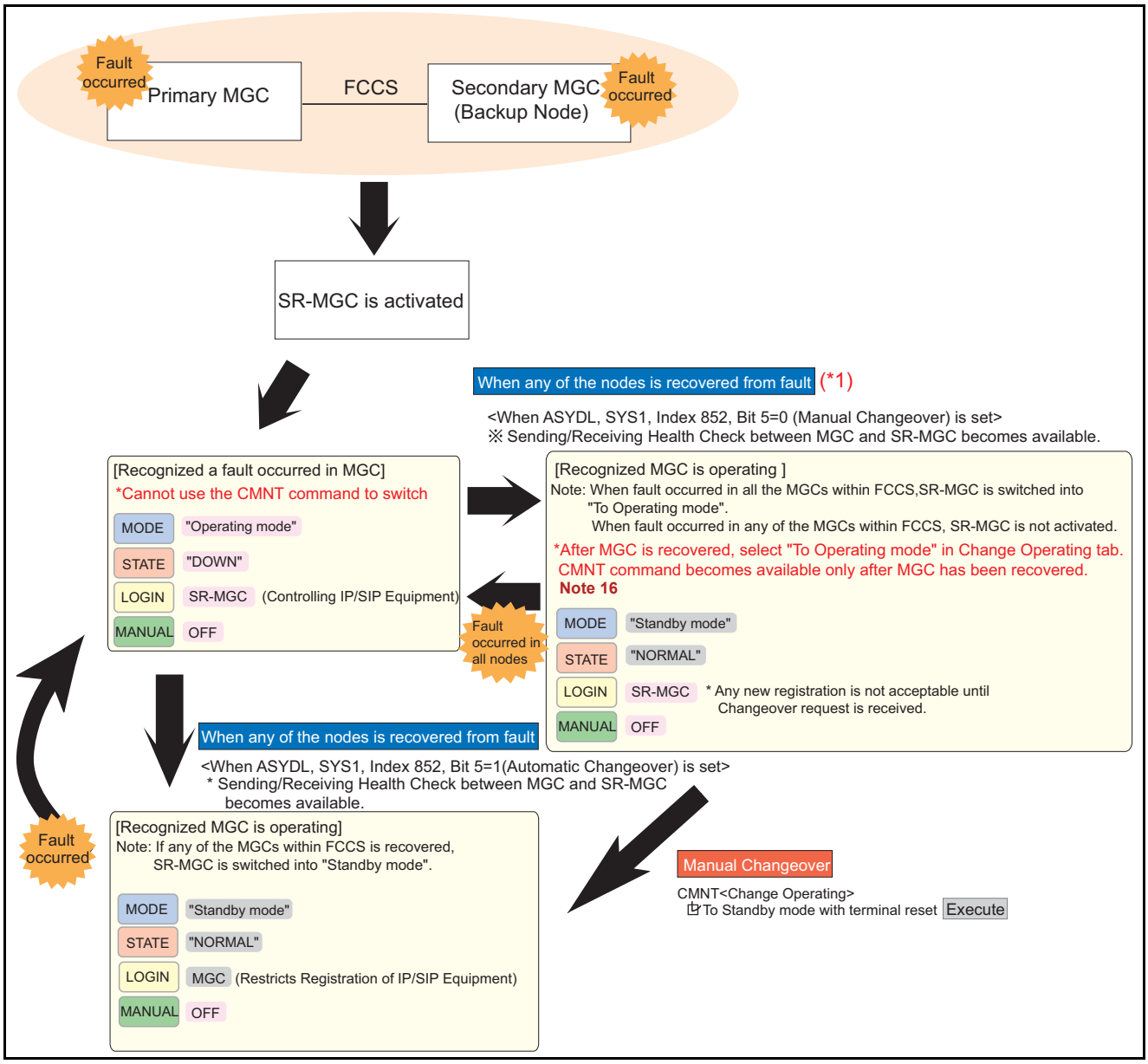
9. When SR-MGC with FCCS Cluster is performed, the operating mode of SR-MGC will be greatly influenced by the communication status between MGC and SR-MGC and the settings of the CMNT command.

   Note: SR-MGC has the following two modes:
   Standby Mode: IP/SIP Equipment cannot register to SR-MGC
   Operating Mode: IP/SIP Equipment can register to SR-MGC
   If IP/SIP equipment registers to SR-MGC in Standby Mode, each equipment operates differently.

   The following are examples when Primary and Secondary MGCs are in Main Office.

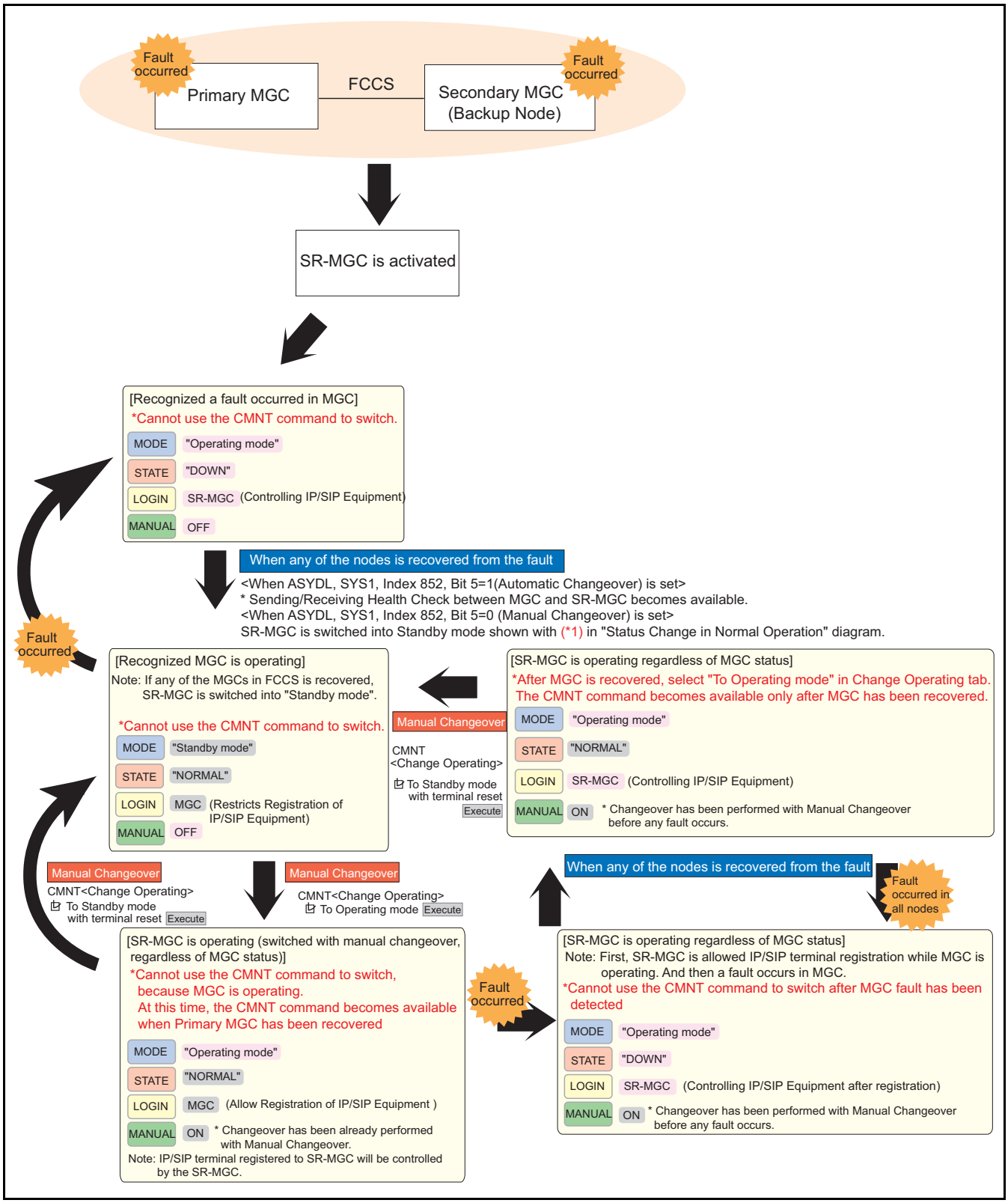


- Status Change in Normal Operation



Note 16: Other than Primary MGC (such as Secondary MGC), the CMNT command is not available.

- Status Change in Test mode (Normally, SR-MGC's status is not changed in the following)



10. This feature is not supported in the UMGi/RNIP system.

Programming (SR-MGC with FCCS Cluster)

For data programming of this feature, refer to the programming part of [S-141] Survivable Remote MGC.

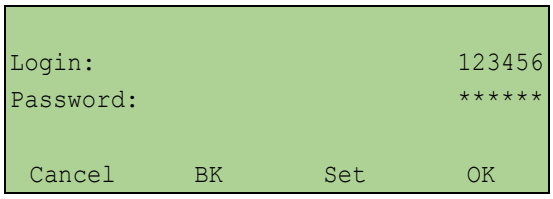
L-56 Login Code

General Description

This feature allows users to log in from IP terminals with Login Code which consists of an arbitrary number, instead of a Station Number, when Login Mode is in service.

Operating Procedure

Example: DtermIP display



STEP 1: Enter the assigned Login Code for "Login" on IP terminal.
STEP 2: Enter the same Login Code you entered in STEP 1 for "Password" on IP terminal. **Note 1**

Note: Password can be changed by terminal operations. For the procedure, refer to DtermIP [D-159] in Data Programming Manual - Business.

Note 1: When logging in from the terminal where the password has not been changed, enter Login Code for Password. If Password has already been changed before registering Login Code by this feature, the Password remains the same. In this case, enter the current Password.

Service Conditions

<Important Notices>

1. This feature is only available when a terminal operates in Login Mode.
2. Telephone Number needs to be assigned to the station using an arbitrary Login Code.
3. Login Code should be a unique number which differs from Station Number.

Example 1:
When the following Office Data is assigned,

	Station Number	Login Code
Station A	30000	50000
Station B	50000	Not assigned
Station C	70000	Not assigned

a.) If Login Code conflicts with an existing Station Number, the Login Code overrides the Station Number.

	If Entered in Login Field...	Login Availability
Station A	50000	Logged in as Station Number 30000
Station B	50000	Logged in as Station Number 30000 **Note 2**
Station C	70000	Logged in as Station Number 70000

Note 2: If Station A is not logged in, the user of Station B can log in as Station Number 30000.

b.) If another user logs in by entering a number which conflicts with the identifiers already in use, the new user will override the previous user.

	If Entered in Login Field...
Station A	50000
Station B	50000

In the above case, when Station A is logged in, Station B will override Station A if the user of Station B logs in by entering "50000" in the Login Field. When the Station B is logged in, Station A will override Station B if the user of Station A logs in by entering "50000" in the Login field.

Note: Make sure that terminals with Login Code do not mix with those without Login Code in the same system. Review an Office Data Setting, if necessary.

4. If Login Code has been assigned to a Station, the user cannot use the Station Number to log in.

Example:
When the following Office Data is assigned,

	Station Number	Login Code
Station A	30000	50000

the login availability for each Station is as follows.

	If Entered in Login Field...	Login Availability
Station A	50000	Logged in as Station Number 30000
Station A	30000	Not available

<Interactions with Other Features>

1. This feature is available for Dial Plan to Support IP Centrex Functionality [D-166]. Login Code should be a unique number which differs from station number.

   Example:
   When the following Office Data is assigned,

	Station Number	Private Number	Login Code
Station A	30000	10020000	90000
Station B	50000	90000	Not assigned
Station C	70000	30040000	Not assigned

   a.) If Login Code conflicts with an existing Private Number, the Login Code overrides the Private Number.

	If Entered in Login Field...	Login Availability
Station A	90000	Logged in as Private Number 10020000
Station B	90000	Logged in as Private Number 10020000**Note 3**
Station C	30040000	Logged in as Private Number 30040000

   Note 3: If Station A is not logged in, the user of Station B can log in as Private Number 10020000.

   b.) If another user logs in by entering a number which conflicts with the identifiers already in use, the new user will override previous user..

	If Entered in Login Field...
Station A	90000
Station B	90000

   In the above case, when Station A is logged in, Station B will override the Station A if the user of Station B logs in by entering “90000” in the Login Field. When the Station B is logged in, the Station A will override the Station B if the user of Station A logs in by entering “90000” in the Login Field.

   Note: Make sure that terminals with Login Code do not mix with those without Login Code in the same system. Review an Office Data Setting, if necessary.

2. If Login Code has been assigned to a Station, the user cannot use the Private Number to log in.

Example:
When the following Office Data is assigned,

	Station Number	Private Number	Login Code
Station A	30000	10020000	90000

the login availability for each Station is as follows.

	If Entered in Login Field...	Login Availability
Station A	90000	Logged in as Private Number 10020000
Station A	10020000	Not available

3. This feature is available for Location Diversity [L-55] as well as for login to Backup Node.
4. When MAC mode is in service, this feature is only available when a user logs in under Login Mode after logging out by pressing the LOGOUT key.
5. This feature is available for Terminal Remote Login [T-66] in an FCCS network.

Note: This feature is not available for Terminal Remote Login [T-66] in CCIS networks.

<Hardware/Software>

1. This feature is available for terminals where Login Mode is in service.
2. This feature is available for login to SR-MGC.
3. To enable a SIP Handler Controlled SIP terminal to log in to the system with the station number which has the Login Code setting, the Login Code needs to be added to the user name in the SIP Handler Controlled SIP terminal side setting.

<Interactions with Networks>

None.

<Conditions Specific to This Feature>

1. Available characters for Login Code are numerals (0-9) as well as “#” and “*” (maximum 16 digits).
2. One Login Code can only be assigned to one Station Number.
3. If the Telephone Number, which the Login Code is already registered for, is deleted by ALGSL/ALGSN command, the registered data of Login Code is also deleted.

<Other Conditions>

None.

Programming

ALGCL/ALGCN - Login Code Data (LDM/NDM)

UGN: User Group Number [1 - 255]
TELN: Telephone Number [0 - 9, # and * (maximum 16 digits)]
LOGIN CODE: Arbitrary Login Code [0 - 9, # and * (maximum 16 digits)]

L-57 Last Login Info Retention

General Description

This feature allows IP terminal and SIP Multiple Line terminal (herein after referred to “IP/SIP terminal”) users to log in to any phone in free seating arrangements with their station number as the login User ID, and continue using the station number after an interruption due to power loss or terminal reset.
If the office data is backed up and restored, they can resume using their phones even after the system initialization while maintaining the last station number used.
For example, after transferring to another section of the company or changing seating arrangements, if an IP/SIP terminal user logs in to a different phone once, the terminal user does not have to log in subsequently even after the terminal reset occurs.

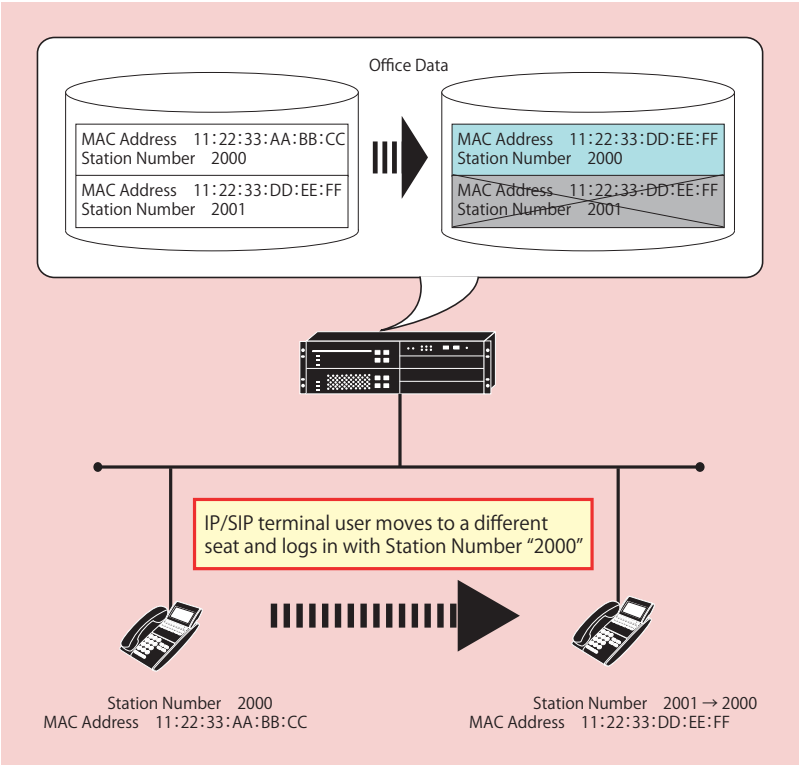
When the IP/SIP terminal user logs in using this feature, the following processes are handled inside of the system:

[Auto-Assignment of MAC Address Information]

When an IP/SIP terminal user enters a User ID and a Password, this process assigns a MAC address of login terminal to the office data.

[Auto-Deletion of existing MAC Address Information]

This process deletes an existing MAC address data when it is identical with a MAC address of login terminal. Also, modifies an existing station number data when it is identical with the login station number.



Operating Procedure

In the Login window, enter a User ID and a Password to log in.

Note: If the IP/SIP terminal has already been in the login state, select “Yes” when “Override?” is displayed in the LCD.

Service Conditions

<Important Notices>

None

<Interactions with Other Features>

1. When a Remote Node and a Local Node which are connected through the FCCS network due to Terminal Remote Login [T-66] use the same UGN number, specify whether to activate this feature with the system data (ASYDL/ASYDN, SYS1, Index 681, Bit 6).
   If the systems are connected through the CCIS network, this feature is not activated.
2. When Dial Plan to support IP Centrex Functionality [D-166] is in service, the availability of this feature depends on the conditions of the new seating position.

Moving Between Nodes	Moving Between UGNs	Last Login Info Retention [L-57]	Reason
No	No	Activated	Moving to a seat whose UGN is the same as before.
No	Yes	Not Activated **Note 1**	Moving to a seat whose UGN is different from before. **Note 3**
Yes	Yes	Not Activated **Note 1**	IP/SIP terminal performs registration to a Remote Node.
Yes	No	Previous Node: Activated **Note 2** Current Node: Not Activated	Moving to a seat whose UGN is the same as before.
			IP/SIP terminal user logs in to a Remote Node.

Note 1: The IP/SIP terminal user can log in but the MAC address is not updated. After the IP/SIP terminal restart or the system initialization, the IP/SIP terminal user needs to log in again.

Note 2: When there is no MAC address information, the MAC address information is input.

Note 3: When the IP/SIP terminal user logs in from a new seating position, it is regarded as an operation of override.

3. When a Physical Station Number is used for a User ID and a Password in the system using Physical Station Number Calling Restriction [P-87], this feature is not activated.

4. When a LN of Location Diversity [L-55] is set as Primary Node, note the following condition:

   If this feature is used when IP/SIP terminals are placed under the Secondary Node due to a failure in the Primary Node, the terminals cannot return to under the Primary Node after Primary Node recovers. To place the terminals back under the Primary Node, the following operations are required:

   - Assign the Primary Node IP address as Telephony Server IP address for the IP/SIP terminal configuration data.
   - Change the MAC address information back to the address before the failure occurs using the AMAC command.

5. The following table shows conditions of Auto-Assignment of MAC address information depending on the system data assignment (ASYDN, SYS1, Index 963, Bit 1: Location Diversity [L-55]).

Feature		Auto-Assignment of MAC Address Information	
Last Login Info Retention [L-57]	Location Diversity [L-55]	AMAC Command Data	AMACN Command Data
In Service	Out of Service	Supported	Supported
	In Service	Unsupported	Supported

Note: When both of Location Diversity [L-55] and Last Login Info Retention [L-57] are in service, the MAC address information remains if it is assigned by the AMAC command. Delete the MAC address information with the AMAC command.

6. When the registration destination of IP/SIP terminals is placed to SR-MGC, if the office data including MAC address has been copied into the SR-MGC, IP/SIP terminals start with the login station number.

<Hardware/Software>

1. When Soft Client SP350 is used, apply this feature with the system data (ASYDL/ASYDN, SYS1, Index 681, Bit 5=1).
2. This feature cannot be used with SIP Handler Controlled Standard SIP terminals.

<Interactions with Networks>

If IP/SIP terminal performs registration to the LN while a link between NCN and LN is disconnected in the FCCS system, MAC address is automatically assigned to the LDM but it is not assigned to the NDM. When the registration destination is changed to NCN or another LN, User ID and Password entries are required on the first login after the registration destination is changed.

<Conditions Specific to This Feature>

1. The following MAC address information is updated through this feature.
   - MAC address assigned with MAC Registration Password (LDM/NDM)
   - MAC address assigned with IP Terminal Automatic Assignment Function **Note 4**

- MAC address assigned with the AMAC/AMACN commands

Note 4: This condition is applied to the case where the checkbox for MAC address is registered is selected in the AEAAL command window.

2. When the following processes are in progress, this feature is not activated.

   - Office Data Backup with the MEM_HDD/MEM_HDD_N commands
   - Office Data Backup with the Routine Diagnosis
   - Broadcast of NDM
   - CPU Changeover

   In the case where this feature is not activated as mentioned above, IP/SIP terminal's registration can be restricted or allowed with the following system data.

   Index 681, Bit 4:
   Terminal's registration to the system when MAC address information update is failed = 0/1: Allow/Restrict

   Note: When IP/SIP terminal's registration to the system is restricted with specifying ASYDL/ASYDN, SYS1, Index 681, Bit 4=1 (Restrict), the following error messages or window are displayed in the LCD of IP/SIP terminal:

System State	IP Terminal	SIP Multiple Line Terminal
Office Data Backup is in progress with the Routine Diagnosis or the MEM_HDD/MEM_HDD_N commands	Unauthorized Retry Yes, No	Login Window is displayed.
	Resource Unavailable	Resource Unavailable
Broadcast of NDM is in progress	PH Unavailable	SIP Server Not Found
	Resource Unavailable	Resource Unavailable
CPU Changeover is in progress	PH Unavailable	SIP Server Not Found
	Resource Unavailable	Resource Unavailable

3. During Auto-Assignment of existing MAC address Information, all existing MAC address information related to the login station number is deleted from the office data.
4. One MAC address is set per station number in the office data.
5. After ASYDL/ASYDN, SYS1, Index 679, Bit 6 is changed from 1 (In Service) to 0 (Out of Service), MAC address information remains in the office data. Delete the remaining MAC address information from the PCPro through the AMAC/AMACN command.
6. If modification of MAC address is attempted from PCPro through the AMAC/AMACN command while a process of Auto-Assignment/Auto-Deletion is in progress, the operation from PCPro is failed. Then "Terminal busy" is displayed in the PCPro window.
7. After the IP/SIP terminal user logs in from the new seat position successfully, the LCD indication of IP/SIP terminal in the previous seat position changes as shown below:

   Connecting Window (LCD: Connecting...) → Login Window

8. In the system operating with Telephone Number, if the Numbering Plan consisting of Telephone Numbers overlaps with the one consisting of Physical Station Numbers, do not assign the same number mutually as a 1st Digit Code in the ANPD command.
Otherwise, when IP/SIP terminal user logs in with a Physical Station Number, the system behaves as the IP/SIP terminal user logs in with a Telephone Number. Then MAC address information is not updated.

   [Example] When the Numbering Plans are set as shown below:

   Numbering Plan Consisting of:
   - Physical Station Number: Four digits number begins with "2" or five digits number begins with "7"
   - Telephone Number: Four digits number begins with "2" or five digits number begins with "7"

Physical Station Number	Telephone Number	Example of Behavior
2000	72000	When IP/SIP terminal user logs in with Station No. 2000, the system regards as the IP/SIP terminal user logs in with Telephone No. 72000.
70000	2700	When IP/SIP terminal user logs in with Station No. 70000, the system regards as the IP/SIP terminal user logs in with Telephone No. 2700.

<Other Conditions>

None.

Programming

STEP 1: **ASYDL/ASYDN** (System Data (LDM/LDN))
SYS1
Index 679
Bit 6=1: Last Login Info Retention [L-57] is in service.
Index 681
Bit 4: Terminal's registration to the system when MAC address information update is failed
0/1: Allow/Restrict
Bit 5: Apply Last Login Info Retention [L-57] to Soft Phone (Soft Client SP350)
0/1: Not apply/Apply
Bit 6: Last Login Info Retention [L-57] with Terminal Remote Login (When Remote Node and Self Node connected via FCCS network have the same UGN number)
0/1: In Service/Out of Service

M-1 Meet-Me Paging

General Description

This feature allows a user dial access to paging equipment, and connects both parties automatically, after the Called Party has answered the page.

Operating Procedure

To page

1. Calling Station dials the Paging Access Code. Receives continuous Ringback Tone for one second.
2. Calling Station pages the desired party.
3. Calling Station remains off-hook or hangs up.

To answer

Called Party dials the Page Answer Code

1. NON-DELAY OPERATION

The party paged is immediately connected to the Calling Party.

2. DELAY OPERATION
   a.) The party paged receives Ringback Tone.
   b.) The Calling Station rings.
   c.) The Calling Station goes off-hook and is immediately connected to the paged Station.

Service Conditions

1. A maximum of six routes can be provided with Paging Trunks.
2. Paging equipment is customer provided.
3. One Paging Trunk will be assigned to each route.
4. Paging access and answer codes are assigned in System Data programming to correspond with each route.
5. The maximum number of digits that can be used as a Paging Access and/or Answer Code is 3.
6. The type of paging service provided on a system basis is programmed via the PCPro into the System database.
7. A page waiting to be answered is automatically canceled after a predetermined program duration. See VARIABLE TIMING PARAMETER [V-1] for other selectable timing choices.
8. The following can be used as a paging trunk interface:
   - COT Circuit Cards
   - MC&MG-COT (4LC2COT)
   - MG-COT (6COT)
   - UG50 (GCD-PGTA) **Note 1 Note 2**

   Note 1: UG50 (GCD-4COTA/B/C) cannot be used as a paging trunk interface.

   Note 2: Available from UG50 software version 2.00.
9. Loop signaling is the recommended signaling type. Refer to PAGING TRANSFER [P-10].

10. When receiving Paging Access Code from CCIS, only the Non-Delay operation is available (Delay operation is not available). Paging Answer and Paging Cancel operations via CCIS are not available. When the Station is paged via CCIS, the Station cannot answer/cancel the page.
11. The billing information is not output to the SMDR after using this feature.

Interactions

1. MEET-ME PAGING and PAGING ACCESS [P-1] can be provided simultaneously in the same system. In ARTD, CDN 6:TCL must be assigned as data "8" for MEET-ME PAGE, while PAGING ACCESS [P-1] routes may be assigned as data "11", General Page, or as data "1", DDD. Separate Access Codes for each feature are required. A MEET-ME PAGING answer and Cancel Code is required for each MEET-ME PAGING zone.
2. MEET-ME PAGING is limited to Stations and Attendant Consoles. Central Office Trunks or REMOTE ACCESS TO SYSTEM [R-2] Calls must have the Attendant Console or user perform the paging operation.
3. The Attendant Console cannot dial the MEET-ME PAGING answer codes
4. BUSY VERIFICATION [B-3] and EXECUTIVE RIGHT-OF WAY [E-1]: These features will be denied for a Connection that includes the Paging route.
5. DELAY OPERATION: Station A is connected to party B. Station A momentarily presses the switch hook and dials the MEET-ME PAGE Route Access Code. When connected to the page, Station A requests Station C to dial the MEET-ME PAGE Answer Code. Station A hangs up. When Station C dials the Answer Code, Station A is rung. Station C and Station A are connected. If Station A hangs up, Party B is connected to Station C. If Station A momentarily presses the switch hook, all three parties are connected together in a THREE-WAY CALLING [T-2] Connection. If Station C does not call the MEET-ME PAGE Answer Code after 30 seconds, Party B rings back to Station A.
6. NON-DELAY OPERATION: Station A is connected to Party B. Station A momentarily presses the switch hook and dials the MEET-ME PAGE Route Access Code. When connected to the page, Station A requests Station C to dial the MEET-ME PAGE Answer Code. Station A stays on the Trunk and is still voice-connected to the paging equipment. Station C dials the MEET-ME PAGE Answer Code. The Paging Trunk becomes idle. Station A is connected to Station C. If Station A hangs up, Party B is connected to Station C. If Station A momentarily presses the switch hook, all three parties are connected together in a THREE-WAY CALLING [T-2] Connection. There is no time-out period if Station C does not dial the MEET-ME PAGE Answer Code. Station A may press the switch hook to disconnect the page Trunk and reconnect to Party B. If desired, Station A may hang up and Station A will be recalled by Party B.
7. Stations allowed Priority Page, SFI = 57 (Priority Page), will come into the paging route, except when the Attendant Console is paging.
8. To answer MEET-ME PAGE, while connected to another party, CALL HOLD [C-6] must be used. THREE-WAY CALLING [T-2] is not allowed.

Programming

STEP 1: **ASYD** - SYS1, Index 73, Bit 0 and Bit 1.

For MEET-ME PAGE, non-Delay service, Bit 0=1, Bit 1=0.

For MEET-ME PAGE, Delay service, Bit 0=0, Bit 1=1.

For MEET-ME PAGE, Delay service with Call Transfer, Bit 0=1, Bit 1=1.

See PAGING TRANSFER [P-10].

SYS1, Index 73, Bit 3. Are PAGE Cancel Codes provided for each route? 0/1: Per route/Common to all routes.

SYS1, Index 74, Bit 0 to Bit 3. Assign the cancel timer for unanswered page.
Assign data 00H for 30 seconds.
(0-F + 1) x 30 = Cancel Timer.
SYS2, Index 1, Bit 0 and Bit 1. Consultation Hold allowed or denied. 00: Denied; 01: Originating and Terminating Calls allowed; Tandem Call Denied; 11: All calls allowed.

STEP 2: **ARTD** - Assign the Paging route as shown below:

RT:1

1-OSGS	:2	2-ONSG	:3	5-TF	:1	6-TCL	:8	7-L/T	:1
8-RLP	:2	15-LSG	:0/1						

Note: To change the data for existing route (RT), initialize the circuit card.

STEP 3: **ATRK** - Assign the LENS, Paging Trunk Route Number, Trunk Number, and Tenant Number.

STEP 4: **MBTK** - Assign the Make Idle status to the Paging Trunks.

STEP 5: **ANPD** - Reserve Levels for Trunk Access and for MEET-ME PAGE answer and Cancel Codes. Select the Trunk Access Level for a Connection Index (CI) of Normal Service (N) or Hooking Service (H) service. Number of necessary digits is usually one or two. Busy Lamp Field is not activated. If the MEET-ME PAGE answer and Cancel Codes reside in a Level reserved for features, then assign according to a predetermined Numbering Plan. If the MEET-ME PAGE answer and Cancel Codes reside in a separate Level, then select a CI of Normal Service (N) only. The number of necessary digits is usually two, and the Busy Lamp Field is OFF.

STEP 6: **ASPA** - For the MEET-ME PAGE route Access Code, assign the Access Code in the Level assigned in ANPD. Type of Service, SRV = "OGC: Outgoing Call" for Outgoing Trunks. Select CIs of "N: Normal" Service or "H: Hooking" Service. For the MEET-ME PAGE route answer and Cancel Codes, assign the Access Code in the Level assigned in ANPD. Type of Service, SRV = "PAGA: Paging Answer" and SRV = "PAGC: Paging Cancel". Assign the proper Route Number associated with this code. Assign for a CI of "N: Normal".

STEP 7: **ARSC** - For Stations allowed access to the paging routes, allow Route Restriction Index (RRI) 3 for the Route Restriction Classes (RSC) assigned to these Stations.

STEP 8: **ASFC** - Assign a Service Feature Restriction Class that allows SFI 38 (Meet-Me Paging) to the Stations that will have MEET-ME PAGE. If Priority Page is used, allow SFI 57 (Priority Page).

STEP 9: **ARRC** - Allow Trunk-to-Trunk Connection between the paging route and the Incoming/Outgoing Route. Assign ARI-A-RES, via the Attendant Console. The Incoming/Outgoing Route would be the party being held at a Station, or Attendant Console wishing to speak to the party for whom the page is intended.

M-2 Miscellaneous Trunk Access

General Description

This feature provides access to all types of external and customer provided equipment/facilities, such as FX [F-2], WATS [W-1], CCSA [C-14], TIE LINE [T-3], and Exchange Network, along with DICTATION [D-3], PAGING [P-1], and CODE CALLING [C-21].

Operating Procedure

MISCELLANEOUS Trunk Access is accomplished via System Data programming at the PCPro.

Service Conditions

1. Access to these Trunk Groups is allowed or denied via CLASS OF SERVICE - INDIVIDUAL [C-15].
2. An Interface Circuit must be provided per Trunk required.

Interactions

If other features are being used (such as SMDR, toll restriction, and/or LCR (S)), refer to those feature descriptions, as other commands may be required that are not listed below.

Programming

STEP 1: **ARTD** - Refer to the Feature listing for the type of Trunk route to be accessed.

STEP 2: **ATRK** - Assign the LENS, Route Number, Trunk Number, and Tenant Number.

STEP 3: **MBTK** - Assign the Make Idle status to the Trunks.

STEP 4: **ANPD** - Assign a Level for Trunk Access. Select Normal Service (N) or Hooking Service (H) from the combo box for Connection Indexes. Number of Necessary Digits is assigned according to a predetermined Numbering Plan. Busy Lamp Field is not activated.

STEP 5: **ASPA** - Assign the Access Code, as indicated, by the Level assigned in ANPD. Type of Service, SRV = “OGC: Outgoing Call” for Outgoing Trunks.

STEP 6: **ARSC** - Assign a Route Restriction Class (RSC) that is allowed in Route Restriction Indexes (RRI) 0, 1, 2 and 3 to the Station that is allowed access to these Trunks.

STEP 7: **ASDT** - Assign an RSC to the Stations that allow access to these Trunks.

M-3 Miscellaneous Trunk Restriction

General Description

This feature provides for certain Stations and certain dial-repeating TIE Trunks to be denied access to particular Trunk Groups, such as FX [F-2], WATS [W-1], CCSA [C-14], TIE LINE [T-3], Exchange Network, DICTATION [D-3] or PAGING [P-1].

Operating Procedure

MISCELLANEOUS TRUNK RESTRICTION is accomplished via System Data programming at the PCPro.

Service Conditions

1. Restricted call attempts can be either:
   a.) intercepted by the Attendant,
   b.) Reorder Tone can be initiated back to the originating Station user,
   c.) Station user can be connected to an announcement informing the user of the restricted call attempt, or
   d.) prompted for Authorization Code.
2. Station Line access to a given miscellaneous Trunk Group is controlled by assigning a Route Restriction Class (RSC) to each Line. There can be a maximum of 15 different RSCs (RSC1-15) per TENANT [T-12], each of which can be programmed to restrict an assigned Station from none, one, or more than one miscellaneous Trunk Group.
3. If the Station user's class of service allows outgoing route completion via the ATTENDANT CONSOLE [A-3], the Station user can dial “0” to reach the Attendant for completion of miscellaneous Trunk Calls.
4. The CALL TRANSFER - ALL CALLS [C-11] feature will operate as follows:
   a.) if the restricted Station is the controller, this Station cannot add a Trunk from which this Station is restricted.
   b.) if the restricted Station Line is not the controller, the controller cannot connect the restricted Station Line to a Trunk from which the Station is normally restricted.
5. Two Outgoing Trunk Group restriction tables are provided per system. The Direct Dial Access table is indexed when a Station user dials the Trunk Group Access Code and the ATTENDANT CONSOLE [A-3] table is indexed on Attendant completed calls. The restriction options per RSC can differ per table, thus allowing Attendant completion of Outgoing Calls for restricted Stations.
6. NIGHT SERVICE [N-1, 2]: Restrictions can be changed for a given Station when the system is switched to the Night mode of operation.
7. CALL TRANSFER - ALL CALLS [C-11]: If a restricted Station user calls another Station, the Called Station cannot use the CALL TRANSFER feature to connect the Calling Station to a Trunk Group to which the restricted Station is denied dial access. If transfer is attempted, the controlling Station is recalled.
8. TIE LINE ACCESS [T-3]: If the Station user accesses a TIE Trunk at a distant Office, access to Outside exchanges can only be controlled by the distant Office.

Interactions

If other services are being used (such as SMDR, toll restriction, and/or LCR (S)), refer to those feature descriptions, as other commands may be required that are not listed.

Programming

STEP 1: **ASYD** - SYS2, Index 1, Bit 4 and Bit 5. Should a restricted Station be routed to the Attendant Console or receive Reorder Tone?

Reorder Tone: Bit 4=0, Bit 5=0.

Attendant Console: Bit 4=1, Bit 5=0.

STEP 2: **ARSC** - Follow the procedure for [M-2] MISCELLANEOUS Trunk Access [M-2]. By assigning the ARSC command, restrict RSCs from receiving certain routes. RRI 3: RES must be "2" if call restriction to go to ATTCON. Then assign the restricted RSCs to Stations via the ASDT or ASCL command. When assigning the ARRC command, restrict Incoming TIE Lines from accessing certain routes.

FCCS Programming

This data setting describes only the additional or replacement commands used to activate FCCS Services. When used as an FCCS feature, these commands replace the non-network commands (for example, AAED is replaced by AAEDN). Note that the data setting for FCCS Service varies depending on the Node.

Note: When using ASYDL or ASYDN to program an FCCS feature, the basic programming of ASYD is required in addition to ASYDL or ASYDN.

ARSCN - Follow the procedure for [M-2] MISCELLANEOUS Trunk Access [M-2]. By assigning the ARSC command, restrict RSCs from receiving certain routes. RRI 3: RES must be "2" if call restriction to go to ATTCON. Then assign the restricted RSCs to Stations via the ASDT or ASCL command. When assigning the ARRC command, restrict Incoming TIE Lines from accessing certain routes.

M-4 Multiple Console Operation

General Description

This feature allows one or more ATTENDANT CONSOLEs [A-3] to operate within the same system.

Operating Procedure

No manual operation is required.

Service Conditions

1. The maximum number of Consoles per PIR is 4.
2. The maximum number of Consoles allowable per system is 16.
3. Each Incoming Call is displayed on an ATTENDANT CONSOLE [A-3], whether idle or busy. If all Attendants are involved in processing calls when another Central Office Call arrives, the CALL WAITING LAMP [C-13] will light on all ATTENDANT CONSOLE [A-3]. This is a Switched Loop Attendant operation.
4. If one of the ATTENDANT CONSOLE [A-3] enters the night mode, all calls addressed to the Attendant will be directed to the other ATTENDANT CONSOLE [A-3]. However, the ATTENDANT CONSOLE [A-3] that has entered the night mode must complete processing of all calls already connected to it's Loops.
5. A Station can be connected to only one Attendant Loop at a time. Any attempt at establishing multiple Connections will result in Reorder Tone being sent to the Attendant.
6. INTER-POSITION TRANSFER [I-5] is used to transfer calls between ATTENDANT CONSOLE [A-3].
7. Two methods of operation are available for placing the system (or Tenant Group) into the night mode:
   a.) All ATTCON Change System Day to Night: When all ATTENDANT CONSOLE [A-3] press the Night key, the system (or associated TENANT [T-12]) enters the night mode. Night to Day: When one ATTENDANT CONSOLE [A-3] releases the Night key, the system (or associated TENANT [T-12]) enters the day mode.
   b.) Primary ATTCON Change System Day to Night: When the designated primary ATTENDANT CONSOLE [A-3] presses the Night key, the system (or associated TENANT [T-12]) enters the night mode. If an ATTENDANT CONSOLE [A-3] other than the primary Console presses the Night key, only that Console enters the night mode. Night to Day: When the primary ATTENDANT CONSOLE [A-3] releases the Night key, the system (or associated TENANT [T-12]) enters the day mode.
8. The automatic distribution of Attendant seeking calls is not provided (see Call Queuing). Any ATTENDANT CONSOLE [A-3] assigned to the same Tenant can answer a particular call; an exception is an INTER - POSITION TRANSFER [I-5] directed call.

Interactions

1. See ATTENDANT CONSOLE [A-3] for information regarding a single Attendant Console and system features provided to the Attendant Consoles. The Attendant Console must first be provided.
2. For each Tenant, only one type of Attendant Console system can be assigned, either normal or Primary Attendant Console system.
3. ASAT must be programmed for INTER-POSITION TRANSFER [I-5].

4. When more than one Attendant Console is installed, a Primary Attendant Console may be assigned (AMAT command). This Attendant Console will control the implementation of DAY/NIGHT CLASS OF SERVICE [D-15].
5. If the Primary Attendant places the system into Night mode, the Night LED on the other Consoles do not light.

Programming

STEP 1: **ASYD** - SYS1, Index 9. Assign the number of Attendant Consoles in the system. This provides information to the ASAT command.

SYS2, Index 2, Bit 2 to Bit 3. ATTCON Day/Night mode change is:

Bit 3	Bit 2	
0	0	= Not required (External Switch effective)
0	1	= by Primary Attendant Console
1	0	= by all Attendant Consoles

STEP 2: **ATRK** - To assign the Attendant Console: LENS = MG, and UNIT. The Line Group is always 20 and the Level is either 0 or 1. Assign as Route 901. Also assign a Service Feature Restriction Class (SFC 0) and a Route Restriction Class (RSC 0) to the Attendant Console.

STEP 3: **ASAT** - Assign a specific Attendant Number to the Attendant Console. See INDIVIDUAL ATTENDANT ACCESS [I-6].

STEP 4: **AMAT** - Assign one of the Attendant Consoles to serve as the Primary Attendant Console. This service is assigned on a Tenant basis. The Attendant Number will correspond to the Trunk Number of that specific Console.

STEP 5: **MBTK** - Make Idle the Attendant Console Trunk.

STEP 6: **ATNR** - Allow Tenant-to-Tenant Connection for Inter- and Intra-Tenant basis. Assign Tenant Restriction Index (TRI) 1 for Station-to-Station Calling; TRI 1 for Call Forwarding, TRI 3 for Call Forwarding set by the Attendant Console; TRI 4 for Incoming Calls to the Attendant Console.

STEP 7: **ASFC** - Allow SFC 0 to all needed features.

STEP 8: **ARSC** - Allow RSC 0 access to all routes via RRI 0, 1, 2 and 3.

M-7 Music On Hold

General Description

This feature allows a party to hear music while in the CALL HOLD [C-6], CALL TRANSFER [C-10, 11], or ATTENDANT CAMP-ON WITH TONE INDICATION [A-1] conditions. Two types of MUSIC-ON-HOLD (MOH) exists; internal MOH and external MOH.

Operating Procedure

No manual operation is required for this software controlled feature.

Service Conditions

<Internal MUSIC-ON-HOLD Conditions>

1. The system is capable of supplying two switch-selectable music channels. Music is generated by 3 synthesizers included in the package.
2. If another music source (tuner, tape-deck, etc.) is desired, it must be customer provided.
3. The Held Call condition can be the result of:

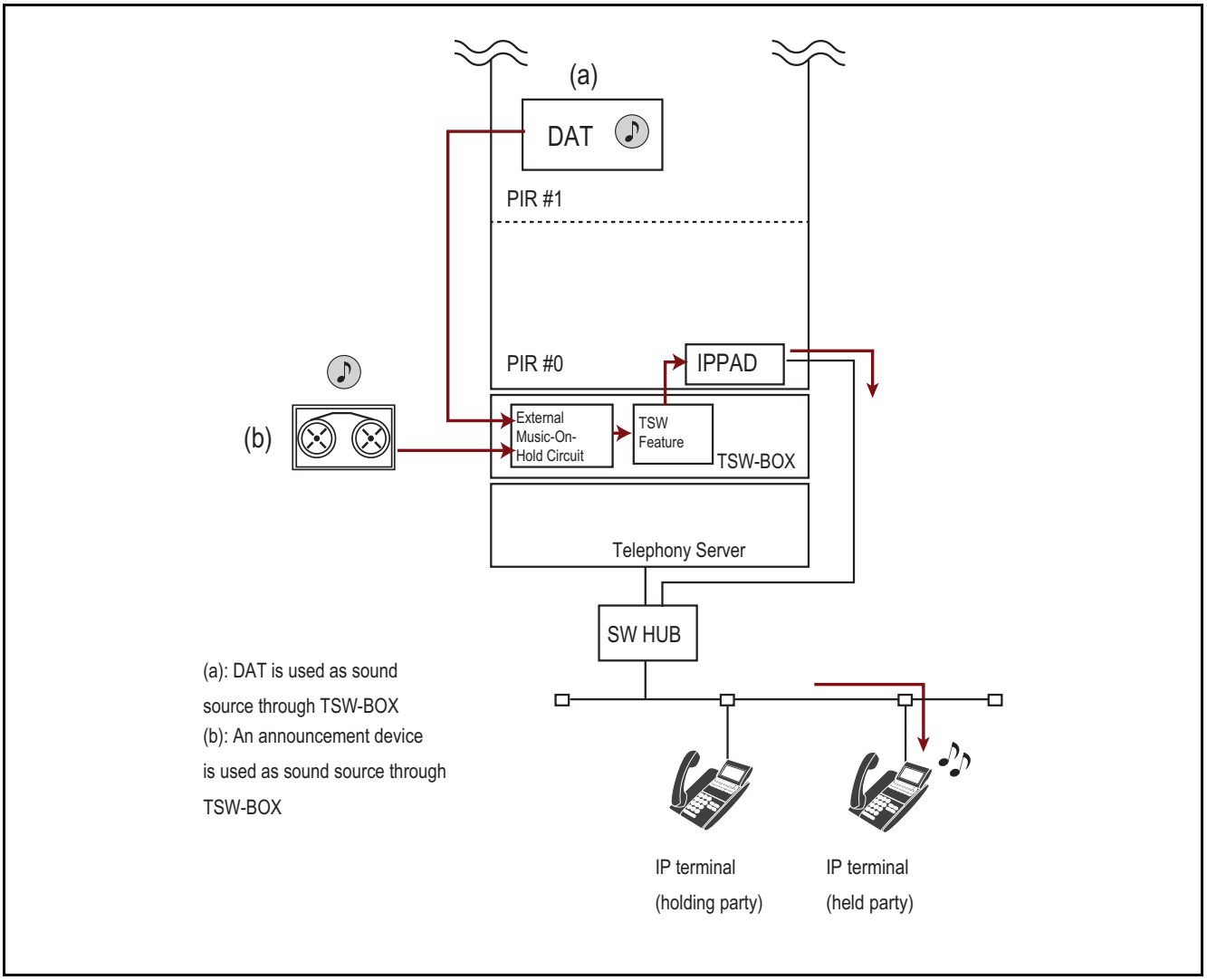
ATTENDANT CAMP-ON WITH TONE INDICATION	[A-1]
CALL HOLD	[C-10]
CONSULTATION HOLD - ALL CALLS	[C-17]
SPLITTING	[S-4]
Attendant Hold	

4. Only one music source at a time, either customer-provided or internal, can be provided per system (system option).
5. ATTENDANT CONSOLE [A-3]: Attendant operations resulting in music to be provided to the Calling Party include the following:
   a.) The Calling Party or the remaining party (not talking with the Attendant) hears music, if the Attendant splits a call prior to completion or after answering an Attendant Recall.
   b.) When Incoming Calls to the Attendant are answered and the Attendant presses the HOLD key, the Held Party hears music.
   c.) When the Attendant camps on a call to a busy Station, the Calling Party is connected to music until the Called Party answers or the Attendant reenters the Switched Loop.
6. When a Station user in a two-party talking Connection switch hook flashes, the Second Party is placed on hold and is connected to music.
7. When activating this feature via the FCCS Network, when either the internal or the external source is used, it is recommended to set the source as the same MOH at each Node for the Node-to-Node Connection, since the Node that controls MOH varies depending on the connection pattern.
8. MG(BRI) cannot use external MOH. To use external MOH with MG(BRI), see EXTERNAL IP MUSIC-ON-HOLD in VS32 Conference Server [V-26].
9. To allow a SIP Handler Controlled SIP terminal to hear Internal Source MOH, enable MUSIC for IP-DTG Tone Send Control Bit through the AUACL/AUACN command.

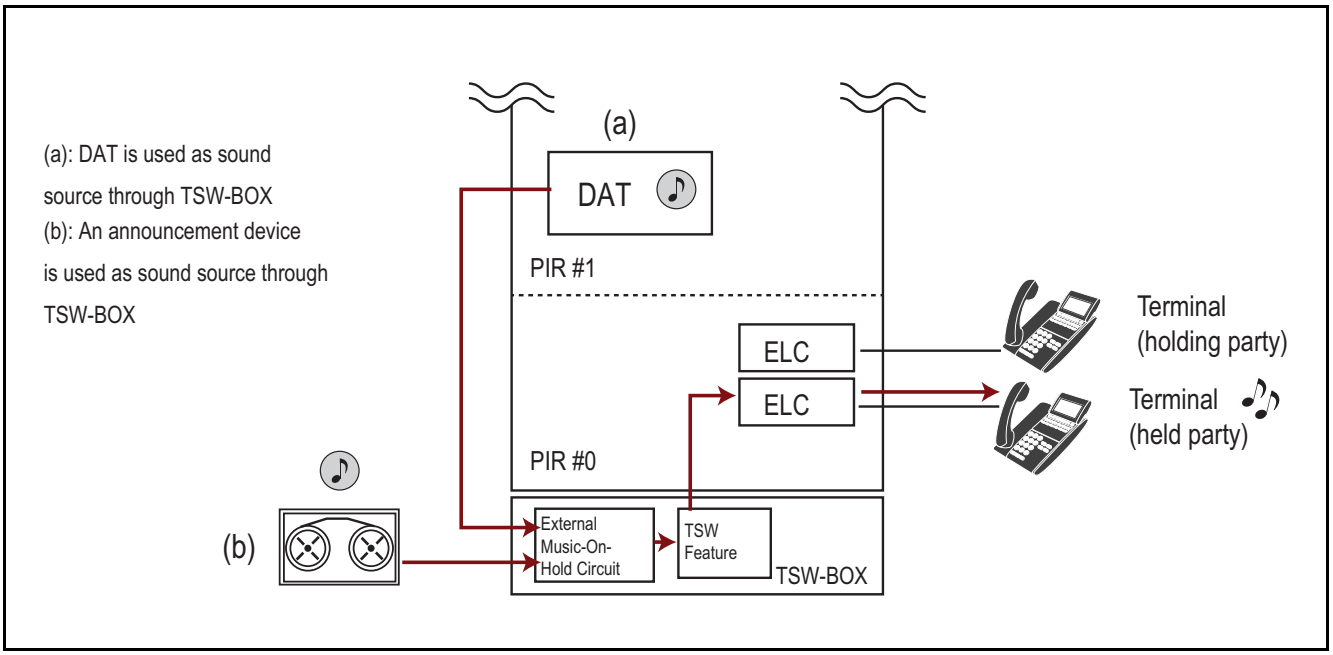
<External MUSIC-ON-HOLD Conditions>

1. By using DAT card or BGM equipment, External MOH source is transmitted from the MUSIC connector (DSUB) to EMA card.
2. External MOH source is connected to DAT card or BGM equipment with the following cable:
   1-IMG: D15 ST-F CA-B (connecting to MUSIC connector on EMA card)
   4-IMG: D15 ST-F CA-B (connecting to MUSIC connector on backboard of PLO card)
3. IPPAD circuit card is required for IP terminals receiving External MOH. If IPPAD is not installed in the system or the card is in link block state such as out of the assigned port number, the IP terminal receives internal MOH.
4. Whenever using external MOH with IP terminal, the port on IPPAD circuit card is hunted for the Connection.
5. External MOH for IP terminal cannot be provided when PHC (Internal PHC) is used.
6. As for External Music source, the electrical resistance for output must be 8 ohm to connect the system. 8 ohm of the music source is a typical one in the input/output.

When Providing External MOH for IP Terminals



When Providing External MOH for Digital Terminals



<Conditions Specific to UMGi/RNIP System>

1. When using UMGi/RNIP System, this feature is available only for stations and trunks in the node connected to External MOH source such as DAT or Announcement Equipment.
2. When this feature is activated at IP Station (IP Enabled Dterm, DtermIP or Analog MC) via IPPAD circuit card, External MOH source must be connected to the node that accommodates the IPPAD.
3. (For 16-IMG System) When using 16-IMG System, this feature is available only for Stations and Trunks in the LMG connected to External MOH source such as DAT or Announcement Equipment.
4. (For 16-IMG System) When this feature is activated at IP Station (IP Enabled Dterm, DtermIP or Analog MC) via IPPAD Circuit Card, External MOH source must be connected to the LMG that accommodates the IPPAD.

Interactions

1. Two types of MUSIC ON HOLD (MOH) exist. Internal Source MOH and External Source MOH.

Programming

<Internal Source MOH>

ASYD - SYS1, Index 64, Bit 7, Is MUSIC-ON-HOLD in service? 0/1: No/Yes.

<External Source MOH (DAT)>

STEP 1: **ASYD** - SYS1, Index 64, Bit 7, Is MUSIC-ON-HOLD in service? 0/1: No/Yes.
STEP 2: **ASYDL** - SYS1, Index 810, Bit 2, IP terminals use External MUSIC-ON-HOLD? 0/1: No/Yes.
STEP 3: **ARTD** - Assign the Route Data of DAT card.
OSGS (CDN 1) = 2
ONSG (CDN 2) = 2
TF (CDN 5) = 3
TCL (CDN 6) = 4
L/T (CDN 7) = 1

Note: To change the data for existing route (RT), initialize or reboot the trunk device such as circuit card, MG, or VS32.

STEP 4: **ATRK** - Assign the Trunk data of the DAT card.
STEP 5: **MBTK** - Cancel the make-busy of the DAT card.

M-11 Meet-Me Paging - Attendant

General Description

This feature allows an Attendant to hold an Incoming Call, page the Called Party, and connect the two after the Called Party has answered the page.

Operating Procedure

PROCEDURE 1:

To page

1. Place the Incoming Call on hold by pressing the HOLD key.
2. The Attendant seizes an Idle Loop.
3. The Attendant dials the Paging Access Code. Receives continuous Ringback Tone for one second.
4. The Attendant pages the desired Party.
5. The Attendant presses the RELEASE key.

To answer

1. The paged party dials the PAGE Answer Code. Receives Ringback Tone.
2. Ringing is heard at Console. After the call is answered (on Loop), the Attendant and paged party converse.
3. The Attendant presses the Loop of the held Outside Call.
4. The Attendant presses the RELEASE key to connect both parties.

PROCEDURE 2:

To page

1. The Attendant answers Incoming Call.
2. The Attendant dials the Paging Access Code. Receives one second of Ringback Tone.
3. The Attendant pages the desired party.
4. The Attendant presses the RELEASE key.

To answer

1. Paged party dials the PAGE Answer Code and is directly connected to the holding party.

   OR

2. Paged party dials the PAGE Answer Code and the Individual Attendant Number and receives Ringback Tone. The Attendant and the paged party converse. The Attendant presses the Loop key of the held Outside Call. The Attendant presses RELEASE key to connect both parties.

Service Conditions

1. If operating Procedure 2 is used, a maximum of six routes can be provided with Paging Trunks.
2. If Procedure 1 is used, the limitation of routes is based on the number of COT cards equipped.
3. Paging equipment is customer provided.
4. One Paging Trunk (COT) will be assigned to each route.
5. UG50 can be used as interface of Paging Trunk by using a PGT card (available for UG50 software version 2.00 or later). However, a COT card installed in UG50 cannot be used as an interface to the Paging Trunk. For details, see Peripheral Equipment Description (UG50).
6. If Procedure 2 is used, only one Held Call Waiting for a paged party can be held at a time at the Console.
7. Paging access and answer codes are assigned to correspond with each route.
8. The maximum number of digits that can be used as a PAGE Access and/or Answer Code is 3.

9. The type of Paging service provided on a system basis is programmed via the PCPro into the System database.
10. A Page waiting to be answered is automatically canceled after a predetermined programmed duration. See VARIABLE TIMING PARAMETERS [V-1] for selectable timing choices.
11. Loop signaling is the recommended signaling type.
    a.) Both MEET-ME PAGING and PAGING ACCESS [P-1] (loudspeaker - basic) can be provided in the same system.
    b.) A separate Access Code for MEET-ME PAGING and PAGING ACCESS [P-1] is required for the same Paging Zone.
    c.) A Paging Answer Code is required for each MEET-ME PAGING zone.
    d.) The maximum number of answer back ports is limited to the number of paging zones (20 max.). These answerback ports are used for RADIO PAGING [R-16] service also.
    e.) ALL ZONE PAGING [A-53] cannot be used with MEET-ME PAGING.
    f.) Paging over the loudspeaker system is limited to paging from a Station or an ATTENDANT CONSOLE [A-3]. Callers on Outside Central Office Trunks must make arrangements with an Attendant or a Station user to have paging performed.
    g.) Direct access (key) to paging from the ATTENDANT CONSOLE [A-3] is not available.
    h.) The Attendant cannot dial the paging answer codes.
12. CALL TRANSFER - ALL CALLS [C-11]: A user can flash and place a Second Party on hold, hear Special Dial Tone, and dial the Paging Access Code. The Paging Transfer (and use of loudspeaker paging system with a party on hold) is allowed, unless denied by the Class of Service of the Station user's Line.
13. Central Office Trunks are used as Interface Circuits to the Paging equipment.
14. BUSY VERIFICATION [B-3]: Busy Verification is denied on a Station Line that is connected to paging equipment. Attempts to busy-verify a Station Line will result in the Attendant hearing Reorder Tone.
15. ATTENDANT-CONTROLLED CONFERENCE [A-2]: When a Conference Call is established, an Attendant cannot connect the Conference Connection to paging loudspeakers.
16. CALL WAITING [C-12, 31]: A call is not allowed to wait on a Line that has accessed loudspeaker paging (making a page). The user hears Reorder Tone.
17. EXECUTIVE RIGHT-OF-WAY [E-1] is denied to a Station Line connected to loudspeaker paging. The user receives Reorder Tone.
18. ATTENDANT CAMP-ON [A-1] is not allowed on a line that has accessed loudspeaker paging. The Station user receives Reorder Tone.
19. THREE-WAY CALLING [T-2] is not allowed after the paged party has been connected to the originating Station.
20. Delayed or non-Delayed operation are the two methods for answer back, either can be selected.
    a.) Delayed Operation: The paging party goes on-hook to wait for the paged party to answer back.
    b.) Non Delayed Operation: The paging party remains off-hook to wait for the paged party to answer back.
21. The Paging Trunk remains in a busy state, while the paging party remains unanswered (the Paging Answer Code has not been dialed). The paging answer back state will be canceled after a programmable time-out period, if the Answer Code is not dialed within the prescribed interval, releasing the Paging Trunk and the paging party as well, if still in the off-hook mode.
22. Procedure 1 must have ASYD, SYS1, Index 73, Bit 7=1.

Interactions

1. MEET-ME PAGING and PAGING ACCESS [P-1] can be provided simultaneously in the same system.

ARTD, CDN 6:TCL must be assigned as data "8" for MEET-ME PAGE, while PAGING ACCESS routes may be assigned as data "11", General Page, or as data "1", DDD. Separate Access Codes for each feature is required. MEET-ME PAGING answer and Cancel Codes are required for each MEET-ME PAGING zone.

2. MEET-ME PAGING is limited to Stations and Attendant Consoles. Central Office Trunks, TIE LINE ACCESS [T-6] or REMOTE ACCESS TO SYSTEM [R-2] Calls must have the Attendant or Station user perform the Paging operation.
3. The Attendant Console cannot dial the MEET-ME PAGING answer codes.
4. BUSY VERIFICATION [B-3] and EXECUTIVE RIGHT OF WAY [E-1]: These features will be denied for a Connection, including the Paging route.
5. DELAY OPERATION: The Attendant Console is connected to party B.
   a.) The Attendant dials the MEET-ME PAGE Route Access Code. When connected to the Page Trunk, the Attendant requests Station C to dial the MEET-ME PAGE Answer Code. The Attendant presses the RELEASE key. When Station C dials the Answer Code, Party B is connected to Station C.
   OR
   b.) The Attendant presses the HOLD key, then seizes an Idle Loop and dials the MEET-ME PAGE Route Access Code. After the paging, the Attendant presses the RELEASE key. When Station C dials the Answer Code, Station C terminates to an Idle Loop key on the Attendant Console. The Attendant answers and places this call on hold, then sequentially presses the two held Loop keys. This connects the Attendant to the party connected to the first pressed Loop key. The Attendant can either press the TALK key, creating a THREE-WAY CALLING [T-2] Connection, or press the RELEASE key, allowing Party B and Station C to be connected. If Station C does not call the MEET-ME PAGE Answer Code within 30 seconds, Party B rings back to the Attendant Console. Normal operating indications are applied.
6. Stations and Attendant Consoles allowed Priority Page, SFI = 57 (Priority Paging), will barge into the Paging route, except when the Attendant Console is paging. An Access Code to Priority Page must be assigned in ASPA, SRV = "SSC: Service Code", SID 40, and assigned for a Connection Index of "N: Normal".
7. To answer the MEET-ME PAGE while connected to another party, CALL HOLD [C-6] must be used. THREE-WAY CALLING [T-2] is not allowed.

Programming

STEP 1: **ASYD** - SYS1, Index 73, Bit 0 and Bit 1.

For MEET-ME PAGE, assign non-Delay service, Bit 0=1, Bit 1=0.

For MEET-ME PAGE, assign Delay service, Bit 0=0, Bit 1=1.

For MEET-ME PAGE, assign Delay service with CALL TRANSFER, Bit 0=1, Bit 1=1. See PAGING TRANSFER [P-10].

Note: ASYD, SYS1, Index 73, Bit 0 and Bit 1 apply to MEET-ME PAGING Station.

SYS1, Index 73, Bit 3. Are PAGE Cancel Codes provided for each route? 0/1: Per route/Common to all routes.

SYS1, Index 73, Bit 7. Attendant Console operation, 0/1: Connect two parties directly when paged party dials PAGE Answer Code/Paged Party terminates on an idle Attendant Console Loop key, then the Attendant presses both Loop keys to connect the two parties.

After the call is answered (on Loop), the Attendant and Paged Party talk. The Attendant presses the

Loop key of the Held Party, then the Attendant presses RELEASE to connect both parties.

SYS1, Index 74, Bit 0 to Bit 3. Assign the cancel timer for unanswered page. Assign data 00H for 30 seconds.

(0-F + 1) x 30 = Cancel Timer.

STEP 2: **ARTD** - Assign the Paging route as shown below:

RT: 1

1-OSGS	:2	2-ONSG	:3	5-TF	:1	6-TCL	:8	7-L/T	:1
8-RLP	:2	15-LSG	:0/1						

Note: To change the data for existing route (RT), initialize or reboot the trunk device such as circuit card, MG, or VS32.

STEP 3: **ATRK** - Assign the LENS, Paging Trunk Route Number, Trunk Number, and Tenant Number.

STEP 4: **MBTK** - Assign the Make Idle status to the Paging Trunks.

STEP 5: **ANPD** - Reserve Levels for Trunk Access and for PAGE answer and Cancel Codes. Select the Trunk Access Level Connection Indexes, CI = N, H; Normal Service (N) or Hooking Service (H). Number of Necessary Digits is usually one or 2. Busy Lamp Field is not activated. If the PAGE answer and Cancel Codes reside in a Level reserved for features, then assign according to a predetermined Numbering Plan. If the PAGE answer and Cancel Codes reside in a separate Level, then select a Connection Index (CI) of Normal Service (N) only. The Number of Necessary Digits is usually two, and the Busy Lamp Field is OFF.

STEP 6: **ASPA** - For the Page Trunk Access Code, assign the Access Code in the Level as assigned in ANPD above. Type of Service, SRV = "OGC: Outgoing Call" for Outgoing Trunks, Assign for Connection Indexes (CI) of "N: Normal" and "H: Hooking". For the Page Route answer and Cancel Codes, assign the Access Code in the Level assigned in ANPD. Type of Service, SRV = "PAGA: Paging Answer" and SRV = "PAGC: Paging Cancel". Assign the proper Route Number that is associated with this Code. Assign a Connection Index of "N: Normal".

STEP 7: **ARSC** - Allow access to the paging route to the RSC of the Attendant Console in Route Restriction Indexes (RRI) 2 and 3. To the RSC of the responding Station, allow access to the Incoming Trunk in RRI 0 and 1.

STEP 8: **ASFC** - Assign a Service Feature Restriction Class (SFC) that allows SFI 38 (Meet-Me Paging), MEET-ME PAGE to the Stations that will answer a MEET-ME PAGE.

STEP 9: **ARRC** - Allow Trunk-to-Trunk Connection between the Paging route and the Incoming/Outgoing Route. Assign for ARI-A-RES, via the Attendant Console. The Incoming/Outgoing Route would be the party being held at a Station or Attendant Console wishing to speak to the party for whom the MEET-ME PAGE is intended.

M-15 Maintenance Printout

General Description

The Telephony Server can provide a hard copy of real-time System Message via the System Printer.

Operating Procedure

Refer to the System Operations and Maintenance Manual.

Service Conditions

1. System Message printout: The system message will be printed automatically, and/or when the command is executed.

Interactions

1. The MAINTENANCE PRINTOUT as described will be assigned to Port 1.
2. For Automatic MAINTENANCE PRINTOUT via the PCPro, assign the PCPro port in System Data 1, Index 116 as data 42H. Assign the destination of System Messages in Index 39 as Port 0. Index 86, Bits 4 and 7 assigned as data "1".
3. Indexes 116 - 123, which were used for IOC port attribute data assignment, are assigned in the AIOC command.

Programming

STEP 1: **ASYD** - System Data 1, Index 38, Bits 0-7. Each Bit represents a corresponding I/O port. Designate by programming which port will accommodate a printer. Port 0, Bit 0 is always an PCPro.

System Data 1, Index 39, Bits 1-7. Each Bit represents a corresponding I/O port. Designate the destination of System Messages by programming which port will receive the Maintenance Printout. Port 0, Bit 0 is always a PCPro.

System Data 1, Index 86, Bit 4. Fault Information automatic printout of System Messages? 0/1: No/Yes.

System Data 1, Index 86, Bit 7. Are simplified or detailed System Messages required? 0/1: Simplified/Detailed.

System Data 1, Index 117-123. Each Index represents an I/O port. Program the following Bits:

0, 1, and 2.

001: Printer*

010: PCPro

Bit 7: Password Function 0/1:-/In Service.

STEP 2: **AIOC** - Assign SYMSCAN (System Message Scan Service) as "In Service" for all ports requiring Automatic System Message Printout. (Assign the data for IOC Port Number 0 when using LAN connection.)

Note: For a System Message Printer, assign these Bits only.

M-16 MFC - Incoming Call

General Description

This service feature allows an MFC incoming call from an MFC trunk to be terminated to a station or an Attendant.

Note: This feature is not available in North America and Australia.

Operating Procedure

No manual operation is required.

Service Conditions

1. This service feature supports the following types of MFC signaling system:
   a.) CCITT-R2 (NEC Standard) Type
   b.) Australia Type
   c.) Thailand Type
   d.) Venezuela Type
   e.) Korea Type
   f.) Singapore/Malaysia Type
   g.) Argentina/Philippines Type
   h.) Brazil Type (digital/analog)
   i.) Chile Type
   j.) Columbia Type
   k.) Mexico Type
   l.) Indonesia Type
   m.) India Type

2. DID incoming calls from MFC trunks can terminate to a station (DTMF, DP and Dterm) or an Attendant.
3. When calling the Attendant Console from MFC trunks, LISTED DIRECTORY NUMBER (LDN) [L-24] is dialed.
4. In case of TRUNK ANSWER FROM ANY STATION (TAS) [T-8] connection, if the called party is not restricted to TAS connection even if it is not registered in TAS, the backward signal “Called Party Idle” will be transmitted to the calling party.

5. When the called party is idle, the incoming call terminates as follows, according to the Group-II signals;
   For CCITT-R2, Korea, Australia, Thailand and Singapore Type

MFC SIGNAL TYPE	GROUP-2 SIGNAL	ROUTE DATA	SYSTEM OPERATION
CCITT-R2 (NEC Standard) Korea	Not Received	0C=0	G-A=6 is transmitted.
	Received	0C=1	G-B=6 transmitted. Call is terminated to the dialed station.

MFC SIGNAL TYPE	GROUP-2 SIGNAL	ROUTE DATA	SYSTEM OPERATION
Australia	1 (Operator) 8 (Overseas Operator)	TR=0	Call is terminated to the Attendant Console.
		TR=1	Call is terminated to the dialed station. **Note 1**
	Except 1 and 8	TR=0 or 1	
Thailand	1 (Toll Operator) 8 (International Operator) 9 (Data Terminal) 10 (Immediate Charge Info)	TR=0	Call is terminated to the Attendant Console.
		TR=1	Call is terminated to the dialed station. **Note 1**
	Except 1,8, 9 and 10	TR=0 or 1	
Singapore	1 (Toll Operator) 7 (Line Test Desk) 8 (International Operator) 9 (Call from Transit Exchange)	TR=0	Call is terminated to the Attendant Console.
		TR=1	Call is terminated to the dialed station. **Note 1**
	Except 1, 7, 8 and 9	TR=0 or 1	

Note 1: When the dialed station is in a STATION HUNTING [S-7, S-8, S-9] group or has set CALL FORWARDING [C-2, C-3, C-5, C-28, C-60, C-74], the incoming call follows the operations specified in each feature.

For Argentina, Chile, Columbia, Mexico, Venezuela, and India Type

When the called party is idle, the incoming call terminates to the dialed party, regardless of the Group-II signals. However, in case the dialed station belongs to a hunting group or has set Call Forwarding, the call follows the operation of each service.

For Brazil Type only

When the called party is idle and the Group-II signal is 1, 2, 4, or 6, the incoming call terminates to the dialed party. However, in case the dialed station belongs to a hunting group or has set Call Forwarding, the call follows the operation of each service. If the Group-II signal is 3, 5, 7, 8, 9, or 10, the call terminates to the Attendant Console in day mode, and to the night station in night mode, regardless of the dialed number.

For Venezuela Type only

When the Group-B signal is 1 (“Subscriber’s line free - change call party release”), the call is released when the calling party releases.

6. The Group-B signals to be sent out according to the called party’s status are as follows:

MFC SIGNAL TYPE / STATUS	CCITT-R2	AUST-RALIA	THAI-LAND	VENE-ZULEA	KOREA	SINGA-PORE/ Malaysia	ARGEN-TINA	CHILE	COLUM-BIA	MEXICO	INDO-NESIA	BRAZIL	INDIA
Idle (Charge)	6	1	1	6	6	1	6	6	6	1	1	1	6

MFC SIGNAL TYPE / STATUS	CCITT-R2	AUST-RALIA	THAI-LAND	VENE-ZULEA	KOREA	SINGA-PORE/ Malaysia	ARGEN-TINA	CHILE	COLUM-BIA	MEXICO	INDO-NESIA	BRAZIL	INDIA
Idle (Charge)-Change Calling Party Release	6	6	1	1	1	1	6	6	6	1	1	6	6
Idle (No Charge)	7	5	5	7	7	1	7	7	7	5	5	5	7
Busy	3	2	2	3	3	2	3	3	3	2	2	2	3
Dead Number/Dead Level	5	2	4	5	5	7	5	5	5	4	7	7	5
Congestion	4	2	4	4	4	4	4	4	4	4	4	4	4
Restriction	8	2	4	8	8	4	8	8	8	4	4	4	4
Lockout/Make Busy	8	2	4	8	8	4	8	8	8	4	8	4	4
LDN (Attendant)	6	1	1	6	6	1	6	6	6	1	1	1	6
Intercept to Attendant (called party restriction)	6	1	1	6	6	1	6	6	6	1	1	1	6
CALL FORWARDING-ALL CALLS [C-5] (to Attendant)	6	1	1	6	6	1	6	6	6	1	1	1	6
CALL FORWARDING-BUSY LINE [C-2] (to Attendant)	6	1	1	6	6	1	6	6	6	1	1	1	6
CALL FORWARDING-ALL CALLS-OUTSIDE [C-28]	6	1	1	6	6	1	6	6	6	1	1	1	6
Night Station	6	1	1	6	6	1	6	6	6	1	1	1	6
TAS Connection	6	1	1	6	6	1	6	6	6	1	1	1	6
Tandem Connection	6	1	1	6	6	1	6	6	6	1	–	1	6

7. Following service features are available for the incoming call from an MFC Trunk:
   - CALL FORWARDING-ALL CALLS [C-5]
   - CALL FORWARDING-BUSY LINE [C-2]
   - CALL FORWARDING-DON'T ANSWER [C-3]
   - CALL FORWARDING-INTERCEPT/ANNOUNCEMENT [C-25] (to a station/Attendant Console)
   - CALL FORWARDING-ALL CALLS-OUTSIDE (except for MFC to MFC) [C-28]
   - CALL FORWARDING-BUSY LINE-OUTSIDE (except for MFC to MFC) [C-60]
   - CALL PICKUP [C-7, 30]
   - CALL QUEUING [C-9]
   - CALL WAITING - TERMINATING [C-12]
   - DISTINCTIVE RINGING [D-10]
   - IMMEDIATE RINGING [I-1]
   - INCOMING CALL IDENTIFICATION [I-2]
   - LDN-NIGHT CONNECTION [L-28]
   - Multi Numbering
   - NIGHT CONNECTION [N-1, 2]
   - PEG COUNT [P-7]
   - REMOTE ACCESS TO SYSTEM [R-2]
   - STATION HUNTING-CIRCULAR [S-7]
   - STATION HUNTING-TERMINAL [S-9]

- STATION HUNTING-SECRETARIAL [S-8]
- TRUNK ANSWER FROM ANY STATION (TAS) [T-8]
- TENANT SERVICE [T-12]
- UNIFORM CALL DISTRIBUTION (UCD) [U-1]
- UNIFORM NUMBERING PLAN [U-4]

8. Optional hardware (MFC Register) is required for this service feature.
9. Incoming calls from the MFC trunks can terminate to the ACD pilot station. The call is handled in the same manner as the ACD incoming call from C.O. line/tie line.
10. When the MFC trunk call terminates to the ACD pilot station, the call can be transferred to the MFC trunk only via the transfer operation of the ACD position Attendant. (The call cannot be transferred to the MFC trunk by CCV programming.)
11. Malaysia Type only
    The services in Malaysia are based on the MFC signaling system in Singapore. When the system receives GII-7 signal from the MFC public exchange, the MFC DID incoming call terminates to the station designated by the GI Signal.

12. For Brazil Type only
    When the DID incoming call from the MFC trunk encounters called station busy, restricted, in make busy state, etc., the call can be transferred to the Attendant Console (when the system is in day mode) or the Night Transfer Station (night mode) in the same tenant as the called station, according to the data assignment. See the table for details.

Called Station Status / ARTD Command	Not Activate Call Transfer to Attendant Console/Night Station		Activate Call Transfer to Attendant Console/Night Station	
	TR=0	TR=1	TR=0	TR=1
Idle (No answer)	System sends B1 signal. CALL FORWARDING-DON'T ANSWER [C-3] to the Attendant Console or the Night Station in the same tenant as the incoming trunk.	System sends B1 signal. Continuous ringing	System sends B1 signal. CALL FORWARDING-DON'T ANSWER [C-3] to the Attendant Console or the Night Station in the same tenant as the called station.	System sends B1 signal. Continuous ringing
Busy	System sends B1 signal. CALL FORWARDING-BUSY LINE [C-2] to the Attendant Console or the Night Station in the same tenant as the incoming trunk.	System sends B2 signal. Caller receives Busy Tone.	System sends B1 signal. CALL FORWARDING-BUSY LINE [C-2] to the Attendant Console or the Night Station in the same tenant as the called station.	System sends B2 signal. Caller receives Busy Tone.

Called Station Status / ARTD Command	Not Activate Call Transfer to Attendant Console/Night Station		Activate Call Transfer to Attendant Console/Night Station	
	TR=0	TR=1	TR=0	TR=1
Restricted from connection/In make busy state	System sends B1 signal. CALL FORWARDING INTERCEPT/ ANNOUNCEMENT [C-75] to the Attendant Console or the Night Station in the same tenant as the incoming trunk.	System sends B4 signal. Caller receives Reorder Tone.	System sends B1 signal. CALL FORWARDING INTERCEPT/ ANNOUNCEMENT [C-75] to the Attendant Console or the Night Station in the same tenant as the called station.	System sends B4 signal. Caller receives Reorder Tone.
Unallocated number (station not assigned)	System sends B1 signal. CALL FORWARDING INTERCEPT/ ANNOUNCEMENT [C-75] to the Attendant Console or the Night Station in the same tenant as the incoming trunk.	System sends B4 signal. Caller receives Reorder Tone.	System sends B1 signal. CALL FORWARDING INTERCEPT/ ANNOUNCEMENT [C-75] to the Attendant Console or the Night Station in the same tenant as the called station.	System sends B7 signal. Caller receives Reorder Tone.

Called Station Status / ARTD Command	Not Activate Call Transfer to Attendant Console/Night Station		Activate Call Transfer to Attendant Console/Night Station	
	TR=0	TR=1	TR=0	TR=1
Unallocated number (station not assigned)	System sends B4 signal. Caller receives Reorder Tone.	System sends B4 signal. Caller receives Reorder Tone.	System sends B1 signal. CALL FORWARDING INTERCEPT/ ANNOUNCEMENT [C-75] to the Attendant Console or the Night Station in the same tenant as the called station.	System sends B7 signal. Caller receives Reorder Tone.
Night Station busy	System sends B2 signal. Caller receives Busy Tone.	System sends B2 signal. Caller receives Busy Tone.	System sends B1 signal. The call is placed in a queue until the Night Station becomes idle.	Return to the status before Call Forwarding was activated. Caller receives Busy Tone.

Called Station Status / ARTD Command	Not Activate Call Transfer to Attendant Console/Night Station		Activate Call Transfer to Attendant Console/Night Station	
	TR=0	TR=1	TR=0	TR=1
GII-3, 5, 7~10 signal received	System sends B1 signal. CALL FORWARDING INTERCEPT/ ANNOUNCEMENT [C-75] to the Attendant Console or the Night Station in the same tenant as the incoming trunk.	Ring the station after sending B1 signal, irrespective of the G-II signal received.	System sends B1 signal. CALL FORWARDING INTERCEPT/ ANNOUNCEMENT [C-75] to the Attendant Console or the Night Station in the same tenant as the called station.	Ring the station after sending B1 signal, irrespective of the G-II signal received.

This service has priority over the following Night Transfer services, in the following order.

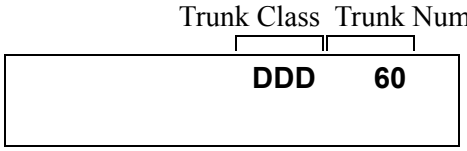
a.) NIGHT CONNECTION-FLEXIBLE [N-2]
b.) LDN NIGHT CONNECTION [L-28]
c.) NIGHT CONNECTION OUTSIDE-SYSTEM [N-17]
d.) TRUNK ANSWER FROM ANY STATION (TAS) [T-8]
e.) BUSY TONE CONNECTION

13. For all the Types except for Singapore/Malaysia Type.
When an MFC incoming call terminates to an OAI terminal (Dterm), the ANI (Automatic Number Identification) information received from the opposite office is sent to the OAI computer via Interface Processor by SMFN facility.
    a.) The ANI information is output when an MFC incoming call is terminated to the OAI terminal, and when the terminal user answers the call.
    b.) The calling number (max. 24 digits) is output as the ANI information.
    c.) When the ANI information is not requested, space (blank) is output to the ANI information area in the SMFN facility format.

14. For all the Types except for Singapore/Malaysia Type.
When an MFC DID incoming call terminates to the Attendant or a UCD station, the ANI (Automatic Number Identification) information received from the opposite office is output to MCI (Message Center Interface). The ANI information consists of category (1 digit) and the calling number (max. 31 digits). When the ANI information is not received, space (blank) is output to the ANI information area.

15. For all the Types except for Singapore/Malaysia Type.
When an MFC DID incoming call is terminated, the system can request the ANI (Automatic Number Identification) information. The ANI information consists of category (1 digit) and the calling number (max. 31 digits). The received ANI number is displayed on the Dterm LCD and output to the SMDR message.
    a.) Display on the Dterm LCD
    -Only the calling number is displayed.

    -When the ANI information is not received, trunk class and number are displayed instead.



In case of the Dterm with 2 or 3-line LCD, the information is displayed on the upper line.

b.) SMDR message
- Both category (1 digit) and the calling number (31 digits) are output.
- When the ANI Information is not received, no data is output to the ANI information area.

c.) NAME DISPLAY-SYSTEM-CCIS [N-37] is not available.

16. For India Type only
Trunk release may be either "Calling party Release" or "First Party Release", according to the route data of the incoming route.

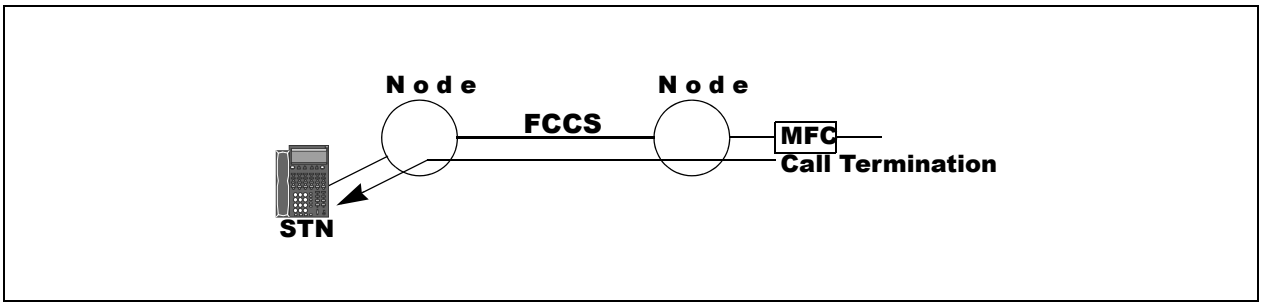
17. For India Type only
The ANI (Automatic Number Identification) information may be requested as follows:
a.) The ANI information is requested on a route basis.
Maximum 30 digits can be requested.

18. For Brazil Type only
When a collect call terminates to a station which is restricted from receiving the collect calls via MFC trunk, the signals notifying the restriction are sent to the office. There are two ways in activating this service, depending on the System Data, as follows:
a.) The backward signal "collect call blocking" is returned to the office.
b.) The signal "collect call blocking" is returned when the called station user answers the call. Reorder Tone is sent to the user.

For more details, refer to the System Data Design Manual.

19. "*" and "#" are available for the Group I (G-I) signal.

FCCS Conditions



a.) Both MFC Receiver and MFC Sender must be accommodated in each node.
b.) When a call is terminated from an MFC trunk in Node B to a station in Node A, MFC Receiver accommodated in Node B is used.
c.) When an MFC incoming call is terminated from an MFC trunk in Node B to a station in Node A, if the connection trunk cannot be used, the backward signal "Congestion" is sent.
d.) Be sure to assign the same data of MFC signaling system for each node.

1. Even if the Tenant service is provided between nodes in the FCCS network, an MFC DID call to an Attendant Console cannot be routed to the Attendant Console in the different node from the terminating trunk.
2. The ANI information is comprised as follows.
   For Indonesia Type only
   a.) When the assigned telephone number is 7 digits or less:
   ANI = Office Code + Telephone Number
   b.) When the assigned telephone number is 8 digits or more/not assigned.
   ANI = Office Code + Physical Station Number

   For the other countries
   a.) When MFC DOD call is originated from an Attendant Console:
   ANI = Office Code + LDN (Listed Directory Number)
   b.) When a call from an ACIS line is routed to an MFC trunk via tandem connection:
   ANI = Office Code of Tandem Office + LDN of Tandem Office
   c.) When a call from an CCIS line is routed to an MFC trunk via tandem connection:
   ANI = Office Code of Originating Office + Telephone Number/Individual ATT Number of Originating Office
   d.) When the system data to send LDN is assigned:
   ANI = Office Code + LDN
   e.) When the system data to send Physical Station Number is assigned:
   - When the assigned telephone number is 7 digits or less:
   ANI = Office Code + Telephone Number
   - When the assigned telephone number is 8 digits or more/not assigned:
   ANI = Office Code + Physical Station Number

NEC Standard (CCITT R-2) Type

Signal	Forward signal		Backward signal	
	G-I	G-II	G-A	G-B
1	Digit “1”	subscriber without priority	send next digit (n+1)	
2	Digit “2”	subscriber with priority	send last but one digit (n-1)	~~send special information tone~~
3	Digit “3”	maintenance equipment	address complete change-over to reception of G-II	subscriber’s line busy
4	Digit “4”		congestion in the national network	congestion, encountered after changeover from G-A to G-B
5	Digit “5”	operator	calling party category	unallocated number
6	Digit “6”	data transmission	address complete change set up speech condition	subscriber’s line free (charge)
7	Digit “7”	subscriber or operator without forward transfer facility	send last but two digits (n-2)	subscriber’s line free (no charge)
8	Digit “8”	data transmission	send last but three digits (n-3)	subscriber’s line out of order
9	Digit “9”	subscriber with priority		
10	Digit “0”	operator with forward transfer facility		
11	Digit “*”			
12	Digit “#”			
13 15				

abcd Signals transmitted and received

abcd Signals received

~~abcd~~ Signals not handled

Australia (Telecom) Type

Signal	Forward signal		Backward signal	
	G-I	G-II	G-A	G-B
1	Digit “1”	operator	send next digit (n+1)	called party idle
2	Digit “2”	subscriber or coin telephone	send first digit	called party busy
3	Digit “3”	test position	address complete change-over to reception of G-B	~~no line out before answer~~
4	Digit “4”	interception center or remote operation center	~~start DP 1st digit~~	~~congestion~~
5	Digit “5”	coin telephone	~~start DP 2nd digit~~	called party idle (no charge)
6	Digit “6”	data telephone	~~start DP 3rd digit~~	called party idle control NO time out before answer
7	Digit “7”	overseas subscriber	~~start DP 4th digit~~	~~called party idle, called party control no charge~~
8	Digit “8”	overseas operators	~~start DP 5th digit~~	~~called party interception~~
9	Digit “9”	subscriber	send calling party access info	~~send calling party access info~~
10	Digit “0”	used by public subscriber	~~send previous digit MFC~~	
11	Digit “*”			
12	Digit “#”			
13 15				

abcd Signals transmitted and received

abcd Signals received

~~abcd~~ Signals not handled

Thailand (Telecom) Type

Signal	Forward signal		Backward signal		
	G-I	G-II	G-A	G-B	G-C
1	Digit “1”	toll operator	send next digit (n+1)	called subscriber’s line free (charge)	send next digit of caller
2	Digit “2”	ordinary subscriber	send first digit of called number	called subscriber’s line busy	send first digit of called number
3	Digit “3”	local coin box	send nature of caller and G-B	~~called subscriber’s line received~~	
4	Digit “4”		congestion	congestion	~~congestion~~
5	Digit “5”	standard coin box		called subscriber’s line free (no charge)	send next digit of called number
6	Digit “6”	test equipment	send number of caller and change G-C	~~last party release~~	
7	Digit “7”	line test desk			
8	Digit “8”	intercepted operator			
9	Digit “9”	reserve for data terminal			
10	Digit “0”	immediate charge information service			
11	Digit “*”				
12	Digit “#”				
13 15					

abcd Signals transmitted and received

abcd Signals received

~~abcd~~ Signals not handled

Venezuela Type

Signal	Forward signal		Backward signal	
	G-I	G-II	G-A	G-B
1	Digit “1”	standard subscriber	send next digit (n+1)	subscriber’s line free (change call party release)
2	Digit “2”	preferential subscriber	send last but one digit (n-1)	~~extension number changed~~
3	Digit “3”	test subscriber	address complete, change-over to reception of G-B signal	subscriber’s line busy
4	Digit “4”	coin phone	congestion	congestion
5	Digit “5”	attendant	send category of calling subscriber	unallocated number
6	Digit “6”	data transmission	address complete, change setup speech condition	subscriber’s line free (with charge)
7	Digit “7”		send last but two digits (n-2)	subscriber’s line free (without charge)
8	Digit “8”		send last but three digits (n-3)	subscriber’s line out of order
9	Digit “9”		send originating subscriber number	
10	Digit “0”			~~data transmitting equipment~~
11	Digit “*”			
12	Digit “#”			
11				
12				
15	end of pulsing			

abcd Signals transmitted and received

abcd Signals received

~~abcd~~ Signals not handled

Korea (KTA) Type

Signal	Forward signal		Backward signal	
	G-I	G-II	G-A	G-B
1	Digit “1”	subscriber without priority	send next digit (n+1)	subscriber’s line free (called party release)
2	Digit “2”	subscriber with priority	send last but one digit (n-1)	~~subscriber number change~~
3	Digit “3”	line test	address complete change-over to reception of Group	subscriber’s line busy
4	Digit “4”	coin box	congestion in the national network	congestion, encountered after changeover from G-A to G-B
5	Digit “5”	operator	send calling party class	unallocated number
6	Digit “6”	data transmission	address complete, change set up speech condition	subscriber’s line free (charge)
7	Digit “7”	international call	send last but two digits (n-2)	subscriber’s line free (no charge)
8	Digit “8”	international call	send last but three digits (n-3)	subscriber’s line out of order or restrict
9	Digit “9”	international call	~~send first digit of called number~~	
10	Digit “0”	international call transfer facility		
11	Digit “*”			
12	Digit “#”			
15	end of origi-nating subscriber			

abcd Signals transmitted and received

abcd Signals received

~~abcd~~ Signals not handled

Singapore (Telecom) Type / Malaysia (Telecom) Type

Signal	Forward signal		Backward signal	
	G-I	G-II	G-A	G-B
1	Digit “1”	operator	send next digit (n+1)	called subscriber’s line free
2	Digit “2”	subscriber		called subscriber’s line busy
3	Digit “3”	local call	address complete change-over to reception of G-B	
4	Digit “4”		congestion	congestion
5	Digit “5”	international call		
6	Digit “6”	test equipment		
7	Digit “7”	line test desk		unallocated number
8	Digit “8”	international operator		
9	Digit “9”	called from transit exchange		
10	Digit “0”			
11	Digit “*”			
12	Digit “#”			
13 15				

abcd Signals transmitted and received

abcd Signals received

Argentina Type

Signal	Forward signal		Backward signal	
	G-I/III	G-II	G-A	G-B
1	Digit “1”	subscribers without priority	send next digit (n+1)	
2	Digit “2”	subscribers with priority	send last but one digit (n-1)	send recorded messages
3	Digit “3”	maintenance equipment	address complete, change-over to G-B signal	subscriber’s line busy
4	Digit “4”	coin phone	congestion	congestion
5	Digit “5”	attendant	send category of calling subscriber	unallocated number
6	Digit “6”	data transmission	address complete, change setup speech conditions	subscriber’s line free with charge

Argentina Type

Signal	Forward signal		Backward signal	
	G-I/III	G-II	G-A	G-B
7	Digit “7”		send last but two digits (n-2)	subscriber’s line free without charge
8	Digit “8”		send last but three digits (n-3)	subscriber’s line out of order
9	Digit “9”		send last digit (n)	
10	Digit “0”		send first digit	
11	Digit “*”			
12	Digit “#”			
11	~~access to test equipment~~			
14				
15	end of pulsing			

abcd Signals transmitted and received

abcd Signals received

~~abcd~~ Signals not handled

Brazil Type

Signal	Forward signal		Backward signal	
	G-I	G-II	G-A	G-B
1	Digit “1”	ordinary subscriber	send next digit (n+1)	called party idle with charging
2	Digit “2”	subscriber with special charging		called party busy
3	Digit “3”	maintenance equipment	address complete, change-over to reception of G-II signal	changed number collect call blocking
4	Digit “4”	local public call box	congestion	congestion
5	Digit “5”	operator	send category of calling subscriber	called party idle without charging
6	Digit “6”	data transmission		called party idle with charging; called party based release
7	Digit “7”	national public call box ordinary subscriber (international working)	send last but two digits (n-2)	unallocated number collect call blocking
8	Digit “8”	data transmission (international working) collect call	send last but three digits (n-3)	
9	Digit “9”	subscriber with priority (international working)	send last but one digit (n-1)	
10	Digit “0”	operator with forward transfer facility (international working)		
11	Digit “*”	call transfer		
12	Digit “#”			
13				
14				
15	end of pulsing			

abcd Signals transmitted and received

abcd Signals received

Chile Type

Signal	Forward signal		Backward signal	
	G-I/III	G-II	G-A	G-B
1	Digit “1”	subscriber without priority	send next digit (n+1)	
2	Digit “2”	subscriber with priority	send last but one digit (n-1)	~~send special information tone~~
3	Digit “3”	maintenance equipment	address complete, change to reception of G-B signal	subscriber’s line busy
4	Digit “4”	subscriber with remote billing	congestion in the national network	congestion, encountered after changeover from G-A signal to G-B signal
5	Digit “5”	operator	send calling subscriber’s category	unassigned number
6	Digit “6”	data transmission	address complete with change	unoccupied subscriber’s line, with billing (charge)
7	Digit “7”		send last but two digits (n-2)	unoccupied subscriber’s line, without billing (without charge)
8	Digit “8”		send last but three digits (n-3)	subscriber’s line out of order
9	Digit “9”			
10	Digit “0”			
11	Digit “*”	unidentifiable subscriber		
12	Digit “#”			
13				
14				
15	end of pulsing			

abcd Signals transmitted and received

abcd Signals received

~~abcd~~ Signals not handled

Columbia Type

Signal	Forward signal		Backward signal	
	G-I	G-II	G-A	G-B
1	Digit “1”	subscriber	send next digit (n+1)	~~put connection under control of incoming side~~
2	Digit “2”	calling party with priority	send last but one digit (n-1)	~~changed number (called line intercepted)~~
3	Digit “3”	maintenance equipment	send nature of caller and change to G-B signals	called subscriber’s line busy
4	Digit “4”		congestion	congestion
5	Digit “5”	operator	send nature (and identify) of caller	dead line or dead level
6	Digit “6”	data transmission	through connect speech path	called subscriber’s line free with metering (charge)
7	Digit “7”	subscriber for international service	send last digit but two (n-2)	called subscriber’s line free without metering (without charge)
8	Digit “8”		send last digit but three (n-3)	called line out of order
9	Digit “9”		~~nationally at disposal~~	
10	Digit “0”	operator for international service	~~nationally at disposal~~	
11	Digit “*”		~~send first digit of international code~~	
12	Digit “#”		~~send first digit of significant number~~	
13	~~access to main. equipment for national service~~		~~send first digit of international code of outgoing country~~	
14	~~insert echo suppressor~~		~~inquiry whether or not echo suppressor should be inserted~~	
15	end of pulsing			

abcd Signals transmitted and received

abcd Signals received

~~abcd~~ Signals not handled

Mexico Type

Signal	Forward signal			Backward signal		
	G-I	G-II	G-III	G-A	G-B	G-C
1	Digit “1”		Digit “1”	send next digit (n+1)	subscriber’s with charge	send G-III signal next digit
2	Digit “2”	ordinary subscriber	Digit “2”	send signal first digit	subscriber’s line busy	send G-I signal first digit and charge receipt G-A
3	Digit “3”	coin box telephone	Digit “3”	send signal of G-II, and change to reception of G-B signal		send G-II signal and charge receipt G-B
4	Digit “4”	time and cost	Digit “4”	congestion	line lock out	congestion
5	Digit “5”		Digit “5”		subscriber idle without charge	send G-I signal next digit and change receipt G-A
6	Digit “6”	maintenance equipment	Digit “6”	send group II-6 and change to receipt G-C signal		send G-I signal same digit and change digit receipt G-A
7	Digit “7”	identification of the No. 2 subscriber in party line	Digit “7”			
8	Digit “8”	identification of the No. 3 subscriber in party line	Digit “8”			
9	Digit “9”	identification of the No. 1 subscriber in party line	Digit “9”			
10	Digit “0”		Digit “0”			
11	Digit “*”					
12	Digit “#”					
15			End of digit			

abcd Signals transmitted and received

abcd Signals received

Indonesia Type

Signal	Forward signal			Backward signal	
	G-I	G-II	G-III	G-A	G-B
1	Digit “1”	national operator	Digit “1”	send next digit (n+1)	subscriber line free (charge)
2	Digit “2”	normal subscriber	Digit “2”	restart from beginning	subscriber busy
3	Digit “3”	local payphone	Digit “3”	address complete change to B signal	~~send special information tone~~
4	Digit “4”	international operator	Digit “4”	congestion	
5	Digit “5”	long distance payphone	Digit “5”	address complete set-up speech condition	subscriber line free (no charge)
6	Digit “6”	national test and maintenance equipment	Digit “6”	send calling party’s category/number	~~malicious call tracing~~
7	Digit “7”	international test equipment	Digit “7”		unallocated national number
8	Digit “8”	cress border operator	Digit “8”	restart with last but one digit (n-1)	line out of service
9	Digit “9”	international payphone	Digit “9”	restart with last but two digits (n-2)	
10	Digit “0”		Digit “0”		
11	Digit “*”				
12	Digit “#”				
13					
14					
15	end of available information		end of available information		

abcd Signals transmitted and received

abcd Signals received

~~abcd~~ Signals not handled

India Type

Signal	Forward signal		Backward signal	
	G-I	G-II	G-A	G-B
1	Digit “1”	ordinary subscriber	send next digit (n+1)	
2	Digit “2”	priority subscriber	restart	changed number
3	Digit “3”		change over to reception of G-B signal	called line busy
4	Digit “4”		calling line identification	congestion
5	Digit “5”	operator	send category of the calling subscriber	unallocated number
6	Digit “6”		switch-through the speech path	normal subscriber, free, with metering
7	Digit “7”		send last but two digits (n-2)	called line free, but not metered
8	Digit “8”		send last but three digits (n-3)	
9	Digit “9”		send last but one digit (n-1)	
10	Digit “0”			
11	Digit “*”			
12	Digit “#”			
13				
14				
15				

abcd Signals transmitted and received

abcd Signals received

Programming

<<For MFC Incoming Connection (without Tandem Connection)>>

STEP 1: **ASYD -** System 1, Index 125, Bits 0 through 3. PAD control of the MFC Sender 0dB--14dB.
Bits 4 through 7. MFC Signaling System CCITT MFC-R2, Venezuela, Australia, Singapore, Thailand.
System 3, Index 10, Bits 0 through 3. Reception sensitivity level of the MFC Register - 28dBm-43dBm.

STEP 2: **ARTD -** Assign the following data to the route of the MFC line.
ONSG=0 and INSG=10
Assigns the parameter OC (originating category) depending of the use of the G-A signal of the backward signal.
OC=0: Originating category is not required.
OC=1: Backward signal G-A=3 (originating category request) is sent after receiving necessary selecting signal.

Note: To change the data for existing route (RT), initialize or reboot the trunk device such as circuit card, MG, or VS32.

STEP 3: **ATRK -** MFC line - Assigns the LENS data of each trunk.
MFC Register/Sender - Assigns the MFC Register/Sender for Incoming (RT=916) to the LENS of the PA-4MFCA circuit card. LV=0-3 can be assigned.

STEP 4: **ARSC -** Releases the connection restriction on the route of the MFC line and station.

STEP 5: **ASFC -** Assigns the backward signal (G-B) against the forward signal (G-II) for each SFC when the called party is idle.
SFI=59: No Charge
SFI=60: Called Party Release
Release between the Assignment of SFI=59,60 and the Backward Signal.

SPECIFICATION	SFI=59	SFI=60	BACKWARD SIGNAL	REMARKS
CCITT MFC-R2	0	0	G-B=6	
	1	0	G-B=7	
VENEZUELA	0	0	G-B=6	
AUSTRALIA	0	0	G-B=1	
	0	1	G-B=6	
	1	0	G-B=5	
SINGAPORE	0	0	G-B=1	
THAILAND	0	0	G-B=1	

Note: Combination of SFI=59 and 60 which are not in the table in each specification must be assigned.

<<For MFC Outgoing/Incoming Connection or MFC Incoming Connection (with Tandem Connection)>>

STEP 1: **ASYD -** System 1, Index 125, Bits 0 through 3. PAD control of the MFC Sender 0dB--14dB.
Bits 4 through 7. MFC Signaling System CCITT MFC-R2, Venezuela, Australia, Singapore, Thailand.
System 3, Index 10, Bits 0 through 3. Reception sensitivity level of the MFC Register - 28dBm-43dBm.

STEP 2: **ANPD -** Assigns the first digit of the MFC line access code.

STEP 3: **ASPA -** Assigns the LCR access code by providing a dummy route.

STEP 4: **AMND -** Assigns the maximum number of necessary digits to the Dial (Area/Office) Code.

STEP 5: **ARNP -** Assigns the access code to the route number. When multiple access code are present by alternative routing, set one of them.
MFC line - Assign the following data, depending on the Trunk type, as shown below.

Trunk Type	ONSG	INSG
OGT	10	0
ICT	0	10
BWT	10	10

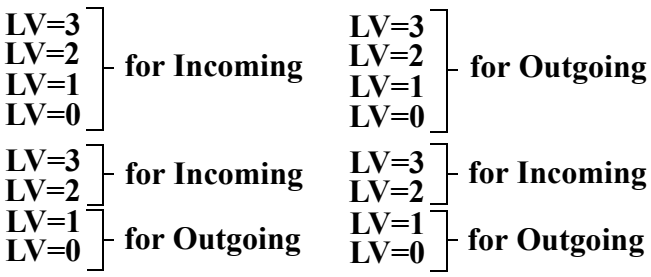
Assign the parameter OC (originating category) depending on the use of the G-A signal of the backward signal.
OC=0: Originating category is not required.
OC=1: For terminating office, backward signal G-A=3 (originating category request) is sent after receiving necessary selecting signal. For originating office, forward signal G-II is sent after receiving backward signal G-A=3 (originating category request).

dummy route - Assign 1 at all times for the AC parameter of CDN 13. The route data of the dummy route is TCL=4, AC=1 and L/T=1. For the others, Assigns 0 in all cases.

STEP 6: **ATRK -** MFC line - Assigns the LENS data of each trunk.
MFC - Register/Sender - Assigns the following data to the LENS of the PA-4MFCA circuit card.
Note 2

RT=916: MFC Register/Sender for Incoming
RT=917: MFC Register/Sender for Outgoing

Note 2: As the use of the 4 circuit on the PA-4MFCA circuit card, only the following 4 patterns are available.



STEP 7: **AFRS -** Assigns the number Pattern Code (NPC) and Outgoing route selection pattern (OPR) to the dummy route.

STEP 8: **AOPR -** Assigns the route number including the alternative route to the outgoing route selection pattern (OPR). Assigns the number of skipped digits and pattern number location (PNL) to each route.

STEP 9: **ARSC -** Releases the dummy route and station connection restriction.

Releases the connection restriction on the route of the MFC line and station.

STEP 10: **ASFC -** Assigns the backward signal (G-B) against the forward signal (G-II) for each SFC when the called party is idle.

SFI=59: No Charge

SFI=60: Called Party Release

Relation between the Assignment of SFI=59, 60 and the Backward Signal.

SPECIFICATION	SFI = 59	SFI = 60	BACKWARD SIGNAL	REMARKS
CCITT MFC-R2	0	1	G-B=6	
	1	0	G-B=7	
VENEZUELA	0	0	G-B=6	
AUSTRALIA	0	0	G-B=1	
	0	1	G-B=6	
	1	0	G-B=5	
SINGAPORE	0	0	G-B=1	
THAILAND	0	0	G-B=1	

Note: Combination of SFI=59 and 60 which are not in the table in each specification must not be assigned.

STEP 11: **ARRC -** For Tandem Connection Service, Release incoming route and outgoing route connection restriction.

M-17 MFC - Outgoing Call

General Description

This service feature allows a station or an attendant to access a Central Office via MFC trunks.

Note: This feature is not available in North America and Australia.

Operating Procedure

No manual operation is required.

Service Conditions

1. This service feature supports the following types of MFC signaling system:
   a.) CCITT-R2 (NEC Standard) Type
   b.) Thailand Type
   c.) Venezuela Type
   d.) Korea Type
   e.) Argentina/Philippines Type
   f.) Mexico Type
   g.) Brazil Type (digital/analog)
   h.) Chile Type
   i.) Columbia Type
   j.) Indonesia Type
   k.) India Type
2. The numbering plan for outgoing calls via MFC trunks uses LCR or LCRS system.
3. Terminals supported for this service feature are DP telephone, DTMF telephone, Dterm and ATTENDANT CONSOLEs [A-3]. (For Indonesia Type, ATTENDANT CONSOLEs are not supported.)
4. This service feature is effective in the following status:
   a.) Direct dialing by a station (For Indonesia Type, available only in this status.)
   b.) CONSULTATION HOLD [C-17] on station
   c.) CONSULTATION HOLD [C-17] on trunk
   d.) Tandem Connection
   e.) Direct dialing by an Attendant
   f.) INDIVIDUAL TRUNK ACCESS [I-4] (not available using LCRS system)
   g.) NON-DELAY OPERATION [N-3]
   h.) Trunk to Trunk Connection via ATTENDANT CONSOLE [A-3]
5. CALL FORWARDING - DON'T ANSWER - OUTSIDE [C-74] to MFC trunk is not available (except for India Type).
6. Outgoing calls to MFC trunks by HOT LINE - OUTSIDE [H-9] are not available (except for India Type).
7. Outgoing calls to MFC trunks by TRUNK LINE APPEARANCE [T-23] are not available (except for India Type).
8. Outgoing calls to MFC trunks by OUTGOING TRUNK QUEUING - ATTENDANT [O-16] are not available (except for India Type).

9. DID calls from MFC trunks cannot be forwarded to MFC trunks by CALL FORWARDING - ALL CALLS - OUTSIDE [C-28] or CALL FORWARDING - BUSY LINE - OUTSIDE [C-60].
10. For CCITT-R2 and Korea Type
    Group-II signal from the Telephony Server is fixed to be "2 (Subscriber with Priority)."
    For Argentina, Brazil, Chile, Columbia, Venezuela, and India Type
    Group-II signal from the Telephony Server may be either 1 or 2, according to the Service Feature Restriction Class of the calling station.
    a.) Group-II = 1 (Subscriber without Priority)
    b.) Group-II = 2 (Subscriber with Priority)
11. For CCITT-R2, Korea, Argentina, Chile, Columbia, Venezuela, Mexico, Brazil and India Type
    When a Group-B signal = "7 (Subscriber's Line Free, No Charge)" is received, the call is not recorded in STATION MESSAGE DETAIL RECORDING [S-10].
12. Even if a Group-B signal = Subscriber's Line Busy is received, the calling party hears Busy Tone.
13. Even if a Group-B signal = "1 (Change Calling Party Release)" is received, trunk release pattern follows the system data programming.
    For Argentina, Philippines, Chile, Columbia, Korea, and Venezuela Type
    Even if a Group-B signal = "1 (Change Calling Party Release)" is received, trunk release may be either "Calling Party Release" or "First Party Release", according to the route data of the originating route.
14. For Argentina, Philippines, Brazil, Chile, Columbia, Mexico, and Venezuela Type
    When the A-1 ("Send Next Digit") signal is received as the backward signal to the last digit of the Group-1 signal, following conditions occur according to the system data.
    a.) End of Digit "G-I = 15" is sent out.
    b.) End of Digit "G-I = 15" is not sent out, but waits for the backward signal.
15. Following service features are available for the outgoing call to an MFC trunk:

    CALL HOLD [C-6]

    CALL FORWARDING - ALL CALLS - OUTSIDE [C-28]
    (except for forwarding to MFC trunks)

    CALL FORWARDING - BUSY LINE - OUTSIDE [C-60]
    (except for forwarding to MFC trunks)

    INCOMING CENTRAL OFFICE CALL TO TIE LINE CONNECTION [I-3]

    INDIVIDUAL TRUNK ACCESS [I-4]

    LAST NUMBER CALL [L-6, L-6D]

    LEAST COST ROUTING (LCR) - 3/6 - DIGIT [L-5]

    OUTGOING TRUNK QUEUING [O-2]

    OFF-HOOK QUEUING [O-7]

    PEG COUNT [P-7]

RESTRICTION FROM OUTGOING CALLS [R-4]

SPEED CALLING - SYSTEM [S-3]

SPEED CALLING - GROUP [S-23]

SPEED CALLING - STATION [S-21]

STATION MESSAGE DETAIL RECORDING [S-10]

STATION MESSAGE DETAIL RECORDING SYSTEM - PRINTOUT [S-48]

TOLL DENIAL/TOLL DIVERSION [T-6]

TOLL RESTRICTION - 3/6 - DIGIT [T-7]

16. The following ANI (Automatic Number Identification) information is sent to the MFC trunks.
    a.) The contents of the information:
    1)Class (1 digit); sent as Calling Party Category (G-II)
    2)Calling Number (office code and telephone number, max. 12 digits).
    The office code is assigned on a tenant basis. For the telephone number, either of the following information can be selected on a tenant basis.
    - calling telephone number
    - listed directory number (LDN) assigned on a tenant basis

    For India Type only
    Class information is not sent.
    When Calling Number can be sent: "1" is sent.
    When Calling Number cannot be sent: "0" is sent

    b.) The information sent in case of the call origination from the Attendant is the listed directory number (LDN) assigned on a tenant basis.
    c.) As the Calling Number, the digits other than the number 0 to 9 will not be sent out (i.e., * and #).
    d.) It can be determined on a tenant basis whether to send the office code and the telephone number, or only the telephone number as the Calling Number.

17. The additional hardware (MFC Register) is required.

18. For India Type only
    Trunk release may be either "Calling Party Release" or "First Party Release", according to the route data of the originating route.
19. When LCRS system is used:
    a.) sending of "End of Digit":
    -when the A-3 ("address complete, changeover to reception of G-B") is received as the backward signal to the last digit of the G-I signal, the calling party class is sent out, and "End of Digit" is not sent out.
    -When the A-1 ("send next digit") is received as the backward signal to the last digit of the G-I

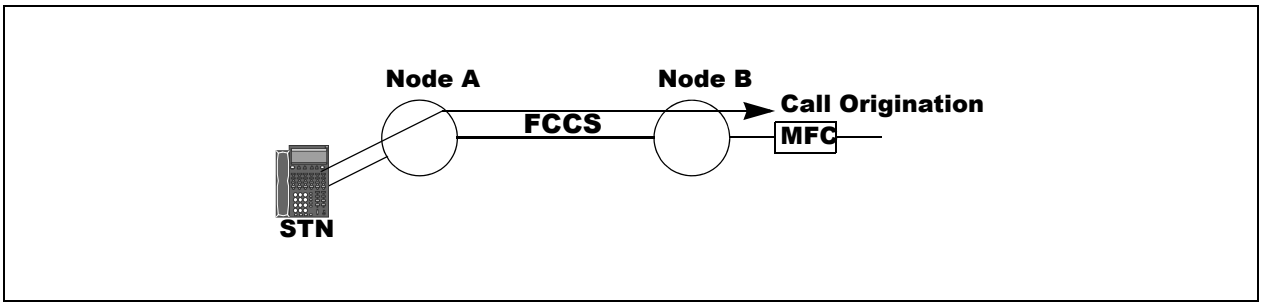
signal, “End of Digit” is sent out.

b.) sending of ANI information
-if ANI information is requested while sending the digits, the rest of the digits are sent out after the ANI information.
-if ANI information is requested while the digits are being received from the station, Attendant Console, or the incoming trunk, the rest of the digits are received after the ANI information.

20. For Indonesia Type only
In Indonesian network, the system is used as Satellite Office of LS. The system is in charge of shaded parts. Note that the system cannot be used as LS or TS.

21. “*” and “#” are available for the Group I (G-I) signal. Both “*” and “#” cannot be used for the “Timing Start”, if those digits are used as G-1 signal.

FCCS Conditions



a.) Both MFC Receiver and MFC Sender must be accommodated in each node.

b.) When an MFC DOD call is originated from a station in Node A through an MFC trunk in Node B via FCCS link, MFC Sender accommodated in Node A is used.

c.) Be sure to assign the same data of MFC signaling system for each node.

NEC Standard (CCITT R-2) Type

Signal	Forward signal		Backward signal	
	G-I	G-II	G-A	G-B
1	Digit "1"	subscriber without priority	send next digit (n+1)	
2	Digit "2"	subscriber with priority	send last but one digit (n-1)	~~send special information tone~~
3	Digit "3"	maintenance equipment	address complete change-over to reception of G-II	subscriber's line busy
4	Digit "4"		congestion in the national network	congestion, encountered after changeover from G-A to G-B
5	Digit "5"	operator	calling party category	unallocated number
6	Digit "6"	data transmission	address complete change set up speech condition	subscriber's line free (charge)
7	Digit "7"	subscriber or operator without forward transfer facility	send last but two digits (n-2)	subscriber's line free (no charge)
8	Digit "8"	data transmission	send last but three digits (n-3)	subscriber's line out of order
9	Digit "9"	subscriber with priority		
10	Digit "0"	operator with forward transfer facility		
11	Digit "*"			
12	Digit "#"			
13 15				

abcd Signals transmitted and received

abcd Signals received

~~abcd~~ Signals not handled

Australia (Telecom) Type

Signal	Forward signal		Backward signal	
	G-I	G-II	G-A	G-B
1	Digit “1”	operator	send next digit (n+1)	called party idle
2	Digit “2”	subscriber or coin telephone	send first digit	called party busy
3	Digit “3”	test position	address complete change-over to reception of G-B	~~no line out before answer~~
4	Digit “4”	interception center or remote operation center	~~start DP 1st digit~~	~~congestion~~
5	Digit “5”	coin telephone	~~start DP 2nd digit~~	called party idle (no charge)
6	Digit “6”	data telephone	~~start DP 3rd digit~~	called party idle control NO time out before answer
7	Digit “7”	overseas subscriber	~~start DP 4th digit~~	~~called party idle, called party control no charge~~
8	Digit “8”	overseas operators	~~start DP 5th digit~~	~~called party intereception~~
9	Digit “9”	subscriber	send calling party access info	~~send calling party access info~~
10	Digit “0”	used by public subscriber	~~send previous digit MFC~~	
11	Digit “*”			
12	Digit “#”			
13 15				

abcd Signals transmitted and received

abcd Signals received

~~abcd~~ Signals not handled

Thailand (Telecom) Type

Signal	Forward signal		Backward signal		
	G-I	G-II	G-A	G-B	G-C
1	Digit “1”	toll operator	send next digit (n+1)	called subscriber’s line free (charge)	send next digit of caller
2	Digit “2”	ordinary subscriber	send first digit of called number	called subscriber’s line busy	send first digit of called number
3	Digit “3”	local coin box	send nature of caller and G-B	~~called subscriber’s line received~~	
4	Digit “4”		congestion	congestion	~~congestion~~
5	Digit “5”	standard coin box		called subscriber’s line free (no charge)	send next digit of called number
6	Digit “6”	test equipment	send number of caller and change G-C	~~last party release~~	
7	Digit “7”	line test desk			
8	Digit “8”	intercepted operator			
9	Digit “9”	reserve for data terminal			
10	Digit “0”	immediate charge information service			
11	Digit “*”				
12	Digit “#”				
13 15					

abcd Signals transmitted and received

abcd Signals received

~~abcd~~ Signals not handled

Venezuela Type

Signal	Forward signal		Backward signal	
	G-I	G-II	G-A	G-B
1	Digit “1”	standard subscriber	send next digit (n+1)	subscriber’s line free (change call party release)
2	Digit “2”	preferential subscriber	send last but one digit (n-1)	~~extension number changed~~
3	Digit “3”	test subscriber	address complete, change-over to reception of G-B signal	subscriber’s line busy
4	Digit “4”	coin phone	congestion	congestion
5	Digit “5”	attendant	send category of calling subscriber	unallocated number
6	Digit “6”	data transmission	address complete, change setup speech condition	subscriber’s line free (with charge)
7	Digit “7”		send last but two digits (n-2)	subscriber’s line free (without charge)
8	Digit “8”		send last but three digits (n-3)	subscriber’s line out of order
9	Digit “9”		send originating subscriber number	
10	Digit “0”			~~data transmitting equipment~~
11	Digit “*”			
12	Digit “#”			
11				
12				
15	end of pulsing			

abcd Signals transmitted and received

abcd Signals received

~~abcd~~ Signals not handled

Korea (KTA) Type

Signal	Forward signal		Backward signal	
	G-I	G-II	G-A	G-B
1	Digit “1”	subscriber without priority	send next digit (n+1)	subscriber’s line free (called party release)
2	Digit “2”	subscriber with priority	send last but one digit (n-1)	~~subscriber number change~~
3	Digit “3”	line test	address complete change-over to reception of Group	subscriber’s line busy
4	Digit “4”	coin box	congestion in the national network	congestion, encountered after changeover from G-A to G-B
5	Digit “5”	operator	send calling party class	unallocated number
6	Digit “6”	data transmission	address complete, change set up speech condition	subscriber’s line free (charge)
7	Digit “7”	international call	send last but two digits (n-2)	subscriber’s line free (no charge)
8	Digit “8”	international call	send last but three digits (n-3)	subscriber’s line out of order or restrict
9	Digit “9”	international call	~~send first digit of called number~~	
10	Digit “0”	international call transfer facility		
11	Digit “*”			
12	Digit “#”			
15	end of origi-nating subscriber			

abcd Signals transmitted and received

abcd Signals received

~~abcd~~ Signals not handled

Argentina/Philippines Type

Signal	Forward signal		Backward signal	
	G-I/III	G-II	G-A	G-B
1	Digit “1”	subscribers without priority	send next digit (n+1)	
2	Digit “2”	subscribers with priority	send last but one digit (n-1)	send recorded messages
3	Digit “3”	maintenance equipment	address complete, change-over to G-B signal	subscriber’s line busy
4	Digit “4”	coin phone	congestion	congestion
5	Digit “5”	attendant	send category of calling subscriber	unallocated number
6	Digit “6”	data transmission	address complete, change setup speech conditions	subscriber’s line free with charge
7	Digit “7”		send last but two digits (n-2)	subscriber’s line free with-out charge
8	Digit “8”		send last but three digits (n-3)	subscriber’s line out of order
9	Digit “9”		send last digit (n)	
10	Digit “0”		send first digit	
11	Digit “*”			
12	Digit “#”			
11	~~access to test equipment~~			
14				
15	end of pulsing			

abcd Signals transmitted and received

abcd Signals received

~~abcd~~ Signals not handled

Mexico Type

Signal	Forward signal			Backward signal		
	G-I	G-II	G-III	G-A	G-B	G-C
1	Digit “1”		Digit “1”	send next digit (n+1)	subscriber’s with charge	send G-III signal next digit
2	Digit “2”	ordinary subscriber	Digit “2”	send signal first digit	subscriber’s line busy	send G-I signal first digit and charge receipt G-A
3	Digit “3”	coin box telephone	Digit “3”	send signal of G-II, and change to reception of G-B signal		send G-II signal and charge receipt G-B
4	Digit “4”	time and cost	Digit “4”	congestion	line lock out	congestion
5	Digit “5”		Digit “5”		subscriber idle without charge	send G-I signal next digit and change receipt G-A
6	Digit “6”	maintenance equipment	Digit “6”	send group II-6 and change to receipt G-C signal		send G-I signal same digit and change digit receipt G-A
7	Digit “7”	identification of the No. 2 subscriber in party line	Digit “7”			
8	Digit “8”	identification of the No. 3 subscriber in party line	Digit “8”			
9	Digit “9”	identification of the No. 1 subscriber in party line	Digit “9”			
10	Digit “0”		Digit “0”			
11	Digit “*”					
12	Digit “#”					
15			End of digit			

abcd Signals transmitted and received

abcd Signals received

Brazil Type

Signal	Forward signal		Backward signal	
	G-I	G-II	G-A	G-B
1	Digit “1”	ordinary subscriber	send next digit (n+1)	called party idle with charging
2	Digit “2”	subscriber with special charging		called party busy
3	Digit “3”	maintenance equipment	address complete, change-over to reception of G-II signal	changed number collect call blocking
4	Digit “4”	local public call box	congestion	congestion
5	Digit “5”	operator	send category of calling subscriber	called party idle without charging
6	Digit “6”	data transmission		called party idle with charging; called party based release
7	Digit “7”	national public call box ordinary subscriber (international working)	send last but two digits (n-2)	unallocated number collect call blocking
8	Digit “8”	data transmission (international working) collect call	send last but three digits (n-3)	
9	Digit “9”	subscriber with priority (international working)	send last but one digit (n-1)	
10	Digit “0”	operator with forward transfer facility (international working)		
11	Digit “*”	call transfer		
12	Digit “#”			
13				
14				
15	end of pulsing			

abcd Signals transmitted and received

abcd Signals received

Chile Type

Signal	Forward signal		Backward signal	
	G-I/III	G-II	G-A	G-B
1	Digit “1”	subscriber without priority	send next digit (n+1)	
2	Digit “2”	subscriber with priority	send last but one digit (n-1)	~~send special information tone~~
3	Digit “3”	maintenance equipment	address complete, change to reception of G-B signal	subscriber’s line busy
4	Digit “4”	subscriber with remote billing	congestion in the national network	congestion, encountered after changeover from G-A signal to G-B signal
5	Digit “5”	operator	send calling subscriber’s category	unassigned number
6	Digit “6”	data transmission	address complete with change	unoccupied subscriber’s line, with billing (charge)
7	Digit “7”		send last but two digits (n-2)	unoccupied subscriber’s line, without billing (without charge)
8	Digit “8”		send last but three digits (n-3)	subscriber’s line out of order
9	Digit “9”			
10	Digit “0”			
11	Digit “*”	unidentifiable subscriber		
12	Digit “#”			
13				
14				
15	end of pulsing			

abcd Signals transmitted and received

abcd Signals received

~~abcd~~ Signals not handled

Columbia Type

Signal	Forward signal		Backward signal	
	G-I	G-II	G-A	G-B
1	Digit “1”	subscriber	send next digit (n+1)	~~put connection under control of incoming side~~
2	Digit “2”	calling party with priority	send last but one digit (n-1)	~~changed number (called line intercepted)~~
3	Digit “3”	maintenance equipment	send nature of caller and change to G-B signals	called subscriber’s line busy
4	Digit “4”		congestion	congestion
5	Digit “5”	operator	send nature (and identify) of caller	dead line or dead level
6	Digit “6”	data transmission	through connect speech path	called subscriber’s line free with metering (charge)
7	Digit “7”	subscriber for international service	send last digit but two (n-2)	called subscriber’s line free without metering (without charge)
8	Digit “8”		send last digit but three (n-3)	called line out of order
9	Digit “9”		~~nationally at disposal~~	
10	Digit “0”	operator for international service	~~nationally at disposal~~	
11	Digit “*”		~~send first digit of international code~~	
12	Digit “#”		~~send first digit of significant number~~	
13	~~access to maint. equipment for national service~~		~~send first digit of international code of outgoing country~~	
14	~~insert echo suppressor~~		~~inquiry whether or not echo suppressor should be inserted~~	
15	end of pulsing			

abcd Signals transmitted and received

abcd Signals received

~~abcd~~ Signals not handled

Indonesia Type

Signal	Forward signal			Backward signal	
	G-I	G-II	G-III	G-A	G-B
1	Digit “1”	national operator	Digit “1”	send next digit (n+1)	subscriber line free (charge)
2	Digit “2”	normal subscriber	Digit “2”	restart from beginning	subscriber busy
3	Digit “3”	local payphone	Digit “3”	address complete change to B signal	~~send special information tone~~
4	Digit “4”	international operator	Digit “4”	congestion	
5	Digit “5”	long distance payphone	Digit “5”	address complete set-up speech condition	subscriber line free (no charge)
6	Digit “6”	national test and maintenance equipment	Digit “6”	send calling party’s category/number	~~malicious call tracing~~
7	Digit “7”	international test equipment	Digit “7”		unallocated national number
8	Digit “8”	cress border operator	Digit “8”	restart with last but one digit (n-1)	line out of service
9	Digit “9”	international payphone	Digit “9”	restart with last but two digits (n-2)	
10	Digit “0”		Digit “0”		
11	Digit “*”				
12	Digit “#”				
13					
14					
15	end of available information		end of available information		

abcd Signals transmitted and received

abcd Signals received

~~abcd~~ Signals not handled

India Type

Signal	Forward signal		Backward signal	
	G-I	G-II	G-A	G-B
1	Digit “1”	ordinary subscriber	send next digit (n+1)	
2	Digit “2”	priority subscriber	restart	changed number
3	Digit “3”		change over to reception of G-B signal	called line busy
4	Digit “4”		calling line identification	congestion
5	Digit “5”	operator	send category of the calling subscriber	unallocated number
6	Digit “6”		switch-through the speech path	normal subscriber, free, with metering
7	Digit “7”		send last but two digits (n-2)	called line free, but not metered
8	Digit “8”		send last but three digits (n-3)	
9	Digit “9”		send last but one digit (n-1)	
10	Digit “0”			
11	Digit “*”			
12	Digit “#”			
13				
14				
15				

abcd Signals transmitted and received

abcd Signals received

Programming

<<For MFC Outgoing/Incoming Connection or MFC Incoming Connection (with Tandem Connection)>>

STEP 1: **ASYD -** System 1, Index 125, Bits 0 through 3. PAD control of the MFC Sender 0dB--14dB.
Bits 4 through 7. MFC Signaling System CCITT MFC-R2, Venezuela, Australia, Singapore, Thailand.
System 3, Index 10, Bits 0 through 3. Reception sensitivity level of the MFC Register - 28dBm-43dBm.

STEP 2: **ANPD -** Assigns the first digit of the MFC line access code.

STEP 3: **ASPA -** Assigns the LCR access code by providing a dummy route.

STEP 4: **AMND -** Assigns the maximum number of necessary digits to the Dial (Area/Office) Code.

STEP 5: **ARTD -** Assigns the access code to the route number. When multiple access code are present by alternative routing, set one of them.
MFC line - Assign the following data, depending on the Trunk type, as shown below.

Trunk Type	ONSG	INSG
OGT	10	0
ICT	0	10
BWT	10	10

Assign the parameter OC (originating category) depending on the use of the G-A signal of the backward signal.
OC= 0: Originating category is not required.
OC=1: For terminating office, backward signal G-A=3 (originating category request) is sent after receiving necessary selecting signal. For originating office, forward signal G-II is sent after receiving backward signal G-A=3 (originating category request).

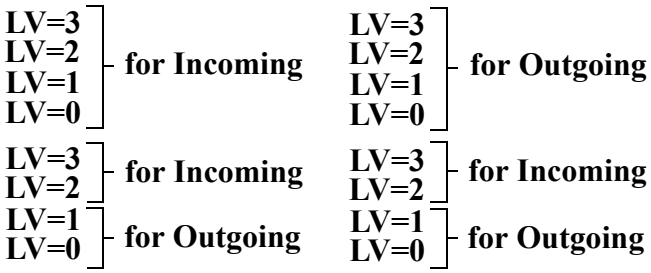
dummy route - Assign 1 at all times for the AC parameter of CDN 13. The route data of the dummy route is TCL=4, AC=1 and L/T=1. For the others, Assigns 0 in all cases.

Note: To change the data for existing route (RT), initialize or reboot the trunk device such as circuit card, MG, or VS32.

STEP 6: **ATRK -** MFC line - Assigns the LENS data of each trunk.
MFC - Register/Sender - Assigns the following data to the LENS of the PA-4MFCA circuit card.
Note 1

RT=916: MFC Register/Sender for Incoming
RT=917: MFC Register/Sender for Outgoing

Note 1: As the use of the 4 circuit on the PA-4MFCA circuit card, only the following 4 patterns are available.



STEP 7: **AFRS -** Assigns the number Pattern Code (NPC) and Outgoing route selection pattern (OPR) to the dummy route.

STEP 8: **AOPR -** Assigns the route number including the alternative route to the outgoing route selection pattern (OPR). Assigns the number of skipped digits and pattern number location (PNL) to each route.

STEP 9: **ARSC -** Releases the dummy route and station connection restriction.

Releases the connection restriction on the route of the MFC line and station.

STEP 10: **ASFC -** Assigns the backward signal (G-B) against the forward signal (G-II) for each SFC when the called party is idle.

SFI= 59: No Charge

SFI=60: Called Party Release

Relation between the Assignment of SFI=59,60 and the Backward Signal.

SPECIFICATION	SFI = 59	SFI = 60	BACKWARD SIGNAL	REMARKS
CCITT MFC-R2	0	1	G-B=6	
	1	0	G-B=7	
VENEZUELA	0	0	G-B=6	
AUSTRALIA	0	0	G-B=1	
	0	1	G-B=6	
	1	0	G-B=5	
SINGAPORE	0	0	G-B=1	
THAILAND	0	0	G-B=1	

Note: Combination of SFI=59 and 60 which are not in the table in each specification must not be assigned.

STEP 11: **ARRC -** For Tandem Connection Service, Release incoming route and outgoing route connection restriction.

M-19 Message Reminder

General Description

This feature allows a Single Line Telephone and a Desktop terminal to leave a MESSAGE REMINDER at the called Dterm set.

Operating Procedure

To send a message from a Desktop terminal

1. Dial the desired Desktop terminal station. Receive Ringback Tone or Busy Tone.
2. Press the MSG key. The LCD displays:



   and a Service Set Tone is heard.
3. At the Called Desktop terminal, the MSG LED will light to indicate a message has been set.

To send a message from a Single Line Telephone

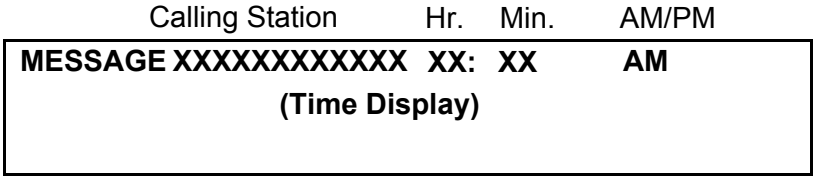
1. Dial the desired internal Dterm Station. Receive Ringback Tone or Busy Tone.
2. Press switch hook. Receive Special Dial Tone.

   Note: The availability of special dial tone is changed depending on the combination of the Step Call - CCIS Restriction (ASYD: Sys 2, Index 6, Bit 2) and SHF method (ASYD: Sys 1, Index 68, Bit 0). Refer to Step Call - CCIS [S-59] in Data Programming Manual - CCIS for the detail.

3. Dial the MESSAGE REMINDER Access Code. Receive Service Set Tone.
4. At the Called Dterm, the MSG LED will light to indicate a message has been set.

To review messages

1. Press the MSG key while the key lights red.
2. The LCD displays:



3. To display the next message, press the MSG key again.

   Note: When the Station Number exceeds 12 digits, MSG is not displayed.

To return the message

1. MSG key lights red.
2. Select a target message by pressing the MSG key.
3. The message is displayed.
4. Lift the handset or press the SPEAKER key.

5. Press the MSG key again.

Service Conditions

When sending a message from a Single Line Telephone or Desktop terminal

1. This feature is available only when the called party is a Dterm set, which has the MSG key.
2. Messages can only be sent if the called station is in the idle or busy condition.
3. Messages cannot be sent while engaged in calling and recalling from CALL TRANSFER - ALL CALLS [C-11] and from CONSULTATION HOLD - ALL CALLS [C-17].
4. Messages can be displayed only when the station is on-hook.
5. A message may be cleared by displaying the message and pressing the "*" or "#" key on the called Desktop terminal's keypad. This is selected via System Data.
6. Messages are cleared when they are obtained via the MSG key.
7. A second message sent by the same calling telephone is registered as a second unique message. The same applies to the third and fourth messages.
8. Messages are sent to the forwarded-to station, which is set by CALL FORWARDING - ALL CALLS [C-5] or CALL FORWARDING - BUSY/DON'T ANSWER [C-2, 3].
9. A maximum of 32 digits can be saved at the called party.
10. To utilize the Telephone Number of the Calling Station, assign System Data 1, Index 641, Bit 0 = 1 by the ASYDN command.

When sending a message from a Desktop terminal

1. If a message is directed to a Desktop terminal that does not have the MESSAGE REMINDER feature, the Calling Station's LCD displays:



2. A maximum of four messages may be received at any one Desktop terminal. If a fifth message is attempted, the LCD will display:



3. Messages cannot be cleared by the Message Sender.
4. If a message is returned to an unattended station, the Telephone Number may be stored as a LAST NUMBER DIALED [L-6], SAVE AND REPEAT [S-24], or a new MESSAGE REMINDER may be sent.
5. Messages cannot be sent to a UCD [U-1] station when all stations in the UCD Group are busy.
6. Messages are sent to a dialed station when all stations in a Station Hunting (Circular) Group are busy.
7. The setting station in the FCCS network must provide Telephone Number for this feature.
8. Message (Telephone Number) is cleared by the circuit card initialization or the system initialization.
9. When a terminal logs out or performs re-registration, all messages stored in the terminal are cleared. To maintain the messages, enable PH Fixed Hunting method by assigning the system data (ASYDL, SYS1, Index 824, Bit 0 = 1).
10. Memory clear is necessary before reprogramming of station data and before using Follow Phone service.

When sending a message from a Single Line Telephone

1. If a message is directed to a Dterm that does not have the MSG key or a Single Line Telephone, the Calling

Station will hear Reorder Tone.

2. A maximum of four messages may be received at any one Dterm. If a fifth message is attempted, the Reorder Tone will be transmitted to the Calling Party.
3. In case the called Dterm station is in a Hunting Group, the message will be set to the first selected Station in the Hunting Group when- all Stations are busy. The message will be set to the ringing station in the Hunting Group when any station is idle.
4. When the call is in the UCD queue, the MESSAGE REMINDER service is not provided.

Interactions

1. If CALL WAITING-ORIGINATING [C-31] automatic operation is in service between two Desktop terminals, it will supersede MESSAGE REMINDER sending a message from Desktop terminal. If a third station calls and receives Busy Tone, MESSAGE REMINDER sending a message from Desktop terminal can be set.
2. A message cannot be sent from or to SIP Handler Controlled SIP terminals.

Programming

To send a message from a Desktop terminal

STEP 1: **ASYD** - System Data 1, Index 80, Bit 1. MESSAGE REMINDER Cancel key (for Desktop terminal) 0/1: "#" key / "*" key.

STEP 2: **AKYD** - Assign the Station according to the following parameters:

TN: 1
STN: Station Number
TP: Type of Desktop Terminal (0-3)
KYN: Key Number (1-40)
KYI: Service Index
0: Key Not Used
→ 1: Feature key
2: Multi Line key
KD: Not assigned for Feature key
FKY: Feature Key Number
→ FKY = 6, MESSAGE REMINDER

STEP 3: **ASFC** - With respect to SFC of the Stations that sets Message Reminder Service, cancel the restriction of SFI 53. (Message Reminder Service)

Note: Desktop terminals with a MESSAGE REMINDER key do not require the Service Feature Restriction Class assignment. Only single-line sets require the ASFC assignment.

To send a message from a Single Line Telephone

STEP 1: **ASYD** - System Data 1, Index 17, Bit 0.

Single Line Stations able to use VOICE CALL [V-2] and MESSAGE REMINDER features to a Dterm while hearing Ringback Tone? 0/1: Restricted/Allowed.

System Data 1, Index 80, Bit 1. Key for message cancel? 0/1: "#" key / "*" key.

STEP 2: **ASFC** - Assign the Single Line Stations a feature class that allows SFI: 53, Message Reminder.

STEP 3: **ANPD** - Reserve a Number Level for Feature Access. Select Normal Service (N), Hooking Service (H) or Busy Service (B) from the combo box for Connection Indexes. Assign NND in accordance with predetermined Numbering Plan.

STEP 4: **ASPA** - Assign an Access Code to Message Reminder; assign SRV = "SSCA: Service Code Appendix", SIDA = 42; assign CI = "H: Hooking and "B: Busy".

FCCS Programming

This data setting describes only the additional or replacement commands used to activate FCCS Services. When used as an FCCS feature, these commands replace the non-network commands (for example, AAED is replaced by AAEDN). Note that the data setting for FCCS Service varies depending on the Node.

Note: When using ASYDL or ASYDN to program an FCCS feature, the basic programming of ASYD is required in addition to ASYDL or ASYDN.

STEP 1: **ANPDL** - Reserve a Number Level for Feature Access. Select Normal Service (N), Hooking Service (H), or Busy Service (B) from the combo box for Connection Indexes. Assign NND in accordance with predetermined Numbering Plan.

STEP 2: **ASPAL** - Assign an Access Code to Message Reminder; assign SRV = "SSCA: Service Code Appendix", SIDA = 42; assign CI = "H: Hooking and "B: Busy".

M-20D Multiple Line Operation -D

General Description

This feature allows for the appearance of Multiple Lines on the Line/Feature keys of the Desktop terminal.

Note: For North America only - Multiple Lines Appearances are supported on an ACD set.

Operating Procedure

No manual operation is required.

Service Conditions

1. Three kinds of lines can appear on Desktop terminals:
   - MY Line (ML): This line is associated with the extension number assigned to the port on the Interface Circuit.
   - Prime Line (PL): This line is given priority for outgoing and Incoming Calls. The Prime Line is automatically seized when going off-hook.
   - Sub Line (SL): This line designates any other line appearing on a Desktop terminal. (The MY Line will serve as a Prime Line, unless differentiated in System Data programming).
2. If a feature is activated that can trigger an AUTOMATIC RECALL [A-8], only the Station being recalled will ring, and the associated line lamp will rapid wink. Other Desktop terminal Stations will display that line as busy.
3. All Sub Lines of a Desktop terminal must be located in the same Module Group as the Desktop terminal MY Line.
4. A Virtual Line (Software Line Appearance [S-33]), Single Line Telephone or Dterm can be assigned to the Sub Line.
5. CALL BACK [C-1] and OUTGOING TRUNK QUEUING -D [O-2] is available only from the My Line of a Desktop terminal.
6. In the calling, when Line key then Answer key are pressed on the system that does not have automatic calling, the Line key holds a call; the Answer key cannot respond to the calling.
7. For North America only - The additional appearance is not configured in the ACD. Therefore, it is not monitored for call progress events, and no information pertaining to call activity will be sent to an attached MIS system. MIS statistics of any kind will not be available on this line.
8. For North America only - ACD features such as “Auto Work on PBX line Ringing/Answer” will not be invoked by call activity on this line.
9. For North America only - Cannot suppress DT from being audible on Dterm stations equipped with a speaker when Multiple line appearances are assigned and ASYD Index 384 Bit 0=1, regardless if a speaker button is not assigned. Dial tone will be produced as a terminal function.
10. Restriction of Call Termination on My Line:
    A Desktop terminal can deny any Incoming Call to its My Line, when a call is made/received via one of its Sub Lines.

    Conditions:

    a.) Restricts a call to My Line only. (A call to Sub Line is not restricted.)
    b.) Available when the caller is a Station, TIE Line, Central Office Line or Attendant Console.
    c.) Available for a call as the destination of Call Forwarding - All Calls.
    d.) When the Original Call on the Sub Line is placed on hold:

• If the terminal is off–hook after placing the call to Sub Line on hold, a call to My Line is restricted.
• If the terminal goes on–hook while the call to Sub Line is still on hold, a call to My Line is not restricted.

11. If a call is terminated at your My Line while you are talking on or placing a call using a Sub Line, LCD does not display the information of the incoming call until you finish talking or placing the call.

Note: Note the following directions about LCD.

- *If you need to see incoming call information while you are talking or placing a call, assign ASYDL/ ASYDN, SYS1, Index 1248, Bit 4=1 (Mixed Display of Caller ID Display-Subline is enabled).*

Interactions

1. Sub Lines can belong to another Station, or be a virtual/software Line Appearance. Sub Lines will operate under the restrictions of the MY Line. Sub Lines must be located within the same Module Group as the MY Line when assigning ASYD, System Data 1, Index 321, Bit 0 as "0."
2. Multi Line service in the Inter-Module Group under the control of one LP is available when assigning ASYD, System Data 1, Index 321, Bit 0 as "1."
3. When SIP Handler Controlled SIP terminals are used, either of Single Line Mode or Multi Line Mode can be selected.
   a.) The following are conditions when Desktop terminal accommodates My Line and Sub Line of SIP Handler Controlled SIP terminals.

   - Desktop terminal can answer a call held by SIP Handler Controlled SIP terminals.
   - When all lines which a SIP Handler Controlled SIP terminal accommodates are used by Desktop terminals, the SIP Handler Controlled SIP terminal cannot place a call.
   - When Desktop terminal uses My Line or Sub Line of the SIP Handler Controlled SIP terminal and puts a call on hold, the SIP Handler Controlled SIP terminal cannot answer the held call.

   b.) The following are conditions when the SIP Handler Controlled SIP terminal is in Single Line Mode.

   - Only My Line can be used when a SIP Handler Controlled SIP terminal user places or receives a call.
   - Single Line Mode supports more features than Multi Line Mode does.

   c.) The following are conditions when the SIP Handler Controlled SIP terminal is in Multi Line Mode.

   - Two lines, My Line and Sub Line, can be used when the SIP Handler Controlled SIP terminal user places or receives a call. The SIP Handler Controlled SIP terminal cannot handle three or more lines.
   - My Line is selected preferentially while placing or receiving a call. If My Line is in use, Sub Line is selected. The SIP Handler Controlled SIP terminal user cannot select a line to place a call. Calling station number displayed on the destination station is either of My Line or Sub Line station number which is seized to place a call.

- The SIP Handler Controlled SIP terminal user cannot recognize which line receives a call.
- The SIP Handler Controlled SIP terminal Multi Line keys are not compliant with the Multiple Line data assigned by the AKYD command. The terminal user recognizes the first call arrives at the same line.
- Multi Line Mode supports more restricted features than Single Line Mode does.

Programming

STEP 1: **ASDT** - LENS equipped with a Desktop terminal Circuit Board must be assigned a Station Number and a Telephone Equipment Class (TEC) of 12.

STEP 2: **AKYD** - Assign a MY Line, a Prime Line, and various Sun Lines to the Desktop terminals.

→TN: 1
→STN: Station Number
TP: Type of Dterm 0/1:
→RING: Assign ringing to the entire phone
0: No Ringing, Day or Night.
1: Ring Day.
2: Ring Night.
3: Not Used.
4: Ring Day & Night, 30 second delay.
5: Ring Day & Night, 20 second delay.
6: Ring Day & Night, 10 second delay.
7: Ring Day & Night, no delay.PL TN STN:Prime Line Tenant and Station Number
KYN: Key Number (1-40)
KYI: Service Index
0: Key Not Used
1: Feature key
→ 2: Multi Line key
KD: Not assigned for Feature key
TN STN: Tenant Number and Station Number of Multi Line
RG: Ringing Assignment per Multi Line Appearance
0: No Ringing, Day or Night.
1: Ring Day.
2: Ring Night.
3: Not Used.
4: Ring Day & Night, 30 second delay.
5: Ring Day & Night, 20 second delay.
6: Ring Day & Night, 10 second delay.
7: Ring Day & Night, no delay.

M-23 Message Waiting Lamp Setting - Attendant

General Description

This feature permits an ATTENDANT CONSOLE [A-3] to set or cancel a Message Waiting indication on a Station provided with this feature.

Operating Procedure

To set a Message Waiting Indication

1. Press an Idle Loop key.
2. Dial the Message Waiting set Code (up to three digits). Receive Special Dial Tone.
3. Dial the Telephone Number. Receive Service Set Tone.
4. Message Waiting indication is set.
5. Press the RELEASE key to return to an idle condition.

To cancel a Message Waiting Indication

1. Press an Idle Loop key.
2. Dial the Message Waiting Cancel Code (up to three digits). Receive Special Dial Tone.
3. Dial the Telephone Number. Receive Service Set Tone.
4. Message Waiting indication is canceled.
5. Press the RELEASE key to return to an idle condition.

Service Conditions

1. The Message Waiting lamp on a Single Line Telephone or a Dterm will be lit when this feature is activated.
2. When Message Waiting is set to a Dterm Station, the LCD display indicates:



3. Message Waiting cannot be set when the ATTENDANT CONSOLE [A-3] is in a consultation condition with a Station or Trunk.
4. An Attendant attempting to set Message Waiting to a Station where Message Waiting has previously been set, will receive Busy Tone.
5. In a MULTIPLE CONSOLE OPERATION [M-4], if Message Waiting is set by one ATTENDANT CONSOLE [A-3], it can be canceled by another.
6. Message Waiting indications are cleared when the system is initialized.

7. When Call Indicator Lamp is controlled by MESSAGE WAITING LAMP SETTING - STATION [M-30], the tone and display indication of the Attendant Console and Call Indicator Lamp of the Station are as follows, depending on the status of the Station.

Status of Preassigned Station		Attendant Console			Call Indicator Lamp Status
		Operation	LCD	Type of Tone	
Unused Telephone Number Dead Level Restricted Station		Canceling	Telephone Number	Reorder Tone	
		Setting			
Call Indicator Lamp OFF	Idle Busy Lock Out Make Busy	Canceling	Telephone Number	Service Set Tone	OFF
		Setting			ON
Call Indicator Lamp OFF set from Attendant	Call Indicator Lamp ON set from Station	Canceling	Telephone Number	Service Set Tone	ON
		Setting			
Call Indicator Lamp ON set from Attendant	Call Indicator Lamp OFF set from Station	Canceling	Telephone Number	Service Set Tone	OFF
		Setting		Busy Tone	ON
	Call Indicator Lamp ON set from Station	Canceling	Telephone Number	Service Set Tone	ON
		Setting		Busy Tone	

Interactions

1. For MESSAGE WAITING LAMP SETTING-ATTENDANT, an 8DLC/16ELC card is required.
2. An external power source is required to light a lamp for a Single Line Telephone. The Dterm uses an 80 VDC power source. NEC stocks Lorain Model HSA3H, stock number 241440.
3. Only the Attendant can set and extinguish the Message Waiting Lamp.

Programming

STEP 1: **ANPD** - Reserve a Number Level for Feature Access and cancel. Select Normal Service (N), Hooking Service (H) or Busy Service (B) from the combo box for Connection Indexes. (applicable to any feature assigned to this Level). Assign NND in accordance with a predetermined Numbering Plan.

STEP 2: **ASPA** - Assign an Access Code to activate MESSAGE WAITING LAMP SETTING-ATTENDANT SRV = "SSCA: Service Code Appendix"; SIDA = 54 and to extinguish the MESSAGE WAITING LAMP SETTING-ATTENDANT SRV = "SSCA: Service Code Appendix"; SIDA = 55. Assign the Connection Status Index (CI) for "N: Normal" service.

STEP 3: **ASFC** - Assign a Service Feature Restriction Class that allows SFI = 74 to the Stations that will receive MESSAGE WAITING LAMP SETTING-ATTENDANT.

FCCS Programming

This data setting describes only the additional or replacement commands used to activate FCCS Services. When used as an FCCS feature, these commands replace the non-network commands (for example, AAED is replaced by AAEDN). Note that the data setting for FCCS Service varies depending on the Node.

STEP 1: **ANPDN** - Reserve a Number Level for Feature Access and cancel. Select Normal Service (N), Hooking Service (H), or Busy Service (B) from the combo box for Connection Indexes. (applicable to any feature assigned to this Level). Assign NND in accordance with a predetermined Numbering Plan.

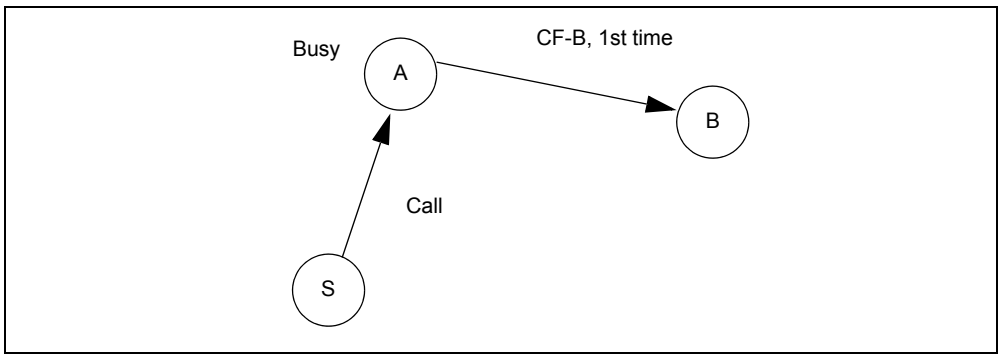
STEP 2: **ASPAN** - Assign an Access Code to activate MESSAGE WAITING LAMP SETTING-ATTENDANT SRV = "SSCA: Service Code Appendix"; SIDA = 54 and to extinguish the MESSAGE WAITING LAMP SETTING-ATTENDANT SRV = "SSCA: Service Code Appendix"; SIDA = 55. Assign the Connection Status Index (CI) for "N: Normal" service.

M-24 Multiple Call Forwarding - Busy Line

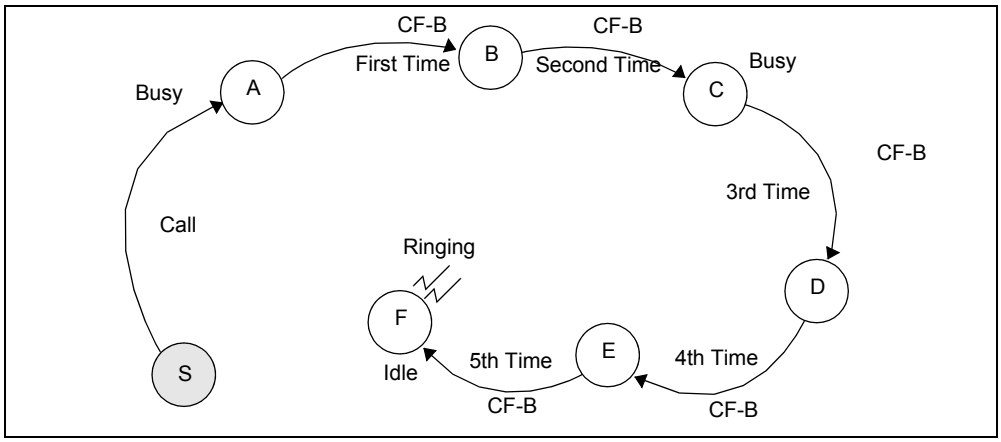
General Description

This feature permits a call to a busy Station to be forwarded, a maximum of five times, to preprogrammed Idle Stations.

Existing CALL FORWARDING - BUSY LINE [C-2] service



MULTIPLE CALL FORWARDING - BUSY LINE [M-24] service



Operating Procedure

To set from an individual Station

1. Lift the handset. Receive Dial Tone.
2. Dial the specific CALL FORWARDING - BUSY LINE Access Code. Receive Special Dial Tone.
3. Dial the desired Target Telephone Number. Receive Service Set Tone.

To cancel from an individual Station

1. Lift the handset. Receive Dial Tone.
2. Dial the specific CALL FORWARDING - BUSY LINE [C-2] Cancel Code; receive Service Set Tone.

To set from the Attendant Console

1. Press an Idle Loop key.
2. Dial the CALL FORWARDING - BUSY LINE [C-2] Access Code. Receive Special Dial Tone.
3. Dial the TENANT SERVICE [T-12] Number (two or three digits).
4. Dial the Originating Telephone Number.

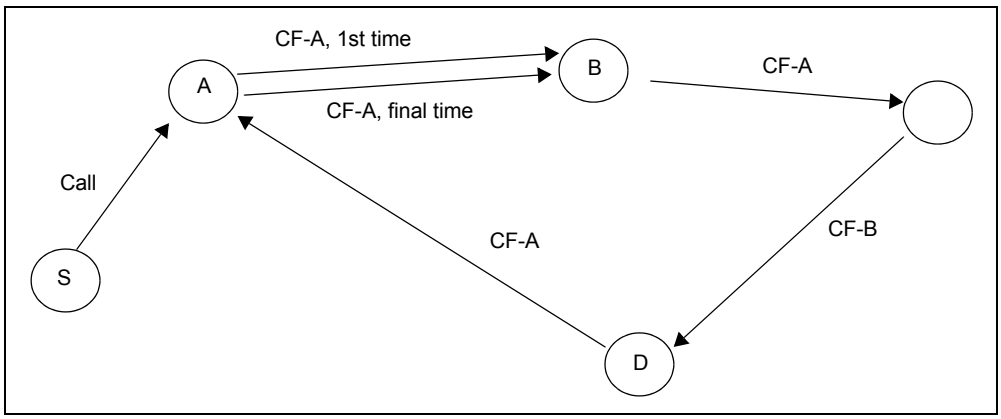
5. Dial the desired Target Telephone Number. Receive Service Set Tone.

To cancel from the Attendant console

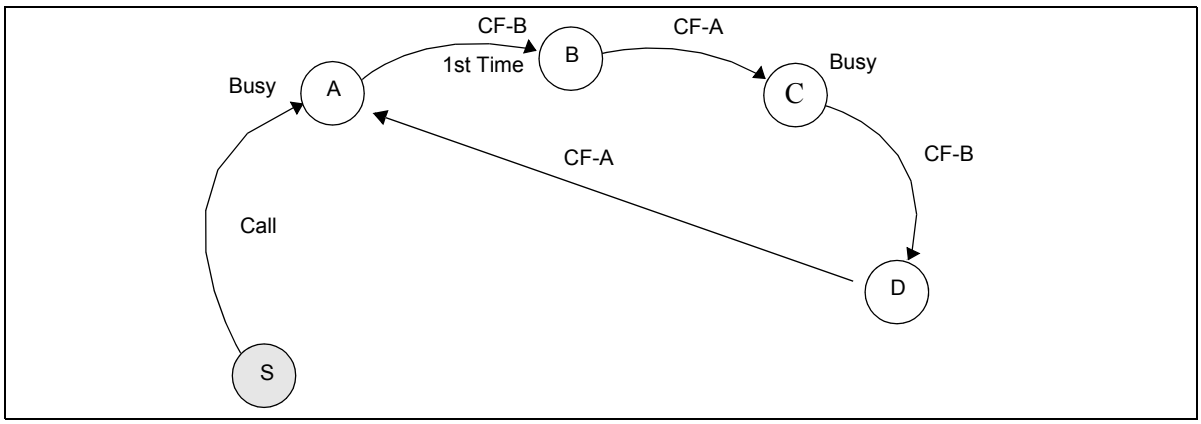
1. Press an Idle Loop key.
2. Dial the CALL FORWARDING - BUSY LINE [C-2] Cancel Code. Receive Special Dial Tone.
3. Dial the TENANT SERVICE [T-12] Number (two or three digits).
4. Dial the Originating Telephone Number. Receive Service Set Tone.

Service Conditions

1. If the Called Station is busy and MULTIPLE CALL FORWARDING - BUSY LINE [M-24] is set at the Called Station, the call is forwarded to the designated Station.
2. A call cannot be forwarded more than five times, even if the fifth Station has CALL FORWARDING - BUSY LINE set.
3. A call cannot be forwarded to the originating Station.
4. If the call is forwarded back to the originally Called Station, or a Station that has already been forwarded to, Call Forwarding will end at that Station.
5. CALL FORWARDING - BUSY LINE [C-2] and CALL FORWARDING - ALL CALLS [C-5] combined:
   a.) Any combination of CALL FORWARDING - BUSY LINE [C-2] and CALL FORWARDING - ALL CALLS [C-5] may be made, up to a total of five forwards.
   b.) If the fifth forwarding destination is busy, the caller receives Busy Tone, and the call will not forward again.
   c.) If a forward destination is the Calling Station, the Calling Station will receive Busy Tone.
   d.) If a forwarding destination is the originally Called Station, and if the originally Called Station forwarded the call with CALL FORWARDING - ALL CALLS [C-5], the ultimate destination is the originally Called Station's CALL FORWARDING - ALL CALLS [C-5] destination.



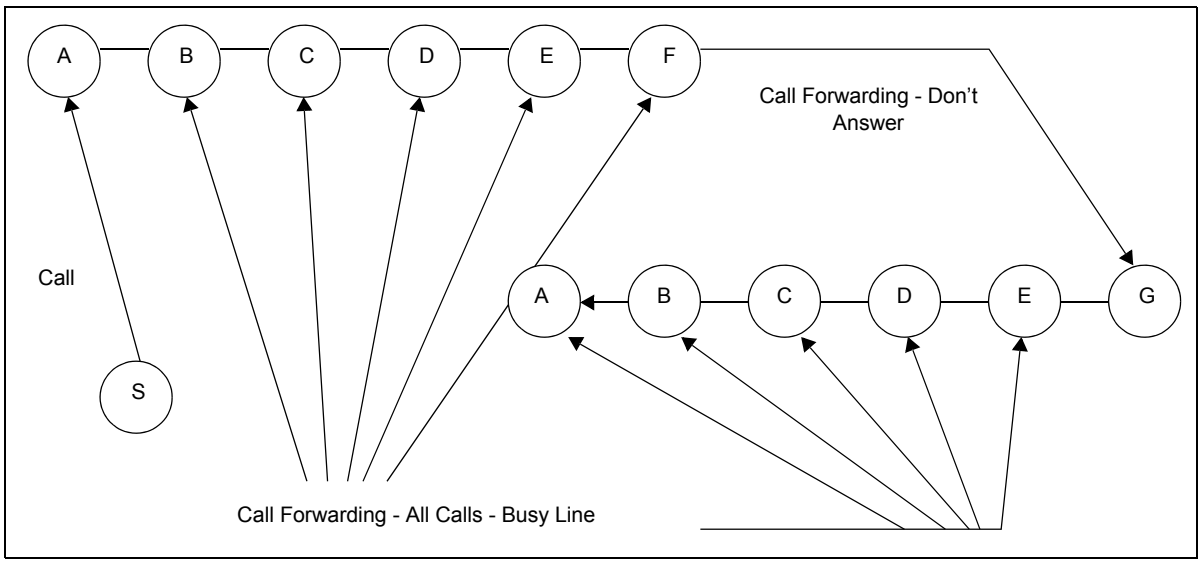
   e.) If a forwarding destination is the originally Called Station, and if the originally Called Station forwarded the call with CALL FORWARDING - BUSY LINE, the ultimate destination is the originally Called Station.



f.) If the call is forwarded to a Station where it has previously been forwarded to, the call will end at that Station.

6. CALL FORWARDING - ALL CALLS [C-5], CALL FORWARDING - BUSY LINE [C-2] and CALL FORWARDING - DON'T ANSWER [C-3] combined:

a.) Although CALL FORWARDING - ALL CALLS [C-5] and CALL FORWARDING - BUSY LINE [C-2] and combinations of the two are limited to five forwards, when a forwarding sequence is divided by CALL FORWARDING - DON'T ANSWER [C-3], CALL FORWARDING - ALL CALLS [C-5] and CALL FORWARDING - BUSY LINE [C-2] can begin again (the only limit is five consecutive forwards).



7. Forwarding to an Attendant Console:

a.) If an Attendant Console is designated as the destination of CALL FORWARDING - ALL CALLS [C-5], the call is forwarded to the Attendant Console that handles the Tenant to which the Calling Station belongs.

b.) The Attendant Console key that lights when a call is forwarded to the Console depends on the forwarding status of the originally Called Station. The following table shows the key that will light (if the forwarding status of the originally Called Station is CALL FORWARDING - BUSY

[C-2], and the call is from a Station, then the BUSY key will light).

Forwarding Status Of Originally Called Station			
	Call Forward -All Calls	**Call Forward -Busy**	**Call Forward -Don't Answer**
Call From Station	ATND	BUSY	NANS
Call From Trunk	Depends on Trunk class	BUSY	NANS

c.) If the call is forwarded by CALL FORWARDING - DON'T ANSWER [C-3] at any point in a forward sequence, the NANS key on the Attendant Console will light, even if the call status of the originally Called Station is CALL FORWARDING - ALL CALLS [C-5] or CALL FORWARDING - BUSY LINE [C-2].

d.) Called Number Display is only provided if a call is forwarded to the Attendant Console by CALL FORWARDING - ALL CALLS [C-5], CALL FORWARDING - BUSY LINE [C-2], or by CALL FORWARDING - DON'T ANSWER [C-3].

8. Forwarding to a Dterm:

a.) The Calling Dterm's display will show the number of the Called Station on the left, and the number of the Station being rung on the right. The display on the ringing Dterm will show the number of the originally Called Station on the left, and the number of the Calling Station on the right.

b.) The forwarding status of the originally Called Station is shown on the far left of each Dterm's display: FDB for CALL FORWARDING - BUSY [C-2] or FDA for CALL FORWARDING - ALL CALLS [C-5].

9. UCD Groups:

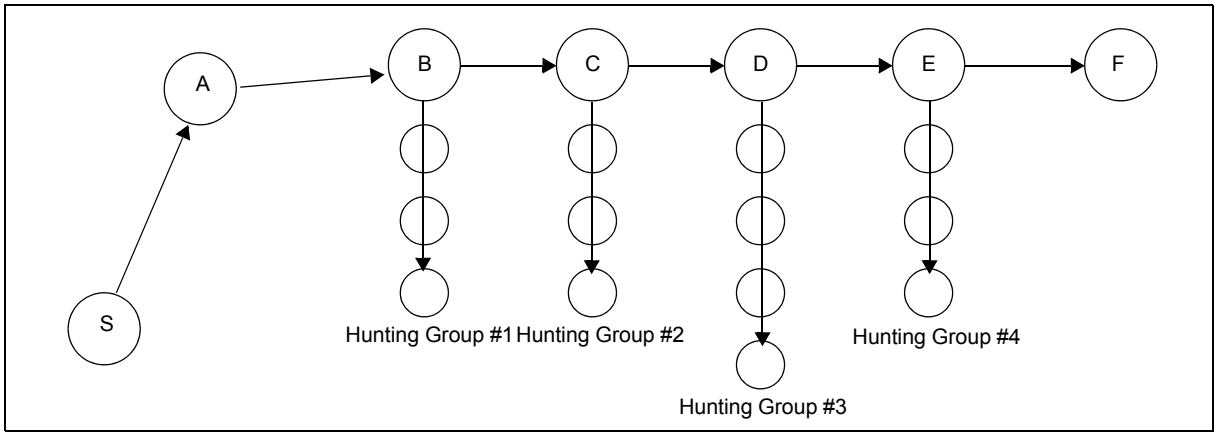
a.) If the destination of CALL FORWARDING - ALL CALLS [C-5], CALL FORWARDING - BUSY [C-2], or CALL FORWARDING - DON'T ANSWER [C-3] is a UCD Group control Station, and CALL FORWARDING - ALL CALLS [C-5] is not set at that Station, the call will be forwarded according to UNIFORM CALL DISTRIBUTION [U-1] procedures.

b.) In the UCD procedure, if CALL FORWARDING - ALL CALLS [C-5] is set on a Station in the group, that Station is skipped.

10. Hunting:

a.) If the ultimate Destination Station is busy, or if some condition prevents the call from being forwarded, a busy line is selected, and a hunting check is performed.)

b.) If a call is forwarded several times by CALL FORWARDING - BUSY LINE [C-2] to Stations that are members of Hunting Groups, a hunting check is performed on the busy line closest to the Called Station or on the busy line closest to the ultimate destination. In the diagram below, a hunting check is performed on Hunt Groups 1 or 4. A hunting check is never performed on groups 2 or 3.



11. Busy Services such as EXECUTIVE RIGHT OF WAY [E-1], CALL BACK [C-1], etc.:
    a.) Busy service is provided to the busy line closest to the originally Called Station.
12. DIT, NIGHT CONNECTION - FIXED [N-1]:
    a.) For CALL FORWARDING - ALL CALLS [C-5] and CALL FORWARDING - BUSY LINE [C-2], a forward check is made up to 5 times by System Data.
    b.) In DIRECT-IN TERMINATION [D-7], if a line is busy, CALL FORWARDING - BUSY LINE [C-2] to the Attendant is not activated. Instead, the destination for the call can be specified by System Data:
    System Data 2 Index = 11
    Bit 7 = 0 - Call is routed to the Attendant Console.
    Bit 7 = 1 - Call is placed in queue until the Called Station becomes idle.
13. A Telephone Number must be set to the Call Forwarding Destination.
14. A call cannot be forwarded more than five times.
15. The number of times is assigned by System Data.
16. The OAI monitor number and the ACD pilot number cannot be assigned as the Call Forwarding target Telephone Number.

Interactions

1. If Access Codes for CALL FORWARDING-BUSY LINE [C-2] and CALL FORWARDING-DON'T ANSWER [C-3] are the same, only one key needs to be assigned in AKYD or ADSL, and only one Access Code needs to be assigned in ASPA.
2. When CALL WAITING [C-12], ATTENDANT CAMP-ON WITH TONE INDICATION [A-1], or CALL BACK [C-1] is directed toward Station A, with CALL FORWARDING-BUSY LINE [C-2] set to Station B, Station A will receive Call Waiting Tones when Station B is also busy.
3. A non-controlling UCD Station is allowed to activate CALL FORWARDING-BUSY LINE [C-2]. When this Station sets CALL FORWARDING-BUSY LINE [C-2], only directly-dialed calls to this Station are call forwarded.
4. If a controlling Station of a UCD Group sets CALL FORWARDING-BUSY LINE [C-2], forwarding will not take place. The Incoming Call will follow the UCD hunt.
5. If an Attendant attempts BUSY VERIFICATION [B-3] to Station A, which has CALL FORWARDING-BUSY LINE [C-2] set to Station B and Station B is busy, BUSY VERIFICATION [B-3] proceeds against Station B. If Station A has set CALL FORWARDING-BUSY LINE [C-2] to the Attendant Console, BUSY VERIFICATION [B-3] is allowed at Station A, if Station A is busy.

6. When CALL FORWARDING-BUSY LINE [C-2] is in effect, the activating Station may be used to originate calls.
7. When a Station is call forwarded, the Destination Station is recorded by STATION MESSAGE DETAIL RECORDING [S-10].

Programming

STEP 1: **ASYD** - System Data 1, Index 5, Bit 0. Are the Access Codes for CALL FORWARDING-BUSY LINE [C-2] and CALL FORWARDING-DON'T ANSWER [C-3] the same or separate? 0/1: Same/ Separate.

System Data 1, Index 5, Bits 3, 2, and 1. Maximum number of MULTIPLE CALL FORWARDING-ALL CALLS [M-44], -BUSY LINE [M-24] occurrences.)

Bit 3	Bit 2	Bit 1	MAXIMUM NUMBER OF JUMPS
0	0	0	1
0	0	1	1
0	1	0	2
0	1	1	3
1	0	0	4
1	0	1	5
1	1	0	5
1	1	1	5

System Data 1, Index 69, Bit 5. Station Hunting after CALL FORWARDING-BUSY LINE [C-2]. Data "0": Hunting takes place in forwarding Station's Group. Data "1": Hunting takes place in Destination Station's Group.

System Data 1, Index 69, Bit 7. MULTIPLE CALL FORWARDING-BUSY LINE, -ALL CALLS [M-44], in service? 0/1: No/Yes.

System Data 2, Index 6, Bit 1. Enable CALL FORWARDING-BUSY LINE [C-2] on a per Tenant basis.

STEP 2: **ANPD** - Reserve a Number Level for Feature Access and cancel. Select Normal Service (N), Hooking Service (H) or Busy Service (B) from the combo box for Connection Indexes. Assign NND in accordance with a predetermined Numbering Plan.

STEP 3: **ASPA** - Assign an Access Code to CALL FORWARDING-BUSY LINE [C-2] entry, SID = 10 and to CALL FORWARDING-BUSY LINE [C-2] cancel SID = 11. Assign a Connection Status Index (CI) for "N: Normal" service.

STEP 4: **ASFC** - Assign a Service Feature Restriction Class that allows SFI = 9 to the Stations that have CALL FORWARDING-BUSY LINE [C-2].

STEP 5: **AKYD** - For Dterm sets, CALL FORWARDING-ALL CALLS [C-5] may be assigned to a Line/ Feature key. Type:Function:CALL FORWARDING-BUSY LINE [1].

STEP 6: **ADSL** - Assign SN = 1 (Originating; DT Connection) and FKY = 1 (Call Forwarding - Busy Line; FDB).

This command is used for assigning the function to a Soft key.

FCCS Programming

This data setting describes only the additional or replacement commands used to activate FCCS Services. When used as an FCCS feature, these commands replace the non-network commands (for example, AAED is replaced by AAEDN). Note that the data setting for FCCS Service varies depending on the Node.

Note: When using ASYDL or ASYDN to program an FCCS feature, the basic programming of ASYD is required in addition to ASYDL or ASYDN.

ASYDN - These data must be set to the NCN.

System Data 1, Index 69, Bit 7. Enable Multiple Call Forwarding-Busy Line/Don't Answer/All Calls. 0/1 = Out of Service/In Service. Index5, Bit 1-3. Assign the number of time for forwarding.

M-25 Multiple Call Forwarding - Don’t Answer

General Description

This feature permits a call to an unanswered Station to be forwarded, multiple times, to preprogrammed Idle Stations that do not have Call Forwarding set.

Operating Procedure

To set from an individual Station

1. Lift the handset. Receive Dial Tone.
2. Dial the specific CALL FORWARDING - DON'T ANSWER [C-3] Access Code. Receive Special Dial Tone.
3. Dial the desired target Telephone Number. Receive Service Set Tone.

To cancel from an individual Station

1. Lift the handset. Receive Dial Tone.
2. Dial the specific CALL FORWARDING - DON'T ANSWER [C-3] Cancel Code. Receive Service Set Tone.

To set from the Attendant Console

1. Press an Idle Loop key.
2. Dial the specific CALL FORWARDING - DON'T ANSWER [C-3] Access Code. Receive Special Dial Tone.
3. Dial the TENANT [T-12] number (two or three digits).
4. Dial the originating Telephone Number.
5. Dial the desired target Telephone Number. Receive Service Set Tone.

To cancel from the Attendant Console

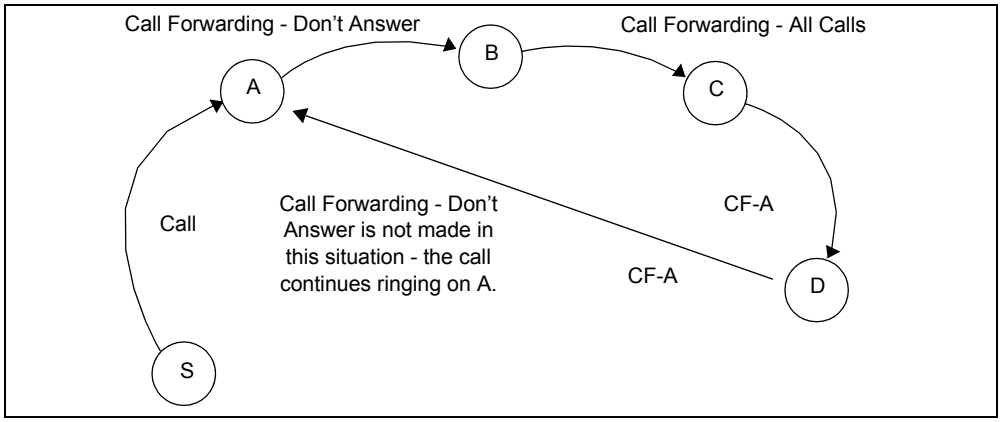
1. Press an Idle Loop key.
2. Dial the specific CALL FORWARDING - DON'T ANSWER [C-3] Cancel Code. Receive Special Dial Tone.
3. Dial the TENANT SERVICE [T-12] Number (two or three digits).
4. Dial the originating Telephone Number. Receive Service Set Tone.

Service Conditions

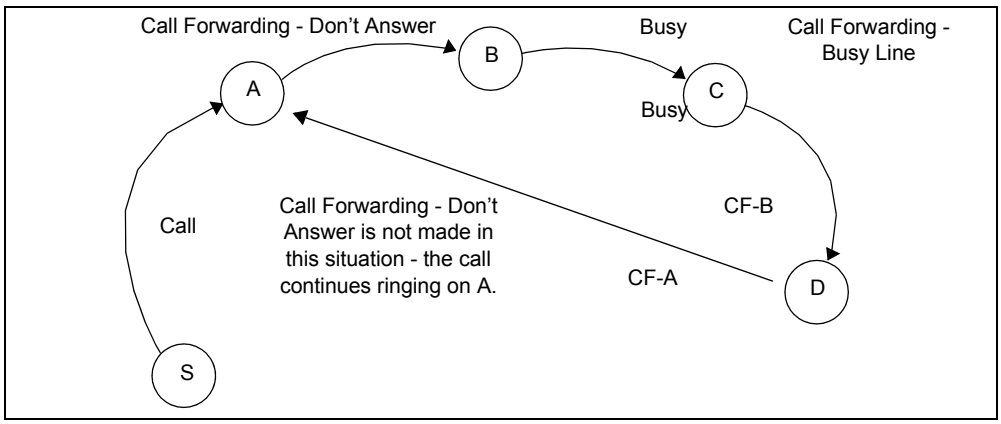
1. If the Called Station does not answer and MULTIPLE CALL FORWARDING - DON'T ANSWER [M-25] is set at the Called Station, the call is forwarded to the designated Station.
2. There is no limit to the number of times a call can be forwarded by MULTIPLE CALL FORWARDING - DON'T ANSWER [M-25].
3. A call cannot be forwarded to the originating Station.
4. CALL FORWARDING - DON'T ANSWER [C-3], CALL FORWARDING - BUSY LINE [C-2] and CALL FORWARDING - ALL CALLS [C-5] combined:
   a.) After a CALL FORWARDING - DON'T ANSWER [C-3] forward has been made, up to five CALL FORWARDING - ALL CALLS [C-5] or CALL FORWARDING - BUSY LINE [C-2] forwards may be made.
   b.) If the fifth or last CALL FORWARDING - ALL CALLS [C-5] or CALL FORWARDING - BUSY LINE [C-2] destination after a CALL FORWARDING - DON'T ANSWER [C-3] is busy, the originally Called Station will continue ringing.

c.) If a Station has CALL FORWARDING - BUSY LINE [C-2] or CALL FORWARDING - ALL CALLS [C-5] to the originating Station set, the call will not be forwarded to the originating Station. The originally Called Station is rung.

d.) If the forwarding sequence, after a CALL FORWARDING - DON'T ANSWER [C-3] goes through several Stations with CALL FORWARDING - ALL CALLS [C-5] set and ends up at the original CALL FORWARDING - DON'T ANSWER [C-3] Station, CALL FORWARDING - DON'T ANSWER [C-3] is not activated.



e.) If the forwarding sequence, after a CALL FORWARDING - DON'T ANSWER [C-3] goes through several busy Stations with CALL FORWARDING - BUSY LINE [C-2] set and ends up at the original CALL FORWARDING - DON'T ANSWER [C-3] Station, CALL FORWARDING - DON'T ANSWER [C-3] is not activated.



5. If a call is transferred blind to a Station that is CALL FORWARDED - DON'T ANSWER [C-3] to another destination (internal or external), the call will not forward but will recall back to the transferee Station. This recall parameter is controlled by System Data.

6. Forwarding to an Attendant Console:

a.) If an Attendant Console is the destination of MULTIPLE CALL FORWARDING - DON'T ANSWER [M-25], the call is forwarded to the Attendant Console that handles the Tenant to which the CALL FORWARDING - DON'T ANSWER [C-3] Station belongs.

b.) The Attendant Console key that lights when a call is forwarded to the Console depends on the forwarding status of the originally Called Station (except in case c). The following table shows the key that will light (e.g., if the forwarding status of the originally Called Station is CALL

FORWARDING - BUSY LINE [C-2], and the call is from a Station, then the BUSY key will light).

Forwarding Status Of Originally Called Station			
	Call Forward -All Calls	**Call Forward -Busy**	**Call Forward -Don't Answer**
Call From Station	ATND	BUSY	NANS
Call From Trunk	Depends on Trunk class	BUSY	NANS

c.) If the call is forwarded by CALL FORWARDING - DON'T ANSWER [C-3] at any point in a forward sequence, the NANS key on the Attendant Console will light, even if the call status of the originally Called Station is CALL FORWARDING - ALL CALLS or CALL FORWARDING - BUSY LINE.

7. Forwarding to a Dterm:
   a.) The Calling Dterm's display will show the number of the Called Station on the left, and the number of the Station being rung on the right. The display on the ringing Dterm will show the number of the originally Called Station on the left, and the number of the Calling Station on the right.
   b.) The forwarding status of the originally Called Station is shown on the far left of each Dterm's display: FDB for CALL FORWARDING - BUSY LINE or FDA for CALL FORWARDING - ALL CALLS.
8. UCD Groups:
   a.) If the destination of a CALL FORWARDING - DON'T ANSWER [C-3] is a UCD Group control Station, the call will be forwarded into the UCD Group.
   b.) If the destination of a CALL FORWARDING - BUSY LINE [C-2] or a CALL FORWARDING - ALL CALLS [C-5] forward is a UCD Group control Station, the call will be forwarded into the UCD Group.
9. Hunting:
   a.) If the final Destination Station of a MULTIPLE CALL FORWARDING-DON'T ANSWER is a controlling Station in a STATION HUNTING-TERMINAL [S-9] or a member Station of a STATION HUNTING-CIRCULAR [S-7], then hunting will take place if that Destination Station is busy.
10. An Applications Floppy Disk is needed to permit the CALL FORWARDING-DON'T ANSWER [C-3] call to be transferred up to five times by CALL FORWARDING-ALL CALLS [C-5] and CALL FORWARDING-BUSY LINE [C-2].
11. For the FCCS Network, assign ASYDN System Data 1, Index 69, Bit 7 = 1 (MULTIPLE CALL FORWARDING-DON'T ANSWER is in service).
12. The OAI monitor number and the ACD pilot number cannot be assigned as the Call Forwarding target Telephone Number.

Interactions

1. If Access Codes for CALL FORWARDING-BUSY LINE [C-2] and CALL FORWARDING-DON'T ANSWER [C-3] are the same, only one key needs to be assigned in AKYD or ADSL, and only one Access Code needs to be assigned in ASPA.

2. Any UCD Station is allowed to activate CALL FORWARDING-DON'T ANSWER [C-3]. When this Station sets CALL FORWARDING-DON'T ANSWER [C-3], Calls directed to this Station through UCD hunting or directly-dialed will ring and then follow the destination of the CALL FORWARDING-DON'T ANSWER [C-3].
3. CALL FORWARDING-DON'T ANSWER [C-3] to an Attendant Console will not operate when the system is in Night mode. If ATTENDANT NIGHT TRANSFER [A-19] is provided, the call is forwarded to the night transfer Station.
4. If the CALL FORWARDING-DON'T ANSWER [C-3] target Station is a Dterm with DO NOT DISTURB [D-21D] activated, Call Forwarding is not activated. The forwarding Station will ring beyond the timer set in ASYD, System Data 1, Index 139.
5. When the CALL FORWARDING-DON'T ANSWER [C-3] feature is set from Station A to Station B, if Station A is idle and Station B is busy, Station A will ring beyond the timer set in ASYD, System Data 1, Index 139.
6. When a call is call forwarded, the Destination Station is recorded for STATION MESSAGE DETAIL RECORDING [S-10].

Programming

STEP 1: **ASYD** - System Data 1, Index 5, Bit 0. Access Codes for CALL FORWARDING-BUSY LINE [C-2] and CALL FORWARDING-DON'T ANSWER [C-3] are same or separate? 0/1: Same/Separate.

System Data 1, Index 69, Bit 2. Is a Short Tone set to alert person receiving the call that this is a CALL FORWARDING-DON'T ANSWER [C-3] Call? 0/1: No/Yes.

System Data 1, Index 69, Bit 7. MULTIPLE CALL FORWARDING-BUSY [M-24], -ALL CALLS [M-44], in service? 0/1: No/Yes.

System Data 1, Index 139. Assign the No-Answer timer for Station-to-Station, DID and TIE Line Calls. For 30 seconds, assign 00H (RAM Data is 3FH).

System Data 1, Index 141. Assign the No Answer timer for Incoming Calls via the Attendant Console. It will then be forwarded to the next Station according to the System Data 1, Index 145 timer. For 10 seconds, assign 00H. (RAM Data is 35H.)

System Data 2, Index 6, Bit 2. Enable CALL FORWARDING-DON'T ANSWER [C-3] on a per Tenant basis.

System Data 2, Index 6, Bit 7. Enable CALL FORWARDING-DON'T ANSWER [C-3] via Attendant Console.

STEP 2: **ANPD** - Reserve a Number Level for Feature Access and cancel. Select Normal Service (N), Hooking Service (H) or Busy Service (B) from the combo box for Connection Indexes. Assign NND in accordance with a predetermined Numbering Plan.

STEP 3: **ASPA** - Assign an Access Code to CALL FORWARDING-DON'T ANSWER [C-3] entry, SID = 12 and to CALL FORWARDING-DON'T ANSWER [C-3] cancel SID = 13. Assign Connection Status Index (CI) for "N: Normal" service.

STEP 4: **ASFC** - Assign the Stations to activate CALL FORWARDING-DON'T ANSWER [C-3] a Service Feature Restriction Class that allows SFI = 8.

STEP 5: **AKYD** - For Dterm sets, CALL FORWARDING-ALL CALLS [C-5] may be assigned to a programmable Line/Feature key. Assign Type:Function:CALL FORWARD-DON'T ANSWER [22].

STEP 6: **ADSL** - Assign SN = 1 (Originating; DT Connection) and FKY = 22 (Call Forwarding - Don't Answer; FDN).

This command is used for assigning the function to a Soft key.

FCCS Programming

This data setting describes only the additional or replacement commands used to activate FCCS Services. When used as an FCCS feature, these commands replace the non-network commands (for example, AAED is replaced by AAEDN). Note that the data setting for FCCS Service varies depending on the Node.

Note: When using ASYDL or ASYDN to program an FCCS feature, the basic programming of ASYD is required in addition to ASYDL or ASYDN.

ASYDN - These data must be set to the NCN.

System Data 1 Index 69, Bit 7. Enable Multiple Call Forwarding-Busy Line/Don't Answer/All Calls. 0/1 = Out of Service/In Service

M-26 Message Center Interface

General Description

This feature provides an interface to an external CPU for Message Center information when a specific UCD [U-1] Group or ATTENDANT CONSOLE [A-3] is called. This interface allows external control of Call Indicator Lamp indications on equipped the SV9500 Stations. This feature also provides a socket interface (LAN interface) for Connection to an external LAN.

Note: Call Indicator Lamp on Desktop terminal is used as the Message Waiting Lamp.

The MESSAGE CENTER INTERFACE (MCI) can provide one of two modes of operation. These modes are described as Feature 1 and Feature 2. The functions of each feature are as follows:

Feature 1:

a.) Incoming Call information is sent to the MCI when an Attendant answers.
b.) Control of Call Indicator Lamp based on data from the external CPU.

Feature 2:

a.) Incoming Call information is sent to the MCI when a UCD Station is called.
b.) Dual Call Forwarding service is possible.
c.) Control of Call Indicator Lamp based on data from the external CPU.

Note: Feature 2 is not available when a Hotline Call terminates to a UCD Station.

Feature 1

1. When the Attendant answers a call, the System transmits a call handling record to the MCI. The external CPU, based on the received data, will display a profile of the Station/Trunk being handled by the Attendant on the CRT.
2. Call Indicator Lamp control is based on data from the external CPU. Through the external CPU terminal, when information is registered, the Host/CPU system forwards Call Indicator Lamp control data according to the Call Indicator Lamp On/Off control information sent.
3. The System, upon receiving this control data from the MCI, will extinguish or illuminate the Call Indicator Lamp, depending on the subscriber displayed on the Attendant's monitor/display.
4. The Called Station will be able to receive a message from the Attendant by dialing “0”, or a computer terminal by entering an Identification Number.
5. The external CPU, upon receiving retrieved message information, will transmit a data record requesting the System to extinguish the Call Indicator Lamp of the corresponding Station.

Feature 2

1. This provides an interface to a Voice Mail System, or a Message Center Computer that is capable of sending Call Indicator Lamp control data to the System.
2. To perform the data transfer to the external CPU, a Station/Trunk is connected to a designated UCD [U-1] Group. The Stations of the UCD [U-1] Group are connected to the Voice Mail System/Message Center System in order to collect call Connection status of Call Indicator Lamp control information. The block diagram of this concept follows on the next page.
3. The System, upon receiving this control data from the MCI, extinguishes or illuminates the Call Indicator Lamp depending upon the information displayed on the Attendant's monitor/display.
4. The Called Station will be able to receive a message by dialing “0”, or by a computer terminal by entering an Identification Number.

5. The external CPU, upon receiving retrieved message information, will transmit a data record requesting the System to extinguish the Call Indicator Lamp of the corresponding Station.

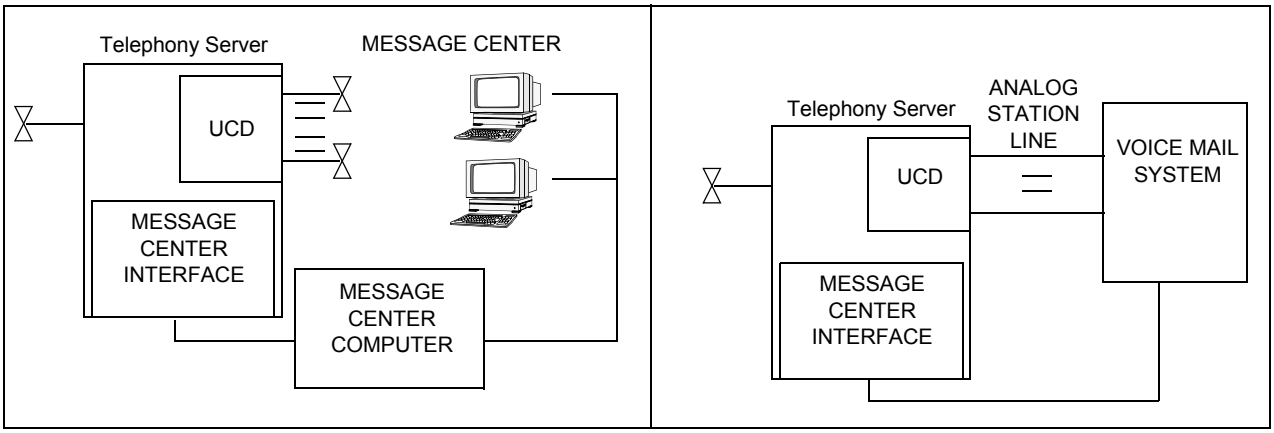
Service Conditions

<RS-232C Interface Conditions>

1. The call type is sent to the external CPU via the MCI interface, this call record will pertain to the following services:

ATTENDANT CAMP-ON	[A-1]
AUTOMATIC RECALL	[A-8]
CALL FORWARDING - DON'T ANSWER	[C-3]
CALL FORWARDING - BUSY LINE	[C-2]
CALL FORWARDING - ALL CALLS	[C-5]
Call returned from Hold	
CALL TRANSFER - ALL CALLS **Note 1**	[C-11]
CENTRALIZED ATTENDANT SERVICE (CAS)	[C-20]
DIAL ACCESS TO ATTENDANT	[D-2]
DID to the Specific UCD Group	
DIT to the Specific UCD Group	
Intercept Call to the Attendant	
INTER-POSITION TRANSFER	[I-5]
LDN Call	
OFF-HOOK ALARM	[O-6]
Recall	
Recall from a Series Call	
Series Call Re-Entry	
Tandem Transferring/Hold Entry	

Note 1: Available for Inter-Office Station transfer calls only.



2. MESSAGE CENTER INTERFACE [M-26] can control the display of a Desktop terminal for Message Indication, Message Waiting in a Message Center application.
3. If an ATTENDANT CONSOLE [A-3] Calls a UCD [U-1] Group directly, for Feature 2, the System will not generate a call record to the MCI.

4. If the Host/CPU malfunctions and stops collecting call records, the System can store up to 32 call records. If the maximum of 32 call records is stored and a 33rd is generated, the system will write over the oldest stored record, thus giving the stack the most recently generated call records.
5. MCI data will not be output when a Hotline Station is terminated to a Station that is forwarded to a Voice Mail System.
6. It can be determined on a UCD Group basis whether to send the Incoming Call information of the call terminating to a Station in the UCD Group to the message center CPU. When OVERFLOW - UCD [O-7] is activated, the MCI operation of the overflow UCD Group is followed.
7. When the Calling Party number is received via the Central Office line/ISDN line/CCIS line, the number is sent to the MCI. The Calling Number is output to the additional MCI text format.
8. The maximum of 2 Message Centers (MC) per standalone system is available. When 2 MCs are connected to IOC, each MC information is transmitted to both of them. On condition that one of them is in the event of failure, another can transmit/receive the MC information.

FCCS Conditions

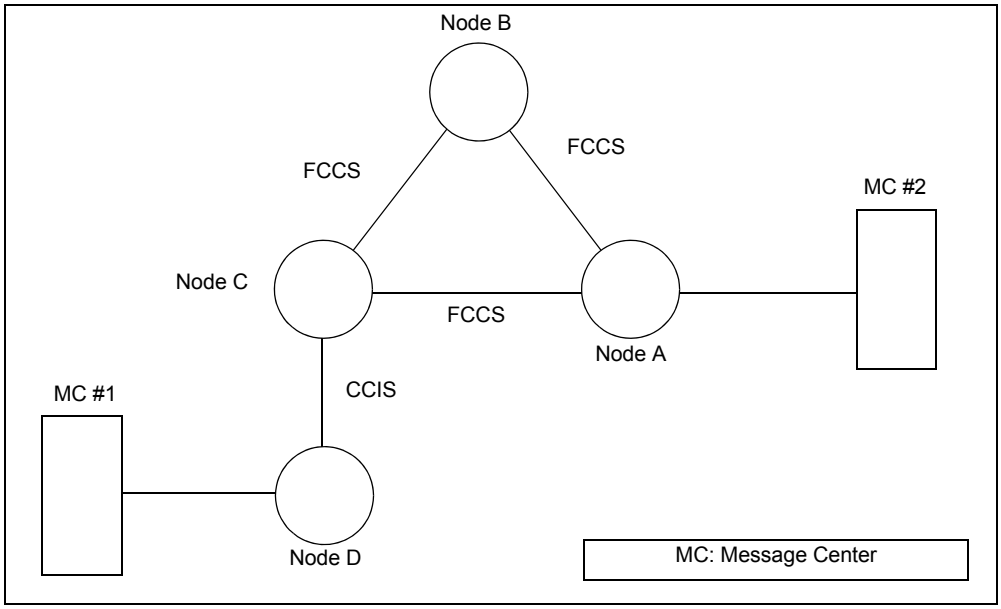
1. Station Number type (Physical Station Number or Telephone Number) to be output for MCI ICS format may be designated by the System Data.
2. Multiple Nodes within FCCS Network can accommodate the MC.
3. A maximum of two MCs per Node is available.
4. The MC Node can control the MWLs of all Station in a maximum of 32 Nodes.

SOCKET Interface (LAN interface) Conditions

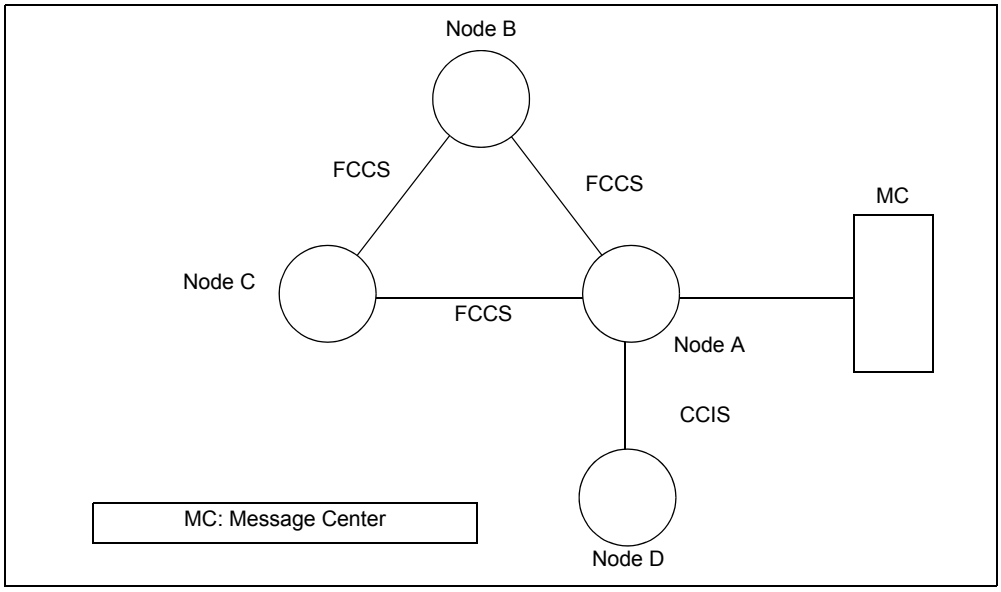
1. LAN interface and IOC interface cannot be used concurrently for MCI.
2. The socket interface will be used for transfer data to and from the Message Center connected to the LAN.
3. After the link has been released, it will always be the client which presents a link establishment request.
4. The MCI messages which are transferred via the LAN interface is IMX format messages only.
5. When 2 MCs are connected to LAN Interface, the MC information is transmitted to both MCs. When transmitting the MC information to lower number port only, assign ASYDL System Data 1 Index 833 Bit 2 = 1 (Single Output).

Interactions

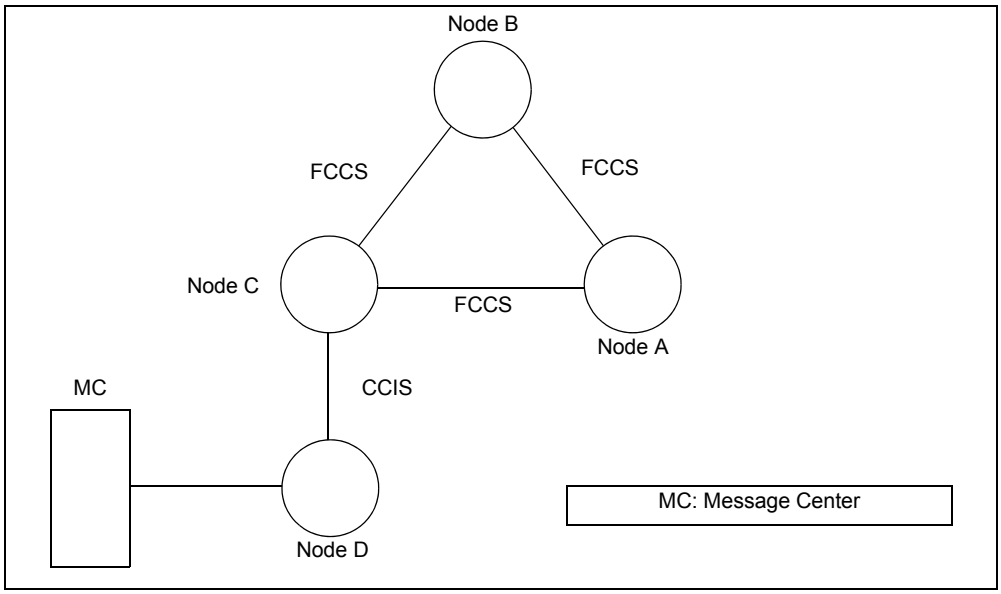
1. In case a Message Center exists on each side of the FCCS Network and CCIS Network:
   a.) Message data will be transmitted to the MC on the side of the CCIS Network (in the figure below, MC 1), provided that a call from any of the FCCS Network Nodes (in the figure, Node A/B/C) terminates to an ATTCON/UCD Station of the non-FCCS Node (in the figure, Node D).
   b.) Message data will be transmitted to the MC on the side of the FCCS Network (in the figure below, MC 2), provided that a call from the non-FCCS Node (in the figure, Node D) terminates to the ATTCON/UCD Station of any of the FCCS Network Nodes (in the figure, Node A, B, C).



2. In case a Message Center is shared by both CCIS and FCCS Networks:
   a.) Message data will be transmitted to the shared MC (in the figure below, assumed to be connected to the FCCS Node A), given that a call from the CCIS Node (in the figure, Node D) terminates to the ATTCON/UCD Station of any of the FCCS Network Nodes (in the figure, Node A, B, C).
   b.) The shared Message Center also can let the Message Waiting Lamps for both FCCS and non-FCCS (CCIS) network Stations go ON and OFF to inform the telephone user of the message arrival.

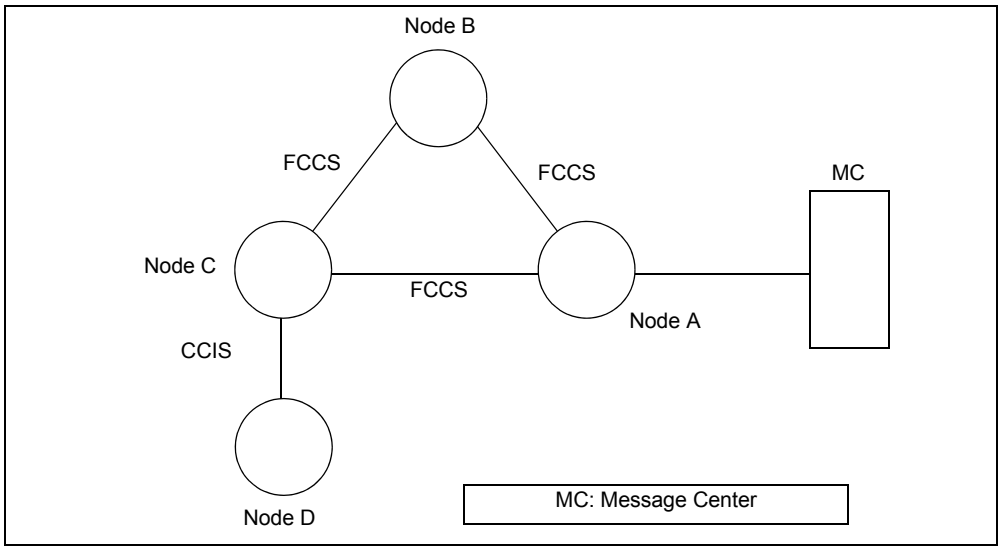


3. In case the Message Center exists only on the side of CCIS network:
   a.) Message data will be transmitted to the offered MC (in the figure below, assumed to be connected to Node D), given that a call from any of the FCCS Network Nodes (in the figure, Node A/B/C) terminates to an ATTCON/UCD Station of the CCIS Node (in the figure, Node D).
   b.) The MC, though provided Outside the FCCS Network, can also control the Message Waiting Lamp for each Station on the FCCS Service link, as well as ON/OFF indication of the lamps for Stations on the CCIS link.



4. In case the Message Center exists only on the side of FCCS Network:
   a.) Message data will be transmitted to the offered MC (in the figure below, assumed to be connected to Node A), given that a call from the CCIS Node (in the figure, Node D) terminates to the ATTCON/UCD Station of any of the FCCS Network Nodes (in the figure, Node A, B, C).

b.) The MC, though provided only Inside the FCCS Network, can also control the Message Waiting Lamp for each Station on the CCIS link, as well as ON/OFF indication of the lamps for Stations within the FCCS link.



Programming

There are two procedures for the programming depending on used interface;

- <RS-232C Interface Programming>
- <SOCKET Interface (LAN interface) Programming>

<RS-232C Interface Programming>

STEP 1: **ASYD** - System Data 1, Index 28, Bits 0-4. Assign a Miscellaneous Timer Counter (MTC) used to calculate the message-sending Guard Timer for Message Center.

Timer = MTC x 128 milliseconds. Normally a Guard Timer is not required. If not required, assign data "0" to these Bits.

System Data 1, Index 28, Bit 5. Is Message Waiting Lamp setting from the Message Center to be used? 0/1: No/Yes.

System Data 1, Index 29, Bits 0-7. Assign which I/O port will act as the MESSAGE CENTER INTERFACE. Bit 0 = Not used; Bit 1 = Port 1; Bit 2 = Port 2; Bit 3 = Port 3; Bit 4 = Port 4; Bit 5 = Port 5; Bit 6 = Port 6; Bit 7 = Port 7, for a maximum of 2 ports.

System Data 1, Index 34, Bit 1 and 2. Parity check of RS-232C output code.

Bit1, Bit2 = "00"; no parity and output (parity bit fixed to "0").

System Data 1, Index 60, Bit 3. UCD Queuing must be required. Assign Bit 3 as data "0".

System Data 1, Index 70, Bit 0. Called Number Display on the Console for DID and TIE Line Calls must be enabled. Assign Bit 0 as data "1" when System Data 2, Index 6, Bit 7 is also enabled.

System Data 1, Index 78, Bit 0. CALLING NUMBER DISPLAY -D [C-24D] must be enabled. Assign Bit 0 as data "1".

System Data 1, Index 78, Bit 1. CALLING STATION STATUS DISPLAY -D [C-22D] must be enabled. Assign Bit 1 as data "1".

System Data 1, Index 244, Bit 2. MCI-CCIS output packet change for originating party's information when the last digit is "*" (This data is valid when the originating party is a station).
System Data 1, Index 370, Bit 1: MCI Office Code Elimination is required? 0/1:No/Yes
When "1" is set office code elimination in MCI output is in service (ARNP Route 0 is referred).

Note: This feature is available in North America and Australia.

System Data 1, Index 400, Bit 2: ANI (CLI in Australia) information is sent when a DID Incoming Call terminates to ATT or a UCD station? 0/1=No/ANI (CLI in Australia) information is sent to MCI.

Note: Index 116, Port 0 is assigned as data 02H, PCPro.

System Data 2, Index 6, Bit 0. Is MCI service with UCD Groups to be enabled? 0/1: No/Yes.
System Data 2, Index 7, Bit 1. Is MCI service for calls via the Attendant Console to be enabled? 0/1: No/Yes.
System Data 1, Index 214, Bit 5. Will eliminate office codes from J record, in expanded field.

Caution: Flagging will remove office codes from phone display.

Note: Flagging will not remove office codes from SMDR records.

Note: Do not flag Index 214, Bit 5, when Index 370, Bit 1=1, as it will cause double office code elimination.

System Data 1, Index 240, Bit 4. This bit defines the meaning of the Station Information in the office that has VMS/Host Computer within the CCIS Network. 0/1: Station Information = Office Code + Station Number/Station Information = Station Number

Note: Do not flag Index 240, Bit 4, when Index 370, Bit 1=1, as it will cause double office code elimination.

STEP 2: **ASYDL** - System Data 1, Index 833, Bit 0. Interface type for MCI is "0" = RS-232C interface.
System Data 1, Index 833, Bit 1. Text format for MCI. 0/1 = ICS/IMX format.
System Data 1, Index 833, Bit 4. Output the LDN number 0/1 = ICS/IMX format.
(Called Party Information Output on Call Type 30 when call was received on ATTCON)

Note: When more than 5 digits' Telephone Number is assigned, designate "1 = IMX format".

STEP 3: **AIOC** - Assign the function and attribute data of the IOC ports.

STEP 4: **ASHU/ASHUL** - Create a group of lines.

STEP 5: **AUCD** - Assign UCD-related data to transfer the call record to the UCD group to MCI.
MCI = 0 (The call record to the UCD group is transferred to MCI.)

<SOCKET Interface (LAN interface) Programming>

STEP 1: **ASYD** - System Data 1, Index 28, Bits 0-4. Assign a Miscellaneous Timer Counter (MTC) used to calculate the message-sending Guard Timer for Message Center.

Timer = MTC x 128 milliseconds. Normally a Guard Timer is not required. If not required, assign data "0" to these Bits.

System Data 1, Index 28, Bit 5. Is Message Waiting Lamp setting from the Message Center to be used? 0/1: No/Yes.

System Data 1, Index 29, Bits 0-7. Assign which I/O port will act as the MESSAGE CENTER INTERFACE. Bit 0 = Not used; Bit 1 = Port 1; Bit 2 = Port 2; Bit 3 = Port 3; Bit 4 = Port 4; Bit 5 = Port 5; Bit 6 = Port 6; Bit 7 = Port 7, for a maximum of 2 ports.

System Data 1, Index 60, Bit 3. UCD Queuing must be required. Assign Bit 3 as data "0".

System Data 1, Index 70, Bit 0. Called Number Display on the Console for DID and TIE Line Calls must be enabled. Assign Bit 0 as data "1" when System Data 2, Index 6, Bit 7 is also enabled.

System Data 1, Index 78, Bit 0. CALLING NUMBER DISPLAY -D [C-24D] must be enabled. Assign Bit 0 as data "1".

System Data 1, Index 78, Bit 1. CALLING STATION STATUS DISPLAY -D [C-22D] must be enabled. Assign Bit 1 as data "1".

System Data 1, Index 244, Bit 2. MCI-CCIS output packet change for originating party's information when the last digit is "*" (This data is valid when the originating party is a station).

System Data 1, Index 370, Bit 1: MCI Office Code Elimination is required? 0/1:No/Yes

When "1" is set office code elimination in MCI output is in service (ARNP Route 0 is referred).

Note: This feature is available in North America and Australia.

System Data 1, Index 400, Bit 2: ANI (CLI in Australia) information is sent when a DID Incoming Call terminates to ATT or a UCD station? 0/1=No/ANI (CLI in Australia) information is sent to MCI.

Note: Index 116, Port 0 is assigned as data 02H, PCPro.

System Data 2, Index 6, Bit 0. Is MCI service with UCD Groups to be enabled? 0/1: No/Yes.

System Data 2, Index 7, Bit 1. Is MCI service for calls via the Attendant Console to be enabled? 0/1: No/Yes.

System Data 1, Index 214, Bit 5. Will eliminate office codes from J record, in expanded field.

Caution: Flagging will remove office codes from phone display.

Note: Flagging will not remove office codes from SMDR records.

Note: Do not flag Index 214, Bit 5, when Index 370, Bit 1=1, as it will cause double office code elimination.

System Data 1, Index 240, Bit 4. This bit defines the meaning of the Station Information in the office that has VMS/Host Computer within the CCIS Network. 0/1: Station Information = Office Code + Station Number/Station Information = Station Number.

Note: Do not flag Index 240, Bit 4, when Index 370, Bit 1=1, as it will cause double office code elimination.

STEP 2: **ASYDL** - System Data 1, Index 529, Bits 0 and 1. Parity check method of SMDR/MCI with LAN interface.

Bit 1, Bit 0: 00 = Odd Parity (Default)
01 = Odd Parity
10 = Even Parity
11 = No Parity

System Data 1, Index 833, Bit 0. Interface type for MCI is "1" = LAN interface.

System Data 1, Index 833, Bit 1. Text format for is "1" = IMX format.

System Data 1, Index 833, Bit 2. Assignment of how to output the Message Center Information for LAN Connection.

0/1 = Dual output/Single Output

System Data 1, Index 833, Bit 4. Output the LDN number 0/1 = ICS/IMX format.

(Called Party Information Output on Call Type 30 when call was received on ATTCON)

Note: When more than 5 digits' Telephone Number is assigned, designate "1 = IMX format".

System Data 1, Index 834, Bit 0. 0/1 = MC0 for LAN is not mounted/mounted.
System Data 1, Index 834, Bit 1. 0/1 = MC1 for LAN is not mounted/mounted.

STEP 3: **ADTM** - Specify the Port Number used for each MCI within the range of 1024 to 65534. When "0" is set, the default value (60020) is automatically used.

STEP 4: **ASHU/ASHUL** - Create a group of lines.

STEP 5: **AUCD** - Assign UCD-related data to transfer the call record to the UCD group to MCI.
MCI = 0 (The call record to the UCD group is transferred to MCI.)

FCCS Programming

This data setting describes the additional FCCS Service data after the non-FCCS Service data has been programmed. Note that the data setting for FCCS Service varies depending on the Node.

STEP 1: **ASYDL**

System Data 1, Index 641 Bit 1. Designate output numbers for MCI ICS format.

0/1 = Physical (Station) number/Telephone Number

System Data 1, Index 832. Assign the FPC (Fusion Point Code) of the Node providing the MC.

STEP 2: **AMWF** **Note 2**

Assign FPC (Fusion Point Code) of the Node for being controlled the MWL on the Desktop terminal by the MC.

Note 2: Assign this data to the Node accommodating the MC.

STEP 3: **ASHUN** - Create a group of lines.

STEP 4: **AUCDN** - Assign UCD-related data to transfer the call record to the UCD group to MCI.

MCI = 0 (The call record to the UCD group is transferred to MCI.)

M-30 Message Waiting Lamp Setting - Station

General Description

This feature allows a Station to set/cancel a Message Waiting indication or lamp. Any Single Line Telephone or Desktop terminal can set/cancel Message Waiting indication to any Single Line Telephone equipped with a 90v neon lamp, or a Desktop terminal.

Note: Call Indicator Lamp on Desktop terminal is used as the Message Waiting Lamp.

Operating Procedure

For example:

Station 200: User's Station
Station 201: Message Agent*

Note: A Message Agent is a preassigned Station allowed to set/cancel a Message Waiting indication.

1. Station 200 sets Call Forwarding to Station 201.
2. An Incoming Call to Station 200 is call forwarded to Station 201.
3. Station 201 receives a message from the Calling Party.
4. Station 201 hangs up.
5. Station 201 sets the Message Waiting indication for Station 200. (Procedure 1)
6. Station 200 sees the Message Waiting indication.
7. Station 200 calls Station 201.
8. Station 200 receives the message from Station 201.
9. The call is terminated.
10. Station 201 cancels the Message Waiting indication for Station 200. (Procedure 2)

Procedure 1:

To set a Message Waiting Indication from a Single Line Telephone

1. Lift the handset. Receive Dial Tone.
2. Dial the Message Waiting Set Access Code (up to three digits). Receive Special Dial Tone.
3. Dial the desired Telephone Number. Receive Service Set Tone.
4. Message Waiting indication is set.

To set a Message Waiting Indication from a Desktop terminal or MY Line

1. Lift the handset or press the SPEAKER key. Receive Dial Tone.
2. Press the MESSAGE WAITING Set key. Receive Special Dial Tone.
3. Dial the desired Telephone Number. The LCD displays:



   Service Set Tone is heard.
4. Message Waiting indication is set.

Procedure 2:

To cancel a Message Waiting Indication from a Single Line Telephone

1. Lift the handset. Receive Dial Tone.
2. Dial the Message Waiting Cancel Access Code (up to three digits). Receive Special Dial Tone.
3. Dial the desired Telephone Number. Receive Service Set Tone.
4. Message Waiting indication is canceled.

To cancel a Message Waiting Indication from a Desktop terminal or MY Line

1. Lift the handset or press the SPEAKER key. Receive Dial Tone.
2. Press the MESSAGE WAITING Cancel key. Receive Special Dial Tone.
3. Dial the desired Telephone Number. The LCD displays, and Service Set Tone is heard.:

MW CANCEL **(Time Display)**

4. Message Waiting indication is canceled.

Service Conditions

1. Any number of Desktop terminal stations within the system can be assigned to activate this feature.
2. Any attempt to set Message Waiting to a Station that already has a Message Waiting indication will receive Busy Tone.
3. Each Station can cancel the Message Waiting indication.
4. This feature can be restricted by CLASS OF SERVICE - INDIVIDUAL [C-15].
5. Message Waiting for one Station of the network can be controlled from the ATTENDANT CONSOLE [A-3] and any other Station at the same time.
6. The Message Waiting indication on a Station appears as follows:

Type of Station Set	Condition
Single Line Telephone with MWL Desktop terminal series without display	Message Waiting Lamp is ON
Desktop terminal series with display	Display shows: **MW** **(Time Display)**

7. The LCD on the Desktop terminal station displays the following messages:

		LCD	
MESSAGE WAITING already set	SETTING	**MW BUSY** **(Time Display)**	Busy Tone
	CANCELLING	**MW CANCEL** **(Time Display)**	Service Set Tone

		LCD	
MESSAGE WAITING not set	SETTING	**MW SET** **(Time Display)**	Service Set Tone
Restricted Station, unused Telephone Number, Dead Level, restriction	SETTING	**MW REST** **(Time Display)**	Reorder Tone
	CANCELLING	**MW REST** **(Time Display)**	Reorder Tone

Interactions

1. For MESSAGE WAITING LAMP SETTING-STATION, the 8DLC/16ELC card is required.
2. An external power source is required to light a Single Line Telephone lamp. The non-display Desktop terminals use an 80 VDC power source. NEC stocks Lorain Model HSA3H.
3. The use of the following features has no effect on this feature. MESSAGE WAITING LAMP SETTING-STATION can still be set to a Station that has previously activated any of the following features:
   - DO NOT DISTURB -D [D-21D].
   - CALL BACK -D [C-1] set to Called Station.
   - CALL BACK -D [C-1] set from Called Station.
   - CALL FORWARDING-ALL CALLS [C-5] will not follow forwarding data.
   - CALL FORWARDING-DON'T ANSWER [C-3] will not follow forwarding data.
   - CALL FORWARDING-BUSY LINE [C-2] will not follow forwarding data.
4. When an Incoming Call is answered by the Sub Line Appearance on another Station, the following services may be activated either by the answering Station or by another Station towards the answering Station. These services will not interfere with the answering Station's ability to leave a Message Waiting indication.

PRIVACY -D	[P-14D]
CONSULTATION HOLD-ALL CALLS	[C-17]
ATTENDANT CAMP-ON WITH TONE INDICATION (set to answering Station)	[A-1]
CALL WAITING-ORIGINATING (set to answering Station)	[C-31]
CALL WAITING-ORIGINATING (set from answering Station)	[C-31]
THREE-WAY CALLING	[T-2]
NON-EXCLUSIVE HOLD -D	[N-7D]
EXCLUSIVE HOLD -D	[E-4D]
DUAL HOLD	[D-32]

5. Message Waiting can be set to yourself if you wish to leave yourself a reminder.
6. Numbering Plan data (ANPD and ASPA) is not required if only Digital Terminal sets use this feature.

Programming

STEP 1: **ANPD** - Reserve a Number Level for Feature Access and cancel. Select Normal Service (N), Hooking Service (H) or Busy Service (B) from the combo box for Connection Indexes. (applicable to any feature assigned to this Level). Assign NND in accordance with a predetermined Numbering Plan.

STEP 2: **ASPA** - Assign an Access Code for setting the Message Waiting Lamp. Set SRV = "SSCA: Service Code Appendix"; SIDA = 54. Cancel Message Waiting Lamp SRV = "SSCA: Service Code Appendix"; SIDA = 55. Assign Connection Status Index (CI) for "N: Normal" service.

STEP 3: **ASFC** - Assign a Service Feature Restriction Class that allows SFI = 74 and SFI = 82 to the Stations that receive the Message Waiting Lamp setting feature.

STEP 4: **AKYD** - Assign Message Waiting Lamp set feature to a Line/Feature key. Assign Type: Function: BOSS - SECRETARY MSG WAIT-SET [9] and a MESSAGE WAITING LAMP Cancel key, Type: Function: BOSS - SECRETARY MSG WAIT-CANCEL [11].

Assign SN = 1 (Originating; DT Connection) and FKY = 11 (Boss - Secretary - Message Waiting Lamp Control; Cancel).

STEP 5: **ADSL** - Assign SN = 1, 2, 3, 4, 5 and FKY = 9 (Boss-Secretary-Message Waiting Lamp Control; Set).

Assign SN = 1, 2, 3, 4, 5 and FKY = 11 (Boss-Secretary-Message Waiting Lamp Control; Cancel).

This command is used for assigning the function to each Soft key.

FCCS Programming

This data setting describes only the additional or replacement commands used to activate FCCS Services. When used as an FCCS feature, these commands replace the non-network commands (for example, AAED is replaced by AAEDN). Note that the data setting for FCCS Service varies depending on the Node.

STEP 1: **ANPDL** - Reserve a Number Level for Feature Access and cancel. Select Normal Service (N), Hooking Service (H), or Busy Service (B) from the combo box for Connection Indexes. (applicable to any feature assigned to this Level). Assign NND in accordance with a predetermined Numbering Plan.

STEP 2: **ASPAL** - Assign an Access Code for setting the Message Waiting Lamp. Set SRV = "SSCA: Service Code Appendix"; SIDA = 54. Cancel Message Waiting Lamp SRV = "SSCA: Service Code Appendix"; SIDA = 55. Assign Connection Status Index (CI) for "N: Normal" service.

M-44 Multiple Call Forwarding - All Calls

General Description

This feature permits a call to a CALL FORWARDING - ALL CALLS Station to be forwarded multiple times to a predesignated Idle Station.

Operating Procedure

To set and cancel from an individual Station using the same Access Code Note 1

- For setting:

1. Lift the handset. Receive Dial Tone.
2. Dial the CALL FORWARDING - ALL CALLS entry Access Code. Receive Special Dial Tone.
3. Dial the desired Forwarding Destination Number. Receive Service Set Tone.

- For cancellation:

1. Lift the handset. Receive Dial Tone.
2. Dial the CALL FORWARDING - ALL CALLS entry Access Code. Receive Special Dial Tone.

Note 1: Available from FP95-114 V4 Issue 2.0 with system data ASYDL, SYS1, Index 686, Bit 0= 1.

To set and cancel from an individual Station using different Access Codes

- For setting:

1. Lift the handset. Receive Dial Tone.
2. Dial the CALL FORWARDING - ALL CALLS entry Access Code. Receive Special Dial Tone.
3. Dial the desired Forwarding Destination Number. Receive Service Set Tone.

- For canceling:

1. Lift the handset. Receive Dial Tone.
2. Dial the CALL FORWARDING - ALL CALLS cancel Access Code. Receive Service Set Tone.

To set from the Attendant Console

1. Press an Idle Loop key.
2. Dial the CALL FORWARDING - ALL CALLS entry Access Code. Receive Special Dial Tone.
3. Dial the TENANT [T-12] number (two or three digits).
4. Dial the originating Telephone Number.
5. Dial the desired Forwarding Destination Number. Receive Service Set Tone.

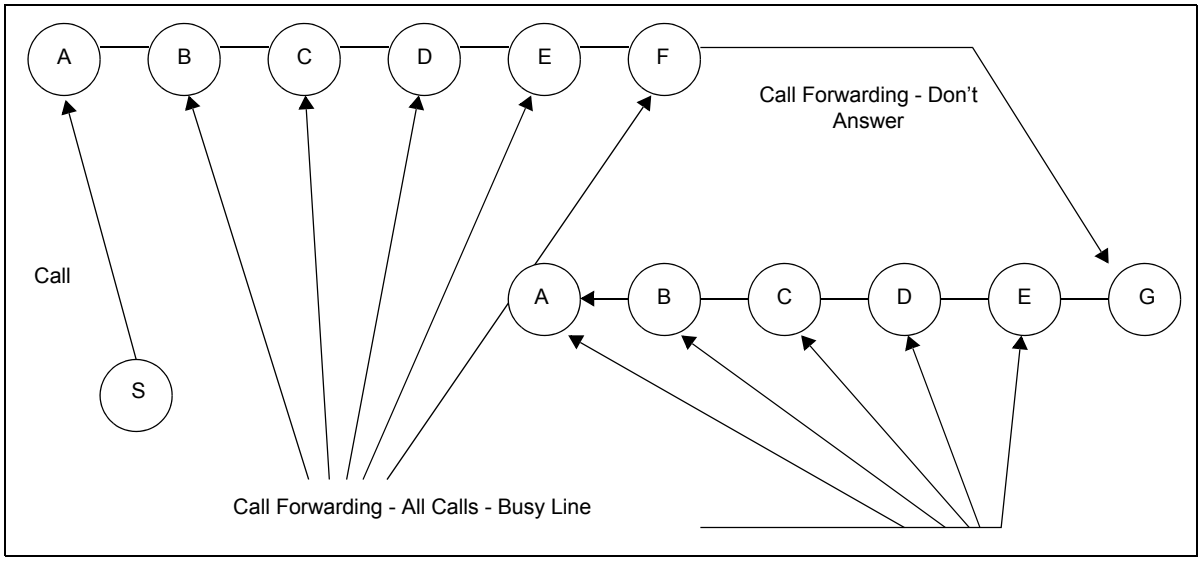
To cancel from the Attendant Console

1. Press an Idle Loop key.
2. Dial the CALL FORWARDING - ALL CALLS cancel Access Code. Receive Special Dial Tone.
3. Dial the TENANT [T-12] number (two or three digits).
4. Dial the originating Telephone Number. Receive Service Set Tone.

Service Conditions

1. If CALL FORWARDING - ALL CALLS [C-5] is set at the Called Station, the call is forwarded to a designated Station that has set CALL FORWARDING-ALL CALLS [C-5] or CALL FORWARDING-BUSY LINE [C-2].
2. A call can be forwarded up to five times, maximum, by MULTIPLE CALL FORWARDING - ALL CALLS [C-5], and only by the voice port of a Dterm, or by a Single Line Telephone.
3. A call will not forward any farther than the Station to where it has been forwarded to.
4. If a forward destination is busy, and if neither CALL FORWARDING - BUSY LINE [C-2] nor CALL FORWARDING - ALL CALLS [C-5] is set, the caller hears a Busy Tone.
5. If the originating Station is designated as the forward destination, the call will not be forwarded.
6. CALL FORWARDING - BUSY LINE [C-2] and CALL FORWARDING - ALL CALLS [C-5] combination:
   a.) A maximum of five forwarding operations can be performed by any combination of CALL FORWARDING-ALL CALLS [C-5] and CALL FORWARDING-BUSY LINE [C-2].
   b.) If the fifth forward destination is busy, the call will forward if CALL FORWARDING - DON'T ANSWER [C-3] is set. Otherwise, the caller will hear a Busy Tone.
   c.) If a sequence of CALL FORWARDING - ALL CALLS [C-5] and CALL FORWARDING - BUSY LINE [C-2] forwards the call back to the originating Station, the forward is not provided.
   d.) If a sequence of CALL FORWARDING - ALL CALLS [C-5] and CALL FORWARDING - BUSY LINE [C-2] forwards the call back to the originally Called Station, and if that Station forwards the call with CALL FORWARDING - BUSY LINE [C-2], the ultimate destination will be the originally Called Station. If, however, the originally Called Station forwards the call with CALL FORWARDING - ALL CALLS [C-5], the ultimate destination will be the Station it has designated as its CALL FORWARDING - ALL CALLS [C-5] destination.
   e.) If the call is forwarded back to the Station where it has been forwarded to, the CALL FORWARD will terminate at that Station.
7. CALL FORWARDING - ALL CALLS [C-5], CALL FORWARDING - BUSY LINE [C-2] and CALL FORWARDING - DON'T ANSWER [C-3] combination:

   Although CALL FORWARDING - ALL CALLS [C-5], CALL FORWARDING - BUSY LINE [C-2] and combinations of the two are limited to forwarding five times, when a sequence of forwards is divided by CALL FORWARDING - DON'T ANSWER [C-3], CALL FORWARDING - ALL CALLS [C-5] and CALL FORWARDING - BUSY LINE [C-3], forwards can begin again (the only limit is shown below).

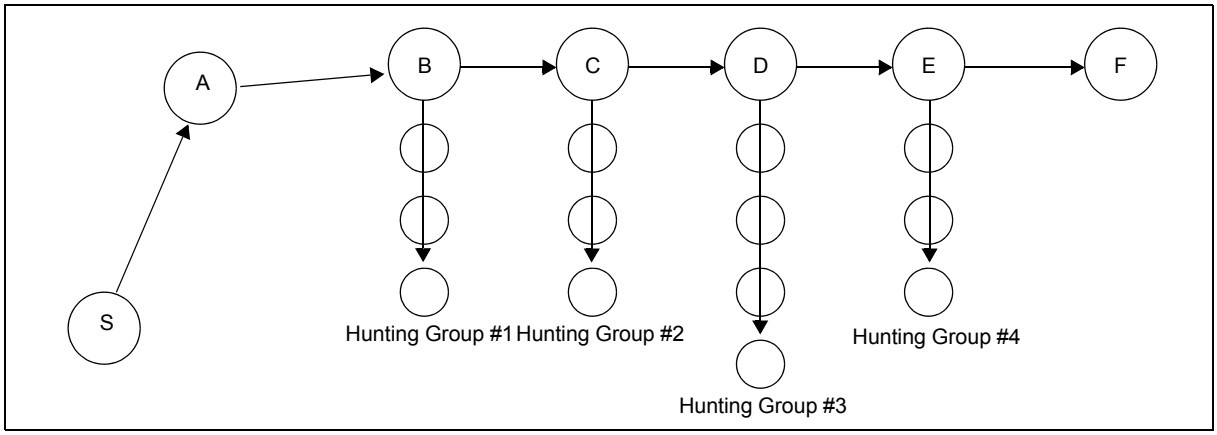


8. Forwarding to an Attendant Console:
   a.) If an Attendant Console is designated as the destination of CALL FORWARDING - ALL CALLS [C-5], the call is forwarded to the Attendant Console handling the Tenant to which the Calling Station belongs.
   b.) The Attendant Console key that lights when a call is forwarded to the Console depends on the forwarding status of the originally Called Station (except in paragraph c). The following table shows the key that will light; e.g., the BUSY key will light, if the forwarding status of the originally Called Station is CALL FORWARDING - BUSY LINE [C-2], and the call is from a Station.

	Forwarding Status Of Originally Called Station		
	Call Forward -All Calls	**Call Forward -Busy**	**Call Forward -Don't Answer**
Call From Station	ATND	BUSY	NANS
Call From Trunk	Depends on Trunk class	BUSY	NANS

   c.) If the call is forwarded by CALL FORWARDING - DON'T ANSWER [C-3] at any point in a forwarding sequence, the NANS key on the Attendant Console will light, even if the call status of the originally Called Station is CALL FORWARDING - ALL CALLS or CALL FORWARDING - BUSY LINE.
   d.) Called Number Display is only provided if a call is forwarded to the Attendant Console by CALL FORWARDING - ALL CALLS [C-5] or by CALL FORWARDING - BUSY LINE [C-2].
9. Forwarding to a Dterm:
   a.) The Calling Dterm's display will show the number of the Called Station on the left, and the number of the Station being called on the right. The display on the ringing Dterm will show the number of the originally Called Station on the left, and the number of the Calling Station on the right.
   b.) The forward status of the originally Called Station is shown on the far left of each Dterm's display: FDB for CALL FORWARDING - BUSY LINE [C-2]; FDA for CALL FORWARDING - ALL CALLS [C-5].

10. UCD Groups:
    a.) If the destination of a CALL FORWARDING - ALL CALLS [C-5], CALL FORWARDING - BUSY LINE [C-2], or CALL FORWARDING - DON'T ANSWER [C-3] is a UNIFORM CALL DISTRIBUTION (UCD) [U-1] Group control Station, and CALL FORWARDING - ALL CALLS [C-5] is not set at that Station, the call will be forwarded according to UCD [U-1] procedures.
    b.) In the UCD procedure, if CALL FORWARDING - ALL CALLS [C-5] is set on a Station in the group, that Station is skipped.
11. Hunting:
    a.) If the final Destination Station is busy, or if some condition prevents the call from being forwarded, a busy line is selected, and a hunting check is performed.
    b.) If a call is forwarded by CALL FORWARDING - BUSY LINE [C-2] to Stations that are members of Hunting Groups, a hunting check is performed on the busy line closest to the Called Station, or on the busy line closest to the final forward destination. In the diagram below, a hunting check is performed on Hunt Groups 1 or 4. A hunting check is never performed on groups 2 or 3.



12. Busy Services, such as, EXECUTIVE RIGHT OF WAY [E-1], CALL BACK [C-1], etc.:
    a.) Busy service is provided to the busy line closest to the originally Called Station.
13. DIT, NIGHT CONNECTION - FIXED [N-1]:
    a.) For CALL FORWARDING - ALL CALLS [C-5] and CALL FORWARDING - BUSY LINE [C-2], a forwarding check is made up to 5 times by System Data.
    b.) If a line is busy in DIRECT-IN TERMINATION [D-7], CALL FORWARDING - BUSY LINE [C-2] to the Attendant is not activated. Instead, the destination for the call can be specified by System Data:
    System Data 2 Index = 11
    Bit 7 = 0 - Call is routed to the Attendant Console
    Bit 7 = 1 - Call is placed in queue until the Called Station becomes idle.
14. When activating this feature via FCCS, a Telephone Number must be set to the Call Forwarding Destination.
15. When activating this feature via FCCS, a call cannot be forwarded more than five times.
16. When activating this feature via FCCS, the number of times is assigned by System Data.
17. The OAI monitor number and the ACD pilot number cannot be assigned as the Call Forwarding target Telephone Number.

18. By registering the Multiple Call Forwarding - All Calls Access Code + the desired Forwarding Destination Number to the CF Setting button of a Line/Feature key on a Desktop terminal, it is possible to enable and disable MULTIPLE CALL FORWARDING - ALL CALLS [M-44] just by pressing the CF Setting button. For a detailed explanation, see CALL FORWARDING (CF) BUTTON - ONE TOUCH [C-208].

Note: CALL FORWARDING (CF) BUTTON - ONE TOUCH [C-208] is available from FP95-114 V4 Issue 2.0.

Interactions

1. Station users are unaccustomed to hearing Ringback Tone when responding to their own Station's ring. Be sure to inform users when providing Ringback Tone with MULTIPLE CALL FORWARDING-ALL CALLS in System Data 1, Index 69, Bit 1.
2. MULTIPLE CALL FORWARDING can only be activated from an Idle Station.
3. When CALL WAITING [C-12, 31] ATTENDANT CAMP-ON WITH TONE INDICATION [A-1], and CALL BACK [C-1] is directed toward Station A with CALL FORWARDING set to Station B, Station B will receive CALL WAITING tones.
4. When a Station in a STATION HUNTING [S-7, 8, 9] Group initiates MULTIPLE CALL FORWARDING-ALL CALLS, calls normally destined for this Station will not be received. The Hunt Group operates as if the Station were busy.
5. A non-controlling UCD Station is allowed to activate CALL FORWARDING [C-5]. With this Station call forwarded, only calls dialed directly to this Station are call forwarded. The member Station is removed from the UCD Group under the conditions of this Call Forwarding feature.
6. If an Attendant attempts BUSY VERIFICATION [B-3] to Station A, which has MULTIPLE CALL FORWARDING-ALL CALLS set to Station B, BUSY VERIFICATION proceeds as if the Attendant had attempted to verify Station B. If Station A has set MULTIPLE CALL FORWARDING-ALL CALLS to the Attendant Console [A-3], BUSY VERIFICATION [B-3] is allowed at Station A, if Station A is busy.
7. When MULTIPLE CALL FORWARDING-ALL CALLS is in effect, the activating Station may be used to originate calls. See System Data 2, Index 6, Bit 4.
8. When a Station is call forwarded, the Destination Station is recorded for STATION MESSAGE DETAIL RECORDING [S-10].
9. MULTIPLE CALL FORWARDING-ALL CALLS can be overridden, allowing Station B to initiate or transfer calls back to Station A.
10. STATION-TO-STATION CALLING [S-11] must be provided first.

Programming

STEP 1: **ASYD** - System Data 1, Index 4, Bit 6. One burst of ringing at forwarding Station when MULTIPLE CALL FORWARDING-ALL CALLS is in service? 0/1: No/Yes.

System Data 1, Index 69, Bit 1. A burst of Ringback Tone to alert the person receiving a call that this is a MULTIPLE CALL FORWARDING-ALL CALLS Call. In service? 0/1: No/Yes. Normally assigned as data "0".

System Data 1, Index 69, Bit 7. MULTIPLE CALL FORWARDING - BUSY LINE [M-24], - ALL CALLS [M-44], in service? 0/1: No/Yes.

System Data 2, Index 6, Bit 4. Enable Stations set for MULTIPLE CALL FORWARDING-ALL CALLS be allowed to use their phones normally. If data "0" is assigned, the Station will only be able to call the Attendant Console. Assign on a per Tenant basis.

STEP 2: **ASYDL/N** - SYS1, Index 686, Bit 0=1.
Disabling Call Forwarding - All Calls with entry Access Code is in service.

Note: Available from FP95-114 V4 Issue 2.0.

STEP 3: **ANPD** - Reserve a Number Level for Feature Access and cancel (**Note 2**). Select Normal Service (N), Hooking Service (H) or Busy Service (B) from the combo box for Connection Indexes. Assign NND in accordance with a predetermined Numbering Plan.

STEP 4: **ASPA** - Assign an Access Code to MULTIPLE CALL FORWARDING-ALL CALLS entry, SID = 8 and to MULTIPLE CALL FORWARDING-ALL CALLS cancel SID = 9 (**Note 2**). Assign a Connection Status Index (CI) for "N: Normal" service.

Note 2: When SYS1, Index 686, Bit 0=1 (Disabling Call Forwarding - All Calls with entry Access Code is in service), Call Forwarding - All calls cancel is not required because the cancellation can be performed with the entry Access Code.

STEP 5: **ASFC** - Assign the Stations to activate MULTIPLE CALL FORWARDING-ALL CALLS a Service Feature Restriction Class that allows SFI = 7.

STEP 6: **AKYD** - For Dterm sets, MULTIPLE CALL FORWARDING-ALL CALLS may be assigned to a Line/Feature key. AssignType:Function:CALL FORWARDING-ALL CALLS [2].

STEP 7: **ADSL** - SN = 1 (Originating; DT Connection) and FKY = 2 (Call Forwarding - All Calls; FDA).

This command is required only for assigning the function to each Soft key.

STEP 8: **ATNR** - Allow Tenant-to-Tenant Connection for Inter and Intra-Tenant Connections. Assign Tenant Restriction Index (TRI) = 1. For this feature to be set by the Attendant Console, assign Inter and Intra-Tenant Connection via the Attendant Console TRI = 3. Also allow TRI = 0, Station-to-Station Calling.

FCCS Programming

This data setting describes only the additional or replacement commands used to activate FCCS Services. When used as an FCCS feature, these commands replace the non-network commands (for example, AAED is replaced by AAEDN). Note that the data setting for FCCS Service varies depending on the Node.

Note: When using ASYDL or ASYDN to program an FCCS feature, the basic programming of ASYD is required in addition to ASYDL or ASYDN.

ASYDN - These data must be set to the NCN.

System Data 1 Index 69, Bit 7. Enable Multiple Call Forwarding-Busy Line/Don't Answer/All Calls. 0/1 = Out of Service/In Service. Index5, Bit 1-3. Assign the number of time for forwarding.

M-47 Manual Signaling

General Description

This feature permits Desktop terminal users to send a one-second ring to a predetermined Station. The manual intercom SIGNAL key (SIG key) is operational at all times. An Intercom Call is not required. If the signaled Station is ringing from another call, the manual intercom signal will interrupt that ringing.

Operating Procedure

Press the SIG key. Send a one-second ring to the predetermined Station.

Service Conditions

1. When MANUAL SIGNALING is initiated by a Station user, this one-second ring takes priority over any other Incoming Call indications from the Multi Line Appearances.
2. Only Dterm can be assigned a manual SIG key.
3. One Destination Station can be assigned for manual signaling through the PCPro.
4. Stations that can signal each other must be assigned in the same Module Group. **Note 1**
5. A Station can manually signal one other Dterm.
6. MANUAL SIGNALING is an operation of MANUAL INTERCOM [A-30] and is used independently of the MANUAL INTERCOM [A-30] feature.

Note 1: This is also applied to FCCS Service.

Programming

STEP 1: **ASYD** - System Data 1, Index 59, Bit 7. Is AUTOMATIC/MANUAL INTERCOM [A-30] in service? 0/1: No/Yes.

STEP 2: **AICD** - For a Station to manually signal another Station, an AUTOMATIC/MANUAL INTERCOM [A-30] Group must be assigned. Bridge Call is not required. The maximum number of Stations per group may be assigned. Assign the intercom Station Number as the signaling Station and the signaling Station Number as the signaled Station.

MG: Module Group
G-ID: Group ID Number per MG
A/M: Automatic/Manual Signaling (0/1)
0: Automatic
→ 1: Manual
BCR: Bridge Call Restriction (0/1)
→ 0: Restricted
1: Allowed
TN: Tenant Number
→ ICM STN: Intercom Station Number
→ SIG STN: Signaling Station Number

STEP 3: **AKYD** - Assign to the Station an Intercom Line according to the following parameters:

TN: 1
STN: Station Number

TP: Type of Dterm (0-3)
PRI: Form of Line Preference (0-3)
KYN: Key Number (1-40)
KYI: Service Index
0: Key Not Used
1: Feature key
→ 2: Multi Line key
KD: Kind of Line
0: Multi Line
→ 1: Intercom Line
2: DSS Console key
3: -
4: Multiple Line Appearance
→ ICM: Kind of Intercom
0: Automatic Intercom
1: Manual Intercom
2: Dial Intercom

If MANUAL SIGNALING is used, assign a programmable Line/Feature key as Type: Function: MANUAL SIGNALING KEY [60].

STEP 4: **ADSL** - Assign SN = 0, 1, 2, 3, 4, 5, 6, 7, 15 and FKY = 60 (Manual Signaling key; SIG).
This command is used only for assigning the function to each Soft key.

M-48 Multi Channel Recording - DAT

General Description

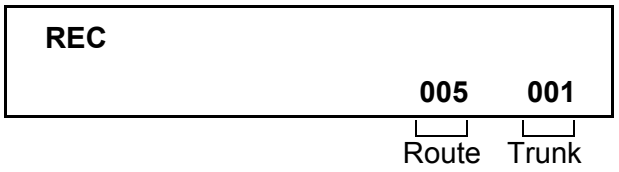
This feature allows a Station user to record a message to multiple Digital Announcement Trunks (DATs) in a single operation. The message recorded in a Trunk is automatically copied to other Trunks one after another.

Operating Procedure

1. Recording by designating RT and TK
   a.) Lift a handset. Receive Dial Tone.
   b.) Dial the MULTI CHANNEL RECORDING (RECORDING) Access Code from the Station. Receive Special Dial Tone.



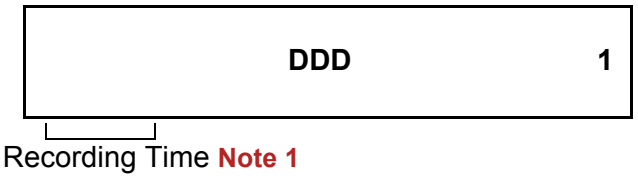
   c.) Enter three-digit Route Number and three-digit Trunk Number. The Special Dial Tone stops.



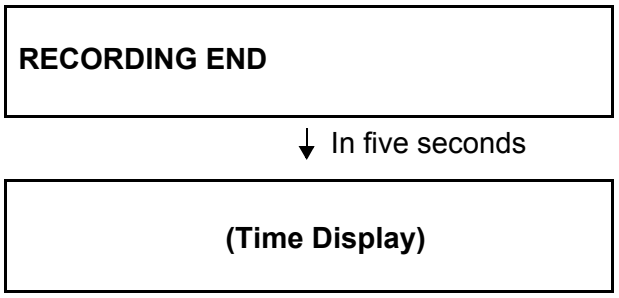
   Hear Service Set Tone when # is pressed. The Service Set Tone continues for three seconds.



   d.) Start your recording when the Service Set Tone is stopped.



   e.) Replace the handset to finish recording.

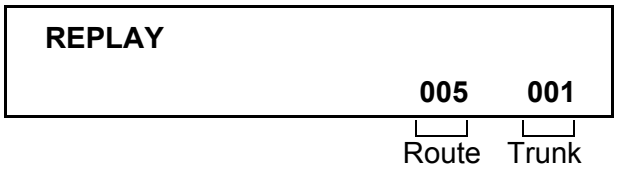


Note 1: The recording length is a multiple of 2 (= 2n seconds).

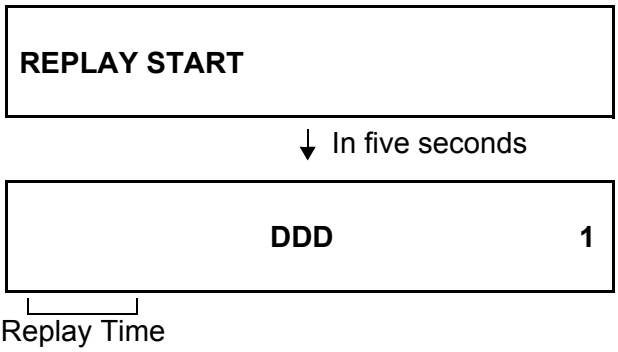
2. Playback by designating RT and TK
   a.) Lift a handset. Receive Dial Tone.
   b.) Dial the MULTI CHANNEL RECORDING (REPLAY) Access Code from the Station. Receive Special Dial Tone.



   c.) Enter three-digit Route Number and three-digit Trunk Number. The Special Dial Tone stops.



   d.) A recorded message of the designated Trunk is automatically played back.



   e.) Replace the handset to finish playback of the recorded message.

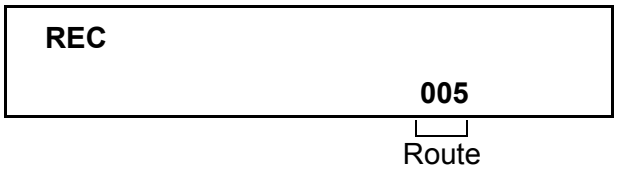


Note: Even though recording of a message has been finished, playback of the Trunk cannot be accepted until dubbing of the message to the remaining Trunks is complete.

3. Recording by designating RT
   (All Trunks within the designated route are recorded.)
   a.) Lift a handset. Receive Dial Tone.
   b.) Dial the MULTI CHANNEL RECORDING (RECORDING) Access Code from the Station. Receive Special Dial Tone.



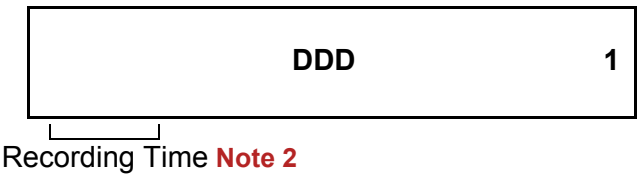
c.) Enter three-digit Route Number and press #; the Special Dial Tone stops.



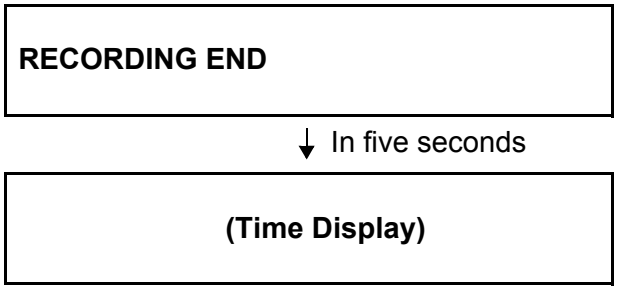
Hear Service Set Tone when # is pressed. The Service Set Tone continues for three seconds.



d.) Start your recording when the Service Set Tone stops.



e.) Replace the handset to finish recording.

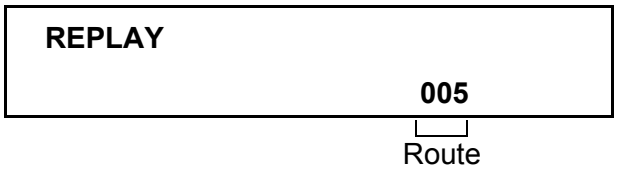


Note 2: The recording length is a multiple of 2 (= 2n seconds).

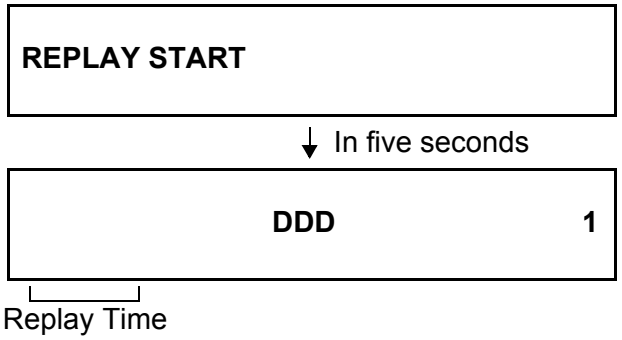
4. Playback by designating RT
   a.) Lift a handset. Receive Dial Tone.
   b.) Dial the MULTI CHANNEL RECORDING (REPLAY) Access Code from the Station. Receive Special Dial Tone.)



   c.) Enter Route Number and press #; the Special Dial Tone stops.



d.) A recorded message of the designated route is automatically played back.



e.) Replace the handset to finish playback of the recorded message.



Note: Even though the recording of a message has been finished, playback of the Trunk cannot be accepted until dubbing of the message to the remaining Trunks is complete.

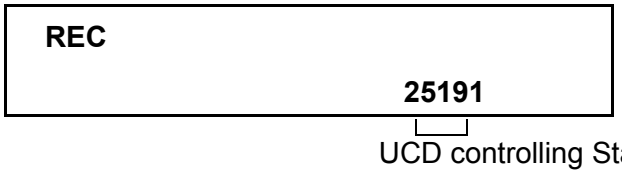
5. Recording of an announcement for a UCD Group

a.) Lift a handset. Receive Dial Tone.

b.) Dial the MULTI CHANNEL RECORDING (RECORDING) Access Code from the Station. Receive Special Dial Tone.



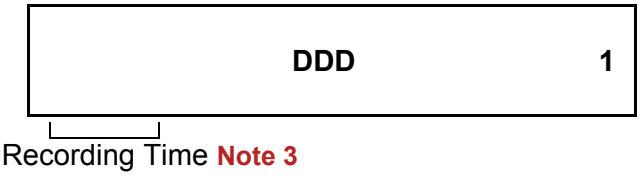
c.) Press * and enter the UCD controlling Station Number.



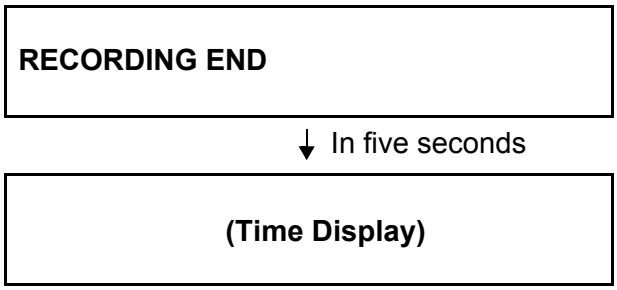
d.) Wait until you hear Service Set Tone. The waiting time depends upon Register Inter Digit Timer (normally six seconds).
The Service Set Tone continues for three seconds.



e.) Start your recording when the Service Set Tone stops.



f.) Replace the handset to finish recording.



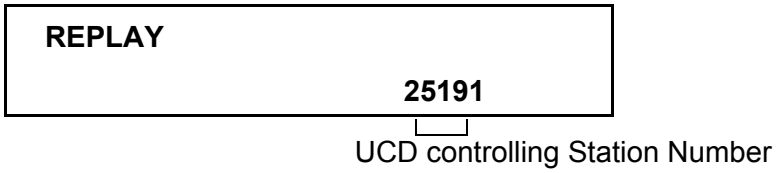
Note: Programming of Delay Announcement-UCD[D-31] needs to be assigned when performing this operation.

Note 3: The recording length is a multiple of 2 (= 2n seconds).

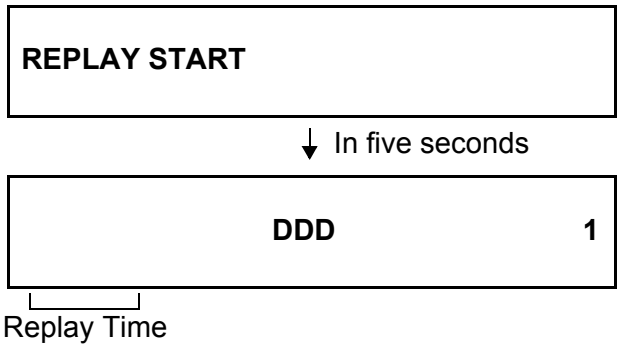
6. Playback of a message recorded for a UCD Group
   a.) Lift a handset. Receive Dial Tone.
   b.) Dial the MULTI CHANNEL RECORDING (REPLAY) Access Code from the Station. Receive Special Dial Tone.



   c.) Press * and enter the UCD controlling Station Number. The Special Dial Tone stops.



   d.) A recorded message of the designated UCD Group is automatically played back.



   e.) Replace the handset to finish playback of the recorded message.



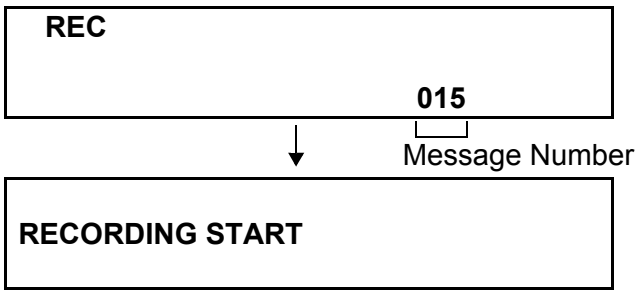
Note: Even though the recording of a message has been finished, playback of the Trunk cannot be accepted until dubbing of the message to the remaining Trunks is complete.

Note: Programming of Delay Announcement-UCD[D-31] needs to be assigned when performing this operation.

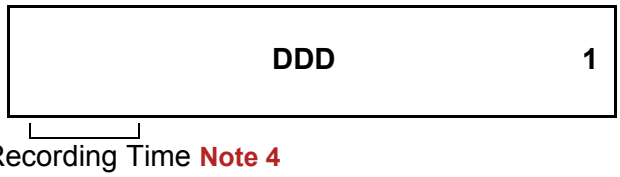
7. Recording of a message for ACD
   a.) Lift a handset. Receive Dial Tone.
   b.) Dial the MULTI CHANNEL RECORDING (RECORDING) Access Code from the Station. Receive Special Dial Tone.



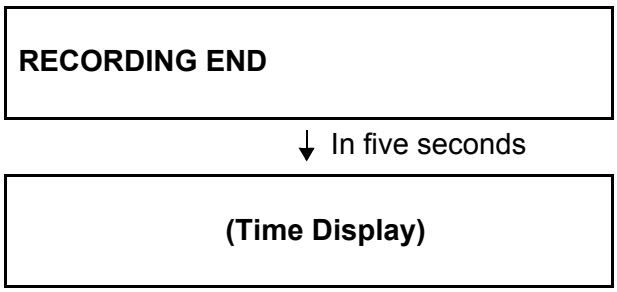
   c.) Press # and enter the three-digit message number that has been assigned in the AADT command. Receive Service Set Tone. The Service Set Tone continues for three seconds.



   d.) Start your recording when the Service Set Tone stops.



   e.) Replace the handset to finish recording.



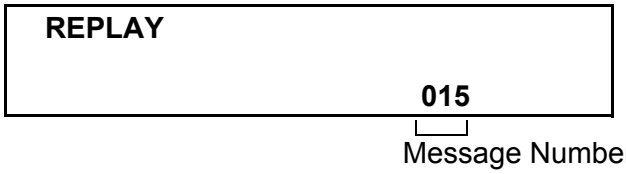
Note 4: The recording length is a multiple of 2 (= 2n seconds).

8. Playback of a message for ACD
   a.) Lift a handset. Receive Dial Tone.
   b.) Dial the MULTI CHANNEL RECORDING (REPLAY) Access Code from the Station. Receive

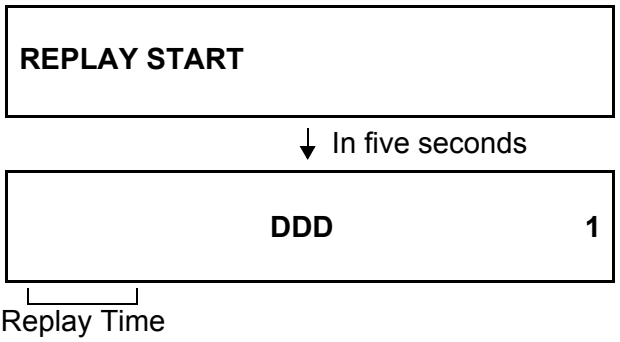
Special Dial Tone.



c.) Press # and enter the Message Number that has been assigned in the AADT command.



d.) A recorded message of the designated Message Number for ACD is automatically played back.



e.) Replace the handset to finish playback of the recorded message.



Note: Even though the recording of a message has been finished, playback of the Trunk cannot be accepted until dubbing of the message to the remaining Trunks is complete.

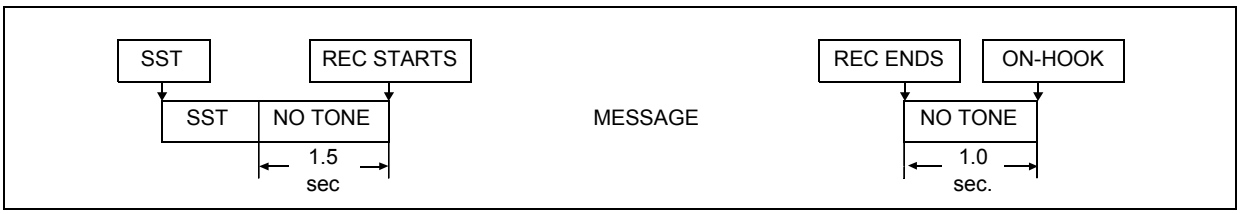
Service Conditions

1. Any entry of RT, TK, UCD Controlling Station, and Message Number that has not been assigned results in Reorder Tone (ROT) Connection.

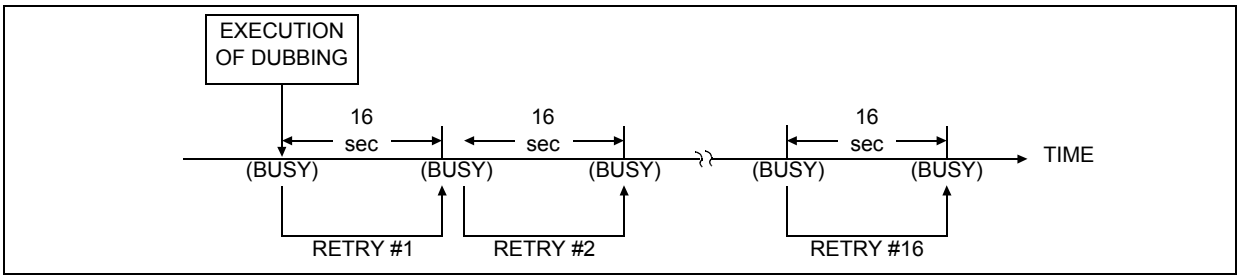
   The values that each parameter can take fall into the ranges:

   RT: 001 - 999
   TK: 000 - 255
   Message Number: 001 - 058
2. The maximum recording length is 60 seconds (30 seconds in the case of UCD).
3. This feature is not effective for Trunks being dubbed or placed in busy status.
4. A Station user cannot record a message to a Trunk in which a recording/dubbing is being made.
5. When a Station user encounters that all Digital Announcement Trunk are busy, after dialing RT/TK/UCD Controlling Telephone Number/Message Number from the terminal, the Station user hears a Busy Tone.

6. After completion of dialing RT/TK/UCD Controlling Telephone Number/Message Number, a Station user hears the Service Set Tone. In this instance, if the designated Trunk is making a dub or being dubbed, the dubbing is stopped. Also, when the Station user goes on-hook during the Service Set Tone, the dubbing is also stopped.
7. A message is recorded in the following sequence. As illustrated below, 1.5 seconds after the Service Set Tone and one second before going on-hook are not recorded.



8. When recording messages to Announcement Trunk assigned to UCD group or playing its recorded message, programming of Delay Announcement-UCD[D-31] needs to be assigned.
9. When the system encounters a busy Trunk in an attempt to make a dubbing of the message to the remaining Trunks, the Trunk is retried after a 16 second interval. Each Trunk is attempted a maximum of 16 times. If the final attempt is rejected because of busy, no further dubbing attempts will be made.



10. Only the portion in which a message has been recorded is played back.
11. The dubbing time is calculated by the following equation: (Dubbing Time) = (Recording time + two seconds) x (Number of Trunk). Refer to the following table to obtain the approximate time for dubbing:

Dubbing Time Calculation

Recording Time	Number of Trunk					
	1	10	15	20	63	n
6 seconds	8	80	120	160	504	8n
16 seconds	18	180	270	360	1134	18n
30 seconds	32	320	480	640	2016	32n
46 seconds	48	480	720	960	3024	48n
60 seconds	62	620	930	1240	3906	62n

12. This feature is available to access a 4-DAT Trunk through FCCS link.
13. When the first Trunk (CNT1), assigned by AADT/AUDT command is busy, the user cannot record or play a message.

14. When a CH-DATA card is used, trunks can be registered to the second half of Levels (from Level 4 to Level 7). However, recording messages to trunks registered to these levels is not possible. To perform multi channel recording with a CH-DATA card, follow one of the operations below.

    - Perform 3. Recording by designating RT by registering trunks only from Level 0 to Level 3.
    - Perform 5. Recording of an announcement for a UCD Group by registering trunks only from Level 0 to Level 3.
    - Perform 7. Recording of a message for ACD by registering trunks only from Level 0 to Level 3.
    - After the recording in any of the operations above is completed, perform additional recording operations to trunks registered to the second half of Levels (from Level 4 to Level 7) with 1. Recording by designating RT and TK.

Programming

STEP 1: **ANPD** - Assign Number of Necessary Digits (NND) for the first digit of the Access Code of MULTI CHANNEL RECORDING.

STEP 2: **ASPA** - Assign Access Codes for the following Service Code Appendix (SSCA).

SRV "SSCA: Service Code Appendix"
SIDA 66 (= MULTI CHANNEL RECORDING <REC>)
68 (= MULTI CHANNEL RECORDING <RPY>)

STEP 3: **ASFC** - For this feature, Service Feature Restriction data must be entered on a Service Feature Restriction Class (SFC) basis.

SFI
87 (= MULTI CHANNEL RECORDING <REC>)
90 (= MULTI CHANNEL RECORDING <REPLAY>)

STEP 4: **ARTD** - Assign Route Class Data for Announcement as shown below.

RT: 1

1- OSGS:2	2-ONSG:3	5-TF:1	6-TCL:1	7-L/T:1	8-RLP:2
15-LSG:1	34-GUARD:1 **Note 5**		44-PRV:1		

Note: To change the data for existing route (RT), initialize the circuit card.

Note 5: When recording by designating route, GUARD must equal "1".

STEP 5: **ATRK** - Assign the Announcement Trunk data.

STEP 6: **MBTK** - Assign the Make Idle status to the Announcement Trunk.

STEP 7: **APAD** - Assign the PAD control data.

KIND
T (= Assignment of PAD data for dubbing)
S (= Assignment of PAD data for playback)

Note: When recording messages to Announcement Trunk assigned to UCD group or playing its recorded message, programming of Delay Announcement-UCD[D-31] needs to be assigned.

M-53 MFC - Incoming Call

General Description

This feature allows an MFC incoming call from an MFC trunk to be terminated to a station or an Attendant.

Note: This feature is available in Australia only.

Operating Procedure

No manual operation is required.

Service Conditions

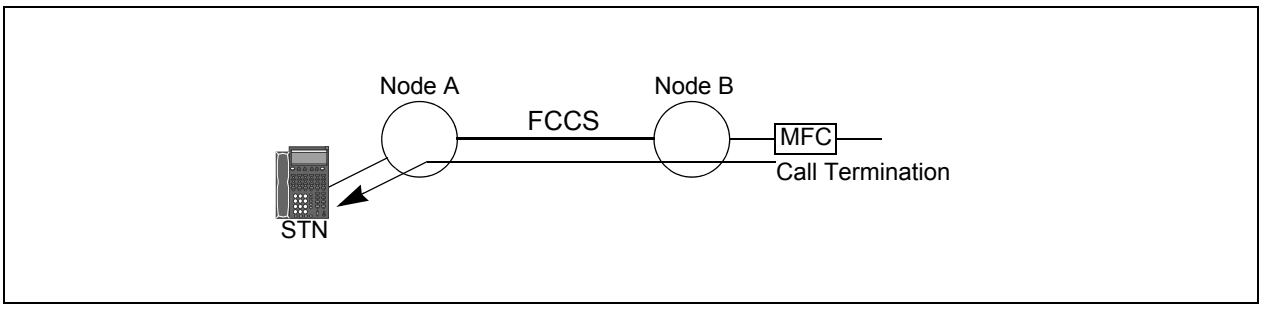
1. DID incoming calls from MFC trunks can terminate to a station (PB, DP and Dterm) or an Attendant.
2. When calling the attendant console from MFC trunks, LISTED DIRECTORY NUMBER (LDN) [L-24] is dialed.
3. CALL WAITING - TERMINATING [C-12] is not available when an incoming call from an MFC trunk is terminated.
4. In case the incoming call from MFC trunk has terminated to a station in the UCD [U-1] group, and that station is busy, UCD Queuing service is available.
5. In case of TRUNK ANSWER FROM ANY STATION (TAS) [T-8] connection, if the called party is not restricted to TAS connection even if it is not registered in TAS, backward signal “called party idle” (Group-B, signal B-1) will be transmitted to the calling party.
6. When a called party is idle, an incoming call is terminated as follows, according to the Group-2 signals:
   a.) If Group-2 signal is 1 (“operator”) or 8 (“overseas operator”) and the route information is TR=0, the call will be connected to the Attendant.
   b.) If Group-2 signal is other than 1 or 8, the call will be connected to the dialed called party. However, calls of LDN or Call Forwarding are terminated to the transfer destination. The call is released notwithstanding Group-B B-6 signal (“called party idle control”), when the calling party has been released.
7. The Group-B signals to be sent out according to the called party's status are as follows:

Called party idle (charge)	Group-B=1
LDN (ATT)	1
Intercept to ATT (Called Party REST)	1
CALL FORWARDING - ALL CALLS to ATT	1
CALL FORWARDING - BUSY LINE to ATT	1
Night station	1
Called party busy	2
Lock out/Make busy	2
Dead number	2

Congestion	2
Restriction	2
Called party idle (no charge)	5
Called party idle (charge)	5
Change calling party release	6
TAS connection	1

8. Following features are available for the incoming call from an MFC trunk:
    - CALL FORWARDING - ALL CALLS [C-5]
    - CALL FORWARDING - BUSY LINE [C-2]
    - CALL FORWARDING - DON'T ANSWER [C-3]
    - CALL FORWARDING - INTERCEPT/ANNOUNCEMENT [C-25]
    - (to a station/Attendant)
    - CALL PICKUP [C-7, 30]
    - CALL QUEUING [C-9]
    - DISTINCTIVE RINGING [D-10]
    - IMMEDIATE RINGING [I-1]
    - INCOMING CALL IDENTIFICATION [I-2]
    - LDN - NIGIT CONNECTION [L-28]
    - NIGHT CONNECTION [N-1, 2]
    - PEG COUNT [P-7]
    - REMOTE ACCESS TO SYSTEM [R-2]
    - STATION HUNTING - CIRCULAR [S-7]
    - STATION HUNTING - SECRETARIAL [S-8]
    - STATION HUNTING - TERMINAL [S-9]
    - TENANT SERVICE [T-12]
    - TRUNK ANSWER FROM ANY STATION (TAS) [T-8]
    - UNIFORM CALL DISTRIBUTION (UCD) [U-1]
    - UNIFORM NUMBERING PLAN [U-4]
    - Multi numbering

FCCS Conditions



1. Both MFC Receiver and MFC Sender must be accommodated in each node.

2. When a call is terminated from an MFC trunk in Node B to a station in Node A, MFC Receiver accommodated in Node B is used.
3. When an MFC incoming call is terminated from an MFC trunk in Node B to a station in Node A, if the connection trunk cannot be used, the backward signal "Congestion" is sent.
4. Be sure to assign the same data of MFC signaling system for each node.
5. Even if the Tenant service is provided between nodes in the FCCS network, an MFC DID call to an Attendant Console cannot be routed to the Attendant Console in the different node from the terminating trunk.
6. The ANI information is comprised as follows.
   a.) When MFC DOD call is originated from an Attendant Console:
   ANI = Office Code + LDN (Listed Directory Number)
   b.) When a call from an ACIS line is routed to an MFC trunk via tandem connection:
   ANI = Office Code of Tandem Office + LDN of Tandem Office
   c.) When a call from an CCIS line is routed to an MFC trunk via tandem connection:
   ANI = Office Code of Originating Office + Telephone Number/Individual ATT Number of Originating Office
   d.) When the system data to send LDN is assigned:
   ANI = Office Code + LDN
   e.) When the system data to send Physical Station Number is assigned:

   - When the assigned telephone number is 7 digits or less:
   ANI = Office Code + Telephone Number

   - When the assigned telephone number is 8 digits or more/not assigned:
   ANI = Office Code + Physical Station Number

Australia (Telecom) Specifications

Signal	Forward signal		Backward signal	
	G-1	G-2	G-A	G-B
1	Digit "1"	operator	send next digit (n + 1)	called party idle
2	Digit "2"	subscriber or coin telephone	send first digit	called party busy
3	Digit "3"	test position	address complete change-over to reception of G-B	no line out before answer
4	Digit "4"	interception center or remote operation center	start DP 1st digit	congestion
5	Digit "5"	coin telephone	start DP 2nd digit	called party idle (no charge)
6	Digit "6"	data telephone	start DP 3rd digit	called party idle control NO time out before answer
7	Digit "7"	overseas subscriber	start DP 4th digit	called party idle, called party control no charge

Australia (Telecom) Specifications

Signal	Forward signal		Backward signal	
	G-1	G-2	G-A	G-B
8	Digit “8”	overseas operators	start DP 5th digit	called party interception
9	Digit “9”	subscriber	send calling party access info	send calling party access info
10	Digit “10”	used by public subscriber	send previous digit MFC	
11 15				

abcd Signals received

abcd Signals Telephony Server transmits and receives.

~~abcd~~ Not handled

Programming

STEP 1: **ASYD -** System Data 1, Index 125, Bits 0 through 3 (Designation of PAD value)
This PAD value is used for adjusting transmission level of MFCS (=MFC Sender).

Input Data	PAD Value (dB)	Input Value	PAD Value (dB)	Remarks
0	0	4	-8	
1	-2	5	-10	
2	-4	6	-11.5	
3	-6	7	-14	

System Data 1, Index 125, Bits 4 through 7 (Type of MFC): Assign 1 (Australia (D1301G)).
System Data 3, Index 10, Bits 0 through 3 (Reception sensitivity level of the MFC Register)
System Data 3, Index 10,Bits7 (ACT): 0/1 = -38dbm (fixed)/depends on input data (Index 10, Bits 0 through 3)

T	Sensitivity Level (dbm)	T	Sensitivity Level (dbm)	Remarks
0	-28	8	-36	
1	-29	9	-37	
2	-30	A	-38	
3	-31	B	-39	
4	-32	C	-40	
5	-33	D	-41	
6	-34	E	-42	
7	-35	F	-43	

STEP 2: **ARTD -** Assign Route Data for MFC lines paying attention to the following CDNs.
<CDN2> ONSG (Outgoing Select Signal): 10 (=MFC)
<CDN4> INSG (Incoming Select Signal): 10 (=MFC)
<CDN23> TR (GII Signal Override (MFC)): 0/1 = -/ICPT by GII signal is not available. **Note 1**

Note: To change the data for existing route (RT), initialize or reboot the trunk device such as circuit card, MG, or VS32.

Note 1: When TR is assigned as "1". the system does not execute ICPT transfer in accordance with Group-II signals/station status.

<CDN 24> OC (Originating Category Request): 0/1 = Out/In Service.
Assign 1 (When reception of Address Signals complete, the System is placed into the next status to receive Group-II signals).

STEP 3: **ASFC -** Assign Service Feature Restriction Class of calling parties.
SFI59 (No charge for IC Call (MFC)): 0/1: = -/No charge
SFI60 (Called party release for IC Call (MFC)): 0/1 = -/Called Party Release **Note 2**

Note 2: This data is used when Group-B signals are edited. When data is assigned in both SFI159 and SFI160, SFI159 has preference.

STEP 4: **ATRK -** Assign MFC Outgoing/Incoming trunks and MFC Receiver (=MFCR) and MFC Sender (MFCS).
MFCR: RT=916
MFCS: RT=917
The 4MFC circuit card (=PA-4MFCA), which can be used as both MFCR and MFCS, must be assigned in accordance with one of the following types.

- TYPE 1 - All four circuits are assigned as MFCR. (Assign MFCS on the other 4MFCA card.)
- TYPE 2 - All four circuits are assigned as MFCS. (Assign MFCS on the other 4MFCA card.)
- TYPE 3 - The first two circuits are assigned as MFCR. The remaining two circuits are assigned as MFCS.
- TYPE 4 - The first two circuits are assigned as MFCR. The remaining two circuits are assigned as MFCS.

M-59 MFC - Incoming Call To Tie Line

General Description

This service feature permits an incoming call from an MFC trunk to be terminated to a station on a distant Office which is connected by a TIE line.

Note: This feature is available in Australia only.

Operating Procedure

No manual operation is required.

Service Conditions

1. The numbering plan for outgoing calls from non-MFC trunks uses OGC, OGCA, LCR and LCRS system.
2. If a called number is the uniform number, the call will be connected to the station.
3. If the terminating station has set following service, tandem connection is activated.
   - CALL FORWARDING - ALL CALLS - OUTSIDE [C-28]
   - CALL FORWARDING - BUSY LINE - OUTSIDE [C-60]
4. If the terminating station has set the following service, tandem connection is not activated.
   - CALL FORWARDING - DON'T ANSWER - OUTSIDE [C-74]
5. If the call origination is restricted by TOLL DIVERSION [T-6], the call will be connected to the attendant console.
6. In case of restrictions other than item 4 above, backward signal “called party busy” (Group-B, signal B-2) will be transmitted to the terminating MFC trunk.
7. When the tandem connection is available, backward signal “called party idle” (Group-B, signal B-1) will be transmitted to the terminating MFC trunk.
8. OUTGOING TRUNK QUEUING [O-2] is not available for MFC-to-non-MFC connection.
9. The Group-B signals to be sent out according to the called party's status are as follows:

Intercept to Attendant (called party REST)	Group B = 1
Dead number	2
Congestion	2
Restriction	2
Tandem Connection	1
CALL FORWARDING - ALL CALLS - OUTSIDE	1

10. Following service features interact with MFC tandem connection.
    - CALL FORWARDING - ALL CALLS - OUTSIDE [C-28]
    - (MFC-to-MFC transfer not available)
    - CALL FORWARDING - BUSY LINE - OUTSIDE [C-60]
    - (MFC-to-MFC transfer not available)
    - CALL FORWARDING - DON'T ANSWER - OUTSIDE [C-74]
    - (MFC-to-MFC transfer not available)
    - CALL FORWARDING - INTERCEPT/ANNOUNCEMENT [C-25]

- (to station/Attendant)
- LDN NIGHT CONNECTION - OUTSIDE [L-30]
- PEG COUNT[P-7]
- TENANT SERVICE[T-12]
- TRUNK-TO-TRUNK CONNECTION [T-10]
- Multi numbering
- Restricted tandem connection

Australia (Telecom) Specifications

Signal	Forward signal		Backward signal	
	G-1	G-2	G-A	G-B
1	Digit “1”	operator	send next digit (n + 1)	called party idle
2	Digit “2”	subscriber or coin telephone	send first digit	called party busy
3	Digit “3”	test position	address complete change-over to reception of G-B	no line out before answer
4	Digit “4”	interception center or remote operation center	start DP 1st digit	congestion
5	Digit “5”	coin telephone	start DP 2nd digit	called party idle (no charge)
6	Digit “6”	data telephone	start DP 3rd digit	called party idle control NO time out before answer
7	Digit “7”	overseas subscriber	start DP 4th digit	called party idle, called party control no charge
8	Digit “8”	overseas operators	start DP 5th digit	called party interception
9	Digit “9”	subscriber	send calling party access info	send calling party access info
10	Digit “10”	used by public subscriber	send previous digit MFC	
11 15				

abcd Signals received

abcd Signals received and transmitted by Telephony Server.

~~abcd~~ Not handled

Programming

STEP 1: **ASYD -** System Data 1, Index 125, Bits 0 through 3 (Designation of PAD value): This PAD value is used for adjusting transmission level of MFCS (= MFC Sender).

Input Data	PAD Value (dB)	Input Data	PAD Value (dB)	Remarks
0	0	4	-8	
1	-2	5	-10	
2	-4	6	-11.5	
3	-6	7	-14	

System Data 1, Index 125, Bits 4 through 7 (Type of MFC): Assign 1 (Australia (D1301G)).

System Data 3, Index 10, Bits 0 through 3 (Designation of Sensitivity Level) (=T) of MFC Receiver)

System Data 3, Index 10, Bits 7 (ACT) 0/1 = -38dbm (fixed)/depends on input data (Index 10, Bits 0 through 3)

STEP 2: **ARTD -** Assign Route Data for MFC lines paying attention to the following CDNs.

<CDN2> ONSG (Outgoing Select Signal): 10 (=MFC)

<CDN4> INSG (Incoming Select Signal): 10 (=MFC)

<CDN23> TR (GII Signal Override (MFC)): 0/1 = -/ICPT by GII signal is not available. **Note 1**

Note: To change the data for existing route (RT), initialize or reboot the trunk device such as circuit card, MG, or VS32.

Note 1: When TR is assigned as “1,” the system does not execute ICPT transfer in accordance with Group-II signals/station status.

<CDN 24> OC (Originating Category Request): 0/1 = Out/In Service.

Assign 1 (When reception of Address Signals complete, the System is placed into the next status to receive Group-II signals).

STEP 3: **ASFC -** Assign Service Feature Restriction Class of calling parties.

SFI59 (No charge for IC Call (MFC)): 0/1: = -/No charge

SFI60 (Called party release for IC Call (MFC)): 0/1 = -/Called Party Release **Note 2**

Note 2: This data is used when Group-B signals are edited. When data is assigned in both SFI159 and SFI160, SFI159 has preference.

STEP 4: **ATRK -** Assign MFC Outgoing/Incoming trunks and MFC Receiver (=MFCR) and MFC Sender (MFCS).

MFCR: RT=916

MFCS: RT=917

The 4MFC circuit card (=PA-4MFCA), which can be used as both MFCR and MFCS, must be assigned in accordance with one of the following types.

- TYPE 1 - All four circuits are assigned as MFCR. (Assign MFCS on the other 4MFCA card.)
- TYPE 2 - All four circuits are assigned as MFCS. (Assign MFCS on the other 4MFCA card.)
- TYPE 3 - The first two circuits are assigned as MFCR. The remaining two circuits are assigned as MFCS.
- TYPE 4 - The first two circuits are assigned as MFCR. The remaining two circuits are assigned as MFCS.

M-84 MF Signaling - DID

General Description

In the Associated Channel Interoffice Signaling system, this feature allows use of selective codes of Multi frequency (MF) signals.

MF - (ANSI) codes: A total of 15 codes are available by combining two out of six frequencies.

Code	CCITT/ BELL	0 (700Hz)	1 (900Hz)	2 (1100Hz)	4 (1300Hz)	7 (1500Hz)	K (1700Hz)
1	1	X	X				
2	2	X		X			
3	3		X	X			
4	4	X			X		
5	5		X		X		
6	6			X	X		
7	7	X				X	
8	8		X			X	
9	9			X		X	
10	0				X	X	
11	C_{11}	X					X
12	C_{12}		X				X
13	KP_1			X			X
14	KP_2				X		X
15	ST					X	X

Note: This feature is available in North America only.

Operating Procedure

No manual operation is required.

Service Conditions

1. When using MF signaling system, Least Cost Routing (LCR) or Register Sender LCR (LCRS) must be assigned.
2. Even when LCRS is assigned, digit outpulsing to the distant office is not started until all digits have been received when the signaling is MF. If there is DP or PB lines among alternate routes, an Outgoing Call may be originated by seizing a DP or PB line, even when the route advance sequence for that line is lower than MF lines. When the route advance sequence of MF to DP or PB is predetermined, LCR must be assigned in the Numbering Plan.
3. Unless otherwise specified, MP starting at termination of an Incoming Call will be activated upon receiving an ST signal.
4. Connection patterns to which MF signaling system can be applied are as follows:
   a.) Individual Call Origination by Station
   b.) Attendant-to-Attendant Connection
   c.) Individual Call Origination by Attendant
   d.) Call origination by Attendant via Station/Trunk
   e.) MF line Connection from CONSULTATION HOLD [C-17]
   f.) Call transfer to MF line via Call Forwarding.
5. Connection to the following services is not available:
   a.) Call origination services to MF lines, such as, OUTGOING TRUNK QUEUING
   b.) Call termination services from MF lines, such as, UCD [U-1]
   c.) Call Waiting, REMOTE ACCESS TO SYSTEM [R-2].
6. Assignment of MF signaling system or DP, PB signaling system can be determined on a per route basis.
7. 8RST card is required for performing this service.

Programming

STEP 1: **ASYD** - System 1, Index 234, Bit 0.
(0) No reverse-type Connection acknowledgment

STEP 2: **ANPD** - Assign the first digit of the MF line Access Code.

STEP 3: **ASPA** - Assign the LCR or LCR (S) Access Code by providing a dummy route.

STEP 4: **AMND** - Assign the maximum number of necessary digits to the Dial (Area/Office) Code.

STEP 5: **ARNP** - Assign the Access Code to the Route Number. When multiple Access Codes are present by alternative routing, set one of them.

STEP 6: **ARTD** - MF line- Assign the following data, depending on the Trunk type, as shown below:

Trunk Type	OSGS	ONSG	ISGS	INSG	LSG	Display
OGT	7	4	0	0	5	1
ICT	0	0	7	4	5	1
BWT	7	4	7	4	5	1

Dummy route-Assign 1 at all times for CDN 13:AC. The Route Data of the dummy route is TCL = 1, AC = 1 and L/T = 1.

For the others, assign 0 in all cases.

Assign the following Route Class Data to the route of the MF Line.
TCMN = 0, TCMC = 0, MFSP: MF Sending Speed.
KPST: Duration of KP sending,
KPPT: Pause after KP sending,
CNI: CNI Calling Number Identification format (= 1)
Assign CDN 22:CNI = 1.

Note: To change the data for existing route (RT), initialize or reboot the trunk device such as circuit card, MG, or VS32.

STEP 7: **ATRK** - MF line-Assign the LENS data of each Trunk.
MF Register/Sender-Assign the following data to the LENS of the 8RST card.
RT = 904: MF Register
RT = 905: Sender

STEP 8: **AFRS** - Assign the number Pattern Code (NPC) and Outgoing route selection pattern (OPR) to the dummy route.

STEP 9: **AOPR** - Assign the Route Number, including the Alternative Route to the outgoing route selection pattern (OPR). Assigns the number of skipped digits and pattern number location (PNL) to each route.

STEP 10: **ARSC** - Release the dummy route and Station Connection restriction.
Release the Connection restriction on the route of the MF line and Station.

FCCS Programming

This data setting describes only the additional or replacement commands used to activate FCCS Services. When used as an FCCS feature, these commands replace the non-network commands (for example, AAED is replaced by AAEDN). Note that the data setting for FCCS Service varies depending on the Node.

Note: When using ASYDL or ASYDN to program an FCCS feature, the basic programming of ASYD is required in addition to ASYDL or ASYDN.

STEP 1: **ANPDL** - Assign the first digit of the MF line Access Code.

STEP 2: **ASPAL** - Assign the LCR or LCR (S) Access Code by providing a dummy route.

STEP 3: **ARNPL** - Assign the Access Code to the Route Number. When multiple Access Codes are present by alternative routing, set one of them.

STEP 4: **ARTDN** - MF line- Assign the following data, depending on the Trunk type, as shown below:

Trunk Type	OSGS	ONSG	ISGS	INSG	LSG	Display
OGT	7	4	0	0	5	1
ICT	0	0	7	4	5	1
BWT	7	4	7	4	5	1

Dummy route-Assign 1 at all times for CDN 13:AC. The Route Data of the dummy route is TCL = 1, AC = 1 and L/T = 1.

For the others, assign 0 in all cases.

Assign the following Route Class Data to the route of the MF Line.

TCMN = 0, TCMC = 0, MFSP: MF Sending Speed.

KPST: Duration of KP sending,

KPPT: Pause after KP sending,

CNI: CNI Calling Number Identification format (= 1)

Assign CDN 22:CNI = 1.

STEP 5: **AFRSL** - Assign the number Pattern Code (NPC) and Outgoing route selection pattern (OPR) to the dummy route.

STEP 6: **AOPRL** - Assign the Route Number, including the Alternative Route to the outgoing route selection pattern (OPR). Assigns the number of skipped digits and pattern number location (PNL) to each route.

STEP 7: **ARSCN** - Release the dummy route and Station Connection restriction.

Release the Connection restriction on the route of the MF line and Station.

M-85D Message Waiting Indication -D

General Description

This feature allows Message Waiting (MW) information to be displayed on the middle line (24 digits) of the LCD.

Operating Procedure

No manual operation is required.

Service Conditions

1. The LCD displays the following information, when MESSAGE WAITING INDICATION -D is activated:

Initial Display (Time Display)	**10:00 AM FRI 01 JAN 1999**
When controlled by MCI, Attendant or Station	**MW**

2. When this service is assigned to a Dterm with a One-line LCD, the service is not available, and the display remains unchanged.
3. This service can be controlled via PCPro command program.
4. This service is available only when OAI service is not available (System Data 1, Index 79, Bit 6 = 1). When OAI service is available (System Data 1, Index 79, Bit 6 = 0), the information cannot be displayed on the lower line of LCD, but will be displayed on the upper line.
5. While Message Waiting information control is being executed to a Dterm equipped with a Two-line LCD by this service, time information is not displayed on the LCD. When Message Waiting information control is canceled, the time information display is resumed.
6. For the FCCS Network, this feature is available for a Telephone Number having less than 7 digits.

Programming

STEP 1: **ASYD** - System Data 1, Index 79, Bit 6 = 1.

STEP 2: **AKYD** - Assign the following data to the Dterm that executes this service.
(MWD = 1: Message Waiting Indication in Service)

M-92 MFC - Tandem Switching

General Description

This service feature allows trunk-to-trunk connections relating to MFC signaling through the System, without the need for any Attendant assistance or control.

The following connection patterns are provided with this service feature:

- MFC Trunk to Non-MFC Trunk Connections
- Non-MFC Trunk to MFC Trunk Connections
- MFC Trunk to MFC Trunk Connections
- MFC Trunk to PHC Connections
- PHC to MFC Trunk Connections

Note: This feature is available for the countries indicated below.

Operating Procedure

No manual operation is required.

Service Conditions

MFC Trunk to Non-MFC Trunk Connections

1. This type of connection is supported by the following types of MFC signaling system:
   a.) CCITT-R2 (NEC Standard) Type
   b.) Australia Type
   c.) Thailand Type
   d.) Venezuela Type
   e.) Korea Type
   f.) Singapore/Malaysia Type
   g.) Argentina/Philippines Type
   h.) Mexico Type
   i.) Brazil Type (digital/analog)
   j.) Chile Type
   k.) Columbia Type
   l.) India Type

2. For CCITT-R2, Australia, Thailand, Korea, and Singapore Type
   Outgoing calls to non-MFC trunks use the LCR or LCRS system numbering plan.

   For Argentine, Brazil, Chile, Columbia, Mexico, Venezuela, and India Type
   Outgoing calls to non-MFC trunks use the LCR system numbering plan.
3. If the called number is the uniform number, the call will be connected to the corresponding station.
4. If the terminating station has set the following services, tandem connection is activated.
   CALL FORWARDING-ALL CALLS-OUTSIDE [C-28]
   CALL FORWARDING-BUSY LINE-OUTSIDE [C-60]
5. If the terminating station has set the following service, tandem connection is not activated.
   CALL FORWARDING-DON'T ANSWER-OUTSIDE [C-74]

6. If the call origination is restricted by TOLL DIVERSION [T-6], the call will be connected to the Attendant Console.
7. In case of restrictions other than item 5 above, Group-B signal relating to restrictions is transmitted to the terminating MFC trunk.
8. When the tandem connection is available, Group-B signal relating to Tandem Connection is transmitted to the terminating MFC trunk.
9. INTER-OFFICE OFF-HOOK QUEUING [I-11] is not available for MFC to non-MFC trunk connection.
10. The Group-B signals to be sent out according to the called party's status are as follows:

MFC SIGNAL TYPE / STATUS	CCITT-R2	AUSTRAL-IA	THAILAND	VENEZU-ELA	KOREA	SINGA-PORE	ARGEN-TINA/ Philippines	CHILE	COLUMBIA	MEXICO	INDIA
Dead Number	5	2	4	5	5	7	5	5	5	4	5
Congestion	4	2	4	4	4	4	4	4	4	4	4
Restriction	8	2	4	8	8	4	8	8	8	4	4
Intercept to Attendant (called party restriction)	6	1	1	6	6	1	6	6	6	1	6
CALL FORWARDING-ALL CALLS-OUTSIDE [C-28]	6	1	1	6	6	1	6	6	6	1	6
Tandem Connection	6	1	1	6	6	1	6	6	6	1	6

11. Following service features interact with MFC -to non-MFC trunk connection:
    CALL FORWARDING-ALL CALLS-OUTSIDE [C-28]
    CALL FORWARDING-BUSY LINE-OUTSIDE [C-60]
    CALL FORWARDING-DON'T ANSWER-OUTSIDE [C-74]
    CALL FORWARDING-INTERCEPT/ANNOUNCNEMENT [C-25]
    (to a station/Attendant Console)
    LDN-NIGHT CONNECTION-OUTSIDE [L-30]
    Multi Numbering
    PEG COUNT [P-7]
    Restricted Tandem Connection
    STATION MESSAGE DETAIL RECORDING [S-10]
    TENANT SERVICE [T-12]
    TOLL DENIAL/TOLL DIVERSION [T-6]
    TOLL RESTRICTION-3/6-DIGIT [T-7]
    TRUNK-TO-TRUNK CONNECTION [T-10]
    UNIFORM NUMBERING PLAN [U-4]
12. <u>For CCITT, Thailand, Venezuela, Korea, Argentina, Brazil, Chile, Columbia, Mexico Type</u>
    The ANI information sent to the MFC trunks is as follows:
    a.) when the calling party is an analog line: the number assigned at the tandem office.
    b.) when the calling party is a station/Attendant Console in the CCIS network: the RT number and the telephone number/individual Attendant Console number assigned at the calling office.
    c.) when the calling party is a trunk in the CCIS network: the number assigned at the tandem office.
13. The additional hardware (MFC Register Sender) is required for this feature.

14. In the following cases, the ANI (Automatic Number Identification) information received from the opposite office is sent to the OAI computer via Interface Processor by SMFN facility. The information is output when an MFC incoming call is terminated to the OAI terminal (Dterm), and when the terminal user answers the call. When the ANI information is not requested, space (blank) is output to the ANI information area in the SMFN facility format.
    a.) When a call is terminated from an MFC public exchange via CCIS line, the calling number (max. 16 digits) is output as the ANI information.
    b.) When a call terminated from an MFC public exchange is transferred to the OAI terminal in another office via CCIS line, the forwarding party number (max. 16 digits) is output.
15. When an MFC DID incoming call is terminated, the system can request the ANI (Automatic Number Identification) information. The ANI information consists of category (1 digit) and the calling number (max. 31 digits). The received ANI number is output to the SMDR message.
    c.) SMDR messages
    -When an MFC incoming call is terminated via CCIS line, the ANI information is output to SMDR of the tandem office as the billing information for the incoming call and for tandem connection.
    The ANI information from the MFC line is output to SMDR of the terminating office as billing information for the incoming call. However, only the calling number (max. 16 digits) is displayed; space is output for category area.
    -When an MFC incoming call is transferred to another office via CCIS line, the ANI information from the MFC line is output to SMDR of the tandem office, and the ANI information of the transferring station is output to SMDR of the terminating office. However, only the calling number (max. 16 digits) is displayed; space is output for category area.
    -When the ANI information is not received, no data is output to the ANI information area.
    d.) NAME DISPLAY-SYSTEM-CCIS [N-37] is not available.

Non-MFC Trunk to MFC Trunk Connections

1. This type of connection is supported by the following types of MFC signaling system:
    a.) CCITT-R2 (NEC Standard) Type
    b.) Venezuela Type
    c.) Korea Type
    d.) Thailand Type
    e.) Argentina/Philippines Type
    f.) Brazil Type (digital/analog)
    g.) Chile Type
    h.) Columbia Type
    i.) Mexico Type
    j.) India Type
2. Outgoing calls to MFC trunks use the LCR or LCRS system numbering plan.
3. If the terminating station has set following service, tandem connection is activated.
    CALL FORWARDING-ALL CALLS-OUTSIDE [C-28]
    CALL FORWARDING-BUSY LINE-OUTSIDE [C-60]
4. If the terminating station has set following service, tandem connection is not activated.
    CALL FORWARDING-DON'T ANSWER-OUTSIDE [C-74]
5. <u>For Mexico Type only</u>
    The Group-II signal transmitted to MFC trunk is fixed to be "2".

For Other Types

The Group-II signal transmitted to MFC trunk may be determined on an incoming route basis.

Group-II 1: Subscriber without Priority
2: Subscriber with Priority

When the calling party is a station user or an Attendant in a CCIS network, the Group-II signal sent to MFC trunk is determined according to the Service Feature Restriction Class of the originating station/Attendant.

6. Even if a Group-B=Subscriber's Line Busy is received, the calling party hears Busy Tone.
7. Following service features interact with Non-MFC-to MFC trunk connection:
   CALL FORWARDING-ALL CALLS-OUTSIDE [C-28]
   CALL FORWARDING-BUSY LINE-OUTSIDE [C-60]
   INTER-OFFICE OFF-HOOK QUEUING [I-11]
   LDN-NIGHT CONNECTION-OUTSIDE [L-30]
   PEG COUNT [P-7]
   Restricted Tandem Connection
   STATION MESSAGE DETAIL RECORDING [S-10]
   TOLL DENIAL/TOLL DIVERSION [T-6]
   TOLL RESTRICTION-3/6-DIGIT [T-7]
   TRUNK-TO-TRUNK CONNECTION [T-10]
8. For India Type Only
   The ANI (Automatic Number Information) information sent to the MFC trunks is as follows:
   a.) When the calling party is a station/Attendant Console in the CCIS network:
   The office code of the originating office and the telephone number/individual Attendant Console number.
   b.) When the calling party is a station/Attendant Console in the tie line network: the office code of the tandem office and the Listed Directory number.

MFC Trunk to MFC Trunk Connections

1. This type of connection is supported by the following types of MFC signaling system:
   a.) CCITT-R2 (NEC Standard) Type
   b.) Venezuela Type
   c.) Korea Type
   d.) Thailand Type
   e.) Argentina/Philippines Type
   f.) Chile Type
   g.) Columbia Type
   h.) Mexico Type
   i.) India Type
2. Outgoing calls to MFC trunks use the LCR or LCRS system numbering plan.
3. If the terminating station has set following service, tandem connection is activated.
   CALL FORWARDING-DON'T ANSWER-OUTSIDE [C-74]
4. Abbreviated dialing from incoming MFC trunks to outgoing MFC trunks is not available.
5. When outgoing trunk routes composed of MFC trunk routes and non-MFC trunk routes configure ALTERNATE ROUTING [A-16] group, the MFC trunk routes have priority over the non-MFC trunk routes in tandem call routing.
6. Following service features interact with MFC-to-MFC trunk connection:

PEG COUNT [P-7]
Restricted Tandem Connection

7. INTER-OFFICE OFF-HOOK QUEUING [I-11] is not available for MFC-to-MFC trunk connection.
8. In the following cases, the ANI (Automatic Number Identification) information received from the opposite office is sent to the OAI computer via Interface Processor by SMFN facility. The information is output when an MFC incoming call is terminated to the OAI terminal (Dterm), and when the terminal user answers the call. When the ANI information is not requested, space (blank) is output to the ANI information area in the SMFN facility format.
   a.) When a call is terminated from an MFC public exchange via MFC tie line, the calling number (max. 31 digits) is output as the ANI information.
   b.) When a call terminated from an MFC public exchange is transferred to the OAI terminal in another office via MFC tie line, one of the following is output depending upon the data assignment in the tandem office.
   - telephone number of the forwarding party
   - office code and telephone number
   - pilot number
   - space when no data is assigned.
9. When an MFC DID incoming call is terminated, the system can request the ANI (Automatic Number Identification) information. The ANI information consists of category (1 digit) and the calling number (max. 31 digits). The received ANI number is output to the SMDR message.
   a.) SMDR message
   -Both category (1 digit) and the calling number (max. 31 digits) are output.
   -When an MFC incoming call is terminated via MFC tie line, the ANI information is output to SMDR of the tandem office as the billing information for the incoming call and for tandem connection. The ANI information from the MFC line is output to SMDR of the terminating office as billing information for the incoming call.
   -When an MFC Incoming call is transferred to another office via MFC tie line, the ANI information from the MFC line is output to SMDR of the tandem office, and the ANI information of the transferring station is output to SMDR of the terminating office.
   -When the ANI information is not received, no data is output to the ANI information area.
   b.) NAME DISPLAY-SYSTEM-CCIS [N-37] is not available.
10. <u>For India Type only</u>
    Following services are not available in MFC-to-MFC connection:
    CALL FORWARDING - ALL CALLS - OUTSIDE [C-28]
    CALL FORWARDING - BUSY LINE - OUTSIDE [C-60]
11. "*" and "#" are available for the Group I (G-I) signal. Both "*" and "#" cannot be used for the "Timing Start", if those digits are used as G-1 signal.

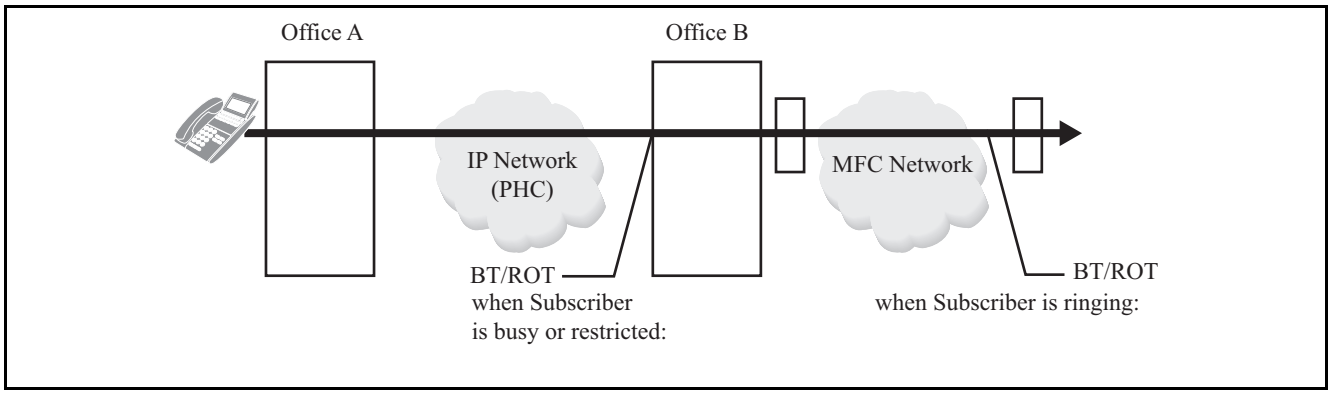
MFC Trunk/PHC to PHC/MFC Trunk Connections

1. The following call forwarding features can be activated:
   - Call Forwarding - All Calls [C-5]
   - Call Forwarding - All Calls - Outside [C-28]
   - Call Forwarding - Busy Line [C-2]
   - Call Forwarding - Busy Line - Outside [C-60]
   - Logged Out IP Station - Call Destination [L-53]
   - Night Transfer
   - Night Connection - Flexible [N-2]
2. (**Tandem connection from MFC to PHC**)

a.) Basically, assign OC = 1 (ARTD) for MFC line. For some specifications, OC = 0 can be assigned.
b.) When OGT/CCS is seized by Tandem connection, the backward signal is returned to the originating office from the terminating office. As in the case of Tandem connection to CCIS, ACM/UBM signal received after the call origination to PHC is not reflected in the backward signal.
c.) The subscriber may hear Reorder Tone (ROT) after receiving Ringback Tone (RBT).
d.) When all routes are busy, queuing is not available. At this time, B4 (Congestion) signal is sent to the originating office.
e.) ANI information is transmitted to the called party. Category is not transmitted.

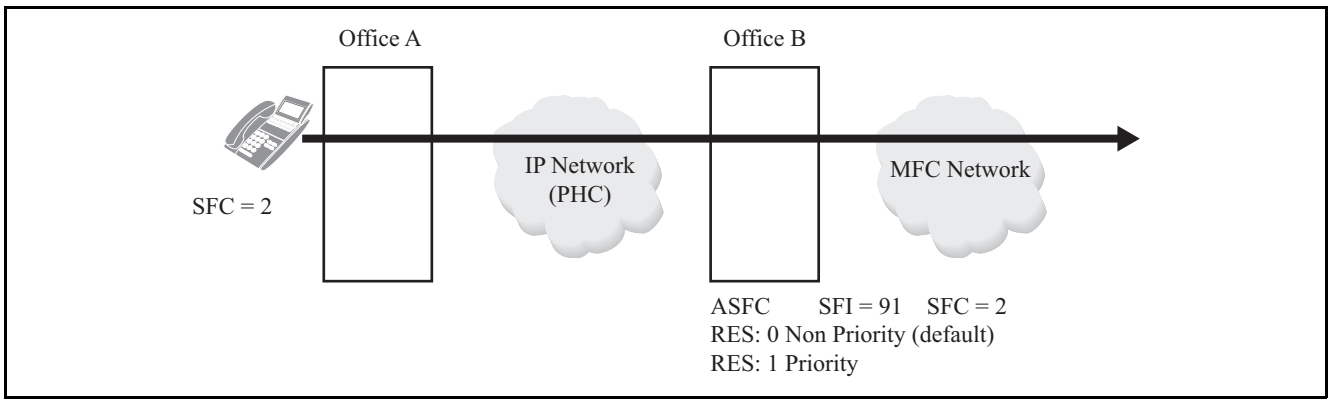
3. (**Tandem Connection from PHC to MFC**)

a.) When OGT/MFCS is seized by Tandem connection, ACM is returned to the originating office.
- Backward signal (Restrict, Busy) received after the call origination to MFC line is not reflected in the originating office.
- ROT (Reorder Tone) and BT (Busy Tone) are sent from Tandem office.

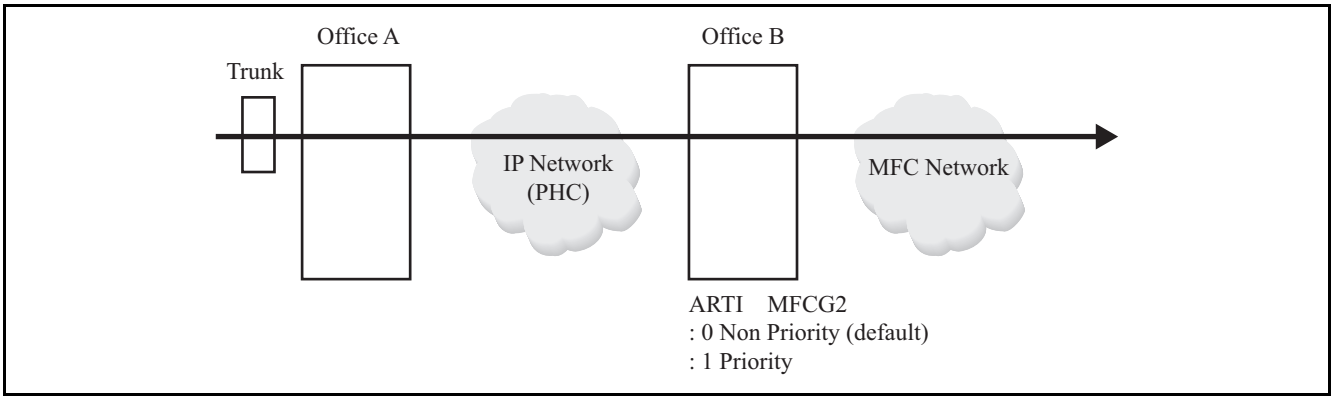


b.) ANI Information
- ANI information received from PHC can be transmitted. However if "#" or "*" is included in the calling number, ANI information cannot be transmitted.
- MFC Category is edited by Tandem office.

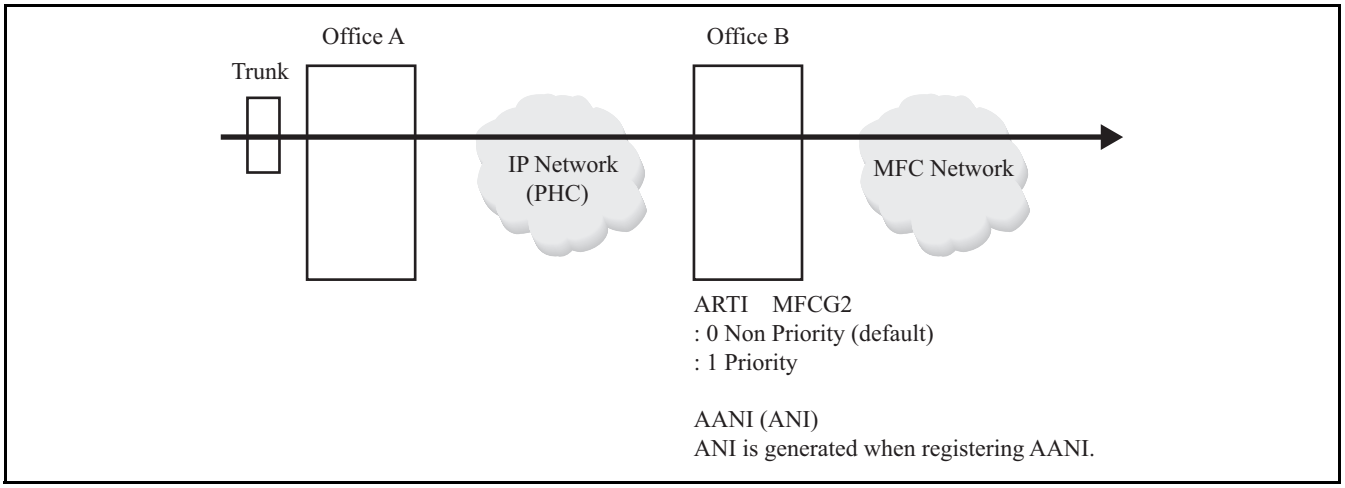
Example 1: When SFC data can be received:
Determines GII signal (Category) by SFC data in Tandem Office.



Example 2: When SFC data cannot be received:
Determines GII signal (Category) by MFCG2 (ARTI) of ICT in Tandem Office.



Example 3: When both SCF data and ANI information cannot be received:
When the calling party is trunk and ANI information cannot be received, Tandem Office determines GII signal (Category) and ANI generation.



4. The following table shows the availability of this feature in each country:

Country		DID (MFC-PHC Tandem)		DOD (PHC-MFC Tandem)
		OC = 0 (ARTD)	OC = 1 (ARTD)	
0	ITU-T	Available	Available	Available
1	Australia	**Note 1**	Available	Not Applicable
2	Thailand	**Note 1**	Available	Available
3	Venezuela	Available	Available	Available
4	Korea	Available	Available	Available
5	Singapore/Malaysia	**Note 1**	Available	Not Applicable
6	China	Not Applicable	Not Applicable	Not Applicable
7	Argentina/Philippines	Available	Available	Available
8	Indonesia	Not Applicable	Not Applicable	Not Applicable
9	Mexico	**Note 1**	Available	Available
10	Brazil	**Note 1**	Available	Not Applicable
11	Chile	Available	Available	Available
12	Colombia	Available	Available	Available
13	India	Available	Available	Available
14	Kuwait	**Note 1**	Available	Available

Note 1: Not Available (This feature doesn't support "A6: Address Complete, Change Setup Speech Conditions" of the backward signal.)

5. This feature is not available for the following specifications:
   - Chain Type
   - Australia Type [Direct Outward Dialing (DOD) to MFC is not available.]
   - Singapore Type [Direct Outward Dialing (DOD) to MFC not available.]
   - Malaysia Type [Direct Outward Dialing (DOD) to MFC not available.]
   - Indonesia Type [Call origination to MFC can be performed by station only]
6. For Australia/Thailand Type only
   <Tandem connection from MFC to PHC>
   GII signal (Subscriber with priority) has precedence over the signal (FW signal: GI) received from MFC when transferring to Attendant Console. To give a higher priority to Tandem connection to PHC, assign TR=1 (ARTD) for the route class of MFC line.
7. For Singapore/ Malaysia Type only
   <Tandem connection from MFC to PHC>
   GII signal (Subscriber with priority) has precedence over the signal (FW signal: GI) received from MFC when transferring to Attendant Console. To give a higher priority to Tandem connection to PHC, assign TR=1 (ARTD) for the route class of MFC line.

   When receiving collect calls of GII signal (International call: GII-9), assign TR=1 (ARTD) for the PHC route of Tandem connection. (To transfer to Attendant Console, assign TR=0)

8. For Brazil Type only
<Tandem connection from MFC to PHC>
GII signal (Subscriber with priority) has precedence over the signal (FW signal: GI) received from MFC when transferring to Attendant Console. To give a higher priority to Tandem connection to PHC, assign TR=1 (ARTD) for the route class of MFC line.

Collect Call Blocking is not available for PHC network.

9. For India Type only
<Tandem connection from PHC to MFC>
A maximum of 6 digits ANI can be output. Thus, the calling number must be within 6 digits.

10. This feature (Tandem connection) is available for a Remote Access to System [R2] call terminated form PHC or MFC.
11. Tandem connection via FCCS network is available.
12. When a call originated from MFC to PHC encounters the IP network failure, alternating routing to another trunk is available.

NEC Standard (CCITT R-2) Type

Signal	Forward signal		Backward signal	
	G-I	G-II Note 2	G-A	G-B
1	Digit "1"	subscriber without priority	send next digit (n+1)	
2	Digit "2"	subscriber with priority	send last but one digit (n-1)	~~send special information tone~~
3	Digit "3"	maintenance equipment	address complete change-over to reception of G-II	subscriber's line busy
4	Digit "4"		congestion in the national network	congestion, encountered after changeover from G-A to G-B
5	Digit "5"	operator	calling party category	unallocated number
6	Digit "6"	data transmission	address complete change set up speech condition	subscriber's line free (charge)
7	Digit "7"	subscriber or operator without forward transfer facility	send last but two digits (n-2)	subscriber's line free (no charge)

abcd Signals transmitted and received

abcd Signals received

~~abcd~~ Signals not handled

NEC Standard (CCITT R-2) Type

Signal	Forward signal		Backward signal	
	G-I	G-II Note 2	G-A	G-B
8	Digit "8"	data transmission	send last but three digits (n-3)	subscriber's line out of order
9	Digit "9"	subscriber with priority		
10	Digit "0"	operator with forward transfer facility		
11	Digit "*"			
12	Digit "#"			
13 15				

abcd Signals transmitted and received

abcd Signals received

~~abcd~~ Signals not handled

Note 2: Only when the category of Attendant is to be transmitted, GII-1 through GII-15 can be used.

Australia (Telecom) Type

Signal	Forward signal		Backward signal	
	G-I	G-II	G-A	G-B
1	Digit "1"	operator	send next digit (n+1)	called party idle
2	Digit "2"	subscriber or coin telephone	send first digit	called party busy
3	Digit "3"	test position	address complete change-over to reception of G-B	~~no line out before answer~~
4	Digit "4"	interception center or remote operation center	~~start DP 1st digit~~	~~congestion~~
5	Digit "5"	coin telephone	~~start DP 2nd digit~~	called party idle (no charge)

abcd Signals transmitted and received

abcd Signals received

~~abcd~~ Signals not handled

Australia (Telecom) Type

Signal	Forward signal		Backward signal	
	G-I	G-II	G-A	G-B
6	Digit “6”	data telephone	~~start DP 3rd digit~~	called party idle control NO time out before answer
7	Digit “7”	overseas subscriber	~~start DP 4th digit~~	~~called party idle, called party control no charge~~
8	Digit “8”	overseas operators	~~start DP 5th digit~~	~~called party interception~~
9	Digit “9”	subscriber	send calling party access info	~~send calling party access info~~
10	Digit “0”	used by public subscriber	~~send previous digit MFC~~	
11	Digit “*”			
12	Digit “#”			
13 15				

abcd Signals transmitted and received

abcd Signals received

~~abcd~~ Signals not handled

Thailand (Telecom) Type

Signal	Forward signal		Backward signal		
	G-I	G-II	G-A	G-B	G-C
1	Digit “1”	toll operator	send next digit (n+1)	called subscriber’s line free (charge)	send next digit of caller
2	Digit “2”	ordinary subscriber	send first digit of called number	called subscriber’s line busy	send first digit of called number
3	Digit “3”	local coin box	send nature of caller and G-B	~~called subscriber’s line received~~	
4	Digit “4”		congestion	congestion	~~congestion~~

abcd Signals transmitted and received

abcd Signals received

~~abcd~~ Signals not handled

Thailand (Telecom) Type

Signal	Forward signal		Backward signal		
	G-I	G-II	G-A	G-B	G-C
5	Digit “5”	standard coin box		called subscriber’s line free (no charge)	send next digit of called number
6	Digit “6”	test equipment	send number of caller and change G-C	~~last party release~~	
7	Digit “7”	line test desk			
8	Digit “8”	intercepted operator			
9	Digit “9”	reserve for data terminal			
10	Digit “0”	immediate charge information service			
11	Digit “*”				
12	Digit “#”				
13 15					

abcd Signals transmitted and received

abcd Signals received

~~abcd~~ Signals not handled

Venezuela Type

Signal	Forward signal		Backward signal	
	G-I	G-II Note 3	G-A	G-B
1	Digit "1"	standard subscriber	send next digit (n+1)	subscriber's line free (change call party release)
2	Digit "2"	preferential subscriber	send last but one digit (n-1)	~~extension number changed~~
3	Digit "3"	test subscriber	address complete, change-over to reception of G-B signal	subscriber's line busy
4	Digit "4"	coin phone	congestion	congestion
5	Digit "5"	attendant	send category of calling subscriber	unallocated number
6	Digit "6"	data transmission	address complete, change setup speech condition	subscriber's line free (with charge)
7	Digit "7"		send last but two digits (n-2)	subscriber's line free (without charge)
8	Digit "8"		send last but three digits (n-3)	subscriber's line out of order
9	Digit "9"		send originating subscriber number	
10	Digit "0"			~~data transmitting equipment~~
11	Digit "*"			
12	Digit "#"			
13				
14				
15	end of pulsing			

abcd Signals transmitted and received

abcd Signals received

~~abcd~~ Signals not handled

Note 3: Only when the category of Attendant is to be transmitted, GII-1 through GII-15 can be used.

Korea (KTA) Type

Signal	Forward signal		Backward signal	
	G-I	G-II Note 4	G-A	G-B
1	Digit "1"	subscriber without priority	send next digit (n+1)	subscriber's line free (called party release)
2	Digit "2"	subscriber with priority	send last but one digit (n-1)	~~subscriber number change~~
3	Digit "3"	line test	address complete change-over to reception of Group	subscriber's line busy
4	Digit "4"	coin box	congestion in the national network	congestion, encountered after changeover from G-A to G-B
5	Digit "5"	operator	send calling party class	unallocated number
6	Digit "6"	data transmission	address complete, change set up speech condition	subscriber's line free (charge)
7	Digit "7"	international call	send last but two digits (n-2)	subscriber's line free (no charge)
8	Digit "8"	international call	send last but three digits (n-3)	subscriber's line out of order or restrict
9	Digit "9"	international call	~~send first digit of called number~~	
10	Digit "0"	international call transfer facility		
11	Digit "*"			
12	Digit "#"			
15	end of originating subscriber			

abcd Signals transmitted and received

abcd Signals received

~~abcd~~ Signals not handled

Note 4: Only when the category of Attendant is to be transmitted, GII-1 through GII-2 can be used.

Singapore (Telecom) Type/Malaysia (Telecom) Type

Signal	Forward signal		Backward signal	
	G-I	G-II	G-A	G-B
1	Digit “1”	operator	send next digit (n+1)	called subscriber’s line free
2	Digit “2”	subscriber		called subscriber’s line busy
3	Digit “3”	local call	address complete change-over to reception of G-B	
4	Digit “4”		congestion	congestion
5	Digit “5”	international call		
6	Digit “6”	test equipment		
7	Digit “7”	line test desk		unallocated number
8	Digit “8”	international operator		
9	Digit “9”	called from transit exchange		
10	Digit “0”			
11	Digit “*”			
12	Digit “#”			
13 15				

abcd Signals transmitted and received

abcd Signals received

Argentina/Philippines Type

Signal	Forward signal		Backward signal	
	G-1	G-II Note 5	G-A	G-B
1	Digit “1”	subscribers without priority	send next digit (n+1)	
2	Digit “2”	subscribers with priority	send last but one digit (n-1)	send recorded messages
3	Digit “3”	maintenance equipment	address complete, change-over to G-B signal	subscriber’s line busy
4	Digit “4”	coin phone	congestion	congestion
5	Digit “5”	attendant	send category of calling subscriber	unallocated number
6	Digit “6”	data transmission	address complete, change setup speech conditions	subscriber’s line free with charge
7	Digit “7”		send last but two digits (n-2)	subscriber’s line free without charge
8	Digit “8”		send last but three digits (n-3)	subscriber’s line out of order
9	Digit “9”		send last digit (n)	
10	Digit “0”		send first digit	
11	Digit “*”			
12	Digit “#”			
13	~~access to test equipment~~			
15	end of pulsing			

abcd Signals transmitted and received

abcd Signals received

~~abcd~~ Signals not handled

Note 5: Only when the category of Attendant is to be transmitted, GII-1 through GII-15 can be used.

Mexico Type

Signal	Forward signal			Backward signal		
	G-I	G-II	G-III	G-A	G-B	G-C
1	Digit “1”		Digit “1”	send next digit (n+1)	subscriber’s with charge	send G-III signal next digit
2	Digit “2”	ordinary subscriber	Digit “2”	send signal first digit	subscriber’s line busy	send G-I signal first digit and charge receipt G-A
3	Digit “3”	coin box telephone	Digit “3”	send signal of G-II, and change to reception of G-B signal		send G-II signal and charge receipt G-B
4	Digit “4”	time and cost	Digit “4”	congestion	line lock out	congestion
5	Digit “5”		Digit “5”		subscriber idle without charge	send G-I signal next digit and change receipt G-A
6	Digit “6”	maintenance equipment	Digit “6”	send group II-6 and change to receipt G-C signal		send G-I signal same digit and change digit receipt G-A
7	Digit “7”	identification of the No. 2 subscriber in party line	Digit “7”			
8	Digit “8”	identification of the No. 3 subscriber in party line	Digit “8”			
9	Digit “9”	identification of the No. 1 subscriber in party line	Digit “9”			
10	Digit “0”		Digit “0”			
11	Digit “*”					
12	Digit “#”					
15			End of digit			

abcd Signals transmitted and received

abcd Signals received

Brazil Type

Signal	Forward signal		Backward signal	
	G-I	G-II Note 6	G-A	G-B
1	Digit “1”	ordinary subscriber	send next digit (n+1)	called party idle with charging
2	Digit “2”	subscriber with special charging		called party busy
3	Digit “3”	maintenance equipment	address complete, change-over to reception of G-II signal	changed number
4	Digit “4”	local public call box	congestion	congestion
5	Digit “5”	operator	send category of calling subscriber	called party idle without charging
6	Digit “6”	data transmission		called party idle with charging; called party based release
7	Digit “7”	national public call box ordinary subscriber (international working)	send last but two digits (n-2)	unallocated number
8	Digit “8”	data transmission (international working)	send last but three digits (n-3)	
9	Digit “9”	subscriber with priority (international working)	send last but one digit (n-1)	
10	Digit “0”	operator with forward transfer facility (international working)		
11	Digit “*”	call transfer		
12	Digit “#”			
15	end of pulsing			

abcd Signals transmitted and received

abcd Signals received

Note 6: Only when the category of Attendant is to be transmitted, GII-1 through GII-15 can be used.

Chile Type

Signal	Forward signal		Backward signal	
	G-I/III	G-II Note 7	G-A	G-B
1	Digit “1”	subscriber without priority	send next digit (n+1)	
2	Digit “2”	subscriber with priority	send last but one digit (n-1)	~~send special information tone~~
3	Digit “3”	maintenance equipment	address complete, change to reception of G-B signal	subscriber’s line busy
4	Digit “4”	subscriber with remote billing	congestion in the national network	congestion, encountered after changeover from G-A signal to G-B signal
5	Digit “5”	operator	send calling subscriber’s category	unassigned number
6	Digit “6”	data transmission	address complete with change	unoccupied subscriber’s line, with billing (charge)
7	Digit “7”		send last but two digits (n-2)	unoccupied subscriber’s line, without billing (without charge)
8	Digit “8”		send last but three digits (n-3)	subscriber’s line out of order
9	Digit “9”			
10	Digit “0”			
11	Digit “*”	unidentifiable subscriber		
12	Digit “#”			
15	end of pulsing			

abcd Signals transmitted and received

abcd Signals received

~~abcd~~ Signals not handled

Note 7: Only when the category of Attendant is to be transmitted, GII-1 through GII-15 can be used.

Columbia Type

Signal	Forward signal		Backward signal	
	G-I	G-II Note 8	G-A	G-B
1	Digit “1”	subscriber	send next digit (n+1)	~~put connection under control of incoming side~~
2	Digit “2”	calling party with priority	send last but one digit (n-1)	~~changed number (called line intercepted)~~
3	Digit “3”	maintenance equipment	send nature of caller and change to G-B signals	called subscriber’s line busy
4	Digit “4”		congestion	congestion
5	Digit “5”	operator	send nature (and identify) of caller	dead line or dead level
6	Digit “6”	data transmission	through connect speech path	called subscriber’s line free with metering (charge)
7	Digit “7”	subscriber for international service	send last digit but two (n-2)	called subscriber’s line free without metering (without charge)
8	Digit “8”		send last digit but three (n-3)	called line out of order
9	Digit “9”		~~nationally at disposal~~	
10	Digit “0”	operator for international service	~~nationally at disposal~~	
11	Digit “*”		~~send first digit of international code~~	
12	Digit “#”		~~send first digit of significant number~~	
13	~~access to maint. equipment for national service~~		~~send first digit of international code of outgoing country~~	
14	~~insert echo suppressor~~		~~inquiry whether or not echo suppressor should be inserted~~	
15	end of pulsing			

abcd Signals transmitted and received

abcd Signals received

~~abcd~~ Signals not handled

Note 8: Only when the category of Attendant is to be transmitted, GII-1 through GII-15 can be used.

India Type

Signal	Forward signal		Backward signal	
	GI	G-II Note 9	G-A	G-B
1	Digit “1”	ordinary subscriber	send next digit (n+1)	
2	Digit “2”	priority subscriber	restart	changed number
3	Digit “3”		change over to reception of G-B signal	called line busy
4	Digit “4”		calling line identification	congestion
5	Digit “5”	operator	send category of the calling subscriber	unallocated number
6	Digit “6”		switch-through the speech path	normal subscriber, free, with metering
7	Digit “7”		send last but two digits (n-2)	called line free, but not metered
8	Digit “8”		send last but three digits (n-3)	
9	Digit “9”		send last but one digit (n-1)	
10	Digit “0”			
11	Digit “*”			
12	Digit “#”			
13				
14				
15				

abcd Signals transmitted and received

abcd Signals received

Note 9: Only when the category of Attendant is to be transmitted, GII-1 through GII-15 can be used.

Programming

When activating this feature via FCCS, use NDM command in NCN.

STEP 1: **ARRC/ARRCN** - Remove the restriction of the connection from MFC/PHC to PHC/MFC and Dummy Route.

STEP 2: **ASFC** - Assign the following Service Feature Restriction Class data if necessary.
SFI = 91 (MFC Outgoing Class of Service)
RES: 0 (Non priority)
RES: 1 (Priority)

Note: When SFC can be received through Tandem connection (from PHC to MFC), the Tandem office determines Category of GII signal by this data. When SFC cannot be received, Category of GII signal is determined by MFCG2 of ARTI/ARTIN.

STEP 3: **ARTI/ARTIN** - Assign the following Trunk Application data regarding to PHC route if necessary.
CDN 43 (MFCG2): 0 (Non priority)/1(Priority)

Note: When SCF cannot be received through Tandem connection, Category of GII signal is determined by this data.

STEP 4: **ARTD/ARTDN** - Assign the following Route Class data if necessary.
CDN: 23
TR: GII Signal Override
0: - / 1: ICPT Transfer by GII Signal is not performed.

Note: Assign this data according to each specification (Australia, Thailand, Singapore, Malaysia, and Brazil.

M-99 Multiple Music-on Hold

General Description

This feature allows the customer leeway to select an appropriate MUSIC ON HOLD [M-7] among a maximum of 64 kinds of music sources. Tenant Number of a Station user who places a call on hold and/or Route Number of a Held Trunk can be used as factors to select the type of MUSIC-ON-HOLD.

Operating Procedure

None.

Service Conditions

1. Maximum of 64 musics can be assigned.
2. The held party is not always connected at the beginning of the music.
3. This service is not available for approximately five minutes after the system initialization and the deletion of data by AHMS command.
4. When assigning Multiple Music-on-Hold Trunk to Group 21-23, change the setting of TSW key to "Disable CFT".
5. The music is assigned to correspond to the following data, in the following priority.
   a.) The tenant number of the station holding the call.
   b.) The route number of the held trunk.
   c.) Combination of the tenant number and the route number.

   The pattern c is not available when either pattern a or b is assigned. Note the following conditions:

   - When the music is assigned to correspond to the tenant number of the station holding the call (pattern a) or the route number of the held trunk (pattern b), the same tenant number or the route number cannot be used to assign the data using pattern c.
   - When the music is assigned using the data pattern c, in order to reassign the data for the same tenant number or the same route number using either pattern a or b, all the previously assigned data must be cleared.
6. When there is no data assigned for neither the route number nor the tenant number, MUSIC ON HOLD [M-7] is not activated.
7. DAT circuit card or BGM equipment can be used as an external music source.
8. VS32 cannot be used as an external music source.
9. When using DAT circuit card as an external music source, refer to Circuit Card Description to select the appropriate key setting.
10. When using DAT circuit card as an external music source, if the music is going to be changed, delete the data in the AHMS command before performing the change. After changing the music source, register the data with the AHMS command.
11. This feature is not available in P2P FCCS network. When using this feature in Legacy FCCS network, setting the data for this feature and for the external music source is required in each node.
12. Do not assign other service data to Group24-31 of unit 2 on each MG.

Programming

When MUSIC ON HOLD [M-7] and MULTIPLE MUSIC-ON HOLD [M-99] have not yet been assigned.
When MUSIC ON HOLD [M-7] data has already been assigned
When MULTIPLE MUSIC-ON HOLD [M-99] has already been assigned.

1. When MUSIC ON HOLD [M-7] and MULTIPLE MUSIC-ON HOLD [M-99] have not yet been assigned.

STEP 1: **ASYD** - System Data 1, Index 251, Bit 7: Multiple Music-on-Hold. Assign 1 (In service).
STEP 2: **ARTD** - Assign Route Data of Trunk (s) which is used for Multiple Music-on-Hold.

Note: To change the data for the existing route (RT), initialize or reboot the trunk device such as circuit card, MG, or VS32.

STEP 3: **ATRK** - Assign the Trunk data by entering LENS data.

Note: When using a **COT** trunk for this feature, change it to a state of MB=1 using the MBTK command and/or flipping the MB switch on the circuit card.

STEP 4: **AHMS** - Assign LENS data of the appropriate Trunk on a HMSC basis, then specify Extension Hold Music Data Kind.

Note: This procedure takes about five minutes before the changes are implemented.

Note: Both settings 1) and 2) are required.

1) Establish a link between the Tenant Number and the HMSC number.
TYPE: Select “Extended Hold Music”.
KIND: Select an Operation Kind from the following:
- Tenant Number of Holding Station
- Route Number of Held Trunk
- Tenant Number of Holding Station and Route Number of Held trunk
TN: Enter a Tenant Number (1-63) to which Holding Station belongs.
RT: Enter an Incoming Route Number (1-255).
HMSC: Specify an Extended Hold Music source (0-63).
Specified Music on Hold is sent to station: Place a checkmark in the checkbox to send the MOH for Station-to-Station call.

2) Establish a link between the HMSC number and the Trunk LENS that is used for the music source.
TYPE: Select “Sound Source LENS”.
HMSC: Specify an Extended Hold Music source (0-63).

Note: Assign the same HMSC number as in the TYPE: Extended Hold Music.

LENS: LENS of the Extended Hold Music source.

2. When MUSIC ON HOLD [M-7] data has already been assigned

Delete the data for MUSIC ON HOLD [M-7] and assign data for this feature.

To delete MUSIC ON HOLD [M-7] data:

STEP 1: **AHMS** -

1) Delete the entry linking the Tenant Number to the HMSC data:
TYPE: "Extended Hold Music".
KIND:
- Tenant Number of Holding Station
- Route Number of Held Trunk
- Tenant Number of Holding Station and Route Number of Held trunk
TN: Tenant Number (1-63) to which Holding Station belongs.
RT: Incoming Route Number (1-255).
HMSC: Extended Hold Music source (0-63).
Specified Music on Hold is sent to station: checkbox to send the MOH for Station-to-Station call.

2) Delete the HMSC data:
TYPE: "Sound Source LENS".
HMSC: Extended Hold Music source (0-63).
LENS: LENS of the Extended Hold Music source.

STEP 2: **ASYD** - System Data 1, Index 42, Bit 3: Trunk Hold Tone sending from Circuit #0 of DAT card. Assign 0.
STEP 3: Perform a system initialization

To assign data for this feature:

STEP 1: **ASYD** - System Data 1, Index 251, Bit 7: Multiple Music-on-Hold. Assign 1 (In service).
STEP 2: **AHMS** - Assign the Trunk LENS that is used for the music source to a HMSC number; then specify the KIND For the Extended Hold Music.

Note: This procedure takes about five minutes before the changes are implemented.

Note: Both settings 1) and 2) are required.

1) Establish a link between the Tenant Number and the HMSC number.
TYPE: Select "Extended Hold Music".
KIND: Select an Operation Kind from the following:
- Tenant Number of Holding Station
- Route Number of Held Trunk
- Tenant Number of Holding Station and Route Number of Held trunk
TN: Enter a Tenant Number (1-63) to which Holding Station belongs.
RT: Enter an Incoming Route Number (1-255).

HMSC: Specify an Extended Hold Music source (0-63).
Specified Music on Hold is sent to station: Place a check mark in the checkbox to send the MOH for Station-to-Station call.

2) Establish a link between the HMSC number and the Trunk LENS that is used for the music source.
TYPE: Select "Sound Source LENS".
HMSC: Specify an Extended Hold Music source (0-63).

Note: Assign the same HMSC number as in the TYPE: Extended Hold Music.

LENS: LENS of the Extended Hold Music source.

3. When MULTIPLE MUSIC-ON HOLD [M-99] has already been assigned.

Delete the data for this feature and then assign new data.

To delete data for this feature:

STEP 1: **AHMS** -

1) Delete the entry linking the Tenant Number to the HMSC data:
TYPE: "Extended Hold Music".
KIND:
- Tenant Number of Holding Station
- Route Number of Held Trunk
- Tenant Number of Holding Station and Route Number of Held trunk

TN: Tenant Number (1-63) to which Holding Station belongs.
RT: Incoming Route Number (1-255).
HMSC: Extended Hold Music source (0-63).
Specified Music on Hold is sent to station: checkbox to send the MOH for Station-to-Station call.

2) Delete the HMSC data:
TYPE: "Sound Source LENS".
HMSC: Extended Hold Music source (0-63).
LENS: LENS of the Extended Hold Music source.

To assign data for this feature:

STEP 1: **AHMS** - Assign the Trunk LENS that is used for the music source to a HMSC number; then specify the Kind for the Extended Hold Music.

Note: This procedure takes about five minutes before the changes are implemented.

Note: Both settings 1) and 2) are required.

1) Establish a link between the Tenant Number and the HMSC number.
TYPE: Select “Extended Hold Music”.
KIND: Select an Operation Kind from the following:
- Tenant Number of Holding Station
- Route Number of Held Trunk
- Tenant Number of Holding Station and Route Number of Held trunk

TN: Enter a Tenant Number (1-63) to which Holding Station belongs.
RT: Enter an Incoming Route Number (1-255).
HMSC: Specify an Extended Hold Music source (0-63).
Specified Music on Hold is sent to station: Place a checkmark in the checkbox to send the MOH for Station-to-Station call.

2) Establish a link between the HMSC number and the Trunk LENS that is used for the music source.
TYPE: Select “Sound Source LENS”.
HMSC: Specify an Extended Hold Music source (0-63).

Note: Assign the same HMSC number as in the TYPE: Extended Hold Music.

LENS: LENS of the Extended Hold Music source.

M-101 MF ANI to SMDR

General Description

This feature allows an office receiving ANI information by means of MF signal to output the information to the expanded SMDR area.

Note: This feature is available in North America only.

Operating Procedure

No manual operation is required.

Service Conditions

1. Output conditions are same as current conditions of MF ANI. (i.e., charging information for Incoming Calls is output only).
2. About the expanded output format, refer to SMDR OUTPUT EXPANSION-ANI/CPN [S-129].
3. When the tandem Connection is established through CCIS, the maximum digits of ANI information output to the SMDR port are 16.

Programming

Also see TRUNK LINE APPEARANCE [T-23] for information on how to accommodate Central Office line on a Line/Feature key.

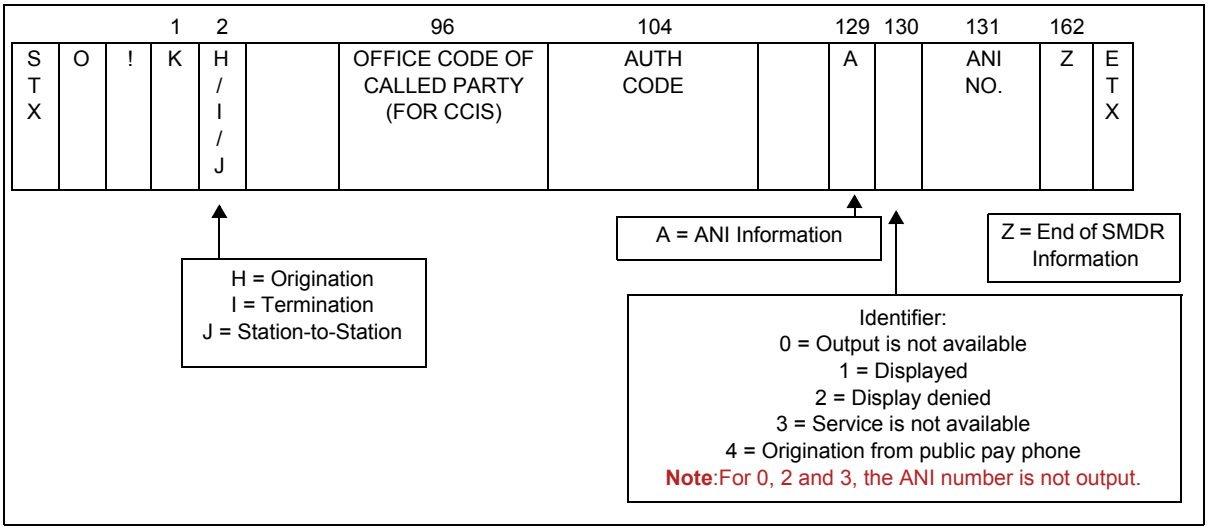
STEP 1: **ASYD** - System Data 1, Index 241, Bit 4. 0/1 = No ANI output to SMDR/ANI output to SMDR.

System Data 1, Index 241, Bit 6. 0/1 = When the length of Caller ID exceeds the limit, only the former/latter part of data appears on the LCD.

System Data 1, Index 390, Bit 0. Caller ID Display Pattern on the LCD. 0/1 = Calling Number/Calling Name has priority.

System Data 1, Index 390, Bit 2. when using Dterm Two-line LCD, the lower LCD can be used for displaying Caller ID. 0/1 = Out/In service.

When Calling Number is output to SMDR, the following expanded format is used.



System Data 1, Index 288, Bit 5. Expanded SMDR Output for SMDR A Apparatus. 0/1 = Out/In Service.

System Data 1, Index 290, Bit 5. Expanded SMDR Output for SMDR B Apparatus. 0/1 = Out/In Service.

System Data 1, Index 292, Bit 5. Expanded SMDR Output for SMDR C Apparatus. 0/1 = Out/In Service.

System Data 1, Index 294, Bit 5. Expanded SMDR Output for SMDR D Apparatus. 0/1 = Out/In Service.

System Data 1, Index 296, Bit 0. Output of Calling Number (ANI Information). 0/1 = Out/In Service. Assign data "1".

System Data 1, Index 300, Bit 0. 0/1 = Billing information is sent to the Main Node in Expanded SMDR format if CENTRALIZED BILLING - CCIS [C-55].

STEP 2: **ASFC** - Assign a Service Feature Restriction Class that allows SFI 14 to Stations allowed Station MESSAGE DETAIL RECORDING [S-10].

STEP 3: **ARTD** - Assign Route Class Data for the Central Office Trunk.

RT: 1
15-LSG :3 50-DPLY :1
When Calling Number is output to SMDR:
RT: 1
10-SMDR :1
Assign Route Class Expansion Data: CNI - 1. SVI: 1645 = 1

When My Line is Prime Line, CALL ID CHANGE DISPLAY Feature key is effective while terminating or in conversation.

When My Line is not Prime Line, CALL ID CHANGE DISPLAY Feature key is effective while in conversation.

However, Display Pattern returns to the original setting when answering, holding, or transferring the call.

STEP 4: **ATRK** - Assign Trunk data for the Central Office Trunk.

STEP 5: **MBTK** - Cancel the make busy of the Central Office Trunk.

STEP 6: **AKYD** - Assign CALL ID CHANGE DISPLAY Feature key. Assign Type:Function:CALL ID CHANGE DISPLAY [113]. When CALL ID CHANGE DISPLAY Feature key is pressed, Display Pattern on the LCD assigned by ASYD command changes according to the conditions above.

FCCS Programming

This data setting describes only the additional or replacement commands used to activate FCCS Services. When used as an FCCS feature, these commands replace the non-network commands (for example, AAED is replaced by AAEDN). Note that the data setting for FCCS Service varies depending on the Node.

Note: When using ASYDL or ASYDN to program an FCCS feature, the basic programming of ASYD is required in addition to ASYDL or ASYDN.

STEP 1: **ASYDL** - System Data 1, Index 576, Bit 0 = 1. Enable central SMDR polling method for FCCS.

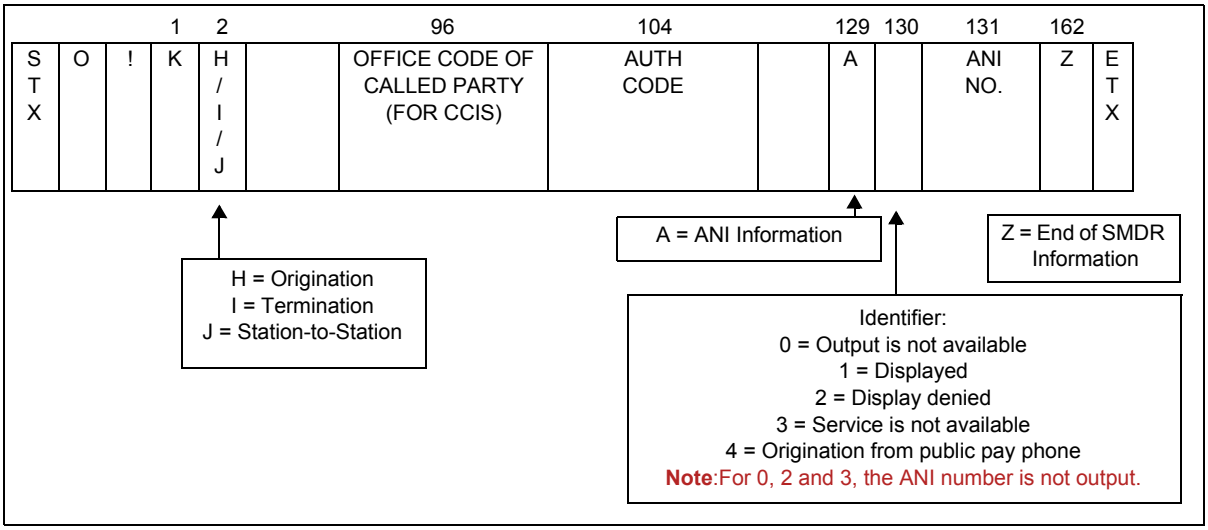
System Data 1, Index 241, Bit 4. 0/1 = No ANI output to SMDR/ANI output to SMDR.

System Data 1, Index 241, Bit 6. 0/1 = When the length of Caller ID exceeds the limit, only the former/latter part of data appears on the LCD.

System Data 1, Index 390, Bit 0. Caller ID Display Pattern on the LCD. 0/1 = Calling Number/Calling Name has priority.

System Data 1, Index 390, Bit 2. when using Dterm Two-line LCD, the lower LCD can be used for displaying Caller ID. 0/1 = Out/In service.

When Calling Number is output to SMDR, the following expanded format is used.



System Data 1, Index 288, Bit 5. Expanded SMDR Output for SMDR A Apparatus. 0/1 = Out/In Service.

System Data 1, Index 290, Bit 5. Expanded SMDR Output for SMDR B Apparatus. 0/1 = Out/In Service.

System Data 1, Index 292, Bit 5. Expanded SMDR Output for SMDR C Apparatus. 0/1 = Out/In Service.

System Data 1, Index 294, Bit 5. Expanded SMDR Output for SMDR D Apparatus. 0/1 = Out/In Service.

System Data 1, Index 296, Bit 0. Output of Calling Number (ANI Information). 0/1 = Out/In Service. Assign data "1".

System Data 1, Index 300, Bit 0. 0/1 = Billing information is sent to the Main Node in Expanded SMDR format if CENTRALIZED BILLING - CCIS [C-55].

STEP 2: **ARTDN** - Assign Route Class Data for the Central Office Trunk.

RT: 1
15-LSG :3 50-DPLY :1

When Calling Number is output to SMDR:

RT: 1
10-SMDR :1

Assign Route Class Expansion Data: CNI - 1. SVI: 1645 = 1

When My Line is Prime Line, CALL ID CHANGE DISPLAY Feature key is effective while terminating or in conversation.

When My Line is not Prime Line, CALL ID CHANGE DISPLAY Feature key is effective while in conversation.

However, Display Pattern returns to the original setting when answering, holding, or transferring the call.

M-104 Message Waiting-Special Dial Tone

General Description

This feature allows the system to send Special Dial Tone when the handset is off-hook, indicating that Message Waiting has been set to the Station.

Note: This feature is not available in North America.

Operating Procedure

None.

Service Conditions

Even if this feature is set, normal Dial Tone is received when the CALL HOLD [C-6] Access Code is dialed.

Programming

STEP 1: **ASFC** - Assign a Service Feature Restriction Class that allows SFI = 142 to the Stations using MESSAGE WAITING-SPECIAL DIAL TONE.

M-105 Multiple Music on Hold Source

General Description

This feature is programmed to let the system use the so-called "Voice Prompt" functions instead of the audible tone generally supplied from the Digital Tone Generator (DTG) or music ROM. When the system is equipped with an Announcement Trunk, such as DAT, COT or EMT, the user can activate this alternative tone provision service on condition that the announcement port and Speech Path Memory (SPM) in the following LENS are catenated by nailed-down Connections.

Note: This feature is not available in North America.

[Locations of Speech Path Memory]

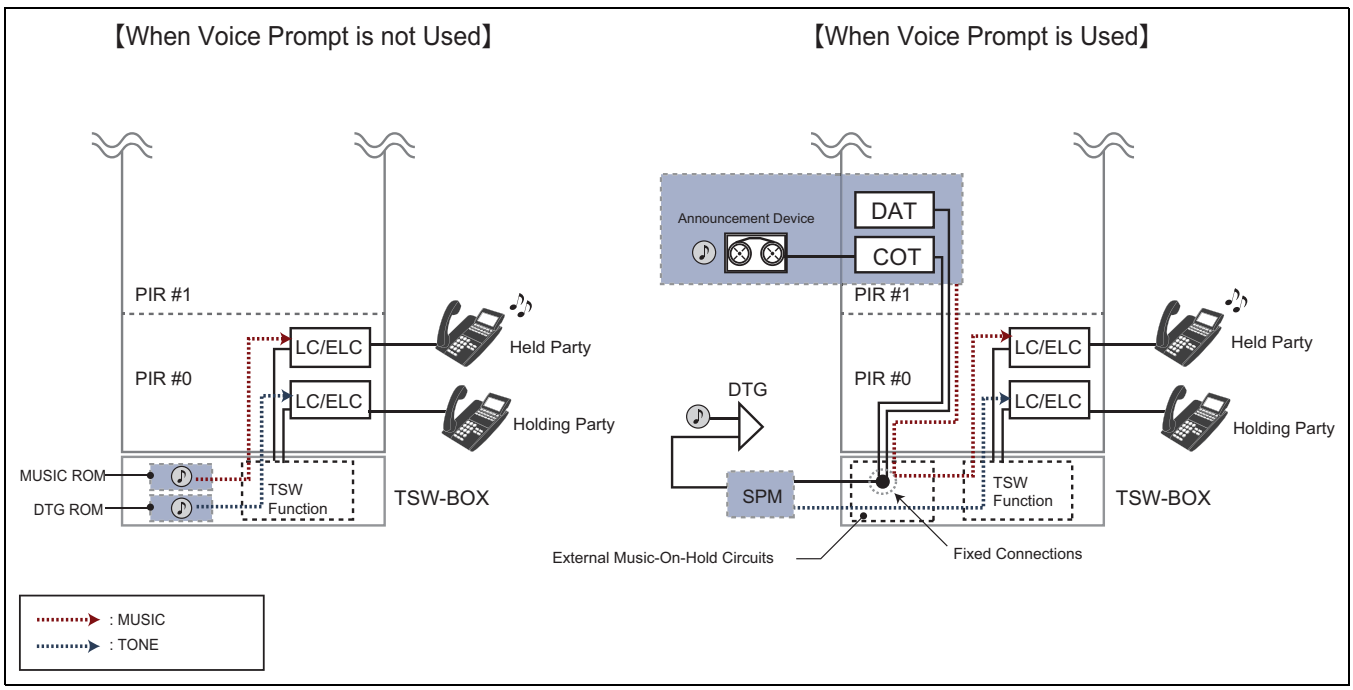
MG = 00, Unit = 2, Group = 24, Level = 0~7
MG = 00, Unit = 2, Group = 25, Level = 0~7] SPM for TONE channel 16~31

MG = 00, Unit = 0, Group = 24, Level = 0~7
MG = 00, Unit = 0, Group = 25, Level = 0~7] SPM for TONE channel 00~15

MG = 01, Unit = 2, Group = 24, Level = 0~5
MG = 01, Unit = 2, Group = 25, Level = 0~7] SPM for MUSIC channel 16~29

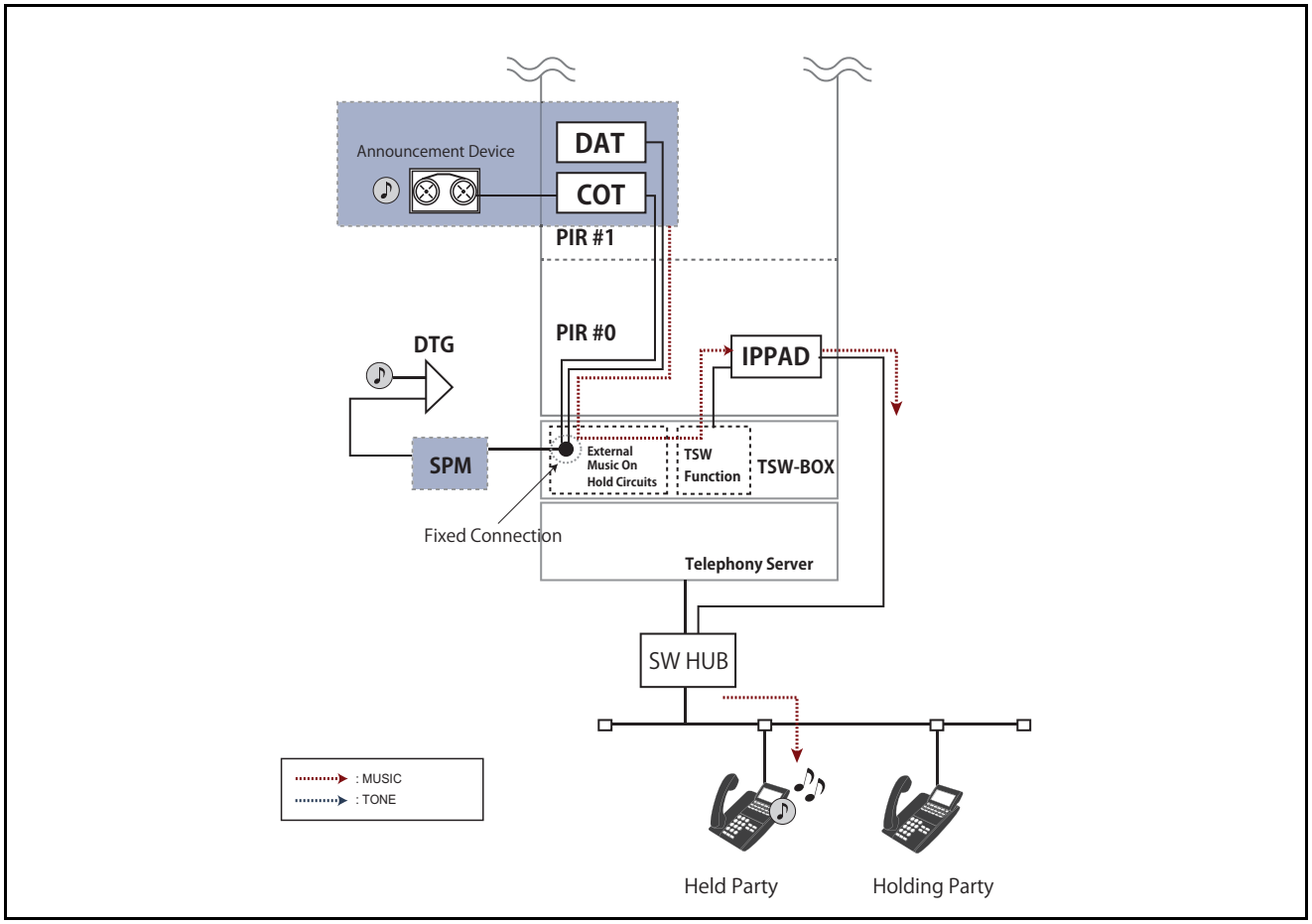
MG = 01, Unit = 0, Group = 24, Level = 0~7
MG = 01, Unit = 0, Group = 25, Level = 0~7] SPM for MUSIC channel 00~15

As shown in the figure below, the TONE ROM accommodated on the TSW card generally contains the audible tone, and EMA/PLO card provide the Hold Tone (music). However, if a DAT/COT/EMT card is accommodated, the system can provide audible tone (Voice Prompt) by fixed Connection established between SPM and the card's Announcement Trunk(s).



IPPAD card is required for IP terminals receiving MOH from Multiple Music on Hold source. If IPPAD card is not installed in the system or the card is link block state such as out of the assigned port number, the IP terminal receives internal MOH.

When Providing Multiple Music On Hold Source to IP terminal



The following shows the tone kinds corresponding to the tone output channels #00-15

CH NO.	KIND OF TONE	REMARKS
00	Dial Tone (DT)	
01	Special Dial Tone (SPDT)	
02	Ringback Tone (RBT)	
03	Continuous Ringback Tone (CRBT)	
04	Busy Tone (BT)	
05	Reorder Tone (ROT)	
06	Service Set Tone (SST)	
07	Second Dial Tone (SDT)	
08	-	
09	Sender Transmitting Tone (SDTT)	
10	Call Waiting Ringback Tone (CWRBT)	
11	-	
12	-	

CH NO.	KIND OF TONE	REMARKS
13	Test Tone (TSTT)	
14	-	
15	-	

The following shows the music kinds corresponding to the MUSIC output channels #00-15.

CH NO.	KIND OF MUSIC	REMARKS
00	Music On Hold (MSC)	
01 ʃ 15	Not used	

Operating Procedure

No manual operations required.

Service Conditions

1. This feature is not available in North America.
2. For the activation of this feature, dedicated hardware, such as DAT/COT/EMT card, is required.
3. LENS (PIM/PIR) needs to be accommodated for SPM (Speech Path Memory). System capacity license software for more than three PIM/PIR is required.
4. Restriction of Announcement Machine Connection by Route Restriction Class (ARSC command) is not available.
5. After system initialization, it takes some specific time to complete the nailed-down Connection between the SPM and the designated Announcement Trunk(s). During the pending time, the Voice Prompt will not be activated (i.e. the service will only be effectuated on the strict condition that the Connection has been firmly established).
6. The system does not cover the Head Start function for announcement guidance.
7. This feature cannot be used with FCCS.

<Conditions for IP terminals>

8. TONE is not available for IP terminals.
9. Whenever using external MOH with IP terminal, the port on IPPAD is hunted for the Connection.
10. External MOH for IP terminal cannot be provided when PHC (Internal PHC) is used.

<Conditions Specific to MMG/UMGi/RNIP System>

1. External MOH can be provided only for Stations and Trunks in the LP accommodating Speech Path Memory (SPM).
2. When receiving External MOH at IP Stations (IP Enabled Dterm, DtermIP or Analog MC), both IPPAD circuit card and Speech Path Memory (SPM) must be accommodated in the same LP.

Interactions

This feature is not available for ISDN Trunk, ISDN terminal and PS.

Programming

STEP 1: **ASYD** - System Data 1, Index 60, Bit 4 = 1 (Nailed Down Connection is effective)

STEP 2: **ASYDL** - System Data 1, Index 642-649, Bit 0-7

Voice Prompt: Each channel's tone source data.

As shown in the table below, each Bit represents the TONE output channel (a total of 32ch). When "1" is set, the TONE source for the channel is specified as Announcement Trunk(s), such as DAT, COT, EMT (Voice Prompt). If the data is "0", the source will be the Tone ROM.

Index	b7	b6	b5	b4	b3	b2	b1	b0
642	ch7	ch6	ch5	ch4	ch3	ch2	ch1	ch0
643	ch15	ch14	ch13	ch12	ch11	ch10	ch9	ch8
644	ch23	ch22	ch21	ch20	ch19	ch18	ch17	ch16
645	ch31	ch30	ch29	ch28	ch27	ch26	ch25	ch24

0/1 = Tone ROM/SPM

Index	b7	b6	b5	b4	b3	b2	b1	b0
646	ch7	ch6	ch5	ch4	ch3	ch2	ch1	ch0
647	ch15	ch14	ch13	ch12	ch11	ch10	ch9	ch8
648	ch23	ch22	ch21	ch20	ch19	ch18	ch17	ch16
649			ch29	ch28	ch27	ch26	ch25	ch24

0/1 = MUSIC ROM/SPM

STEP 3: **ARTD** - Assign data on the route for Announcement Trunk(s) to be used for Voice Prompt functions. (The Announcement Trunk can be DAT, COT or EMT, etc.) Refer to the figure below.

Note: To change the data for existing route (RT), initialize or reboot the trunk device such as circuit card, MG, or VS32.

The data assignment for the Digital Announcement Trunk (DAT) is shown below as an example:

```
RT:1
1-OSGS  :2   2-ONSG  :0   3-ISGS  :0   4-INSG  :0   5-TF    :0
6-TCL   :4   7-L/T   :1   8-RLP   :0   9-TQ    :0   10-SMDR :0
11-TD   :0   12-DR   :0   13-AC   :0   14-TNT   :0   15-LSG  :0
16-SMDR2 :0  17-H/M  :0   18-MC   :0   19-EANI  :0   20-D    :0
21-MSB  :0   22-MSW  :0   23-TR   :0   24-OC    :0   25-R/L  :0
26-RSVD :0   27-TL   :0   28-ANS  :0   29-TELP  :0   30-PAD  :0
31-OGRL :0   32-ICRL :0   33-HD   :0   34-GUARD :0   31-WINK :0
36-VAD  :0   37-CLD  :0   38-FA   :0   39-BC    :0   40-TCM  :0
41-TDMQ :0   42-TPQ  :0   43-BT   :0   44-PRV   :0   45-A/D  :0
46-CW   :0   47-TPQ  :0   48-BL   :0   49-TRKS  :0   50-DPLY :0
51-ACD  :0   52-2W/4W :0  53-FAAT  :0   54-GW    :0
```

STEP 4: **ATRK** - Assign data on Announcement Trunk(s) to be used for Voice Prompt service.

STEP 5: **MBTK** - Let the Trunk(s) assigned in STEP 2 be workable for the Voice Prompt carrier. Cancel the Make-busy status applied to the Trunk(s) by assigning "0 (Make idle)" to the parameter "MB."

STEP 6: **AFCD** - Assign the data related to Fixed Connections between the Announcement Trunk(s) and the Speech Path Memory. LENS data for the Announcement Trunk is assigned in the LENS-A parameter and LENS data for the Speech Path Memory is assigned in the LENS-B parameter.

M-107 Message Audible Alert

(Series 8700 software feature)

General Description

Messages (like voice mail or message waiting lamp control) can be realized by MCI or digital voice integration. This feature with MCI is an adjunct to MESSAGE CENTER INTERFACE [M-26] and MESSAGE CENTER INTERFACE - CCIS [M-67], which provides normal voice mail service. When MCI or digital voice integration sends a new voice mail notification to a station, a chime (audible alert) information will also sent to the station.
Also, This feature with MCI lamp control on control is an adjunct to MESSAGE WAITING LAMP ON - ATTENDANT [M-23] and MESSAGE WAITING LAMP ON - STATION [M-30]. When a lamp on notification is sent to a station, a chime (audible alert) information will also send to the station. The station will have a light flash followed a chime after receiving the notification.

Note: This feature is available in North America only.

Operating Procedure

Normal voice mail operation procedure is same as MESSAGE CENTER INTERFACE [M-26], MESSAGE CENTER INTERFACE - CCIS [M-67] or digital voice integration. Normal Message Waiting Lamp on procedure is same as MESSAGE WAITING LAMP ON - ATTENDANT [M-23] or MESSAGE WAITING LAMP ON - STATION [M-30]. This feature can be controlled on a system basis or assigned to a Feature key or Access Code.

Service Conditions

1. The message audible alert key can be assigned by Feature key (AKYD), Soft key (ADSL) or Access Code (ANPD, ASPA).
2. If the ASYDL, System Data 1, Index 805, Bit 3 and Bit 4 is on, the Station will chime each time when it receives a new message notification.
3. If the ASYDL, System Data 1, Index 805, Bit 3 is on and Bit 4 is off, if message audible alert key is set on, the station will chime each time when it receives a message notification. If the key is set off, the Station will not chime.
4. If the ASYDL, System Data 1, Index 805, Bit 3 is off and Bit 4 is on, the Station will chime only one time when it receives multiple message notifications.
5. If the ASYDL, System Data 1, Index 805, Bit 3 is off and Bit 4 is off, if message audible alert key is set on, the Station will chime only one time when it receives multiple message notifications. If the key is set off, the Station will not chime.
6. The feature works for MCI, AD40 MCI and AD120, not for AD40 voice digital integration.
7. The message audible alert key is set or cancelled when a Station is at idle condition.
8. If the ASYDL, System Data 1, Index 805, Bit 4 is off and the message audible alert key on a Station is set on, then the administrator turn the SFC off, the key can not be cancelled and a restriction “RST” will be displayed. Resetting the circuit card will clear the message audible alert key.
9. Only one Feature key can be assigned per Dterm station for this feature. Multiple Feature key assignments have unpredicted results.
10. When the Station is busy, if it is a single line, it will lamp on without audible alert after the Station hangs up. It is also true for Dterms with MCI. But for Dterms with digital voice integration, it will lamp on with audible alert.

11. With outside power source (80V), a single line will chime (short ring burst-audible alert) with lamp on when a new message arrives it. Without outside power source, a single line will chime (short ring burst-audible alert) with when a new message arrives.
12. Firmware SP3295 or later version is needed for this feature work properly for Dterm Stations.
13. PA-16LC Cards that do not support Audible Alert when MW is ON: PA-16LCBY, PA-16LCBW, and PA-16LCBV.

Interactions

1. This feature is based on MESSAGE CENTER INTERFACE [M-26], MESSAGE CENTER INTERFACE - CCIS [M-67] and digital voice integration specification. All interactions related to them are effective for this feature.
2. This feature also supports tmm, fmm, hm, mci notification like vmm. When a Station receives those notifications, it will chime like vmm (voice mail).
3. When a Station receives a Waiting Lamp on from an Attendant [M-23] or a station [M-30], it will chime like vmm.

Programming

All programming in M-23, M-26, M-30, M-67or digital voice integration are applied on this feature. The following is added programming:

STEP 1: **ASYDL/ASYDN** - System Data 1, Index 805, Bit 3: 0/1 = Audible Alert for first message/Audible Alert for all messages.

System Data 1, Index 805, Bit 4: 0/1 = Audible Alert by SFC and FKY /Audible Alert for all Stations in the system.

STEP 2: **AKYD** - Assign Message Audible Alert key [149].

STEP 3: **ADSL** - Assign Message Audible Alert key [149].

STEP 4: **ASPA** - Assign an Access Code for set Message Audible Alert: SRV = “SSCA: Service Code Appendix”, SIDA = 140 and an Access code for cancel Message Audible Alert: SRV = “SSCA: Service Code Appendix”, SIDA = 141.

STEP 5: **ASFC** - Assign Service Feature Restriction Class of the Station.
SFI = 201 (Message Audible Alert)
RES = 1 (Allowed)

M-111 Media Gateway SIP

General Description

Media Gateway-Session Initiation Protocol is an interface that enables your system to build multi-service and reliable Voice over IP (VoIP) networks using Session Initiation Protocol (SIP) defined by IETF. SIP is an application-layer control protocol that can initiate call setup, routing, authentication and transferring other feature messages to endpoints.

The MG-SIP supports the SIP standard terminals based on RFC3261. Thereby, the MG-SIP, which is controlled by Internal PHI-BRI of the Telephony Server, provides a full array of the call routing capabilities to maximize network performance in the packet voice network (SIP network). The MG-SIP provides the system with a flexible and cost-efficient network environment without wasting the accumulated assets of the NEC products.

"MG-SIP" refers to the following devices.

DEVICE TYPE		NUMBER OF USABLE CHANNELS Note 1	MG GROUP/MG NUMBER Note 3
MG-SIP128 [MG-128SIPMGL-A] Note 2		1-128ch	1-15
MG-SIP128 [MG-128SIPMGG-B/MG-128SIPMGG]			
MG-SIP96		1-96ch	1-4
MG-SIP16		1-16ch	1-4
MG-SIP16 [SCA-16SIPMGA/MG-16SIPMGA]			1-15
MG-SIP16 [SCA-16SIPMG(US)-B/SCA-16SIPMG(US)]			1-15
Software-based MG-SIP Note 4	SP-4085 MGSIPVM PROG-E Note 2	1-254ch Note 5	1-15
	SP-4078 MGSIPVM PROG-B	1-128ch	1-15

Note: Notice the following:
- Connection test with common carrier is required.
- For using MG-SIP TIE Line, MG-SIP128 [MG-128SIPMGJ-B/MG-128SIPMGJ] or Software-based MG-SIP [SP-4080 MGSIPVM PROG-D] is required. For details of MG-SIP TIE Line, refer to MG-SIP TIE Line [M-122].

Note 1: The number of usable channels varies depending on the connecting carrier.

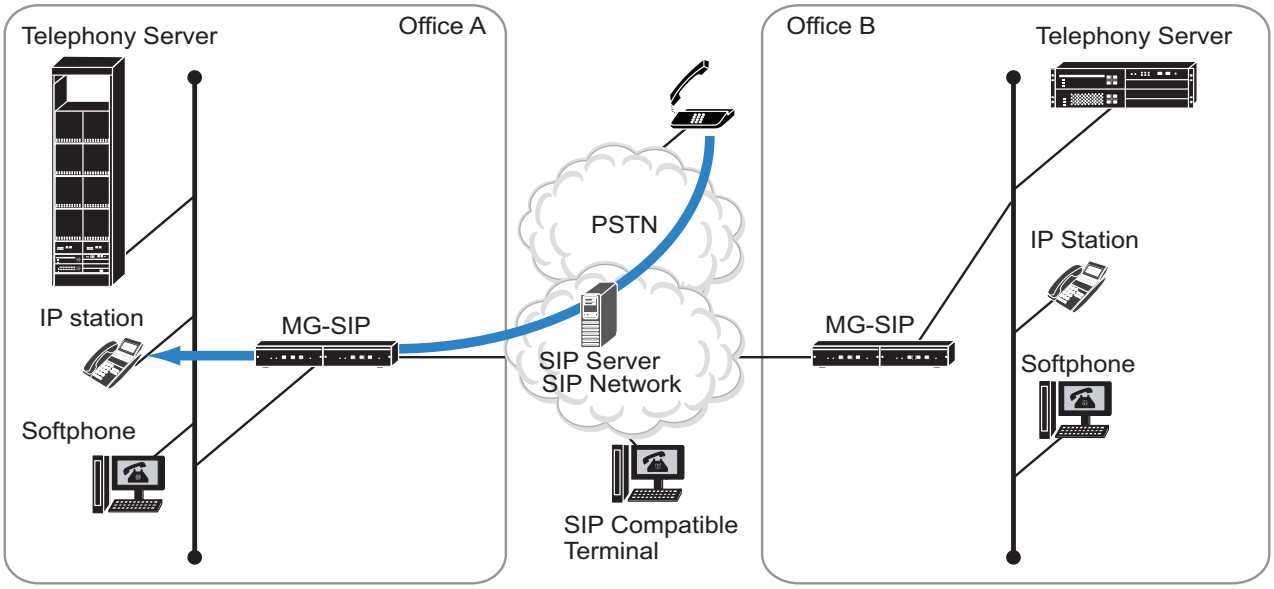
Note 2: From FP95-114 V4 Issue 2.0, RTP encryption on Telephony Server network side is supported. RTP encryption on SIP network side is not supported. For details, see VOIP ENCRYPTION [V-27].

Note 3: MG Group is the group which is comprised of multiple MG-SIPs for alternate routing. Alternate routing can be executed between MG-SIPs registered to the same MG Group. For details, refer to Multiple Number Service (MG-SIP) [M-113].

Note 4: Notice the following:
- Available since FP95-112 V2.
- Software-based MG-SIP is a virtual machine installed in SV9500 (Software Model). See Software-based MG-SIP chapter in Software Model Peripheral Equipment Description.

Note 5: Available since FP95-114 V4.

The following illustration shows the system configuration when MG-SIP is accommodated.



Operating Procedure

None.

Service Conditions

1. Internal PHI-BRI is required to perform this feature. Internal PHI-BRI is built into the Telephony Server and it controls the Voice Procedure of the MG-SIP via TCP/IP.
2. The number of available Internal PHI-BRIs is one per system.

   Note: In UMGi system, the data assignment of Internal PHI-BRI is required per node (CCN/RN).
3. Virtual Trunk is registered to Physical PIR or Virtual PIR.

4. The maximum number of System Capacity is 6144 channels. The Maximum number of devices and channels per system is equivalent to the capacity of the system. The number of devices varies depending on the total accommodated channels. When 6144 channels are occupied only by MG-SIP, other devices cannot be connected. The following table shows the details on the accommodation limitations only for MG-SIP.

	Capacity of IP Components	Maximum Number of Simultaneous Connections	Maximum Number of Speech Channels	Maximum Number of Ports to be able to use
MG-SIP128	44 devices	1840 ch (460ch/LP) **Note 6**	5632 ch	6132 ch
MG-SIP96	60 devices		5760 ch	
MG-SIP16	188 devices		3008 ch	

Note 6: The maximum number of simultaneous speech channels available includes other ISDN channels.

Note: Notice the following:
- Software-based MG-SIP is an application for MG-SIP installed in the virtual machine. When using software-based MG-SIP, the number of devices and channels that can be accommodated in the system includes software-based MG-SIP channels.
- The number of accommodated capacity can be up to four times in the UMGi system.

5. Speech channel is established by Virtual trunk. When the MG-SIP Virtual Speech channels are registered to a HW (4-Group), channels other than Virtual Speech channels can not be registered to the same HW. To use more than 32-channels, register the channels per 32-channels from top channel on a HW. However, Virtual Speech channels are not required to register for a consecutive HWs.

6. Only Voice, FAX, Video, and DTMF (PB Signaling) can be established by connections via MG-SIP. This feature does not support Unrestricted digital, Modem connection, SIP Instant Message [S-151] etc.

Item	Specification	Note
Voice	G.711	64k [bps]: PCM μ-law/A-law
	G.729a	8k [bps]: CS-ACELP (Reduce complexity)
FAX **Note 8**	G.711 (PassThrough)	64k [bps]: PCM μ-law/A-law
	T.38	**Note 7**
Video **Note 9**	G.711	**Note 7**
	G.729a	
DTMF	G.711 (PassThrough)	
	Proprietary method	Toward the Telephony Server network
	RTP (RFC2833)	Toward the Telephony Server network and SIP network For DTMF Digits, Telephony Tones and Telephony Signals

Note 7: Supported from FP95-114 V4. The following MG-SIPs and firmwares are required:
- MG-SIP128 [MG-128SIPMGL-A]
- Software-based MG-SIP: SP-4085 MGSIPVM PROG-E Issue 1 or later

Note 8: For Fax data transmission using MG-SIP, see IP CONTROL-FAX [I-44] in this manual.

Note 9: For Video Call using MG-SIP, see VIDEO CALL [V-31] in this manual.

Note: Depending on the IP equipment or the condition of SIP network, some of the above conditions may not be available for MG-SIP.

Note: In case DTMF is sent via RFC2833, RTP Warning Tone may play back at the reception side.

<DTMF relay method>

DTMF relay method on the SIP network side is set by the configuration command "dtmf_mode" on the MG-SIP. DTMF relay method on the Telephony Server network side is automatically set according to the one on the MG-SIP network side, as follows.

DTMF relay system on the SIP network side	DTMF relay system on the Telephony Server network side	Remarks
RFC2833 (out-band)	Proprietary (out-band)	Conversions between "RTP (DTMF: RFC2833) packets on SIP network" and "UDP (DTMF: Proprietary) packets on Telephony Server network"
RFC2833 (out-band) **Note 10**, **Note 11**		When the setting of RFC2833 packet transparency is enabled, pass-through of "RTP (DTMF: RFC2833) packets on SIP network and Telephony Server network" are available. For details, refer to command section (set rfc2833_pass) of each MG-SIP in Peripheral Equipment Description (IP Devices).
G.711 PassThrough (in-band)		Conversions between "RTP packets on SIP network" and "RTP packets on Telephony Server network"

Note 10: The following MG-SIPs with firmware are required for RFC2833 packet transparency:
- MG-SIP16 [SCA-16SIPMG (US)-B/SCA-16SIPMG (US)]: SP-3988 MGSIP (16) PROG-G Issue 10 or later
- MG-SIP128 [MG-128SIPMGL-A]: SP-4089 MGSIP(MM) PROG-A Issue 1 or later
- Software-based MG-SIP: SP-4085 MGSIPVM PROG-E Issue 1 or later

Note 11: RFC2833 packet transparency is available only when the terminals on the Telephony Server network side and SIP network side support RFC2833.

7. MG-SIP makes a connection using SIP Server, which helps to create multi-location connection. SIP server can use Proxy Server and Registry Server. User Agent (UA) has been registered to SIP Server.
8. IP Masquerade feature can make a VoIP connection between Private IP Address and Global IP Address.

   IP Masquerade feature translates UDP and TCP Ports, thereby, it can connect multiple Private IP Addresses to single Global IP Address.

   No special equipment is necessary for performing IP Masquerade. However, the internet that may cause delay or does not guarantee bandwidth cannot be used (Only the operation on an intranet is guaranteed).
9. When third party SIP-based terminals (such as SIP Gateway, SIP terminal, SIP Server) are used in the SIP network, test these terminals in advance and make sure that they are available to use (also, if there is any related additional service that you want to use, confirm that is available). Note that the terminal cannot be managed as a station under the Telephony Server.
10. Tone and holding tone to SIP network is generated by MG-SIP. When MGC (Internal PHI-BRI) external holding tone is selected by ASYDL command, the information is provided via IPPAD card.

11. Caller ID will not be sent to the forwarding destination when a call is transferred toward the public SIP network by forwarding features such as Call Forwarding - Don't Answer [C-3], Call Forwarding - Busy Line [C-2], and Call Forwarding - All Calls [C-5].

    Note: When ISDN CLI WITH CALL FORWARDING OUTSIDE [I-50] feature is activated, it is possible to send Caller ID to SIP network. (Available in Australia only.)

12. A single route can be shared by multiple MG-SIPs (maximum 8 devices). For details, refer to MG-SIP Shared Route [M-123].
13. MG Registration Status can be displayed by using the DIMG command.
14. Transferring only Voice RTP Payload, MGC has no effect on the jitter-buffer size or Echo Canceller, which are controlled by the endpoint devices.
15. MG-SIP can receive incoming calls through AUTOMATED ATTENDANT [A-82]/REMOTE ACCESS TO SYSTEM [R-2] under the following conditions.
    a.) AUTOMATED ATTENDANT [A-82]/REMOTE ACCESS TO SYSTEM [R-2] may not work properly, depending on the system configuration of SV9500 network and the conditions for DTMF specifications in SIP network. **Note 12**
    The conditions of system configuration are as follows:

DTMF in SIP Network	Telephony Server	SR-MGC	System Configuration
Fax PassThrough	Available*	Unavailable	Set ARTI/ARTIN CDN77(RA_RT) = 1 (Register card is used). *PIR is required to use IPPAD. *IPPAD is required.
RFC2833	Available	Available	Set ARTI/ARTIN CDN77(RA_RT) = 0 (Virtual Register is used). *PIR is not required because Virtual Register is used.

    Note 12: The specifications for DTMF in SIP network varies, depending on the network carrier.

    b.) PB Receiver becomes invalid while receiving an announcement for AUTOMATED ATTENDANT [A-82]/REMOTE ACCESS TO SYSTEM [R-2].
    c.) After dialing a target telephone number, a beep may be heard before Ringback Tone (RBT), but it's not abnormal.

16. SERIAL CALL -D [S-127D] is not available for incoming call via MG-SIP.
17. MG-SIP does not support ring down feature.
18. To use VLAN feature of MG-SIP, connect switch supporting Port VLAN feature (Tag VLAN is not supported).
19. Alternate routing to PSTN line can be activated in the case of the following connection failure on IP network.
    - LAN cable failure on MG-SIP's LAN side (Telephony Server network) connector → Alternate routing is activated after health check time-out detection.

- Network failure between the Telephony Server and MG-SIP → Alternate routing is activated after health check time-out detection.
- MG-SIP failure (such as power-off, etc.) → Alternate routing is activated after health check time-out detection.
- LAN cable failure on MG-SIP's WAN side (SIP network) connector → Alternate routing is activated on detecting the cable failure.
- Network failure between MG-SIP and carrier's SIP server → Alternate routing is activated after originating connection time-out (6 seconds).
- MG-SIP receives error response from carrier's SIP server → Alternate routing is activated after receiving error response (400, 408, 481, 500, 503).
- All the available channels are busy or in make-busy state.

20. Designated Connection Test (PCPro) with the TOGC command is available for Media Gateway SIP.
21. MG-SIP supports Name Display service equivalent to CALLING NAME IDENTIFICATION PRESENTATION (CNIP)/CONNECTED NAME IDENTIFICATION PRESENTATION (CONP) [C-153]. For proper functioning of this feature, see service condition listed in CALLING NAME IDENTIFICATION PRESENTATION (CNIP)/CONNECTED NAME IDENTIFICATION PRESENTATION (CONP) [C-153] in Data Programming Manual - ISDN and the following.

(a) The following firmware is required for MG-SIP.

MG-SIP16 [SCA-16SIPMG(US)]	SP-3988 MGSIP (16) PROG-G Issue 2 or later
MG-SIP96	SP-3987 MGSIP PROG-B Issue 2 or later
MG-SIP128 [MG-128SIPMGL-A]	SP-4089 MGSIP(MM) PROG-A Issue 1 or later
MG-SIP128 [MG-128SIPMGG/ MG-128SIPMGG-B]	SP-4051 MGSIP PROG-G Issue 1 or later SP-4058 MGSIP PROG-J Issue 1 or later
Software-based MG-SIP	SP-4078 MGSIPVM PROG-B Issue 1 or later SP-4085 MGSIPVM PROG-E Issue 1 or later

(b) Configure the set cpn_pattern command to obtain cpn (calling party number) from the P-Asserted-Identity header, not from the Displayname field. If the set cpn_pattern command is not configured or the P-Asserted-Identity header is not added, the caller name is not available.

(c) When more than one P-Asserted-Identity header is added, this feature uses only the first header.

(d) The following single-byte alphanumerics (JIS8 code) are used as Name Display data. When double quotation marks (") and backward slash marks are used in the Displayname fields, these marks can be escaped (displayed) by putting a backward slash mark before them.

	2	3	4	5	6	7
0	a blank space	0	@	P	`	p
1	!	1	A	Q	a	q
2	“	2	B	R	b	r

	2	3	4	5	6	7
3	#	3	C	S	c	s
4	$	4	D	T	d	t
5	%	5	E	U	e	u
6	&	6	F	V	f	v
7	‘	7	G	W	g	w
8	(	8	H	X	h	x
9	)	9	I	Y	i	y
A	*	:	J	Z	j	z
B	+	;	K	[	k	{
C	,	<	L	¥	l	|
D	-	=	M	]	m	}
E	.	>	N	^	n	
F	/	?	O	_	o	

(e) Up to 16 characters can be displayed on the LCD. If a name has more than 16 characters, only the first 16 characters are displayed.

(f) In a network whose opposing point is MG-SIP, a destination number is not notified because “P-Asserted-Identity” header is not added.

(g) When messages received from the SIP network contains a “Privacy: id” header, the caller name is not available.

(h) This feature does not support “BusyName” notified when a called party is in busy state.

(i) Conditions for CALLING NAME IDENTIFICATION PRESENTATION (CNIP):

- CNIP supports “ini-INVITE” only.
- When Caller IDs are not notified, Name IDs are not provided regardless of the setting of the “set privacy pattern” command.
- To obtain the Displayname field as Name information, you need to select a method of obtaining cpn other than "Displayname" by the “set cpn pattern” command.

(j) Conditions for CONNECTED NAME IDENTIFICATION PRESENTATION (CONP):

- CONP supports “180 Ringing” and “200 OK” only.
- When 183 Session Progress (without SDP) arrives while the set 183rbt command is enabled, ALERT is transmitted, but the caller name is not available.

22. MG-SIP supports Route Advance service equivalent to ROUTE ADVANCE [R-6] and PRI/PRA FAILSAFE ROUTING [P-49/50]. When a call is originated via a MG-SIP and the system receives "503 Service Unavailable" from the SIP network, the call can take an alternate MG-SIP by these services. MG-SIP requires the following firmware.

MG-SIP16 [SCA-16SIPMG(US)]	SP-3988 MGSIP (16) PROG-G Issue 2 or later
MG-SIP96	SP-3987 MGSIP PROG-B Issue 2 or later
MG-SIP128 [MG-128SIPMGL-A]	SP-4089 MGSIP(MM) PROG-A Issue 1 or later
MG-SIP128 [MG-128SIPMGG]/ MG-SIP128 [MG-128SIPMGG-B]	SP-4051 MGSIP PROG-G Issue 1 or later SP-4058 MGSIP PROG-J Issue 1 or later
Software-based MG-SIP	SP-4078 MGSIPVM PROG-B Issue 1 or later SP-4085 MGSIPVM PROG-E Issue 1 or later

For proper functioning of this feature, see service condition listed in ROUTE ADVANCE [R-6] in Data Programming Manual - Business and PRI/PRA FAILSAFE ROUTING [P-49/50] in Data Programming Manual - ISDN.

23. When originating a tandem call to ACIS line via MG-SIP, if a destination station is busy, you can choose to hear Busy Tone (BT) or Ringback Tone (RBT) through the system data setting (ASYD, SYS1, Index 372, Bit 5). **Note 13**

Note 13: When Busy Tone (BT) is selected, billing is performed for the outgoing call.

24. When the Telephony Server side terminal releases during a conversation after a temporary communication failure between the Telephony Server and MG-SIP, MG-SIP may not send disconnection signals to the SIP network.
25. In this feature the system processes the timing of Guard Timer for Originating Call. Configure the settings for the timer via the system data (ASYD, SYS1, Index 474). Otherwise the settings are set to "256 milliseconds".
26. When a global IP Address of NAT is assigned to the MG-SIP, MG-SIP supports NAT. For the details, refer to command section (set nataddress) of each MG-SIP in Peripheral Equipment Description (IP Devices).

Note: The following firmware is required for MG-SIP:
- MG-SIP16 [SCA-16SIPMG (US)-B/SCA-16SIPMG (US)]: SP-3988 MGSIP (16) PROG-G Issue 10 or later
- MG-SIP128 [MG-128SIPMGL-A]: SP-4089 MGSIP(MM) PROG-A Issue 1 or later
- Software-based MG-SIP: SP-4085 MGSIPVM PROG-E Issue 1 or later

27. Feature Services available in the SIP network are described in page 1421.

<Tone Code by Countries>

MG-SIP plays Inband Tone by using Software Source. Tone Code by Countries is available under the following conditions.

(a) CODECs supported by TONE are: G.711μ-law, G.711A-law, and G.729a only.

(b) "No Tone" data is set to TONE data for other CODECs (G.723.1(5.3K) and G.723.1(6.3K)).

(c) CODECs supported by DTMF are: G.711μ-law, G.711A-law only.

(d) Music on Hold (MOH) is not supported.

(e) Short tone is not supported.

(f) Tone Code by Countries is available for the following services.
- Blind Transfer
- Calling to the tandem destination (Only in this case, Inband Tone RBT is sent to the station side.)
- Receiving REFER
- Automated Attendant

(g) Inband Tone of MG-SIP128 and software-based MG-SIP can be changed by using ASYDL, SYS1, Index 811.
- To change Tone Code by countries, the configuration command "set country_code" should be set to "0".

<Conditions Specific to UMGi/RNIP System>

Note the following when the Telephony Server type is UMGi/RNIP.

1. Because PHI-BRI function is installed in each Local Processor (LP) of the node, Internal PHI-BRI needs to be registered for each node that uses it.
2. Communication between PHI-BRI function and MG-SIP is performed via TCP/IP communication port of LP.
3. Virtual Trunk and Internal PHI-BRI must be assigned within the same node. Virtual Trunk cannot use Internal PHI-BRIs assigned to the other nodes.
4. Up to 460 speech channels, including other ISDN channels, can be used simultaneously per CCN/RN.
5. In additional dial-in via MG-SIP incoming, the virtual registers are seized in the range of the node whose speech channels have been registered. (A virtual register across nodes cannot be seized.) Unless the virtual registers cannot be seized in this way, the registers (Originating Register/Destination Register/Additional Dial-in PB Receiver) will be seized in the range of the node whose speech channels have been registered. If any register in the appropriate nodes cannot be seized in both ways, originating will receive busy.

Programming

Note: Be sure to assign the data in the following order.

STEP 1: **ASYD** - Assign System Data.
SYS1,
Index 64,
Bit 0 = 0/1 (μ-law/A-law for TSW) **Note 1**
Bit 7 = 1 (holding tone sending)
Index 186, Bit 6 = 1 (ISDN)
Index 187=00
Index 220, Bit 6=0 (ISDN is enabled)
Index 474 (Guard Timer for Originating Call)
Bit 0~3: MTC (Miscellaneous Timer Counter)
Bit 4~6: TC (Timer Class)
1: 64 ms period

Note 1: Note the following conditions when this system data is changed.
- This system data determines the compression method for the entire system. When this data is changed, the μ-law/A-law settings that are notified to other MG equipment are changed as well.
- You can specify either μ-law or A-law as the information which the Telephony Server informs MG-SIP128/96 and MG-SIP16 [SCA-16SIPMG(US)].
  For more information, refer to the configuration command for MG-SIP128/96 and MG-SIP16 [SCA-16SIPMG(US)] in Peripheral Equipment Description (IP Devices).

STEP 2: **ASYDL** - Assign the following:
SYS1, Index 810, Bit 2 = 0/1 [Internal Holding Tone (IP terminals)/External Holding Tone (EMA and PLO)] **Note 2**
SYS1, Index 880, Bit 1 = 1 (Internal PHI-PRI/BRI is available)
SYS1, Index 1144, Bit 6 = 1 (Codec Change function is in service) **Note 3**, **Note 4**

Note 2: If External music (2) is selected, refer to Music On Hold [M-7].

Note 3: This system data is available as follows:

MG-SIP16 [SCA-16SIPMG(US)]	SP-3988 MGSIP (16) PROG-G Issue 3 or later
MG-SIP96	SP-3987 MGSIP PROG-B Issue 3 or later
MG-SIP128 [MG-128SIPMGL-A]	Any version
MG-SIP128 [MG-128SIPMGG]/ MG-SIP128 [MG-128SIPMGG-B]	
Software-based MG-SIP	

Note 4: This system data is required to enable Codec Change function and G.711 Fax renegotiation in addition to a configuration command setting with "set codec_change" in the MG-SIP side. For more information, refer to the configuration command for MG-SIP128/96 and MG-SIP16 [SCA-16SIPMG(US)] in Peripheral Equipment Description (IP Devices) and for Software-based MG-SIP in Software Model Peripheral Equipment Description.

STEP 3: **ALOCL** - Assign ToS settings as follows:

Note: When a change has been made, it is necessary to initialize the Internal PHI-BRI module with the ADPM command.

TYPE: Select "Network Address".
IP ADDRESS: Specify the IP address (assign 0.0.0.0 when LOC-ID=0).
MASK BIT: Specify the Mask Bit (assign 32 when LOC-ID=0).
LOC-ID = 0
Setting Type of Signaling Packet Service: Specify the Quality of Service (IP PRECEDENCE/ DIFFSERV).

- When IP PRECEDENCE is selected
Specify its priority [0-7] and the type (LOW DELAY/THROUGH PUT/RELIABILITY/MIN COST).

- When DIFFSERV is selected
Specify its priority [0-63].

Note: LOC-ID=0 is essential to be assigned, however, assign other network's LOC-ID according to the configuration.

Note: One of the following is required to allow the changes to Setting Type of Signaling Packet Service data for already-operating equipment (such as PH and IP station) to take effect.

- Initialize the Internal PHI-BRI (Initialize the Internal PHI-BRI module with the ADPM command) or system
- Re-register the IP station

STEP 4: **ADTM** - Assign port numbers if necessary [default value = 64000].

Note: When you change the default port number with this command, you need to initialize the Internal PHI-BRI module with the ADPM command.

STEP 5: **ADPM** - Initialize the MGC (Internal PHI-BRI).
PH TYPE = Internal PHI (BRI)
Menu = Initialize

STEP 6: **ALIDL** - Specify the LENS (five digits) of Internal PHI-BRI.
PH LENS: Specify an even-numbered Group Number. The LENS can be allocated on either a Virtual PIR or a Physical PIR.
PH TYPE = Internal PHI
REGMAX: Specify the registration capacity (1-6144).

Note: When the total number of channels using PHI-BRI exceeds 256, increase the value of REGMAX.

STEP 7: **ARTD/ARTDN** - Specify the route class data on Virtual Speech Channel as follows. It is not necessary to register attributes for the Virtual Signaling Channel.

CDN2 (ONSG) = 2 (PB, 60 milliseconds Interruption, CCIS, or ISDN)
CDN4 (INSG) = 2 (PB, 60 milliseconds Interruption, or CCIS)
CDN5 (TF) = 3 [Bothway Trunk (BWT)]
CDN6 (TCL) = 1 (DDD Line)
CDN7 (L/T) = 1 (Trunk)
CDN8 (RLP) = 2 [First Party Release (either station or trunk side)]
CDN15 (LSG) = 12 [Speech Line (for CCIS or B-Channel of ISDN)]
CDN28 (ANS) = 1 (Answer signal is provided)
CDN30 (PAD) = 7 [PAD OFF(0db)]
CDN31 (OGRL) = 1 (224 milliseconds)
CDN32 (ICRL) = 1 (224 milliseconds)
CDN34 (GUARD) = 1 (512 milliseconds)
CDN45 (A/D) = 0 (Analog)
CDN49 (TRKS) = 0 (Select from the trunk which becomes idle first)
CDN50 (DPLY) = 1 (Caller ID display is enabled **Note 5**)
CDN60 (TC/EC) = 1 (Echo Canceller)
CDN65 (INT) = 10 [Q-Sig, (ETS 300 172)/IS-11572]
CDN66 (DC) = 1-15 (DID Addressing number)
Assign the CDN111(ADVPRA)=1 (ISDN PRI Failure Routing Service) to dummy route.

Note: To change the data for existing route (RT), initialize or reboot the trunk device such as MG-SIP.

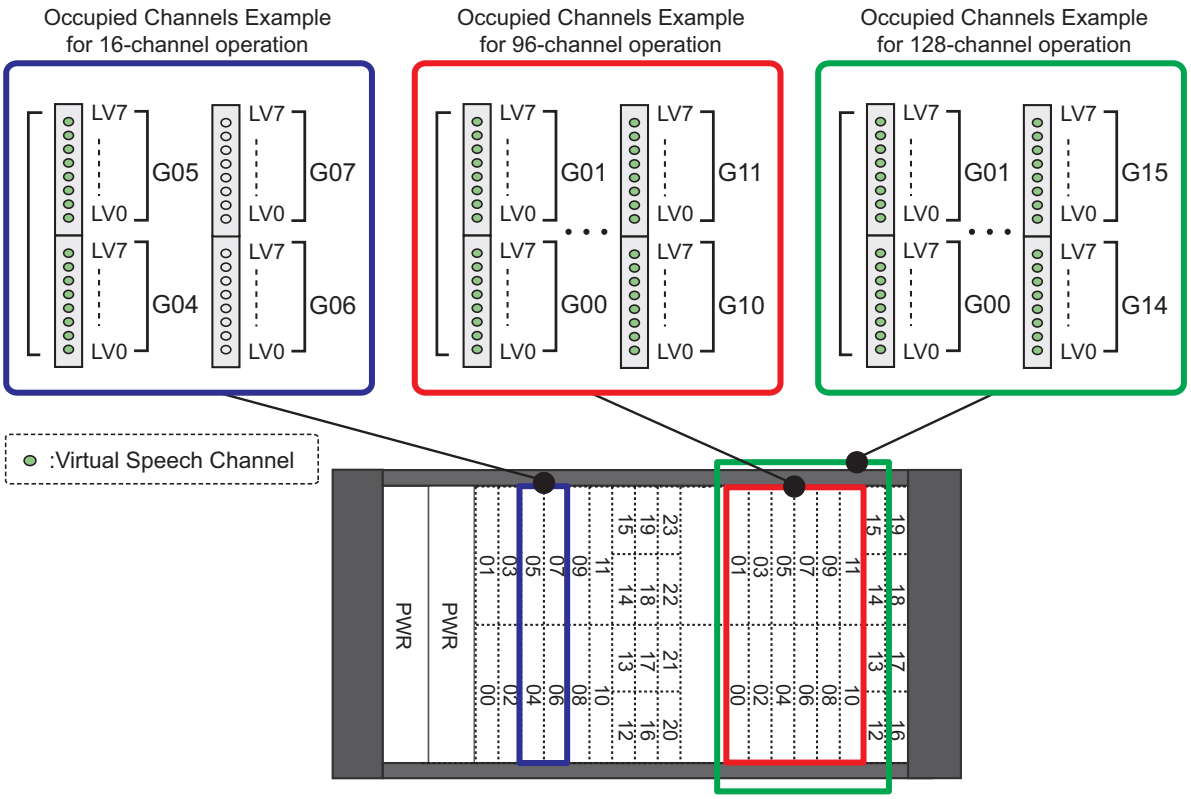
Note 5: Up to 16 digits of a Caller ID can be shown on the display.

STEP 8: **ARTI** - Specify the application data for trunk.
CDN47: INTD = 2 (SIP)
CDN68: VIR = 2 (MG connection)
CDN77: RA_RT = 0/1 (Virtual Register is used/Register card is used)
Assign the following parameters for FAX use:
CDN71: FXD = 1 (IP FAX data is in service)
CDN72: FXJS (Jitter Buffer Size for IP FAX [1-30] × 10 ms) ex. Enter “15” for 150 ms.
CDN73: FXPT = 1 (Payload Type for IP FAX is G.711)
CDN74: FXPS (Payload Size for IP FAX [1-4] × 10 ms) ex. Enter “2” for 20 ms.

Note: To change the data for existing route (RT), initialize or reboot the trunk device such as MG-SIP.

STEP 9: **ATRK** - Specify trunk data on Virtual Speech Channel and Virtual Signaling Channel.
RT: Specify Route Number [1-255]
TK: Specify Trunk Number [1-255]
LENS: Specify Accommodated Location (Six digits)

Example: In the case of 128/96/16-channel operation using MG-SIP



Data Setting Example for 16-channel operation

Virtual Speech Channel

RT (1-255)	TK (1-255)	LENS			
		MG 00-07	U 0-3	G 00-23	LV 0-7
1 0	0 1	0 2	0	0 4	0
⋮	⋮	⋮	⋮	⋮	⋮
1 0	1 6	0 2	0	0 5	7

Data Setting Example for 96-channel operation

RT (1-255)	TK (1-255)	LENS			
		MG 00-07	U 0-3	G 00-23	LV 0-7
2 0	0 1	0 2	1	0 0	0
⋮	⋮	⋮	⋮	⋮	⋮
2 0	9 6	0 2	1	1 1	7

Data Setting Example for 128-channel operation

Virtual Speech Channel

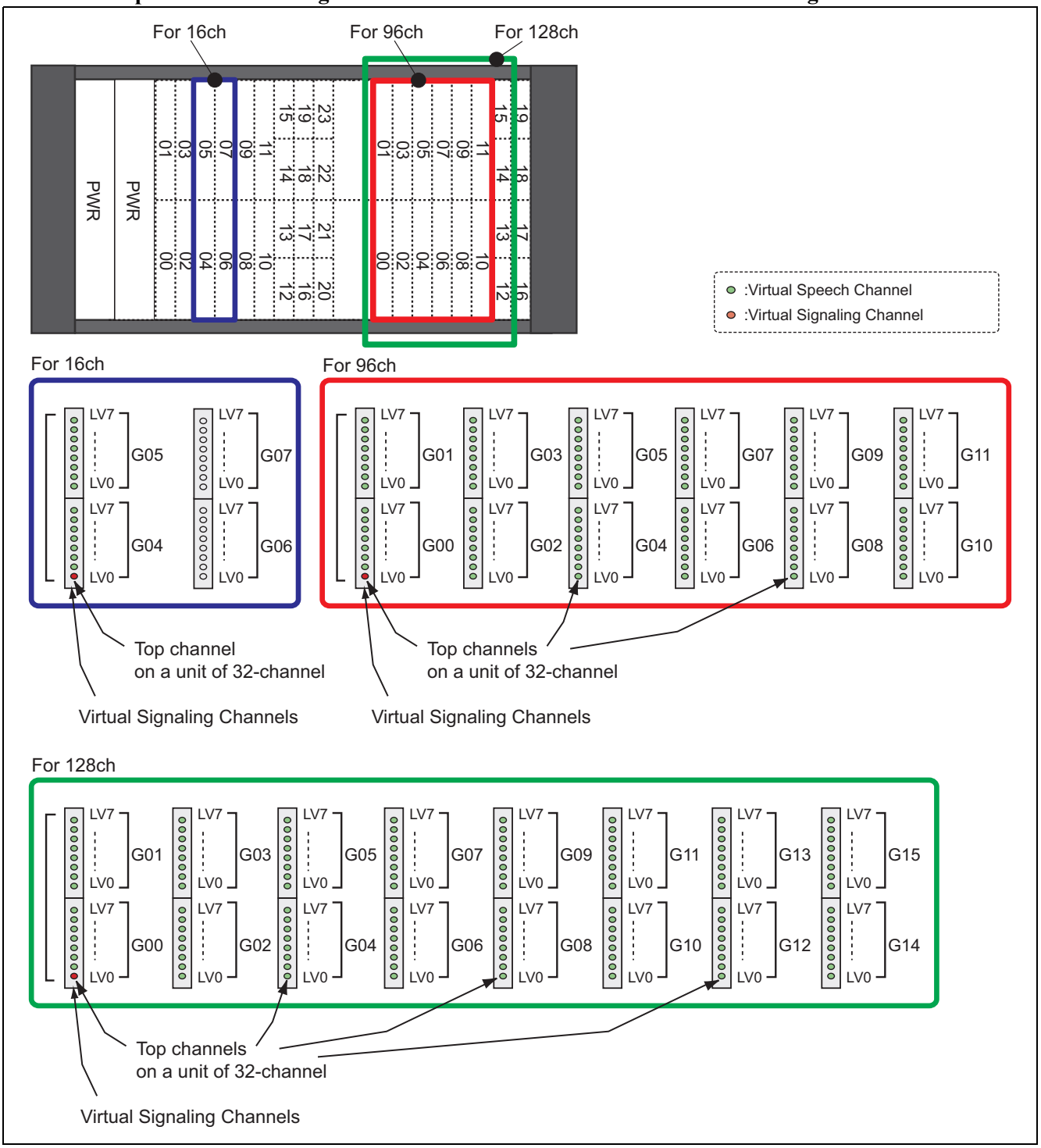
RT (1-255)	TK (1-255)	LENS			
		MG 00-07	U 0-3	G 00-23	LV 0-7
2 0	0 1	0 2	1	0 0	0
⋮	⋮	⋮	⋮	⋮	⋮
2 0	1 2 8	0 2	1	1 5	7

Note: The conditions when this command is assigned are as follows.
- Be sure to register the virtual communication channel per four groups.
- Other than MG-SIP virtual communication channel cannot be registered for HW (4 group) that is registered one or more MG-SIP virtual communication channel.
- The virtual communication channel is assigned by using the AMGIL command.
- Do not assign the LENS that belongs to Even-numbered module group, Unit=0 and Group=0 to a trunk, because not supported by MG-SIP.

STEP 10: **ADPC** - Register Point Code on Virtual Speech Channel and Virtual Signaling Channel.
RT: Specify a Route Number [1-255]
PC: Specify unused Point Code, not Point Code used in SS7 [1-16383]

STEP 11: **ACSC** - Assign the top number of 32-channel LENS.
CSCG: Specify the CSC Group Number
GROUP: Specify the CIC Group Number
CCH: Specify the lead LENS of a unit.

Example: When bundling 128/96/16-channel with a unit of 32-channel using MG-SIP



For 16ch

CSCG	CICG	CCH		
		MG	U	G
130	1	02	0	04
	2	02	0	04
	3	02	0	04
	4	02	0	04
	5	02	0	04
	6	02	0	04
	7	02	0	04

For 96ch

CSCG	CICG	CCH		
		MG	U	G
130	1	02	1	00
	2	02	1	04
	3	02	1	08
	4	02	1	00
	5	02	1	00
	6	02	1	00
	7	02	1	00

For 128ch

CSCG	CICG	CCH		
		MG	U	G
130	1	02	1	00
	2	02	1	04
	3	02	1	08
	4	02	1	12
	5	02	1	00
	6	02	1	00
	7	02	1	00

STEP 12: **ACIC1** - Specify the accommodated location of Virtual Control Channel on Point Code as follows:
PC: Enter Point Code specified by ADPC
CSCG: Enter CSC group specified by ACSC

STEP 13: **ACIC2** - Specify the accommodated location of Virtual Control Channel on Point Code as follows:
PC: Enter Point Code specified by ADPC
CIC: Enter CSC numbers in sequential number
LENS: Enter accommodated locations of Speech Channel and B-Channel

STEP 14: **MBTK** - Cancel Make Busy of Virtual Speech Channel.
RT: Enter Route Number
TK: Enter Trunk Number
MB = 0 (Make idle)

STEP 15: **AMGIL** - Specify information regarding to MG-SIP as follows:
MG-ID: Enter MAC Address (6 bytes)
FUNCTION = Setting
KIND = SIP-MG
LINE = 0 (fixed)
CH = 0 (fixed)
LENS: Enter Basic Accommodated Location (the first number of HW)
Retry = 0
Setting Type of Signaling Packet Service: Specify QoS settings of Signaling that is used when Internal PHI-BRI connects MG-SIP.
Usable CH Number: Enter the number of channels to be used per MG-SIP (1-128).
Service of Authentication Exists: Select the check box if registration is required per number.
MG Group: Enter MG-SIP Group Number 1-15 to make an alternate route for redundancy configuration. **Note 6**
MG Number: Enter MG-SIP Number 1-15 belonging to the MG-SIP Group. **Note 6**

Note: Whether to use Service of Authentication is determined by the carrier's selected services. For more information about this feature, refer to Multiple Number Service (MG-SIP) [M-113].

Note: For MG-SIP16/MG-SIP16 [MG-16SIPMGA/SCA-16SIPMGA/SCA-16SIPMG(US)], 1-16 channels can be assigned.

Note 6: This parameter only appears if "Service of Authentication exists" check box is selected.

STEP 16: **AMGVL** - Register the voice control data when connecting to MG-SIP. If the settings are not specified by this command, the values assigned by the AIVCL command will be set automatically.
A RT: Specify Physical Route Number (MG-SIP).
TYPE: Select "Route Specified" or "Location ID Specified".
B LOC-ID: Displayed when TYPE = Location ID Specified. Specify Destination Location ID (Destination MG-SIP). **LOC-ID=0 must be set when IPPAD card is mounted.*
B RT: Displayed when TYPE = Route Specified is selected. Specify Physical Route Number (MG-SIP) in the case of tandem connection between MGs.
Setting Type of Voice Packet Service: Specify QoS settings of Voice Speech
TYPE: Displayed when the check box of "Setting of Payload" is ON. Specify Payload type. **G.711 and 20ms must be set for public SIP network.*
SIZE: Displayed when the check box of "Setting of Payload" is ON. Specify Payload size.

STEP 17: **ACNAL** - Assign the Calling Number Pattern (CNP) on an Access Code basis. **When this command is used, assign ARTD command CDN66 to DC=15.*

- Settings for a Call Originated to SIP Network
  Specify Calling Number Pattern against route for Call-ID Notification and Access Code.

  IC/OG = OG (Outgoing Call)
  RT: Enter External Route Number to be sent Call-ID Notification per Access Code.
  ACC: Enter Access Code.
  CNP: Enter Calling Number Pattern.

  Note: To modify CNP set by the ACNAL command, use the ACND command; ACNDN command cannot be used.

- Settings for a Call Arrived from SIP Network
  Specify Calling Number Pattern against route for Call-ID Notification and Access Code.

  IC/OG = IC (Incoming Call)
  RT: Enter External Route Number to be arrived.
  ACC: Enter Access Code.
  DC: Dialed Number Confirmation

STEP 18: **ACND** - Assign the data for number change to CNP has been specified by the ACNAL command
CNP: Calling Number Pattern
SKIP: Number of Skip Digits
ADD: Number of Digits of the Additional Number
DC: Additional Digit Information

STEP 19: **ACDD** - Change Digit Code according to DID Addressing.
DAY/NIGHT: Day/Night Mode
RT: Enter the B-Channel Route Number.
DC: Enter Number of digits for this DC must match the data in DC of the ARTD command.
CDC: Enter Change Digit Code [Maximum 24 digits]
AD: Enter Additional Dialing
XFR: Enable/Disable Transfer Service

STEP 20: After the registration has been completed, reset MG-SIP (ON→OFF→ON).

When using Name Display service equivalent to CALLING NAME IDENTIFICATION PRESENTATION (CNIP)/CONNECTED NAME IDENTIFICATION PRESENTATION (CONP) [C-153], the following data setting is required. To send Calling Party Information, the data setting on the ANDD command is required in addition to the following setting.

STEP 1: **ASYD** - Assign System Data.
SYS1,
Index 31=04
Index 78, Bit 3: 0/1(Station Number with Name Display on Desktop terminal is out of service/in service.)
Index 78, Bit 5=1 (Name Display Service is in service.)
Index 79, Bit 5: 0/1 (Name Display Enhance Service is out of service/in service.)
*When Index 78, Bit 3=1 and Bit 5=1:
0 = Called party's number
1 = Dialed number
Index 79, Bit 6=0 (OAI/ACD in service.)
Index 241, Bit 1: The number of digits displayed in Name Display service. 0/1 (8-digit/16-digit)
Index 390, Bit 0: Caller ID display pattern on the LCD (Top line). 0/1 (Calling Name/Calling Number has priority.)

STEP 2: **ARTD** - Assign route class data for the MG-SIP route.
CDN 6 (TCL) = 4 (Tie Line)
CDN 43 (BT) = 1 (Inter-office Busy Service is in service.)
CDN 50 (DPLY) = 1 (Caller ID display is enabled. **Note 7**)
CDN 98 (CI) = 0 (ISDN transmitting information is out of service.)
CDN 112 (IND) = 1 (Inter-office Name Display is in service.)

Note: To change the data for existing route (RT), initialize or reboot the trunk device such as MG-SIP.

Note 7: Up to 16 digits of a Caller ID can be shown on the display.

When using Route Advance service equivalent to ROUTE ADVANCE [R-6] and PRI/PRA FAILSAFE ROUTING [P-49/50], the following data setting is required.

STEP 1: **ASYD** - Assign System Data.
SYS1,
Index 65: 01-FF (RT 1-255) Assign a maximum number of routes.
Index 368, Bit 2=1 (Allows alternate routing to CCIS line when a call to ISDN Network (first line) is not available.)

STEP 2: **ARTD** - Assign CDN 111 (ADVPRA) = 1 to enable ISDN PRI Failure Routing service. This data is assigned to dummy routes.

Note: To change the data for existing route (RT), initialize or reboot the trunk device such as MG-SIP.

STEP 3: **ANPD**
1stDC: The first digit of LCR Access Code.
CI: N (Normal Service) or H (Hooking Service)
NND: 1=Necessary (minimum) number of digits for Access Code.

STEP 4: **ASPA**
ACC: LCR Access Code.
CI= "N: Normal" or "H: Hooking"
SRV= "LCR: Least Cost Routing"/"LCRS: Register Sender LCR"
RT: Dummy Route Number.

STEP 5: **AFRS**
RT: Dummy Route Number.
NPC: Number Pattern Code.
OPR: Outgoing Route Selection Number.

STEP 6: **AOPR**
TDPTN: Time of Day Change Pattern.
OPR: OPR assigned by the AFRS command.
<1st Route>
RA: 0 (route seizure order)
E: 1 (More Route Advance assignments are required)
RT: MG-SIP route number which is seized first.
<2nd Route>
RA: 1 (route seizure order)
E: 1 (More Route Advance assignments are required)
RT: MG-SIP route number which is seized second.
<Last Route>
RA: 2 (route seizure order)
E: 0=Route advance ends (Last route)
RT: MG-SIP route number which is seized last.

When using Call-ID Notification service equivalent to ISDN CLI WITH CALL FORWARDING OUTSIDE [I-50], the following data setting is required.

STEP 1: **ARTI** - Assign Trunk Application Data to MG-SIP (Virtual Speech Channel) of Outgoing Route.
CDN: 78 CLICF = 1 (The calling number generated in own node is informed to SIP network.)

Note: While the feature is activated (ARTI CDN: 78 CLICF = 1), caller ID to SIP network from station call and call forward from station-to-station call (Call Transfer - All Calls, Blind Transfer to Attendant/Station and each call forwarding feature) are same condition.

STEP 2: **ASYD** - SYS1, Index 20, Bit7
Specify the billing party when calls are transmitted by Call Forwarding - Outside features.
0/1 = The originating station of the transferred call./The station that has set Call Forwarding.

To change Inband Tone of MG-SIP128 and software-based MG-SIP, the following data setting is required.

STEP 1: **ASYDL** - Assign System Data for LDM.
SYS1, Index 811, Bit 0-7 (Tone Code by countries.)

ASYDL SYS 1 INDEX 811			
TONE CODE	**Basic Tones**	**TONE CODE**	**EMEA Tones**
00	Not used **Note 8**	C8	Germany
01	Japan (default)	C9	Italy
02	North America	CA	Netherlands
03	Australia	CC	Denmark
04	Other countries	CD	Sweden
05	Hong Kong	D3	Spain
06	Malaysia	D4	Austria
07	Singapore	D5	Belgium
08	England	D6	England [For EMEA]
09	Mexico	D7	Greece
0A	Taiwan	D8	Switzerland
0B	New Zealand	D9	South Africa
0C	Korea	Other Values	Not used **Note 8**
0D	Brazil		
0E	China		
0F	Thailand		
10	Thailand (EGAT)		
11	Sri Lanka		

Note: To apply this change, assign the configuration command “set country_code” to “0”.

Note 8: When this Tone Code is assigned to “0” or “Not used”, and the configuration command “set country_code” is assigned to “0”, the tone is set as Japan (default).

- **OG/IC CALL FEATURES**

x: Available N: Not Available C: Conditionally available

Feature Name	Availability	Remarks
Direct-In Termination (DIT) [D-7]	N	
External Music On Hold	X	IPPAD card is required.
Multiple Music-on Hold [M-99]	X	IPPAD and DAT cards are required.
Distinctive Ringing [D-10]	X	Distinctive Ringing is available for an incoming call from the SIP network.
Internal Music On Hold	X	Music on Hold can be sent from the MG-SIP16 to the SIP network.
Alternate Routing [A-16]	X	SIP network can be specified as an alternate route.
Speed Calling - Station [S-21] Speed Calling - Group [S-23]	X	Available to access the SIP network by using this feature.
Direct Outward Dialing (DOD) [D-9]	X	Available to access the SIP network by dialing an Access Code + destination telephone number.
Speed Calling - System [S-3] Consecutive Speed Calling - System [C-62]	X	Available to access the SIP network by using this feature.
Toll Restriction - 3/6-Digit [T-7]	X	Available to restrict/allow Outgoing Calls to the SIP network according to specific area.
Outgoing Trunk Queuing [O-2]	X	Available when MG-SIP is in all-busy state.
Off-Hook Queuing [O-7]	X	Available when MG-SIP is in all-busy state.
Miscellaneous Trunk Restriction [M-3]	X	Call origination to SIP network can be restricted according to the Station's route restriction class.

- **ISDN FEATURES**

x: Available N: Not Available C: Conditionally available

Feature Name	Availability	Remarks
ISDN CLI with Call Forwarding Outside [I-50]	X	Available when connecting to the SIP network. **Note:** Available in Australia only.
SID to Network-Present [S-91]	X	Available to deliver SID to the SIP network. It is necessary to confirm specifications of the SID service.
SID to Network-Privacy [S-92]	C	Available to control the transfer of SID to the SIP network. It is necessary to confirm specifications of the SID service. **Note:** Available by using the configuration command (184toprivacy).

- **OTHER FEATURES**

x: Available N: Not Available C: Conditionally available

Feature Name	Availability	Remarks
Authorization Code [A-20]	X	Available to originate a call to the SIP network after temporarily changing the Station's class by this feature.
Account Code [A-18]	X	By dialing the Access Code + Account Code when Station originates a call to the SIP network, the appropriate account code can be added to the billing information.
Call Forwarding - Intercept / Announcement [C-25]	X	Available to route an incoming call from the SIP network to the Announcement trunk. IPPAD and DAT cards are required.
Call Park [C-29]	X	Available to park an incoming call from the SIP network.
Call Park Group [C-194]	X	Available for an incoming call from the SIP network.
Call Park Group -D [C-176D]	X	Available for an incoming call from the SIP network
Announcement Service [A-15]	X	Available to route an incoming call from the SIP network to the Announcement trunk. IPPAD and DAT cards are required.
Route Restriction - Announcement [R-27]	X	If call origination to the SIP network is restricted, an originated call to the SIP network is automatically routed to the Announcement trunk.
Uniform Call Distribution (UCD) [U-1]	X	Available to distribute an incoming call from the SIP network to a UCD Group.
Overflow - UCD [O-13]	X	An incoming call from the SIP network to a UCD Group can be transferred to an Overflow UCD Group when all Stations in the first UCD Group are busy.
Call Waiting Lamp - UCD [C-76]	X	An LED indication is available when an incoming call from the SIP network is waiting in the UCD Queue.
Delay Announcement - UCD [D-31]	X	An incoming call from the SIP network to a UCD Group can be transferred to the Announcement trunk when all Stations in the UCD Group are busy.
IP Control-FAX [I-44]	X	
Video Call [V-31]	X	

- **MULTI-FUNCTION TELEPHONE FEATURES**

x: Available N: Not Available C: Conditionally available

Feature Name	Availability	Remarks
Speed Calling - One Touch -D [S-26D]	X	A call can be originated to SIP network by using the Programmable Line/Feature key.
Call Forwarding - All Calls [C-5]	X	Available for an incoming call from the SIP network.
Elapsed Time Display -D [E-3D]	X	LCD display of the time elapsed is available while an Station is connected to the SIP network.
Call Pickup - Group [C-7] Selectable Station Call Pickup [S-137]	X	Available to pick up an incoming call from the SIP network.
Call Hold [C-6]	X	Available when connecting to the SIP network.
Three-Way Calling [T-2]	X	Available when connecting to the SIP network.
Exclusive Hold -D [E-4D]	X	Available when connecting to the SIP network.
Save And Repeat -D [S-24D]	X	Available when connecting to the SIP network.
Call Transfer - All Calls [C-11]	X	Available for the Station connected with the SIP network.
Serial Call -D [S-127]	N	
Call Forwarding - Don't Answer [C-3]	X	Available for an incoming call from the SIP network.
Non-exclusive Hold -D [N-7D]	X	Available when connecting to the SIP network.
Call Forwarding - Busy Line [C-2]	X	Available for an incoming call from the SIP network.
Tone Block [T-63]	X	Available when connecting to the SIP network.

- **STATION FEATURES**

x: Available N: Not Available C: Conditionally available

Feature Name	Availability	Remarks
Station-Controlled Conference [S-6]	X	Available when connecting to the SIP network.
Call Forwarding - All Calls - Outside [C-28]	X	SIP network can be specified as a forwarding destination.
Call Forwarding - Don't Answer - Outside [C-74]	X	SIP network can be specified as a forwarding destination.
Call Forwarding - Busy Line - Outside [C-60]	X	SIP network can be specified as a forwarding destination.
Blind Transfer To Station [B-18]	X	Available when connecting to the SIP network.
Call Waiting - Terminating [C-12] Call Waiting - Answer -D [C-27D] Call Waiting - Originating [C-31]	X	Available when connecting to the SIP network
Call Pickup - Direct [C-30]	X	Available to pick up an incoming call from the SIP network by dialing a CALL PICKUP - DIRECT code.
Split Call Forwarding [S-99]	X	Available for an incoming call from the SIP network and the other trunks.
Periodic Time Indication Tone [P-11]	X	Available to provide a Tone to the Station user who has made an Outgoing Call to the SIP network.
Boss - Secretary Override -D [B-5D] Boss - Secretary - Message Waiting Lamp Control [B-8] Boss - Secretary Transfer [B-11]	X	Available when connecting to the SIP network.
Call Forwarding - All Calls - Announcement [C-69]	X	Available to route an incoming call from the SIP network to the Announcement trunk.
Last Number Called - Single - Line Station [L-6]	X	Available when connecting to the SIP network.
Remote Call Forwarding Control [R-54]	X	Available to set/cancel the forwarding features from the SIP network.
Call Forwarding Intercept/Announcement [C-25]	X	Available for an incoming call from the SIP network.

- **OPERATIONS AND MAINTENANCE FEATURES**

x: Available N: Not Available C: Conditionally available

Feature Name	Availability	Remarks
Traffic Measurement	X	Available when connecting to the SIP network.
Individual Trunk Access [I-4]	X	
Individual Trunk Access (PCPro)	X	

- **NETWORK FEATURES**

x: Available N: Not Available C: Conditionally available

Feature Name	Availability	Remarks
Least Cost Routing - 3/6-Digit [L-5]	X	Alternating Routing to the SIP network via LCR is available.
Outgoing Trunk Busy Announcements [O-21]	X	When MG-SIP16 is in all-busy state, an originated call to the SIP network is automatically routed to the Announcement trunk.
Inter-Office Off-Hook Queuing [I-11]	X	Available when a call is routed from a different trunk to the SIP network via tandem connection.
Tandem Switching of TIE Trunks - 2/4-Wire [T-1]	X	Tandem connection from a different trunk to the SIP network is available.
LCR-Special Line Warning Tone	X	A Warning Tone can be sent to the Station user who has made an outgoing call to the SIP network.
Hotline - Outside [H-9]	X	Available to access the SIP network by using Hotline service.

- **IP SERVICE FEATURES**

x: Available N: Not Available C: Conditionally available

Feature Name	Availability	Remarks
Peer to Peer Bandwidth Control [P-75]	X	Available when connecting with the SIP network.
ToS Control	X	ToS value for RTP packets can be assigned.
Logged Out IP Station- Call Destination [L-53]	X	Available for an incoming call from the SIP network

- **ADDITIONAL FEATURES**

x: Available N: Not Available C: Conditionally available

Feature Name	Availability	Remarks
Alternating Routing when SIP network error occurs	X	Available when the SIP server sends back no response.
Alternating Routing when MG-SIP is all-busy	X	Alternate routing to another trunk is available.
SIP network fault management	X	Available to restrict a call origination to the SIP network when the system detects MG-SIP failure or LAN cable disconnection.
Tandem connection with COT included	X	It is necessary to confirm specifications of the SIP network side.

M-113 Multiple Number Service (MG-SIP)

General Description

This feature allows MG-SIP to make Register Data (SIP-URI, User ID, Telephone Number, IP Address of Equipment and Password) per Telephone Number against SIP server. Thereby, MG-SIP is able to support multiple number service even if the SIP network (IP network) that MG-SIP connected does not support Dial-in/Hunting Features.

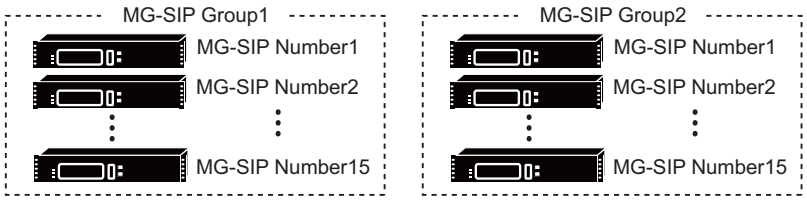
Operating Procedure

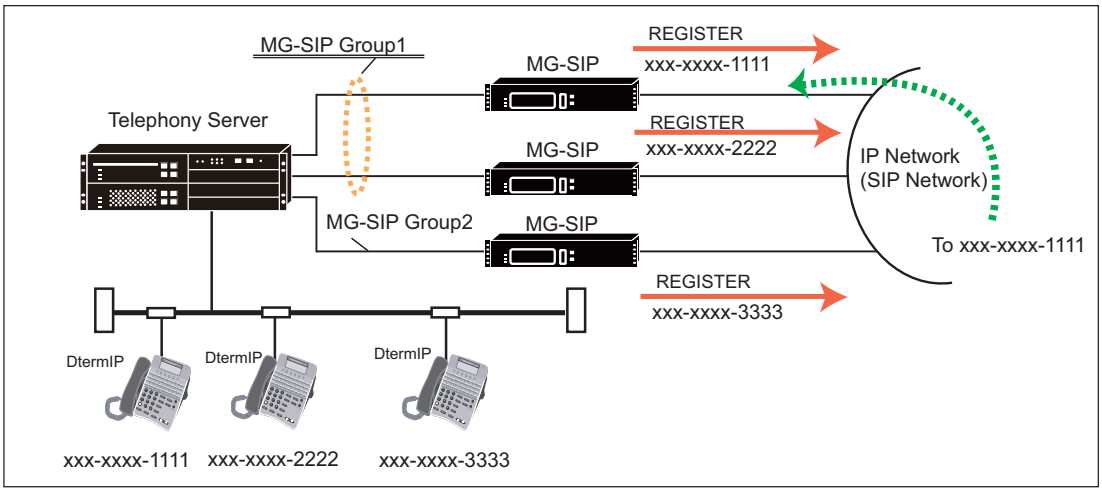
None.

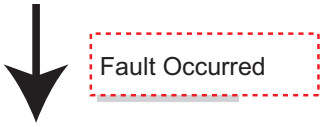
Service Conditions

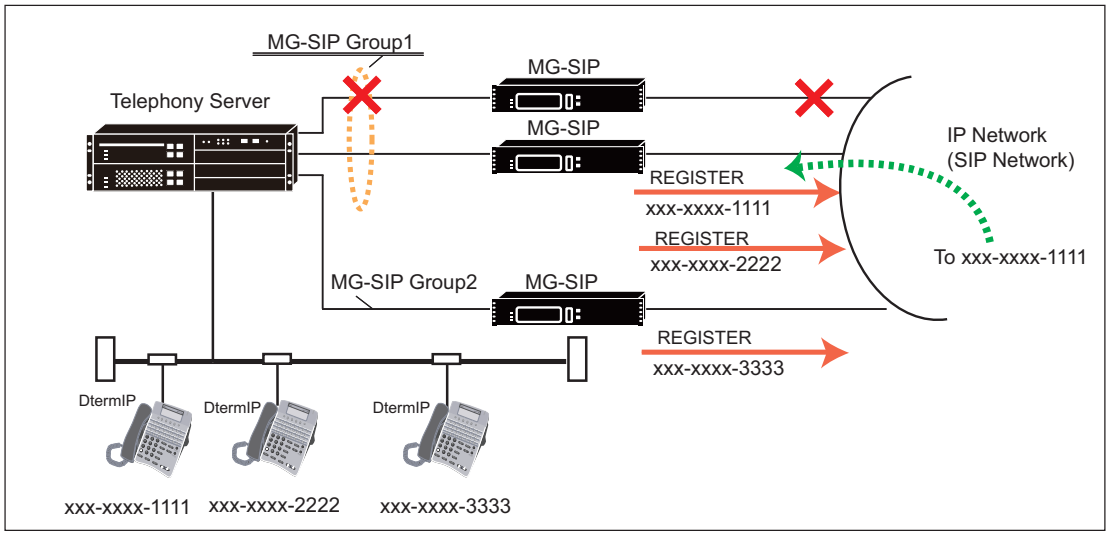
1. To perform this feature, the following firmware or later are required for each MG-SIP.
   - Software-based MG-SIP: SP-4078, SP-4085
   - MG-SIP128 [MG-128SIPMGL-A]: SP-4089
   - MG-SIP128 [MG-128SIPMGG/MG-128SIPMGG-B]: SP-4051, SP-4058
   - MG-SIP96: SP-3987
   - MG-SIP16: SP-3843
   - MG-SIP16 [SCA-16SIPMGA/MG-16SIPMGA]: SP-3905
   - MG-SIP16 [SCA-16SIPMG(US)]: SP-3988
2. Up to 4000 can be specified per system for Register Data.

3. When making Redundancy Configuration, the system needs MG-SIP Group Number to make an alternative route. The following describes MG-SIP Group.









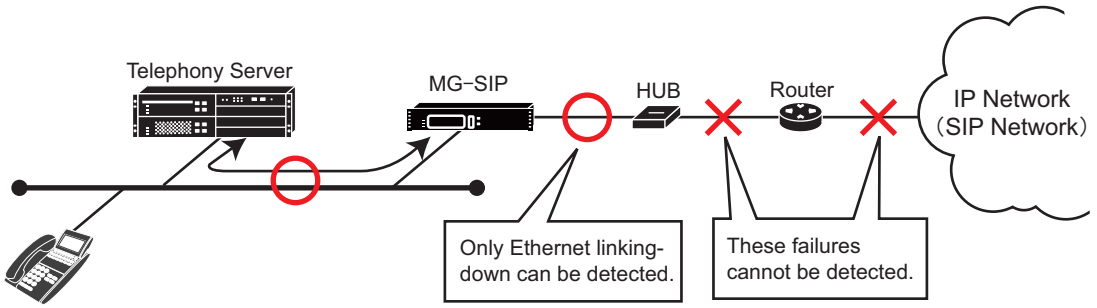
- The following MG-SIP can be assigned up to 15 per MG-SIP Group, and up to 15 Groups per system.

Software-based MG-SIP	Any version
MG-SIP128 [MG-128SIPMGL-A]	
MG-SIP128 [MG-128SIPMGG/ MG-128SIPMGG-B]	
MG-SIP16 [SCA-16SIPMGA/MG-16SIP-MGA]	SP-3905 MGSIP (16) PROG-A Issue 6 or later
MG-SIP16 [SCA-16SIPMG(US)]	SP-3988 MGSIP (16) PROG-G Issue 4 or later

- Up to 225 MG-SIPs (15 devices×15 groups) can be registered per system.
- To accommodate MG-SIPs, be careful for the total number of channels for each MG-SIP to be registered to one LP not to exceed the maximum number of simultaneous speech channels per LP (460 channels including ISDN channels).
  For example, when only MG-SIP16s are accommodated and all channels are registered, up to 112 MG-SIPs can be registered (up to 28 MG-SIPs can be registered to each LP).
  When the number of ISDN channels including MG-SIP registered to an LP exceeds the maximum simultaneous speech channels, simultaneous communication cannot be executed at all MG-SIPs registered to the LP.
- MG-SIP128/96/16 and software-based MG-SIP can coexist in the same group.
- If a network fault occurs, MG-SIP located on an alternative route sends Register Data.
- MG-SIPs belonging to a single MG-SIP Group must be accommodated in the same LP.

4. Up to 1,000 Register Data can be assigned per MG-SIP128/96 and software-based MG-SIP. Up to 100 Register Data per MG-SIP16.
5. When a failure occurs, a calls is re-routed to an alternative route within the same MG-SIP Group. Alternate routing is not available for outgoing calls made between MG-SIPs across different MG-SIP Groups.
6. When an MG-SIP Group has only a single MG-SIP number, alternate routing function is not available.
7. MG-SIP sends Register (performs authentication) when in the following conditions:
   - When MG-SIP is initialized
   - When an alternative route is taken
   - When a fault is recovered
   - When the user data are updated (add/delete)

8. Alternate route is available when a failure (Ethernet linking-down only) is detected in the SIP network or a communication failure occurs between the Telephony Server and MG-SIP.



Note: It takes 90 to 250 seconds to detect a communication failure between the Telephony Server and MG-SIP.

Note: In software-based MG-SIP, Ether link down in the SIP network is not detected.

9. Alternate routing of Register Data takes the following time.
The following table shows the time it takes to perform alternate routing of Register Data when the number of Register Data per MG-SIP is assigned to 16 and the Register sending cycle of MG-SIP is assigned to 1 second (default).

Note: The values in this table are given for reference only. The actual time depends on the carrier and the network that you use.

Alternate routing is available when:	The Number of Failed Devices in the same MG-SIP Group (Registering 15 MG-SIPs to an MG-SIP Group)			
	1 device	4 devices	8 devices	14 devices
Communication failure occurs between the Telephony Server and MG-SIP	About 160 sec.	About 200 sec.	About 240 sec.	About 420 sec.
MG-SIP requests alternate routing for Register data	About 35 sec.	About 100 sec.	About 200 sec.	About 300 sec.

10. During alternate routing of Register Data, if the destination MG-SIP is deleted by AMGIL command, the Register Data of both source and destination MG-SIPs become unavailable.
11. The Register Data associated with the MG-SIP that has never been online is not the target of alternate routing.
12. In an FCCS network, up to 15 MG-SIP Groups can be assigned to each node.
13. In an FCCS network, alternate routing is not available for outgoing calls made between MG-SIPs across different nodes (across different FPCs).

14. Alternate routing by Register is the feature that provides the alternate routing of an incoming call to MG-SIP. For alternate routing of an originating call, be sure to assign PRI/PRA FAILSAFE ROUTING [P-49/50]. For details, refer to PRI/PRA FAILSAFE ROUTING [P-49/50].
15. When the Telephony Server side terminal releases during a conversation after a temporary communication failure between the Telephony Server and MG-SIP, MG-SIP may not send disconnection signals to the SIP network.

Programming

Before setting this feature, it is required to set basic settings for MG-SIP. For more information about the MG-SIP settings, refer to MEDIA GATEWAY SIP [M-111].

STEP 1: **AMGIL** - Assign information regarding to MG-SIP as follows:
MG-ID: Enter MAC Address (six bytes)
FUNCTION = Setting
KIND = SIP-MG
LINE = 0 (fixed)
CH = 0 (fixed)
LENS: Enter Accommodated Location of Virtual Signaling Channel.
Retry = 0
Setting Type of Signaling Packet Service: Assign QoS settings of Signaling that is used when PHI connects MG-SIP.
Usable CH Number: Assign Channel Number 1-254. **Note 1**
Service of Authentication Exists: Select the box if a registration is required per number, and then assign the following:
- MG Group: Enter MG-SIP Group Number 1-15 to make an alternative route for redundancy configuration. **Note 2**
- MG Number: Enter MG-SIP Number 1-15 belonging to the MG-SIP Group. **Note 2**

Note 1: The parameter range is limited by the number of usable channels of each type of MG-SIP. For an explanation, refer to the General Description section of MEDIA GATEWAY SIP [M-111].

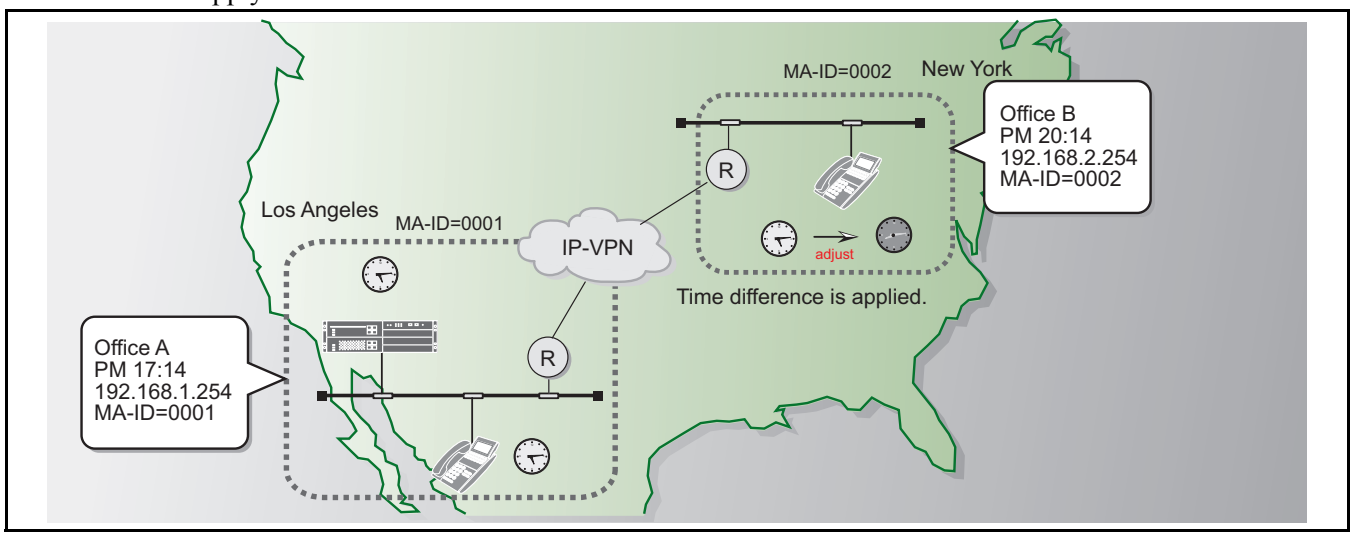
Note 2: This parameter only appears if “Service of Authentication exists” check box is selected.

STEP 2: **ASRIL** - Assign the information per Telephone Number.
MG Group: Enter MG-SIP Group Number 1-15.
MG Number: Enter MG-SIP Number 1-15 belonging to the MG-SIP Group.
CNT (START-END): Specify a range of Register Data to be read out.
Register Data
CNT: Counter to manage Register Data
Calling Num: Telephone Number to be used in the SIP network
SIP URI: SIP URI to be used in the SIP network
UserID: User ID to be authenticated in the SIP network
Password: Password to be used in the SIP network
Reg: Enable/Disable settings for MG-SIP to be authenticated in the SIP network.

M-114 Multiple Time Zone

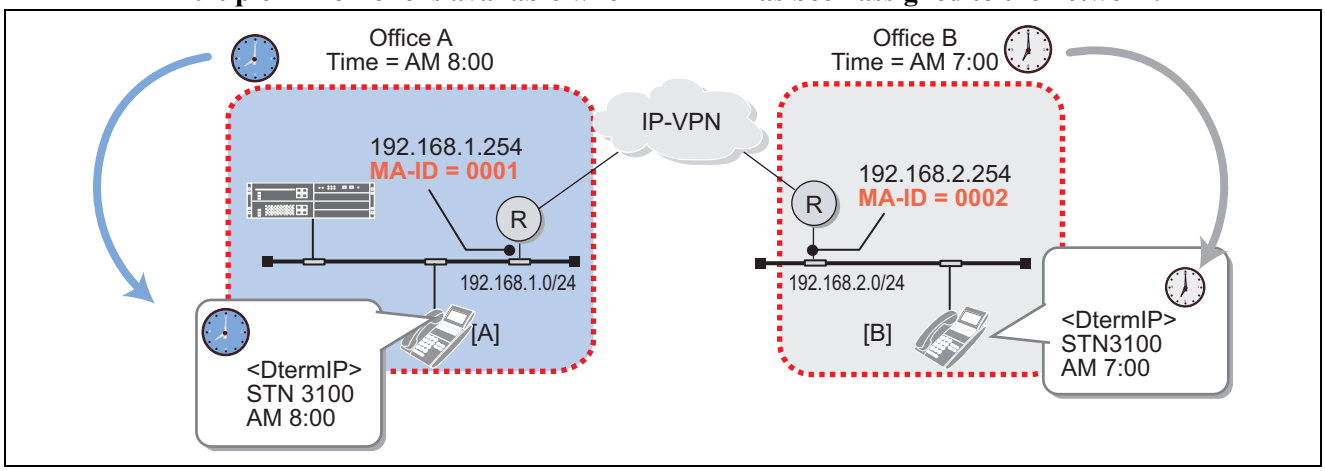
General Description

This feature allows DtermIP and DtermIP INASET users to adjust the date and time information on the LCD to the local time by using MA-ID data. When the above terminals are used over multiple time zones, the date and time information on the LCD can be automatically adjusted to the local time. If this feature is not used, the LCD of DtermIP and DtermIP INASET displays the date and time information of the Telephony Server. Note that this feature does not apply to other features that involve time information.



Note: For this feature, MA-ID must be assigned on a network basis.

Multiple Time Zone is available when MA-ID has been assigned to the network.



[A] STN 3100 logs in at Office A. The LCD of the DtermIP displays AM 8:00.

[B] STN 3100 logs out at Office A, and then STN 3100 logs in at Office B. In this instance, the LCD of the DtermIP displays AM 7:00 (Local Time of Office B).

Operating Procedure

No manual operation is required.

Service Conditions

1. This feature does not apply to other features that involves time information such as SMDR.
2. This feature applies to DtermIP and DtermIP INASET only.
3. This feature applies to time display information of the above terminals only.
4. The availability of this feature depends on ASYDL SYS1, Index 831, Bit 0.
5. This feature does not apply to SoftPhone since it uses the date and time information of the personal computer.
6. A maximum of 128 Network Areas can be used for this feature.
7. If there is any time difference in the FCCS network, assign the time difference data by using ATDF in advance and adjust the time by using ATIMN.
8. In an environment where two nodes are connected via FCCS, use AMAIL command to define the area information when one of the following cases applies:

    - When there is any time difference between two nodes
    - When only one node uses daylight saving service and the other node does not

9. When time difference data has been assigned to an MA-ID by using AMAIL/AMAIN, be sure to assign the MA-ID to the network by using AMAA.
10. If AMAA is used to assign an MA-ID that has not been assigned by AMAIL/AMAIN to the network, the MA-ID data cannot be assigned.
11. The date and time that are displayed on the LCD reflect the time difference and daylight saving time.
12. When the time difference data is changed by AMAIL/AMAIN while this service is used, the date and time information of terminals that are connected to the network of the corresponding MA-ID will also be changed.
13. MA-ID for the network must be assigned to all of the network.
14. When the Telephony Server is placed in daylight saving time mode, the date and time of all terminals will be automatically updated. However, if the Telephony Server is installed in node that does not apply daylight saving time, the date and time of terminals installed in a remote office will not be automatically updated even if the remote office applies daylight saving time.
15. The date and time will be updated at the time of terminal registration.
16. The date and time will be automatically updated once a day.
17. The date and time will be updated when those of the Telephony Server is changed by ATIM.
18. If daylight saving service has not been assigned at the Telephony Server, the time difference of daylight saving time cannot be assigned.
19. If the time zones use different daylight saving time, this feature cannot be used.

Note: Daylight saving time can be set only one per system (ATIM). This feature sets the time difference based on the date and time of the Telephony Server. For this reason, the correct date and time cannot be displayed on the LCD if the time zones use different daylight saving time.

Programming

Note: It is assumed that date and time information have already been assigned by ATIM.

STEP 1: **ASYDL** - System Data 1, Index 831, Bit 0: Multiple Time Zone. Assign 1 (In Service).

STEP 2: **AMAIL** - Assign the following parameters.
MA-ID: Enter the Message Area ID [1-9999]
AREA NAME: Enter the Area Name (Maximum 16 letters)
MGC Time Zone: Select a time zone for the MGC from the list box.
MA-ID Time Zone: Select a time zone for the MA-ID from the list box.
Daylight Saving Time is automatically applied: Check this box when Daylight Saving Time is used (When checked, Daylight Saving Time is applied to the time zone of the user.)

STEP 3: **AMAA** - Assign the following parameters. Select Network in KIND.
TYPE: Select Individual.
FPC: Assign the Fusion Point Code.
IP ADDRESS: Assign the IP address of the terminal where MA-ID is assigned.
MASK BIT: Select Mask Bit. [1-31]
MA-ID: Enter the MA-ID that has been assigned by AMAIL/AMAIN. [1-9999] **Note 1**

Note 1: MA-ID=0 is used for deleting the data.

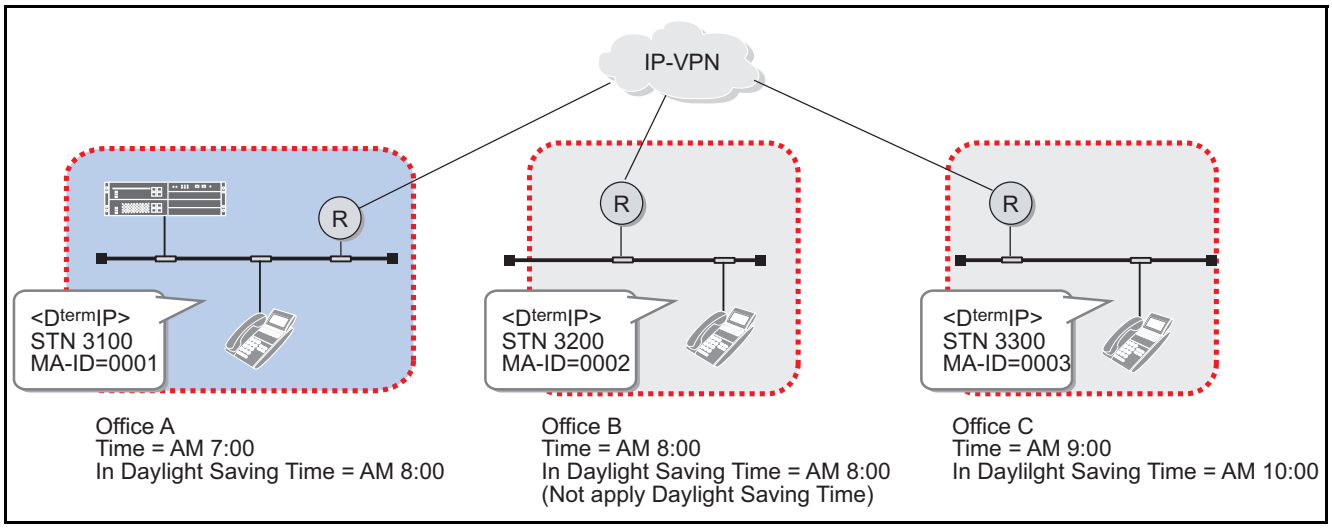
FCCS Programming

Use AMAIN command at STEP 2 in place of AMAIL command.

Reference:

The following shows a sample data assignment for this feature.

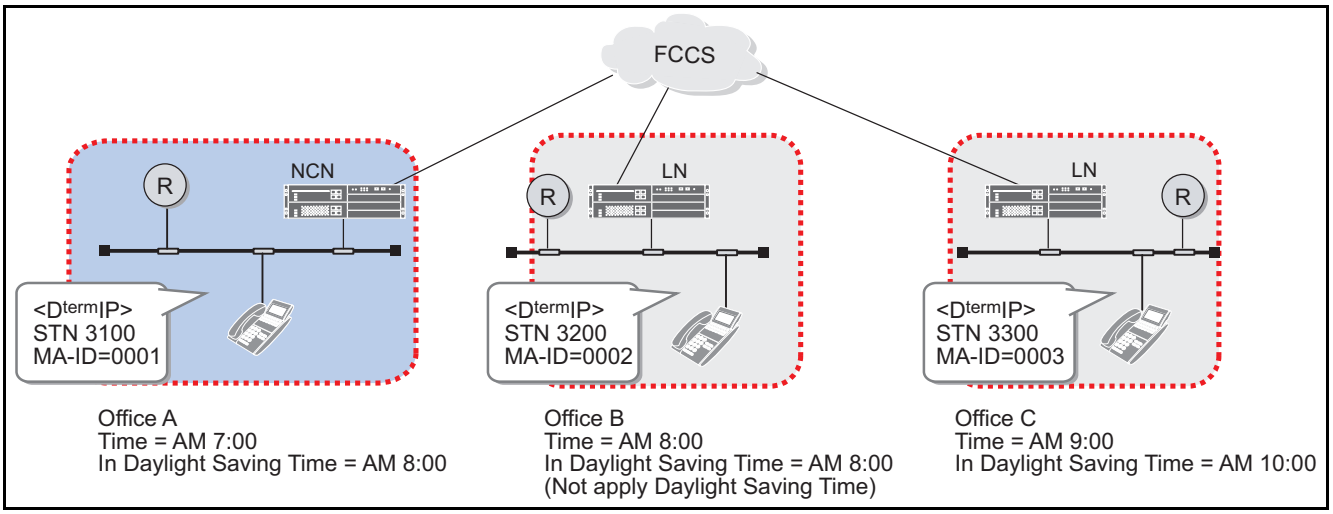
When AMAIL is used:



Time Difference Assignment Example (AMAIL)

Station Number	Time Zone of MGC	Time Zone of User	Check Box
3100	Office A	Office A	checked
3200		Office B	not checked
3300		Office C	checked

When AMAIN is used:



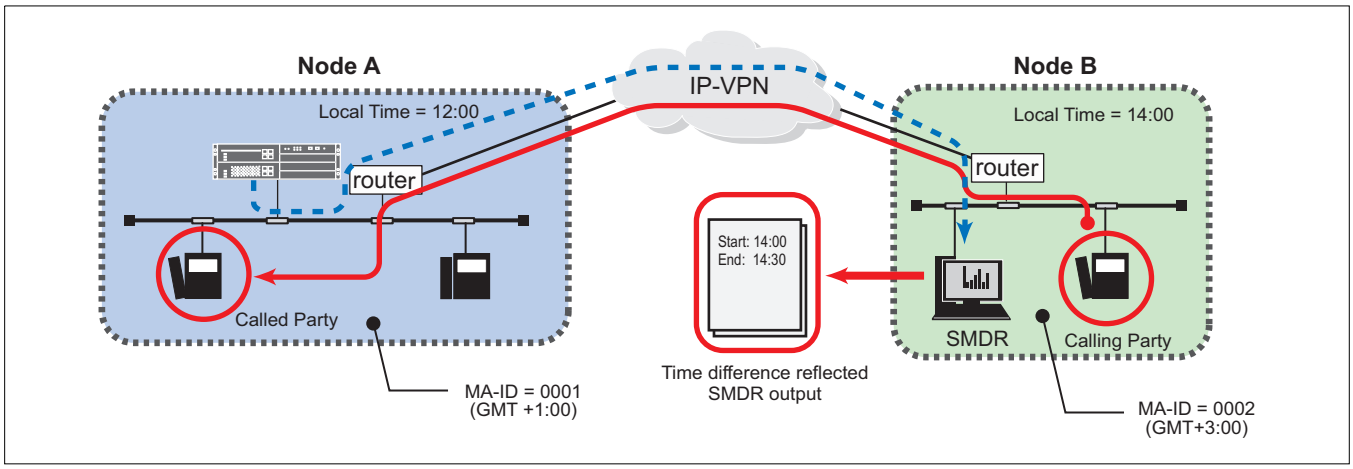
Time Difference Assignment Example (AMAIN)

Station Number	Time Zone of MGC	Time Zone of User	Check Box	Remarks
3100	Office A	Office A	checked	-
3200		Office B	not checked	When STN 3200 belongs to Office A
3200	Office C		not checked	When STN 3200 belongs to Office C
3300		Office C	checked	-

SMDR - Multiple Time Zone

General Description (for SMDR - Multiple Time Zone)

Based on [S-10] STATION MESSAGE DETAIL RECORDING (SMDR) and [S-144] SMDR MULTI AREA ID VIA IP, this feature allows the system to output SMDR information that takes into account time difference when a call is made over different time zones.



The following table shows basic formula for reflecting time difference.

Note: For more detailed information of time difference reflected SMDR output, see Reference 2 (SMDR Output Sample for SMDR - Multiple Time Zone) at the end of this feature description.

Call Start Time	=	Call Start Time in Telephony Server	+	Time Difference **Note 2**
Call End Time	=	Call End Time in Telephony Server	+	Time Difference **Note 2**
Trunk Arrival/Sending Time **Note 3**, **Note 4**, **Note 5**	=	Trunk Arrival/Sending Time in Telephony Server	+	Time Difference **Note 2**

Note 2: Time difference between the Telephony Server and the MA-ID to which the SMDR subject telephone user belongs.

Note 3: Trunk Arrival/Sending Time is included in SMDR information of SMDR for Incoming/Outgoing Trunk Calls.

Note 4: Trunk Arrival Time is available in EMEA from FP95-111 V1 and available in other countries from FP95-113 V3.

Note 5: Trunk Sending Time is available from FP95-113 V3.

Operating Procedure (for SMDR - Multiple Time Zone)

No manual operation is required.

Service Conditions (for SMDR - Multiple Time Zone)

1. The availability of this feature depends on ASYDL, System Data 1, Index 589 Bit 5 (SMDR - Multiple Time Zone).
2. This feature is available when both [S-144] SMDR MULTI AREA ID VIA IP and [M-114] MULTIPLE TIME ZONE are in service.
3. This feature is available for IP-based terminals of the Telephony Server.
4. For Soft Phone/Soft Phone (SIP), time information is displayed using that of the PC. For this reason, the PC time should be correctly set. If incorrect time is set for the PC, discrepancy may arise between the time of the Soft Phone and output SMDR information.
5. When this feature is used for a legacy terminal, the time information of the Telephony Server applies.
6. This feature is applied for Call Start Time, Call End Time, and Trunk Arrival/Sending Time of SMDR information.
7. When this feature is in service, Time Difference Reflected Call Time and MA-ID will be output in the following format.
   - KK message (Trunk Outgoing Billing: IMX format)
   - KL message (Trunk Incoming Billing: IMX format)
   - KM message (Station-to-Station Billing: IMX format)
8. When this feature is in service and SMDR is output in the following format, MA-ID will not be output. However, Time Difference Reflected Call Time will be output for Call Start Time and Call End Time:
   - KA/KH message (Trunk Outgoing Billing: standard/expanded format)
   - KE/KI message (Trunk Incoming Billing: standard/expanded format)
   - KB/KJ message (Station-to-Station Billing: standard/expanded format)
9. The following shows MA-ID Area information according to the call type:
   - Trunk Outgoing Billing: Time Difference of calling party's MA-ID Area information
   - Trunk Incoming Billing: Time Difference of called party's MA-ID Area information
   - Station-to-Station Billing: Time Difference of calling party's MA-ID Area information
10. This feature is available when billing subject party is one of the following:
    - Station
    - Attendant Console
    - Trunk
11. When this feature is in service, it applies to terminals or trunks that are connected to the Telephony Server.
12. MA-ID must be assigned for all networks that have Stations and/or Trunks.
13. If any 'MA-ID not assigned network' exists, the MA-ID of Stations and/or Trunks that belong to the network will be notified to SMDR as MA-ID = 0.
14. If MA-ID = 0, this feature becomes invalid. The system assumes that there is no time difference between the network and the Telephony Server.

15. If Time Difference data has not been assigned to MA-ID, the system assumes that there is no time difference between the network and the Telephony Server.
16. If the Time Difference data of MA-ID Area Information is modified during a call in progress, Time Difference Reflected Call Time will be edited by using the Time Difference data of MA-ID Area Information that had been set when the call is started.
17. When split billing is in service, MA-ID of the calling party of each SMDR record will be used.
18. When total billing is in service and billing subject party is set to transferring party by System Data, the MA-ID of the transferring party will be used.
19. When Three-Way Calling, 8-Party Conference (Meet Me/Progressive) is in service, MA-ID of the calling party of each SMDR record will be used.
20. When a trunk becomes billing subject party because of a direct tandem connection, MA-ID of the trunk will be used for outputting Time Difference Reflected Call Time.
21. When this service is used and SMDR is sent to Center Node of Centralized Billing - CCIS, the SMDR information that is sent from the Center Node does not contain the MA-ID. However, Time Difference Reflected Call Time will be output for Call Start Time and Call End Time.
22. When this service is used and the following message output is attempted, the MA-ID will not be output. However, Time Difference Reflected Call Time will be output as the time information.

    - CS Report
    - ETF (Call Event Transfer Facility) service in OAI

23. A maximum of 128 areas can be used for the network areas of Multiple Time Zone. The maximum numbers of areas is also 128 areas for this feature.
24. When RS-232C is used for SMDR interface, output format is shown below. Note that when using multiple ports, the format can be specified on a port basis by using System Data.

    - KA/KH/KK message (Trunk Outgoing Billing: standard/expanded/IMX format)
    - KE/KI/KL message (Trunk Incoming Billing: standard/expanded/IMX format)
    - KB/KJ/KM message (Station-to-Station Billing: standard/expanded/IMX format)

25. When LAN is used for SMDR interface, output format is shown below. Note that when using LAN interface, SMDR is output in IMX format only even if multiple ports are used for SMDR.

    - KK message (Trunk Outgoing Billing: IMX format)
    - KL message (Trunk Incoming Billing: IMX format)
    - KM message (Station-to-Station Billing: IMX format)

26. SMDR call base table will not be changed for this feature.

Interactions (for SMDR - Multiple Time Zone)

<[S-144] SMDR MULTI AREA ID VIA IP related Interactions>

1. This feature is available for MA-IDs that are assigned to network. Not available for MA-ID assigned to LENS.
2. A single MA-ID can be set to multiple networks. However multiple MA-IDs cannot be set to a single network.
3. When a call is made using a subline, the MA-ID of the phone will be output
4. When a call is made using a virtual line, the MA-ID of the phone will be output.
5. This feature can be valid irrespective of the availability of [S-144] SMDR MULTI AREA ID VIA IP. When [S-144] SMDR MULTI AREA ID VIA IP is in service, SMDR will be output only when a Station-to-Station call is made over different MA-ID networks.
6. The maximum number of terminals that can belong to a single MA-ID is the same as that of terminals that can be assigned within the network of the MA-ID.

<[M-144] MULTIPLE TIME ZONE related Interactions>

1. [M-144] MULTIPLE TIME ZONE is available for DtermIP, DtermIP(SIP), DtermIP INASET. Note that SMDR-Multiple Time Zone is available for equipment to which Multi Time Zone is not available (i.e. Analog MC).
2. When using daylight saving time, first enable it by ATIM command. In addition, set daylight saving time when assigning MA-ID.

<FCCS related Interactions>

1. When using this feature in FCCS network, be sure to assign MA-ID to all Telephony Servers and networks. If there is any area to which MA-ID has not been assigned, condition of (e) A call between different MA-IDs (when 'MA-ID not assigned network' exists) will be applied.
2. This feature is also valid in a Centralizing Billing - FCCS environment.

<Abandoned Call related Interactions>

1. If an incoming/outgoing trunk call is abandoned, when the call destination can be specified, the MA-ID of the destination will be used to output Time Difference Reflected Call Time.
2. In the above case, when the call destination can not be specified, the system assumes MA-ID=0, and the time of the Telephony Server will be used.
3. If the above case occurs when UCD or Station Hunting is used, the output will be as follows:
   When UCD/Station Hunting has been set between terminals which have different MA-ID, the MA-ID of the called station will be used to output Time Difference Reflected Call Time.

<Daylight Saving Time (DST) related Interactions>

1. When a call is established between the Telephony Server (without DST) and a remote node (with DST), Daylight Saving Time cannot be used. In this instance, the Telephony Server must be installed in a location where DST is used.
2. To use DST, the term (START and END) should be specified by ATIM command.
3. Only one DST can be set per system. For this reason, when the switchover to DST is not performed automatically, manual operation is required.

4. If DST is applied during a call in progress, the SMDR will be output using the DST-not-applied-time. Note that when split billing is in service, if a new SMDR record is created by some telephone operation such as call transfer, DST is applied for the new SMDR output.

<PS related Interactions>

1. If roaming occurs during the call in progress, the time difference of MA-ID of the area where the call started is used.

<SR-MGC related Interactions>

The following shows interactions when IP terminals attempt to log in to the SR-MGC in case of network failure.

1. MA-ID basis time difference represents time difference between the area and the Telephony Server. Any SMDR record that is made while the terminal logs in to the SR-MGC, Time Difference Reflected Call Time is used.
2. Between the Telephony Server and SR-MGC, clock synchronization is performed when the SR-MGC is activated or health check is attempted. When clock information has been changed at the Telephony Server, synchronization will not take place if the network is disconnected before attempting health check.

Programming (for SMDR - Multiple Time Zone)

STEP 1: Assign SMDR data referring to [S-10] SYSTEM MESSAGE DETAIL RECORDING (SMDR).

Note: When assigning SMDR data, ASYDL System Data 1, Index 589, Bit 5 (SMDR - Multiple Time Zone) must be set to 1 (= in service).

STEP 2: Assign MA-ID data referring to [S-144] SMDR MULTI AREA ID VIA IP.

Note: Skip AMAIL/AMAIN, AMAA data assignment. These should be assigned when assigning MULTIPLE TIME ZONE data.

STEP 3: Assign [M-114] MULTIPLE TIME ZONE data referring to Programming for Multiple Time Zone. **Note 6**, **Note 7**

Note 6: To delete the data, perform the above procedure in reverse order.

Note 7: To disable SMDR - Multiple Time Zone feature only, set ASYDL System Data 1, Index 589, Bit 5 (SMDR - Multiple Time Zone) to 0 (= out of service). Do not delete other data.

Reference 2 (SMDR Output Sample for SMDR - Multiple Time Zone)

This reference provides SMDR output samples under the following headings:

(a) A call within a same MA-ID area

(b) A call between different MA-IDs (Station-to-Station Call)

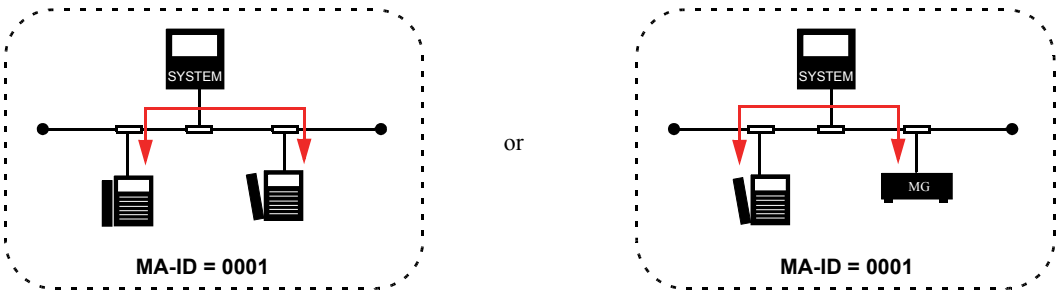
(c) A call between different MA-IDs (Trunk Call)

(d) A call between different MA-IDs (when 'time difference not assigned MA-ID' exists)

(e) A call between different MA-IDs (when 'MA-ID not assigned network' exists)

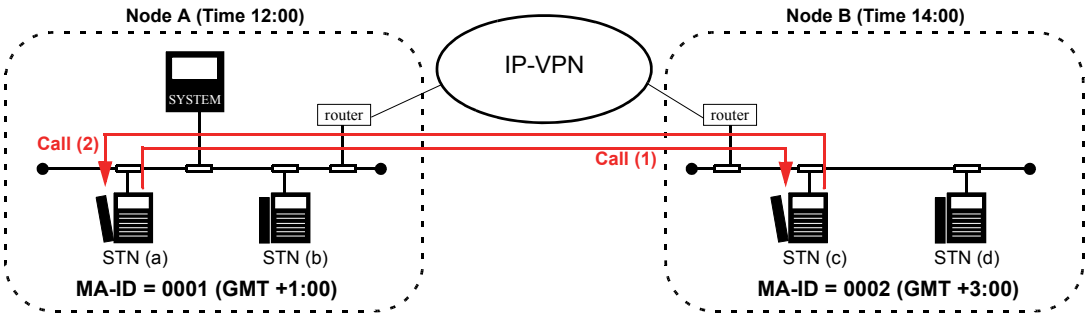
(f) A call within a same MA-ID area

In this case, no time adjustment will take place on the ground that the call is made within a same area.



(g) A call between different MA-IDs (Station-to-Station Call)

- Type 1



In this case, the SMDR output is as follows for a Station-to-Station Call between Station (a) and Station (c) during 12:00 - 12:30 (Node A Local Time).

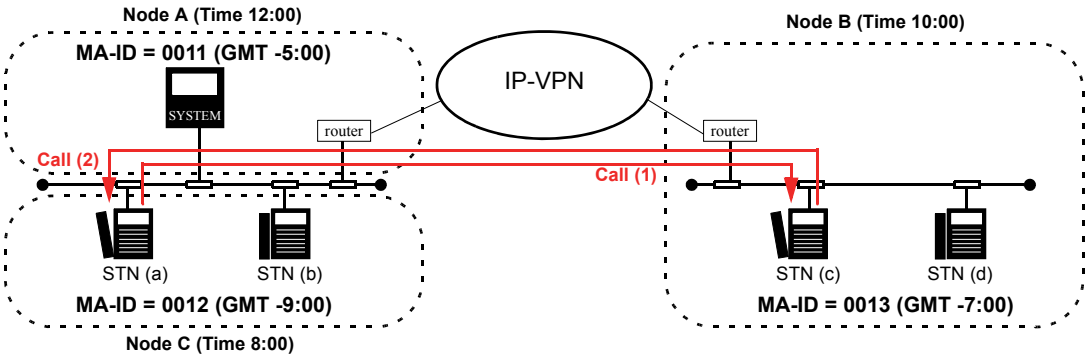
SMDR output for Call (1): Call from STN (a) to STN (c)

```
Call Start Time:
  Call End Time:
```

SMDR output for Call (2): Call from STN (c) to STN (a)

```
Call Start Time:
  Call End Time:
```

- Type 2



In this case, the SMDR output is as follows for a Station-to-Station Call between Station (a) and Station (c) during 12:00 - 12:30 (Node A Local Time).

SMDR output for Call (1): Call from STN (a) to STN (c)

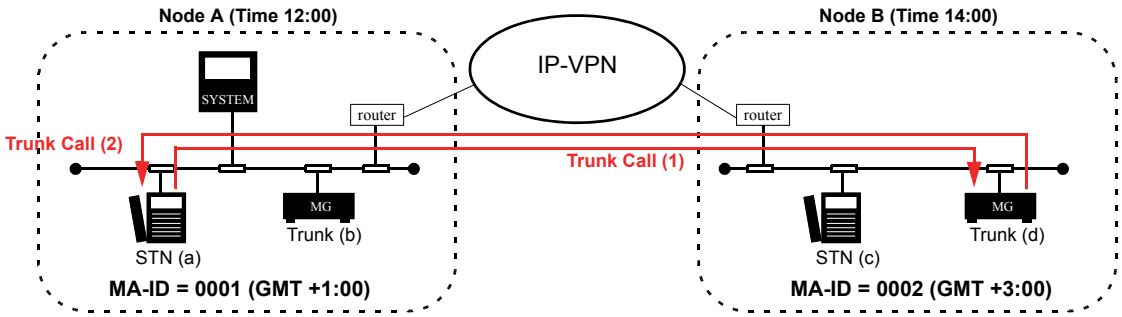
```
Call Start Time:
  Call End Time:
```

SMDR output for Call (2): Call from STN (c) to STN (a)

```
Call Start Time:
  Call End Time:
```

(h) A call between different MA-IDs (Trunk Call)

- Type 1



In this case, the SMDR output is as follows for a Trunk Call between Station (a) and Trunk (d) during 12:00 - 12:30 (Node A Local Time).

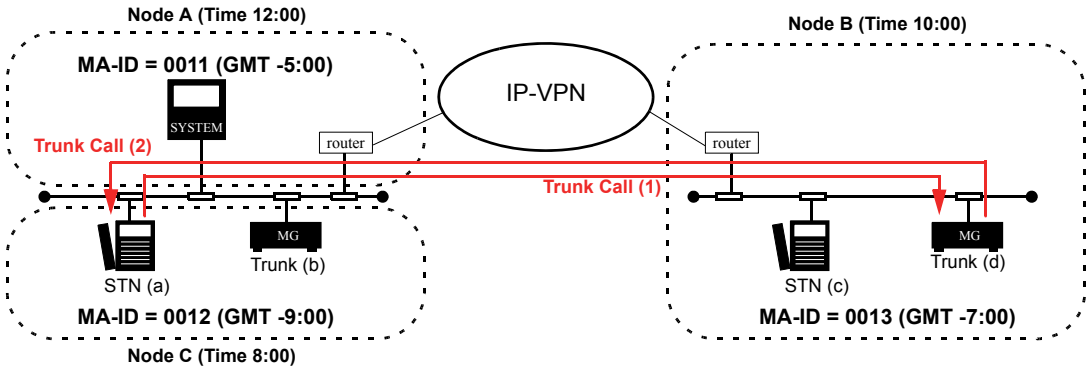
SMDR output for Trunk Call (1): Call from STN (a) to Trunk (d)

```
    Call Start Time:
      Call End Time:
```

SMDR output for Call (2): Call from Trunk (d) to STN (a)

```
Trunk Arrival Time:
    Call Start Time:
      Call End Time:
```

- Type 2



In this case, the SMDR output is as follows for a Station-to-Station Call between Station (a) and Trunk (d) during 12:00 - 12:30 (Node A Local Time).

SMDR output for Trunk Call (1): Call from STN a to Trunk d

```
    Call Start Time:
      Call End Time:
```

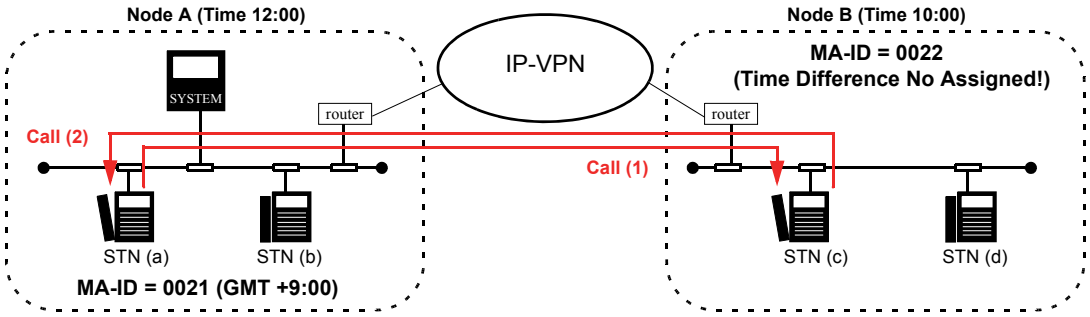
SMDR output for Trunk Call (2): Call from Trunk d to STN a

```
Trunk Arrival Time:
    Call Start Time:
```

SMDR output for Trunk Call (2): Call from Trunk d to STN a

```
      Call End Time:
```

(i) A call between different MA-IDs (when 'time difference not assigned MA-ID' exists)



In this case, time difference data has not been assigned for MA-ID = 0022. Under this condition, the SMDR output is as follows for a Station-to-Station Call between STN (a) and STN (c) during 12:00 - 12:30 (Node A Local Time). **Note 8**

SMDR output for Call (1): Call from STN (a) to STN (c)

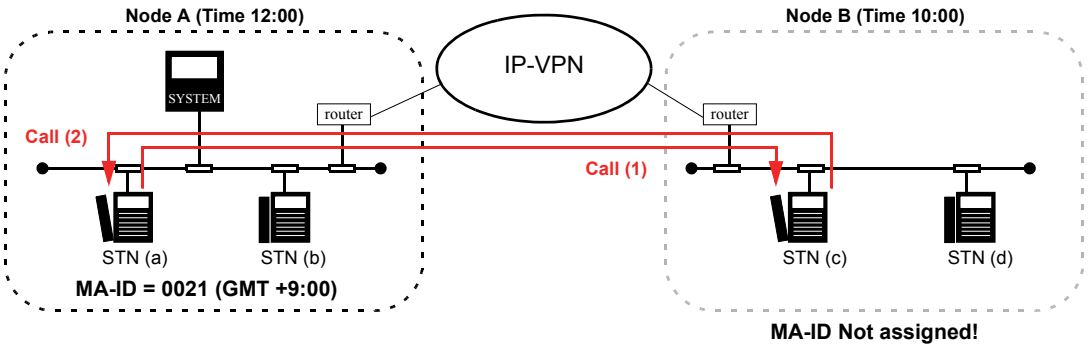
```
    Call Start Time:
      Call End Time:
```

SMDR output for Call (2): Call from STN (c) to STN (a)

```
    Call Start Time:
      Call End Time:
```

Note 8: If any network exists whose time difference data has not been assigned, the system assumes that there is no time difference between the Telephony Server and the network.

(j) A call between different MA-IDs (when 'MA-ID not assigned network' exists)



In this case, there is a network to which MA-ID has not been assigned. Under this condition, the SMDR output is as follows for a Station-to-Station Call between STN (a) and STN (c) during 12:00 - 12:30 (Node A Local Time). **Note 9**

SMDR output for Call (1): Call from STN (a) to STN (c)

```
Call Start Time:
  Call End Time:
```

SMDR output for Call (2): Call from STN (c) to STN (a)

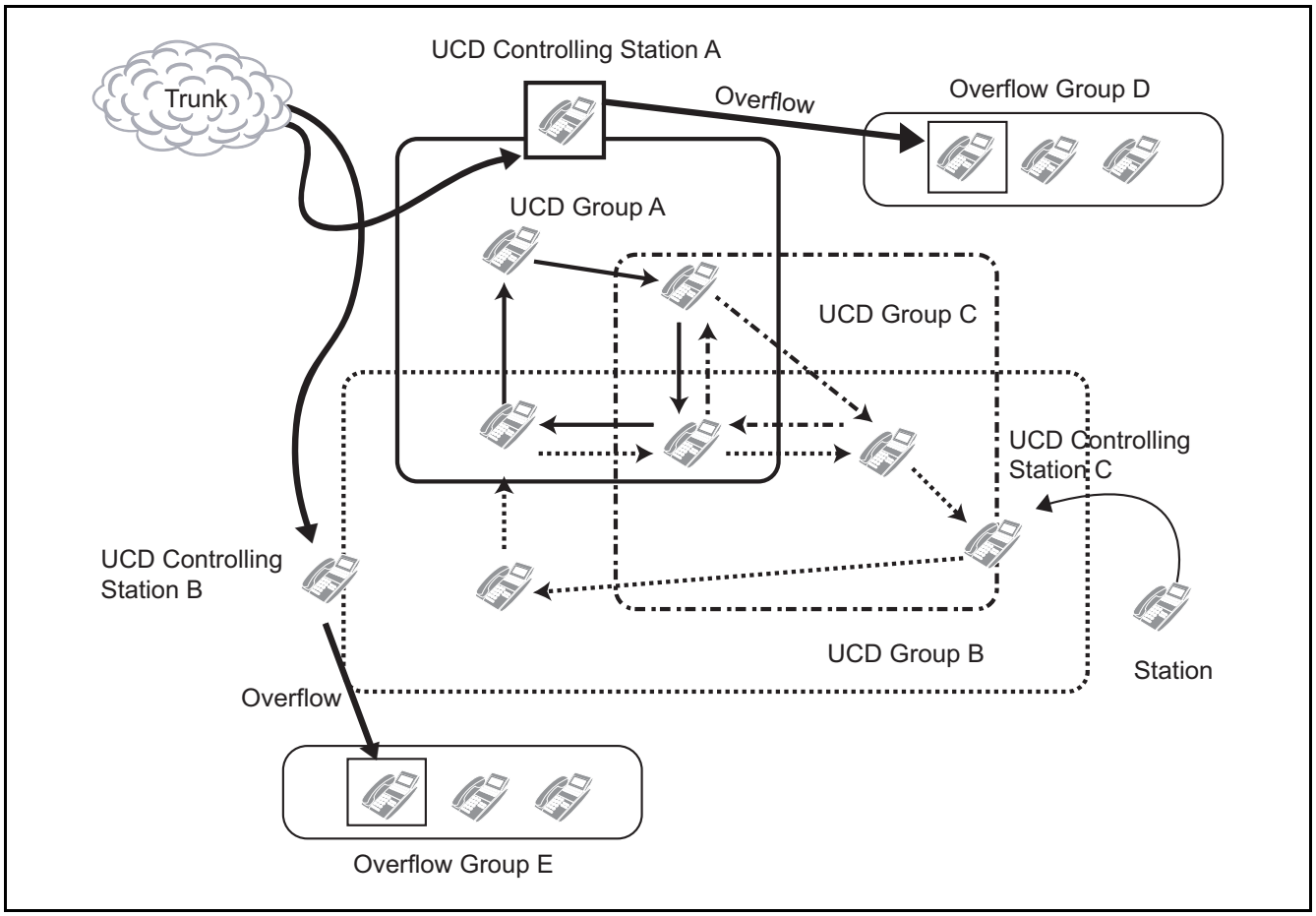
```
Call Start Time:
  Call End Time:
```

Note 9: If any network exists whose MA-ID data has not been assigned, the system assumes that there is no time difference between the Telephony Server and the network.

M-115 Multi Group Station - UCD

General Description

This feature allows a UCD Controlling Station/Non-controlling Station to belong to multiple UCD groups.



Operating Procedure

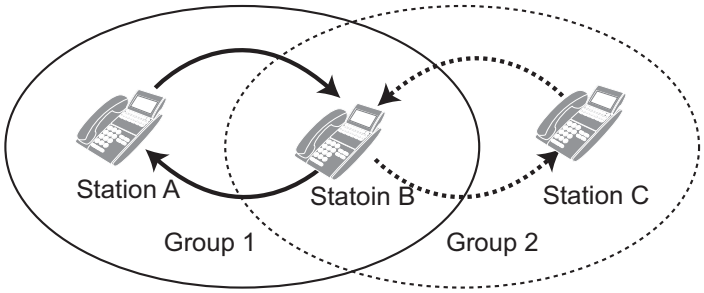
The operating procedure is the same as for UNIFORM CALL DISTRIBUTION (UCD) [U-1]. Refer to UNIFORM CALL DISTRIBUTION (UCD) [U-1].

Service Conditions

Service conditions of this feature is same as those of UNIFORM CALL DISTRIBUTION (UCD) [U-1]. The following shows additional service conditions specific to this feature.

1. This feature is available only in FCCS network. A Controlling Station/Non-controlling Station can belong to multiple UCD groups across a maximum of four nodes.
2. When UCD Group Station belongs to multiple groups, Call Waiting lamp supports only one Controlling Station. For Non-Controlling Station, Call Waiting lamp supports Stations in the UCD Group registered for the last time.

3. Traffic Measurement (UCD Peg Count) is available as usual.
4. Incoming calls are distributed among the member Stations in each UCD group. Load sharing for the Station which belongs to multiple UCD groups (Station B in the following figure) is not performed.



Station B may have twice the traffic volume of Station A and Station C.

5. Station that has already been assigned as a member of Station Hunting group (Circular/Terminal/Priority/ Priority with Switchback) cannot be assigned as a member of UCD group, and vice versa.
6. When a Station Number has already been assigned by ASHUN command, delete the ASHUN data, and then reassign the Station Number by ASHUEN command.
7. When using Location Diversity [L-55], the Telephone Number of User Profile Data must be assigned to the station by using ASHUEN command.

Programming

STEP 1: **ASYD** - System Data 1, Index 60 Bit 3. Selection of UCD Queuing:
0/1 = UCD Queuing is provided/Not provided
System Data 1, Index 70, Bit 6. Designation of Announcement Trunk used for Delay Announcement-UCD service: 0/1 = Common/Per UCD Group

STEP 2: **ANPDN** - Assign and Number of Necessary Digits (NND) for the 1st Digit Code (1stDC) of Access Code for temporary leaving from UCD Group.

STEP 3: **ASPAN** - Assign Access Code for temporary leaving from UCD Group.
CI="N: Normal"
SRV="SSCA: Service Code Appendix"
SIDA= 50 (UCD Busy out; Entry)

CI="N: Normal"
SRV="SSCA: Service Code Appendix"
SIDA=51 (UCD Busy out; Cancel)

Note: When using a Line/Feature key:
AKYD (Dterm Key Data)
FKY: 19 (UCD Busy Out)

STEP 4: **ASHUEN** - Assign the UCD Group Hunting data.

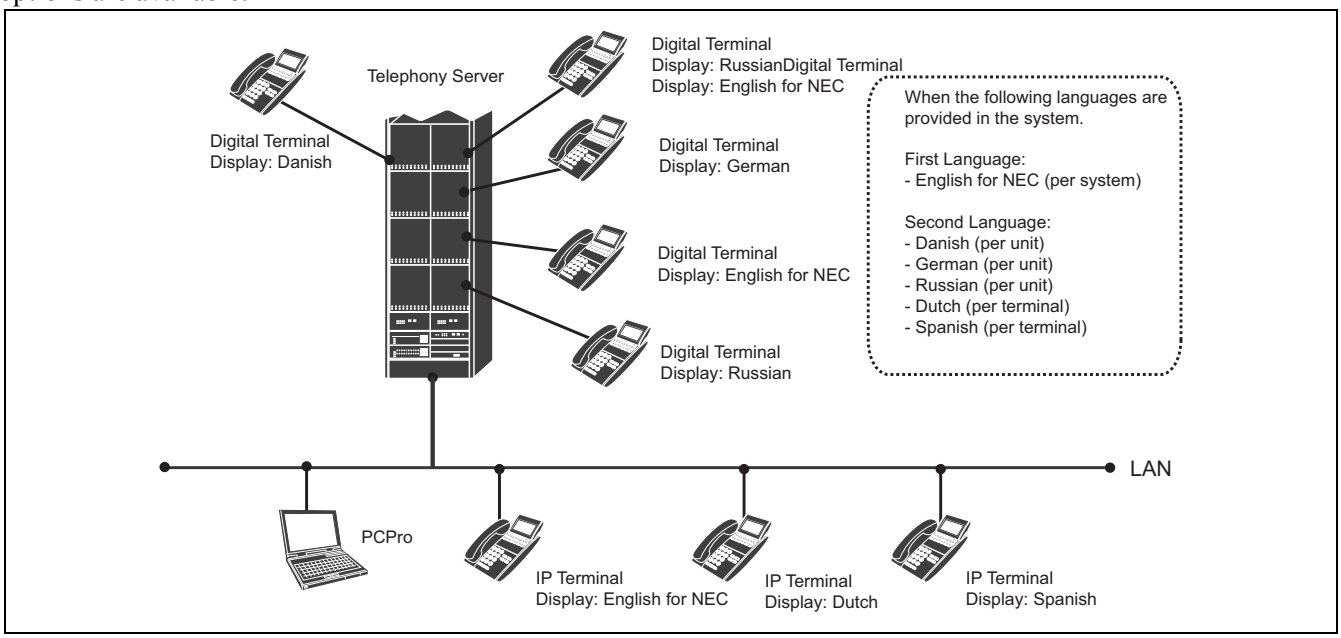
UGN: User Group Number
TELN: Assign Telephone Number of Controlling Station.
Kind of Group 0: Multi Group Station - UCD
1: Multi-Boss/Multi-Secretary Group
[Station List in Group]
TELN: Telephone Number of UCD Member Station
FPC: Fusion Point Code to which Member Station belongs is displayed.

Note: When the Station Number has already assigned by ASHUN, delete the ASHUN data once, and then assign the Station Number by ASHUEN.

M-117 Multilingual Display

General Description

This feature allows the system to provide the display in a maximum of ten different languages per system from sixteen language selections on Digital terminal/IP terminal (Proprietary Protocol) and on SIP Multiple Line terminal. For IP Station, either of the provided languages can be specified per terminal. For Digital terminal, either of First Language set per system or Second Language set per Unit can be specified.
With Extension Languages configuration of OPTIONAL LANGUAGE PACK LOADING [O-42], more language options are available.



Operating Procedure

The operation procedure for language change with Soft key/Feature key is as follows.

To change language

1. Press the LANGUAGE CHANGE Feature key (LANG). The LCD displays menu and the language in use.
2. Press "#" key to select the language. When "#" is pressed once, the LCD displays First Language.
3. Press "#" key again to display Second Language. To display next language, press "#" key again.
4. Press "1" key to set the language.

To cancel language change operation

Press "0" key during the operation. Operation stops and LCD returns to idle state.

To display default language

Press "1" key when LCD displays menu.

Note: To display menu during the operation, press "*". To display menu from the idle state, press the LANGUAGE CHANGE Feature key (LANG).

Service Conditions

1. Up to ten different languages per system can be selected from the following for the Multilingual Display.

Language	Time and Date Display	DT300 Series (DT330) Menu Display Note 1
Japanese	Available	Available
English for NEC	Available	Available
Portuguese for NEC	Available	Available
Spanish for NEC	Available	Available
Canadian French for NEC	Available	Available
German	Available	Available
Italian	Available	Available
Dutch	Available	Available
Danish	Available	Available
Swedish	Available	Available
Greek	Available in English for NEC	Available in English for NEC
English	Available	Available
Portuguese	Available in Portuguese for NEC	Available in Portuguese for NEC
Spanish	Available in Spanish for NEC	Available in Spanish for NEC
French	Available in Canadian French for NEC	Available in Canadian French for NEC
Russian	Available	Available

Note: Other languages in addition to the above are available with the use of the OPTIONAL LANGUAGE PACK LOADING [O-42]. For details, refer to the "Optional Language Pack Loading [O-42]" in this manual.

Note 1: Bluetooth Handset of Cordless Terminal (DT330 (12BT)) can display only English and Japanese.

2. 24-hour display or 12-hour display can be selected as the Digital terminal time display format by data setting (ASYDL SYS1 Index 677 Bit 1).

Note: This service is available for Digital terminals only. IP Stations can provide only 12-hour display.

3. IP Station differs in its display depending on the kind of terminal. Day of the week and month display on the Dterm Series *i* and DT300 Series can be adjusted to that of the IP Station by data setting (ASYDL SYS1 Index 677 Bit 2).
   Types for the day of the week and month display are shown below:
   - Type 1: Display for Europe
   - Type 2: Standard
4. DtermIP INASET and Soft Phone are not supported.
5. Billing information display is not supported.
6. This feature does not support OAI, ACD or Hotel displays.
7. The following are the conditions when Russian is supported.
   a.) The following Desktop terminals are required to provide the display in Russian.
   - DTR-8D-1R TEL
   - DTR-16D-1R TEL
   - DTR-32D-1R TEL
   - DT300 Series

   b.) The display in Russian can be specified per system or per Unit.
   c.) IP Stations are not supported.
8. The Following shows the conditions for language change with Soft key/Feature key on the terminal.
   a.) Language change with Soft key/Feature key is available when Second Language is assigned by ASLG command. If the terminal is assigned only First Language, language cannot be changed.
   b.) Language change is available only for the languages assigned to the target terminal.
   c.) The LANGUAGE CHANGE Feature key is available when the terminal is in idle.
   d.) Language change operation is available even if a call terminates to the operating station. When the terminating call is answered, language operation is closed.
   e.) Although a call terminates to the language change operating station, operating display does not close. When language change operation is performed on the call terminating station, the language on the display is changed in the following timing.
   - Digital terminal
     The language on the display is changed after executing MBST command or initializing the corresponding ELC card.
   - IP Station
     The language on the display is changed by terminal initialization after call is finished. When initialization is failed, initialization is performed again after next call is finished.

   f.) When keys except Soft key, Dial key, and Feature key assigned as LANGUAGE CHANGE Feature key are pressed during the language change operation, the operation stops and starts to follow operation of the pressed key. When the FEATURE key and Dial key except “1” are pressed together, operation can be continued. Display does not change though ringer changes.
   g.) Language change display closes after no operation is performed for 30 seconds.
   h.) LCD is displayed in English during the language change operation regardless of the assigned language on the terminal.
   i.) Pressed Dial key during the language change operation is indicated for a moment.
   j.) If the LANGUAGE CHANGE Feature key is pressed during language change operation, the operation closes as well as pressing CANCEL key.
   k.) When the language change operation is performed, the Office Data back up from the Telephony Server Memory to the Telephony Server Flash Card is required by MEM_HDD command before Routine Diagnosis Program Start Time.

l.) When a terminal used as a handset for SOFT PHONE WIRED-PHONE HANDSET [S-147] is assigned the non-English language display with this service, the DtermSP30 connecting the terminal is still displayed in English.

m.) The language cannot be changed while back up the Office Data.When language change operation is performed on the backing up Office Data, the language change operation is closed and the terminal displays an error message.

n.) The terminals as which the corresponding ELC card is the same cannot set the language at the same time. When two terminals set the language at the same time, only one terminal can change the language. The other terminal is closed language operation and displays an error message.

9. After the kinds of language per system are changed, reboot SP. When Digital terminal is used, initialize the corresponding ELC card (by flipping its MB key OFF→ON→OFF).

10. The following are the conditions for Multilingual Display on a menu window of DT300 Series (DT330).

a.) Execute the initialization of ELC card after data assignment (by flipping its MB key OFF→ON→OFF).

b.) When ELC card fails to download data from the Telephony Server, a system message (33-F: PKG Download Failure Notification, kind of fault: ELCN-PKG Menu download failure for DT300) is output. (**Note 2**) In that case, the card that detected the failure should be reset manually, or initialized by executing make-busy and make-busy cancel operation with MBPM command. **Note 3**

c.) When a terminal fails to download data from ELC card, a system message (33-F: PKG Download Failure Notification, kind of fault: Menu download failure between ELCN-PKG and terminal) is output. (**Note 2**) In that case, the terminal that detected the failure should be reset manually, or initialized by executing make-busy and make-busy cancel operation with MBST command. **Note 3**

Note 2: System messages can be displayed on the screen of maintenance PC by executing DFTD command. For details of System messages, refer to Chapter 3 in Operations and Maintenance Manual.

Note 3: For details of maintenance commands, refer to Operations and Maintenance Manual.

11. To activate this feature, the following firmware is required on each equipment.

Equipment	Firmware	Remarks
PA-16ELCJ-B	SP-3887 16ELCJB PROG-K Issue 1 or later	
PA-16ELCN	SP-3956 16ELCN PROG-A Issue 1 or later	
	SP-3977 16ELCN PROG-B Issue 1 or later **Note 4**	To enable multilingual menu displays of DT300 Series (DT330)
PA-16ELCN-A	SP-4013 16ELCN-A PROG-A Issue 1 or later	
	SP-4034 16ELCN-A PROG-B Issue 1 or later	To enable multilingual menu displays of Cordless Terminal (DT330 (12BT))
CH-16ELCA	Any version	

Note 4: To use the OPTIONAL LANGUAGE PACK LOADING [O-42] feature on Dterm Series i, SP-3977 16ELCN PROG-B Issue 1 or later is needed.

Interactions

1. This feature is not effective for display that registered by system data such as ANDD, ADRTL, ADRTN, ACID, and AFDD.
2. When preset dialing is operated with setting of the specified language, Bluetooth Handset of Cordless Terminal (DT330 (12BT)) can display only up to 20 digits. Thus, the following messages in brackets (21 digits or more) cannot display on the Bluetooth Handset.

Language	Displayed Message
Portuguese for NEC	Introduzir Discar Nù (mero)
Spanish for NEC	Entrar Marcado Numer (o)
Canadian French for NEC	Entre Numero Compose (r)
Spanish	Introducir Numer Mar (car)

3. When a language other than Japanese is assigned as the displayed language, Bluetooth Handset of Cordless Terminal (DT330 (12BT)) displays in English.
4. When SIP Handler Controlled SIP terminal is used, Tone Region of IP-DTG can be selected with the ASLG command.
5. When SIP Handler Controlled SIP terminal is used, Display Language cannot be selected from the Telephony Server data assignment.

Programming

To activate Multilingual Display

STEP 1: **ASYDL** - System Data 1, Index 677, Bit 1. Specify the time display format of Digital terminal whether 12-hour display or 24-hour display.
0=12-hour display
1=24-hour display

Note: This service is available for Digital terminal only. IP Station can provide only 12-hour display.

System Data 1, Index 677, Bit 2. Specify the display type for day of the week and month in order to adjust display on the Digital terminal to that on the IP Station.
0=Type 1 (Display for Europe)
1=Type 2 (Standard)

Note: This data setting is available for Dterm Series i and DT300 Series.

STEP 2: (Optional) When using OPTIONAL LANGUAGE PACK LOADING [O-42], assign the following system data, and go to the next step:
ASYDL - System Data 1, Index 679, Bit 2: (Optional Language Pack Loading)=1 (in service)
System Data 1, Index 686, Bit 7: Setting for Azerbaijani/Turkish **Note 1**
0=Out of Service
1=In Service **Note 2**

Note 1: Available from FP95-114 V4 Issue 4.0.

Note 2: Assign 1 only if Azerbaijani/Turkish is used.

STEP 3: **ASLG** - Assign the language display.

Note: Azerbaijani/Turkish requires to set Russian as First Language and Azerbaijani/Turkish as Second Language.

a.) (Optional) If Extension Languages are used, set their Language numbers.

- **[EXTENSION] tab**

  Choose the desired language files from the [Select Language] list as Extension Languages, and assign Extension Language numbers [18-27] to each of them by entering the numbers in the [LANG NO] field of the [Select Language] list.

b.) Specify the First Language used per system.

- **[FIRST] tab -** Basic Data Setting

  First Language Data field
  LANG NO: Specify the First Language used per system. For the details of LANG NO, see <Language Number (LANG NO)>.

Note: This parameter is in conjunction with the system data [ASYD: SYS 1, Index 78, Bit 4 and Index 389, Bit 0-4].

Note: When the first language is changed from Japanese to language other than Japanese, the originating method of appropriate station will be in pre-line selected state.

Note: The first language can be assigned by the system data [ASYD: SYS 1, Index 389, Bit 0-4 (Display Language on Digital terminal LCD)] as well. Refer to “Command Manual” for the detail. The first language assigned last by ASLG or ASYD will be used.

c.) Specify the Second Language.

- **[SECOND] tab -** Second Language Data

  Second Language field: Select either of the radio buttons.
  System Setting
  Select Terminal

LANG NO: Specify the Second Language used per system or per terminal. For the details of LANG NO, see <Language Number (LANG NO)>.

d.) Specify the language used per terminal.

- **[TERMINAL] tab -** Terminal Data Setting

Line Number Type: Specify the radio button for Line Number Type.

- STN
  - TN START/END: Enter the appropriate range of Tenant Numbers.
  - STN START/END: Enter the appropriate range of Station Numbers.
  - KIND: Specify terminal kind.
- TELN(LDM)
  - UGN: User Group Number
  - TELN START/END: Enter the appropriate range of Telephone Numbers.
  - KIND: Specify terminal kind.

Change Data Type: Check the check box for "LANG".

- LANG: Specify the language. **Note 3**, **Note 4**, **Note 5**
- TONE: This feature is not used. **Note 6**

Note 3: When "System Setting" is registered in Second Language on SECOND tab, selecting "DTERMIP" in KIND makes "First Language" and "Second Language" selectable.

Note 4: When "Select Terminal" is registered in Second Language on SECOND tab, selecting "DTERMIP" in KIND makes "First Language" and "Language registered by SECOND tab" selectable.

Note 5: When "LEGACY" is selected in KIND, "First Language" or "Second Language" can be selected.

Note 6: For the detailed setting, refer to MULTIPLE TONE -D [M-119D].

e.) Specify the language used per Unit.

- **[UNIT] tab**

Select the Module Group Number and Unit Number, then specify languages used per Unit.

MG: Assign the Module Group Number of the node.

U: Assign the Unit Number of the node.

LANG: Specify the language used per Unit. For the details of LANG NO, see <Language Number (LANG NO)>.

STEP 4: **MEM_HDD** - Back up the data to the hard disk.

Direction Select: Choose "Memory to Hard Disk".

Data Type Select: Choose "Data Memory"

LP Select: Specify the LP Number.

To change language with Soft key

STEP 1: **ADSL** - Assign the LANGUAGE CHANGE to Soft key. SKP: 0-15 (Soft key Pattern) SN: 0 (Idle), SKN: 0-15 (Soft key number appears on the PCPro), FKY: 177 (LANGUAGE CHANGE key [LANG]).

STEP 2: **ADKS** - Assign the following to use a Soft key on the Desktop terminal for Multilingual Display.
TN: Tenant Number
STN: Station Number
SKP: 0-15 (Software Key Pattern assigned in STEP 1)
LKP: 1-3 (Location of Key Pattern)
PG: 0/1 (Soft key/Feature key is allocated for PAGE CHANGE key)

To change language with Feature key

STEP 1: **AKYD** - Assign to a Line/Feature key for Language Change key (LANG). KYI: 1 (Feature Key), FKY: 177 (Language Change Key [LANG]).

To delete

The deletion procedure is the reverse order of assigning the feature.

FCCS Programming

To activate Multilingual Display

STEP 1: **ASYDN** - System Data 1, Index 677, Bit 1. Specify the time display format of Digital terminal whether 12-hour display or 24-hour display. **Note 7**, **Note 8**
0=12-hour display
1=24-hour display

Note 7: This service is available for Digital terminal only. IP Station can provide only 12-hour display.

Note 8: This data needs to be assigned to each Local Node.

System Data 1, Index 677, Bit 2. Specify the display type for day of the week and month in order to adjust display on the Digital terminal to that on the IP Station. **Note 9**, **Note 10**
0=Type 1 (Display for Europe)
1=Type 2 (Standard)

Note 9: This data needs to be assigned to each Local Node.

Note 10: This data setting is available for Dterm Series i and DT300 Series.

STEP 2: (Optional) To use OPTIONAL LANGUAGE PACK LOADING [O-42], assign the following system data, and go to the next step:
ASYDN - System Data 1, Index 679, Bit 2 (Optional Language Pack Loading)=1 (in service)
System Data 1, Index 686, Bit 7: Setting for Azerbaijani/Turkish **Note 11**
0=Out of Service
1=In Service **Note 12**

Note 11: Available from FP95-114 V4 Issue 4.0.

Note 12: Assign 1 only if Azerbaijani/Turkish is used.

STEP 3: **ASLG -** Assign the language display.

Note: Azerbaijani/Turkish requires to set Russian as First Language and Azerbaijani/Turkish as Second Language.

a.) (Optional) If Extension Languages are used, set their Language numbers.

- **[EXTENSION] tab**

  Choose the desired language files from the [Select Language] list as Extension Languages, and assign Extension Language numbers [18-27] to each of them by entering the numbers in the [LANG NO] field of the [Select Language] list.

b.) Specify the First Language used per system.

- **[FIRST] tab -** Basic Data Setting

  First Language Data field
  LANG NO: Specify the First Language used per system. For the details of LANG NO, see <Language Number (LANG NO)>.

Note: This data needs to be assigned to each Local Node.

Note: This parameter is in conjunction with the system data [ASYD: SYS1, Index 78, Bit 4 and Index 389, Bit 0-4].

Note: When the first language is changed from Japanese to language other than Japanese, the originating method of appropriate station will be in pre-line selected state.

Note: The first language can be assigned by the system data [ASYD: SYS1, Index 389, Bit 0-4 (Display Language on Digital terminal LCD)] as well. Refer to “Command Manual” for the detail. The first language assigned last by ASLG or ASYD will be used.

c.) Specify the Second Language.

- **[SECOND] tab -** Second Language Data

  Second Language field: Select either of the radio buttons.
  System Setting
  Select Terminal
  LANG NO: Specify the Second Language used per system or per terminal. For the details of LANG NO, see <Language Number (LANG NO)>.

Note: This data needs to be assigned to each Local Node.

d.) Specify the language used per terminal.

- **[TERMINAL] tab -** Terminal Data Setting

  Line Number Type: Specify the radio button for Line Number Type.
  - STN
    TN START/END: Enter the appropriate range of Tenant Numbers.
    STN START/END: Enter the appropriate range of Station Numbers.
    KIND: Specify terminal kind.
  - TELN(NDM)
    FPC: Fusion Point Code
    UGN: User Group Number
    TELN START/END: Enter the appropriate range of Telephone Numbers.
    KIND: Specify terminal kind.

  Change Data Type: Check the check box for "LANG".
  - LANG: Specify the language. **Note 13**, **Note 14**, **Note 15**
  - TONE: This feature is not used. **Note 16**

Note 13: When "System Setting" is registered in Second Language on SECOND tab, selecting "DtermIP" in KIND makes "First Language" and "Second Language" selectable.

Note 14: When "Select Terminal" is registered in Second Language on SECOND tab, selecting "DtermIP" in KIND makes "First Language" and "Language registered by SECOND tab" selectable.

Note 15: When "Legacy" is selected in KIND, "First Language" or "Second Language" can be selected.

Note 16: For the detailed setting, refer to MULTIPLE TONE -D [M-119D].

e.) Specify the language used per Unit.

- **[UNIT] tab**

  Enter the Module Group Number and Unit Number, then specify languages used per Unit.
  MG: Assign the Module Group Number of the node respectively.
  U: Assign the Unit Number of the node respectively.
  LANG: Specify the language used per Unit. For the details of LANG NO, see <Language Number (LANG NO)>.

Note: This data needs to be assigned to each Local Node.

STEP 4: **MEM_HDD_N** - Back up the data to the hard disk.
Direction Select: Choose "Memory to Hard Disk".
Data Type Select: Choose "Data Memory"
LP Select: Specify the LP Number.

To delete

The deletion procedure is the reverse order of assigning the feature.

<LANGUAGE NUMBER (LANG NO)>
This table shows kind of Language for each LANG NO. For LANG NO, set the data according to this table.

LANGUAGE	LANG NO	LANGUAGE	LANG NO
Japanese	1	Danish	10
English for NEC	2	Swedish	11
Portuguese for NEC	3	Greek	12
Spanish for NEC	4	English	13
Canadian French for NEC	5	Portuguese	14
German	6	Spanish	15
Italian	7	French	16
Dutch	8	Russian	17
Not used	9	Extension Languages 1-10	18-27

M-118 Mobility Access

General Description

This feature allows users to use their mobile phones or home phones as a Station accommodated in the System. By using this feature, the users can access the Telephony Server's service features via any phone, anywhere, and at any time (even when they are out of office). This feature is software based and does not require any additional hardware.

Note: This feature is available in EMEA only.

The following explains the terms used in this feature description.

Terms	Description
MA (Mobility Access) Number	Station Number that is given to outside-line telephone by this feature
Mobile Phone	Outside-line telephone having Mobility Access Number
Station	Telephone accommodated in the System
MA (Mobility Access) Station	Station to which Mobility Access data is set
Outside Party	Outside-line telephone having no Mobility Access Number
MA Mode	A Station operates in conjunction with a Mobile Phone
MA Termination	Call Termination from Mobile Phone
MA Call Forwarding	MA Station automatically forwards received calls to Mobile Terminal

DID Number for each MA Station is to be displayed. However, when DID Number for each MA Station is not set, the Telephony Server Pilot Number or Contract Line Number is displayed.

The following table shows the list of the number display patterns.

Making a call to Station (Mobile Phone ® Station)		
	Display	
Calling Party : Mobile Phone	Dialed Number	0471859999+2222
Called Party : Station	MA Station Number	1111
Making a call to outside party (Mobile Phone ® Outside party)		
	Display	
Calling Party : Mobile Phone	Dialed Number	0471859999+0+0678981234
Called Party : Outside telephone	DID Number of MA Station	0471851111
Receiving a call from Station (Station ® Mobile Phone)		
	Display	
Calling Party : Station	Dialed Number	1111
Called Party : Mobile Phone	DID Number of Calling Station	0471852222
Receiving a call from Outside Party (Outside party ® Mobile Phone)		
	Display	
Calling Party : Outside telephone	Dialed Number	0471851111
Called Party : Mobile Phone	DID Number of MA Station	0471851111
Receiving a call from another Mobile Phone (Mobile Phone ® Mobile Phone)		
	Display	
Calling Party : Outside telephone	Dialed Number	0471851111
Called Party : Mobile Phone	DID Number of MA Station	0471852222

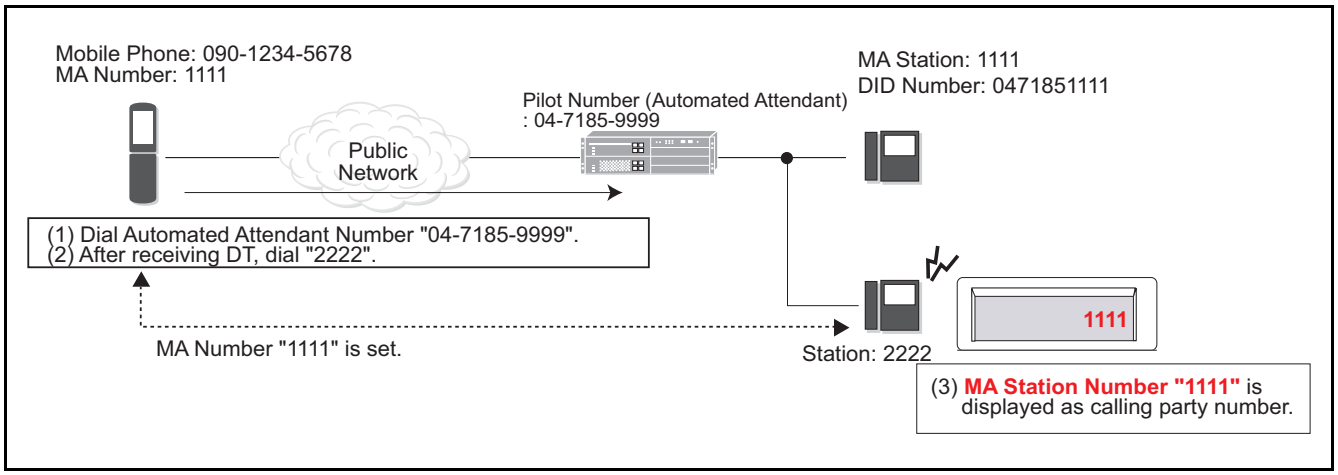
Making a call to Station (Mobile Phone → Station)

When a call is made from Mobile Phone to Station, Mobility Access Number is sent to the called Station.

<Example>

In the case of the Automated Attendant:

Step 1: Mobile Phone dials the Automated Attendant Number to connect to the system.
Step 2: After receiving DT (Dial Tone), dials Station Number “2222”.
Step 3: The called Station answers the call originated from Mobile Phone. **Mobility Access Number “1111”** is sent to the Station).



Making a call to outside party (Mobile Phone → Outside party):

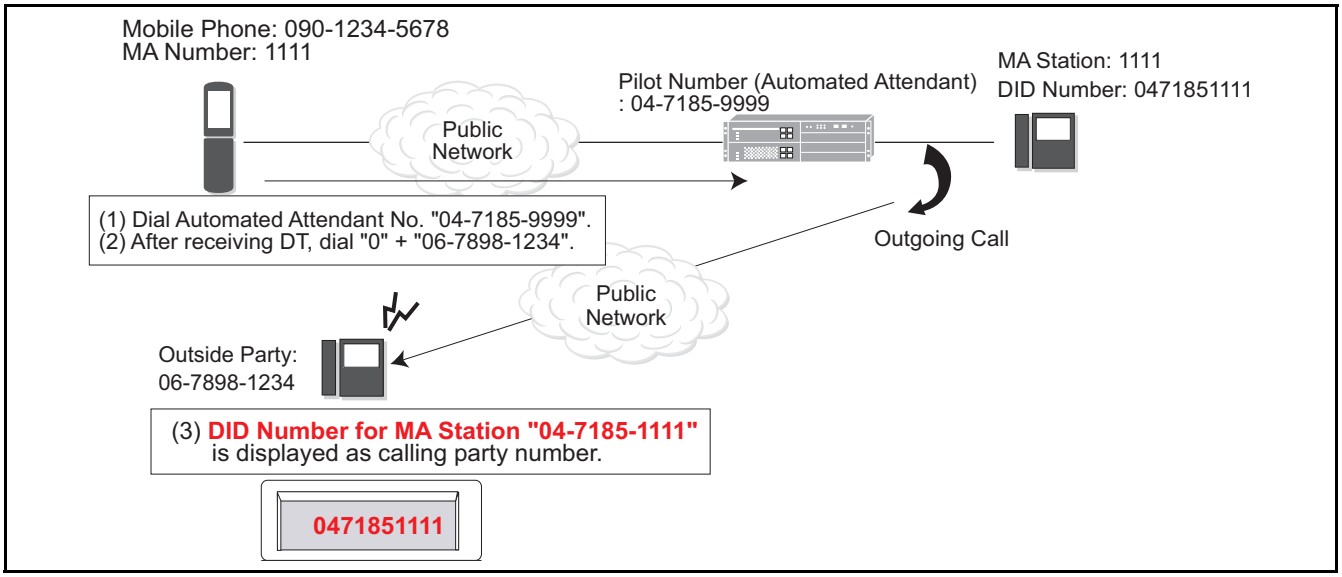
<Example>

In the case of the Automated Attendant:

Step 1: Mobile Phone dials the Automated Attendant Number to connect to the system.

Step 2: After receiving DT (Dial Tone), dials Outgoing Access Code + destination outside party's telephone number.

Step 3: The called outside party answers the call originated from Mobile Phone. **DID Number for MA Station "04-7185-1111"** is sent to the called party (See Figure).

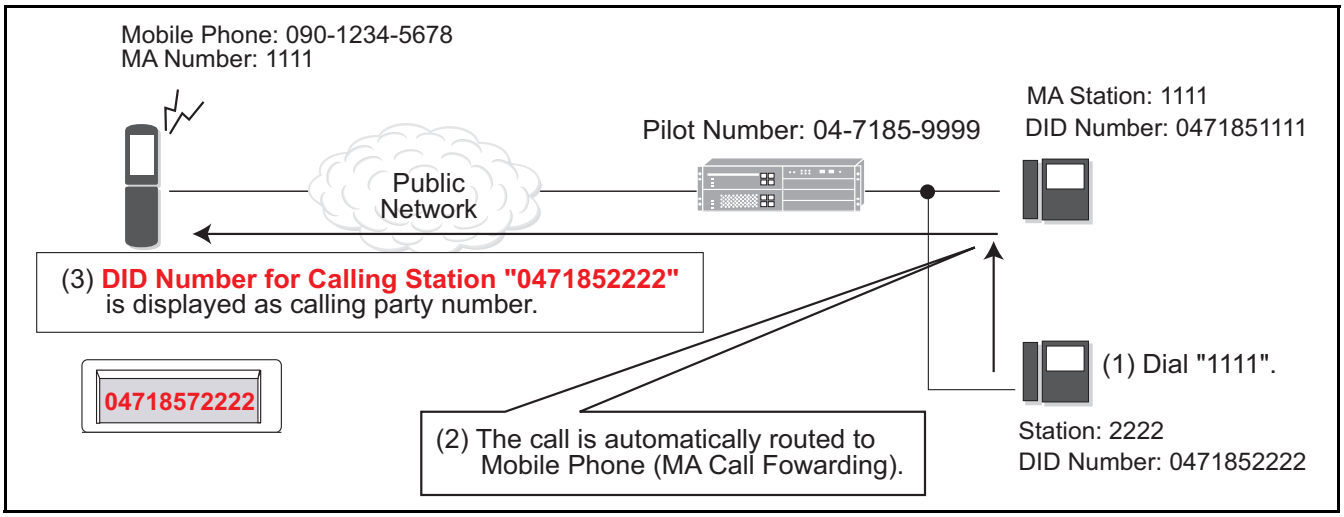


Receiving a call from Station (Station → Mobile Phone)

When an incoming call is terminated to MA Station, MA Station is not rung and the call is forwarded to Mobile Phone immediately. This is called “MA Call Forwarding”. In MA Call Forwarding, call forwarding process is hidden from the called party and it looks like a station call.

<Example>

Step 1: Station (2222) dials the MA Station Number “1111”.
Step 2: The call is automatically forwarded to the particular Mobile Phone (MA Call Forwarding).
Step 3: Mobile Phone answers the call. **DID Number for Calling Station “04-7185-2222”** is sent to Mobile Phone.



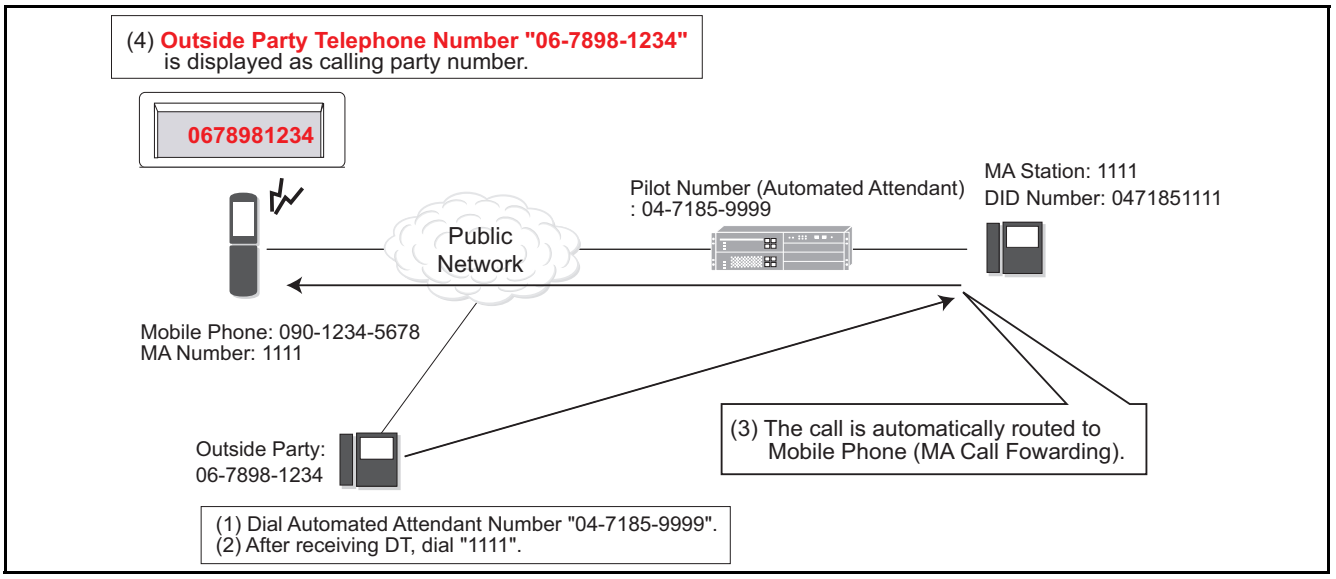
Receiving a call from Outside Party (Outside party → Mobile Phone)

When a call is made from outside party to MA Station, the call is automatically routed to the particular Mobile Phone (MA Call Forwarding). At this time, MA Station is not rung.

<Example>

In the case of the Automated Attendant:

Step 1: Outside party dials the Automated Attendant Number to connect to the system.
Step 2: After receiving DT (Dial Tone), dials Station Number of MA Station “1111”.
Step 3: The call is automatically forwarded to the particular Mobile Phone (MA Call Forwarding).
Step 4: Mobile Phone answers the call. **Outside Party Telephone Number “060-7898-1234”** is sent to Mobile Phone.



Receiving a call from another Mobile Phone (Mobile Phone → Mobile Phone)

When a call is made from another Mobile Phone to MA Station, the call is automatically routed to the particular Mobile Phone. At this time, MA Station is not rung.

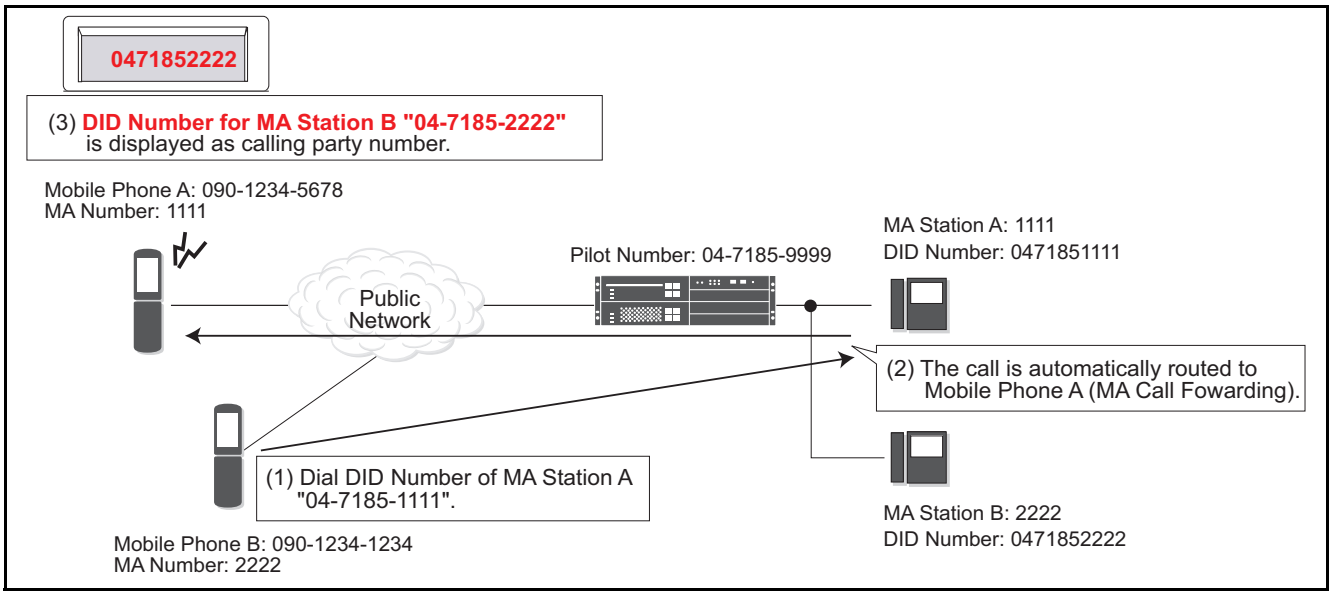
<Example>

In the case of the Direct Inward Dialing (DID)

Step 1: Mobile Phone B dials DID Number of MA Station A.

Step 2: The call is automatically forwarded to Mobile Phone A. (MA Call Forwarding).

Step 3: Mobile Phone A answers the call. **DID Number for Calling Station "04-7185-2222"** is sent to Mobile Phone.



Call Back-Out of Zone:

When a call is forwarded to Mobile Phone from MA Station, the call may be disconnected according to the status of the Mobile Phone (busy or Out of Zone, etc.). In this case, if Service Feature Restriction Class that allows SFI 236 (Call Back-Out of Zone) is assigned to the MA Station, CALL BACK can be activated. The setting of ASYDL, SYS1, Index 1088, Bit 1 determines a destination of CALL BACK.

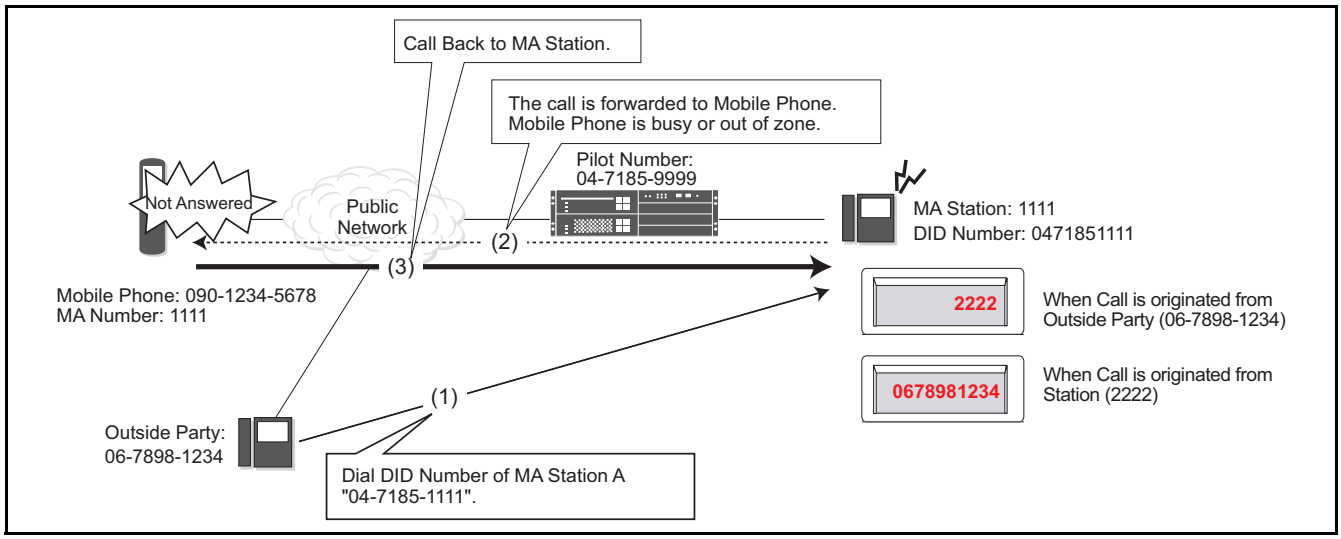
<Example>

(1) Outside party dials DID Number of MA Station.

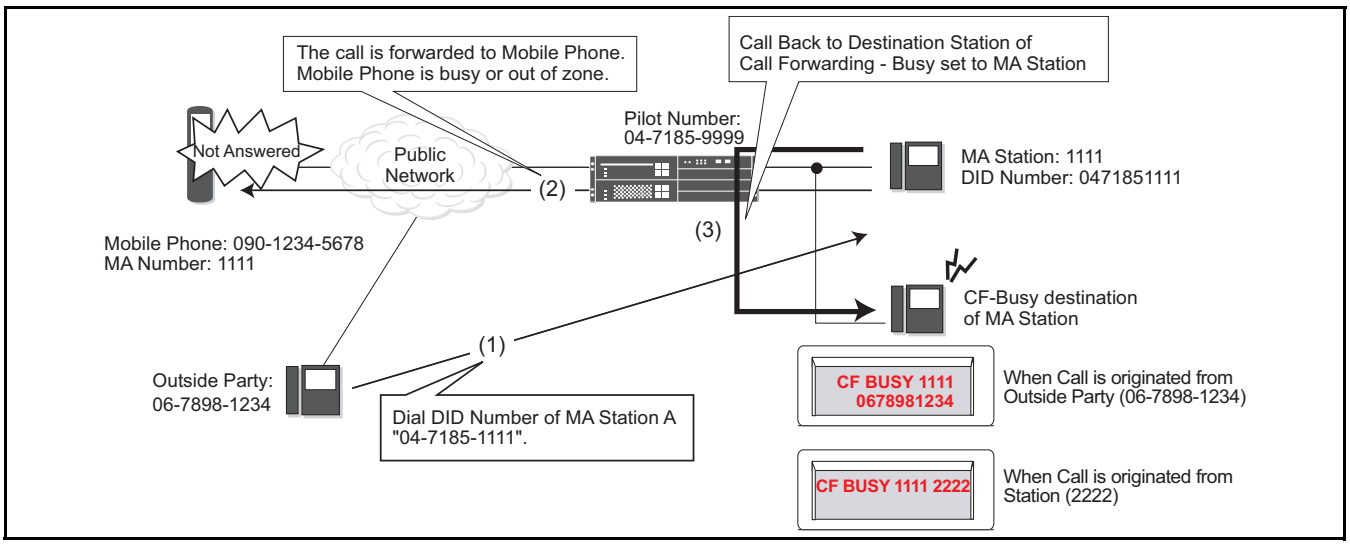
(2) The call is automatically forwarded to the particular Mobile Phone.

(3) Mobile Phone does not answer the call; the call is disconnected and a predetermined CALL BACK destination is rung.

- When ASYDL, SYS1, Index 1088, Bit 1 = 0: MA Station



- When ASYDL, SYS1, Index 1088, Bit 1 = 0: Destination Station of Call Forwarding - Busy Line set to MA Station.



En-block Access

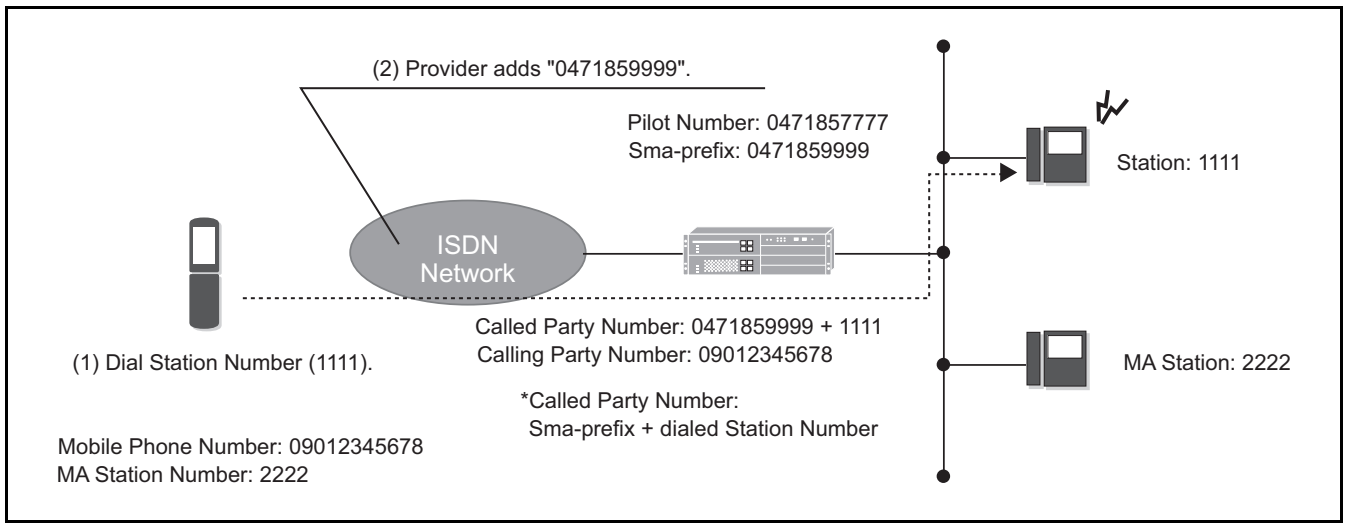
Mobile Phone can make a Station Call/Central Office Call/TIE Line Call by dialing provider specific number and destination telephone number, such as Station Number. Mobile Phone users can use their existing phone book when making these calls. It is required to ink a contract with provider to add sma-prefix.

<Example>

Step 1: Mobile Phone dials Station Number (1111).
Step 2: The originated call is routed to the System according to the following information:

Called Party Number: sma-prefix (0471859999) + Station Number (1111).
Calling Party Number: Mobile Phone Number (09012345678)

Step 3: Station (1111) is rung.



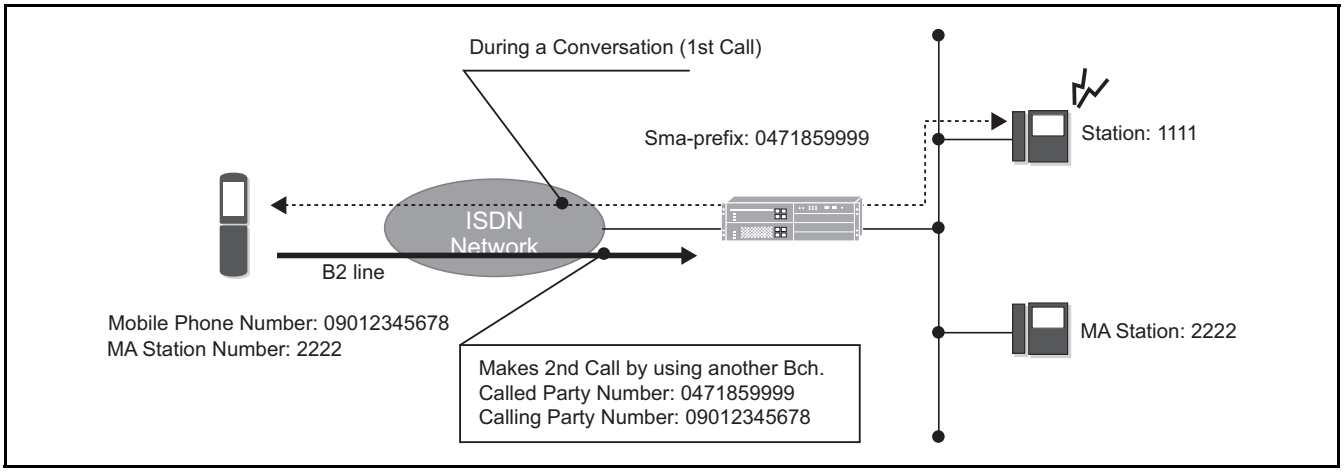
2nd Call

2nd Call service enables hooking on Mobile Phone.

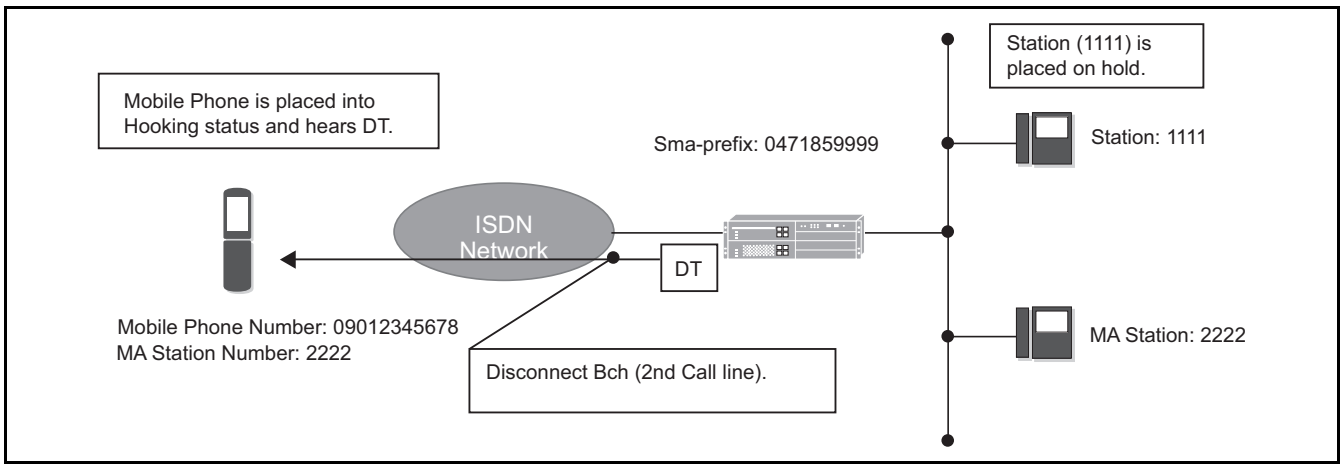
- **2nd Call without En-block Access**

<Example>

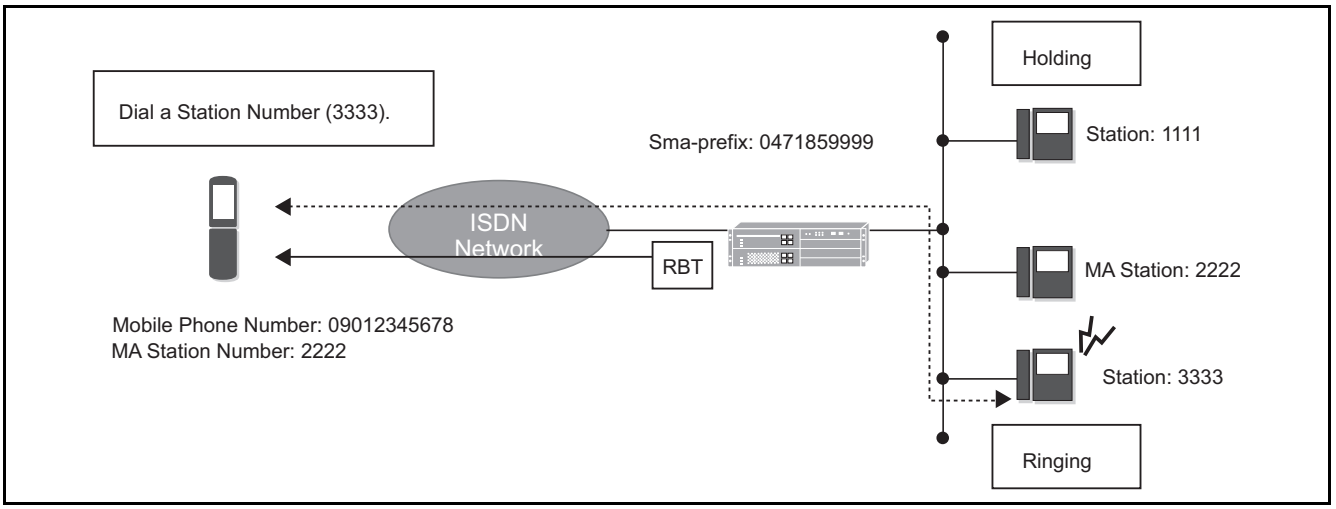
Step 1: While Mobile Phone is engaged in a two-party connection with a Station (1111), the Mobile Station makes another call (2nd Call) to sma-prefix by using B2 line.



Step 2: When the System confirms that 2nd Call is made, the Mobile Phone is placed into Hooking status and the Station (1111) is placed on hold. Dial Tone (DT) is sent to the Mobile Phone and the B2 line is disconnected.



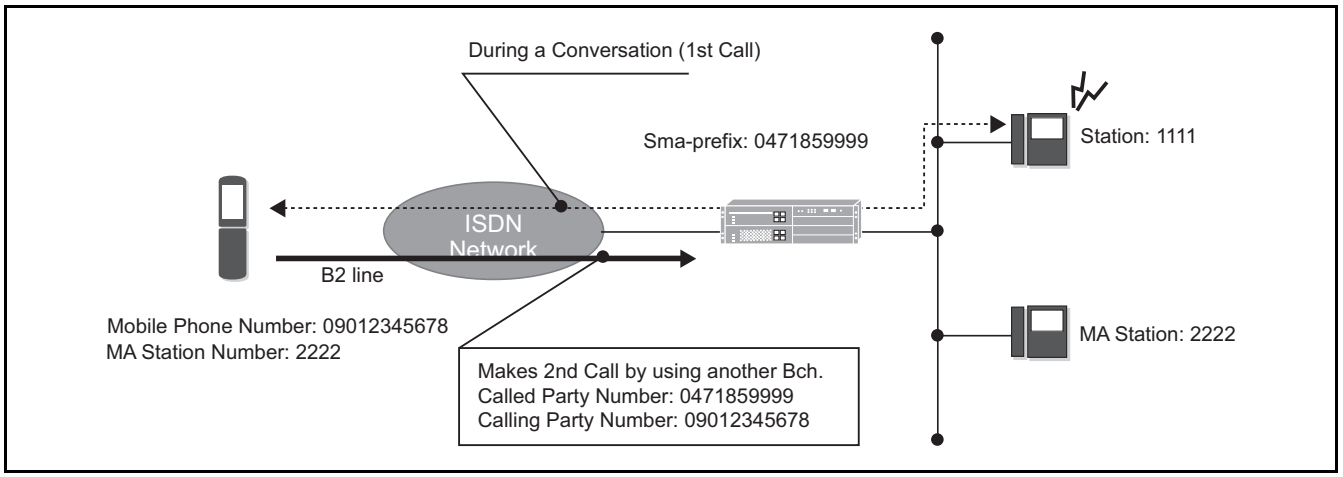
Step 3: The Mobile Phone makes a Station Call/Central Office Call/TIE Line Call by dialing a number.



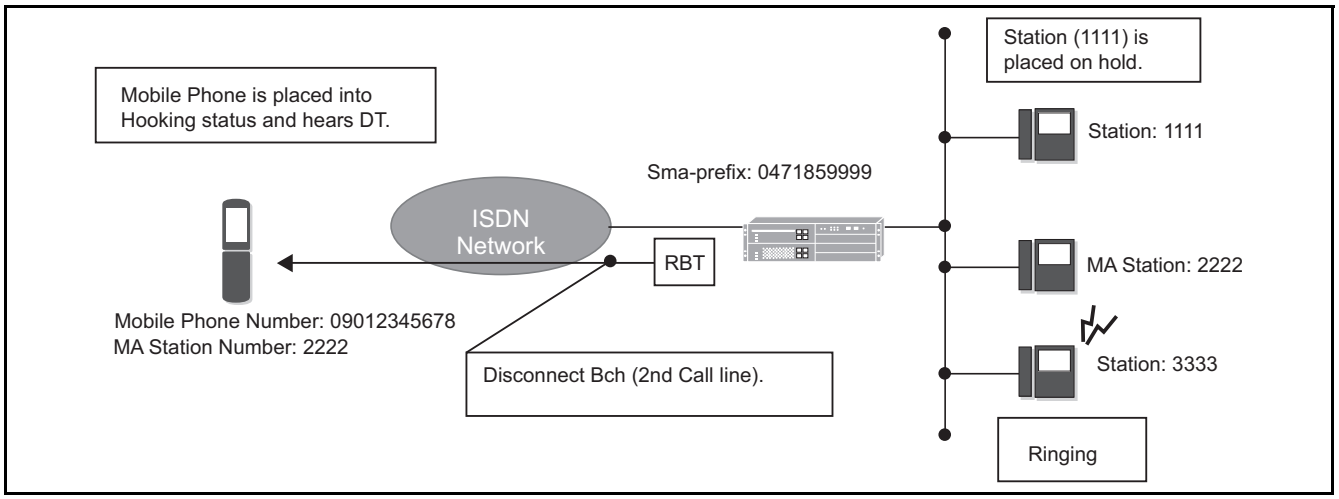
- **2nd Call with En-block Access**

<Example>

Step 1: While Mobile Phone is engaged in a Two-party Connection with Station (1111), the Mobile Station makes another call (2nd Call) to sma-prefix by using B2 line.



Step 2: If the System confirms that 2nd Call is made, the Mobile Phone is placed into Hooking status and the Station (1111) is placed on hold. The Mobile Phone makes a Station Call/ Central Office Call/TIE Line Call according to additional number included in destination party number (In the following figure, Station Number “3333” is dialed). The B2 line is disconnected.



Operating Procedure

This feature can be set/canceled/confirmed not only from PCPro (AMAD command) but from MA Station and Mobile Phone.

Operating Terminal of Mobility Access

Operating Terminal	Set		Cancel	
	Access Code	Feature/Soft key	Access Code	Feature/Soft key
MA Station	-	-	X	X
Mobile Phone	X	-	X	-

X: Available -: Not available

- Set Mobility Access feature from Mobile Phone (by using Access Code):

1. Dial the Automated Attendant/Remote Access Number to connect to the system. Receive Ringback Tone (RBT).
2. About six seconds later, receive Second Dial Tone (2nd DT).
3. Dial the Remote Call Forwarding Control Access Code. Receive Ringback Tone (RBT).
4. About six seconds later, receive Special Dial Tone (SPDT).
5. Dial the Station ID Code. Receive Special Dial Tone (SPDT).
6. Dial the target Station Number. Receive Dial Tone (DT).
7. Dial the Mobility Access Entry Code. Receive Service Set Tone (SST).
8. The LED of MOBILITY ACCESS Feature key on the Station flashes red. (In the case of Soft key, "MACC" flashes.).

Note: If Mobility Access data has already been set, the previous data will be overwritten with new one.

<Example of Operation>
Automated Attendant Number: 04-7185-7000
Remote Call Forwarding Control Access Code: 2999
Station ID Code: 111 (If the code is less than 10 digits, a "#" must be dialed as the last digit.)
Station Number: 2000
Mobility Access Entry Code: 153

When Announcement service is used:
04-7185-7000 → <Announcement> → <RBT> → <2nd DT> → 2999 → <RBT> → 111# → 2000 → 153

When Announcement service is not used:
04-7185-7000 → <RBT> → <2nd DT> → 2999 → <RBT> → 111# → 2000 → 153

- Cancel Mobility Access feature from Mobile Phone (by using Access Code):

1. Dial the Automated Attendant/Remote Access Number to connect to the system. Receive Ringback Tone (RBT).
2. About six seconds later, receive Second Dial Tone (2nd DT).
3. Dial the Remote Call Forwarding Control Access Code. Receive Ringback Tone (RBT).

4. About six seconds later, receive Special Dial Tone (SPDT).
5. Dial the Station ID Code. Receive Special Dial Tone (SPDT).
6. Dial the Mobility Access Number. Receive Dial Tone (DT).
7. Dial the Mobility Access Cancel Code. Receive Service Set Tone (SST).
8. The LED of the MOBILITY ACCESS Feature key on the Station goes off. (In the case of Soft key, "MACC" becomes steady.).

<Example of Operation>
Automated Attendant Number: 04-7185-7000
Remote Call Forwarding Control Access Code: 2999
Station ID Code: 111 (If the code is less than 10 digits, a "#" must be dialed as the last digit.)
Station Number: 2000
Mobility Access Cancel Code: 154

When Announcement service is used:
04-7185-7000 → <Announcement> → <RBT> → <2nd DT> → 2999 → <RBT> → 111# → 2000 → 154

When Announcement service is not used:
04-7185-7000 → <RBT> → <2nd DT> → 2999 → <RBT> → 111# → 2000 → 154

- Cancel Mobility Access feature from Station (by using Access Code):

1. Lift the handset or press the SPEAKER key. Receive Dial Tone (DT).
2. Dial the Mobility Access Cancel Code. Receive Service Set Tone (SST).
3. Replace the handset or press the SPEAKER key.

- Cancel Mobility Access feature from Station (by using Dterm Feature/Soft key):

1. Lift the handset or press the SPEAKER key. Receive Dial Tone (DT).
2. Press the MOBILITY ACCESS Feature key. Receive Service Set Tone (SST).
3. The LED of the MOBILITY ACCESS Feature key goes off. (In the case of Soft key, "MACC" becomes steady.).
4. Replace the handset or press the SPEAKER key.

- Confirm the registered Mobility Access feature data from Station (by using a Feature/Soft key):

1. Press the MOBILITY ACCESS Feature key (in the on-hook status).
2. Number of Mobile Phone (including OG Access Code) is displayed.

- To confirm the registered Mobility Access feature via PCPro command (AMAD)

1. Choose a data type in the TYPE parameter.
2. Enter each parameter, and then click "Get" button.

    - For NDM:
      Assign UGN and TELN.

- For LDM:
  Assign FPC, UGN and TELN.
- For STN:
  Assign FPC, TN and STN.

Note: For more details on each parameter setting, see the Programming section of this feature.

3. The registered MA service data (Mobile Phone Number, Trunk Access Code) appears.

- To set the Mobility Access feature via PCPro command (AMAD)

1. Choose a data type in the TYPE parameter.
2. Enter each parameter, and then click “Get” button.

- For NDM:
  Assign UGN and TELN.
- For LDM:
  Assign FPC, UGN and TELN.
- For STN:
  Assign FPC, TN and STN.

Note: For more details on each parameter setting, see the Programming section of this feature.

3. Enter Mobile Phone Number into the MA parameter.
4. Enter the number of digits for Trunk Access Code into the TACC parameter.
5. Click “Set” button.

- To cancel the Mobility Access feature via PCPro command (AMAD)

1. Choose a data type in the TYPE parameter.
2. Enter each parameter, and then click “Get” button.

- For NDM:
  Assign UGN and TELN.
- For LDM:
  Assign FPC, UGN and TELN.
- For STN:
  Assign FPC, TN and STN.

Note: For more details on each parameter setting, see the Programming section of this feature.

3. Check to see that MA feature has been set.
4. Click “Del” button.

Service Conditions

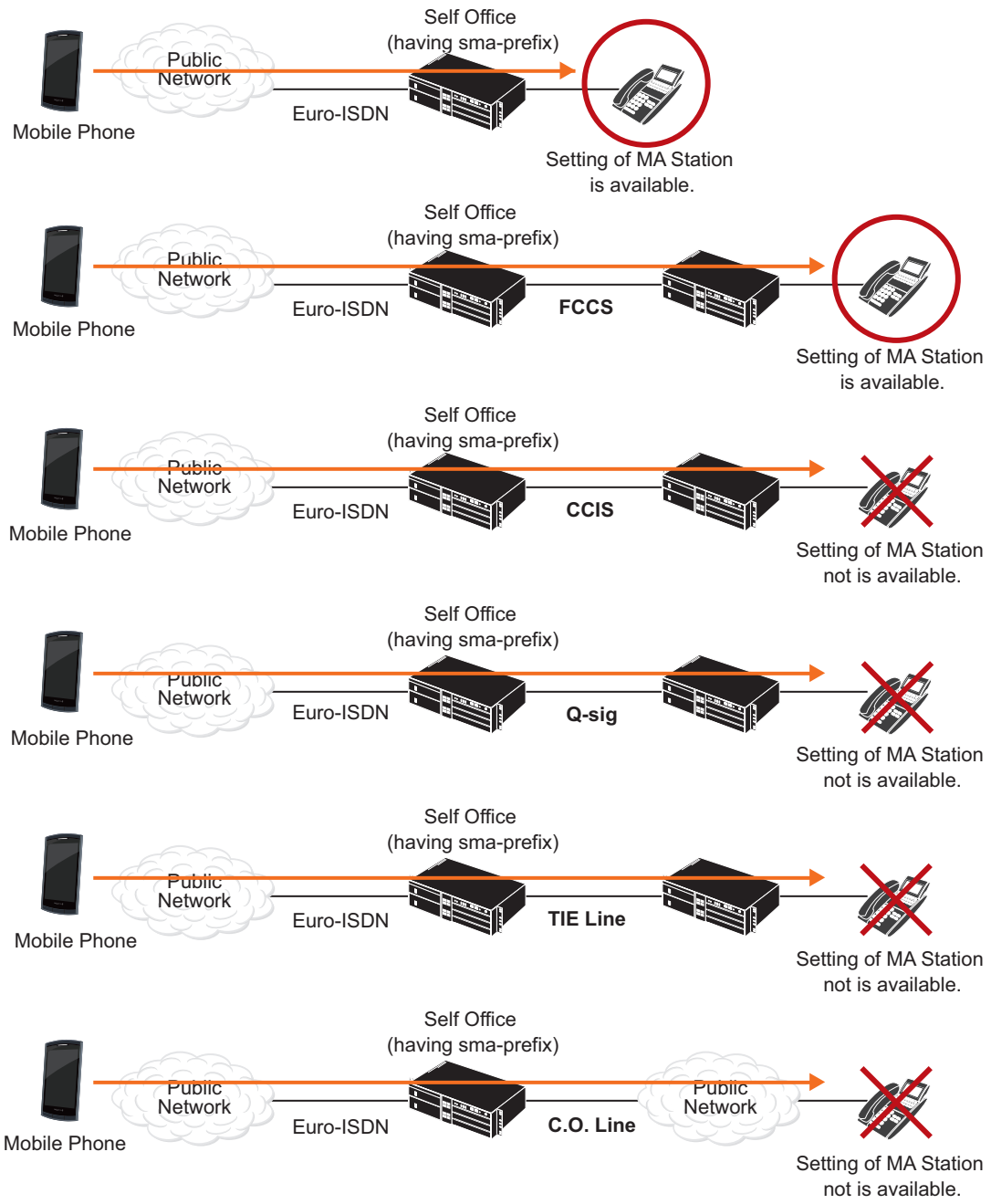
1. This feature can be set/canceled from Mobile Phone via Euro-ISDN (Transfer Rate: 64 Kbps, Bearer Capability: Speech or 3.1 kHz Audio) only. The following table shows the available network configuration for this feature.

	Self Office Note 1	via FCCS	via CCIS	via G-sig	via TIE line	via C.O. line
Setting of Mobility Access	X	X	N	N	N	N
Call to Mobile via MA Station	X	X	X	X	X	X
Incoming Call from Mobile	X	X	X	X	X	X

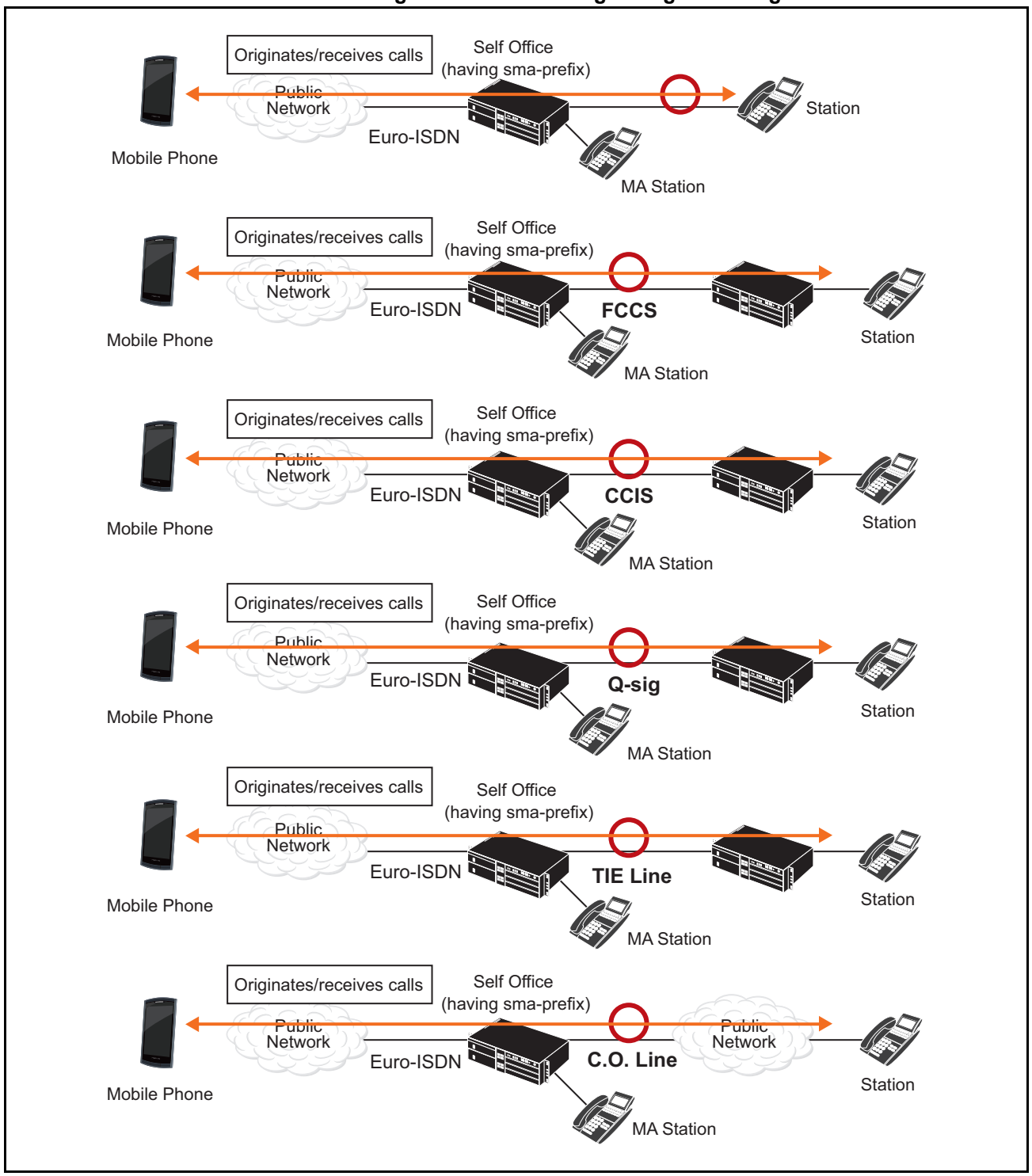
X: Available N: Not Available

Note 1: The office accommodating Euro-ISDN line with sma-prefix.

Network Configuration for Mobility Access Setting



Network Configuration for Call Originating/Receiving



2. Legacy Dterm, DtermIP and Analog telephone can be set as MAS Station. Standard SIP terminal, PS, Attendant Console and ISDN terminal cannot be set.
3. This feature is not available for Hotel/OAI/ACD System.

4. This feature is not available via FCCS link. In the FCCS network, this feature can be activated on a node basis. If Mobile Phone places a call to the other node Station, the call is handled as "Trunk Incoming Call", and the Station Number is not displayed on the called party.
5. A Mobility Access Number (Station) can be allocated to an outside party's number.
6. Up to 16,000 Stations can be set this feature.
7. A maximum number of digits for Mobility Access Number is 24.
8. To set/cancel this feature from Mobile Phone, the Mobile Phone needs to access the System via Automated Attendant [A-82]/Remote Access to System [R-2] feature.
9. Make the Reversed Number (ARNP) of incoming route coincide with Outgoing Access Code.
10. When using this feature, it is necessary to set the Station ID Code from the terminal. The following shows the conditions concerning the Station ID Code:
    a.) A maximum number of digits for the Station ID Code is 10.
    b.) When dialing the Station ID Code that is less than 10 digits, a "#" must be dialed as the last digit. If a "#" is not dialed, the dialed Station ID Code becomes invalid.
    c.) If the Station ID Code is not assigned, Station Number is applied by default.
    d.) The assigned Station ID Codes can be saved in Flash Card by PCPro operation or Call Forwarding Data Backup of periodical diagnostic function.
11. Backup of the Mobility Access data and the Station ID Code data is available by PCPro operation or Call Forwarding Data Backup of periodical diagnostic function. Therefore, they can be maintained even after the system initialization with office data loading.
12. Mobile Phone and carrier must support Caller ID Notification Service. If Caller ID Notification Service is disabled by the setting on the Mobile Phone, this feature does not work.
13. ISDN network must support Caller ID Notification Service.
14. When using Distinctive Ringing [D-10]:
    If an Mobile Phone places a call to a Station, Ringer Pattern for Internal Incoming Call is applied.
15. When a call is terminated to a Dterm Station from an Mobile Phone, the display on the Dterm LCD is as follows:

    - If this feature is valid, Mobility Access Number is displayed.
    - If this feature is invalid, the calling party number is displayed. If calling party number is not delivered from the network, incoming trunk type and trunk number are displayed.

16. Elapsed Time Display -D [E-3D] can be provided when a Station answers the call from an Mobile Phone.

17. When multiple call forwarding features are set to a Station:
The setting of SFI 209 and SFI 240 (ASFC command) determines priorities of Logged Out IP Station - Call Destination, Call Forwarding - All Calls and MA Call Forwarding. See the table below.

Call Forwarding features										
(1) Number Sharing (Sub-station is set as forwarding destination) **(2) Logged Out IP Station - Call Destination*** **(3) Call Forwarding - All Calls*** **(4) MA Call Forwarding*** (5) Uniform Call Distribution (UCD) (6) Call Forwarding - Busy Line (7) Station Hunting (8) Do Not Disturb -D (9) Call Forwarding - Don't Answer										
		High ←								→ Low
SFI 209 = 0	SFI 240 = 0	(1)	**(2)**	**(3)**	**(4)**	(5)	(6)	(7)	(8)	(9)
	SFI 240 = 1	(1)	**(4)**	**(2)**	**(3)**	(5)	(6)	(7)	(8)	(9)
SFI 209 = 1	SFI 240 = 0	(1)	**(3)**	**(2)**	**(4)**	(5)	(6)	(7)	(8)	(9)
	SFI 240 = 1	(1)	**(4)**	**(3)**	**(2)**	(5)	(6)	(7)	(8)	(9)

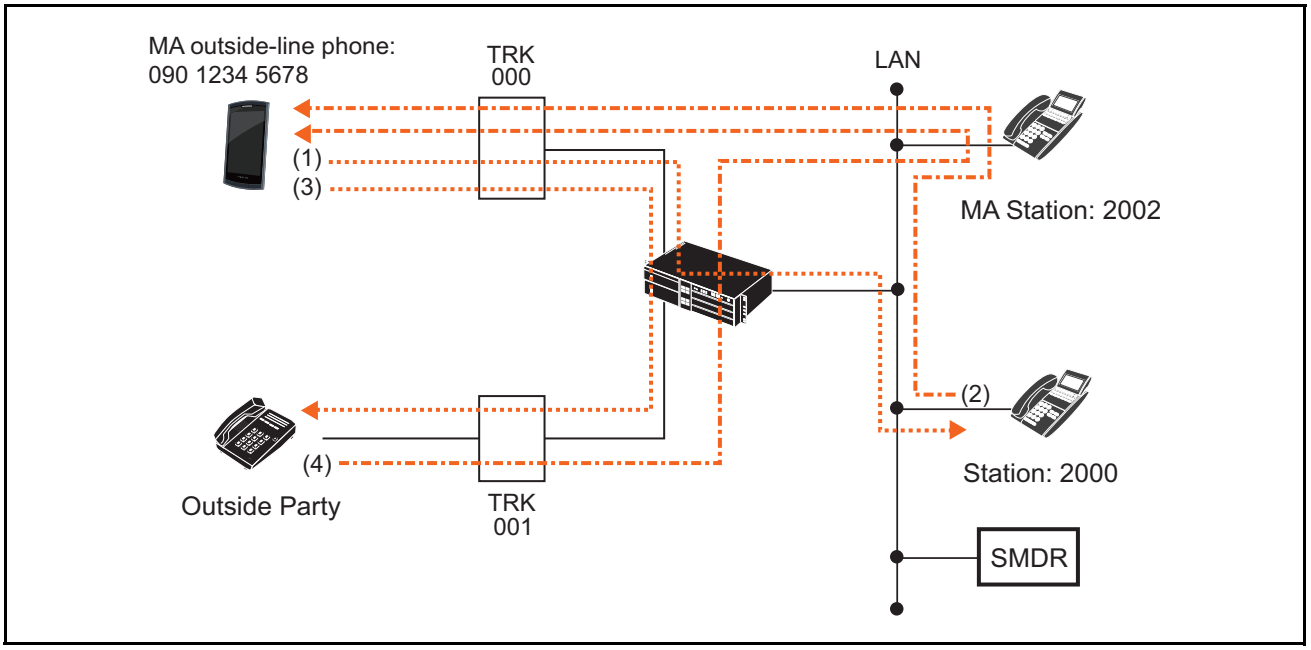
***Variable**

SFI 209 (Priorities of Logged Out IP Station - Call Destination and CF-All Calls)
RES 0 = Logged Out IP Station - Call Destination > CF-All Calls/1 = CF-All Calls > Logged Out IP Station - Call Destination

SFI 240 (Priorities of MA Transfer and Logged Out IP Station - Call Destination & CF-All Calls)
RES 0 = Logged Out IP Station - Call Destination & CF-All Calls > MA Transfer/1 = MA Transfer > Logged Out IP Station - Call Destination & CF-All Calls

18. If MA Station receives a call from the Station that is not allowed to place an outgoing call, the call is not forwarded to the Mobile Phone (restricted).
19. SMDR output information concerning this feature is as follows:

	Connection Pattern	SMDR Output
(1)	Mobile Phone makes a call to a Station	Trunk Incoming (TRK000 -> STN2000)
(2)	Mobile Phone receives a call from a Station	Trunk Outgoing (STN2000 -> TRK000) or Trunk Outgoing (STN2002 -> TRK000) * Depending on the System data assignment
(3)	Mobile Phone makes a call to an outside party	Tandem Outgoing (TRK000 -> TRK001)
(4)	Mobile Phone receives a call from an outside party	Tandem Outgoing (TRK001 -> TRK000)



20. Busy Lamp Field (BLF) display indicates the status of MA Station (not Mobile Phone). Therefore, if the target Mobile Phone is in busy state while the Busy Lamp Field (BLF) display indicates as "idle", Attendant Console hears Reorder Tone (ROT).
21. The following features cannot be used in conjunction with this feature because Mobile Phone can bypass the System and establish a call.

    - Busy Verification [B-3]
    - Call Back [C-1]
    - Call Forwarding - Busy Line [C-2]
    - Call Waiting - Terminating [C-12]
    - Executive Right-of-Way [E-1]

22. When Mobile Phone makes a call to Attendant Console, the call is terminated to the C.O. line key on the Attendant Console. In this case, Station Number of the MA Station is displayed on the Attendant Console's LCD.
23. If an analog telephone that is set as MA Station is called, Warning Tone (One Burst of Ringing) can be provided before the call is forwarded to the target Mobile Phone, depending on the system data assignment (ASYD, SYS1, Index 4, Bit 6).
24. The following shows the conditions concerning Call Back-Out of Zone:
    a.) Call Back-Out of Zone is available for the call to MA Station only.
    b.) Assign a Service Feature Restriction Class that allows SFI 236 to Stations allowed Call Back-Out of Zone.
    c.) Call Back-Out of Zone works only when receiving the following call disconnection reasons (DISC/REL/REL_COMP) from the counter office. If the other reasons are received, ROT/BT is sent to the calling party.

Indication Value (in decimal)	Description
17	Destination User busy
18	No Response from Destination User
19	Destination User does not answer the call during ringing
31	Other normal class
34	No line/channel available
41	Temporary fault

Note: If Mobile Phone is out of zone and the calling party is connected to announcement of carrier, the call is not forwarded because DISC/REL/REL_COMP is not received.

d.) Call Back-Out of Zone works only when MA Station receives the following calls. If any calls other than the following are reached to MA Station, ROT/BT is sent to the calling party.

- Station Call
- Incoming Call from an outside party
- Incoming Call from a Mobile Phone via Automated Attendant [A-82]/Remote Access to System [R-2]

e.) Even if Inband Tone is available, Call Back-Out of Zone works without connecting to Inband Tone.

f.) If it is impossible to call back to MA Station (line lockout state, etc.), ROT/BT is sent to the calling party.

g.) The destination of Call Back-Out of Zone can be determined by the following System data.
ASYDL, SYS1, Index 1088, Bit 1: 0/1 = MA Station/Destination Station of CF-Busy set to MA Station.

h.) A maximum number of Multiple Call Forwarding occurrences can be determined by the System data (ASYD, SYS1, Index 486, Bit 3-Bit 1).

b3	b2	b1		b3	b2	b1	
0	0	0	once	1	0	0	four times
0	0	1	once	1	0	1	five times
0	1	0	twice	1	1	0	five times
0	1	1	three times	1	1	1	five times

i.) In Call Back-Out of Zone, even if Service Display -D is available, forwarding information of multi-hops is not taken over.

j.) When Call Back-Out of Zone is allowed, Camp-on service cannot be used.

25. A newly set MA License is required to activate this feature.
    a.) A lineup of MA Licenses includes the following:
    10/50/100/500/1000 Licenses
    b.) Licences can be added by using License Merge Software (addition method).
    c.) In a node, a maximum of 4,000 ports can be used for this feature.
    d.) In the FCCS network, a maximum of 16,000 ports can be used for this feature.
    e.) When assigning MA Station data, a license is required for each Mobile Terminal. The assigned license is released when MA setting data is canceled.
26. Station Classes which can be assigned to MA Station are TEC=1-4 (Analog Telephone) and TEC=12 (Dterm). MA Station does not support TEC=18 (Virtual Line Appearance), TEC=23 and 24 (ISDN Terminal) and TEC=28 (PHS).
27. If a MA Station is included in Station Hunting group, the MA Station is skipped in hunting.
28. Direct Inward Dialing (DID) [D-8] data is required to be set per Station for Calling Party Number information to be sent from the System.
29. If no data is set, the Telephony Server Pilot Number or Contract Line Number is notified when making/ receiving a call. Calling Party Number is transformed to Subscriber Number (Pilot Number) when it is notified to the Public network. (Status Display (reason#100: Invalid information element contents) is sent from the Public network.)
30. The following table shows the necessary number of the contracts for DID Number.

	Number of Contracts	Method
Number to access system	-	Automated Attendant
Number to be allocated to MA Station	The number of MA Stations (recommended)	Direct Inward Dialing (DID)

31. If the line for each Station cannot be prepared, the Telephony Server Pilot Number or Contract Line Number is notified.
32. Even if calling party encounters the restriction on the Outgoing Trunk Connection, call origination (MA Call Forwarding) to Mobile Phone is allowed. (MA Call Forwarding is handled as “Station Call”.) In this case, Billing Party is determined by the following System data.

    ASYD, SYS1, Index 20, Bit 7: 0/1 = Calling Party/ Transfer Operating Party (Determination of Billing Party for Call Forwarding service)

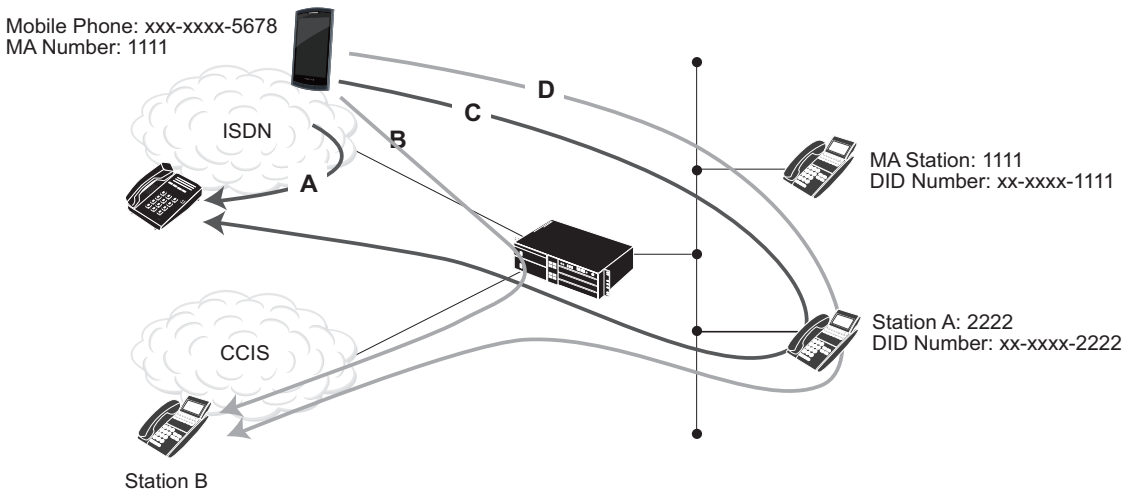
33. When making a call to outside line or CCIS by using MA function, Calling Party Number to be sent to the network varies according to the connection patterns.

Calling Party Number information for each connection pattern is shown in Table 1 and 2.

Calling Party Number to be Sent to the Network When Making a Call to Outside Line after MA Termination

	Connection Pattern	Calling Party Number Information
A	Call origination to outside line after MA Termination	Calling Party Number of MA Station = 0471851111
B	Call origination to CCIS after MA Termination	Calling Party is MA Station = 1111, and the Calling Party Number (telephone number of Mobile Phone) is not notified. DID Number of MA Station (1111) is notified.
C	Call origination to station after MA Termination –> Call transfer to outside line (destination station of Call Forwarding - All Calls/Busy/Don't Answer outside set to MA Station)	Calling Party Number of MA Station = 0471851111
D	Call origination to station after MA Termination –> Call transfer via CCIS	Calling Party is MA Station = 1111, and the Calling Party Number is not notified. Station Number of MA Station is notified.

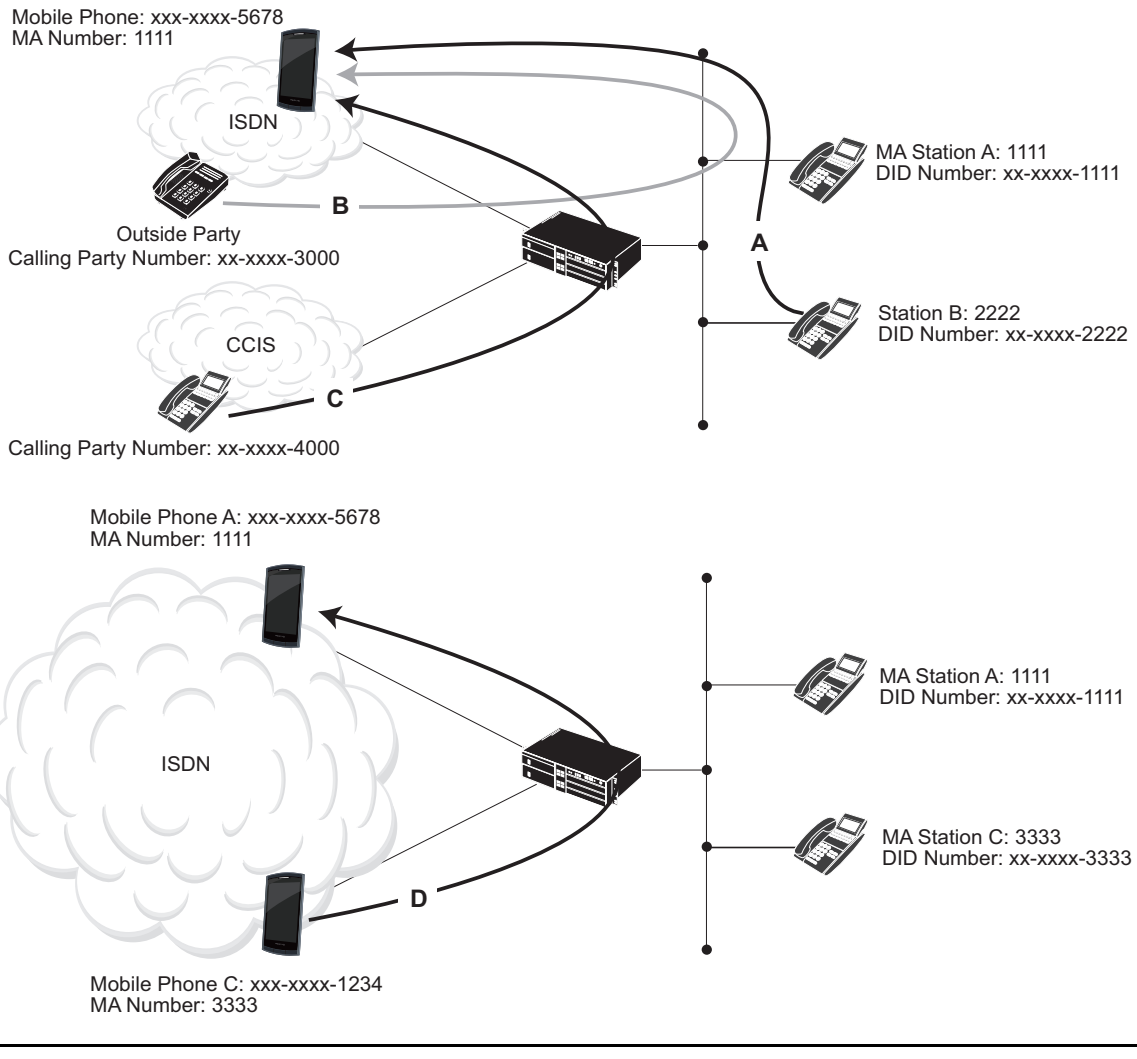
Call Origination to Outside Line after MA Termination



Calling Party Number to be Sent to Mobile Phone by MA Call Forwarding

	Connection Pattern	Calling Party Number Information
A	MA Call Forwarding after call termination from Station	Calling Party Number of caller-side Station = 0345672222
B	MA Call Forwarding after call termination from Outside Line	Calling Party Number of MA Station = 0698763000
C	MA Call Forwarding after call termination from CCIS	Calling Party Number received from CCIS Counter Office = 0698764000
D	Call origination to Station after MA Termination -> MA Call Forwarding	Calling Party Number of caller-side MA Station = 0345672222

Call Origination to Mobile Phone by MA Call Forwarding



34. The following tables show service feature availability and conditions for this feature.

OG Call Features Availability

Code	Service Feature	To Mobile Phone	From Mobile Phone	Remarks
D-8	Direct Inward Dialing (DID)	-	-	
M-7	Music On Hold (External Music on Hold)	N	N	Call Hold from Mobile Phone varies depending on the carrier.
M-99	Multiple Music-on Hold	N	N	Call Hold from Mobile Phone varies depending on the carrier.
D-7	Direct-In Termination (DIT)	-	-	
S-106	Slumber Time - Do Not Disturb	N	-	Mobile Phone is called.
P-3	Power Failure Transfer	-	-	
D-10	Distinctive Ringing	-	N	
A-82/R-2	Automated Attendant/Remote Access to System	-	-	
M-7	Music On Hold (Internal Music on Hold)	N	N	Call Hold from Mobile Phone varies depending on the carrier.
N-1	Night Connection - Fixed	-	-	

×: Available N: Not Available -: Not Applicable

Multiple Line Telephone Features Availability

Code	Service Feature	To Mobile Phone	From Mobile Phone	Remarks
O-37D	Off-Hook Line Number Display -D	-	-	
A-30	Automatic/Manual Intercom	N	-	
I-21	Internal Zone Paging	N	N	
C-7	Call Pickup - Group	N	X	
C-6	Call Hold	N	N	
S-25D	Service Display -D	-	-	
L-6D	Last Number Called -D	-	-	
S-24D	Save And Repeat -D	-	-	
D-165	Dialed Number Privacy Mode	-	-	
S-162	Selective Dialing	-	-	
O-2	Outgoing Trunk Queuing	N	N	
N-28	Name Display - System	N	N	
F-3D	Flash Button -D	-	-	
P-13D	Prime Line Pickup -D	-	-	
N-8D	Non-square Line Assignment -D	-	-	
S-143	Software Multiple My Line Appearance	-	-	
M-20D	Multiple Line Operation -D	-	-	
M-19	Message Reminder	N	N	
R-49	Ringer Mute	-	-	
B-6	Brokerage - Hotline	-	-	
S-26D	Speed Calling - One Touch -D	-	-	
V-2	Voice Call	N	N	
I-10D	Intermediate Station Number Display -D	N	N	
S-21/S-23	Speed Calling - Station/Speed Calling - Group	-	-	
C-5	Call Forwarding - All Calls	-*	X	*This feature has precedence over MA Forwarding.
E-3D	Elapsed Time Display -D	-	-	
T-2	Three-Way Calling	-	N	Three-Way Calling cannot be established from Mobile Phone.
T-18D	Time Display -D	-	-	
E-4D	Exclusive Hold -D	N	N	

Code	Service Feature	To Mobile Phone	From Mobile Phone	Remarks
C-11	Call Transfer - All Calls	N	N	This feature cannot be set from Mobile Phone.
S-140D	Side Tone On/Off -D	-	-	
D-11D	Do Not Disturb -D	N	N	
E-1	Executive Right-of-Way	N	N	
C-1	Call Back	N	N	
B-5D/B-8/ B-11	Boss - Secretary Override -D/Boss - Secretary - Message Waiting Lamp Control/Boss - Secretary Transfer	N	N	
C-22D	Called Station Status Display -D	-	-	
C-3	Call Forwarding - Don't Answer	-*	N	*This feature has precedence over MA Forwarding.
N-7D	Non-exclusive Hold -D	N	N	
C-2	Call Forwarding - Busy Line	-*	N	*This feature has precedence over MA Forwarding.
P-14D	Privacy -D	N	N	

×: Available N: Not Available -: Not Applicable

Attendant Console Features Availability

Code	Service Feature	To Mobile Phone	From Mobile Phone	Remarks
A-8	Automatic Recall	-	-	
A-6	Attendant Loop Release	-	-	
S-15	Serial Call	-	X	
I-4	Individual Trunk Access	-	-	
C-10	Call Transfer - Attendant	-	X	
B-17	Blind Transfer To Attendant	X	X	
C-8	Call Processing Indication	-	-	
T-18	Time Display	-	-	
C-13	Call Waiting Lamp	-	N	
A-1	Attendant Camp-On with Tone Indication	-	N	
N-3	Non-delay Operation	-	-	
I-5	Inter Position Transfer	-	-	
I-2	Incoming Call Identification	-	N*	*Identified as Incoming Central Office Call.
C-9	Call Queueing	-	X	
A-55	Automatic Change of Night Service	-	-	
S-13	Step Call	N	-	
D-114	Delay Announcement - Attendant	-	-	
-	Attendant Console Monitor Function	N	N	
L-1	Lamp Check	-	-	
A-98	Answer Hold - Attendant	-	-	
T-9	Trunk Group Busy Lamp	-	-	
I-6	Individual Attendant Access	-	X	
A-7	Attendant Override	-	-	
A-19	Attendant Night Transfer	-	-	
B-3	Busy Verification	N	N	
B-2	Busy Lamp Field - Flexible	N	N	
-	Semi-Auto Attendant Camp-On	-	N	
S-4	Splitting	-	-	

×: Available N: Not Available -: Not Applicable

Station Features Availability

Code	Service Feature	To Mobile Phone	From Mobile Phone	Remarks
D-156	Dual Station Call	N*	X	When MA Station is set as a sub-station, MA Station is called.
O-6	Off-Hook Alarm	-	-	
D-15	Day/Night Class of Service	-	-	
-	Expanded Call Pickup - Group	N	X	
C-12 C-27D C-31	Call Waiting - Terminating Call Waiting - Answer -D Call Waiting - Originating	N	N	Call Waiting services cannot be activated from Mobile Phone.
C-7	Call Pickup - Group	N	X	
C-6	Call Hold	-	-	
S-13	Step Call	N	N	
F-1	Flexible Numbering of Stations	-	-	
-	Howler Tone sending service	-	-	
S-62	Station 5dB PAD	N	N	This feature is applied to the trunk.
P-4	Pushbutton Calling	-	-	
P-9	Priority Call	-	N	
I-1	Immediate Ringing	-	X	
L-6	Last Number Called - Single-Line Station	-	-	
-	Single Number Service	-	-	PS is not available.
C-25	Call Forwarding - Intercept/ Announcement	-	X	
S-6	Station-Controlled Conference	X	N*	*Mobile Phone can be the Conference Leader.
R-54	Remote Call Forwarding Control	-	N	
C-28	Call Forwarding - All Calls - Outside	-*	X	*This feature has precedence over MA forwarding.
C-74	Call Forwarding - Don't Answer - Outside	N	X	
C-60	Call Forwarding - Busy Line - Outside	N	X	
C-5	Call Forwarding - All Calls	-*	X	*This feature has precedence over MA forwarding.

Code	Service Feature	To Mobile Phone	From Mobile Phone	Remarks
E-11	Emergency Call	-	N	
T-2	Three-Way Calling	N	N	Three-Way Calling cannot be established from Mobile Phone.
T-62	Timed Reminder	N*	-	*MA Station is called.
C-30	Call Pickup - Direct	-	X	
C-11	Call Transfer - All Calls	N	N	This feature cannot be set from Mobile Phone.
S-99	Split Call Forwarding	-	N*	*Handled as a trunk Incoming call.
H-1	Hotline	-	-	
P-11	Periodic Time Indication Tone	N*	-	*Periodic Time Indication Tone is sent to the calling party.
C-81	Call Forwarding - I'm Here	-	-	
-	Tone sent out when the handset is off-hook at the station on where CF - All Calls service is set.	-	N*	*Special Dial Tone is not sent to Mobile Phone.
-	Automatic Call Back Cancel	-	-	
C-15	Class of Service - Individual	-	-	
-	Line Circuit Reverse Relay Control	-	-	
-	Facility Restriction Class	X	N	
E-1	Executive Right-of-Way	N	N	
S-7 S-8 S-9 S-138 S-139	Station Hunting - Circular Station Hunting - Secretarial Station Hunting - Terminal Station Hunting - Priority Station Hunting - Priority with Switchback	-*	X	*MA forwarding has precedence over this feature. If a MA Station is included in Station Hunting group, the MA Station is skipped in hunting
C-1	Call Back	N	N	
B-5D/B-8/B-11	Boss - Secretary Override -D/ Boss - Secretary - Message Waiting Lamp Control/Boss - Secretary Transfer	N	N	
C-3	Call Forwarding - Don't Answer	-*	X	*MA forwarding has precedence over this feature.
C-69	Call Forwarding - All Calls - Announcement	N	X	
C-2	Call Forwarding - Busy Line	-*	X	*MA forwarding has precedence over this feature.

Code	Service Feature	To Mobile Phone	From Mobile Phone	Remarks
P-14D	Privacy -D	N	N	

×: Available N: Not Available -: Not Applicable

Operation and Maintenance Features Availability

Code	Service Feature	To Mobile Phone	From Mobile Phone	Remarks
T-65	Terminal Claim Handling	X	-	
C-160	Call Trace	-	X*	*Mobility Access Number is notified.

×: Available N: Not Available -: Not Applicable

Network Features Availability

Code	Service Feature	To Mobile Phone	From Mobile Phone	Remarks
-	LCR-Special Line Warning Tone	N	-	

×: Available N: Not Available -: Not Applicable

SIP Features Availability

Code	Service Feature	To Mobile Phone	From Mobile Phone	Remarks
S-151	SIP Instant Message	N	-	

×: Available N: Not Available -: Not Applicable

35. The following shows the conditions concerning the setting/canceling/confirming of this feature.
    a.) This feature can be set/canceled/confirmed via AMAD command.
    b.) This feature can be canceled/confirmed from MA Station. MA Station cannot set this feature.
    c.) The new setting from Mobile Phone/PCPro overwrites the existing setting. See the table below.

    • When Mobile Phone A has set Station B as its MA Station, Mobile Phone C tried to set Station B as its MA Station:

	Setting before overwriting		Setting after overwriting
Mobile Phone A	Setting with Station B		No Setting
Station B	Setting with Mobile Phone A	->	Setting with Mobile Phone C
Mobile Phone C	No Setting		Setting with Station B
Station D	No Setting		No Setting

- When Mobile Phone A has set Station B as its MA Station, Mobile Phone A tried to set Station D as its MA Station:

	Setting before overwriting		Setting after overwriting
Mobile Phone A	Setting with Station B		Setting with Station D
Station B	Setting with Mobile Phone A	->	No Setting
Mobile Phone C	No Setting		No Setting
Station D	No Setting		Setting with Mobile Phone A

- When Mobile Phone A has set Station B as its MA Station and Mobile Phone C has set Station D as its MA Station, Mobile Phone A tried to set Station D as its MA Station:

	Setting before overwriting		Setting after overwriting
Mobile Phone A	Setting with Station B		Setting with Station D
Station B	Setting with Mobile Phone A	->	No Setting
Mobile Phone C	Setting with Station D		No Setting
Station D	Setting with Mobile Phone C		Setting with Mobile Phone A

d.) This feature can be set/canceled only when MA Station is idle. This feature cannot be set/canceled when MA Station is busy or ringing, etc.
e.) This feature cannot be set/deleted from Mobile Phone and PCPro (AMAD command) during this feature call. Confirmation of this feature setting is available.
f.) This feature cannot be set with Uniform Numbering.
g.) Legacy Dterm, DtermIP and Analog telephone can be set as MA Station. Standard SIP terminal, PS, Attendant Console and ISDN terminal cannot be set.

36. The following shows the conditions concerning En-block Access.
a.) Only Station Number and outside line number can be used as Addition Number in En-block Access.
b.) The System determined whether or not the receiving call is En-block Access call by the presence or absence of Additional Number.
c.) En-block Access activates by receiving “Single Complete” information element.

		Single Complete Information Element	
		Yes	No
Additional Number	Yes	En-block Access	En-block Access by Inter Digit Timer time-out
	No	Automated Attendant/Remote Access to System	Automated Attendant/Remote Access to System

d.) Both Preset Dialing and Overlap Dialing are available in En-block Access.
e.) If the dialed Additional Number is not suited for the predetermined Numbering Plan, the number is handled as dead number or dead level.

f.) The form of the received Called Party Number must be one of the following:

sma-prefix + Station Number

sma-prefix + Number for placing a call to C.O. line

sma-prefix + Number for placing a call to TIE line

g.) Only the forms of Called Party Number and Called Party Number receiving supported by Euro-ISDN are available.
h.) Provider must add sma-prefix to Called Party Number automatically as Prefix by the special service setting of the domain.
i.) Even if a called party is in busy state, Call Back is not available.
j.) When ETSI ISDN ADDRESSING [E-32] Extended Addressing is enabled (ASYDL, SYS1, Index 1088, Bit 5=1), registering with the ACODL command is also required.
k.) The following tables show service feature availability for En-block Access.

Note: Access Codes cannot be used in En-block Access. The following features cannot be set/canceled from Mobile Phone. Any features other than the following are not available.

Multiple Line Terminal Features Availability

Code	Service Feature	Availability in En-block Access	Remarks
I-21	Internal Zone Paging	N	
O-2	Outgoing Trunk Queuing	N	
N-28	Name Display - System	N	User information is not displayed.
V-2	Voice Call	N	
C-5	Call Forwarding - All Calls	X	Call Forwarding set on Called Party is available.
D-11D	Do Not Disturb -D	X	Do Not Disturb set on Called Party is available.
E-1	Executive Right-of-Way	N	
C-1	Call Back	N	
C-3	Call Forwarding - Don't Answer	X	Call Forwarding set on Called Party is available.
C-2	Call Forwarding - Busy Line	X	Call Forwarding set on Called Party is available.

×: Available N: Not Available -: Not Applicable

Incoming Call Features Availability

Code	Service Feature	Availability in En-block Access	Remarks
D-10	Distinctive Ringing	X	Ringer Pattern for Station Call is provided.

×: Available N: Not Available -: Not Applicable

Attendant Console Features Availability

Code	Service Feature	Availability in En-block Access	Remarks
C-13	Call Waiting Lamp	X	Counted as “trunk call”.
I-2	Incoming Call Identification	N	Identified as “trunk call”.
C-9	Call Queueing	X	
I-6	Individual Attendant Access	X	
B-2	Busy Lamp Field - Flexible	N	Station engaged in this feature call is handled as “idle”.

×: Available N: Not Available -: Not Applicable

Station Features Availability

Code	Service Feature	Availability in En-block Access	Remarks
C-12 C-27	Call Waiting - Terminating DCall Waiting - Answer -D	N	
S-13	Step Call	N	
C-25	Call Forwarding - Intercept/Announcement	X	
C-28	Call Forwarding - All Calls - Outside	X	Call Forwarding set on Called Party is available.
C-74	Call Forwarding - Don't Answer - Outside	X	Call Forwarding set on Called Party is available.
C-60	Call Forwarding - Busy Line - Outside	X	Call Forwarding set on Called Party is available.
S-99	Split Call Forwarding	N	Handled as trunk incoming call.
-	Tone to be sent out when the handset is off-hook at the Station on which Call Forwarding - All Calls Service is set.	N	SPDT is not sent to Mobile Phone.

Code	Service Feature	Availability in En-block Access	Remarks
C-15	Class of Service - Individual	N	Handled as trunk incoming call (Restriction Class of MA Station is ignored).
-	EPN Facility Restriction	N	According to Restriction Classes of Incoming Trunk and Station.
S-7 S-8 S-9 S-138 S-139	Station Hunting - Circular Station Hunting - Secretarial Station Hunting - Terminal Station Hunting - Priority Station Hunting - Priority with Switch-back	X	Station Hunting set on Called Party is available.
B-5D B-8 B-11	Boss - Secretary Override -D Boss - Secretary - Message Waiting Lamp Control Boss - Secretary Transfer	X	Boss Station and Secretary Station cannot be used as MA Station.
C-69	Call Forwarding - All Calls - Announce-ment	X	

×: Available N: Not Available -: Not Applicable

37. The following shows the conditions concerning 2nd Calling.
    a.) 2nd Call is available only when the line of 1st Call is in busy state.
    b.) 1st Call hears no tone until 2nd Call is terminated to the destination party. Therefore, time period of no tone status becomes longer in case of overlap dialing.
    c.) Provider must add sma-prefix to Called Party Number automatically as Prefix by the special service setting of the domain.
    d.) Mobile Phone must support two-line calling.
    e.) The form of the received Called Party Number must be one of the following:

        sma-prefix + Station Number

        sma-prefix + Number for placing a call to C.O. line

        sma-prefix + Number for placing a call to TIE line

        Note: Available Station Number: Physical Station Number in Self office and Telephone Number. Uniform Number is not available.
        Available Number for placing a call to C.O. line: Number for placing a LCR call to ISDN (Preset dialing).

    f.) One of hooking, retuning to two-party calling and calling party switching is available in 2nd Call.
    g.) Only the 2nd Call from Euro-ISDN line is available.

h.) 2nd Calling is available only when 1st Call is engaged in a conversation or consultation hold. Also, Mobile Phone must be engaged in a conversation with one of the available parties in the following table.

Party On the Other End	Availability
Euro-ISDN (Self Office/via FCCS)	X
ISDN Trunk (Self Office/via FCCS)	X
Qsig Trunk (Self Office/via FCCS)	X
Dterm (Self Office/via FCCS)	X
DtermIP (Property protocol) (Self Office/via FCCS)	X
DtermIP(SIP) (Self Office/via FCCS)	X
Analog Station (Self Office/via FCCS)	X
CCIS Trunk (Self Office/via FCCS)	N
Analog Trunk	N
PHS (Self Office/via FCCS)	N
Attendant Console (Self Office/via FCCS)	N
Standard SIP terminal (Self Office/via FCCS)	N
ISDN terminal (Self Office/via FCCS)	N
Other Trunks (Self Office/via FCCS)	N
Other Stations (Self Office/via FCCS)	N

×: Available N: Not Available

i.) The following parties are available as destination of 2nd Call.

Destination Party	Availability
Euro-ISDN (Self Office/via FCCS)	X
ISDN Trunk (Self Office/via FCCS)	X
Qsig Trunk (Self Office/via FCCS)	X
Dterm (Self Office/via FCCS)	X
DtermIP (Property protocol) (Self Office/via FCCS)	X
DtermIP(SIP) (Self Office/via FCCS)	X
Analog Station (Self Office/via FCCS)	X
CCIS Trunk (Self Office/via FCCS)	N
Analog Trunk	N
PHS (Self Office/via FCCS)	N
Attendant Console (Self Office/via FCCS)	N
Standard SIP terminal (Self Office/via FCCS)	N
ISDN terminal (Self Office/via FCCS)	N
Other Trunks (Self Office/via FCCS)	N
Other Stations (Self Office/via FCCS)	N

×: Available N: Not Available

Note: Hooking by 2nd Calling under the invalid conditions will be ignored.

j.) The related terminal can only press the RELEASE/CONF key.

k.) If Mobile phone (intermediate party) releases a call while it is in Consultation Hold state, the following occur:

Before transfer destination party is rung: All calls are released

While transfer destination party is ringing: Blind Transfer

After transfer destination party answers: Call Transfer - All Calls

l.) If the number of digits of the dialed Additional Number (Station Number) is not enough to call the destination party, the Additional Number is discarded and DT (**Note 2**) is sent to the Calling Party. The Calling Party is prompted to redial the all digits of the required number.

Note 2: In this case, 2nd Call is not connected to Announcement Trunk. Also, when the destination party of 2nd Call is unused or busy, the call is not connected to Announcement Trunk.

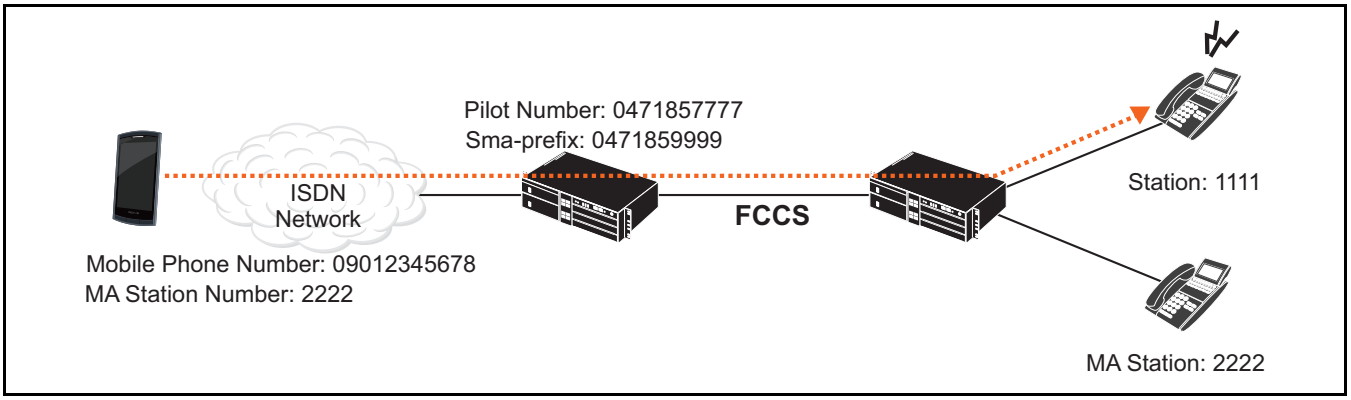
m.) The following tables show service feature availability for 2nd Calling.

Note: Access Codes cannot be used in 2nd Calling. Any features other than the following are not available.

Code	Service Feature	Availability in En-block Access	Remarks
C-11	Call Transfer - All Calls	X	
B-18	Blind Transfer To Station	X	
T-2	Three-Way Calling	N	

×: Available N: Not Available

38. This feature is available via FCCS network.



a.) In Refresh function, MA Control Node (MA_NCN) and MA Local Node (MA_LN) update the Mobile Number registered in Common Memory (CM) at designated time to unify the Mobile Numbers among all nodes.
Refresh process is activated at the designated time and MA_NCN and MA_LN operate as follows:
MA_NCN:
Update the Mobility Numbers in the self-common memory.
Broadcast the updated Mobility Numbers to each MA_LN.
Each MA_LN updates its common memory.
MA_LN:
Transfer the Mobility Numbers to MA_NCN.
MA_NCN receives the Mobility Numbers from each MA_LN and broadcasts the Mobility Numbers to each MA_LN after updating the self-common memory.

b.) MA Control Node (MA_NCN) is designated by the following system data: ASYDN, System Data 1, Index 1058.

c.) Call origination/termination via this feature is available even while the communication with Control Node is interrupted.

d.) Assignment/deletion of this feature data is not available while the communication with Control Node is interrupted.

e.) For more details on the service conditions for FCCS network, see Data Programming Manual - FCCS.
f.) It takes approximately 90 seconds to update MA data after the system starts operating (depending on the load condition on the network.). MA feature cannot be activated until the update of MA data is completed.
g.) A maximum of 64 Mobility Numbers can be transferred/broadcast.
h.) If an error occurs while Mobility Numbers are being transferred from MA Local Node (MA_LN) to MA Control Node (MA_NCN), Mobility Numbers are re-transferred (until the transfer is completed successfully).
i.) This feature does not support Location Diversity [L-55].

39. The following are conditions only when Euro-ISDN trunk used as this feature is intermediate party. For other conditions, see the conditions of Three-Way Calling.
    a.) Three-Way Calling is available only when the CONF key on Desktop terminal is pressed.
    b.) Only the Dterm can establish a Three-Way Calling.
    c.) Three-Way Calling cannot be established when Conference Equipment, PS or ISDN terminal is included.
40. When ETSI ISDN ADDRESSING [E-32] is enabled, set the total number of trunk access code and international/national prefix number to six digits or less. If this total number exceeds seven digits, this feature is not available.

Note: This condition is valid not only when Extended Addressing (ASYDL, SYS1, Index 1088, Bit 5=1) is enabled, but also when Existing Addressing (ASYDL, SYS1, Index 1088, Bit 5=0) is enabled.

Programming

To use this feature, the data assignments of ETSI ISDN, Automated Attendant [A-82]/Remote Access to System [R-2], Station ID Code and Mobility Access are required.

Data Assignment for ETSI ISDN:

See ETSI ISDN DATA ASSIGNMENT in the Data Programming Manual - ISDN.

Data Assignment for Automated Attendant/Remote Access to System:

See Automated Attendant [A-82]/Remote Access to System [R-2].

Data Assignment for Station ID Code:

The following shows how to assign the Station ID Code data.

STEP 1: **ANPD** - Assign the minimum necessary number of digits for the first number of Station ID Number Entry Access Code.

TN:	Tenant Number
1stDC:	The first digit of service Access Code. (0-9, *, #)
CI:	Select Normal Service (N) from the combo box for Connection Indexes.
NND:	Number of Necessary Digits for the Access Code (up to 6 digits)
BLF:	Busy Lamp Field Display Activated on ATT (0/1 = Out of Service/In Service)

STEP 2: **ASPA** - Assign Station ID Number Entry Access Code.

TN:	Tenant Number
ACC:	Access Code for this feature (up to 6 digits)
CI:	Connection status Index = “N: Normal”
SRV:	Kind of Service = “SSCA: Service Code Appendix”
SIDA:	Service Index A = 100 (ID Number Entry)

STEP 3: **ARAC** - Assign the Remote Call Forwarding Control Access Code.

ICTN:	Tenant Number of Incoming Trunk
ACC:	Access Code (up to 5 digits)
R/M:	M (Manual Roaming)

STEP 4: **ASFC** - Assign a Service Feature Restriction Class that allows SFI 143 to Stations that will use the Station ID Number Entry Access Code.

Data Assignment for Mobility Access:

The following shows how to assign the Mobility Access data.

STEP 1: **ANPD** - Assign the minimum necessary number of digits for the first number of Mobility Access Entry and Cancel Codes.

TN: Tenant Number
1stDC: The first digit of service Access Code. (0-9, *, #)
CI: Select Normal Service (N) from the combo box for Connection Indexes.
NND: Number of Necessary Digits for the Access Code (up to 6 digits)
BLF: Busy Lamp Field Display Activated on ATT (0/1 = Out of Service/In Service)

STEP 2: **ASPA** - Assign Mobility Access Entry and Cancel Codes.

(Mobility Access Entry Code)
TN: Tenant Number
ACC: Access Code for this feature (up to 6 digits)
CI: Connection status Index = "N: Normal"
SRV: Kind of Service = "SSCA: Service Code Appendix"
SIDA: Service Index A = 153 (Mobility Access Entry)

(Mobility Access Cancel Code)
TN: Tenant Number
ACC: Access Code for this feature (up to 6 digits)
CI: Connection status Index = "N: Normal"
SRV: Kind of Service = "SSCA: Service Code Appendix"
SIDA: Service Index A = 154 (Mobility Access Cancel)

STEP 3: **ASFC** - Assign a Service Feature Restriction Class that allows SFI 234 to Stations that will use Mobility Access feature.

STEP 4: **AKYD** - Assign the key data for MOBILITY ACCESS on the Line/Feature key of Desktop terminal.

KYN: Key Number (1-40)
KYI: 1 (Function Key)
FKY: 167 (Mobility Access)

STEP 5: **ADSL** - Assign SN = 1 (Originating; DT Connection) and FKY = 167 (Mobility Access). This command needs to be assigned only when using Soft key.

Data Assignment for Call Back-Out of Zone:

Assign the following Call Back-Out of Zone data if required.

STEP 1: **ARTD** - For the route used for call origination, assign CDN 111: ADVPRA ISDN PRI Failure Routing Service, as data "1".

Note: To change the data for existing route (RT), initialize or reboot the trunk device such as a circuit card or MG.

STEP 2: **ASFC** - For Stations allowed Call Back-Out of Zone, assign a Service Feature Restriction Class that allows SFI 236.

Data Assignment for En-block Access/2nd Call:

- Data Assignment for Automated Attendant/Remote Access to System

  See Automated Attendant [A-82]/Remote Access to System [R-2].

- Data Assignment for Called Party Number Digit

STEP 1: **ACODL** - Assign Country Code and Area Code, etc. per route. For more details, see ETSI ISDN Addressing [E-32] in Data Programming Manual - ISDN.

STEP 2: **ARTI/ARTIN** - (Assignment of Trunk Application Data)

CDN: 92 (EBDC) - Assign the number of digits for Called Party Number (except the number (Area Code and Local Code) assigned by ACODL command) per route.

Note: To change the data for existing route (RT), initialize or reboot the trunk device such as circuit card, MG, or VS32.

Mobility Access [M-118] via FCCS

This feature is available via FCCS network. To activate this feature via FCCS link, the following data assignment is required.

STEP 1: **ASYDN** - System Data 1, Index 1058: Assign FPC (1-253) of the Control Node (01 Hex- FD Hex).
System Data 1, Index 1059: Assign CM data periodic Refresh Time (Hour): 01-24 (using a decimal number)
System Data 1, Index 1060: Assign CM data periodic Refresh Time (Minute): 00-59 (using a decimal number)

Confirming/Setting/Canceling MA data from PCPro

The following shows how to confirm/set/cancel the MA data from the PCPro.

STEP 1: **AMAD** (Assignment of Mobility Access Data)

Note: Even if an incorrect MA Number is assigned, no error message is indicated (the assigned command data can be written into the system). However, in this case, this feature does not work properly.

- When TYPE= 1.Telephone Number(NDM)

  UGN: User Group Number (1-255)
  TELN: Telephone Number of MA Station (Up to 16 digits)
  CFDM: Call Forwarding - Mobility Access Destination Number including Trunk Access Code [Up to 24 digits (0-9)]
  TACC: The number of digits for Trunk Access Code (1-6)
  MA Number: Mobile Phone Number not including Trunk Access Code is displayed.

- When TYPE= 2.Telephone Number(LDM)

  FPC: Fusion Point Code (1-253) **Note 3**
  UGN: User Group Number (1-255)
  TELN: Telephone Number of MA Station (Up to 16 digits)
  CFDM: Call Forwarding - Mobility Access Destination Number including Trunk Access Code [Up to 24 digits (0-9)]
  TACC: The number of digits for Trunk Access Code (1-6)
  MA Number: Mobile Phone Number not including Trunk Access Code is displayed.

- When TYPE= 3.Station Number

  FPC: Fusion Point Code (1-253) **Note 3**
  TN: Tenant Number
  STN: Station Number of MA Station
  CFDM: Call Forwarding - Mobility Access Destination Number including Trunk Access Code [Up to 24 digits (0-9)]
  TACC: The number of digits for Trunk Access Code (1-6)
  MA Number: Mobile Phone Number not including Trunk Access Code is displayed.

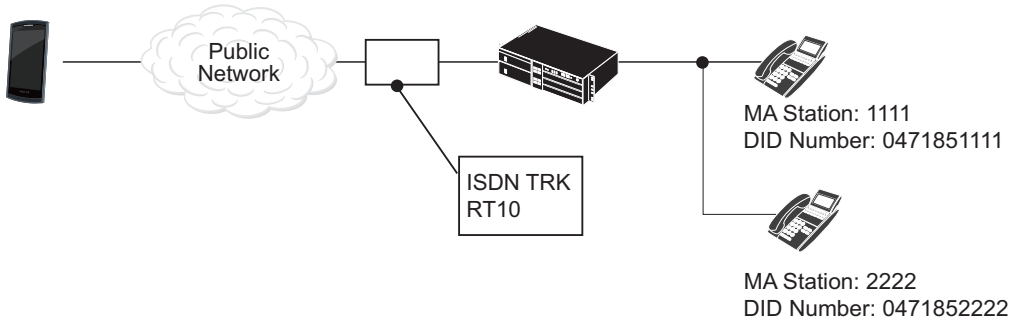
Note 3: This parameter is not displayed when Fusion Point Code (FPC) has not been assigned via System Data.

Direct Inward Dialing (DID)

STEP 1: **ARTD** - Assign the following data to the B-channel of DID line.

- Assign the number of digits to be converted into Station Number to the parameter DC (CDN=66). The number of digits for this DC value must correspond to the DC of ACDD command. The number of digits is counted from the last digit of the received DID number.
- Assign the Basic Data for ISDN. Refer to CHAPTER 3 BASIC DATA ASSIGNMENT FOR ISDNS in Data Programming Manual - ISDN.

Ex.)



Assign DC=4 to terminate a DID call (DID number: 0471-85-2222) to Station 1111. The last four digits for the received number are converted into Station Number.

Note: To change the data for existing route (RT), initialize or reboot the trunk device such as a circuit card or MG.

STEP 2: **ACDD** - Assign the digits of the received DID number to be converted into Station Number.

Ex.) To terminate a DID call (DID number: 0471-85-1111) to Station Number 1111:

D/N = D, RT=10, DC=1111, AD = 0, XFR = 0

SID To Network-Present

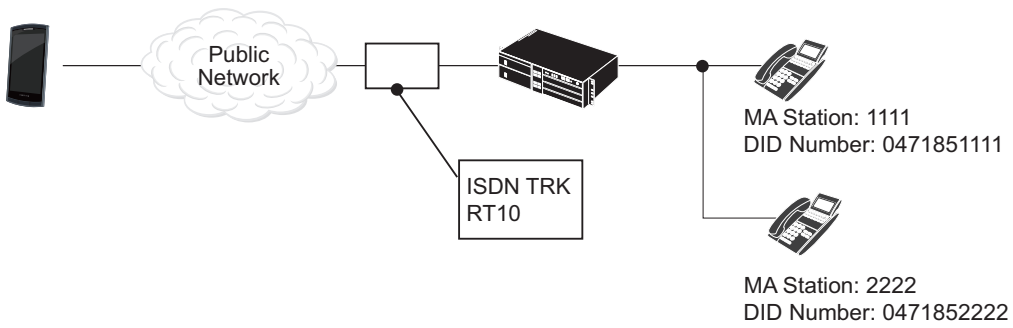
Note: When two or more DID numbers are assigned to a station, the last DID number assigned will be delivered to the ISDN as SID (Station Identification Number). However, when the last DID number assigned has been deleted (even if the other number(s) is/are still in existence), any DID number is not transferred to the network (no SID is transferred to a destination). In this case, reassignment of the DID number(s) is necessary for sending SID to the network.

Note: If using Logical Route Number, assign the data by ACNPN and ACNDN command.

STEP 1: **ACNP** - Assign the Calling Number Pattern (CNP) to the route to be provided SID to Network-Present.
IC/OG = OG: Outgoing Call
RT: External Route Number used for SID to Network -Present
CNP: Calling Number Pattern
Enter a number within the range between 1 and 255.

STEP 2: **ACND** - Assign the following data to Calling Number Pattern (CNP) assigned in ACNP command.
CNP: Calling Number Pattern assigned in ACNP command.
SKIP: Number of Skip Digits from the SID.
ADD: Number of Digits of the Additional Number.
DC: Additional Digit Information

Ex.)



To send DID number (0471851111) of Station 1111 as Calling Party Number:
ACNP: OG: Outgoing Call, RT=10, CNP=2
ACND: CNP=2, SKIP=4, ADD=6, DC=0471851111

Change the destination of Call Back-Out of Zone

ASYDL - System Data 1, Index 1088, Bit 1: Specify the destination of Call Back-Out of Zone. 0/1 = MA Station/Destination Station of Call Forwarding - Busy Line set to MA Station.

Change the priority of MA Call Forwarding

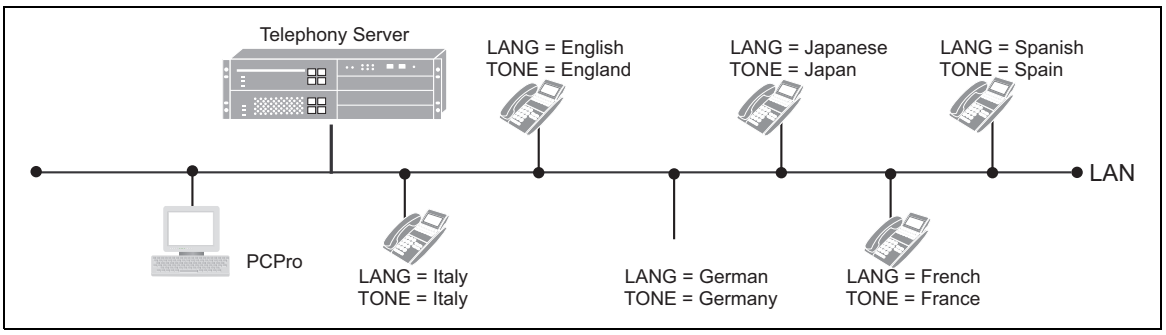
ASFC - Assign Service Feature Restriction Class that allows SFI 240 (Change of the priority for MA Call Forwarding) to the Stations activating this service.

RES: 0 (Logged Out IP Station - Call Destination and Call Forwarding - All Calls > MA Call Forwarding)/1(MA Call Forwarding > Logged Out IP Station - Call Destination and Call Forwarding - All Calls)

M-119D Multiple Tone -D

General Description

This feature can specify the Country Code for tone setting. Country Code can be assigned per terminal by PCPro command.



Note: For information of Language Setting (LANG), refer to Multilingual Display [M-117].

Following shows the list of Country Code for tone setting.

Country Code		Country
Country Code for Basic Tone Setting	1	Japan
	2	North America
	3	Australia **Note 1**
	4	Other Countries
	5	Hong Kong
	6	Malaysia
	7	Singapore
	8	England
	9	Mexico
	10	Taiwan
	11	New Zealand
	12	Korea
	13	Brazil
	14	China
	15	Thailand
	16	Thailand (EGAT)
	17	Sri Lanka
	18-199	Not used

Country Code		Country
Country Code for EMEA Tone Setting	200	Germany
	201	Italy
	202	Netherlands
	203	Not used
	204	Denmark
	205	Sweden
	206-210	Not used
	211	Spain
	212	Austria
	213	Belgium
	214	England [For EMEA]
	215	Greece
	216	Switzerland
	217	South Africa
	218	Brazil
	219-255	Not used

Note 1: An Australian tone cannot be selected by the Multiple Tone function.

Operating Procedure

None

Service Conditions

1. When the tone setting is changed by the PCPro, initialization is executed for the changed terminals. And the data setting will be reflected.

   - Timing of Terminal Initialization by terminal status

Terminal Status	Terminal Initialization Timing
Idle	Executed when the tone data is changed.
Talking/Ringing	Executed when the call is finished.
During Call Origination/ Termination	If the terminal status changes to Talking or Ringing, the terminal is re-set at the end of the call.
	If the attempted call is abandoned at the time of call setup, the terminal is reset at the end of the next call.

- The effects after Terminal Initialization

Message Reminder	Message data will be deleted. (Lamp off)
REDIAL key on the Desktop terminal	Redial data will be deleted.

2. This feature is available for the following terminals.
   - DtermIP (Proprietary Protocol)
   - DtermIP (SIP) Version. 4.0.0.0
   - DT800 Series/DT700 Series
3. Soft Phone is not supported.
4. On MC-MG, Tone Setting should be assigned by the system data [ASYDL - Index 811].
5. When Configuration Setup is available on the terminal, specify Area ID as "Automatic". For Configuration Setup, refer to the Peripheral Equipment Description.
6. When Area ID is specified as "Automatic" and tone setting is not assigned, the tone setting will be specified as "Japan".
7. Available tone depends on the firmware and version of the terminal.
8. By assigning ASYDL/N, System Data 1, Index 677, Bit 5 = 1, this feature can be inactivated. System Initialization is necessary for changing this system data.
9. This service does not support the system to which Australian Tone for O/G Connection (ASYD, System Data 1, Index 6, Bit 3=1) is set.
10. This feature is not available for SR-MGC.

Programming

STEP 1: **ASYDL** - System Data 1, Index 677, Bit 5 (Multiple Tone -D). 0/1 = In service/Out of service.

STEP 2: **ASLG** - Specify the Tone Data used per system.

- **[FIRST] tab -** Basic Data Setting

  System Tone Data
  Enter the Country Code as Tone Number for Tone Setting. For the Country Code, refer to the General Description in this feature.

- **[TERMINAL] tab -** Terminal Data Setting

  Enter the range of the Tenant Numbers and Station Numbers, then specify the tone used per terminal.
  Line Number Type: Specify the radio button for Line Number Type: "STN" or "TELN(LDM)"
  Change Data Type: check the check box for "TONE".

◆ When "STN" is selected in [Line Number Type] field.

START/END TN:	Enter the appropriate range of Tenant Numbers.
START/END STN:	Enter the appropriate range of Station Numbers.
KIND:	Specify terminal kind.
TONE:	Enter the Country Code as Tone Number for Tone Setting. For the Country Code, refer to the General Description in this feature.

◆ When "TELN(LDM)" is selected in [Line Number Type] field.

UGN:	User Group Number
TELN START/END:	Enter the appropriate range of Telephone Numbers.
KIND:	Specify terminal kind.
TONE:	Enter the Country Code as Tone Number for Tone Setting. For the Country Code, refer to the General Description in this feature.

FCCS Programming

STEP 1: **ASYDN** - System Data 1, Index 677, Bit 5 (Multiple Tone -D). 0/1 = In service/Out of service.

STEP 2: **ASLG** - Specify the Tone Data used per system.

- **[FIRST] tab -** Basic Data Setting

  System Tone Data
  Enter the Country Code as Tone Number for Tone Setting. For the Country Code, refer to the General Description in this feature.

- **[TERMINAL] tab -** Terminal Data Setting

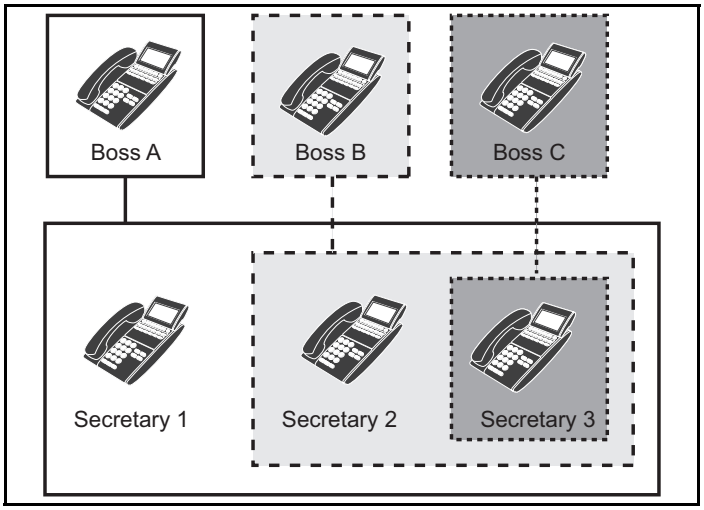
  Enter the range of the Tenant Numbers and Station Numbers, then specify the tone used per terminal.

Line Number Type:	Specify "TELN(NDM)" radio button for Line Number Type.
Change Data Type:	Check the check box for "TONE".
FPC:	Fusion Point Code
UGN:	User Group Number
TELN START/END:	Enter the appropriate range of Telephone Numbers.
KIND:	Specify terminal kind.
TONE:	Enter the Country Code as Tone Number for Tone Setting. For the Country Code, refer to the General Description in this feature.

M-120 Multi-Boss/Multi-Secretary

General Description

This feature enables multiple Secretaries to serve duties for one Boss as well as for one Secretary to serve duties for multiple Bosses.



See the following table for details on features.

Feature	Descriptions	Available for Market
BOSS ABSENT	When the Boss' Station has BOSS ABSENT set, all incoming calls for Boss' Station will arrive at the Secretary's Station by Slide Action. BOSS ABSENT can be set/canceled from a Secretary's Station.	All Countries
SECRETARY GROUPING	A Secretary's Station may belong to multiple Secretary Groups. Within a Secretary Group, Slide Action takes place.	All countries
SECRETARY ABSENT	If a Secretary is away from his or her desk, SECRETARY ABSENT can be set to that Secretary Station. During SECRETARY ABSENT, the Secretary Station gets skipped from Slide Action by all the Secretary Groups it belongs to.	All countries
ENTIRE SECRETARY GROUP IS BUSY	When the entire Secretary Group is busy, busy line features are assigned to Secretary 1 instead of the Boss' Station. Also, when the entire Secretary Group is busy, UCD Queuing is available through data setting.	All countries

Feature	Descriptions	Available for Market

MULTI-BOSS/MULTI-SECRETARY-ONE TOUCH	• Assigning a Sub Line as Boss' Line enables the Secretary's Station to display the Boss' status on the Boss' Line key.
	• Use of the Boss' Line key enables the Secretary to perform the following operations:
	  • Calling the Boss
	  • Answering the Boss' Call
	  • Overriding to the Boss' line
	  • Setting Call Waiting to the Boss' Station.
		All countries

MULTI-BOSS/MULTI-SECRETARY AWAY PER BOSS	A Secretary's Station can temporarily leave its Secretary Group	All countries
GROUP NUMBER EXPANSION	The maximum number of Boss Secretary Groups that can be registered by ASHUEN per a node can be expanded from 63 to 255.	All countries
MULTI-BOSS/MULTI-SECRETARY OVERRIDING FROM A SECRETARY'S CALL FORWARDING DESTINATION	When a Boss' Station sets BOSS ABSENT and a Secretary's Station sets Call Forwarding features (**Note 1**, **Note 2**), an incoming call to Boss' Station is forwarded to the forwarding destination. The forwarding destination can place a call to the Boss' Station by Overriding.	EMEA only
MULTI-BOSS/MULTI-SECRETARY BOSS' CALL FORWARDING ACTIVATE	When a Boss' Station sets Call Forwarding features (**Note 3**) and BOSS ABSENT, the following operations are available. • An incoming call to the Boss' Station slides to a Secretary's Station. • When the Secretary's Station performs switch hook flash and dials the Boss' Station number after receiving the call with the Slide Action, the call is forwarded to a forwarding destination which the Boss' Station sets. • When the Secretary's Station sets Call Forwarding features (**Note 1**, **Note 2**), the call is forwarded to a forwarding destination which the Secretary's Station sets. When the forwarding destination performs switch hook flash and dials Boss' Station number after receiving the call, the call is forwarded to a forwarding destination which the Boss' Station sets.	EMEA only
MULTI-BOSS/MULTI-SECRETARY OVERRIDING FROM A STAND-IN SECRETARY	A Stand-In Secretary is a term referring to a forwarding destination of CALL FORWARDING - ALL CALLS [C-5]/CALL FORWARDING - DON'T ANSWER [C-3] which a Secretary sets. When a Boss' Station has BOSS ABSENT set, the Stand-In Secretary's Station can place a call to the Boss' Station by Overriding. Whether a call is forwarded from the Secretary's Station or not, the Stand-In Secretary's Station can make a call to the Boss' Station which has BOSS ABSENT set.	EMEA only

Feature	Descriptions	Available for Market
MULTI-BOSS/MULTI-SECRETARY MULTI LINE OPERATION	In the case that a Boss (or a Secretary) has two stations, a main station and a substitute station, when the main station line is assigned to a Sub Line of the substitute station, the substitute station can be operated as well as the main station by seizing the main line from the substitute station even if the substitute line does not belong to a Boss Secretary Group.	All countries

Note 1: Call Forwarding features set by the Secretary's Station mentioned here are:
- CALL FORWARDING - ALL CALLS [C-5]
- CALL FORWARDING - DON'T ANSWER [C-3]
- CALL FORWARDING - ALL CALLS - OUTSIDE [C-28]
- CALL FORWARDING - DON'T ANSWER - OUTSIDE [C-74]

Note 2: CALL FORWARDING - ALL CALLS - OUTSIDE [C-28]/CALL FORWARDING - DON'T ANSWER - OUTSIDE [C-74] can be also set but the forwarding destination cannot place a call to the Boss' Station by Overriding.

Note 3: Call Forwarding features set by the Boss' Station mentioned here are:
- CALL FORWARDING - ALL CALLS [C-5]
- CALL FORWARDING - DON'T ANSWER [C-3]
- CALL FORWARDING - ALL CALLS - OUTSIDE [C-28]

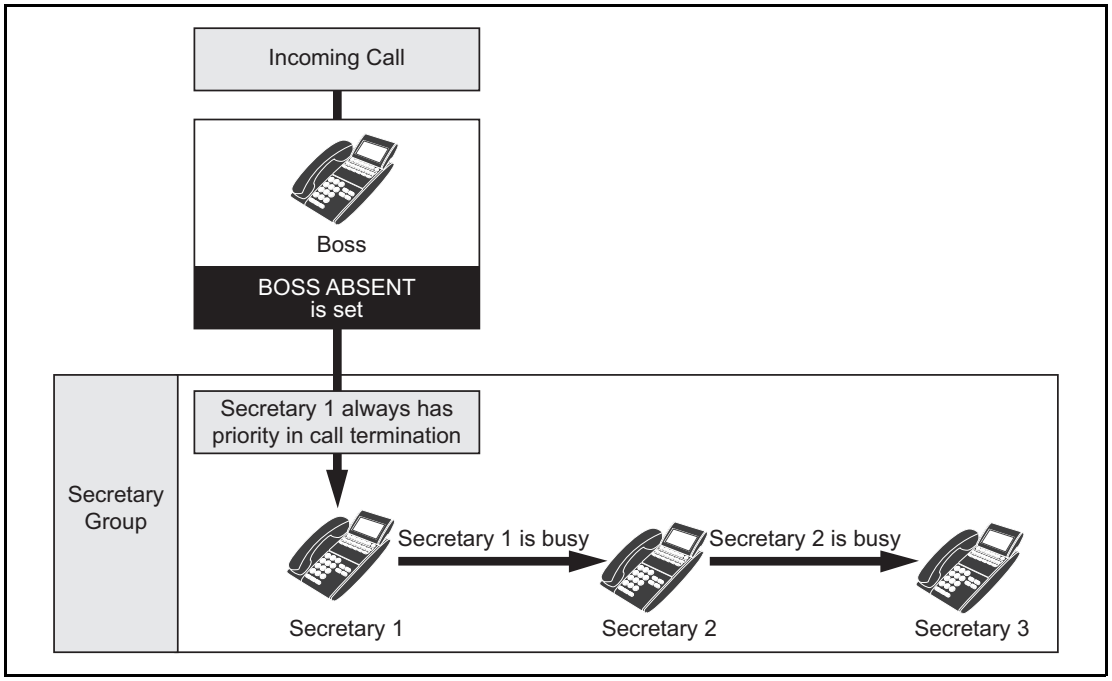
CALL FORWARDING - DON'T ANSWER - OUTSIDE [C-74] cannot be used.

1. BOSS ABSENT

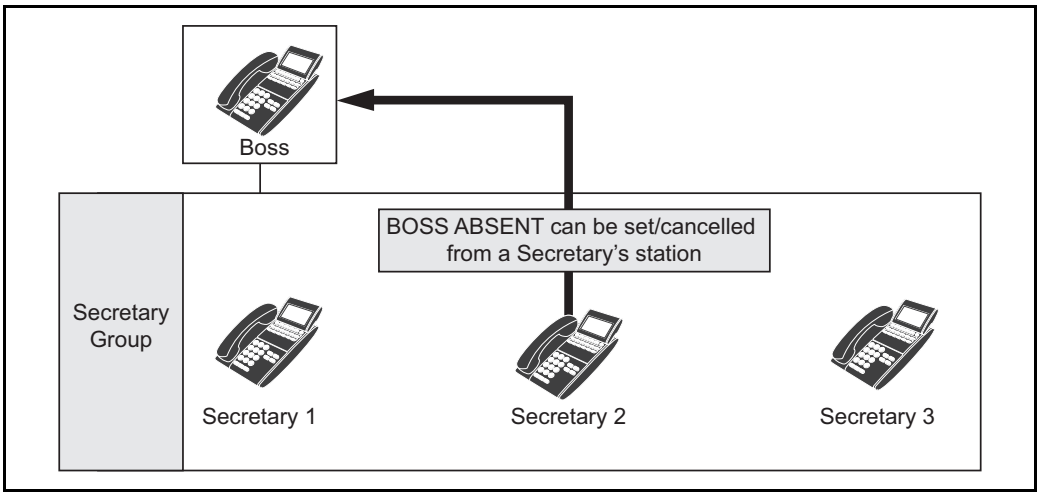
When the Boss' Station has BOSS ABSENT set, all incoming calls for Boss' Station will arrive at the Secretary's Station by Slide Action.

Note: Available for all countries.

Note: Slide Action searches for an idle Secretary's Station where call termination is possible. The priority of the search is as follows: Secretary1→Secretary2→Secretary3. Priority of Slide Action is assigned in the order in which the Secretary's Station was registered through ASHUEN command.



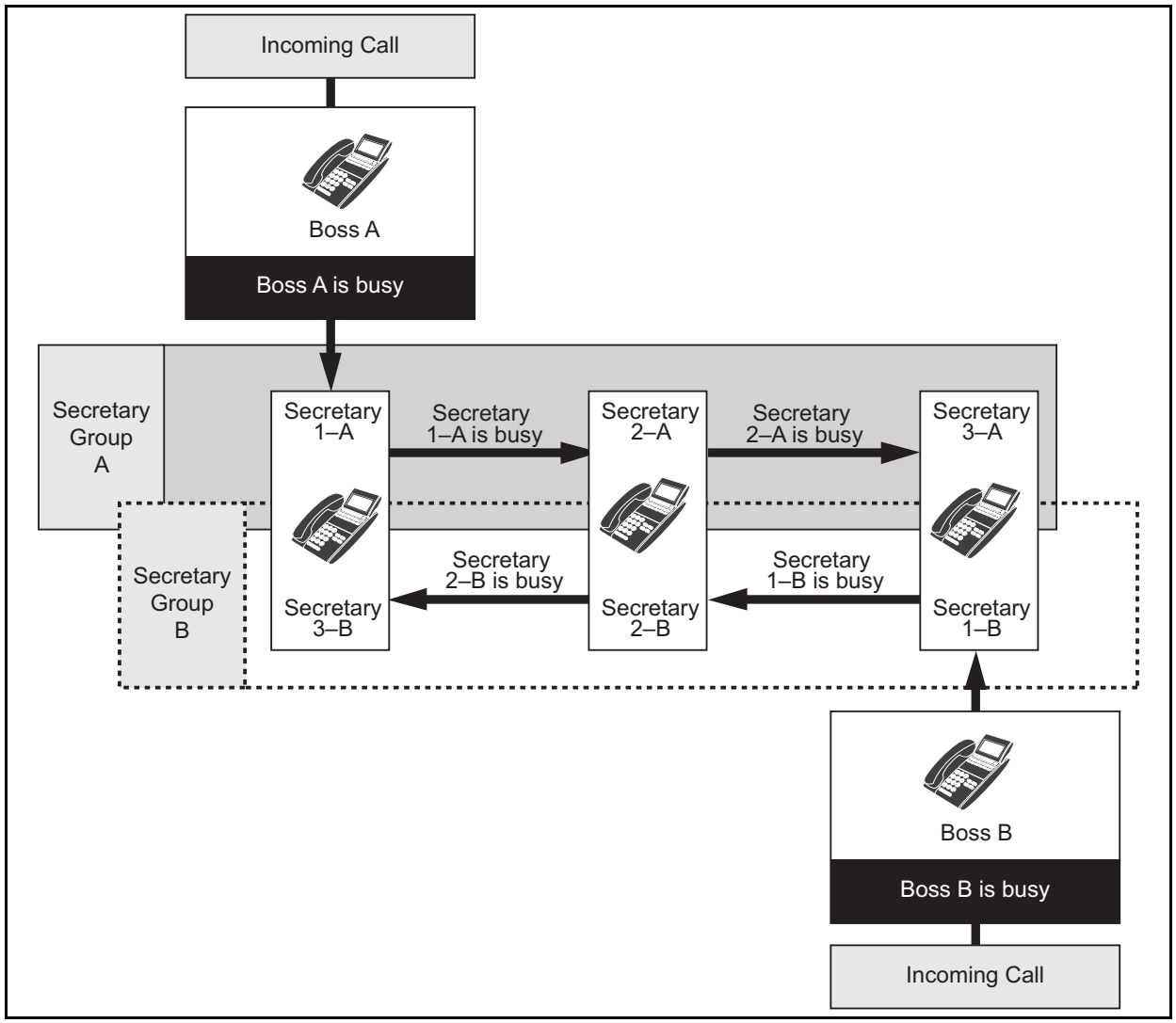
BOSS ABSENT can be set/canceled from a Secretary's Station.



2. SECRETARY GROUPING

A Secretary's Station may belong to multiple Secretary Groups. Within a Secretary Group, Slide Action takes place.

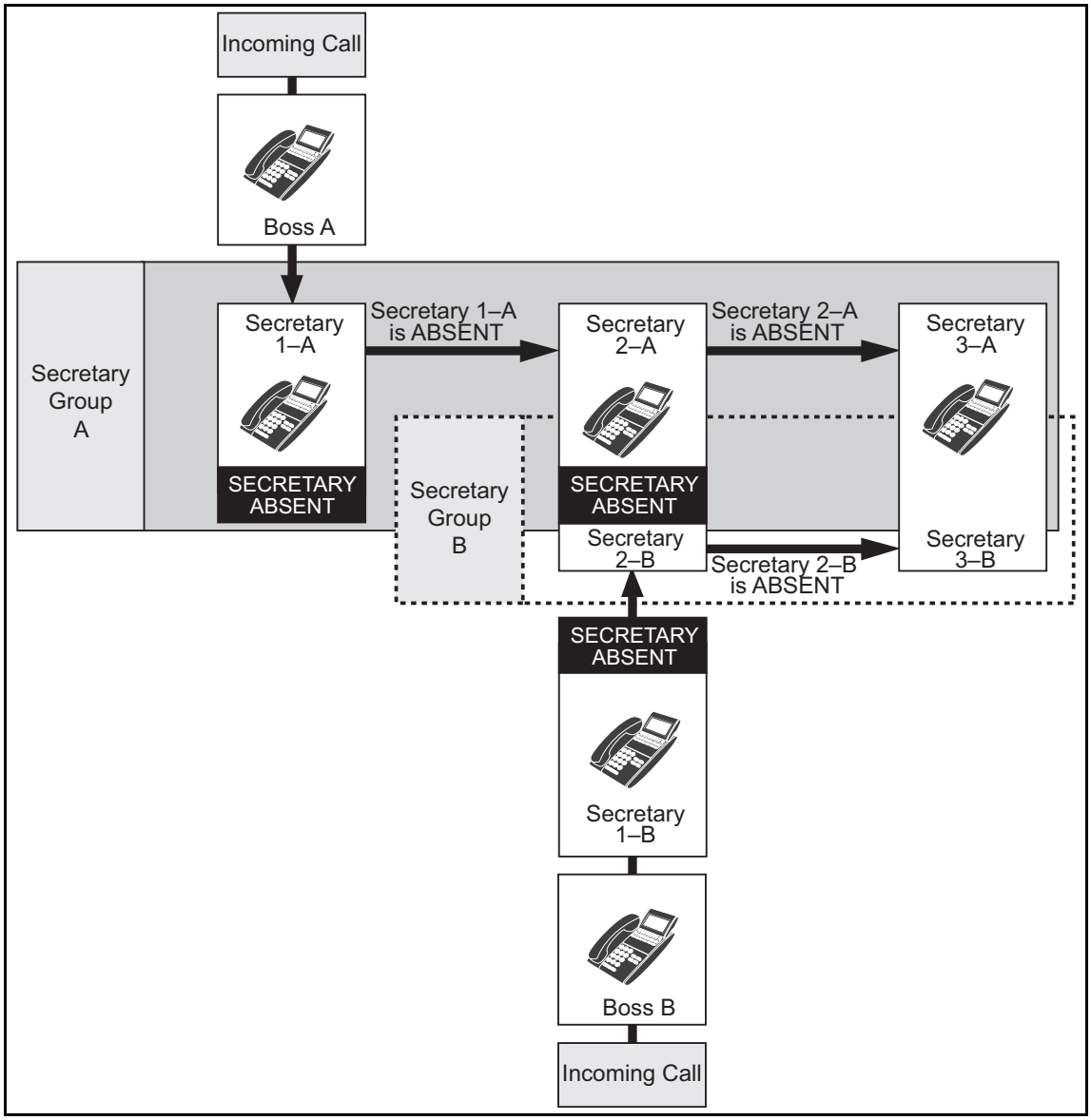
Note: Available for all countries.



3. SECRETARY ABSENT

If a Secretary is away from his or her desk, SECRETARY ABSENT can be set to that Secretary Station. During SECRETARY ABSENT, the Secretary Station gets skipped from Slide Action by all the Secretary Groups it belongs to.

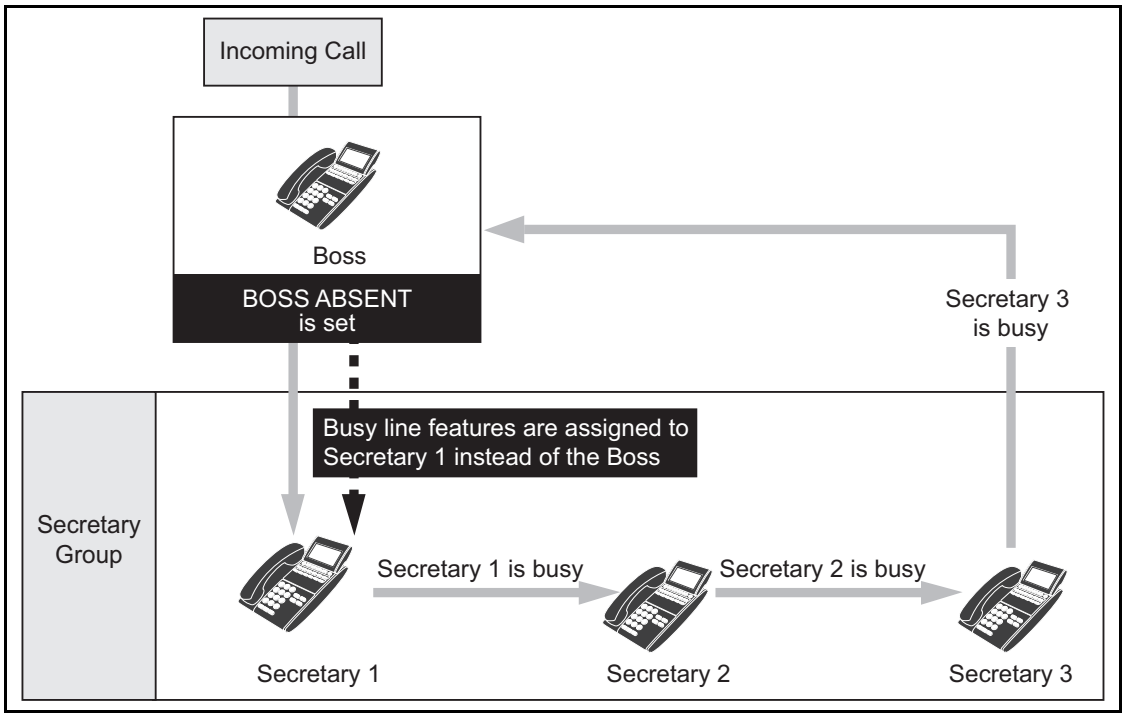
Note: Available for all countries.



4. ENTIRE SECRETARY GROUP IS BUSY

When the entire Secretary Group is busy, busy line features are assigned to Secretary 1 instead of the Boss' Station. Also, when the entire Secretary Group is busy, UCD Queuing is available through data setting.

Note: Available for all countries.

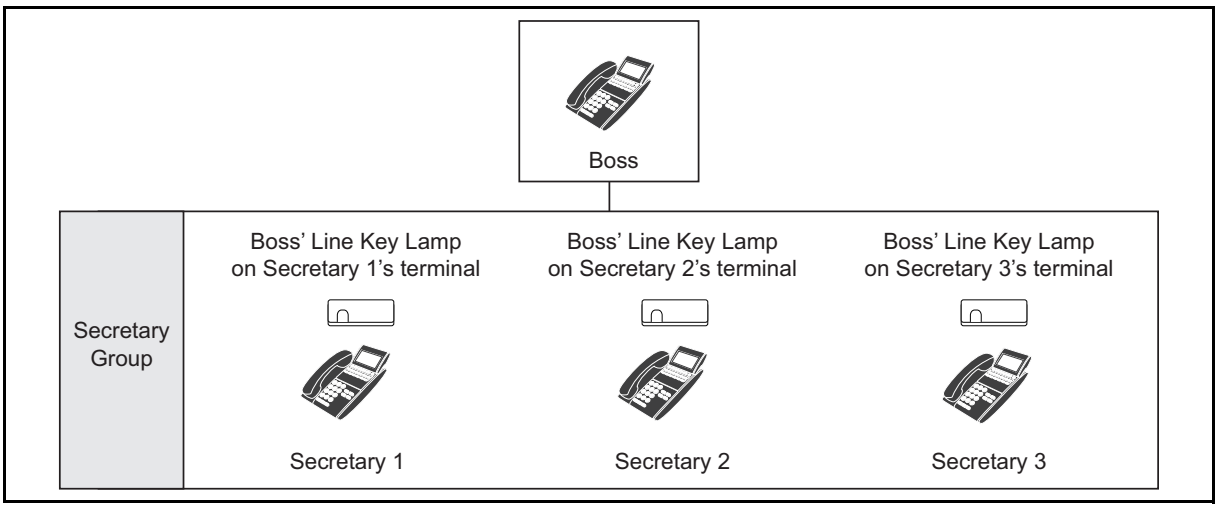


5. MULTI-BOSS/MULTI-SECRETARY-ONE TOUCH

Note: Available for all countries.

5.1 Status Display of Boss' Terminal

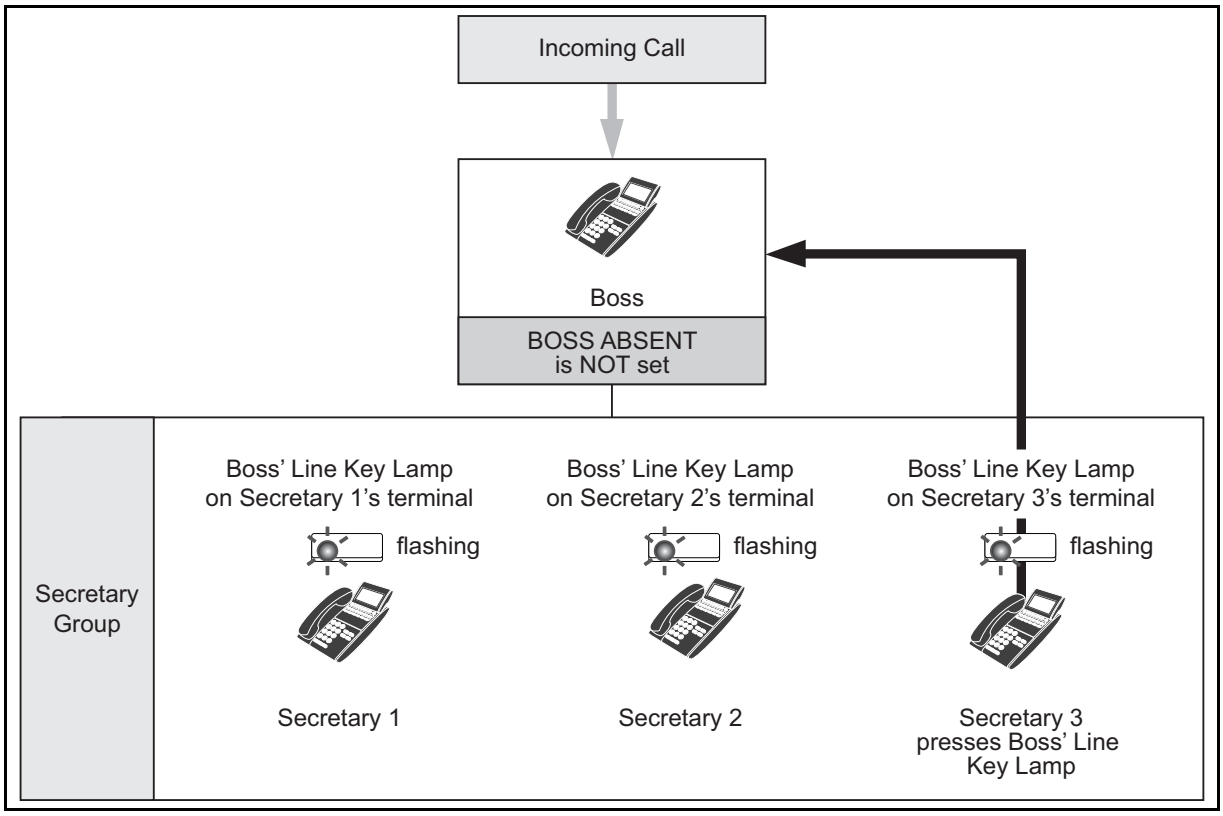
Assigning Boss' Line key as a Sub Line on the Secretary's Terminal enables status display of Boss' Terminal.



Status of Boss' Terminal	Status of Boss' Line Key Lamp on Secretary's Terminal
Busy	Light On
Idle	Light Off
Receiving a call	Flashing

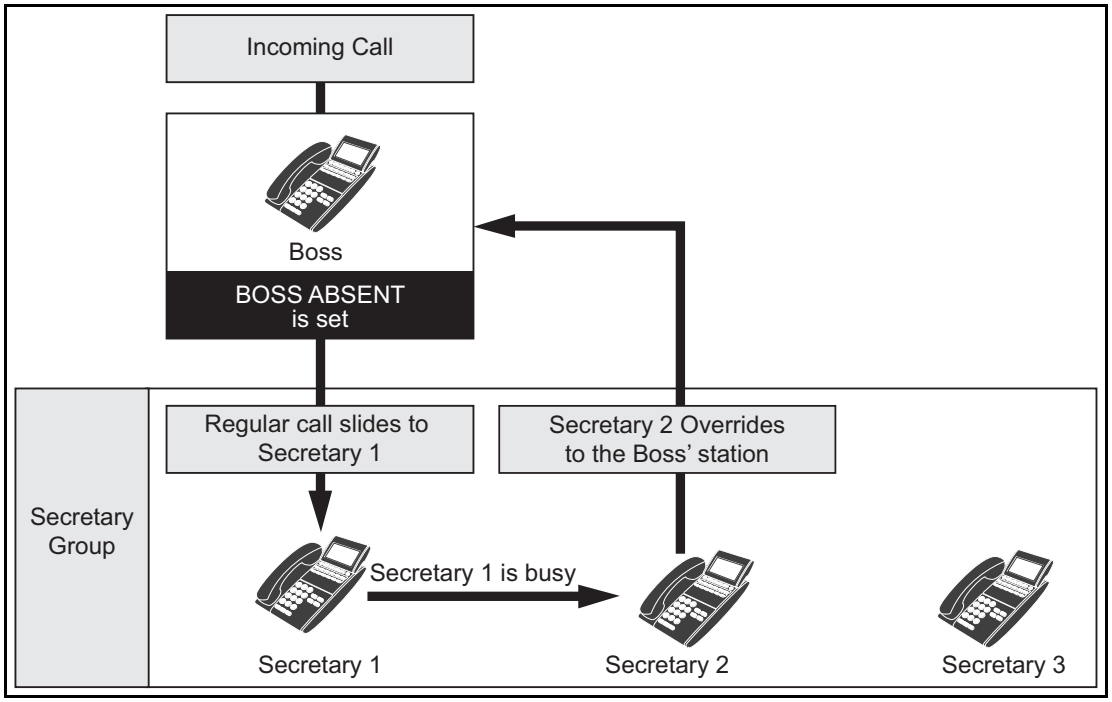
5.2 Secretary's Terminal Answering Boss' Line Call

While the Boss' Terminal is receiving a call, the Secretary is able to answer that call by pressing the flashing Boss' Line key lamp.



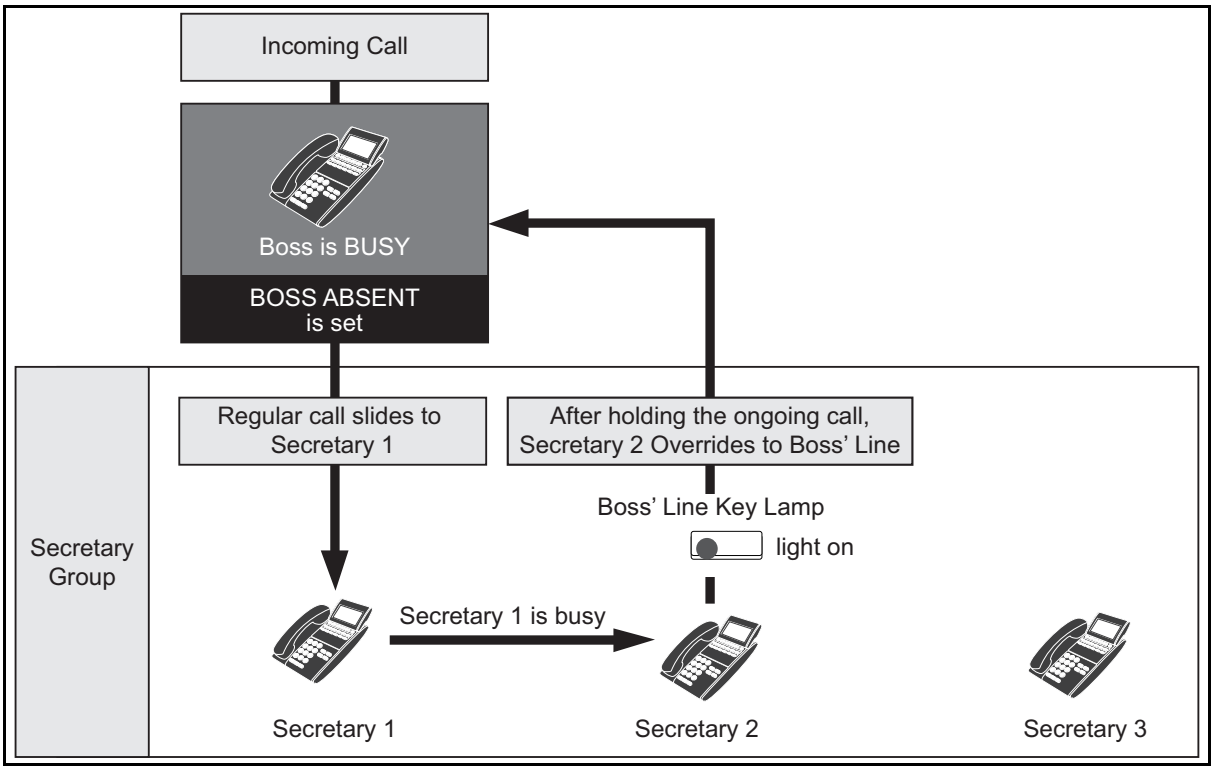
5.3 Overriding from a Secretary's Station

When the Boss' Station has BOSS ABSENT set, all incoming calls to the Boss' Station will arrive at the Secretary's Station by Slide Action. The Secretary's Station can place a call to the Boss' Station by Overriding.



5.4 Overriding from a Busy Secretary's Station

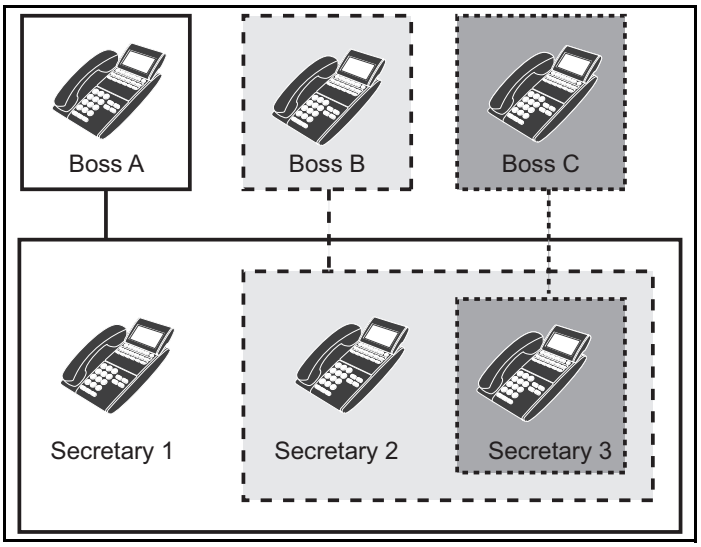
When the Secretary's Terminal is busy, Overriding to the Boss' Station is possible after holding the ongoing call using Call Park.



6. MULTI-BOSS/MULTI-SECRETARY AWAY PER BOSS

This feature enables a Secretary's Station to temporarily leaves its Secretary Group. In other words, a Secretary's Station with this feature assigned will be skipped from call termination.

Note: Available for all countries.



	Secretary 1	Secretary 2	Secretary 3
Boss A	Secretary 1	Secretary 2	Secretary 3
Boss B	Secretary 2	Secretary 3	—
Boss C	Secretary 3	—	—

Secretary 2 sets AWAY PER BOSS from Boss A's group

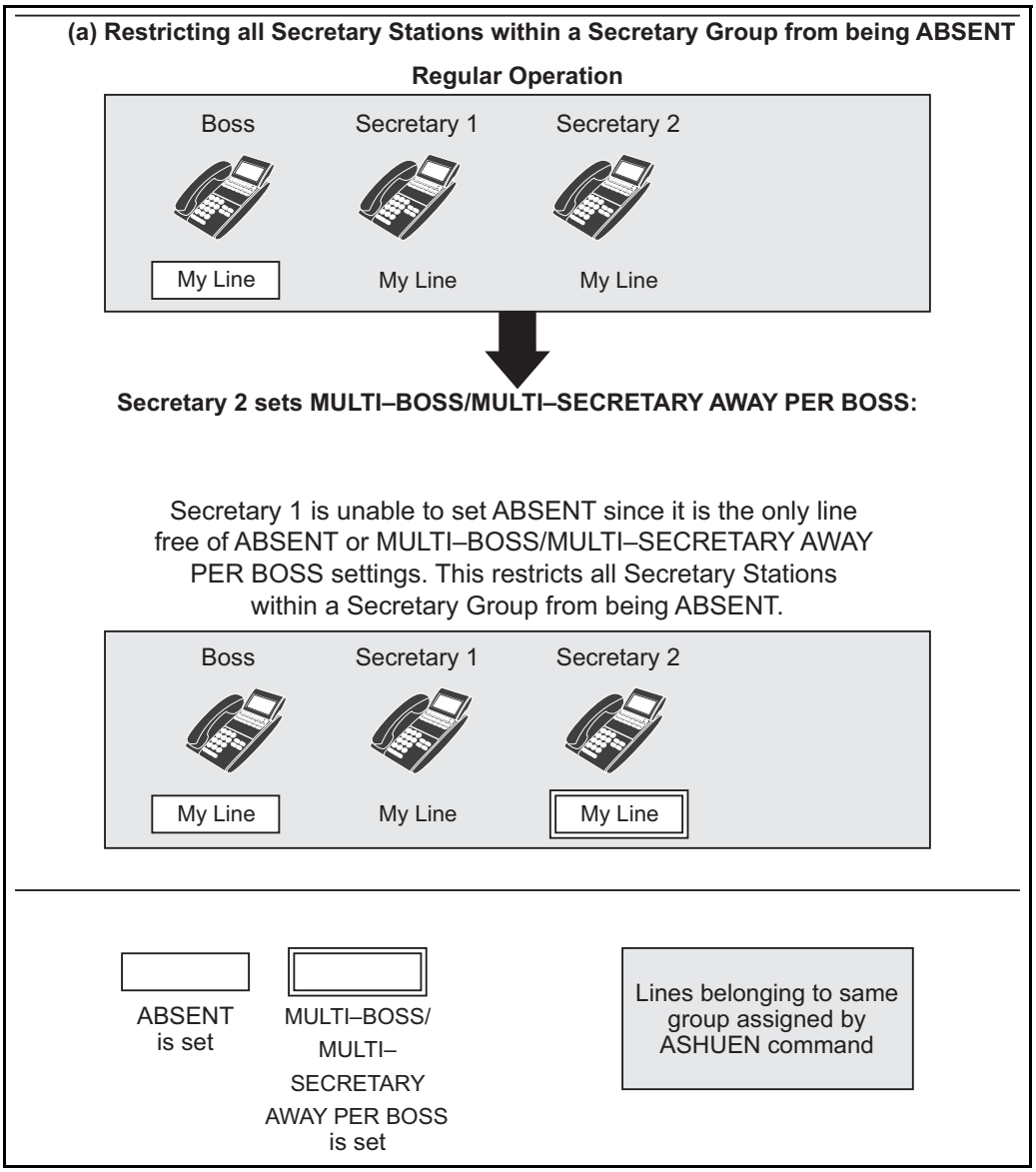
Secretary 2 returns to Boss A's group

	Secretary 1	Secretary 2	Secretary 3
Boss A	Secretary 1	—	Secretary 3
Boss B	Secretary 2	Secretary 3	—
Boss C	Secretary 3	—	—

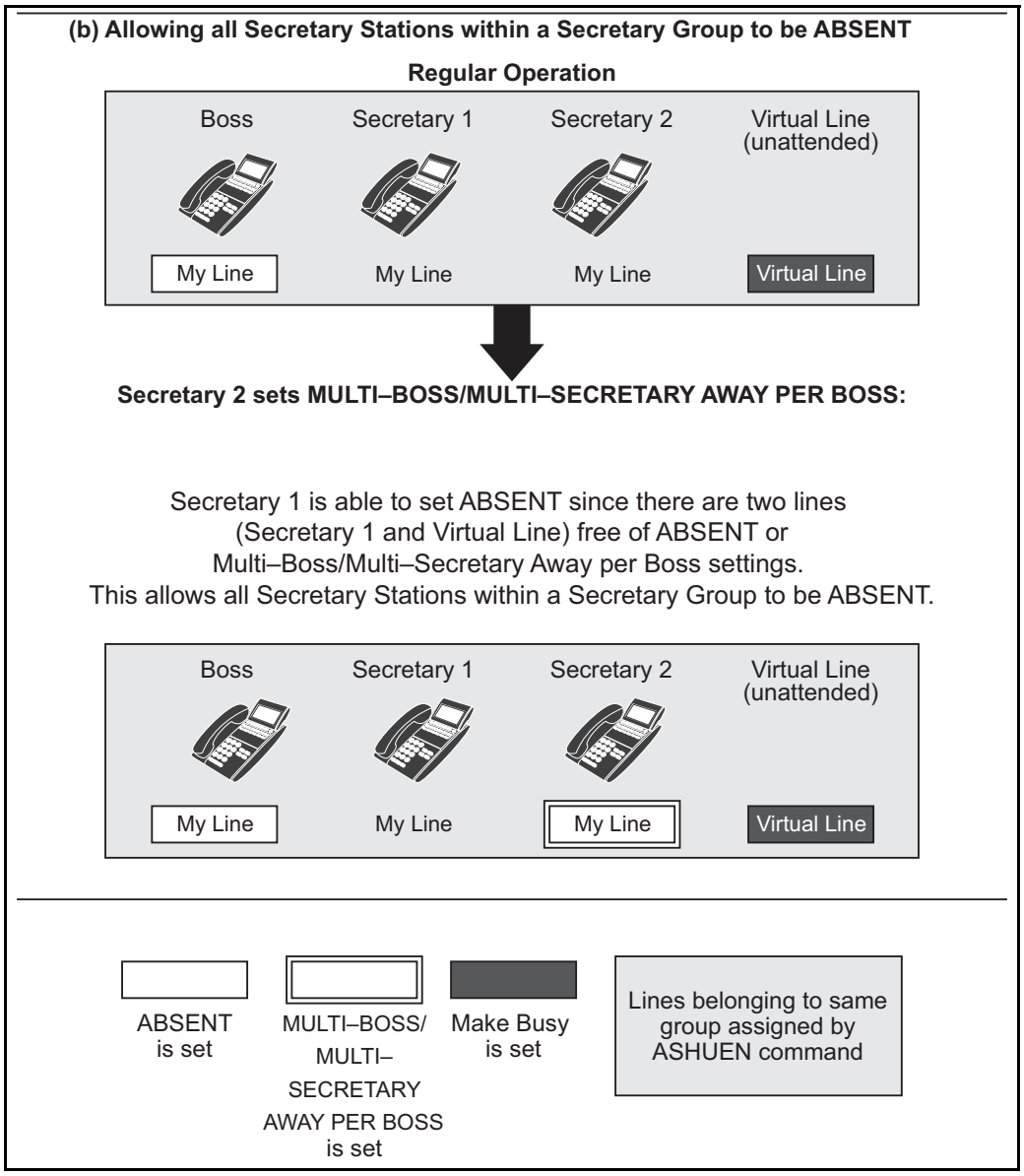
6.1 Allowing All Secretary Stations Within Secretary Group to be ABSENT

Combining MULTI-BOSS/MULTI-SECRETARY with MULTI-BOSS/MULTI-SECRETARY AWAY PER BOSS feature enables all Secretary Stations within a Secretary Group to be ABSENT. This feature applies to Stations with ABSENT set by MULTI-BOSS/MULTI-SECRETARY or MULTI-BOSS/MULTI-SECRETARY AWAY PER BOSS, regardless of its busy/idle status.

(a) Restricting all Secretary Stations within a Secretary Group from being ABSENT



(b) Allowing All Secretary Stations within a Secretary Group to be ABSENT



7. MULTI-BOSS/MULTI-SECRETARY OVERRIDING FROM A SECRETARY'S CALL FORWARDING DESTINATION

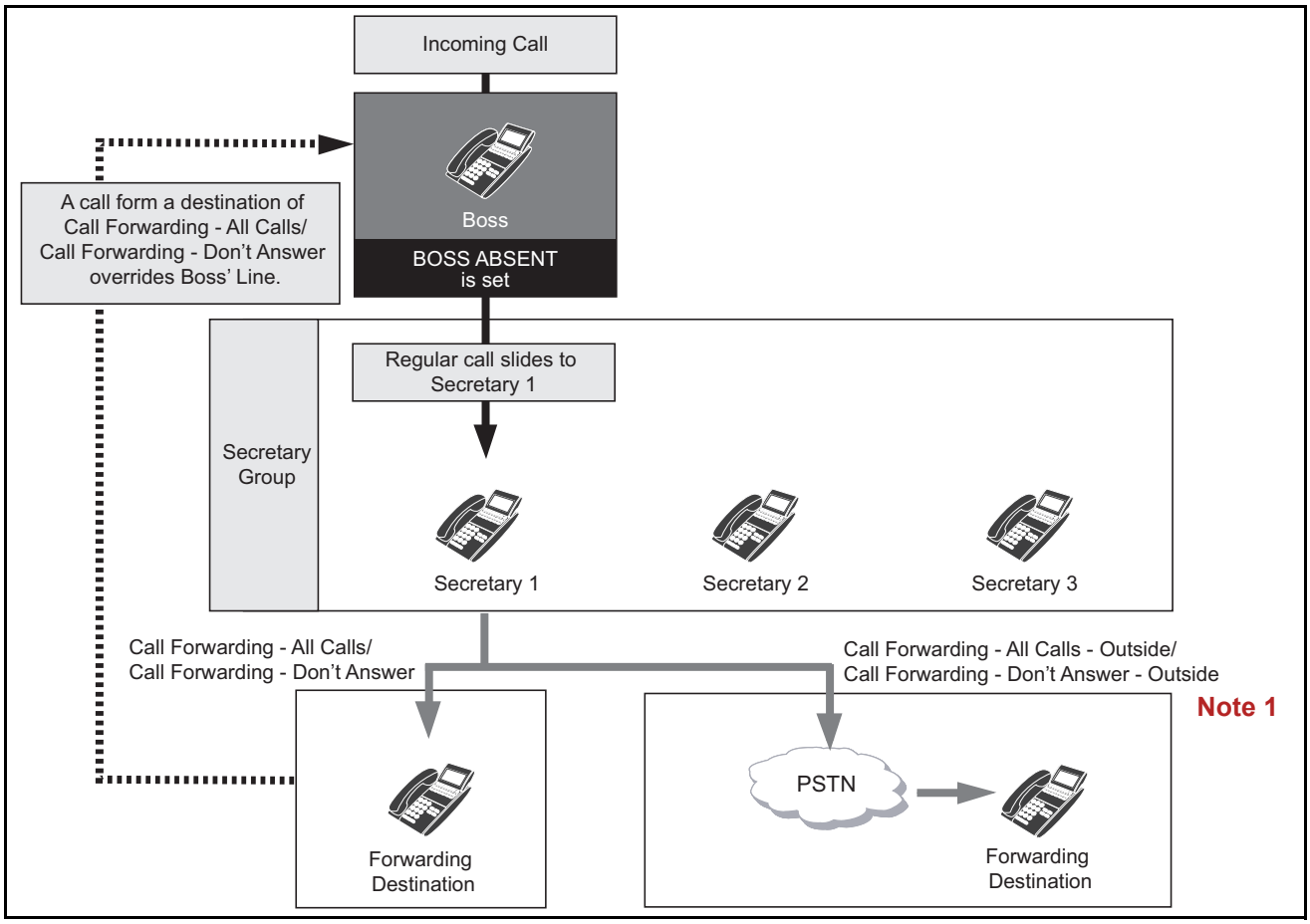
When a Secretary's Station has CALL FORWARDING - ALL CALLS [C-5]/CALL FORWARDING - ALL CALLS - OUTSIDE [C-28] set and a Boss' Station has BOSS ABSENT set, an incoming call to Boss' Station can be forwarded to another station which the Secretary's Station sets.

Note: Available for EMEA only.

Note: If this feature is out of service;

- CALL FORWARDING - ALL CALLS [C-5]/CALL FORWARDING - ALL CALLS - OUTSIDE [C-28] is not activated and the incoming call to Boss' Station slides to the low-order Secretary's Station.
- CALL FORWARDING -DON'T ANSWER [C-3]/CALL FORWARDING - DON'T ANSWER - OUTSIDE [C-74] is activated regardless of this feature. The incoming call to Boss' Station is forwarded to the forwarding destination without sliding to the low-order Secretary's Station.

In the case where Secretary's Station has CALL FORWARDING - ALL CALLS [C-5]/CALL FORWARDING -DON'T ANSWER [C-3], after the forwarding destination answers the call, the destination can place a call to the Boss' Station by Overriding even if the Boss' Station has BOSS ABSENT set.



Note 1: In the case where the Secretary's Station has CALL FORWARDING - ALL CALLS - OUTSIDE [C-28]/CALL FORWARDING - DON'T ANSWER - OUTSIDE [C-74] set, the forwarding destination cannot place a call to Boss' Station by Overriding. The call slides to the Secretary's Station and each Call Forwarding feature is activated again. When CALL FORWARDING - ALL CALLS - OUTSIDE [C-28] is activated, the call is forwarded to the destination again and the destination hears a Busy Tone. When CALL FORWARDING - DON'T ANSWER - OUTSIDE [C-74] is activated, the call is forwarded to the destination again and Don't Answer Timer is reset. Then the Secretary's Station is kept ringing.

8. MULTI-BOSS/MULTI-SECRETARY BOSS' CALL FORWARDING ACTIVATE

When a Boss' Station has CALL FORWARDING - ALL CALLS [C-5]/CALL FORWARDING - DON'T ANSWER [C-3]/CALL FORWARDING - ALL CALLS - OUTSIDE [C-28]/CALL FORWARDING - DON'T ANSWER - OUTSIDE [C-74] set along with BOSS ABSENT, an incoming call to Boss' Station slides to a Secretary's Station.

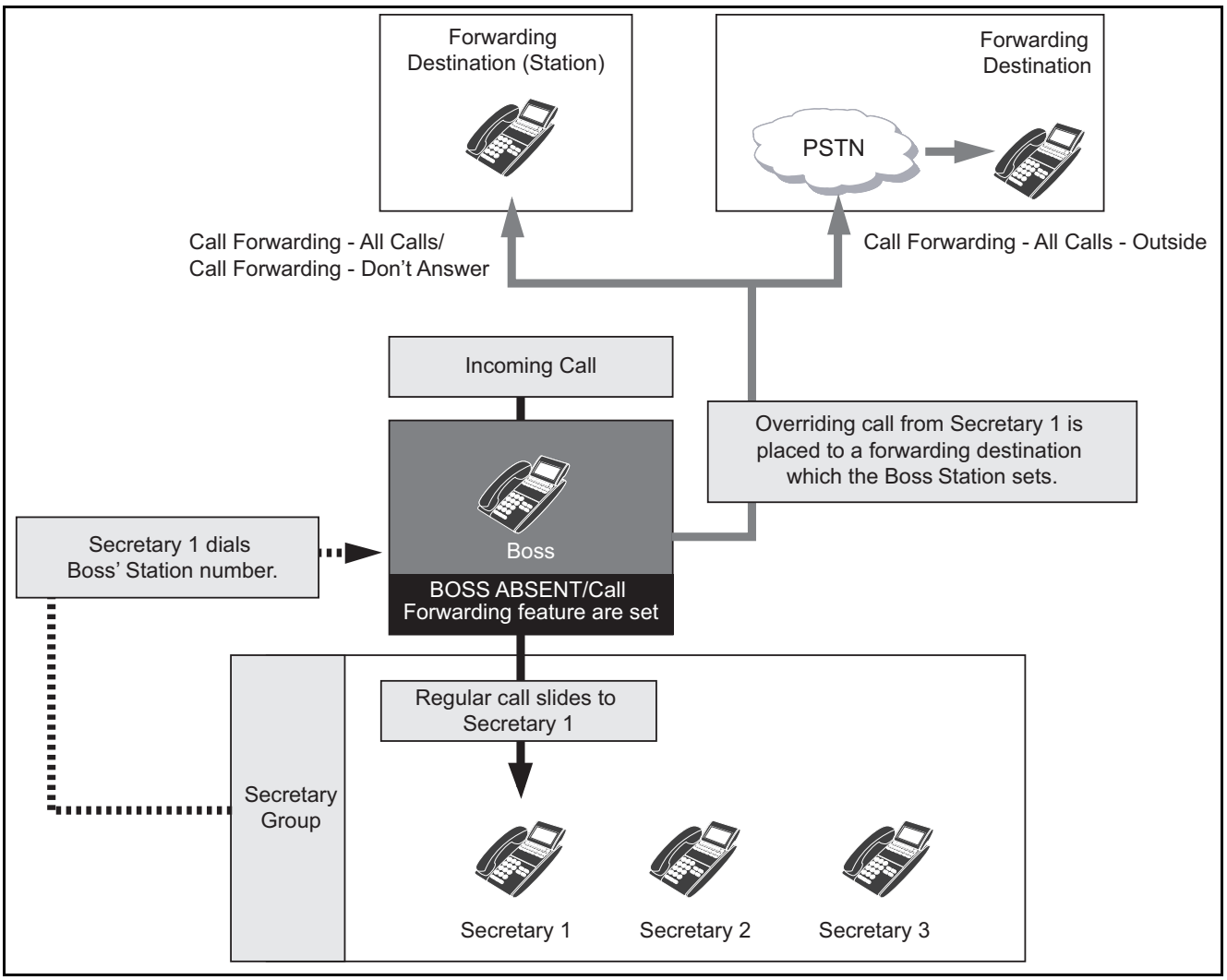
Note: Available for EMEA only.

Note: If this feature is out of service;

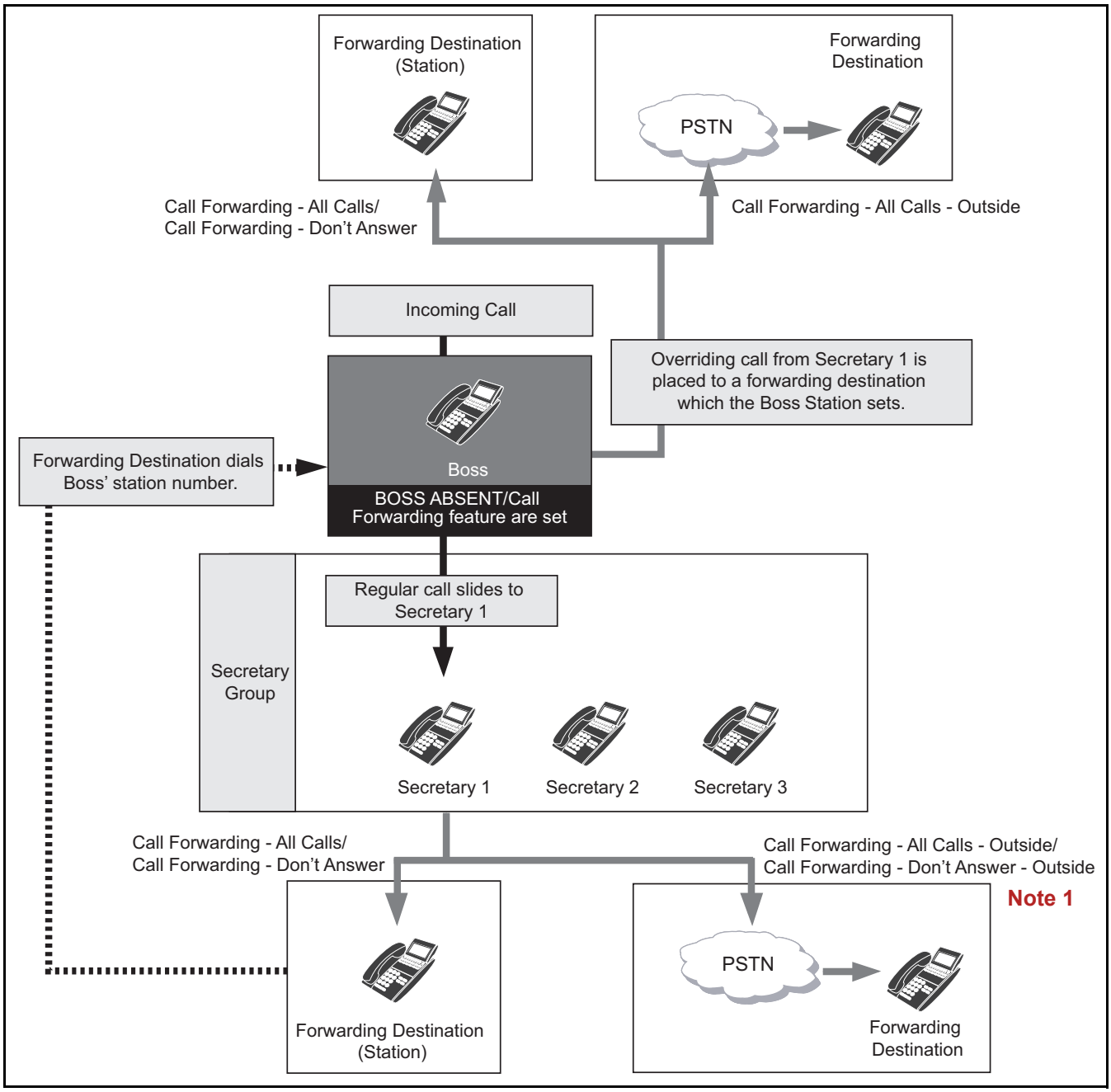
- When Boss' Station has CALL FORWARDING - ALL CALLS [C-5]/CALL FORWARDING - ALL CALLS - OUTSIDE [C-28] set, the incoming call to the Boss' Station is forwarded to the forwarding destination regardless of BOSS ABSENT set.
- When Boss' Station has CALL FORWARDING -DON'T ANSWER [C-3]/CALL FORWARDING - DON'T ANSWER - OUTSIDE [C-74] set, the incoming call to Boss' Station slides to Secretary's Station.

When the Secretary's Station answers the sliding call, performs switch hook flash and then dials the Boss' Station number, the call is forwarded to a forwarding destination which the Boss' Station sets.

Note: When the Boss' Station has CALL FORWARDING - DON'T ANSWER - OUTSIDE [C-74] set, incoming call to the Boss' Station slides to the Secretary's Station. Even if the Secretary's Station answers the sliding call, performs switch hook flash, rings the Boss' Station and the Boss' Station does not answer the call, CALL FORWARDING - DON'T ANSWER - OUTSIDE [C-74] is not activated and the Secretary's Station continues ringing the Boss' Station.



When a Secretary's Station has CALL FORWARDING - ALL CALLS [C-5]/CALL FORWARDING - DON'T ANSWER [C-3] set, the call can be forwarded to another station which the Secretary's Station sets. Then the forwarding destination performs switch hook flash and dials Boss' Station number. The call arrives at another forwarding destination which the Boss' Station sets.



Note 1: In the case where the Secretary's Station has CALL FORWARDING - ALL CALLS - OUTSIDE [C-28]/CALL FORWARDING - DON'T ANSWER - OUTSIDE [C-74] set, the forwarding destination cannot place a call to Boss' Station by Overriding. The call slides to the Secretary's Station and each Call Forwarding feature is activated again. When CALL FORWARDING - ALL CALLS - OUTSIDE [C-28] is activated, the call is forwarded to the destination again and the destination hears a Busy Tone. When CALL FORWARDING - DON'T ANSWER - OUTSIDE [C-74] is activated, the call is forwarded to the destination again and Don't Answer Timer is reset. Then the Secretary's Station is kept ringing.

9. MULTI-BOSS/MULTI-SECRETARY OVERRIDING FROM A STAND-IN SECRETARY

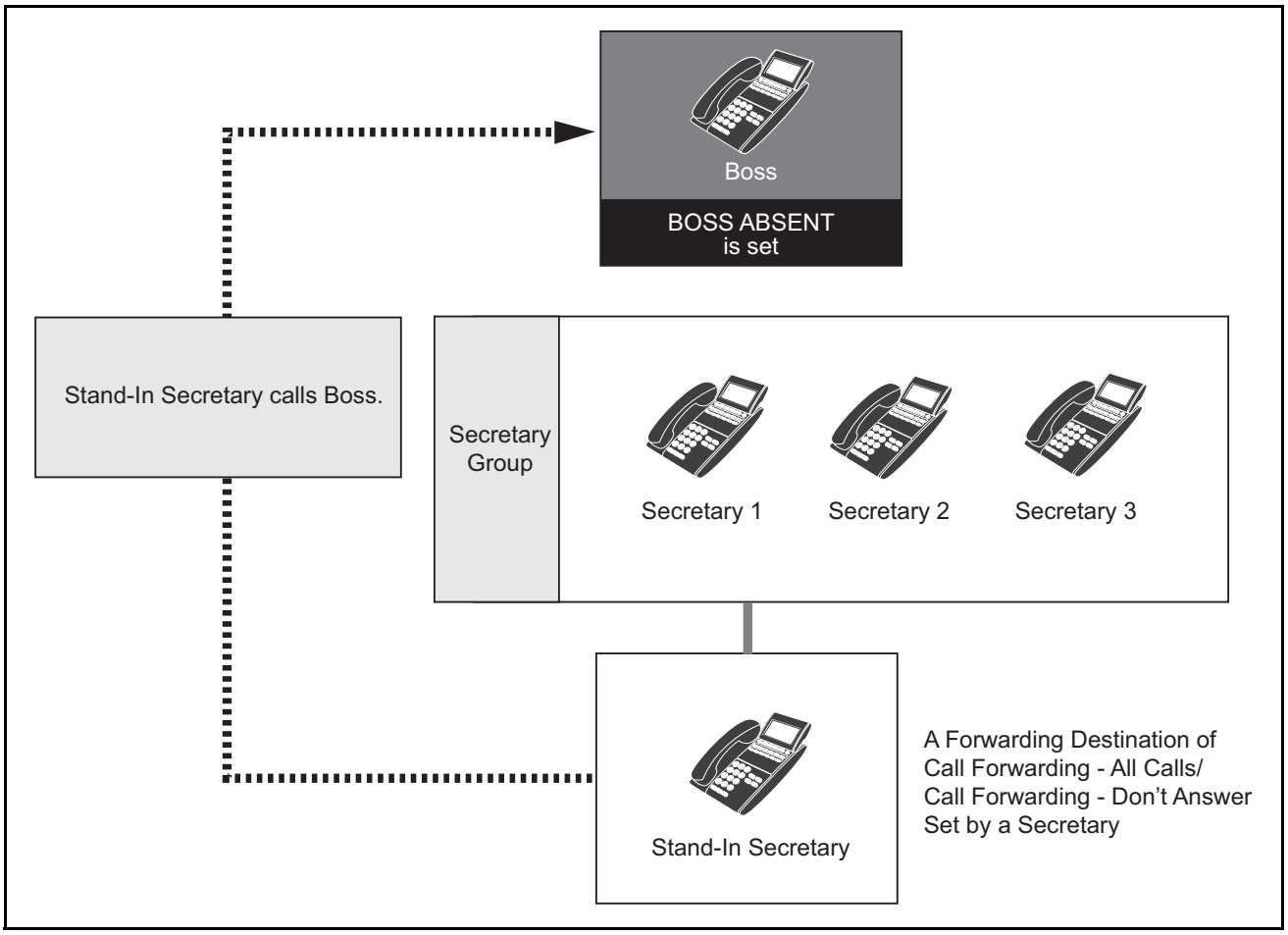
A Stand-In Secretary is a term referring to a forwarding destination station of CALL FORWARDING - ALL CALLS [C-5]/CALL FORWARDING - DON'T ANSWER [C-3] which a Secretary set. This feature is available only between terminals accommodated within the same node or nodes configuring the FCCS network. When a Boss' Station has BOSS ABSENT set, the Stand-In Secretary's Station can place a call to the Boss' Station by Overriding. Whether a call is forwarded from the Secretary's Station or not, the Stand-In Secretary's Station can make a call to the Boss' Station which has BOSS ABSENT set.

Note: Available for EMEA only.

Note: Up to seven Stand-In Secretary's Stations can be set per Boss Secretary group.

Note: If this feature is out of service;
When a Boss' Station has BOSS ABSENT set; a Secretary's Station has CALL FORWARDING - ALL CALLS [C-5]/CALL FORWARDING - DON'T ANSWER [C-3] set and a forwarding destination is a Stand-In Secretary's Station, an incoming call to Boss' Station slides to the Secretary's Station and it is forwarded to the Stand-In Secretary's Station. The Stand-In Secretary's Station cannot place a call to the Boss' Station. The call slides to the Secretary's Station.

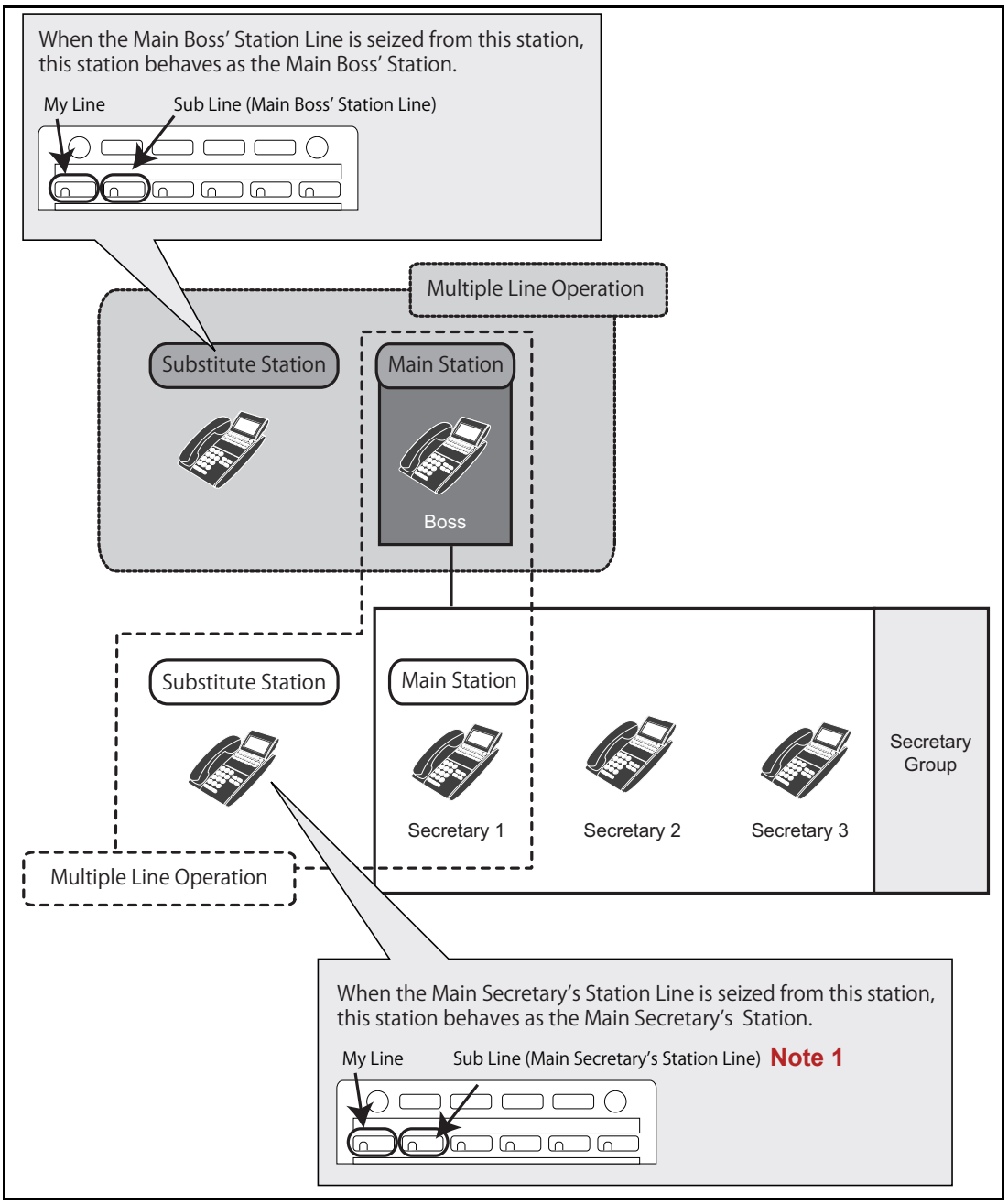
When the Stand-In Secretary's Station rings the idle Boss' Station which has BOSS ABSENT set while talking on the phone with another station/trunk, the Stand-In Secretary's Station can also place a call to the Boss' Station by Overriding without sliding the call to the Secretary's Station.



10. MULTI-BOSS/MULTI-SECRETARY MULTI LINE OPERATION

In the case that a Boss (or a Secretary) has two stations, a main station and a substitute station, when the main station line is assigned to a Sub Line of the substitute station, the substitute station can be operated as well as the main station by seizing the main line from the substitute station even if the substitute line does not belong to a Boss Secretary Group.

Note: Available for all countries.



Note 1: To use MULTI-BOSS/MULTI-SECRETARY ONE TOUCH, assign also the Main Boss' Station Line as a Sub Line.

Operating Procedure

The following shows operating procedure for each feature.

Note: When MULTI-BOSS/MULTI-SECRETARY Multi Line Operation is in service, the following features can be operated not only from Main Boss' Station or Main Secretary's Station but also from a substitute station. Substitute station needs to seize a line of the Boss Station or Secretary Station to use the following features.

- BOSS ABSENT from a Boss' Station
- SECRETARY ABSENT from a Secretary's Station
- MULTI-BOSS/MULTI-SECRETARY-ONE TOUCH:
  - Secretary's Terminal Answering Boss' Line Call
  - Overriding from a Secretary's Station
  - Overriding from a Busy Secretary's Station
- MULTI-BOSS/MULTI-SECRETARY AWAY PER BOSS

1. Setting/Canceling BOSS ABSENT from a Boss' Station

- Setting Procedure

  a.) Setting by Access Code

  STEP 1: Lift Boss' Station's handset.
  As for Substitute Station, press the Main Boss' Station Line key and lift the handset.
  STEP 2: Receive Dial Tone (DT).
  STEP 3: Dial access code to set BOSS ABSENT.
  STEP 4: Receive Service Set Tone (SST): Absent key lamp on Boss' Station lights red.
  STEP 5: Hang up Boss' Station's handset.
  As for Substitute Station, hang up Substitute Station's handset.

  b.) Setting by Assigning Line/Feature key or Soft key

  STEP 1: Lift Boss' Station's handset.
  As for Substitute Station, press the Main Boss' Station Line key and lift the handset.
  STEP 2: Receive Dial Tone (DT).
  STEP 3: Press Line/Feature key or Soft key assigned to set BOSS ABSENT.
  STEP 4: Absent key lamp on Boss' Station lights red.
  STEP 5: Hang up Boss' Station's handset.
  As for Substitute Station, hang up Substitute Station's handset.

- Cancelling Procedure

  a.) Cancelling by Access Code

  STEP 1: Lift Boss' Station's handset.
  As for Substitute Station, press the Main Boss' Station Line key and lift the handset.
  STEP 2: Receive Dial Tone (DT).
  STEP 3: Dial access code to cancel BOSS ABSENT.

STEP 4: Receive Service Set Tone (SST): Absent key lamp on Boss' Station turns off.
STEP 5: Hang up Boss' Station's handset.
As for Substitute Station, hang up Substitute Station's handset.

b.) Canceling Using the Line/Feature key or Soft key

STEP 1: Lift Boss' Station's handset.
As for Substitute Station, press the Main Boss' Station Line key and lift the handset.
STEP 2: Receive Dial Tone (DT).
STEP 3: Press Line/Feature key or Soft key assigned to cancel BOSS ABSENT.
STEP 4: Absent key lamp on Boss' Station turns off.
STEP 5: Hang up Boss' Station's handset.
As for Substitute Station, hang up Substitute Station's handset.

2. Setting/Canceling BOSS ABSENT from a Secretary's Station

a.) Setting by Access Code

STEP 1: Lift Secretary's Station's handset.
As for Substitute Station, press the Main Secretary's Station Line key and lift the handset.
STEP 2: Receive Dial Tone (DT).
STEP 3: Dial access code to set BOSS ABSENT from the Secretary's Station.
STEP 4: Receive Special Dial Tone (SPDT).
STEP 5: Dial Boss' Station number.
STEP 6: Receive Service Set Tone (SST).
STEP 7: Hang up Secretary's Station's handset.
As for Substitute Station, hang up Substitute Station's handset.

b.) Cancelling by Access Code

STEP 1: Lift Secretary's Station's handset.
As for Substitute Station, press the Main Secretary's Station Line key and lift the handset.
STEP 2: Receive Dial Tone (DT).
STEP 3: Dial access code to cancel BOSS ABSENT from the Secretary's Station.
STEP 4: Receive Special Dial Tone (SPDT).
STEP 5: Dial Boss' Station number.
STEP 6: Receive Service Set Tone (SST).
STEP 7: Hang up Secretary's Station's handset.
As for Substitute Station, hang up Substitute Station's handset.

3. Setting/Canceling SECRETARY ABSENT from a Secretary's Station

- Setting Procedure

  a.) Setting by Access Code

  STEP 1: Lift Secretary's Station's handset.
  As for Substitute Station, press the Main Secretary's Station Line key and lift the handset.
  STEP 2: Receive Dial Tone (DT).
  STEP 3: Dial access code to set SECRETARY ABSENT.
  STEP 4: Receive Service Set Tone (SST): Absent key lamp on Secretary's Station lights red.
  STEP 5: Hang up Secretary's Station's handset.
  As for Substitute Station, hang up Substitute Station's handset.

  b.) Setting by Using the Line/Feature key or Soft key

  STEP 1: Press Secretary's Station's monitor/speaker button.
  As for Substitute Station, press the Main Secretary's Station Line key and lift the handset.
  STEP 2: Receive Dial Tone (DT).
  STEP 3: Press Line/Feature key or Soft key assigned to set SECRETARY ABSENT.
  STEP 4: Absent key lamp on Secretary's Station lights red.
  STEP 5: Press Secretary's Station's monitor/speaker button.
  As for Substitute Station, hang up Substitute Station's handset.

- Cancelling Procedure

  a.) Cancelling by Access Code

  STEP 1: Lift Secretary's Station's handset.
  As for Substitute Station, press the Main Secretary's Station Line key and lift the handset.
  STEP 2: Receive Dial Tone (DT).
  STEP 3: Dial access code to cancel SECRETARY ABSENT.
  STEP 4: Receive Service Set Tone (SST): Absent key lamp on Secretary's Station turns off.
  STEP 5: Hang up Secretary's Station's handset.
  As for Substitute Station, hang up Substitute Station's handset.

  b.) Canceling by Using the Line/Feature key or Soft key

  STEP 1: Press Secretary's Station's monitor/speaker button.
  As for Substitute Station, press the Main Secretary's Station Line key and lift the handset.
  STEP 2: Receive Dial Tone (DT).
  STEP 3: Press Line/Feature key or Soft key assigned to cancel SECRETARY ABSENT.
  STEP 4: Absent key lamp on Secretary's Station turns off.
  STEP 5: Press Secretary's Station's monitor/speaker button.
  As for Substitute Station, hang up Substitute Station's handset.

4. Operating Procedure for MULTI-BOSS/MULTI-SECRETARY-ONE TOUCH

4.1 Secretary's Terminal Answering Boss' Line Call

- Operation by Secretary's Station

STEP 1: An idle Secretary's Station presses the flashing Boss' Line key lamp, which indicates that the Boss' Station is receiving a call.
STEP 2: The call originating party and the Secretary are connected.
STEP 3: The above changes the Boss Station's status from busy to idle. **Note 1**

Note 1: If the Boss answers the call before the Secretary, it results in the Secretary's Station Overriding to the busy Boss' Station.

- Operation by Substitute Station

STEP 1: Substitute Station presses the flashing Main Boss' Line key lamp, which indicates that the Main Boss' Station is receiving a call.
STEP 2: Substitute Station goes off-hook. **Note 2**
STEP 3: The call originating party and Substitute Station are connected.

Note 2: You can skip off-hook operation to answer the call by setting ASYD Index 384 System Data 1 Bit 0=1.

- Operation by Substitute Station by Seizing Secretary's Station Line

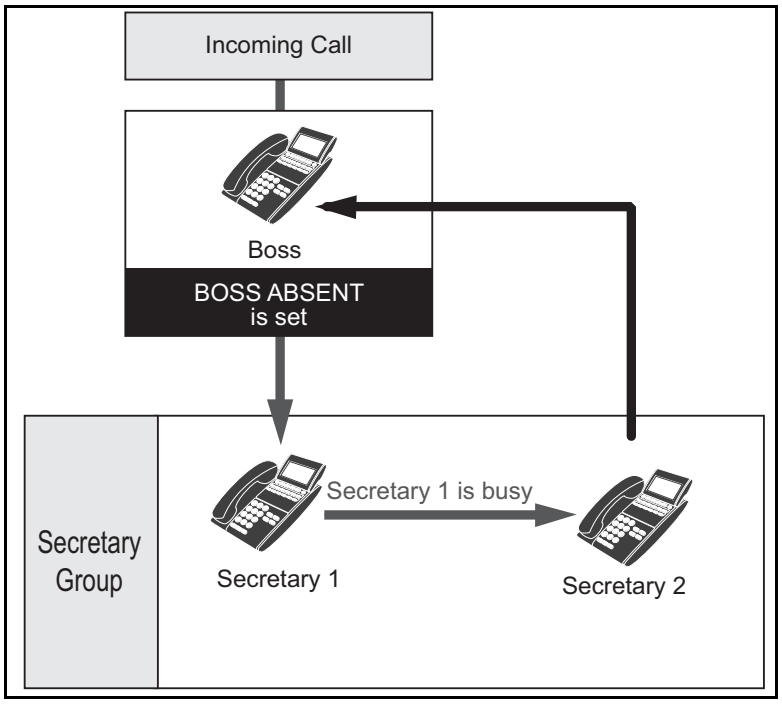
STEP 1: Substitute Station presses the Main Secretary's Station Line key and lifts the handset.
STEP 2: Substitute Station receives Dial Tone (DT).
STEP 3: Substitute Station presses the flashing Main Boss' Station Line key lamp, which indicates that the Main Boss' Station is receiving a call.
STEP 4: By seizing the Main Boss' Station Line, the call originating party and the Main Secretary's Station Line are connected.
STEP 5: The above changes the Main Boss Station's status from busy to idle. **Note 3**

Note 3: If the Main Boss answers the call before the call originating party and the Main Secretary's Station Line are connected, it results in the Substitute Station Overriding to the busy Boss' Station.

4.2 Overriding from a Secretary's Station

a.) Operation for Override: Override from the Call Forwarded Secretary's Station by Slide Action

- Operation by Secretary's Station 2

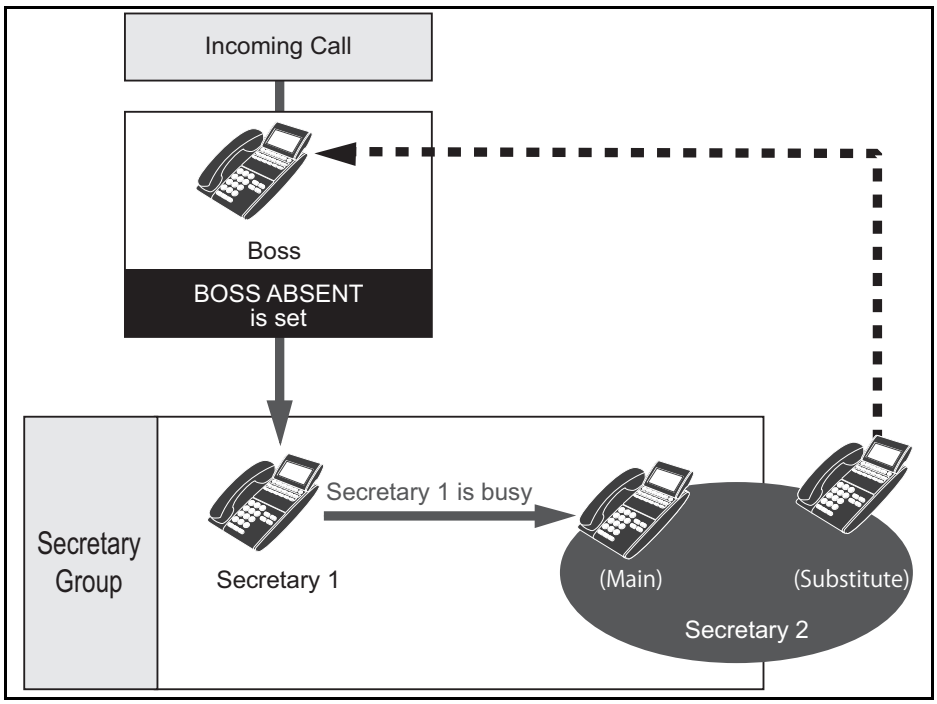


STEP 1: Secretary's Station 2 answers a call forwarded by Slide Action.

STEP 2: The Secretary's Station 2 presses the Boss' Line key:

- If the Boss' Station is in idle status, the Secretary's Station 2 calls the Boss' Station. The incoming call from the call originating party is put on hold.
- If the Boss' Station is in busy status (exception: call via an Attendant Console), Call Waiting is automatically set to the Boss' Station. In this case, Secretary 2 receives Service Set Tone (SST); the call originating party receives Ring Back Tone (RBT).

• Operation by Substitute Station



STEP 1: The Main Secretary's Station 2 receives a call forwarded by Slide Action.

STEP 2: Substitute Station presses the flashing Main Secretary's Station 2 Line key lamp and lift the handset.

STEP 3: The call originating party connects to the Substitute Station.

STEP 4: The Substitute Station presses the Boss' Line key:

- If the Boss' Station is in idle status, the Substitute Station calls the Boss' Station. The incoming call from the call originating party is put on hold.
- If the Boss' Station is in busy status (exception: call via an Attendant Console), Call Waiting is automatically set to the Boss' Station. In this case, Substitute Station receives Service Set Tone (SST); the call originating party receives Ring Back Tone (RBT).

b.) Operation for Override: Direct Override from the Secretary's Station

• Operation by Secretary's Station

STEP 1: An idle Secretary's Station presses the Boss' Line key:

- If the Boss' Station is in idle status, the idle Secretary's Station calls the Boss' Station.
- If the Boss' Station is in busy status (exception: call via an Attendant Console), the Secretary's call Overrides to the Boss' Station.

• Operation by Substitute Station

STEP 1: Substitute Station presses the Main Secretary's Station Line key and lift the handset.

STEP 2: Substitute Station receives Dial Tone (DT).

STEP 3: Substitute Station presses the Main Boss' Line key:

- If the Main Boss' Station is in idle status, the Substitute Station calls the Main Boss' Station.
- If the Main Boss' Station is in busy status (exception: call via an Attendant Console), the Substitute Station's call Overrides to the Main Boss' Station.

4.3 Overriding from a Busy Secretary's Station

a.) Overriding from a busy Secretary's Station

- Operation by Secretary's Station

STEP 1: A busy Secretary's Station confirms that the Boss' Line key lamp is on, indicating that the Boss' Station is busy.
STEP 2: Secretary's Station presses the CALL PARK Feature key (pre-assigned to a Line/Feature key) to put its ongoing call on hold.
STEP 3: Secretary's Station presses the Boss' Line key (lamp is on) to Override to the Boss' ongoing call.

- Operation by Substitute Station

STEP 1: Substitute Station is talking by using the Main Secretary's Line.
STEP 2: Substitute Station using the Main Secretary's Line presses the CALL PARK Feature key (pre-assigned to a Line/Feature key) to put its ongoing call on hold.
STEP 3: Substitute Station presses the Main Boss' Line key (lamp is on) to Override to the Main Boss' ongoing call.

b.) Secretary's Station/Substitute Station Resuming Communication with its Original Caller After Overriding

- Operation by Secretary's Station

STEP 1: Secretary finishes Overriding.
STEP 2: Secretary's line becomes idle.
STEP 3: Boss resumes communication with its original caller.
STEP 4: Secretary initiates a call, then presses the CALL PARK Feature key (pre-assigned to a Line/Feature key) to resume communication with its original caller.

- Operation by Substitute Station

STEP 1: Substitute Station finishes Overriding.
STEP 2: The Main Secretary's line becomes idle.
STEP 3: The Main Boss resumes communication with its original caller.
STEP 4: Substitute Station presses the Main Secretary's Line key and lift the handset.
STEP 5: Substitute Station receives Dial Tone (DT).
STEP 6: Substitute Station dials the Access Code to answer the parked call to resume communication with its original caller.

5. Operating Procedure for MULTI-BOSS/MULTI-SECRETARY AWAY PER BOSS

- Setting MULTI-BOSS/MULTI-SECRETARY AWAY PER BOSS

  LCD number indicates the Boss' number.

  STEP 1: Lift the Secretary's Station's handset.
  As for Substitute Station, press the Main Secretary's Station Line key and lift the handset.
  STEP 2: Receive Dial Tone (DT).
  STEP 3: Dial Access Code to enable SECRETARY AWAY PER BOSS.
  STEP 4: Receive Dial Tone (SPDT).
  STEP 5: Dial the desired Boss' Station number.
  STEP 6: Receive Service Set Tone (SST).
  STEP 7: Hang up Secretary's Station's handset.
  As for Substitute Station, hang up Substitute Station's handset.
  STEP 8: Information is displayed on the LCD.

When the display of maximum 16 digits of Call Forwarding Information on the LCD is unavailable.
(ASYDL, System Data 1, Index 677, Bit 3=0)

Display in 16x2 lines: Alphabet

O	U	T						1	2	3	4	5	6	7	8

Display in 24x3 lines: Alphabet

A	B	S	E	N	T											1	2	3	4	5	6	7	8

When the display of maximum 16 digits of Call Forwarding Information on the LCD is available.
(ASYDL, System Data 1, Index 677, Bit 3=1)

Display in 16x2 lines: Alphabet

O	U	T						1	2	3	4	5	6	7	8

Display in 24x3 lines: Alphabet

A	B	S	E	N	T			1	2	3	4	5	6	7	8	9	0	1	2	3	4	5	6

- Cancelling MULTI-BOSS/MULTI-SECRETARY AWAY PER BOSS

STEP 1: Lift the Secretary's Station's handset.
As for Substitute Station, press the Main Secretary's Station Line key and lift the handset.
STEP 2: Receive Dial Tone (DT).
STEP 3: Dial Access Code to disable SECRETARY AWAY PER BOSS.
STEP 4: Receive Dial Tone (SPDT).
STEP 5: Dial the desired Boss' Station number.
STEP 6: Receive Service Set Tone (SST).
STEP 7: Hang up Secretary's Station's handset.
As for Substitute Station, hang up Substitute Station's handset.
STEP 8: Information is displayed on the LCD.

When the display of maximum 16 digits of Call Forwarding Information on the LCD is unavailable.
(ASYDL, System Data 1, Index 677, Bit 3=0)

Display in 16x2 lines: Alphabet

I	N							1	2	3	4	5	6	7	8

Display in 24x3 lines: Alphabet

P	R	E	S	E	N	T									1	2	3	4	5	6	7	8

When the display of maximum 16 digits of Call Forwarding Information on the LCD is available.
(ASYDL, System Data 1, Index 677, Bit 3=1)

Display in 16x2 lines: Alphabet

I	N							1	2	3	4	5	6	7	8

Display in 24x3 lines: Alphabet

P	R	E	S	E	N	T		1	2	3	4	5	6	7	8	9	0	1	2	3	4	5	6

6. Operating Procedure to Allowing All Secretary Stations Within Secretary Group to be ABSENT

Setting Procedure

a.) Restricting All Secretary Stations within a Secretary Group from being ABSENT

STEP 1: STEP 1: Assign Telephone Number for Boss' Station and Secretary's Station using ASHUEN command.

b.) Allowing All Secretary Stations within a Secretary Group to be ABSENT

STEP 1: Assign Virtual Line to allow ABSENT using ASDT command.

STEP 2: Assign Telephone Number for Boss' Station, Secretary's Station and Virtual Line using ASHUEN command.

STEP 3: Assign Make Busy to Virtual Line using MBST command.

7. Operating Procedure for MULTI-BOSS/MULTI-SECRETARY OVERRIDING FROM A SECRETARY'S CALL FORWARDING DESTINATION

STEP 1: Secretary's Station sets CALL FORWARDING - ALL CALLS/CALL FORWARDING - DON'T ANSWER.

STEP 2: An incoming call to Boss' Station slides to Secretary's Station.

STEP 3: The call is forwarded to the destination which the Secretary's Station sets.

STEP 4: The forwarding destination answers the call.

STEP 5: The forwarding destination performs switch hook flash.

STEP 6: The forwarding destination dials the Boss' Station number.

Note: MULTI-BOSS/MULTI-SECRETARY Overriding from a Secretary's Call Forwarding Destination is available.

8. Operating Procedure for MULTI-BOSS/MULTI-SECRETARY BOSS' CALL FORWARDING ACTIVATE

STEP 1: Boss' Station sets CALL FORWARDING - ALL CALLS/CALL FORWARDING - DON'T ANSWER.

STEP 2: Secretary's Station answers a call arriving at Boss' Station and forwards it to the Boss' Station.

STEP 3: The call is forwarded to the destination which the Boss' Station sets.

9. Operating Procedure for MULTI-BOSS/MULTI-SECRETARY OVERRIDING FROM A STAND-IN SECRETARY

When a Stand-In Secretary's Station makes a call to Boss' Station by Overriding while any other lines are not connected

STEP 1: Secretary's Station sets a Stand-In Secretary's Station number as a destination of CALL FORWARDING - ALL CALLS/CALL FORWARDING - DON'T ANSWER.

STEP 2: Stand-In Secretary's Station rings Boss' Station which has BOSS ABSENT set. **Note 1**

When a Stand-In Secretary's Station makes a call to Boss' Station by Overriding while talking on the phone with another station/trunk

STEP 1: Secretary's Station sets a Stand-In Secretary's Station number as a destination of CALL FORWARDING - ALL CALLS/CALL FORWARDING - DON'T ANSWER.

STEP 2: Stand-In Secretary's Station performs switch hook flash while talking on the phone with another station/trunk, and then rings Boss' Station which has BOSS ABSENT set. **Note 1**

Note 1: The call does not slide to the next Secretary's Station but Stand-In Secretary's Station can talk with Boss' Station.

Service Conditions

1. Common Conditions

1. Terminals available for Boss'/Secretary's Terminal are the following:
   - Analog terminal (LC/MC)
   - Desktop terminal
   - DtermIP (Proprietary Protocol)
   - DtermIP (SIP)
   - Soft Phone
2. When a call arrives at a Boss'/Secretary's Station, the calling/called party's terminal's service name display (LCD) will indicate "UCD."

1.1 Programming Conditions

(a) This feature is available for Stations where Telephone Number is registered using the ALGSN command [Network Data Memory (NDM)].

Note: The maximum number of Secretary Groups combined with UCD Groups is fixed to 63 per node but can be expanded to 255 when using Group Number Expansion; the maximum number of stations within a Secretary Group is 21.

Note: When Group Number Expansion is used, if the Boss Secretary Group has already been registered, delete the existing Boss Secretary Group data and then reregister it by using ASHUEN command. Also, when reregistering, check the "Multi Boss Secretary Group" checkbox of ASHUEN command, by which ON (Boss Secretary Group)/OFF (UCD Group) is displayed per groups for ASHUEN and LSHUEN command. However, if not deleted and reregistered, it will be displayed as OFF (UCD Group) even if it is Boss Secretary Group.

(b) Boss' Station and Secretary's Station must be in the same node.

(c) Boss' Station may not be registered as a Secretary's Station.

(d) Secretary's Station may belong to multiple Secretary Groups. There is no limit to the number of Secretary Groups the Secretary's Station can belong to.

(e) The same Service Feature Restriction Class must be assigned to all Boss/Secretary Stations within a Secretary Group.

Note: For details on language display for the LCD, refer to MULTILINGUAL DISPLAY [M-117].

(f) If more than 255 groups are registered in the combination of Boss Secretary Group and UCD Group, an error window will appear.

(g) When the office data of Multi-Boss/Multi-Secretary is changed, it will takes about 30 seconds to reflect the change. Within this time, if a call is made/received by the Boss/Secretary station for which the data is changed, the former operation is applied for the station.

(h) When registering the Boss/Secretary Group in the ASHUEN command, be sure to check the "Multi Boss Secretary Group" checkbox.

(i) When listing up by using LSHUEN command, all of the UCD Groups and Boss Secretary Groups are output, and ON (Boss Secretary Group)/OFF (UCD Group) is displayed per groups.

(j) When a UCD Group has been registered by ASHU/ASHUL command, this feature is not available.

1.2 Interactions

(a) This feature is not available for Hotel/ACD systems.

(b) This feature does not correspond to OAI features. Do not assign this feature to an OAI terminal.

(c) Existing UCD features (OVERFLOW-UCD [O-13], CALL WAITING LAMP-UCD [C-76], DELAY ANNOUNCEMENT-UCD [D-31], UCD Peg count) are not available for terminals assigned as the Boss' or Secretary's Station.

(d) Soft Phone related services (SOFT PHONE WIRELESS HANDSET [S-145], SOFT PHONE WIRED-PHONE HANDSET [S-147] and SOFT PHONE WLAN HANDSET [S-159]) are not available for Boss'/Secretary's Station.

(e) Station Hunting services (Pilot, Circular, UCD) are not available for the Boss' or Secretary's Station. Any Station Hunting service previously assigned must be deleted prior to assigning this feature.

(f) Do not assign ASFC SFI=181 (No Hunting to Next Station in Group) to a Secretary's Station.

(g) Terminals assigned as Multi-Boss/Multi-Secretary Stations do not work in conjunction with existing Boss Secretary services such as BOSS-SECRETARY OVERRIDE [B-5D], BOSS-SECRETARY - MESSAGE WAITING LAMP CONTROL [B-8], and BOSS-SECRETARY TRANSFER [B-11].

(h) If a Secretary's Station assigned with multiple My Lines is busy, a call does not slide to the Sub Line within the same number, but will skip to the next Secretary Station.

(i) BROKERAGE-HOTLINE [B-6] is not available for terminals with this feature assigned.

(j) Do not assign DUAL STATION CALL [D-156] to a Boss' or Secretary's Station.

(k) In DIAL PLAN TO SUPPORT IP CENTREX FUNCTIONALITY [D-166], data cannot be assigned across different UGNs.

(l) When originating a call from a Secretary's Station, the number assigned to the seized line (My Line or Sub Line) will be displayed on the LCD.

(m) The order of priority is as follows when a Boss' Station has BOSS ABSENT set along with other call forwarding features:

Order of Priority when BOSS ABSENT is Set

Call Forwarding Features	
1. NUMBER SHARING [N-45] (When call forwarding is assigned to a sub-station)	High Priority
2. LOGGED OUT IP STATION-CALL DESTINATION [L-53] **Note 14**	
3. CALL FORWARDING-ALL CALLS [C-5] **Note 14**	
4. MOBILITY ACCESS (MA) [M-118] **Note 14**	
5. MULTI-BOSS/MULTI-SECRETARY [M-120]	
6. CALL FORWARDING-BUSY LINE [C-2]	
7. STATION HUNTING-TERMINAL [S-9]	
8. DO NOT DISTURB -D [D-11D]	
9. CALL FORWARDING-DON'T ANSWER [C-3]	Low Priority

Note 14: Order of priority may be changed by ASFC SFI=209.

2. Conditions to Set BOSS ABSENT

1. When the calling party dials the Boss' Station, the call arrives at the Secretary Group by Pilot service.
2. When the calling party dials the Secretary's Station Number, the call is received according to DIRECT-IN TERMINATION (DIT) [D-7].
3. If the Boss' line is busy and BOSS ABSENT is not set, incoming calls arrive at a Secretary's Station by Pilot service.
4. When a Secretary's Station has the following features set, that Station gets skipped from Slide Action:
   (a) Secretary's Station gets skipped from Slide Action unconditionally when the following features are set:
   - CALL FORWARDING-ALL CALLS [C-5]
   - LOGGED OUT IP STATION-CALL DESTINATION [L-53] **Note 4**
   - SECRETARY ABSENT
   - DO NOT DISTURB -D [D-11D]

   (b) Secretary's Station in busy status with CALL FORWARDING-BUSY LINE [C-2] assigned gets skipped from Slide Action.
5. If a call arrives at a Secretary's Station that has CALL FORWARDING-DON'T ANSWER [C-3] set, the call will be forwarded to the assigned destination. It is not possible to Override to the Boss other than from the forwarded Secretary's Station. Overriding to the Boss is unavailable in this case. Override is available, however, if the call forwarding destination is a Secretary's Station in the Boss' Secretary Group.
6. If none of the Secretary's Station within the Secretary Group is eligible for Slide Action (all lines are busy or all Secretaries have SECRETARY ABSENT set), the entire Secretary Group will be considered busy. **Note 5**
   (a) The following features are available when the entire Secretary Group is busy:
   - CALL BACK [C-1] : Secretary 1 is eligible for this feature.
   - STEP CALL [S-13] : After changing the last digits, the Boss' Station is recalled.
   - EXECUTIVE RIGHT-OF-WAY [E-1] : Secretary 1 is eligible for this feature.
   - CALL WAITING-TERMINATING [C-12] : Secretary 1 is eligible for this feature.

   (b) Busy line features will not be available if Secretary 1 has them restricted.
   (c) It is possible to assign Multi-Boss/Multi-Secretary queuing per system (usage is combined with UCD).
   - The maximum Multi-Boss/Multi-Secretary queuing slots per system are 640 (usage combined with UCD).
   - If incoming calls exceed the number of maximum possible number of queuing, queuing will not occur. The calling party will hear Busy Tone (BT).
   - During Multi-Boss/Multi-Secretary queuing, the calling party will hear Ringing Back Tone (RBT).
   - When the calling party is the Attendant Console, queuing will not occur. Attendant Console will hear Busy Tone (BT).
   - If the calling party has a held call, queuing will not occur. The calling party will hear Busy Tone (BT).
7. If BOSS ABSENT is set, the Boss' Desktop terminal's service name display (LCD) will indicate "BYO SET." **Note 6**
8. If BOSS ABSENT is not set, the Boss' Desktop terminal's service name display (LCD) will indicate "BYO CNCL." **Note 6**

9. Before deleting a Boss Secretary Group using the ASHUEN command, cancel the BOSS ABSENT set to the Boss' Station and the SECRETARY ABSENT set to the Secretary's Station.
10. Setting and canceling BOSS ABSENT are available through the Sub Line of the Substitute Station by the system data (ASYDL SYS1 Index 1089 Bit 6=0) setting.
    When BOSS ABSENT is set, the Sub Line Lamp of the Substitute Station does not light but My Line Lamp of the Main Boss' Station lights.
11. The following show behaviors when BOSS ABSENT is set/cancelled.

BOSS ABSENT is set/cancelled by seizing Main Boss' Station Line from Substitute Station:

Operation		When BOSS ABSENT Is Set/Cancelled
• Dial Access Code • Press pre-programmed Line/ Feature key or Soft key	Set	• "BYO SET" is displayed in the LCD of the Operating Station. • Service Set Tone (SST) is received when Access Code is used. • On the Main Boss' Station, the lamp of the BUSY OUT Feature key (FKY=19) turns on.
	Cancel	• "BYO CNCL" is displayed in the LCD of the Operating Station. • Service Set Tone (SST) is received when Access Code is used. • On the Main Boss' Station, the lamp of the BUSY OUT Feature key (FKY=19) turns off.

Note 4: All Stations in log out status get skipped from Slide Action regardless of set/cancel status for LOGGED OUT IP STATION-CALL DESTINATION [L-53] feature.

Note 5: Actions (for example: service settings, call initiation) vary for each feature depending on the state of Secretary 1 such as the Station's idle/busy status.

Note 6: BYO stands for "Busy Out."

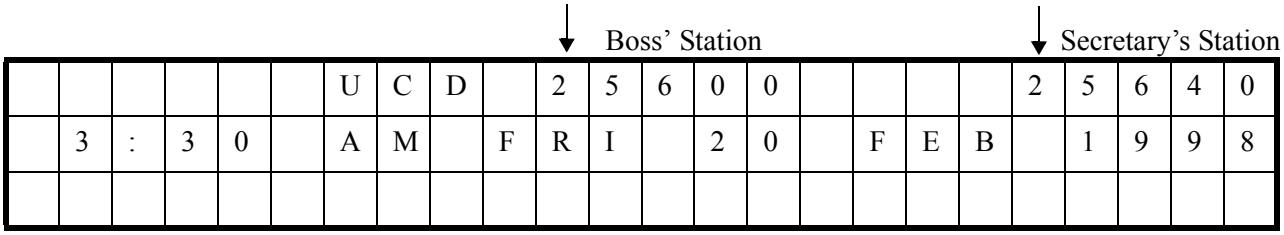
3. Conditions to Set BOSS ABSENT From a Secretary's Station

1. Only the Secretary's Station can set BOSS ABSENT to a Boss' Station. If a different type of Station attempts to set this feature, that Station will receive Reorder Tone (ROT).
2. Only the Boss' Station is qualified to be the destination for BOSS ABSENT. If the Secretary attempts to set a different type of Station to be the destination for this feature, the Secretary will receive Reorder Tone (ROT).
3. The Secretary is only enabled to set/cancel BOSS ABSENT to the Boss' Station associated with its Secretary Group. If the Secretary attempts to set BOSS ABSENT to a Boss' Station not associated with its Secretary Group, the Secretary will receive Reorder Tone (ROT).
4. Setting/cancelling BOSS ABSENT is only available while the Secretary's Station is idle.
5. If BOSS ABSENT is set to a Boss' Station where the service is already set, setting conditions remain the same. The Secretary conducting the setting receives Service Set Tone (SST).
6. If BOSS ABSENT is cancelled on a Boss' Station where the service is already cancelled, cancellation conditions remain the same. The Secretary conducting the setting receives Service Set Tone (SST).

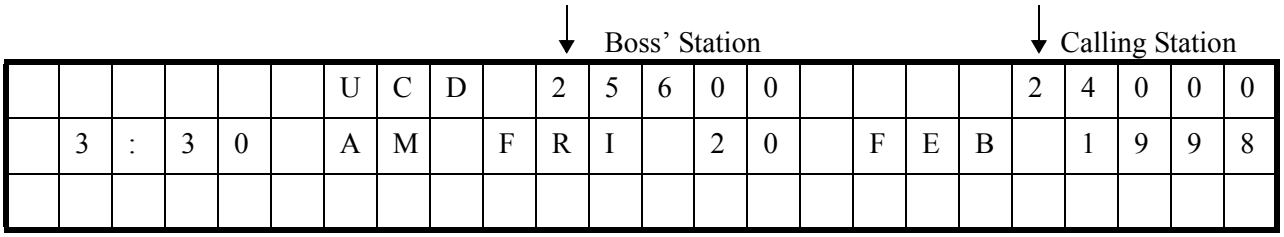
4. Service Name Display (LCD) When BOSS ABSENT is Set

1. Service name display (LCD) when a call to the Boss' Station arrives at the Secretary Group: **Note 3**

(a) Calling Station's LCD Indication

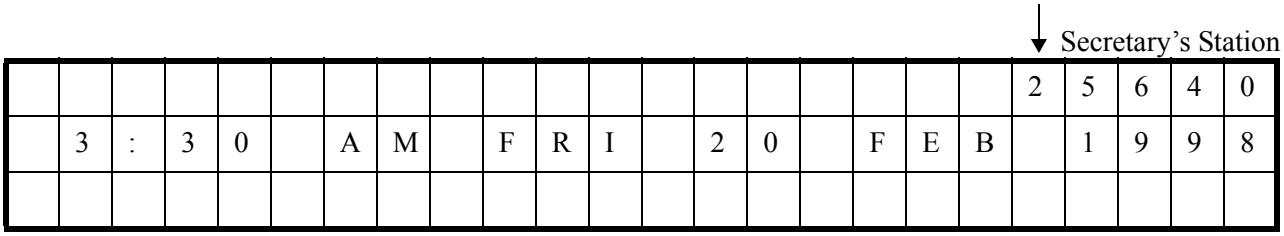


(b) Secretary's Station's LCD Indication

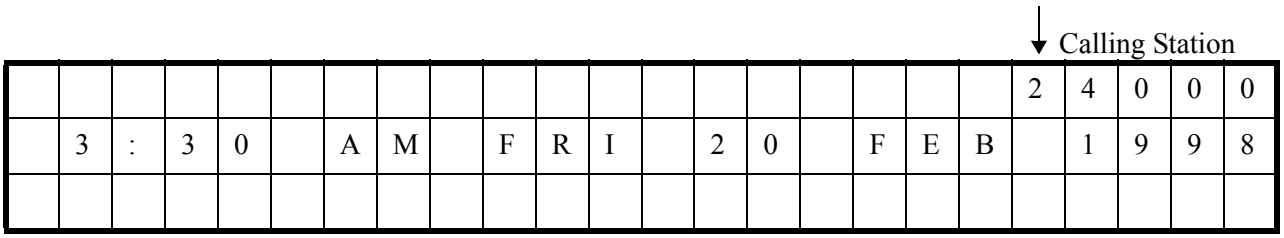


2. Service name display (LCD) when a call is made directly to the Secretary's Station: **Note 3**

(a) Calling Station's LCD Indication



(b) Secretary's Station's LCD Indication



Note 3: For details on changing the availability and position of the display, refer to the section entitled "Display (LCD) Indications" in this manual.

5. Conditions to Set SECRETARY ABSENT

1. Setting SECRETARY ABSENT with an Access Code is only available when the Secretary's Station is idle.
2. Setting SECRETARY ABSENT with a Line/Feature key is only available when the Secretary's Station is idle or engaged in a two-party call.
3. Cancelling SECRETARY ABSENT is only available when the Secretary's Station is idle.
4. When a Secretary Station belonging to multiple Secretary Groups sets/cancels SECRETARY ABSENT, the setting/cancelling will affect all Secretary Groups it belongs to. It is not possible to set/cancel SECRETARY ABSENT for an individual Secretary Group.
5. It is possible to make and receive a call using Secretary's Station number regardless of whether SECRETARY ABSENT is set or canceled.
6. When all Secretaries within a Secretary Group have SECRETARY ABSENT set, Group status will be the same as when all lines are busy.
7. While SECRETARY ABSENT is being set, the Secretary's Desktop terminal's service name display (LCD) will indicate "BYO SET." **Note 1**
8. While SECRETARY ABSENT is being canceled, the Secretary's Desktop terminal's service name display (LCD) will indicate "BYO CNCL." **Note 1**
9. Before deleting a Boss Secretary Group using the ASHUEN command, cancel the BOSS ABSENT set to the Boss' Station and the SECRETARY ABSENT set to the Secretary's Station.
10. Setting and canceling SECRETARY ABSENT are available through the Sub Line of the Substitute Station by the system data (ASYDL SYS1 Index 1089 Bit 6=0) setting.
    When SECRETARY ABSENT is set, the Sub Line Lamp of the Substitute Station does not light but My Line Lamp of the Main Secretary's Station lights.
11. The following show behaviors when SECRETARY ABSENT is set/canceled.

SECRETARY ABSENT is set/canceled by seizing Main Secretary's Station Line from Substitute Station:

Operation		When SECRETARY ABSENT Is Set/Cancelled
• Dial Access Code • Press pre-programmed Line/ Feature key or Soft key	Set	• "BYO SET" is displayed in the LCD of the Operating Station. • Service Set Tone (SST) is received when Access Code is used. • On the Main Secretary's Station, the lamp of the BUSY OUT Feature key (FKY=19) turns on.
	Cancel	• "BYO CNCL" is displayed in the LCD of the Operating Station. • Service Set Tone (SST) is received when Access Code is used. • On the Main Secretary's Station, the lamp of the BUSY OUT Feature key (FKY=19) turns off.

Note 1: BYO stands for "Busy Out."

6. Conditions for MULTI-BOSS/MULTI-SECRETARY-ONE TOUCH

6.1 Interactions

(a) The order of priority is as follows for Overriding when a Boss' Station has set other call forwarding features:

Order of Priority when a Call Overrides to a Boss' Station

Call Forwarding Features	
1. CW and EROW features to the Boss **Note 2**	High Priority
2. NUMBER SHARING [N-45] (When call forwarding is assigned to a sub-station)	
3. LOGGED OUT IP STATION-CALL DESTINATION [L-53] **Note 3**	
4. CALL FORWARDING-ALL CALLS [C-5] **Note 3**	
5. MOBILITY ACCESS (MA) [M-118] (MA CALL FORWARDING) **Note 3**	
6. CALL FORWARDING-BUSY LINE [C-2]	
7. STATION HUNTING-TERMINAL [S-9]	
8. DO NOT DISTURB -D [D-11D]	
9. CALL FORWARDING-DON'T ANSWER [C-3]	Low Priority

Note 2: Call Waiting (CW) and Executive-Right-Of-Way (EROW) indicates CW and EROW for Multi-Boss/Multi-Secretary Feature.
CW and EROW will be restricted when Multi-Boss/Multi-Secretary Feature is unavailable.

Note 3: Order of priority may be changed by ASFC SFI=209.

6.2 Conditions to Status Display of Boss' Terminal

(a) It is necessary to accommodate the Boss' Line as a Sub Line to the Secretary's Desktop terminal.

6.3 Conditions for Secretary's Terminal Answering Boss' Line Call

(a) It is necessary to accommodate the Boss' Line as a Sub Line to the Secretary's Desktop terminal.

(b) This feature is available only when the Secretary's Station is idle.

(c) When the Secretary's Station is busy, answer the Boss' line call through the following procedure:

STEP 1: Press the CALL PICKUP Feature key (pre-assigned to a Line/Feature key).
STEP 2: Press the flashing Boss' Line key.

(d) Office data assignment for CALL PICKUP-GROUP [C-7] is necessary since conditions of this feature are the same as those of general Call Pickup features. See CALL PICKUP-GROUP [C-7] for more information.

(e) An Overriding call from a Secretary's Station cannot be picked up by another Secretary's Station.

6.4 Conditions to Overriding from a Secretary's Station

(a) Only the Secretary's Station can Override to a Boss' Station.

(b) The destination for Secretary Override must be the Boss' Station associated with its Secretary Group.

(c) If the Secretary's Station has Executive Right-of-Way (Calling) restricted (ASFC SFI 3=restricted), when attempting to directly Override to a busy Boss' Station, the Secretary's Station will receive Reorder Tone (ROT).
- Pre-assign office data for Executive Right-of-Way.

(d) Override is restricted when the Boss' Station is in busy status due to reasons other than an ongoing call (exception: call via an Attendant Console).

(e) If a busy Secretary's Station presses the Boss' Line key to set Call Waiting, there will be no resulting operation if Call Waiting is restricted by office data settings. LCD will display the service as restricted.

(f) If Call Waiting is restricted by office data settings, the following Secretary will receive Reorder Tone (ROT):
A busy Secretary's Station is receiving Special Dial Tone (SPDT) after pressing the CALL FORWARDING Feature key. While receiving SPDT, the same Secretary presses the Boss' Line key to set Call Waiting.
- Pre-assign office data for Call Waiting.
- When forwarding a Station call, assign settings to allow Call Waiting-Originating (Calling) to that Station.
- When forwarding a trunk call, assign Call Waiting Service to that Route Class (CDN=46).

(g) MULTI-BOSS/MULTI-SECRETARY is the only feature that allows Override termination. If other call forwarding features are set, those features will function for an incoming call. The incoming call will be forwarded instead of Overriding.

6.5 Conditions for Overriding from a Busy Secretary's Station

(a) Office data assignment of EXECUTIVE RIGHT-OF-WAY [E-1] is necessary since conditions for Override are the same as those of general Override features. See EXECUTIVE RIGHT-OF-WAY [E-1] for more information.

(b) Pressing the CALL PARK DELUXE/RETRIEVE-1 Feature key (FKY=158: pre-assigned to a Line/Feature key) changes a busy Secretary Station's status to idle. Thus, when the Secretary's Station is busy, Override to the Boss' Station through the following procedure:

STEP 1: Press the CALL PARK DELUXE/RETRIEVE-1 Feature key (FKY=158: pre-assigned to a Line/Feature key).

STEP 2: Press the flashing Boss' Line key.

(c) Override to a busy Boss' Station is not available from a Stand-In Secretary's Station and a forwarding destination of MULTI-BOSS/MULTI-SECRETARY Overriding from a Secretary's Call For-

warding Destination while the Stand-In Secretary's Station and the forwarding destination are in busy state or holding another line.

6.6 List of Operations When Boss' Line Key is Pressed

(a) Boss' Line key is used for the following operations:
- Display of Boss' Line status
- Calling the Boss
- Answering the Boss' call
- Overriding to the Boss' Line
- Setting Call Waiting to the Boss' Station

(b) Only the Boss can seize the Boss' Line by pressing the Boss' Line key.

(c) Pressing the Boss' Line key from a terminal other than the Secretary's Terminal voids operations using the Boss' Line key.

(d) Refer to the following table for a list of operations when the Secretary presses the Boss' Line key in idle status.
- The following abbreviations are used in the table below.
  - CF-A= CALL FORWARDING-ALL CALLS
  - CF-B= CALL FORWARDING-BUSY LINE
  - CF-D= CALL FORWARDING-DON'T ANSWER

- "Sending DT" in Status of Secretary's Line (My Line) column includes:
  - state of receiving Dial Tone (DT).
  - state of not receiving DT during dialing.

List of Operations When Boss' Line Key is Pressed

Status of Secre-tary	Status of Secre-tary's Line (My Line)	Status of Boss	Status of Boss' Line Key Lamp	Status of CF Set to Boss' Station	Result of Operation
Idle	Idle/ Sending DT	Idle	Off	Not set	Boss is called using Secretary's My Line.
				CF-A	Call is routed to another Station using Secretary's My Line.
				CF-B	Call is forwarded to Boss' Station using Secretary's My Line.
				CF-D	After Boss is called using Secretary's My Line, predesignated Station is called.
Idle	Idle/ Sending DT	2-Party Call	On	Not set	Override
				CF-A	
				CF-B	
				CF-D	

List of Operations When Boss' Line Key is Pressed

Status of Secre-tary	Status of Secre-tary's Line (My Line)	Status of Boss	Status of Boss' Line Key Lamp	Status of CF Set to Boss' Station	Result of Operation
Idle	Idle/ Sending DT	Receiving Call	Flashing	Not set	Answer by Call Pickup using Secretary's My Line.
				CF-A	
				CF-B	
				CF-D	
Idle	Idle/ Sending DT	Busy (due to reasons other than above)	On/ Flashing	Not set	Restricted
				CF-A	
				CF-B	
				CF-D	
Idle	Other than Idle/ Sending DT	Idle	Off	Not set	No Operation
				CF-A	
				CF-B	
				CF-D	
Idle	Other than Idle/ Sending DT	2-Party Call	On	Not set	No Operation
				CF-A	
				CF-B	
				CF-D	
Idle	Other than Idle/ Sending DT	Receiving Call	Flashing	Not set	No Operation
				CF-A	
				CF-B	
				CF-D	
Idle	Other than Idle/ Sending DT	Busy (due to reasons other than above)	On/ Flashing	Not set	No Operation
				CF-A	
				CF-B	
				CF-D	

(e) Refer to the following table for a list of operations when the Secretary presses the Boss' Line key in call forwarded, hold or busy status.

- The following abbreviations are used in the table below.
  - CF-A= CALL FORWARDING-ALL CALLS

- CF-B= CALL FORWARDING-BUSY LINE
- CF-D= CALL FORWARDING-DON'T ANSWER

- Even if the Boss' Line key is flashing, it is impossible to pick up a call unless the Boss' Line is actually receiving a call.

- Conditions for Overriding and Answering by Call Pickup are the same as general Override features.

- "2-Party Call:" state of 2-Party Call (excluding connection to an Attendant Console) where it is possible to perform switch hook flash.
- Status of Secretary's Line (My Line) column:
  - "TRANSFER key pressed:" TRANSFER key is pressed during a 2-party call and Special Dial Tone is received.
  - "Idle due to Call Park:" the Secretary's line is idle after holding an ongoing call using Call Park.

List of Operations When Boss' Line Key is Pressed

Status of Secretary	Status of Secretary's Line (My Line)	Status of Boss	Display of Boss' Line Key	Status of CF Set to Boss' Station	Result of Operation
Transfer	2-Party Call	Idle	Off	Not set	Ongoing call to Secretary is held. Boss is called.
				CF-A	Ongoing call to Secretary is held. Call is routed to another Station.
				CF-B	Ongoing call to Secretary is held. Call is forwarded to Boss' Station.
				CF-D	Ongoing call to Secretary is held. After Boss is called, predesignated Station is called.
Transfer	2-Party Call	2-Party Call	ON	Not set	Call Waiting is set.
				CF-A	
				CF-B	
				CF-D	
Transfer	2-Party Call	Receiving Call	Flashing	Not set	No Operation
				CF-A	
				CF-B	
				CF-D	

List of Operations When Boss' Line Key is Pressed

Status of Secretary	Status of Secretary's Line (My Line)	Status of Boss	Display of Boss' Line Key	Status of CF Set to Boss' Station	Result of Operation
Transfer	2-Party Call	Busy (due to reasons other than above)	On/ Flashing	Not set	Restricted
				CF-A	
				CF-B	
				CF-D	
Transfer	TRANSFER key pressed	Idle	Off	Not set	Boss is called.
				CF-A	Call is routed to another Station.
				CF-B	Call is forwarded to Boss' Station.
				CF-D	After Boss is called, predesignated Station is called.
Transfer	TRANSFER key pressed	2-Party Call	On	Not set	Call Waiting is set.
				CF-A	
				CF-B	
				CF-D	
Transfer	TRANSFER key pressed	Receiving Call	Flashing	Not set	No Operation
				CF-A	
				CF-B	
				CF-D	
Transfer	TRANSFER key pressed	Busy (due to reasons other than above)	On/ Flashing	Not set	Restricted
				CF-A	
				CF-B	
				CF-D	
Hold	Idle due to Call Park	Idle	Off	Not set	Boss is called using Secretary's My Line.
				CF-A	Call is routed to another Station using Secretary's My Line.
				CF-B	Call is forwarded to Boss' Station using Secretary's My Line.
				CF-D	After Boss is called using Secretary's My Line, predesignated Station is called.

List of Operations When Boss' Line Key is Pressed

Status of Secretary	Status of Secre-tary's Line (My Line)	Status of Boss	Display of Boss' Line Key	Status of CF Set to Boss' Station	Result of Operation
Hold	Idle due to Call Park	2-Party Call	On	Not set	Override using Secretary's My Line.
				CF-A	
				CF-B	
				CF-D	
Hold	Idle due to Call Park	Receiving Call	Flashing	Not set	Answer by Call Pickup using Secretary's My Line.
				CF-A	
				CF-B	
				CF-D	
Hold	Idle due to Call Park	Busy (due to reasons other than above)	On/ Flashing	Not set	Restricted
				CF-A	
				CF-B	
				CF-D	
Busy	Other than above	Idle	Off	Not set	No Operation
				CF-A	
				CF-B	
				CF-D	
Busy	Other than above	Busy	On	Not set	No Operation
				CF-A	
				CF-B	
				CF-D	
Busy	Other than above	Receiving Call	Flashing	Not set	No Operation
				CF-A	
				CF-B	
				CF-D	

List of Operations When Boss' Line Key is Pressed

Status of Secretary	Status of Secre-tary's Line (My Line)	Status of Boss	Display of Boss' Line Key	Status of CF Set to Boss' Station	Result of Operation
Busy	Other than above	Busy (due to reasons other than above)	On/ Flashing	Not set	No Operation
				CF-A	
				CF-B	
				CF-D	

7. Conditions for MULTI-BOSS/MULTI-SECRETARY AWAY PER BOSS

7.1 Conditions

(a) This feature is unavailable for SOFT PHONE WIRED-PHONE HANDSET [S-147].

(b) The maximum number for registered Secretary Terminals per Boss is 21 terminals.

(c) This feature notifies the setting/cancelling status of MULTI-BOSS/ MULTI-SECRETARY AWAY PER BOSS by Service Set Facility Request (SSFR) or by Service Set Facility Notification (SSFN) from OAI.

(d) SECRETARY ABSENT has priority over AWAY PER BOSS when the Secretary is ABSENT from all Secretary Groups it belongs to.

(e) A Secretary can set MULTI-BOSS/MULTI-SECRETARY AWAY PER BOSS to a maximum of 10 Bosses.

(f) If initialization process is executed, the setting for this feature is cancelled.

(g) The setting status for this feature will be saved even if the Terminal is logged out.

(h) Even if the setting were executed using a Sub Line, the feature is set to My Line.

(i) If setting MULTI-BOSS/ MULTI-SECRETARY AWAY PER BOSS is unavailable, due to reasons such as the Secretary not belonging to the Boss' Secretary Group, the user hears Reorder Tone (ROT). The LCD will display an error message.

(j) If a Boss Group becomes non-existent, settings for MULTI-BOSS/ MULTI-SECRETARY AWAY PER BOSS assigned to that Boss becomes void.

(k) If MULTI-BOSS/ MULTI-SECRETARY AWAY PER BOSS is set to a Boss Group which already has that feature set, the user hears Reorder Tone (ROT). LCD will display an error message.

(l) If the cancelling procedure is executed for a Boss Group where MULTI-BOSS/ MULTI-SECRETARY AWAY PER BOSS is not set, the user hears Reorder Tone (ROT). LCD will display an error message.

(m) MULTI-BOSS/ MULTI-SECRETARY AWAY PER BOSS cannot be set to the last available terminal in a Boss Group. When attempted, the user hears Reorder Tone (ROT). LCD will display an error message.

7.2 Interactions

(a) MULTI-BOSS/ MULTI-SECRETARY AWAY PER BOSS is also available for Uniform Call Distribution (UCD) Group by dialing the access code and\ pilot (controlling) Station's number from the member Station. The maximum number of Stations within a UCD Group is 21.

8. Conditions for MULTI-BOSS/MULTI-SECRETARY OVERRIDING FROM A SECRETARY'S CALL FORWARDING DESTINATION

8.1 Conditions

(a) Stations including Standard SIP terminals are available to be used as a forwarding destination.

(b) When the Secretary's Station sets a station as a Call Forwarding destination, Overriding the Boss' Station line from a Secretary's Call Forwarding Destination is activated.
When the Secretary's Station sets a trunk line as a Call Forwarding destination, a call to Boss' Station slides to the Secretary's Station although the destination performs the Overriding operation after answering the call.

(c) When high-order Secretary's Stations have CALL FORWARDING - ALL CALLS [C-5]/CALL FORWARDING - DON'T ANSWER [C-3] set, a call does not slide to low-order Secretary's Stations.

(d) There is no limit to the number of Secretary's Stations which can have CALL FORWARDING - ALL CALLS [C-5]/CALL FORWARDING - DON'T ANSWER [C-3] set.

8.2 Interactions

(a) This feature supports only CALL FORWARDING - ALL CALLS [C-5]/CALL FORWARDING - DON'T ANSWER [C-3]. When a forwarding destination of CALL FORWARDING - BUSY LINE [C-2] answers a call and places the call to the Boss' Station, the call slides to the Secretary's Station.

(b) This feature cannot be used by a Secretary's Station which has SPLIT CALL FORWARDING [S-99] set.

(c) Calls cannot be placed to the Boss' Station in the following cases.

- An incoming call is forwarded to an unexpected destination which is not a forwarding destination of CALL FORWARDING - ALL CALLS [C-5]/CALL FORWARDING - DON'T ANSWER [C-3] set by the Secretary's Station due to MULTIPLE CALL FORWARDING - ALL CALLS [M-44].
- A forwarded incoming call is picked up with the Call Pickup features by an unexpected station which is not a forwarding destination of CALL FORWARDING - ALL CALLS [C-5]/CALL FORWARDING - DON'T ANSWER [C-3] set by the Secretary's Station.

- A forwarded incoming call is answered by an unexpected station which is not a forwarding destination of CALL FORWARDING - ALL CALLS [C-5]/CALL FORWARDING - DON'T ANSWER [C-3] set by the Secretary's Station due to UNIFORM CALL DISTRIBUTION (UCD) [U-1]/Station Hunting features.

(d) Do not set any other Call Forwarding features to a forwarding destination of CALL FORWARDING - ALL CALLS [C-5]/CALL FORWARDING - DON'T ANSWER [C-3] set by the Secretary's Station.

(e) Multi-Boss/Multi-Secretary queuing is not supported by a forwarding destination which the Secretary's Station sets.

(f) When a forwarding destination set by the Secretary's Station is Attendant Console/Night Attendant Console, this feature is not supported. The call placed by the forwarding destination to Boss' Station slides to the Secretary's Station.

(g) MULTI-BOSS/MULTI-SECRETARY-ONE TOUCH: Overriding from a Busy Secretary's Station is available only with Secretary's Stations. Overriding is unavailable from a forwarding destination while holding another line.

(h) When a forwarding destination of CALL FORWARDING - ALL CALLS [C-5] is in busy state or the forwarding destination has DO NOT DISTURB -D [D-11D] set, the Call Forwarding feature is not activated. An incoming call slides to the low-order Secretary's Station.

(i) When the Boss' Station is in busy state and activates BOSS ABSENT, an incoming call slides to the low-order Secretary's Station even if the high-order Secretary's Station has Call Forwarding features set.

(j) When the Boss' Station is in idle state and deactivates BOSS ABSENT, an incoming call arrives at the Boss' Station.

(k) The table shows behaviors depending on the forwarding destination of CALL FORWARDING - ALL CALLS [C-5] set by the Secretary's Station.

<table>
<tr><th>Forwarding Destination State</th><th>Boss Secretary Group which Forwarding Destination Belongs</th><th>Behavior</th></tr>
<tr><td>Boss' Station with BOSS ABSENT set</td><td rowspan="2">Same group as the Secretary's Station.</td><td rowspan="3">The Call Forwarding feature is inactivated.
The call slides to the low-order Secretary's Station in the same Boss Secretary Group.</td></tr>
<tr><td>Secretary's Station with SECRETARY ABSENT set</td></tr>
<tr><td>Boss' Station with BOSS ABSENT set</td><td>Different group from the Secretary's Station.</td></tr>
<tr><td>Secretary's Station with SECRETARY ABSENT set</td><td>Different group from the Secretary's Station.</td><td>Call Forwarding is activated.
The call is forwarded to the forwarding destination.</td></tr>
</table>

(l) The table shows behaviors when CALL FORWARDING - ALL CALLS [C-5]/CALL FORWARDING - DON'T ANSWER [C-3] is set to the Secretary's Station along with any of the following features. Also shows behaviors when the Secretary's Station becomes any of these states.

<table>
<tr><th rowspan="2">Features/State</th><th colspan="2">Behavior</th></tr>
<tr><th>When CALL FORWARDING - ALL CALLS [C-5] is Set</th><th>When CALL FORWARDING - DON'T ANSWER [C-3] is Set</th></tr>
<tr><td>MOBILITY ACCESS (MA) [M-118] (MA CALL FORWARDING)</td><td rowspan="3">CALL FORWARDING-ALL CALLS [C-5] is activated.</td><td rowspan="8">The call slides to a low-order Secretary's Station.</td></tr>
<tr><td>LOGGED OUT IP STATION - CALL DESTINATION [L-53] (Secretary's Station logs out.)</td></tr>
<tr><td>DO NOT DISTURB -D [D-11D]</td></tr>
<tr><td>UNIFORM CALL DISTRIBUTION (UCD) [U-1] (Busy Out)</td><td rowspan="3">The call slides to a low-order Secretary's Station.</td></tr>
<tr><td>MULTI-BOSS/MULTI-SECRETARY AWAY PER BOSS</td></tr>
<tr><td>Secretary’s Station is in lockout.</td></tr>
<tr><td>NUMBER SHARING [N-45]</td><td>CALL FORWARDING-ALL CALLS [C-5] is activated.</td></tr>
<tr><td>Secretary’s Station is in busy state</td><td>The call slides to a low-order Secretary's Station.</td></tr>
</table>

9. Conditions for MULTI-BOSS/MULTI-SECRETARY BOSS’ CALL FORWARDING ACTIVATE

9.1 Conditions

(a) Stations including Standard SIP terminals are available to be used as a forwarding destination.

(b) When a Secretary’s Station sets CALL FORWARDING - ALL CALLS [C-5]/CALL FORWARDING - DON'T ANSWER [C-3] and sets a station as a forwarding destination in the same office, terminating a call to Boss’ Station by Overriding is allowed (See “11. MULTI-BOSS/MULTI-SECRETARY Overriding from a Secretary’s Call Forwarding Destination”). When the Secretary’s Station sets a trunk line as a forwarding destination, terminating a call to Boss’ Station by Overriding is restricted.

(c) When the Boss’ Station has CALL FORWARDING - DON’T ANSWER - OUTSIDE [C-74] set, incoming call to the Boss’ Station slides to the Secretary’s Station. Even if the Secretary’s Station answers the sliding call, performs switch hook flash, rings the Boss’ Station and the Boss’ Station does not answer the call, CALL FORWARDING - DON’T ANSWER - OUTSIDE [C-74] is not activated and the Secretary’s Station continues ringing the Boss’ Station.

9.2 Interactions

(a) This feature cannot be used by a Boss' Station which has SPLIT CALL FORWARDING [S-99] set.

(b) Boss' Station needs to be in login status.

(c) The following shows the order of priority for Call Forwarding features in the Boss' Station when a Boss' Station has set the following features:

Note: Order of priority may be changed by the following data assignments.
- ASFC/ASFCN, SFI=209, RES=1
- (Priority of (3) Call Forwarding-All Calls > (2) Logged Out IP Station - Call Destination)
- ASFC/ASFCN, SFI=240, RES=1
- (Change of the priority for (4) MA Call Forwarding > (2) Logged Out IP Station - Call Destination/Call Forwarding-All Calls)

Order of Priority When BOSS ABSENT is Set
(MULTI-BOSS/MULTI-SECRETARY BOSS' CALL FORWARDING ACTIVATE is In Service)

- When a Third Party Places a Call to Boss' Station

Call Forwarding Features	
(1) NUMBER SHARING [N-45] (When call forwarding is assigned to a sub-station) (2) LOGGED OUT IP STATION-CALL DESTINATION [L-53] (6) MULTI-BOSS/MULTI-SECRETARY [M-120] **Note 4** (3) CALL FORWARDING-ALL CALLS [C-5] (4) MOBILITY ACCESS (MA) [M-118] (MA CALL FORWARDING) **Note 6** (5) SURVIVABLE REMOTE MGC [S-141] (Automatic Re-routing when SR-MGC is in Operating Mode) (7) UNIFORM CALL DISTRIBUTION (UCD) [U-1] (8) CALL FORWARDING-BUSY LINE [C-2] (9) STATION HUNTING-TERMINAL [S-9] (10) DO NOT DISTURB -D [D-11D] (11) CALL FORWARDING-DON'T ANSWER [C-3]	**High Priority**  **Low Priority**

- When Secretary's Station Places a Call to Boss' Station

Call Forwarding Features	
(1) NUMBER SHARING [N-45] (When call forwarding is assigned to a sub-station) (2) LOGGED OUT IP STATION-CALL DESTINATION [L-53] (3) CALL FORWARDING-ALL CALLS [C-5] **Note 5** (6) MULTI-BOSS/MULTI-SECRETARY [M-120] (4) MOBILITY ACCESS (MA) [M-118] (MA CALL FORWARDING) **Note 6** (5) SURVIVABLE REMOTE MGC [S-141] (Automatic Re-routing when SR-MGC is in Operating Mode) (7) UNIFORM CALL DISTRIBUTION (UCD) [U-1] (8) CALL FORWARDING-BUSY LINE [C-2] (9) STATION HUNTING-TERMINAL [S-9] (10) DO NOT DISTURB -D [D-11D] (11) CALL FORWARDING-DON'T ANSWER [C-3]	**High Priority** ↕ **Low Priority**

Note 4: When a third party places a call to the Boss' Station, CALL FORWARDING - ALL CALLS [C-5] set to the Boss' Station is not activated, but the call slides to the Secretary's Station according to MULTI-BOSS/MULTI-SECRETARY [M-120].

Note 5: When the Secretary's Station places a call to the Boss' Station, CALL FORWARDING - ALL CALLS [C-5] set to the Boss' Station is activated without Sliding Action.

Note 6: MOBILITY ACCESS (MA) [M-118] (MA CALL FORWARDING) is not activated when the Boss' Station receives an incoming call by DIRECT-IN TERMINATION (DIT) [D-7].

Order of Priority When BOSS ABSENT is Set
(MULTI-BOSS/MULTI-SECRETARY BOSS' CALL FORWARDING ACTIVATE is Out of Service)

Call Forwarding Features	
(1) NUMBER SHARING [N-45] (When call forwarding is assigned to a sub-station) (2) LOGGED OUT IP STATION-CALL DESTINATION [L-53] (3) CALL FORWARDING-ALL CALLS [C-5] (4) MOBILITY ACCESS (MA) [M-118] (MA CALL FORWARDING) **Note 6** (5) SURVIVABLE REMOTE MGC [S-141] (Automatic Re-routing when SR-MGC is in Operating Mode) (6) MULTI-BOSS/MULTI-SECRETARY [M-120] (7) UNIFORM CALL DISTRIBUTION (UCD) [U-1] (8) CALL FORWARDING-BUSY LINE [C-2] (9) STATION HUNTING-TERMINAL [S-9] (10) DO NOT DISTURB -D [D-11D] (11) CALL FORWARDING-DON'T ANSWER [C-3]	**High Priority** ↕ **Low Priority**

See the following table when the order of priority is changed by the ASFC/ASFCN SFI=209/240 Data Assignment.

Call Forwarding Features
(1) NUMBER SHARING [N-45] (When call forwarding is assigned to a sub-station)
(2) LOGGED OUT IP STATION-CALL DESTINATION [L-53]*
(3) CALL FORWARDING-ALL CALLS [C-5]*
(4) MOBILITY ACCESS (MA) [M-118] (MA CALL FORWARDING) **Note 6**
(5) SURVIVABLE REMOTE MGC [S-141] (Automatic Re-routing when SR-MGC is in Operating Mode)
(6) MULTI-BOSS/MULTI-SECRETARY [M-120]*
(7) UNIFORM CALL DISTRIBUTION (UCD) [U-1]
(8) CALL FORWARDING-BUSY LINE [C-2]
(9) STATION HUNTING-TERMINAL [S-9]
(10) DO NOT DISTURB -D [D-11D]
(11) CALL FORWARDING-DON'T ANSWER [C-3]

***Variable**

ASFC/ ASFCN SFI=209	ASFC/ ASFCN SFI=240	Boss' Station Status (Note 7)	MULTI-BOSS/MULTI-SECRETARY BOSS' CALL FORWARDING ACTIVATE	
			In Service (ASYDL/ASYDN Index 1089 Bit 5=1)(Note 8)	Out of Service (ASYDL/ASYDN Index 1089 Bit 5=0)
			Priority (High -> Low)	Priority (High -> Low)
SFI 209 = 0 (2) > (3)	SFI 240 = 0 (2)/(3) > (4)	Login	(1)**(6)(3)**(4)(7)(8)(9)(10)(11)	(1)**(3)**(4)**(6)**(7)(8)(9)(10)(11)
		Logout	(1)**(2)(3)(6)**(4)(5)(7)(9)(10)	(1)**(2)(3)**(4)(5)**(6)**(7)(9)(10)
	SFI 240 = 1 (4) > (2)/(3)	Login	(1)(4)**(6)(3)**(7)(8)(9)(10)(11)	(1)(4)**(3)(6)**(7)(8)(9)(10)(11)
		Logout	(1)(4)**(2)(3)(6)**(5)(7)(9)(10)	(1)(4)**(2)(3)**(5)**(6)**(7)(9)(10)
SFI 209 = 1 (3) > (2)	SFI 240 = 0 (2)/(3) > (4)	Login	(1)**(6)(3)**(4)(7)(8)(9)(10)(11)	(1)**(3)**(4)**(6)**(7)(8)(9)(10)(11)
		Logout	(1)**(6)(3)(2)**(4)(5)(7)(9)(10)	(1)**(3)(2)**(4)(5)**(6)**(7)(9)(10)
	SFI 240 = 1 (4) > (2)/(3)	Login	(1)(4)**(6)(3)**(7)(8)(9)(10)(11)	(1)(4)**(3)(6)**(7)(8)(9)(10)(11)
		Logout	(1)(4)**(6)(3)(2)**(5)(7)(9)(10)	(1)**(3)(2)**(4)(5)**(6)**(7)(9)(10)

Note 7: See the line of "Login" in the "Boss' Station Status" column when the Boss' Secretary Station is a non-IP terminal.

Note 8: When a third party places a call to the Boss' Station, CALL FORWARDING - ALL CALLS [C-5] set to the Boss' Station is not activated, but the call slides to the Secretary's Station according to MULTI-BOSS/MULTI-SECRETARY [M-120].
When the Secretary's Station places a call to the Boss' Station, CALL FORWARDING - ALL CALLS [C-5] set to the Boss' Station is activated without Sliding Action.

(i) Here shows a condition when a Boss' Station has CALL FORWARDING - ALL CALLS [C-5]/ CALL FORWARDING - DON'T ANSWER [C-3] set and the forwarding destination is another Boss' Station belonging to a different Boss Secretary Group.

When there are multiple Boss Secretary Groups in the following states:

[Group A]

- Member: Boss-A, Secretary 1-A, Secretary 2-A, Secretary 3-A, Stand-In Secretary-A
- Boss-A has CALL FORWARDING-ALL CALLS [C-5] set (Forwarding Destination: Boss-B).

[Group B]

- Member: Boss-B, Secretary 1-B, Secretary 2-B, Secretary 3-B
- Boss-B has BOSS ABSENT set.

Secretary 1-A/Stand-In Secretary-A places a call to the Boss-A, CALL FORWARDING-ALL CALLS [C-5] set to Boss-A is activated and the call is forwarded to Boss-B. The forwarded call to the Boss-B slides to the Secretary 1-B due to BOSS ABSENT set to Boss-B. **Note 9**

Note 9: When MULTI-BOSS/MULTI-SECRETARY Overriding from a Secretary's Call Forwarding Destination is in service, the call arrives at Secretary 1-B can be forwarded to a destination of CALL FORWARDING-ALL CALLS [C-5]/CALL FORWARDING - ALL CALLS - OUTSIDE [C-28] which the Secretary 1-B sets.

10. Conditions for MULTI-BOSS/MULTI-SECRETARY OVERRIDING FROM A STAND-IN SECRETARY

10.1 Conditions

(a) Up to seven (two by default) Stand-In Secretary's Stations per Boss Secretary group can be set by the system data assignment (ASYDL/ASYDN SYS1 Index 1089 Bit 0-2).

(b) Do not include a station accommodated in a different node in a Boss Secretary group.

10.2 Interactions

(a) This feature cannot be used by a Secretary's Station which has SPLIT CALL FORWARDING [S-99] set.

(b) Multi-Boss/Multi-Secretary queuing does not support Stand-In Secretary's Station.

(c) MULTI-BOSS/MULTI-SECRETARY-ONE TOUCH: Overriding from a Busy Secretary's Station is available for the Secretary's Station only. Overriding from a Stand-In Secretary's Station in a busy state is not available.

11. Conditions for MULTI-BOSS/MULTI-SECRETARY MULTI LINE OPERATION

11.1 Conditions

(a) To set BOSS or SECRETARY ABSENT by pressing a Line/Feature key or a Soft key after seizing the Main Boss' or the Main Secretary's Station Line on the Substitute Station, assign the key for UCD Busy Out (FKY=19) to a Line/Feature key or a Soft key of the Substitute Station.

(b) To enable the MULTI-BOSS/MULTI-SECRETARY-ONE TOUCH: Overriding from a Busy Secretary's Station feature by seizing the Main Secretary's Station Line from the Substitute Station, assign a Call Park Group Number (GPN) to the Substitute Station using the ACPSN command.

(c) Do not assign the Main Boss' Station Line to the Main Secretary's Station as a Prime Line.

11.2 Interactions

(a) This feature cannot be used in combination with PS Line Appearance on Dterm [P-71] and SIP Multiple Line Operation [S-160].

(b) Even if Boss Secretary Multi Line Operation is enabled, a Boss' Station Line cannot be seized from a Secretary's Station if the Boss and the Secretary belong to different Boss Secretary Groups.
If a Secretary presses a Boss' Line whose Boss belongs to a different Boss Secretary Group, key operation becomes invalid (there is no reaction).

Note: If a Secretary presses a Boss' Station Line whose Boss and the Secretary belong to the same Boss Secretary Group, MULTI-BOSS/MULTI-SECRETARY-ONE TOUCH is activated.

Hardware and Firmware Conditions

None

Programming

1. Common Programming

STEP 1: **ASYDL/ASYDN** - System Data 1, Index 1140, Bit 2 (Group Number Expansion for Multi Boss Secretary Group): 0/1 = Out of service/In service

STEP 2: **ASYD**-System Data 1, Index 60, Bit 3=0/1: UCD Queuing is provided/Not provided.
Select UCD queuing if necessary. **Note 1**

STEP 3: **ASHUEN [Network Data Memory (NDM)]**-Assign Telephone Number for Secretary Group.

UGN: User Group Number (1-255)
Assign "1" if Dial Plan to support IP Centrex Functionality is not used.

TELN: Telephone Number (maximum 16 digits)
Assign Telephone Number of Boss' Station. **Note 2**

Kind of Group 0: Multi Group Station - UCD
1: Multi-Boss/Multi-Secretary Group

[Station List in Group]

TELN: Telephone Number of UCD Member Station

FPC: Fusion Point Code (1-253)
FPC where each Station belongs is displayed.

Note 1: Refer to UCD Queuing service assignment regarding queuing features.

Note 2: Turn ON checkbox for "Multi Boss Secretary Group".

2. Programming for Boss' Station

STEP 1: **ASFC/ASFCN** - Allow Service Feature Restriction Class (SFC) for this feature.
SFI 252=1 (MULTI-BOSS/MULTI-SECRETARY)

STEP 2: **ANPD/ANPDN** - Assign the minimum number of digits that is required to the first digit received (pre-translation) to determine that the Access Code is for BOSS ABSENT.

STEP 3: **ASPA/ASPAN** - Assign Access Code to set BOSS ABSENT. **Note 1**

- CI="N: Normal"
  SRV="SSCA: Service Code Appendix"
  SIDA=50 (Set BOSS ABSENT)

- CI="N: Normal"
  SRV="SSCA: Service Code Appendix"
  SIDA=51 (Cancel BOSS ABSENT)

Note 1:
- To use the Line/Feature key:
  AKYD (Assignment of key data for Desktop Terminal)
  FKY=19 (Assign/Cancel UCD Busy Out)

- To use the Soft key:
  ADSL [Assignment of Desktop Terminal Soft key on LCD (LDM)]
  SN=0/4 (Line is idle/Line is busy)
  FKY=19 (Assign/Cancel UCD Busy Out)

3. Programming for Secretary's Station

STEP 1: **ASFC/ASFCN** - Allow Service Feature Restriction Class (SFC) for this feature.
SFI 252=1 (MULTI-BOSS/MULTI-SECRETARY)

STEP 2: **ANPD/ANPDN** - Assign the minimum number of digits that is required to the first digit received (pre-translation) to determine that the Access Code is to set SECRETARY ABSENT or BOSS ABSENT from Secretary's Station.

STEP 3: **ASPA/ASPAN**

Assign Access Code for SECRETARY ABSENT. **Note 1**

- CI="N: Normal"
  SRV="SSCA: Service Code Appendix"
  SIDA=50 (Set SECRETARY ABSENT)

- CI="N: Normal"
  SRV="SSCA: Service Code Appendix"
  SIDA=51 (Cancel SECRETARY ABSENT)

Assign Access Code for BOSS ABSENT from Secretary's Station.

- CI="N: Normal"
  SRV="SSCA: Service Code Appendix"
  SIDA=164 Set BOSS ABSENT from Secretary's Station

- CI="N: Normal"
  SRV="SSCA: Service Code Appendix"
  SIDA=165 Cancel BOSS ABSENT from Secretary's Station

Note 1:
- To use the Line/Feature key:
  AKYD (Assignment of Key Data for Desktop Terminal)
  FKY=19 (Assign/Cancel UCD Busy Out)

- To use the Soft key:
  ADSL [Assignment of Desktop Terminal Soft key on LCD (LDM)]
  SN=0/4 (Line is idle/Line is busy)
  FKY=19 (Assign/Cancel UCD Busy Out)

4. Programming for MULTI-BOSS/MULTI-SECRETARY-ONE TOUCH

STEP 1: **AKYD/AKYD_T** - Assign Key Data for Desktop terminal.

- This command assigns Boss' Station number in addition to My Line for Secretary's Station's multi-line.
- Assigning another Secretary's line as a multi-line is also possible.
- The above assumes that the Telephone Numbers for Boss' Station and Secretary's Station are pre-assigned.

STEP 2: **ASHUEN [Network Data Memory (NDM)]**-Assign Telephone Number for Secretary Group.

UGN: User Group Number (1-255)
Assign "1" if Dial Plan to support IP Centrex Functionality is not used.

TELN: Telephone Number (maximum 16 digits)
Assign Telephone Number of Boss' Station. **Note 1**

Kind of Group 0: Multi Group Station - UCD
1: Multi-Boss/Multi-Secretary Group

[Station List in Group]

TELN: Telephone Number of UCD Member Station

FPC: Fusion Point Code (1-253)
FPC where each Station belongs is displayed.

STEP 3: **ASYDL/ASYDN** - System Data 1, Index 1140, Bit 1=1: Assignment of Boss Line to Secretary Terminal's Sub Line enabled. **Note 2**, **Note 3**

Note 1: Turn ON checkbox for "Multi Boss Secretary Group."

Note 2: When executing Direct Call Pickup, Override and Call Waiting features using the Boss' Line key, refer to each feature entry regarding office data settings such as system data or SFI. Assign if necessary.

Note 3: Keep in mind the following when assigning office data for Call Waiting features:

- When Call Forwarding a Station call, allow Service Feature Restriction Class (SFC) for Call Waiting-Originating (Calling) to that Station.
- When Call Forwarding a trunk call, allow Call Waiting to its route class (CDN=46).

4.1 Overriding from a Secretary's Station

STEP 1: **ASFC/ASFCN** - Allow Service Feature Restriction Class (SFC) for this feature.

- Allow Service Feature Restriction Class (SFC) for Overriding.
  - Assignment for Boss' Station:
    SFI 4=1 [Executive Right-of-Way (Called)]
  - Assignment for Secretary's Station:
    SFI 3=1 [Executive Right-of-Way (Calling)]
- Allow Service Feature Restriction Class (SFC) for Call Park Deluxe feature.
  - Assignment for Secretary's Station:
    SFI 220=1 (Call Park Deluxe Entry)
  - Assignment for Station being Call "Parked:"
    SFI 68=1 (Call Park-Called)

STEP 2: **AKYD** - Assign Key Data for Desktop terminal.
- Assignment for Secretary's Station:
  FKY=158 (Call Park Deluxe-Hold/Retrieve-1)

STEP 3: **ASCS/ACPSL/ACPSN** - Assignment of Call Park Deluxe Group for Data Memory.

STN: Enter the Secretary's Station number.
GPN: Assignment of Call Park Group Number.
If one GPN is assigned per Secretary, the Line/Feature Key number for every Secretary's Desktop terminal can be unified to "158."

STEP 4: **ASYD** - System Data 1, Index 142: Assign Automatic Recall Timer for Call Park Deluxe service.
When "0" (default) is assigned, the timer is set to 500 seconds.

4.2 Overriding from a Busy Secretary's Station

To use this feature, assign the same data as "4.1 Overriding from a Secretary's Station"

4.3 Secretary's Terminal Answering Boss' Line Call

STEP 1: **ASFC/ASFCN** - Allow Service Feature Restriction Class (SFC) for Direct Call Pickup.
- Assignment for Secretary's Station:
  SFI 29=1 (Direct Call Pickup)

4.4 Call Waiting

STEP 1: **ASFC/ASFCN** - Allow Service Feature Restriction Class (SFC) for Call Waiting-Originating.
- Assignment for Boss' Station:
  SFI 6=1 [Call Waiting Originating (Called)]
  SFI 11=1 (Data Line Security)
- Assignment for Station communicating with the Secretary's Station:
  SFI 5=1 [Call Waiting Originating (Calling)]

STEP 2: **ARTD/ARTDN** - Assign Route Class Data.
- Assignment for Station communicating with the Secretary's Station:
  Parameter CDN=46
  CW=1 (Call Waiting= In Service)

5. Programming for MULTI-BOSS/MULTI-SECRETARY AWAY PER BOSS

Setting MULTI-BOSS/MULTI-SECRETARY AWAY PER BOSS

STEP 1: Pre-assign data settings for MULTI-BOSS/ MULTI-SECRETARY.

STEP 2: **ASYDL/ASYDN** - System Data 1, Index 1088, Bit 4=1: MULTI-BOSS/MULTI-SECRETARY AWAY PER BOSS enabled.

STEP 3: **ANPD/ANPDN** - Assign Numbering Plan.
This command assigns the minimum number of digits needed to determine the service (Telephone Number) that is required by the first digit received (pre-translation).

STEP 4: **ASPA/ASPAN** - Assign Special Access Code.
- CI="N: Normal"
  SRV="SSCA: Service Code Appendix"
  SIDA=166 (MULTI-BOSS/ MULTI-SECRETARY AWAY PER BOSS=Set)

- CI="N: Normal"
  SRV="SSCA: Service Code Appendix"
  SIDA=167 (MULTI-BOSS/ MULTI-SECRETARY AWAY PER BOSS=Cancel)

Canceling MULTI-BOSS/MULTI-SECRETARY AWAY PER BOSS

STEP 1: Delete data by executing the above setting process in reverse order.

STEP 2: Delete data for MULTI-BOSS/ MULTI-SECRETARY.

6. Programming for MULTI-BOSS/MULTI-SECRETARY OVERRIDING FROM A SECRETARY'S CALL FORWARDING DESTINATION

STEP 1: Pre-assign data settings for MULTI-BOSS/ MULTI-SECRETARY.

STEP 2: **ASYDL/ASYDN** - System Data 1, Index 1089, Bit 4=1: MULTI-BOSS/MULTI-SECRETARY Overriding from a Secretary's Call Forwarding Destination is in service.

7. Programming for MULTI-BOSS/MULTI-SECRETARY BOSS' CALL FORWARDING ACTIVATE

STEP 1: Pre-assign data settings for MULTI-BOSS/ MULTI-SECRETARY.

STEP 2: **ASYDL/ASYDN** - System Data 1, Index 1089, Bit 5=1: MULTI-BOSS/MULTI-SECRETARY Boss' Call Forwarding Activate is in service.

8. Programming for MULTI-BOSS/MULTI-SECRETARY OVERRIDING FROM A STAND-IN SECRETARY

STEP 1: Pre-assign data settings for MULTI-BOSS/ MULTI-SECRETARY.

STEP 2: **ASYDL/ASYDN** - System Data 1, Index 1089,

Bit 3=1:
MULTI-BOSS/MULTI-SECRETARY Overriding from a Stand-In Secretary is in service.

Bit 0-2:
Specify the number of Stand-In Secretary's Stations to be used with MULTI-BOSS/ MULTI-SECRETARY Overriding from a Stand-In Secretary **Note 1**

BIT2	BIT1	BIT0	Stand-In Secretaries
0	0	0	Secretaries 1 - 2 (Default)
0	0	1	
0	1	0	
0	1	1	Secretaries 1 - 3
1	0	0	Secretaries 1 - 4
1	0	1	Secretaries 1 - 5
1	1	0	Secretaries 1 - 6
1	1	1	Secretaries 1 - 7

Note 1: This data is valid when ASYDL/ASYDN, System Data 1, Index 1089 Bit 3=1 is set.

9. Programming for MULTI-BOSS/MULTI-SECRETARY MULTI LINE OPERATION

STEP 1: Pre-assign data settings for MULTI-BOSS/ MULTI-SECRETARY.

STEP 2: Pre-assign data settings for the Main Boss' Station number or/and the Main Secretary's Station number to Multi Line of Substitute Station. **Note 1**, **Note 2**

STEP 3: **ASYDL/ASYDN** - System Data 1, Index 1089, Bit 6=0:
MULTI-BOSS/MULTI-SECRETARY Multi Line Operation is in service.

Note 1: To use the Line/Feature key, assign the key data of the Substitute Station:
AKYD (Assignment of Key Data for Desktop Terminal)
FKY=19 (Assign/Cancel UCD Busy Out)

Note 2: To use the Soft key, assign the key data of the Substitute Station:
ADSL [Assignment of Desktop Terminal Soft key on LCD (LDM)]
SN=0/4 (Line is idle/Line is busy)
FKY=19 (Assign/Cancel UCD Busy Out)

M-122 MG-SIP TIE Line

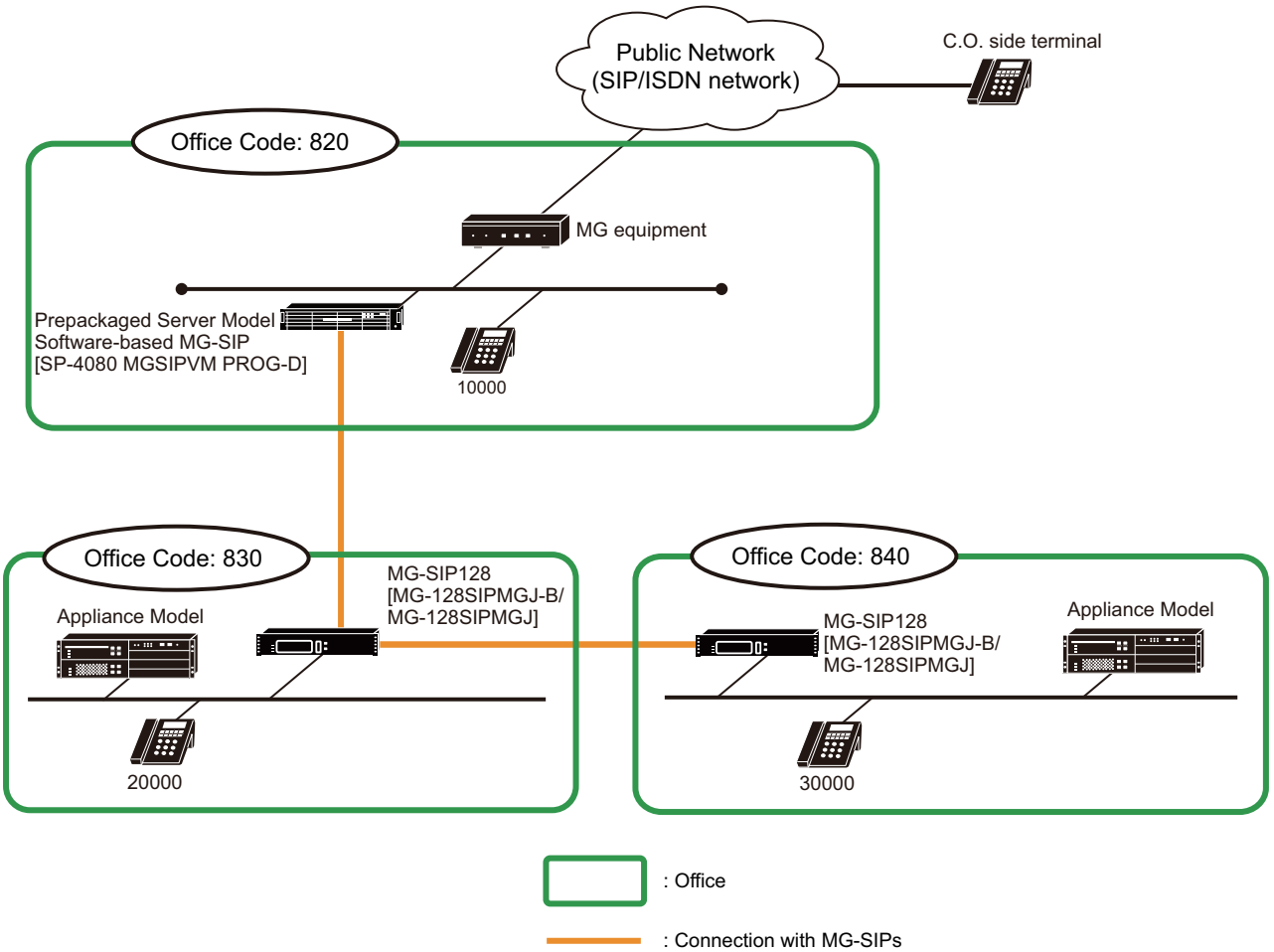
General Description

This feature allows multiple offices to be connected by TIE Line through MG-SIPs.
With this feature, user information of the incoming C.O. calls can be shared with all offices that are connected with MG-SIP TIE Line.

Note: For using this feature, MG-SIP128 [MG-128SIPMGJ-B/MG-128SIPMGJ] or Software-based MG-SIP [SP-4080 MGSIPVM PROG-D] is required.

Note: Note the following terms in this context.

- A C.O. line refers to PSTN (ISDN/SIP network) in this feature.
- A SIP network refers to a network connected with MG-SIPs that is available for TIE Line in this feature.
- MG-SIP refers to MG-128SIPMGJ-B/MG-128SIPMGJ or Software-based MG-SIP [SP-4080 MGSIPVM PROG-D] in this feature.



Operating Procedure

None.

Service Conditions

<Important Notice>

1. The MG-SIP128 [MG-128SIPMGJ-B/MG-128SIPMGJ] or Software-based MG-SIP [SP-4080 MG-SIPVM PROG-D] where this feature is enabled cannot connect to PSTN and SIP server. For using this feature, be sure to use an MG-SIP128 [MG-128SIPMGJ-B/MG-128SIPMGJ] or Software-based MG-SIP [SP-4080 MGSIPVM PROG-D] as an opposite device.
2. This feature does not support ETSI OVERLAP SENDING AND RECEIVING [E-35] from ISDN network.

<Interactions with Other Features>

1. This feature cannot be used with SIP Register or Addressing/COLP feature of ETSI ISDN ADDRESSING [E-32].
2. This feature can be used with MG-SIP Shared Route [M-123]. However, both MG-SIP128 [MG-128SIP-MGJ-B/MG-128SIPMGJ] and MG-SIP connected to C.O. line cannot be registered for the same single route.
3. This feature can be used with Call-ID Notification on SMFN for SIP. However, when autonomous alternate routing by MG-SIP is used with MG-SIP Multipoint feature, Call-ID is not changed.
4. From FP95-114 V4 Issue 2.0, RTP encryption on Telephony Server network side is supported. RTP encryption on SIP network side is not supported. For details, see VOIP ENCRYPTION [V-27].

<Hardware/Software>

1. For using this feature, MG-SIP128 [MG-128SIPMGJ-B/MG-128SIPMGJ] is required.
2. This feature supports incoming call from only the following C.O. trunks.
    - MG(BRI)
    - MG(PRI)
    - MG(SIP)
    - BRT
    - PRT

<Interactions with Networks>

1. This feature does not support the tandem connection or call forwarding with TIE Lines (such as CCIS) other than TIE Line via MG-SIP.
2. Consider the following when making a tandem connection from MG-SIP TIE Line to an external trunk (ISDN).
   a.) When ARTI/ARTIN INBAND (CDN:29)=0 (In-band information is relayed):
   When a station user attempts to transfer a call to an external trunk and then performs transfer switchback operation while hearing audible sound such as In-band Busy Tone or announcement, the station user may continue to hear the audible sound even after the transfer switchback.
   b.) When ARTI/ARTIN INBAND (CDN:29)=1 (In-band information is not relayed):
   Since In-band tone such as Busy Tone or announcement from a public network becomes difficult to be transmitted to MG-SIP TIE Line, the calling party may not hear the announcement from the public network.

<Conditions Specific to This Feature>

1. Internal PHI-BRI is essential to perform this feature. Internal PHI-BRI is built into the Telephony Server and controls the Voice Procedure of the MG-SIP via TCP/IP.
2. The number of available Internal PHI-BRIs is one per system.
   Note: In UMGi system, the data assignment of Internal PHI-BRI is required per node (CCN/RN).
3. Virtual Trunk can be registered to Physical PIR or Virtual PIR.
4. The maximum number of MG-SIP an accommodated per system and available channels are as follows. However, note that a maximum number of simultaneous speech channels available per LP, including other ISDN channels, is 460.

DEVICE TYPE	MAXIMUM ACCOMMODATION PER SYSTEM		
	1 LPs	4 LPs	Channels per system
MG-SIP	11 devices	44 devices	6144 channels

Note: The maximum number of devices and channels per system is equivalent to capacity of the system. The number of devices varies depending on the total accommodated channels.

Note: When 6144 channels are occupied only by MG-SIP, other devices cannot be connected.

5. Only Voice, FAX, DTMF (PB Signaling) communication can be established by connections via MG-SIP. This feature does not support Unrestricted digital, Modem connection, SIP Instant Message [S-151] etc.

ITEM	SPECIFICATION	NOTE
Voice	G.711	64k [bps]: PCM μ-law/A-law
FAX	G.711 (PassThrough)	64k [bps]: PCM μ-law
DTMF	G.711 (PassThrough)	
	Proprietary method	Toward the Telephony Server network
	RTP (RFC2833)	Toward SIP network For DTMF Digits, Telephony Tones and Telephony Signals

Note: Depending on IP equipment or conditions of SIP network, the above conditions may not be applied for MG-SIP.

Note: Be sure to use G.711 for voice CODEC.

Note: When DTMF is sent via RFC2833, RTP Warning Tone may play back to the reception side.

Note: DTMF relay method on the SIP network side is set by the configuration command "dtmf_mode" on the MG-SIP. DTMF relay method on the Telephony Server network side is automatically set according to the one on the MG-SIP network side, as follows.

DTMF relay system on the SIP network side	DTMF relay system on the Telephony Server network side	Remarks
RFC2833 (out-band)	Proprietary (out-band)	Conversions between "RTP (DTMF: RFC2833) packets on SIP network" and "UDP (DTMF: Proprietary) packets on Telephony Server network"
G.711 PassThrough (in-band)	G.711 PassThrough (in-band)	Conversions between "RTP packets on SIP network" and "RTP packets on Telephony Server network"

6. Additional services (such as REFER) on the SIP procedure are not provided in this system.
7. Tone and holding tone to SIP network is generated by MG-SIP. When MGC (Internal PHI-BRI) external holding tone is selected by the ASYDL command, the information is provided via IPPAD card.
8. A single route can be shared by multiple MG-SIPs (maximum 2 devices). For details, refer to MG-SIP Shared Route [M-123].
9. MG Registration Status can be displayed by using the DIMG command.
10. Because MGC transfers only Voice RTP Payload, it is not concerned for Jitter-Buffer Control, Echo Canceler and PAD Control. The connection is controlled by end points.

11. MG-SIP can receive incoming calls through AUTOMATED ATTENDANT [A-82]/REMOTE ACCESS TO SYSTEM [R-2] under the following conditions.

    - AUTOMATED ATTENDANT [A-82]/REMOTE ACCESS TO SYSTEM [R-2] may not work properly, depending on the system configuration of the SV9500 network and the conditions for DTMF specifications in SIP network. (**Note 1**)
      The conditions of system configuration are as follows:

DTMF in SIP Network	Telephony Server	SR-MGC	System Configuration
Fax PassThrough	Available*	Unavailable	Set ARTI/ARTIN CDN77(RA_RT) = 1 (Register card is used). *PIR is required to use IPPAD. *IPPAD is required.
RFC2833	Available	Available	Set ARTI/ARTIN CDN77(RA_RT) = 0 (Virtual Register is used). *PIR is not required because Virtual Register is used.

Note 1: The specifications for DTMF in a SIP network vary depending on a carrier.

    - PB Receiver becomes invalid while receiving an announcement for AUTOMATED ATTENDANT [A-82]/REMOTE ACCESS TO SYSTEM [R-2].
    - After dialing a target telephone number, a beep may be heard before Ringback Tone (RBT), but it's not abnormal.

12. SERIAL CALL -D [S-127D] is not available for incoming call via MG-SIP.
13. MG-SIP does not support ring down feature.
14. To use VLAN feature of MG-SIP, connect switch supporting Port VLAN feature (Tag VLAN is not supported).
15. To use FAX communication feature of MG-SIP, specify the FAX settings by the ARTI/ARTIN command. The FAX communication has limitations as follows:
    * For details, refer to Command Manual.

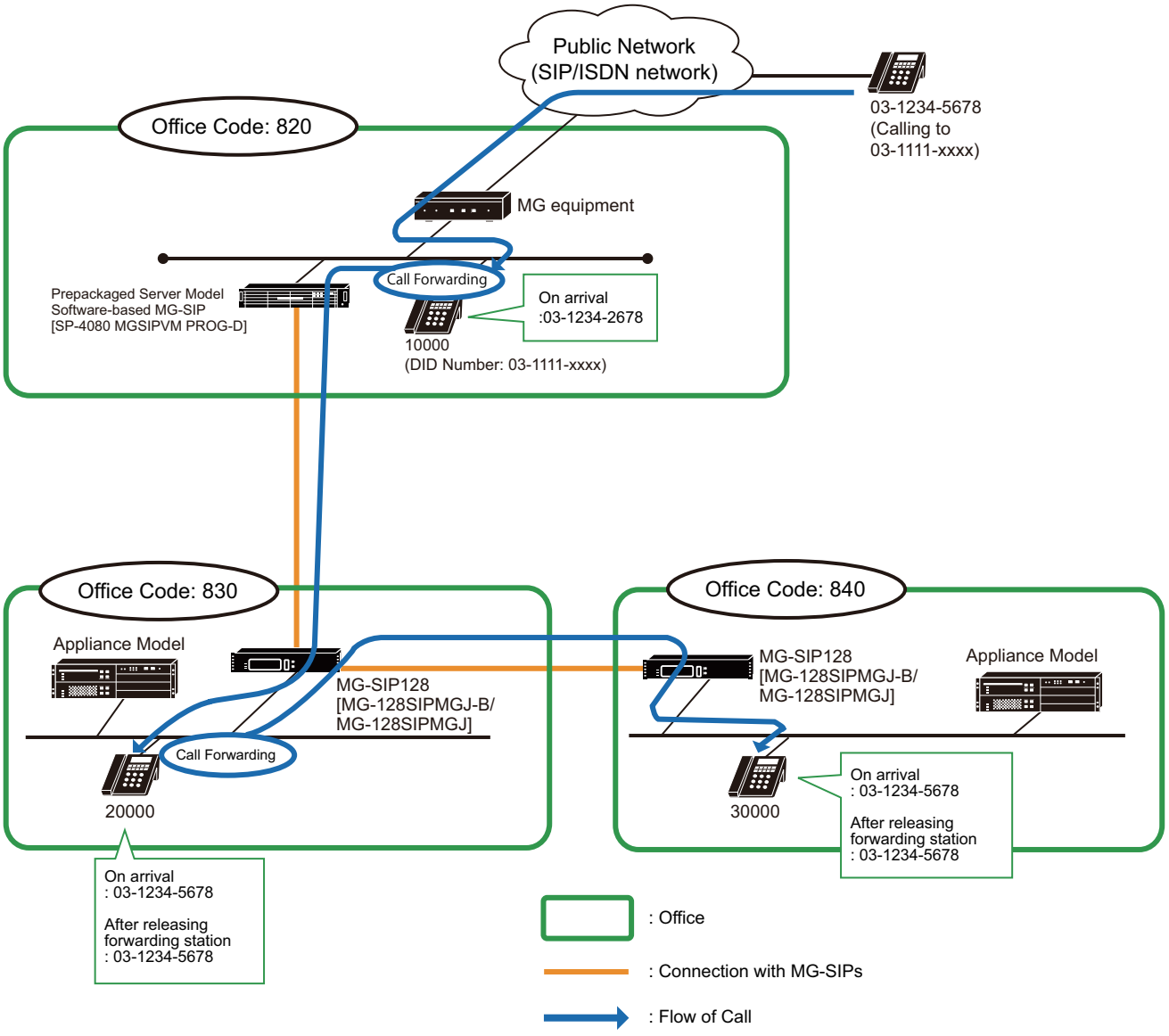
    - FAX communication can be established only when G.711 is assigned to payload for Voice and FAX.
    - FAX communication can be established only when Payload Sizes for Voice and FAX are the same.
    - To establish FAX communication with SIP network via MG-SIP, be sure to assign G.711 as Payload Type for all the FAX accommodated in the system.

16. When connection failure on IP network occurs, alternate routing to PSTN line can be activated. However, because user information cannot be notified, this feature is not supported.
17. MG-SIP supports Individual Trunk Access [I-4] (through both Access Code and PCPro).
18. When the Telephony Server side terminal releases during a conversation after a temporary communication failure between the Telephony Server and MG-SIP, MG-SIP may not send disconnection signals to the SIP network.

19. Speech channel is established by Virtual trunk. When the MG-SIP Virtual Speech channels are registered to a HW (4-Group), channels other than Virtual Speech channels can not be registered to the same HW. To use more than 32-channels, register the channels per 32-channels from top channel on a HW. However, Virtual Speech channels are not required to register for a consecutive HWs.
20. This feature does not support the routing protocol.
21. The following figures show the conditions of the display content on LCD when a call is received.

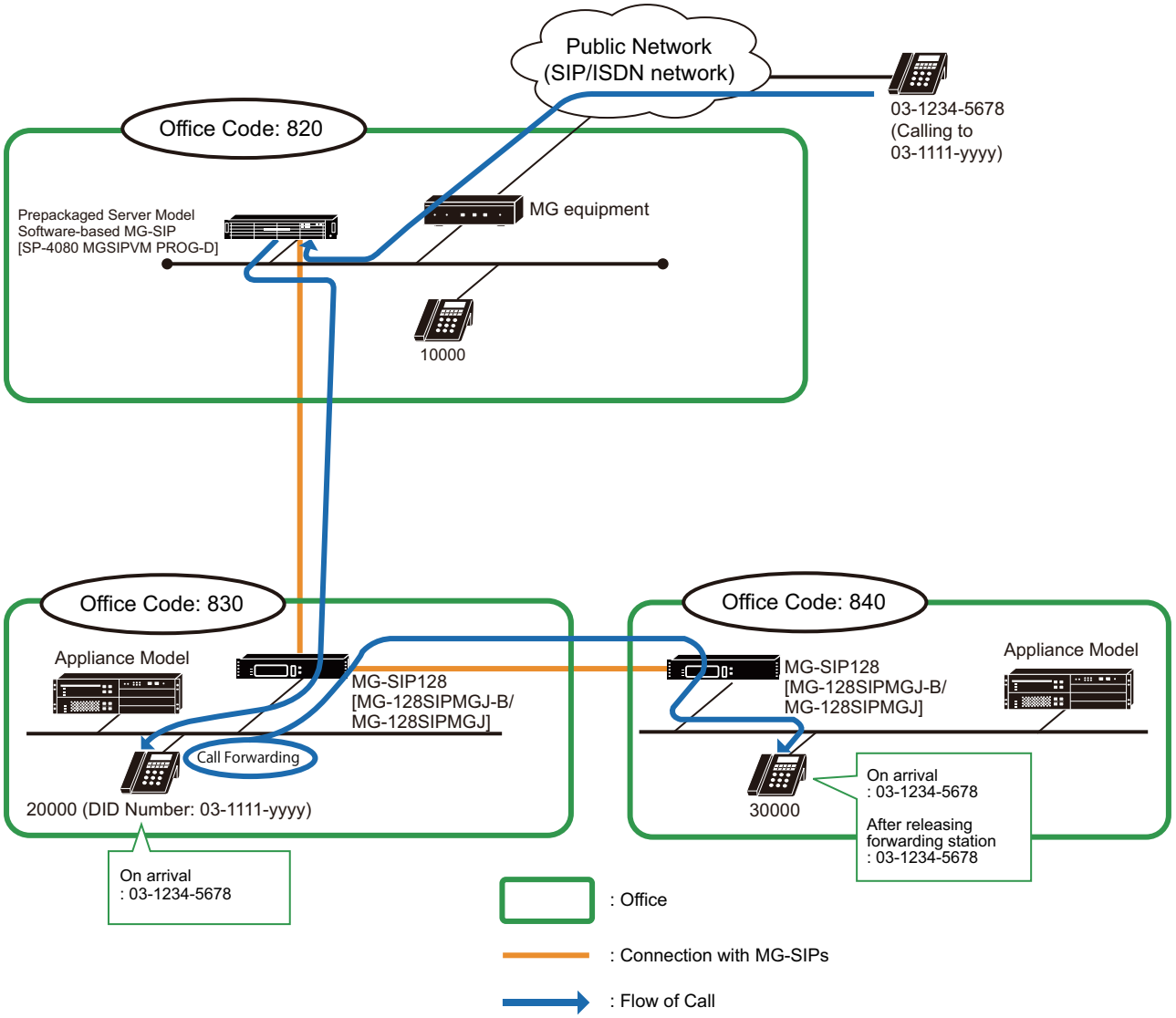
(a) Terminating a C.O. call

When an incoming C.O. call is forwarded to other office, even before a forwarding station releases the call, the Caller ID received from the C.O. line is displayed on the received terminal.



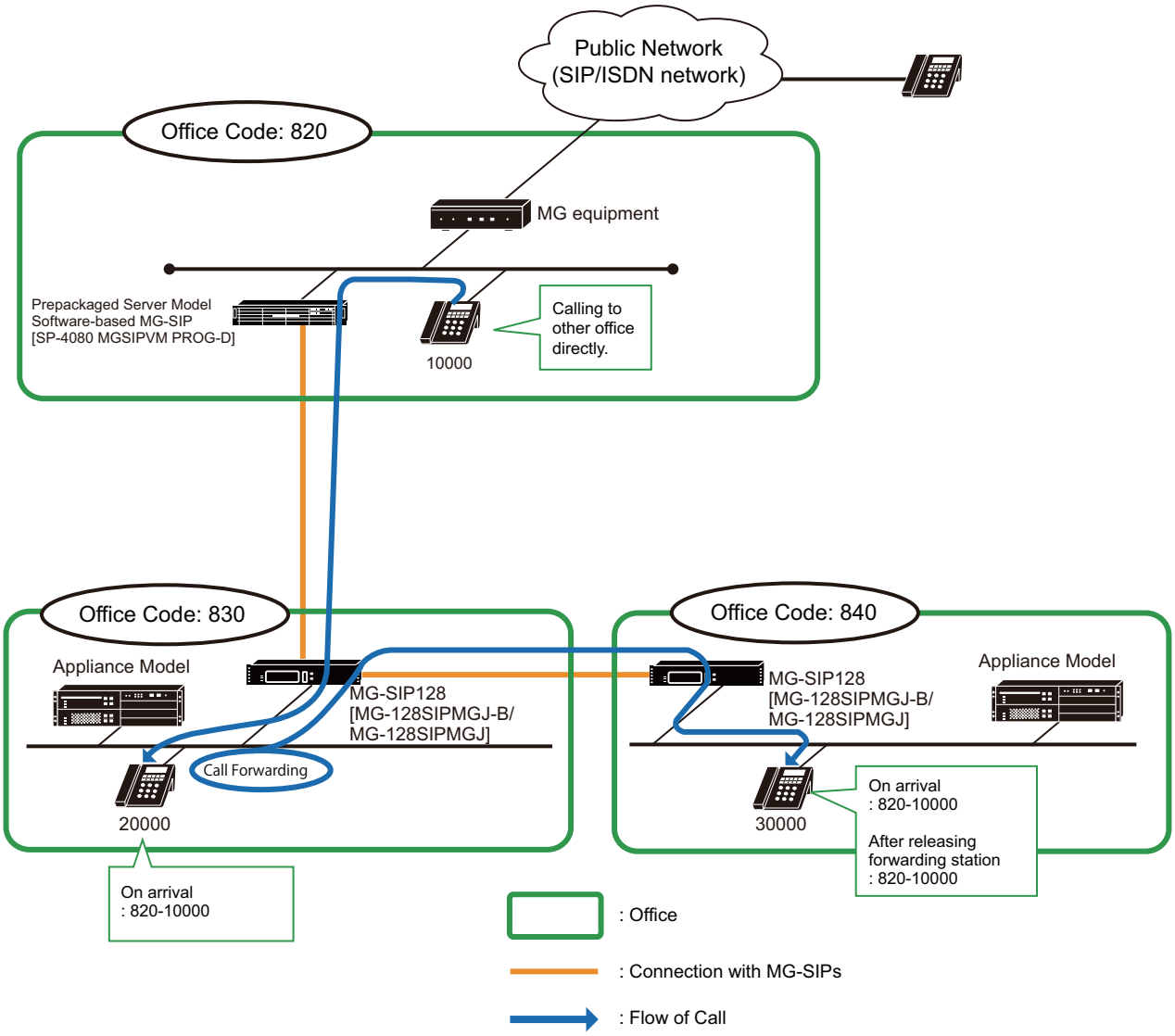
(b) Terminating a C.O. tandem call

When a C.O. tandem call is forwarded to other office, even before released the call by a forwarding station, the Caller ID received from C.O. line is displayed on the received terminal.



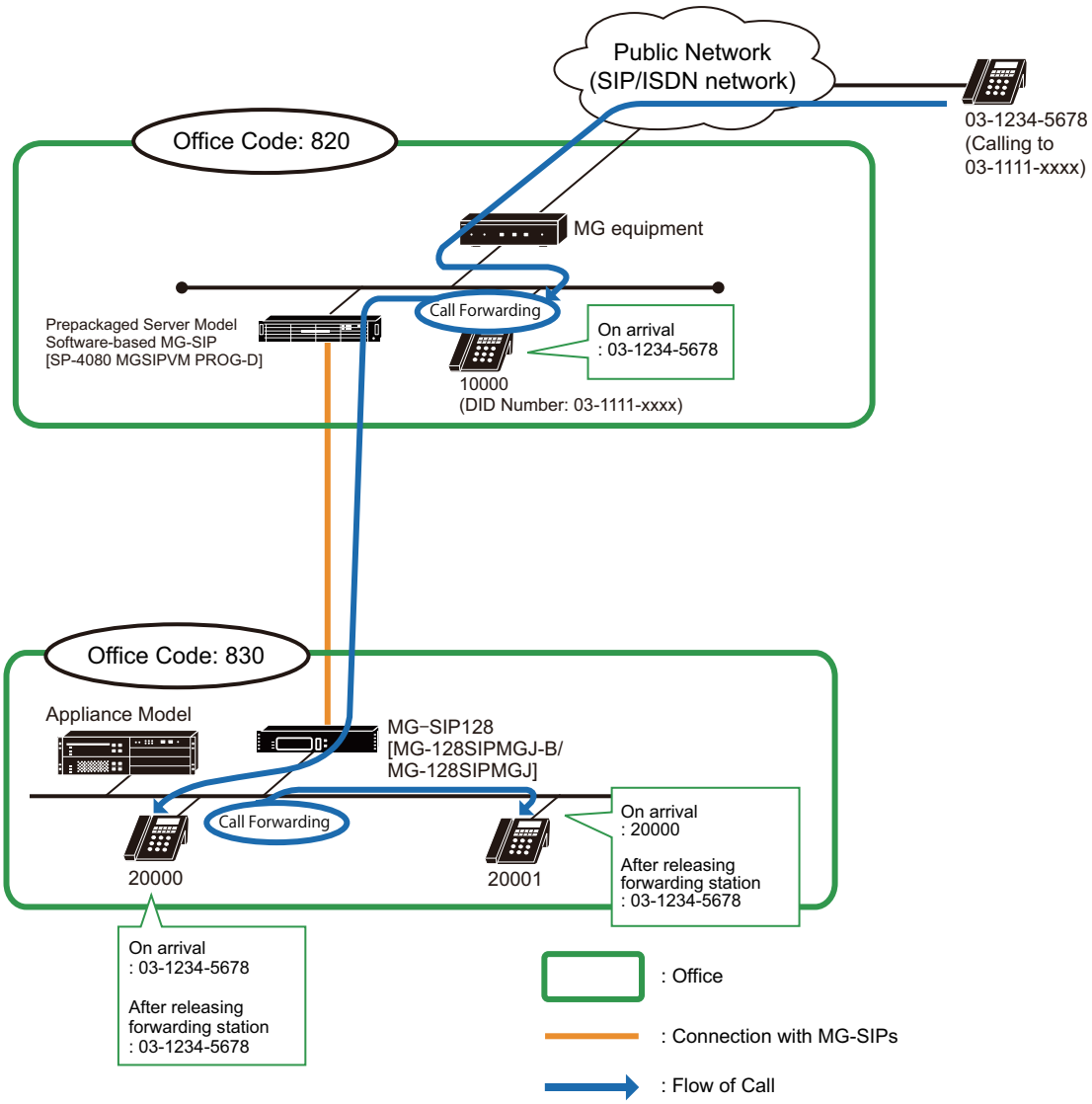
(c) Terminating a station call

When a call is originated from a station to opposite office, “Office Code + Station Number” is set as the Caller ID, and displayed on LCD. If the call is forwarded to another office, operations (a) or (b) on the previous pages apply.



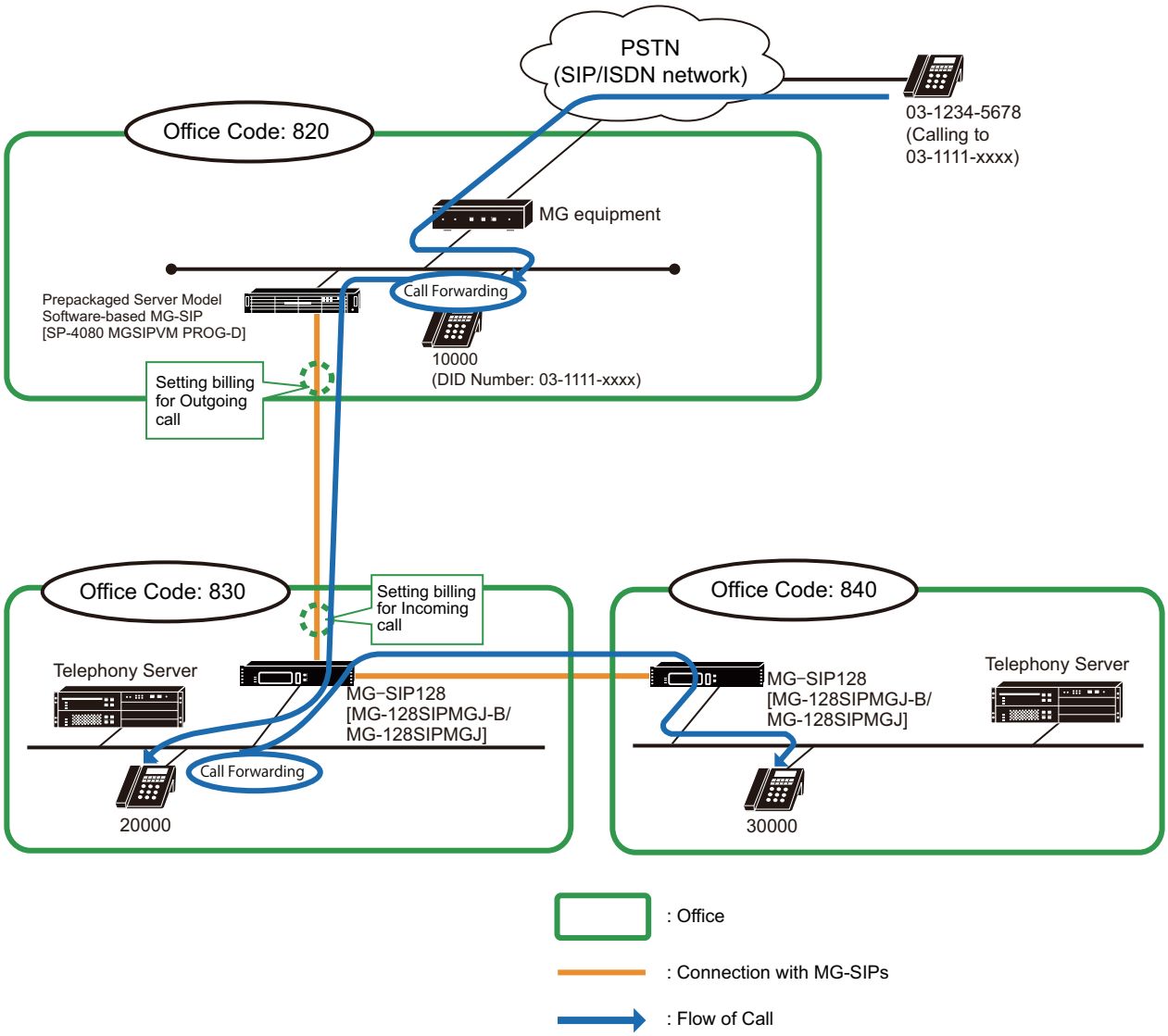
(d) Terminating a C.O. call (Call Forwarding to a terminal within its own office)

When a call is forwarded to a terminal within the same office, the forwarding station number is displayed on LCD at the time of termination.



22. Because of above Service Condition 21. for LCD, when billing is set to MG-SIP for TIE Line, the number in the displayed billing information is as follows.

- Example when holding a C.O. call



(a) Billing for Incoming Call

- Holding a C.O. call

  Caller ID received from C.O. line (“03-1234-5678” in above figure), and not a forwarding station number (“10000” in above figure), is notified. Thus, the Caller ID “03-1234-5678” received from C.O. line is displayed in the billing information for incoming call.

- Holding a station call

  The held station number, other than a forwarding station number, is notified. Thus, the held station number is displayed in the billing information for incoming call.

  Note: Be sure to note the following conditions.
  - The forwarding station number is not displayed in the billing information.
  - Whether you actually talk with the terminal having the notified number or not, the number is displayed to the billing information.

(b) Billing for Outgoing Call

This feature does not have effect on Billing for Outgoing Call.

23. Because of the above Service Condition 21. for LCD, when billing is set for MG-SIP TIE Line, applications that display Call History cannot display the forwarding station number.
24. When an attendant console originates a call to the opposite office by using this feature, "OPE" is not displayed on the LCD of the called station. Instead, the LCD shows either the specific Attendant Number if it has been configured, or "TIE" if not.
25. When a tandem connection for other office is used with MG-SIP TIE Line, the Caller ID received from C.O. line is notified to the destination office.
26. If a C.O. call whose Caller ID is unknown is forwarded, the forwarding station number (Office Code + station number) is displayed on the destination station.
27. In this feature the system processes the timing of Guard Timer for Originating Call. Configure the settings for the timer via the system data (ASYD, SYS1, Index 474). Otherwise the settings are set to "256 ms".

<Tone Code by Countries>

MG-SIP128 plays Inband Tone by using Software Source. Tone Code by Countries is available under the following conditions.

(a) CODECs supported by TONE are: G.711μ-law/A-law only. MG-SIP does not support G.729a.

(b) "No Tone" data is set to TONE data for other CODECs (G.723.1(5.3K) and G.723.1(6.3K)).

(c) CODECs supported by DTMF are: G.711μ-law/A-law only.

(d) Music on Hold (MOH) is not supported.

(e) Short tone is not supported.

(f) This feature is available for the following services.
- Blind Transfer
- Calling to the tandem destination (Only in this case, Inband Tone RBT is sent to the station side.)
- Receiving REFER
- Automated Attendant

(g) Inband Tone of MG-SIP can be changed by using ASYDL, SYS1, Index 811.

- To change Inband Tone of MG-SIP128 is available when the configuration command "set country_code" is set to "0".

<Interactions with Other Features>

None.

Programming

<Programming of MG-SIP>

Note: To use Multipoint Feature, be sure to set MG-SIP side. For details of Multipoint Feature, see "Multipoint Feature" in Peripheral Equipment Description (IP Devices).

Note: Be sure to assign the data in the following order.

STEP 1: **ASYD** - Assign System Data.
SYS1,
Index 64,
Bit 0 = 0/1 (μ-law/A-law for TSW) **Note 8**, **Note 9**
Bit 7 = 1 (holding tone sending)
Index 186, Bit 6 = 1 (ISDN)
Index 187=00
Index 220, Bit 6=0 (ISDN is enabled)
Index 474 (Guard Timer for Originating Call)
Bit 0~3: MTC (Miscellaneous Timer Counter)
Bit 4~6: TC (Timer Class)
1: 64 ms period

Note 8: You can specify either μ-law or A-law as the information which the Telephony Server informs MG-SIP.

Note 9: This data setting is influenced to the entire system.

STEP 2: **ASYDL** - Assign the following:
System Data 1 Index 810 Bit 2 = 0/1 [Internal Holding Tone (IP terminals)/External Holding Tone (EMA and PLO)] **Note 10**
System Data 1 Index 880 Bit 1 = 1 (Internal PHI-PRI/BRI is available)

Note 10: If External music (2) is selected, refer to Music On Hold [M-7].

STEP 3: **ASYDL/ASYDN** - Assignment of System Data into LDM/NDM.
SYS1 Index 1145 Bit 4=1 (C.O. call Information during using MG-SIP TIE Line is stored)

STEP 4: **ALOCL** - Assign ToS settings as follows:

Note: When a change has been made, it is necessary to initialize the Internal PHI-BRI module with the ADPM command.

TYPE: Select "Network Address".
IP Address: Specify the IP address (assign 0.0.0.0 when LOC-ID=0).
MASK BIT: Specify the Mask Bit (assign 32 when LOC-ID=0).

LOC-ID = 0
Setting Type of Signaling Packet Service: Specify the Quality of Service (IP PRECEDENCE/ DIFFSERV).

- When DIFFSERV is selected
Specify its priority [0-7] and the type (LOW DELAY/THROUGH PUT/RELIABILITY/MIN COST).

- When Diffserv is selected
Specify its priority [0-63].

Note: LOC-ID=0 is essential to be assigned, however, assign other network's LOC-ID according to the configuration.

Note: One of the following is required to allow the changes to Setting Type of Signaling Packet Service data for already-operating equipment (such as PH and IP station) to take effect.
- Initialize the Internal PHI-BRI (Initialize the Internal PHI-BRI module with the ADPM command) or system
- Re-register the IP station

STEP 5: **ADTM** - Assign port numbers if necessary [default value = 64000].

Note: When you change the default port number with this command, you need to initialize the Internal PHI-BRI module with the ADPM command.

STEP 6: **ADPM** - Initialize the MGC (Internal PHI-BRI).
PH TYPE = Internal PHI (BRI)
Menu = Initialize

STEP 7: **ALIDL** - Specify the LENS (five digits) of Internal PHI-BRI.
PH LENS: Specify an even-numbered Group Number. The LENS can be allocated on either a Virtual PIR or a Physical PIR.
PH TYPE = Internal PHI
REGMAX: Specify the registration capacity (1-6144).

Note: When the total number of channels using PHI exceeds 256, increase the value of REGMAX.

STEP 8: **ARTD/ARTDN** - Assignment of Route Class Data.

Note: Assign MG-SIP template data with Template Feature.

Assign Route Class Data of MG-SIP for TIE Line.
TCL (CDN:6) =4 (Tie Line)
DC (CDN:66) =15 (Dialed Number Confirmation is set to all digits.)

Note: When changing this data for the existing route, be sure to reboot MG-SIP.

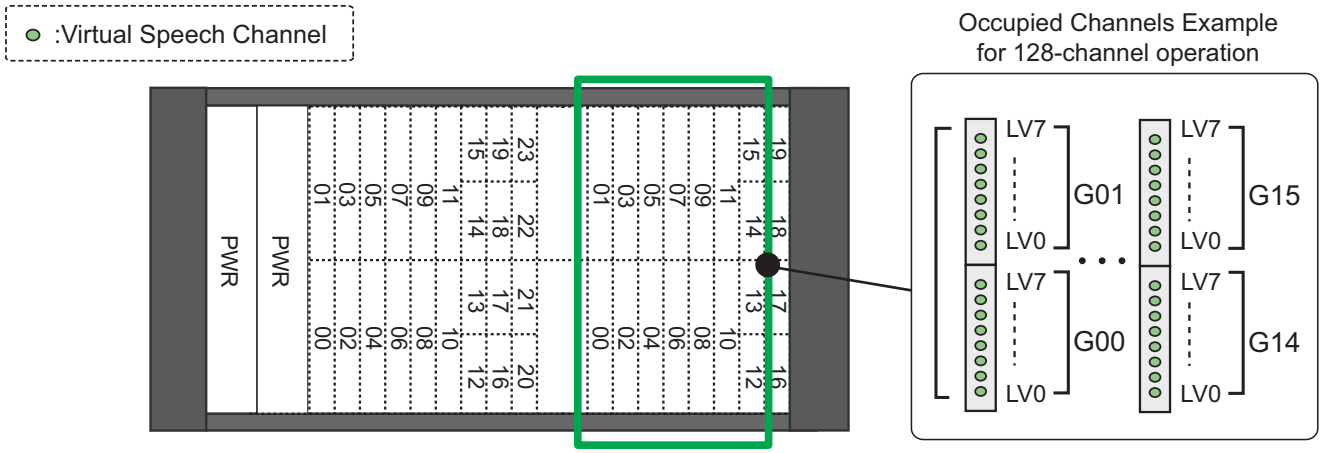
Note: Assign the CDN111(ADVPRA)=1 (ISDN PRI Failure Routing Service) to dummy route.

STEP 9: **ARTI/ARTIN** - Assignment of Trunk Application Data.
Assign Trunk Application Data of MG-SIP for TIE Line.
CDN47: INTD = 2 (SIP)
CDN68: VIR = 2 (MG connection)
CDN77: RA_RT = 0/1 (Virtual Register is used/Register card is used)
CDN96: MGSIPSV =3: Additional service for TIE Line Connection
Assign the following parameters for FAX use:
CDN71: FXD = 1 (IP FAX data is in service)
CDN72: FXJS (Jitter Buffer Size for IP FAX [1-30] × 10 ms) ex. Enter "15" for 150 ms.
CDN73: FXPT = 1 (Payload Type for IP FAX is G.711)
CDN74: FXPS (Payload Size for IP FAX [1-4] × 10 ms) ex. Enter "2" for 20 ms.

Note: When changing this data for the existing route, be sure to reboot MG-SIP.

STEP 10: **ATRK** - Specify trunk data on Virtual Speech Channel and Virtual Signaling Channel.
RT: Specify Route Number [1-255]
TK: Specify Trunk Number [1-255]
LENS: Specify Accommodated Location (Six digits)

Example: In the case of 128-channel operation using MG-SIP128



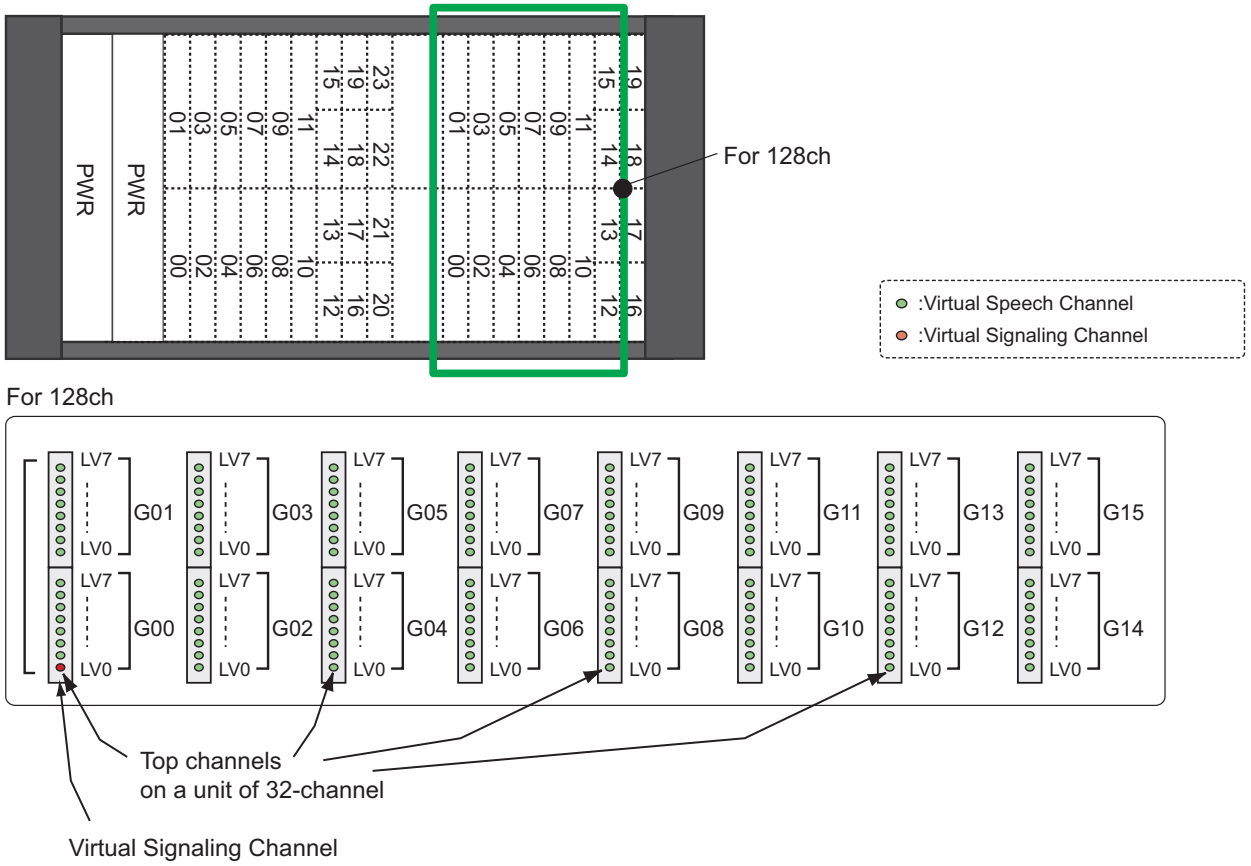
Data Setting Example for 128-channel operation

	RT (1-255)	TK (1-255)	LENS MG 00-07	LENS U 0-3	LENS G 00-23	LENS LV 0-7
Virtual Speech Channel	1 0	0 1	0 2	1	0 0	0
	⋮	⋮	⋮	⋮	⋮	⋮
	1 0	1 2 8	0 2	1	1 5	7

STEP 11: **ADPC** - Register Point Code on Virtual Speech Channel and Virtual Signaling Channel.
RT: Specify a Route Number [1-255]
PC: Specify unused Point Code, not Point Code used in SS7 [1-16383]

STEP 12: **ACSC** - Assign the top number of 32-channel LENS.
CSCG: Specify the CSC Group Number
GROUP: Specify the CIC Group Number
CCH: Specify the top LENS of a unit.

Example: When bundling 128-channel with a unit of 32-channel using MG-SIP128



STEP 13: **ACIC1** - Specify the accommodated location of Virtual Control Channel on Point Code as follows:
PC: Enter Point Code specified by ADPC
CSCG: Enter CSC group specified by ACSC

STEP 14: **ACIC2** - Specify the accommodated location of Virtual Control Channel on Point Code as follows:
PC: Enter Point Code specified by ADPC
CIC: Enter CSC numbers in sequential number
LENS: Enter accommodated locations of Speech Channel and B-Channel

STEP 15: **MBTK** - Cancel Make Busy of Virtual Speech Channel.
RT: Enter Route Number
TK: Enter Trunk Number
MB = 0 (Make idle)

STEP 16: **AMGIL** - Specify information regarding to MG-SIP128 as follows:
MG-ID: Enter MAC Address (6 bytes)
FUNCTION = Setting
KIND = SIP-MG
LINE = 0 (fixed)
CH = 0 (fixed)
LENS: Enter Basic Accommodated Location (the first number of HW)
Retry = 0
Setting Type of Signaling Packet Service: Specify QoS settings of Signaling that is used when Internal

PHI-BRI connects MG-SIP128.
Usable CH Number: Enter the number of channels to be used per MG-SIP128 (1-128).
Service of Authentication Exists: Clear the check box.

STEP 17: **AMGVL** - Register the voice control data when connecting to MG-SIP128. If the settings are not specified by this command, the values assigned by the AIVCL command will be set automatically.
A RT: Specify Physical Route Number (MG-SIP128).
B LOC-ID: Displayed when TYPE = Location ID Specified. Specify Destination Location ID (Destination MG-SIP128). **LOC-ID=0 must be set when IPPAD card is mounted.*
B RT: Displayed when TYPE = Route Specified is selected. Specify Physical Route Number (MG-SIP128) in the case of tandem connection between MGs.
Setting Type of Voice Packet Service: Specify QoS settings of Voice Speech
TYPE: Displayed when the check box of "Setting of Payload" is ON. Specify Payload type.
SIZE: Displayed when the check box of "Setting of Payload" is ON. Specify Payload size.

STEP 18: **ANPD/ANPDL/ANPDN** - Assign the minimum number of digits needed to determine the service required to the first digit received.
TN: Specify a Tenant Number.
1stDC: Specify the first digit of service access code.
CI: Choose a Connection Status Index.
NND: Specify the minimum necessary number of digits for each Access Code.

STEP 19: **ASPA/ASPAL/ASPAN** - Define the Access Codes for each opposite office.
TN: Specify a Tenant Number
ACC: Specify an Access Code. [Maximum 6 digits]
CI: Choose a Connection Status Index.
SRV = LCR (Least Cost Routing)
LGRT: Specify a Dummy Route Number.

STEP 20: **AFRS/AFRSL/AFRSN** - Assign a Number Pattern Code (NPC) and Outgoing Route Selection Number (OPR) to the Dummy Route.
TN: Specify a Tenant Number.
RT: Specify a Dummy Route Number.
NPC: Define a Number Pattern Code. [Maximum 16 digits]
OPR: Define an Outgoing Route Selection Number. [0-4000]

STEP 21: **AOPR/AOPRL/AOPRN** - Assign an LCR/LCRS routing pattern for each Outgoing Route Selection Pattern Routing (OPR) number.
TDPTN: Specify a Time of Day Change Pattern Number. [0-7]
OPR: Specify an Outgoing Route Selection Number. [0-4000]
RA: Specify the Route Advancing Order. [0-7]
E: Specify Route Advancing End Indication. [0/1] **Note 11**
RT: Specify an Outside Route Number. [1-255]
SKIP: Specify the number of digits to be skipped from dialed numbers. **Note 12**
PNL: Specify a Pattern Number Location. [0-999]

Note 11: Be sure to assign 0 at the last RA.

Note 12: If an Access Code is to be removed from the dialed number before the call is sent out, specify the number of digits of the Access Code for SKIP. Otherwise, incorrect number will be passed to the MG-SIP. When the Multi-Point connection is enabled on the MG-SIP, change the setting on the MG-SIP, accordingly.

STEP 22: **AMND**/**AMNDL/AMNDN** - Specify the number of maximum necessary digits for a Destination Code (DC).
TN: Specify a Tenant Number
DC: Specify a Destination Code. [Maximum 10 digits]
MND: Specify maximum necessary digits for the DC. [1-24]

STEP 23: **ARNP/ARNPL/ARNPN** - Assign an Access Code (ACC) to be used for Self Office Identification Code and each individual routing.
RT: Specify a Route Number. [0-255]
ACC: Specify an Access Code. [Maximum 4 digits]

Note: For an Open Numbering network, the following data should be assigned:
RT = 0
ACC: The Office Code of the self office (including ACC for the originating call)

Note: All Offices within an Open Numbering network must have an Office Code with the same number of digits.

STEP 24: **ACNAL** - Assign the Calling Number Pattern (CNP) on an Access Code basis. **When this command is used, assign ARTD command CDN66 to DC=15.*

- Settings for a Call Originated to SIP Network
  Specify Calling Number Pattern against route for Caller ID Notification and Access Code.

  IC/OG = OG (Outgoing Call)
  RT: Enter External Route Number to be sent Caller ID Notification per Access Code.
  ACC: Enter Access Code.
  CNP: Enter Calling Number Pattern.

  Note: To modify CNP set by the ACNAL command, use the ACND command; ACNDN command cannot be used.

- Settings for a Call Arrived from SIP Network
  Specify Calling Number Pattern against route for Caller ID Notification and Access Code.

  IC/OG = IC (Incoming Call)
  RT: Enter External Route Number to be arrived.
  ACC: Enter Access Code.
  DC: Dialed Number Confirmation

STEP 25: **ACND** - Assign the data for number change to CNP has been specified by the ACNAL command
CNP: Calling Number Pattern
SKIP: Number of Skip Digits
ADD: Number of Digits of the Additional Number
DC: Additional Digit Information

STEP 26: **ACDD** - Change Digit Code according to DID Addressing.
DAY/NIGHT: Day/Night Mode
RT: Enter the B-Channel Route Number.
DC: Enter Number of digits for this DC must match the data in DC of the ARTD command.
CDC: Enter Change Digit Code [Maximum 24 digits]
AD: Enter Additional Dialing
XFR: Enable/Disable Transfer Service

STEP 27: After the registration has been completed, reset MG-SIP128 (ON→OFF→ON).

To change Inband Tone of MG-SIP128, the following data setting is required.

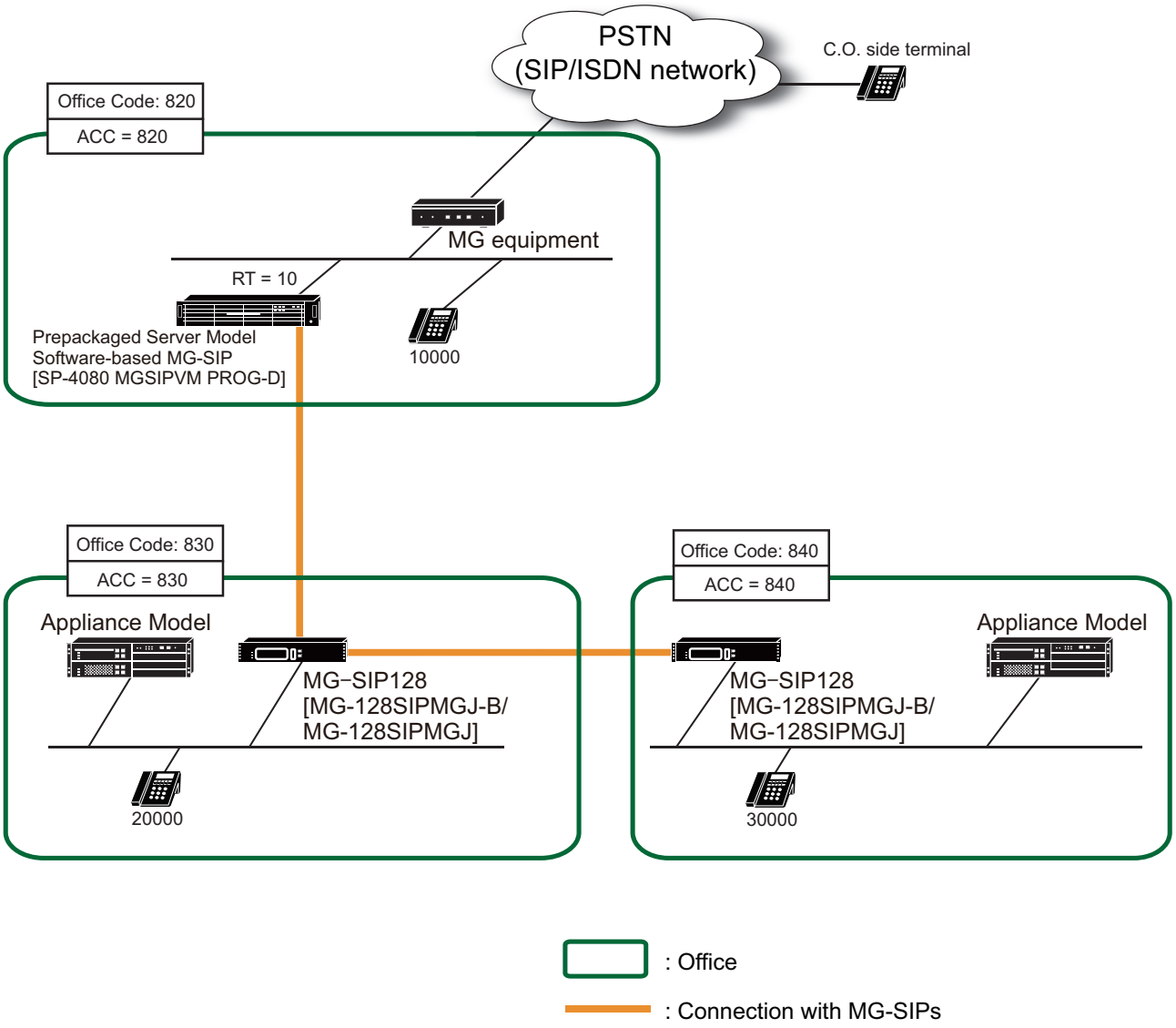
STEP 1: **ASYDL** - Assign System Data into Local DM (LDM).
SYS1, Index 811 Bit 0-7 (Tone Code by countries.)

ASYDL SYS 1 INDEX 811			
TONE CODE	**Basic Tones**	**TONE CODE**	**EMEA Tones**
00	Not used **Note 13**	C8	Germany
01	Japan (default)	C9	Italy
02	North America	CA	Netherlands
03	Australia	CC	Denmark
04	Other countries	CD	Sweden
05	Hong Kong	D3	Spain
06	Malaysia	D4	Austria
07	Singapore	D5	Belgium
08	England	D6	England [For EMEA]
09	Mexico	D7	Greece
0A	Taiwan	D8	Switzerland
0B	New Zealand	D9	South Africa
0C	Korea	Other Values	Not used **Note 13**
0D	Brazil		
0E	China		
0F	Thailand		
10	Thailand (EGAT)		
11	Sri Lanka		

Note: To apply this change, assign the configuration command “set country_code” to “0”.

Note 13: When this Tone Code is assigned to “0” or “Not used”, and the configuration command “set country_code” is assigned to “0”, the tone is set as Japan (default).

[Example of Setup]
As an example, the following procedure describes how to configure this feature at the office with Office Code 820 shown in the figure below:



Note: This procedure assumes that basic configuration of Office Data is already completed.

STEP 1: **ASYDL/ASYDN** - Assign System Data.
SYS1 Index 1145 Bit 4=1 (C.O. call Information during using MG-SIP TIE Line is stored)

STEP 2: **ARTD/ARTDN** - Assignment of Route Class Data.
Assign Route Class Data of MG-SIP for TIE Line.
RT = 10
TCL (CDN:6) = 4 (Tie Line)
DC (CDN:66) =15 (Dialed Number Confirmation is set to all digits)

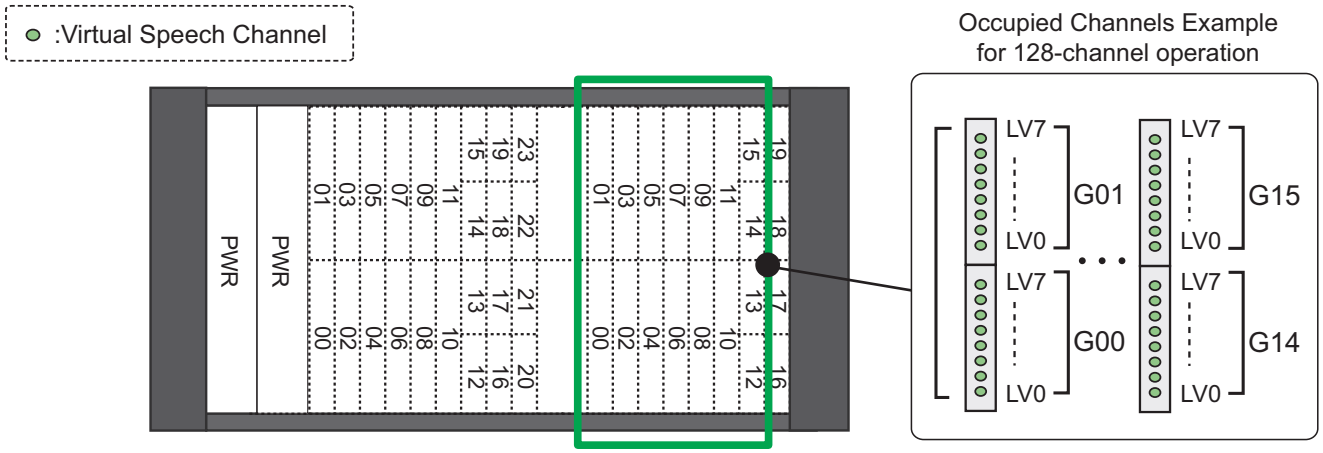
Note: When changing this data for the existing route, be sure to reboot MG-SIP.

STEP 3: **ARTI/ARTIN** - Assignment of Trunk Application Data.
Assign Trunk Application Data of MG-SIP for TIE Line.
RT = 10
CDN 47: INTD = 2 (SIP)
CDN 68: VIR = 2 (MG connection)
CDN 77: RA_RT = 0/1 (Virtual Register is used/Register card is used)
CDN 96: MGSIPSV = 3 (Additional service for TIE Line Connection)

Note: When changing this data for the existing route, be sure to reboot MG-SIP.

STEP 4: **ATRK** - Specify trunk data on Virtual Speech Channel and Virtual Signaling Channel.
RT = 10
TK: 1-128
LENS: 021000-021157

Example: In the case of 128-channel operation using MG-SIP128



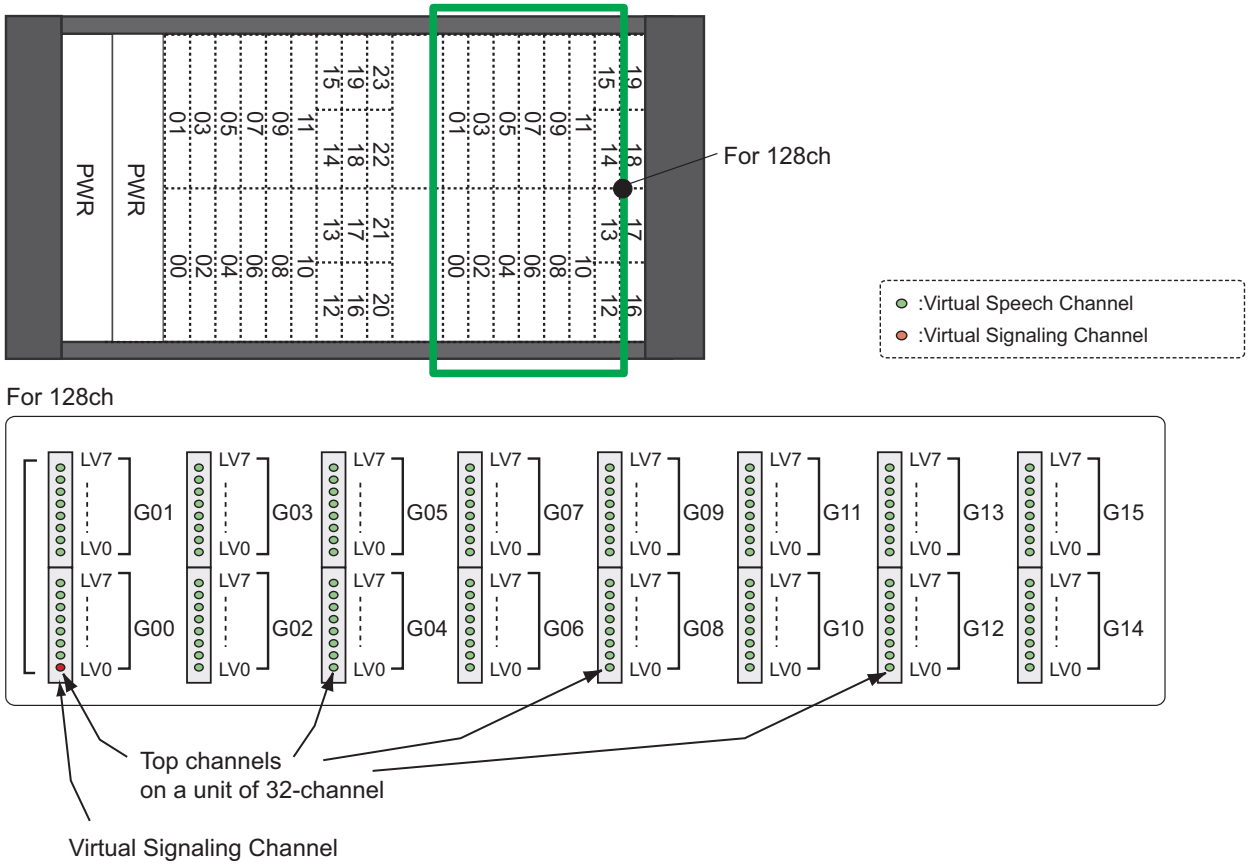
Data Setting Example for 128-channel operation

	RT (1-255)	TK (1-255)	LENS MG 00-07	LENS U 0-3	LENS G 00-23	LENS LV 0-7
Virtual Speech Channel	10	01	02	1	00	0
	⋮	⋮	⋮	⋮	⋮	⋮
	10	128	02	1	15	7

STEP 5: **ADPC** - Register Point Code on Virtual Speech Channel and Virtual Signaling Channel.
RT = 10
PC = 10

STEP 6: **ACSC** - Assign the top number of 32-channel LENS.
CSCG = 130
GROUP: 0-7
CCH: 02100, 02104, 02108, 02112

Example: When bundling 128-channel with a unit of 32-channel using MG-SIP128



CSCG	CICG	CCH		
		MG 00-07	**U** 0-3	**G** **00-23**
130	0	02	1	00
	1	02	1	04
	2	02	1	08
	3	02	1	12
	4	02	1	00
	5	02	1	00
	6	02	1	00
	7	02	1	00

STEP 7: **ACIC1** - Specify the accommodated location of Virtual Control Channel on Point Code as follows:
PC = 10
CSCG = 130

STEP 8: **ACIC2** - Specify the accommodated location of Virtual Control Channel on Point Code as follows:
PC = 10
CIC: 1-128
LENS: 021000-021157

STEP 9: **MBTK** - Cancel Make Busy of Virtual Speech Channel.
RT: Enter Route Number
TK: Enter Trunk Number
MB = 0 (Make idle)

STEP 10: **AMGIL** - Specify information regarding to MG-SIP128 as follows:
MG-ID: Enter MAC Address (6 bytes)
FUNCTION = Setting
KIND = SIP-MG
LINE = 0 (fixed)
CH = 0 (fixed)
LENS = 021000
Retry = 0
Setting Type of Signaling Packet Service: Specify QoS settings of Signaling that is used when Internal PHI-BRI connects MG-SIP128.
Usable CH Number: Enter the number of channels to be used per MG-SIP128 (1-128).
Service of Authentication Exists: Clear the check box.

STEP 11: **ANPD/ANPDL/ANPDN** - Assign the minimum number of digits needed to determine the service required to the first digit received.
TN = 1
1stDC = 8
CI = N or H
NND = 3

STEP 12: **ASPA/ASPAL/ASPAN** - Define the Access Codes for each opposite office.
[For Office Code 830]
TN = 1
ACC = 830
CI = N (Normal) or H (Hooking)
SRV = LCR (Least Cost Routing)
LGRT = 255 (Dummy Route)

[For Office Code 840]
TN = 1
ACC = 840
CI = N (Normal) or H (Hooking)
SRV = LCR (Least Cost Routing)
LGRT = 255 (Dummy Route)

STEP 13: **AFRS/AFRSL/AFRSN** - Assign a Number Pattern Code (NPC) and Outgoing Route Selection

[For Office Code 830]
TN = 1
RT = 255
NPC = 830
OPR = 1

[For Office Code 840]
TN = 1
RT = 255
NPC = 840
OPR = 2

STEP 14: **AOPR/AOPRL/AOPRN** - Assign an LCR/LCRS routing pattern for each Outgoing Route Selection Pattern Routing (OPR) number.

[For Office Code 830]
TDPTN = 0
OPR = 1
RA = 0
E = 0
RT = 10
SKIP = 3 **Note 14**
PNL = 0

[For Office Code 840]
TDPTN = 0
OPR = 2
RA = 0
E = 0
RT = 10
SKIP = 3 **Note 14**
PNL = 0

Note 14: If an Access Code is to be removed from the dialed number before the call is sent out, specify the number of digits of the Access Code for SKIP. Otherwise, incorrect number will be passed to the MG-SIP. When the Multi-Point connection is enabled on the MG-SIP, change the setting on the MG-SIP, accordingly.

STEP 15: **AMND/AMNDL/AMNDN** - Specify the number of maximum necessary digits for a Destination Code (DC).
[For Office Code 830]
TN = 1
DC = 830
MND = 8

[For Office Code 840]
TN = 1
DC = 840
MND = 8

STEP 16: **ARNP/ARNPL/ARNPN** - Assign an Access Code (ACC) to be used for Self Office Identification Code and each individual routing.
Since the network is an Open Numbering network, assign values as follows: **Note 15**, **Note 16**
RT = 0
ACC = 820
Assign an ACC to the external trunk route number.
RT = 10
ACC = 820

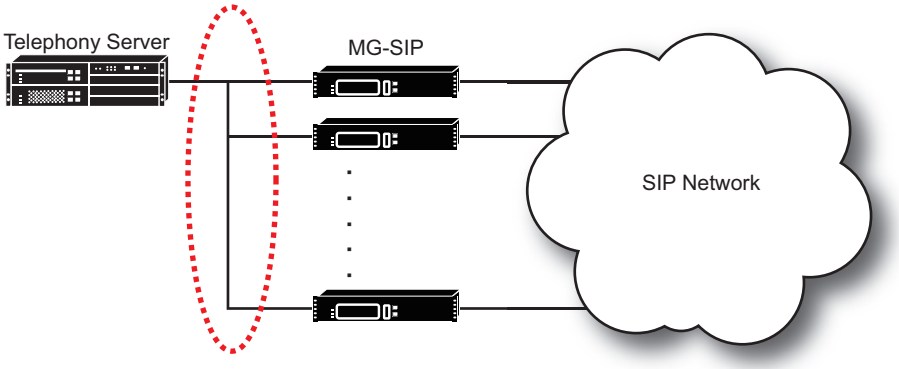
Note 15: For an Open Numbering network, the following data should be assigned:
RT = 0
ACC: The Office Code of the self office (including ACC for the originating call)

Note 16: All Offices within an Open Numbering network must have an Office Code with the same number of digits.

M-123 MG-SIP Shared Route

General Description

With this feature, up to eight MG-SIP devices can share a single route (RT).



Multiple MG-SIP devices are registered to a single route (RT).

Operating Procedure

None.

Service Conditions

<Important Notice>

To make an existing route shared with multiple devices, perform the data settings, and reboot the MG-SIP devices.
To reboot an MG-SIP device, follow any of below steps:

- Delete the configure data by the AMGIL command, and assign again.
- Turn off the power of the MG-SIP device, and then turn on again.
- Execute “reboot” in MG-SIP configuration mode.

<Interactions with Other Features>

None.

<Hardware/Software>

None.

<Interactions with Networks>

None.

<Conditions Specific to This Feature>

1. Accommodate all MG-SIP devices sharing a same route to a single LP.
2. The maximum number of simultaneous speech channels that can be controlled by a single LP is 460 including other ISDN channels.
3. The number of MG-SIPs that can share the same route is as follows.

DEVICES	MAXIMUM NUMBER OF MG-SIPS SHARING THE SAME ROUTE	NUMBER OF USABLE CHANNELS
Software-based MG-SIP	2 devices	255 channels **Note 1**
MG-SIP128 [MG-128SIPMGL-A]		
MG-SIP128 [MG-128SIPMGG-B/ MG-128SIPMGG]		
MG-SIP128 [MG-128SIPMGJ-B/ MG-128SIPMGJ]		
MG-SIP96	2 devices	192 channels
MG-SIP16	8 devices	128 channels
MG-SIP16 [SCA-16SIPMGA/MG-16SIPMGA]		
MG-SIP16 [SCA-16SIPMG (US)]		

Note 1: Because trunk numbers per route can be assigned 1-255, the maximum number of channels that can be shared by the same route is 255.

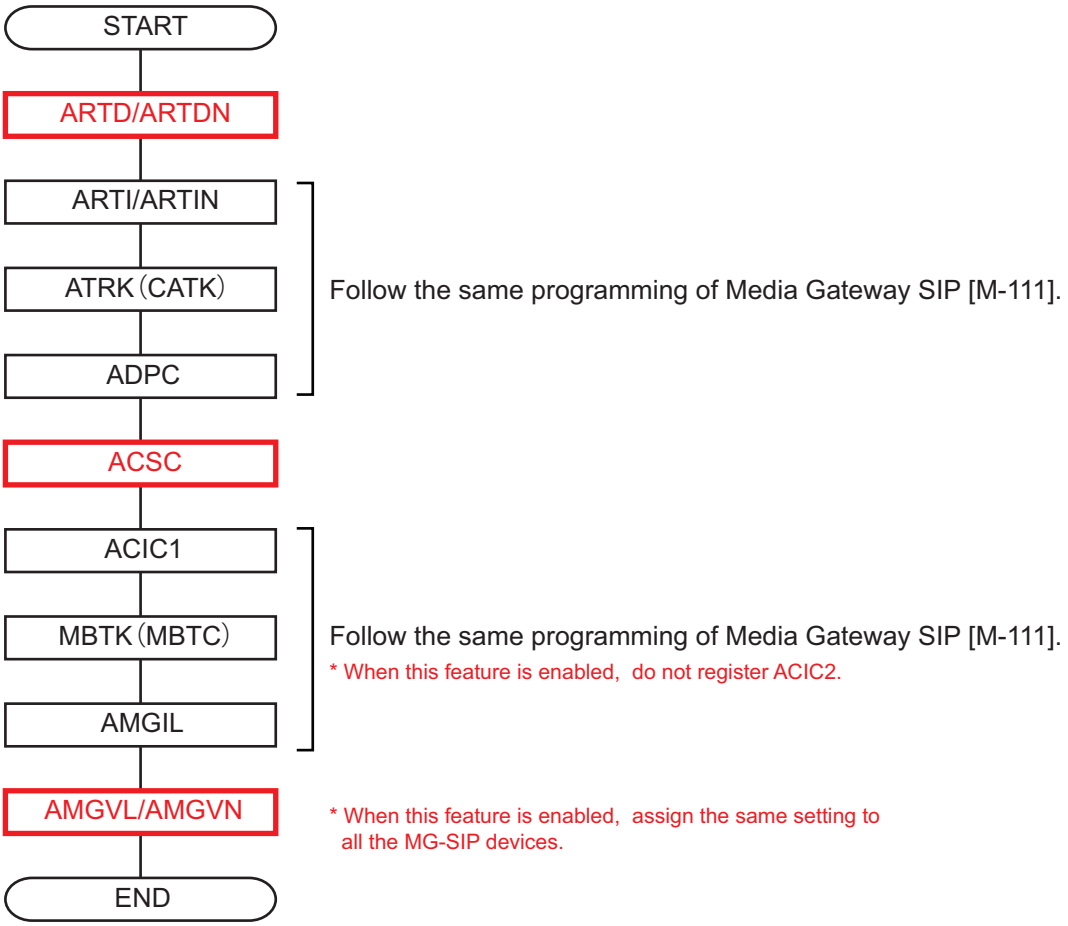
4. Multiple carriers cannot coexist in the same route.
5. Assign the same CODEC to MG-SIPs sharing the same route.
6. Billing Information is output per route. When one route is shared by multiple MG-SIPs, Billing Information cannot be output per MG-SIP device.

<Other Conditions>

None.

Programming

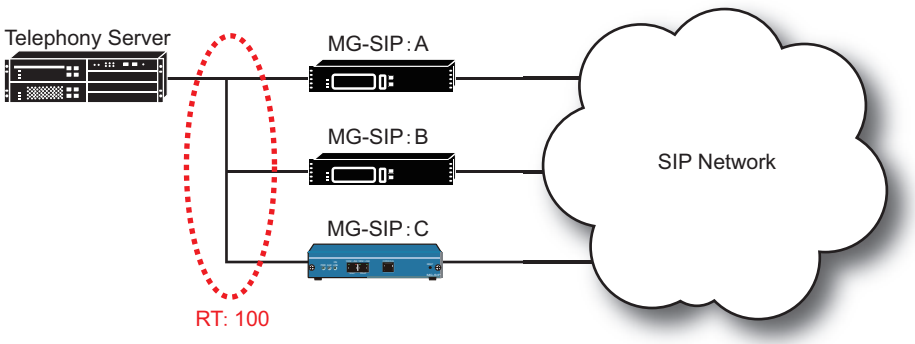
Be sure to assign the data in the following order. The following figure in red shows the required programming for this feature after assigning the programming of Media Gateway SIP [M-111].



Registering a new MG-SIP device with MG-SIP Shared Route

The programming that the MG-SIP is registered with MG-SIP Shared Route is described as follows.

<Configuration Example>



DEVICES		NUMBER OF USABLE CHANNELS	LENS		
			LEAD LENS1	LEAD LENS2	LEAD LENS3
A	MG-SIP96	96ch	020040 (Dch)	020080	020120
B	MG-SIP96	96ch	020160 (Dch)	021040	021080
C	MG-SIP16	8ch	022040 (Dch)	-	-

Note: Following are the system commands that differ from the programming when MG-SIP Route Sharing is not used. For details of the common programming, refer to Media Gateway SIP [M-111].

STEP 1: **ARTD/ARTDN** - Assignment of Route Class Data for DM/NDM.
Assign the following Route Sharing Data for RT: 100.
CDN115 (CMRT)=1 (MG-SIP Shared Route is enabled)

STEP 2: **ACSC** - Assignment of CSC Data.
Assign the lead LENS of MG-SIP by 32-channels (by Highway Block) to even-numbered CSC Group, and the LENS of Virtual Signal Channel to odd-numbered CSC Group.
CSCG: CSC Group Number (even numbers between 130-255 and odd numbers next to them)
GROUP: CIC Group Number
CCH: Lead LENS by 32-channels for even-numbered CSCG
LENS of Virtual Signal Channel for odd-numbered CSCG

<Example of ACSC command assignment>
Assign the lead LENS of MG-SIP by 32-channels to even-numbered CSC Group. Set the LENS of Virtual Signal Channel to the lead CIC Group Number (CICG) of each MG-SIP.

CSC Group Number (CSCG)	CIC Group Number (CICG)	LENS of Signal Channel (CCH)				Corresponding MG-SIP
		MG 00-07	U 0-3	G 00-23	LV 0-7	
130	0	02	0	04	0	A
	1	02	0	08	0	
	2	02	0	12	0	
	3	02	0	16	0	B
	4	02	1	04	0	
	5	02	1	08	0	
	6	02	2	04	0	C
	7	02	0	04	0	**Note 1**

Next, assign the LENS of Virtual Signal Channel to the next CSC Group Number.

CSC Group Number (CSCG)	CIC Group Number (CICG)	LENS of Signal Channel (CCH)				Corresponding MG-SIP
		MG 00-07	U 0-3	G 00-23	LV 0-7	
131	0	02	0	04	0	A
	1	02	0	04	0	
	2	02	0	04	0	
	3	02	0	16	0	B
	4	02	0	16	0	
	5	02	0	16	0	
	6	02	2	04	0	C
	7	02	0	04	0	**Note 1**

Note 1: Assign the same data as CICG=0 to extra CIC Group Number (CICG).

STEP 3: **AMGVL/AMGVN** - Assignment of Media Gateway Voice Control Data for LDM/NDM

Assign Voice Control data when connecting to IP station or MG.

Note: Assign the same data to MG-SIPs for MG-SIP Shared Route.

A RT: Physical Route Number of Virtual Speech Channel of a destination MG.

Type of destination devices:

TYPE: Select “Route Specified” or “Location ID Specified”.

B LOC-ID: Displayed when TYPE = Location ID Specified. Select the Location ID of a destination IP station.

B RT (accommodating MG): Displayed when TYPE = Route Specified is selected. Select the Physical Route Number of Virtual Speech Channel of a destination MG

Note: When IPPAD Card is accommodated, assign LOC-ID to 0.

Setting Type of Voice Packet Service: Data setting of ToS (Type of Service) field for Voice Packet used by MG

Setting of Payload: Click this check box for specifying the following.

TYPE: Select the Payload Type.

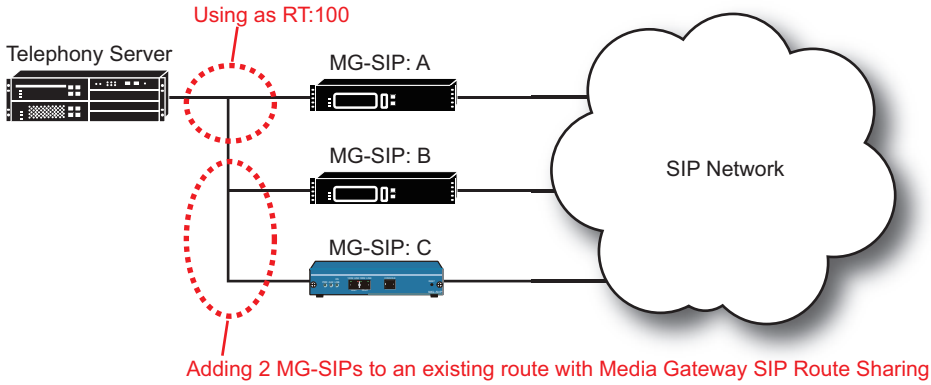
SIZE: Select the Payload Size.

Note: When connecting to the public SIP network, be sure to assign G.711 20 ms.

Change from an existing route to a shared route with MG-SIP Shared Route

To share an existing route:

<Configuration Example>



DEVICES		NUMBER OF USABLE CHANNELS	LENS		
			LEAD LENS1	LEAD LENS2	LEAD LENS3
A	MG-SIP96	96ch	020040 (Dch)	020080	020120
B	MG-SIP96	96ch	020160 (Dch)	021040	021080
C	MG-SIP16	8ch	022040 (Dch)	-	-

Note: Here describes the system commands that differ from the programming when MG-SIP Route Sharing is not used. For details of the common programming, refer to Media Gateway SIP [M-111].

STEP 1: **MBTK/MBTC** - Make-Busy of Trunk/Trunk-Continuous
Assign Make Busy (Bothway) (MB=2) to the trunk used by MG-SIP: A.

STEP 2: **ARTD/ARTDN** - Assignment of Route Class Data
Assign the following Route Sharing Data for RT: 100.
CDN115 (CMRT)=1 (MG-SIP Shared Route is enabled)

STEP 3: **ACSC** - Assignment of CSC Data.
Assign the lead LENS of MG-SIP by 32-channels to even-numbered CSC Group, and the LENS of Virtual Signal Channel to odd-numbered CSC Group.
CSCG: CSC Group Number (even numbers between 130-255 and odd numbers next to them)
GROUP: CIC Group Number
CCH: Lead LENS by 32-channels for even-numbered CSCG
LENS of Virtual Signal Channel for odd-numbered CSCG

<Example of ACSC command assignment>
Assign the lead LENS of MG-SIP by 32-channels to even-numbered CSC Group.

- Before this data assignment

CSC Group Number (CSCG)	CIC Group Number (CICG)	LENS of Signal Channel (CCH)				Corresponding MG-SIP
		MG 00-07	U 0-3	G 00-23	LV 0-7	
130	0	02	0	04	0	A
	1	02	0	08	0	
	2	02	0	12	0	
	3	02	0	04	0	Note 2
	4	02	0	04	0	
	5	02	0	04	0	
	6	02	0	04	0	
	7	02	0	04	0	

- After this data assignment

CSC Group Number (CSCG)	CIC Group Number (CICG)	LENS of Signal Channel (CCH)				Corresponding MG-SIP
		MG 00-07	U 0-3	G 00-23	LV 0-7	
130	0	02	0	04	0	A
	1	02	0	08	0	
	2	02	0	12	0	
	3	02	0	16	0	B
	4	02	1	04	0	
	5	02	1	08	0	
	6	02	2	04	0	C
	7	02	2	04	0	Note 2

Next, assign the LENS of Virtual Signal Channel to the next CSC Group Number.

- Before this data assignment

CSC Group Number (CSCG)	CIC Group Number (CICG)	LENS of Signal Channel (CCH)				Corresponding MG-SIP
		MG 00-07	U 0-3	G 00-23	LV 0-7	
131	No data					

- After this data assignment

CSC Group Number (CSCG)	CIC Group Number (CICG)	LENS of Signal Channel (CCH)				Corresponding MG-SIP
		MG 00-07	U 0-3	G 00-23	LV 0-7	
131	0	02	0	04	0	A
	1	02	0	04	0	
	2	02	0	04	0	
	3	02	0	16	0	B
	4	02	0	16	0	
	5	02	0	16	0	
	6	02	2	04	0	C
	7	02	0	04	0	**Note 2**

Note 2: Assign the same data as CICG=0 to extra CIC Group Number (CICG).

STEP 4: **AMGIL** - Assignment of Media Gateway Information for LDM
Assign the data of MG-SIP: B/C.
Delete and reassign the data of MG-SIP: A.

Note: Be sure to reboot the MG-SIP after this data assignment.

STEP 5: **MBTK/MBTC** - Make-Busy of Trunk/Trunk-Continuous
Cancel the make busy status of the trunk (MG=0).

STEP 6: **ACIC2** - Assignment of CIC Code Data 2
When using Route Sharing, ACIC2 command is not required to assign. Delete the unnecessary data assigned before using Route Sharing.

M-124 My Line Number Display -D

General Description

This feature allows an idle terminal of DT800 Series / DT700 Series / DT400 Series / DT300 Series to display information about the My Line of the terminal (Name and Number) on the LCD. The contents of the display can be changed as listed below by configuring the System Data (ASYDL/ASYDN, SYS1, Indexes 682 and 683).

- Switching the contents of the default display for the line information.

  Four patterns are available: [Name + Number], [Number only], [Name only] and [No display].

- Selecting a line to display the line information.
- Selecting between My Line and Prime Line to display the line information when a terminal is idle.
- Selecting the order of precedence between System Phone Book (APBKN command) and Name Display (ANDD command) to display the registered information when a terminal is receiving Dial Tone (DT) or idle.
- Selecting the display method between "[Number]" (Enclose the number in brackets) and ": Number" (Insert a: colon before the number).
- Selecting the number of characters and the digit number to display.

  When "[Number]" is selected as a number display method, select between [Name 12 characters + Number 10 digits] and [Name 16 characters + Number 6 digits].

  When ": Number" is selected as a number display method, select between [Name 12 characters + Number 11 digits] and [Name 16 characters + Number 7 digits].

- Select whether or not to flash the contents of the display when originating a call.

Below is an example of the display.

[Name + Number]

Name Display [1234567890]

[Number Only] **Note 1**

[1234567890123456]

[Name Only]

Name Display

Note 1: In the case of selecting [Number only], ": Number" (Insert a colon ":" before a number) settings become invalid and only a number will be displayed regardless of the settings. The same condition applies when a terminal is receiving Dial Tone (DT) or idle.

Operating Procedure

Note: The pattern of switching the LCD screen displays is in the order of [Name+ Number]-[Number only]-[Name only]-[No display]. After [No display], it returns to [Name + Number].

- Switch the LCD display with the Soft key or the Feature key.

  The line information shown on the LCD screen display can be switched by pressing the Soft key or the Feature key.

- Switch the LCD screen display with an access code.

1. Lift the handset.
2. Receive the Dial Tone (DT).
3. Dial the access code for switching the screen display of the line information.
4. Receive the Service Set Tone (SST).
5. Place the handset down.

Service Conditions

<Important Notice>

1. This feature is available for DT800 Series/DT700 Series/DT400 Series/DT300 Series.

   Note: This feature is not available for terminals without the LCD screen display.

2. If you use a Digital terminal, CH-16ELCA card is also required.

3. If you change the System Data (ASYDL/ASYDN, SYS1, Indexes 682 and 683) related to this feature, or My Line information being shown on the LCD screen display, the changes will be reflected in the data within 90 seconds.

   Note: The System Data change (except ASYDL/ASYDN, SYS1, Index 682, Bit 0) can be reflected by an off-hook operation as well.

<Interactions with Other Features>

1. The order of precedence in the case of competing with other features is shown below.

2ND LINE (1ST LINE Note 2)	3RD LINE (2ND LINE Note 2)	PRECEDENCE
Help	Help	High
Timer	Timer	
History	History	
OAI	OAI	
Steady	Steady	
Data	Prime Line	
Prime Line	Sub Line	
Sub Line	Extended Steady	
Message	My Line	
My Line	Time	
Blank	-	Low

Note 2: In the case of DT310 6-button type and DT710 6-button type

2. The line information is displayed on the Multi-line telephone side while SOFT PHONE WIRED-PHONE HANDSET [S-147] is being used. When originating a call, the line information is displayed on the soft phone with OFF-HOOK LINE NUMBER DISPLAY-D [O-37D] and on the Multi-line telephone with this feature.

3. The My Line display at time of originating a call will be shown as below if OFF-HOOK LINE NUMBER DISPLAY-D [O-37D] is in use.

[Name + Number]

AAAAA 1234567890

[Number Only]

12345678901

[Name Only]

AAA BBBBBB

4. During Remote Login operation, this feature works according to the data settings on the destination node. For example, if a terminal of Node B logs in to Node A, this feature works by using the data settings on Node A, not on Node B.

5. When Line Pre-selection is in service, the selected line information will be displayed after pressing the Line key. (approx. for 10 seconds)

6. When using OFF-Hook Line Number Display service, regardless of the System Data settings for My line display location “M” or “P” will be displayed at the end of a number.

7. OAI/ACD is not supported.

<Hardware/Software>

If Cordless Terminal (DT430/DT330 (12BT)) is used, only 16 digits are displayed. You can perform a display change operation for this terminal between number only and no display.

<Interactions with Networks>

None.

<Conditions Specific to This Feature>

1. The parameter “NAME” of the ANDD command is displayed as the Name information on the display of the line information.

2. If the Name information has not been set, the display will be shown as follows.

[Name + Number]

Only the number will be displayed.

[1234567890]

[Name Only]

Nothing will be displayed. (Same as the “No display” settings)

3. The key for switching the display for the line information is only valid when the terminal is idle.

4. If the entire Name or the Number of the line information cannot fit in the display, it will display from the start of the Name or the Number.

5. The line information will not be displayed upon originating a call if [No Display] has been selected for the line information. OFF-HOOK LINE NUMBER DISPLAY-D [O-37D] will not operate.

6. Below are examples of a case when the full Name or the Number cannot be displayed.

- If there is any space in the name row of Name Information, the rest of the displayed name after space will be displayed as space. If no space used in the name row, the digits set in the System Data (ASYDL/N, SYS1, Index 682, Bit 6) will be fully displayed from the front.

  Example: Name (with space) [AAAAA BBBBBBB] Number [1234567890]

AAAAA	[1234567890]

  Example: Name (No space) [AAAAABBBBBBCCC] Number[1234567890]

AAAAABBBBBBB	[1234567890]

- If the number is cut off halfway, "*"will be displayed at the end of the displayed number.

  Example: Name [AAA BBBBB] Number [1234567890123456]

AAA BBBBBBB	[123456789*]

7. If the terminal is set to display the line information of Prime Line, when you select a line other than Prime Line and dial the access code, the selected line information is displayed for a given length of time (within 90 seconds).
8. If a terminal is initialized, the display items for the line information will be back to the default settings. The default settings can be configured via the System Data (ASYDL/ASYDN, SYS1, Index 682, Bits 1 and 2).
9. The name of System Phone Book (APBKN command) will not be displayed for the UGN line that is different from the terminal. However, if Name Display (ANDD command) is registered, this command will be valid instead. For example, if Line B of UGN=2 is registered to a Sub Line of Terminal A of UGN=1, the name of Name Display will be shown instead of the name of System Phone Book when Terminal A selects Line B (Line Pre-selection).

<Other Conditions>

None.

Programming

STEP 1: **ASYD** (Assignment of System Data)

SYS1

Index 389

Bit 7=1 (Off-Hook Line Number Display-D is in service)

STEP 2: **ASYDL/ ASYDN** (Assignment of System Data into Local LDM/NDM)

SYS1

Index 682

Bit 0=1 (Line Information Display valid when a terminal is idle)

Bits 1 and 2: The contents of default display of the line information

The default display will be shown as follows according to the each bit settings.

Display Item	Bit 1	Bit 2
Name+ Number	0	0
Number (16 digits)	1	0
Name (24 characters)	0	1
No Display	1	1

Bit 3: The line used for the display of the line information

0/1:My Line/ Prime Line

Bit 4: Name information prioritized to be used for the display of the line information.

0/1:System Phone Book (APBKN command)/ Name Display (ANDD command)

Bit 5: The display format for an idle terminal.

0/1:[Number] (The number in brackets) /: Number (":" before the number)

Bit 6: When selecting Name+ Number for the default display of the line information.

The displayed digit number will be shown as follows according to the settings of Bit 5.

<Bit 5=0, The number in brackets>

0/1: Name 12 characters + Number 10 digits/ Name 16 characters + Number 6 digits

<Bit 5=1, ":" before the number>

0/1: Name 12 characters + Number 11 digits/ Name 16 characters+ Number 7 digits.

Bit 7: Setting for flashing the line information when originating a call.

0/1: No flash/ Flash

Index 683

Bits 6 and 7: The row to display the line information

The displayed row will be as follows according to the terminal to be set.

<DT830/DT750/DT730/DT710 (8-button type)>

Bit 6=0, Bit 7=0: 2nd line

Bit 6=1, Bit 7=0: 3rd line



<DT430/DT330/DT310 (6-button type)/DT710 (6-button type)>

Bit 6=0, Bit 7=0: 1st line

Bit 6=1, Bit 7=0: 2nd line



STEP 3: **AKYD** (Assignment of Key Data for Dterm)
KYN: Key Number
KYI=1 (Function Key)
FKY=193 (Line Information Change Display)

STEP 4: **ADSL/ADSLN** (Assignment of Dterm Soft Key on LCD)
SN=0 (Idle)
FKY=193 (Line Information Change Display)

Note: If switching displays by an access code, use the following commands instead of the AKYD command.

STEP 5: **ANPD/ANPDL/ANPDN** (Assignment of Numbering Plan Data)

Assign the minimum number of digits needed to determine the service required to the first digit received.

STEP 6: **ASPA/ ASPAL/ ASPAN** (Assignment of Special Access Code)
CI=N
SRV=SSCA
SIDA=171 (Line Information Change Display)

M-125 MG External Music-On-Hold

General Description

This feature enables Music-On-Hold to be played to outside callers not only from the Telephony Server but also from an external audio source or an MG internal audio source on an MG device basis.
MG External Music-On-Hold (MOH) is sent when a station places a trunk call on hold during a conversation through an MG.

Operating Procedure

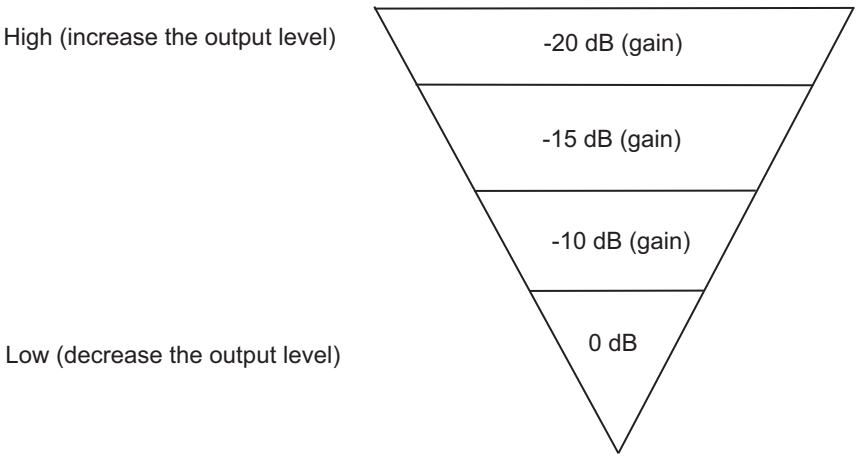
None

Service Conditions

1. This feature is available for the following devices:

   Note: Not available for UG50.

   - MG (BRI)
   - MG (PRI) [SIP]
   - MC&MG-COT
   - MG-6COT
2. This feature is available only when **Music-On-Hold** is enabled (ASYD SYS1 Index 64 Bit 7=1).
3. Use a cable compatible with a 3.5 φ mini-jack (monaural type) to connect audio equipment used for external Music-On-Hold and the MG device.
4. To prevent damage from an external excessive input signal, you should pay careful attention to the output of the external Music-On-Hold equipment. Start with low output level, and then gradually increase it.
5. You can control the sound level of external Music-On-Hold by changing the output level or the input gain while monitoring the audio level on the opposite telephone via ISDN. The output level can be adjusted manually on the device, and the input gain can be adjusted with the AMGIL command from PCPro.
6. Gain adjustment can be selected from four patterns; “0 dB”, “-10 dB (gain)”, “-15 dB (gain)” and “-20 dB (gain)”. See the diagram below.



7. Input impedance from external Music-On-Hold equipment must be within the range of 8Ω to 50kΩ at all times. Do not exceed this range.
8. Induction noise may occur depending on external Music-On-Hold equipment if the equipment and FG on the MG device are not earthed.
9. With this feature, a hold tone can be selected for each MG device with the AMGIL command. This setting overrides the system data settings of External Music-On-Hold (ASYDL SYS1 Index 810 Bit 2:0/1, for IP terminals). If hold tone setting is not configured with the AMGIL command, it will be selected in accordance with the settings of External Music-On-Hold for IP terminals.

Selecting Audio Source of Music-On-Hold

Settings per MG (AMGIL command)	External Music-On-Hold for IP Terminals (ASYDL SYS1 Index 810 Bit 2)	Audio Source of Selected Music-On-Hold
No Setting	Internal Music-On-Hold (IP terminal) (=0)	MG External Music-On-Hold (MG Internal Music-On-Hold 1 is used if External Music-On-Hold equipment is not connected)
	External Music-On-Hold (EMA/PLO) (=1)	Telephony Server External Music-On-Hold In the case of IPPAD PKG not used, the order of precedence to operate is as follows: (1) MG External Music-On-Hold (2) MG Internal Music-On-Hold 1
External Music-On-Hold	- (Regardless of settings)	MG External Music-On-Hold (MG Internal Music-On-Hold 1 is used if External Music-On-Hold equipment is not connected)
Internal Music-On-Hold 1		MG Internal Music-On-Hold 1
Internal Music-On-Hold 2		MG Internal Music-On-Hold 2

10. With Telephony Server External Music-On-Hold configured, if external Music-On-Hold equipment cannot be used, e.g. IPPAD circuit card not available or all busy, MG External Music-On-Hold will be activated instead.
11. If external Music-On-Hold equipment is not connected to AUX on an MG device when MG External Music-On-Hold is selected, Internal Music-On-Hold 1 will be selected instead by the MG device. However, MG External Music-On-Hold can be selected when external Music-On-Hold equipment is connected afterward.
12. Either Telephony Server Music-On-Hold (Extended Hold Music and External Music-On-Hold) or MG External Music-On-Hold can be chosen as a hold tone used on each MG device. Note that Telephony Server External Music-On-Hold, Extended Hold Music, and Multiple Music-On-Hold Source cannot be used in combination.
13. When MG External Music-On-Hold is selected, the gain value of External Music-On-Hold input can be configured per MG device with the AMGIL command. Note that the input value range is different from the gain value for the input value. Follow the conditions of MG devices.
14. For service conditions for UG50, see “Conditions and Limitations” in Peripheral Equipment Description (UG50).

Programming

STEP 1: **ASYD** - Assignment of System Data

SYS1

Index 64

Bit 7=1 (Music-on-Hold in service)

STEP 2: **ASYDL** - Assignment of System Data for LDM

SYS1
Index 810
Bit 2: External Music-on-Hold service for IP terminal. (In the case of MG External Music-on-hold used)
0/1=(Internal Music-on-hold for IP terminal/External Music-on-hold for EMA/PLO)

Note: Conditions for this command.
- This command is only available when Music-on-hold is not configured with the AMGIL command.
- For more information on selecting External Music-on-hold, see “M-7 Music On Hold” in Data Programming Manual - Business.

STEP 3: **AMGIL** - Assignment of Media Gateway Information for LDM

Register information of each MG (Media Gateway).

(a) In the case of MG (BRI) [Proprietary Protocol]

MG-ID: Assign the MAC address of MG (6 bytes)

FUNCTION: Select the function type of MG settings.

KIND: Select ISDN Trunk (BRI)

LINE=0/1 (ISDN Line Setting)

CH=0/1 (Channel B1/ Channel B2) **Note 2**

LENS: Allocate the LENS of Virtual Speech Channel. **Note 3**

Setting Type of Signaling Packet Service:
MG Signaling Channel QoS setting (ToS value used by MG (BRI) for signal control)

Music On Hold Exists:

Music On Hold= External Music On Hold/Internal Music On Hold1/Internal Music On Hold2

External Music On Hold Input Gain Adjustment [0-15]

0: 0 dB (gain)
1: -10 dB (gain)
2: -15 dB (gain)
3: -20 dB (gain)
4-15: Not used

Emergency Telephone= Automatically Cancel (default)/ Manually Cancel

<When FUNCTION =RESET>

ACTION AFTER RESET

= Startup with operation mode (default)/ Startup with emergency telephone mode

Note 2: Ensure that settings for both CH0 and 1 are configured. Do not register just CH0 (even number level) or CH1 (add 1 to the LENS of CH0) only.

Note 3: Before registration, check to ensure that the ATRK command data has been entered.

(b) In the case of MG (BRI) [SIP]

MG-ID: Assign the MAC address of MG (6 bytes)

FUNCTION: Select the function type of MG settings.

KIND: Select ISDN Trunk (BRI) [SIP]

LINE=0/1 (ISDN Line Setting)

CH=0/1 (Channel B1/ Channel B2) **Note 3**

LENS: Allocate the LENS of Virtual Speech Channel. **Note 4**

Setting Type of Signaling Packet Service:
MG Signaling Channel QoS setting (ToS value used by MG (BRI) for signal control)

Music On Hold Exists:

Music On Hold= External Music On Hold/Internal Music On Hold1/Internal Music On Hold2

External Music On Hold Input Gain Adjustment [0-15]

0: 0 dB (gain)
1: -10 dB (gain)
2: -15 dB (gain)
3: -20 dB (gain)
4-15: Not used

Emergency Telephone= Automatically Cancel (default)/ Manually Cancel

<When FUNCTION =RESET>

ACTION AFTER RESET

= Startup with operation mode (default)/ Startup with emergency telephone mode

Note 3: Ensure that settings for both CH0 and 1 are configured. Do not register just CH0 (even number level) or CH1 (add 1 to the LENS of CH0) only.

Note 4: Before registration, check to ensure that the ATRK command data has been entered.

(c) In the case of MG (PRI) [SIP]

MG-ID: Assign the MAC address of MG (6 bytes)

KIND: Select ISDN Trunk (PRI 1.5M) [SIP]/ISDN Trunk (PRI 2M) [SIP]

UG50 Multi-Slot Mode: Select this check box if registering UG50 in Multi-slot mode.

FUNCTION: Select the function type of MG settings.

LINE=0 (fixed)

CH=0 (fixed)

LENS: Allocate the head channel of HW (Highway Block) where Virtual Signal Channel is assigned.

Setting Type of Signaling Packet Service:
MG Signaling Channel QoS setting (ToS value used by MG (PRI) for signal control)

Music On Hold Exists:

Music On Hold= External Music On Hold/Internal Music On Hold1/Internal Music On Hold2

External Music On Hold Input Gain Adjustment [0-15]

Country Code: Select an appropriate country code

Law of A/D Conversion: Appropriate method is displayed according to Country Code.

(d) In the case of MC&MG-COT

Note: The device will be reset when a registration/deletion is executed with this command during operation.

MG-ID: Assign the MAC address of MC&MG-COT (6 bytes)

KIND: Select MC-MG (COT)

UG50 Multi-Slot Mode: Select this check box if registering UG50 in Multi-slot mode.

FUNCTION: Select the function type of MG settings.

LINE=0/1 (MG Line Number)

CH=0 (fixed)

LENS: Allocate the LENS of Virtual Speech Channel.

Setting Type of Signaling Packet Service:
MC&MG-COT Signaling Channel QoS setting (ToS value used by MC&MG-COT for signal control.

Music On Hold Exists:

Music On Hold= External Music On Hold/Internal Music On Hold1/ Internal Music On Hold2

External Music On Hold Input Gain Adjustment [0-15]

0: 0 dB (gain)
1: -10 dB (gain)
2: -15 dB (gain)
3: -20 dB (gain)
4-15: Not used

Country Code: Select an appropriate country code

Law of A/D Conversion: Appropriate method is displayed according to Country Code.

PB Receive Complete Timing of Dial-In [0-15]

False Answer Timing [0-15]

(e) In the case of MG-6COT

Note: The device will be reset when a registration/deletion is executed with this command during operation.

MG-ID: Assign the MAC address of MG-6COT (6 bytes)

KIND: Select MG (6COT)/MG (6COT) [SIP]

UG50 Multi-Slot Mode: Select this check box if registering UG50 in Multi-slot mode.

FUNCTION: Select the function type of MG settings.

LENS Setting: Allocate the LENS of Virtual Speech Channel.

LINE=0-5 (MG Line Number)

CH=0 (fixed)

Setting Type of Signaling Packet Service:
MG-6COT Signaling Channel QoS setting (ToS value used by MG-6COT for signal control.)

Music On Hold Exists:

Music On Hold= External Music On Hold/Internal Music On Hold1/Internal Music On Hold2

External Music On Hold Input Gain Adjustment [0-15]

0: 0 dB (gain)
1: -10 dB (gain)
2: -15 dB (gain)
3: -20 dB (gain)
4-15: Not used

Country Code: Select an appropriate country code

Law of A/D Conversion: Appropriate method is displayed according to Country Code.

PB Receive Complete Timing of Dial-In [0-15]

False Answer Timing [0-15]

M-126 MG-SIP TIE Line Point-to-Multipoint Connection

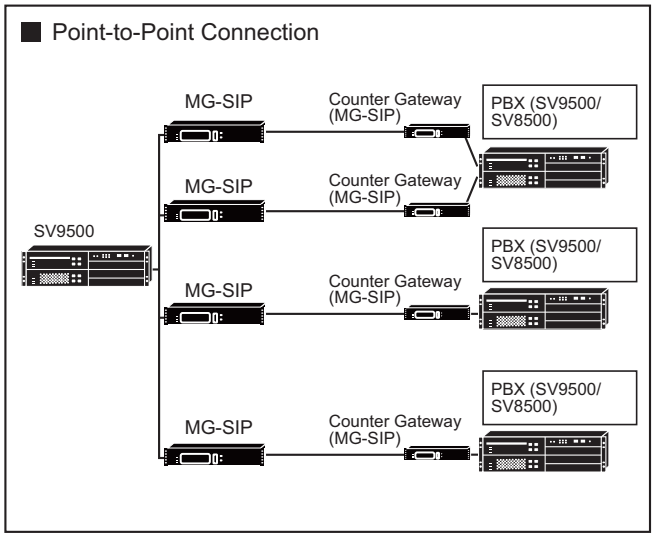
General Description

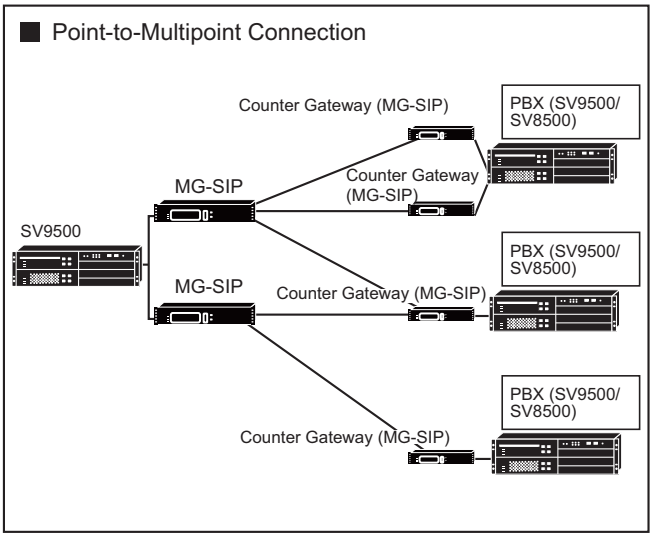
This feature enables Point-to-Multipoint Connection on MG-SIP TIE Line.

Note: This feature is available since FP95-113 V3.

Note: The definitions of the terms used in this feature shall be as follows:

Point-to-Point Connection:	One MG-SIP can be connected to only a single counter gateway. This connection supports, as a counter gateway, only MG-SIPs controlled by SV9500/SV8500 or the SIP gateway devices controlled by SV9300.
Point-to-Multipoint Connection:	One MG-SIP can be connected to multiple counter gateways. Also, multiple MG-SIPs can be connected to multiple counter gateways. This connection supports, as a counter gateway, only MG-SIPs controlled by SV9500/SV8500 or the SIP gateway devices controlled by SV9300.





To implement Point-to-Multipoint Connection, the following features must be configured:

Feature Name	Descriptions
MG-SIP TIE Line Path Splitting [M-127]	A single MG-SIP can be connected to multiple counter gateways.

Feature Name	Descriptions
MG-SIP TIE Line Failure Detection Per Path [M-128]	MG-SIP periodically sends keep-alive packets to each path to verify that the connection is valid. If MG-SIP detects a communication failure or recovery, it sends a failure notification to Telephony Server.
MG-SIP TIE Line Loop Detection [M-129]	When a call routed via MG-SIP loops through nodes, the call termination is restricted.

Operating Procedure

None.

Service Conditions

<Important Notices>

1. This feature is available only for MG-SIP TIE Line. For details on MG-SIP TIE Line, see MG-SIP TIE LINE [M-122] and MG-SIP128 [MG-128SIPMGJ-B/MG-128SIPMGJ] in Peripheral Equipment Description (IP Devices).

<Interactions with Other Features>

None.

<Hardware/Software>

1. This feature is available for MG-128SIPMGJ-B with firmware versions SP-4060 MGSIP PROG-L Issue 3 or later.

<Interactions with Networks>

None.

<Conditions Specific to This Feature>

1. The service conditions for MG-SIP TIE LINE [M-122] also apply to this feature.
2. This feature cannot be used with the Multipoint feature on MG-SIP. When using this feature, disable the Multipoint feature with the configuration command on MG-SIP.

<Other Conditions>

None.

Programming

For data assignments of each feature, see the following:

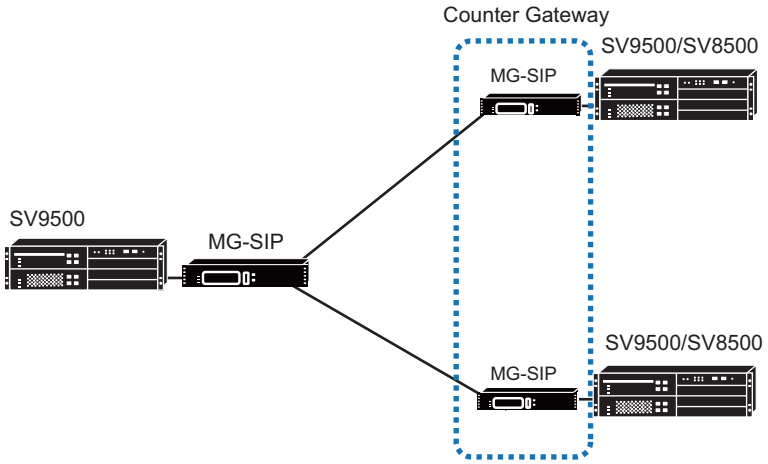
- MG-SIP TIE LINE PATH SPLITTING [M-127]
- MG-SIP TIE LINE FAILURE DETECTION PER PATH [M-128]
- MG-SIP TIE LINE LOOP DETECTION [M-129]

M-127 MG-SIP TIE Line Path Splitting

General Description

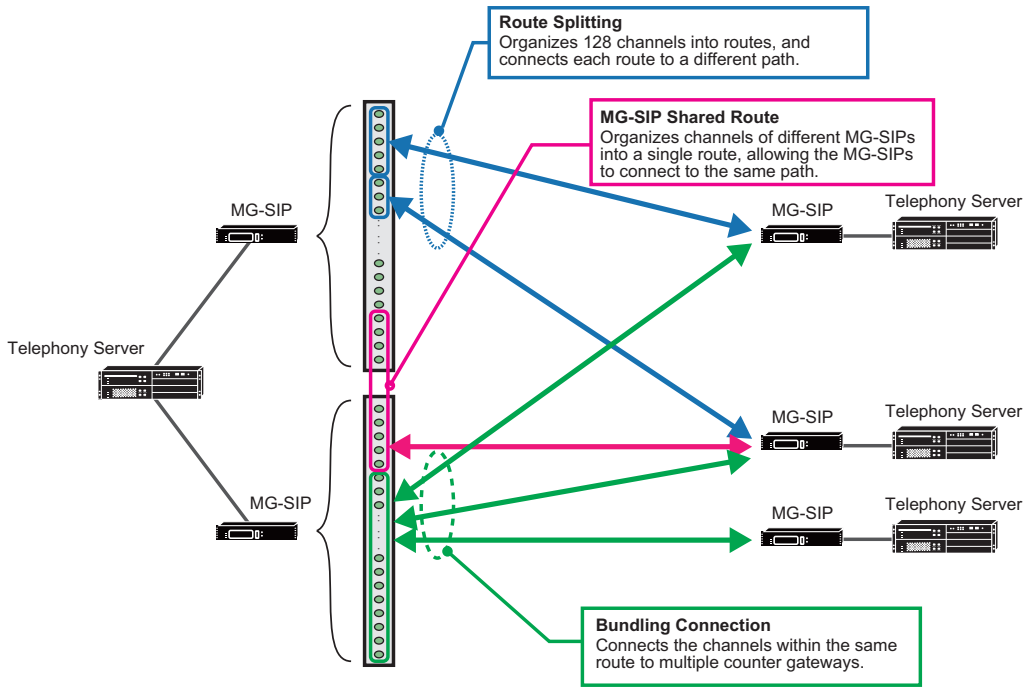
A single MG-SIP can be connected to multiple counter gateways.

Note: This feature is available since FP95-113 V3.



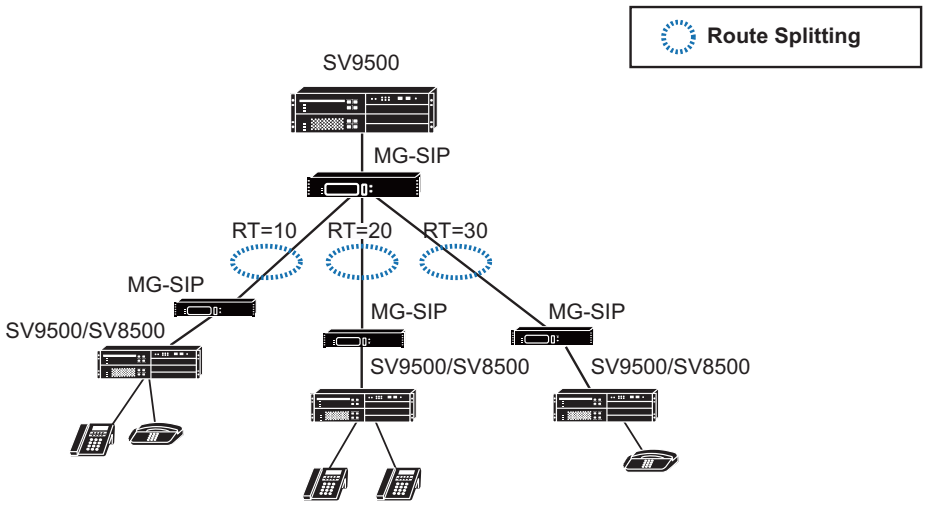
By combining the following functions, speech channels in an MG-SIP or between MG-SIPs can be changed flexibly according to the operating condition.

	Features	Descriptions	Remarks
1	Route Splitting	One MG-SIP can connect to multiple external routes and counter gateways.	
2	Bundling Connection	One MG-SIP can connect to the multiple gateways in the same counter office.	
3	MG-SIP Shared Route	Multiple MG-SIPs can share the same external route.	
4	Combination of Route Splitting and Bundling Connection	One MG-SIP can connect to multiple counter gateways in different external routes.	Combination of 1 and 2
5	Combination of Route Splitting and MG-SIP Shared Route	Some of the external routes used in Route Splitting can be shared by multiple MG-SIPs.	Combination of 1 and 3
6	Combination of Bundling Connection and MG-SIP Shared Route	Some of the external routes used in Bundling Connection can be shared by multiple MG-SIPs.	Combination of 2 and 3
7	Combination of Route Splitting, Bundling Connection and MG-SIP Shared Route	Some of the external routes used in Route Splitting and Bundling Connection can be shared by multiple MG-SIPs.	Combination of 1, 2 and 3



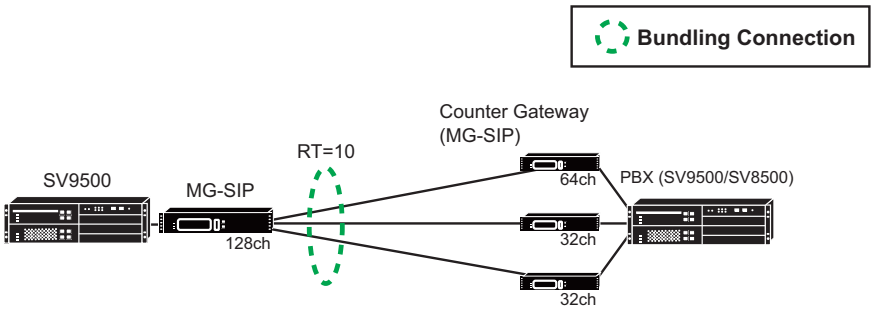
Route Splitting

One MG-SIP can connect to multiple offices.



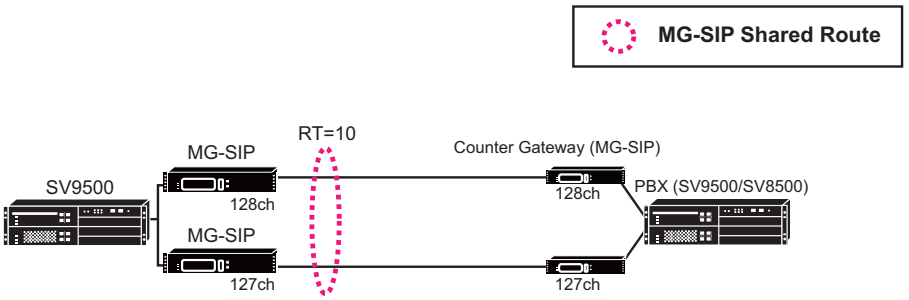
Bundling Connection

One MG-SIP can connect to multiple gateways in the same counter office.



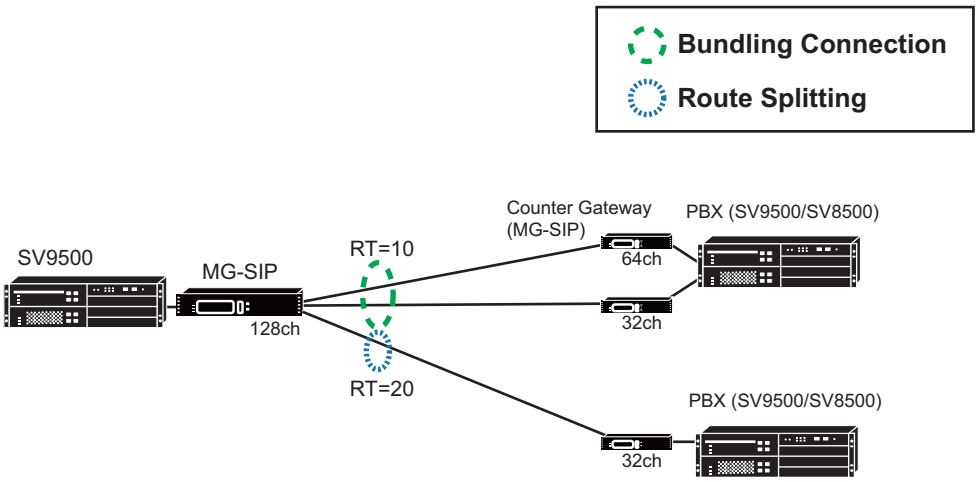
MG-SIP Shared Route

Multiple MG-SIPs can share the same external routes. For more details on MG-SIP Shared Route, see MG-SIP SHARED ROUTE [M-123].



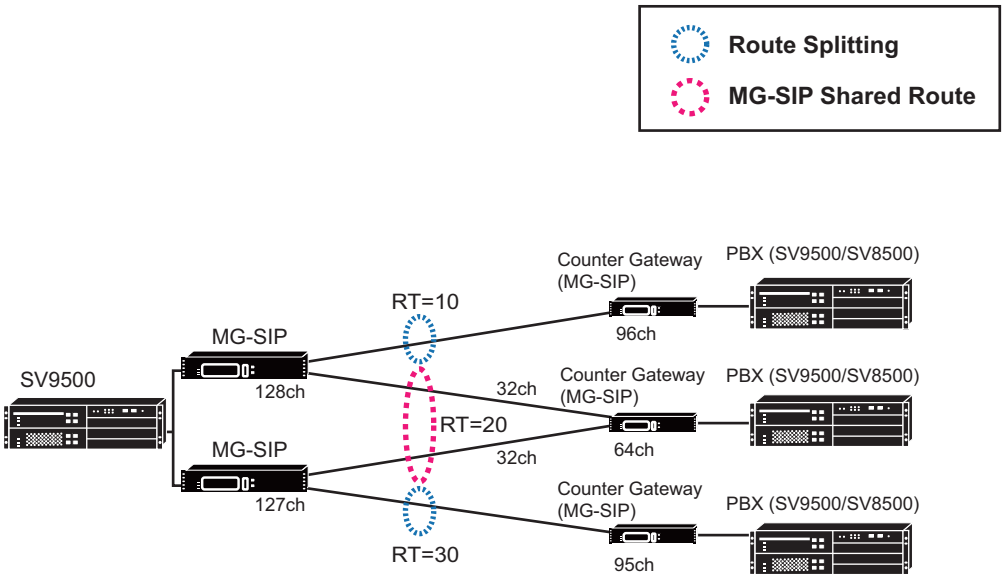
Combination of Route Splitting and Bundling Connections

One MG-SIP can use both Route Splitting and Bundling Connection at the same time.



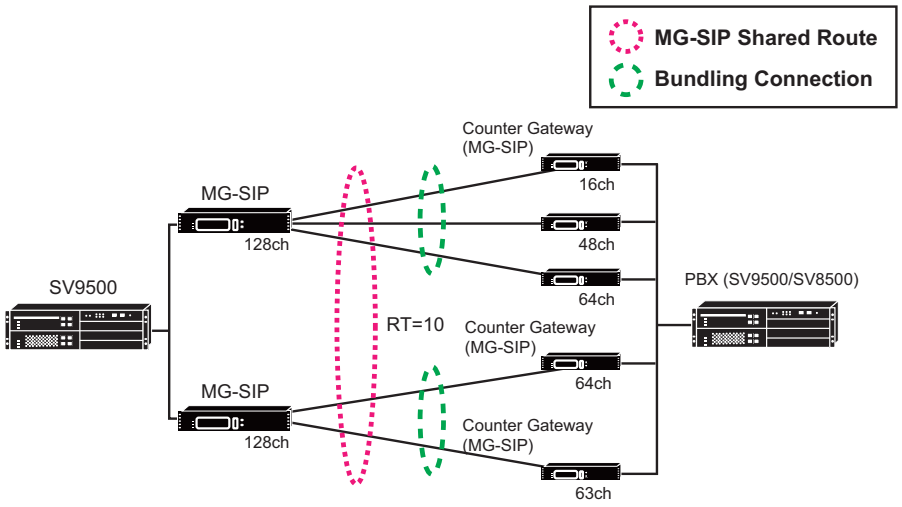
Combination of Route Splitting and MG-SIP Shared Route

One MG-SIP can use both Route Splitting and MG-SIP Shared Route at the same time.



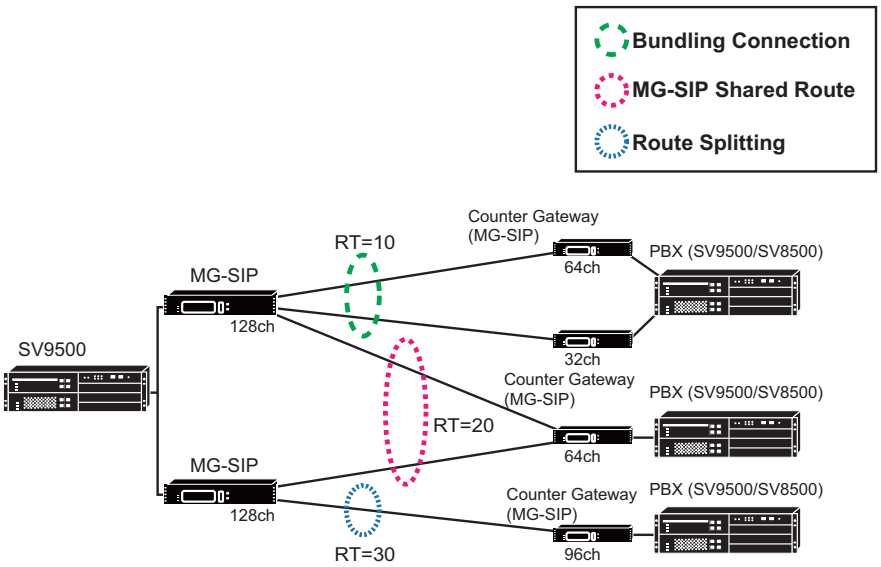
Combination of Bundling Connection and MG-SIP Shared Route

One MG-SIP can use both Bundling Connection and MG-SIP Shared Route at the same time.



Combination of Route Splitting, Bundling Connection and MG-SIP Shared Route

One MG-SIP can use Route Splitting, Bundling Connection and MG-SIP Shared Route at the same time.



Operating Procedure

None.

Service Conditions

<Important Notices>

1. To use this feature, destination information needs to be assigned to Path ID of the AMGIL command per counter SIP gateway. A maximum of 32 Path IDs (1 - 255) can be assigned per MG-SIP.
2. Make sure that Usable CH Number of the AMGIL command is set to 128ch even when number of the registered trunks is less than 128ch. If you set the Usable CH Number to less than 128 and later register additional trunks to exceed the previously set limit value, you need to restart the MG-SIP.
3. Restart of the MG-SIP is not required if the increase in the number of routes/trunks or paths has no effect on the ACSC command.
4. Up to 62 MG-SIPs can be accommodated in a system due to the restriction of CSC Group (CSCG) of the ACSC command (range from 130 to 254).
5. When path data is changed, reset the path with the AMGIL command for the changes to take effect.
6. When the office data for Path Splitting is assigned through the AMGIL command, MG-SIPs that do not support Path Splitting cannot perform the registration with that office data. When MG-SIPs supporting Path Splitting register to the system without Path Splitting office data, they have Point-to-Point connection.
7. The LENS data for virtual signaling channels which has been set through the AMGIL command must not be deleted by the ATRK command during the operation. To delete the LENS data, make the MG-SIP in an unusable state, then delete the trunk data by the ATRK command.
8. Do not set the ACIC2 command data when using this service. When the command data has already been set, be sure to delete it.
9. When using this service, enable MG-SIP TIE Line Path Splitting through the configuration command *set mp_mode* on the MG-SIP.

<Interactions with Other Features>

1. Designated trunk connection assigned by the TOGC command is available, but INDIVIDUAL TRUNK ACCESS [I-4] is not available.

<Hardware/Software>

1. This service is available for MG-128SIPMGJ-B with firmware versions SP-4060 MGSIP PROG-L Issue 3 or later.
2. This feature supports as a counter gateway the following MG-SIPs:

    - MG-128SIPMGJ/MG-128SIPMGJ-B of SV9500/SV8500 (including SV9500/SV8500 that does not support this feature. Also, there is not any firmware condition regarding the counter gateway MG-SIP).
    - SIP gateway devices controlled by SV9300.

<Conditions Specific to This Feature>

1. When a single MG-SIP connects to multiple counter gateways, the maximum number of simultaneous speeches depends on that of the MG-SIP. In the case of MG-128SIPMGJ-B, the maximum number of simultaneous speeches is 128 channels.
2. A single MG-SIP can connect to a maximum of 32 counter gateways.
3. Route can be assigned on a per counter gateway basis. Up to 32 routes are available for a single MG-SIP.
4. All the trunks used by a single MG-SIP need to be accommodated in the same LP.

5. Multiple external routes can be assigned in a highway (four groups). In this case, trunks to use must be assigned to continuous channels necessary for the counter gateway. When an external route does not fit within one highway (four groups), the trunks must be split across multiple highways. (the remaining trunks need to be assigned from the beginning of another group).

   Note: When an already registered trunk is deleted during the operation, the trunk cannot be used to make/receive a call any more. A path to which trunks are not assigned continuously becomes unusable after the registration or the reset of the path.

6. A maximum number of simultaneous speeches between an MG-SIP and a counter gateway depends on the value of the CH parameter in List of Path Data of the AMGIL command. This value must be the same as the number of channels on the counter gateway. When the number of speech channels on the MG-SIP is larger than number of speech channels on the counter gateway, make some trunks on the MG-SIP busy to adjust the number of the channels.
7. When deleting path data, make sure that all the trunks to be deleted are idle. Access the MBTK command and select "2: Make Busy (Bothway)" for the MB parameter to place the trunks in make busy state.
8. A route can be shared by up to two MG-SIPs. A maximum of 255 channels can be used with the same route because up to 255 trunk numbers can be assigned to a route.
9. MG-SIPs sharing a route by using MG-SIP SHARED ROUTE [M-123] need to be assigned in the same LP.
10. When two MG-SIPs connect to the same counter gateway with MG-SIP SHARAED ROUTE [M-123], the gateway must support Bundling Connection.
11. The same IP address cannot be assigned to multiple paths.
12. After path data of MG-SIPs is created or deleted with the AMGIL command during the operation, the MG-SIPs must be restarted and perform re-registration.
13. Path data registered in Telephony Server (AMGIL path data) and registered in MG-SIP will be monitored at a prescribed period (at about 50 second intervals). When difference exists between them, the system message 35-N will be issued and the corresponding path will be busy. In this case, assign path data again through the AMGIL command or reset the path.

    Note: Monitoring of path data is suspended during the registration/deletion/reset of the AMGIL path data. After the process, the monitoring is resumed.

14. Note the following points when adding the AMGIL path data for an MG-SIP which has been already operated.

    - When path data on MG-SIP is successfully registered:

      System message 35-M will be issued.
      If you use the DIMG command (View Type = Media Gateway Information) to verify registration status, it displays:
      - a path number in the Path ID field
      - "Operating" as the line status in the ISDN Status field

    - When registration of path data on MG-SIP failed:

      According to the condition of "13", the system message 35-N will be issued.
      If you use the DIMG command (View Type = Media Gateway Information) to verify registration status, it displays:
      - no path number in the Path ID field
      - "Trouble of remote factor" as the line status in the ISDN Status field

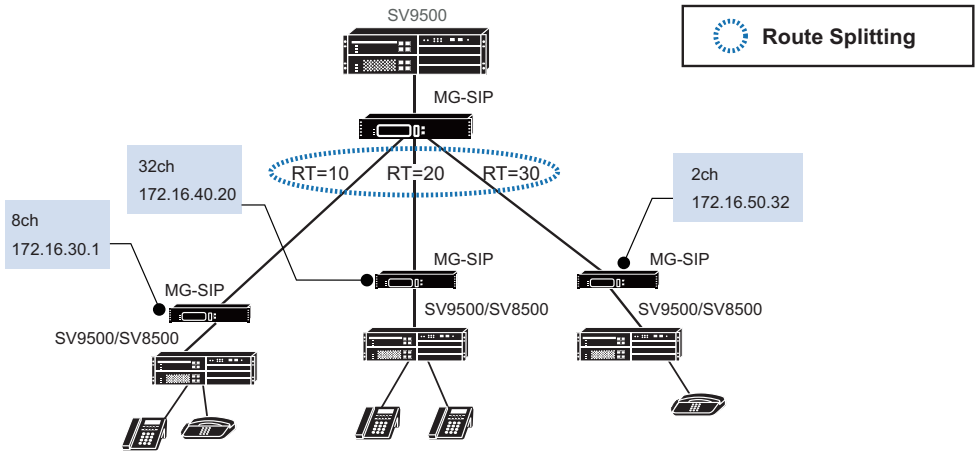
<Other Conditions>

None.

Programming

Route Splitting

Office data settings for Route Splitting are as follows:



STEP 1: **ASYD**
SYS1,
Index 64, Bit 7 = 1 (Music-on-Hold service is in service)
Index 186, Bit 6 = 1 (ISDN is in service)
Index 187 = 00
Index 220, Bit 6 = 0 (ISDN service is in service)
Index 474 (Guard Timer for Originating Call (MTC × TC)
Bit 0-3: MTC (Miscellaneous Timer Counter)
Bit 4-6: TC (Timer Counter) = 1 (64 ms period)

STEP 2: **ASYDL**
SYS1
Index 810, Bit 2 = 0/1 [Internal Holding Tone (IP terminals)/External Holding Tone (EMA and PLO)] **Note 1**
Index 880, Bit 1 = 1 (Internal PHI-PRI/BRI is enabled)

Note 1: When External music is used, see MUSIC ON HOLD [M-7].

STEP 3: **ASYDL/ASYDN** - SYS1, Index 1145, Bit 4 = 1 (Stores C.O. call Information collected during operation of the MG-SIP TIE Line feature.)

STEP 4: **ALOCL** - Assign ToS settings for Internal PHI-BRI. When assigning ToS, be sure to assign LOC-ID = 0.

TYPE: Select “Network Address”.
IP ADDRESS: 0.0.0.0
MASK BIT: 32
NETWORK ADDRESS: Display only.
LOC-ID: 0
Setting Type of Signaling Packet Service: Assign ToS value to control MG-SIP through the Internal PHI-BRI. Click the check box to ON when data setting for Type of Service (ToS) field is required. When this check box remains OFF, the default value is applied.
IP PRECEDENCE/DIFFSERV: Choose either IP PRECEDENCE or DIFFSERV.
Priority (PRECEDENCE /DSCP)
- When IP PRECEDENCE is selected
Specify its priority [0-7] and the type (LOW DELAY/THROUGH PUT/RELIABILITY/MIN COST).
- When DIFFSERV is selected
Specify its priority [0-63].

Note: One of the following is required to allow the changes to Setting Type of Signaling Packet Service data for already-operating equipment (such as PH and IP station) to take effect.
- Initialize the Internal PHI-BRI (Initialize the Internal PHI-BRI module with the ADPM command) or system.
- Re-register the IP station.

STEP 5: **ADTM** - Assign port numbers if necessary (default value = 64000).

Note: When you change the default port number with this command, you need to initialize the Internal PHI-BRI module with the ADPM command.

STEP 6: **ADPM** - Assign ToS settings for Internal PHI-BRI. When assigning ToS, be sure to assign LOC-ID = 0.
PH TYPE = Internal PHI (BRI)
Menu = Initialize

Note: When MG-BRI or other MG-SIPs are used, they are also initialized.

STEP 7: **ALIDL** - Specify the LENS (five digits) of Internal PHI-BRI.
PH LENS: Specify an even-numbered Group Number. The LENS can be allocated on either a Virtual PIR or a Physical PIR.
PH TYPE = Internal PHI
REGMAX: Specify the registration capacity (1-6144).

Note: When the total number of channels using PHI-BRI exceeds 256, increase the value of REGMAX.

STEP 8: **ARTD/ARTDN** - Specify the route class data on Virtual Speech Channel as follows. It is not necessary to register attributes for the Virtual Signaling Channel.
CDN6: TCL = 1 (DDD Line)
CDN65: INT = 10 [Q-Sig, (ETS 300 172)/IS-11572]
CDN66: DC = 1-15 (DID Addressing number)
CDN115: CMRT = 1 (Single route number used by multiple ISDN trunks/MG-SIPs)
Assign the CDN111(ADVPRA) = 1 (ISDN PRI Failure Routing Service) to dummy route.

Note: To change the data for existing route (RT), initialize or reboot the MG-SIP.

STEP 9: **ARTI/ARTIN** - Specify the application data for trunk.
CDN47: INTD = 2 (SIP)

CDN68: VIR = 2 (MG connection)

CDN77: RA_RT (Register type (MG-SIP) = 0 (Virtual Register)/1 (Register card)

Note: When Automated Attendant service is used with MG-SIP while RST card and IPPAD card are accommodated in the system, set the RA_RT parameter to 1.

CDN96: MGSIPSV = 3 (TIE Line connection)

Assign the following parameters for FAX use:

CDN71: FXD = 1 (IP FAX data is in service)

CDN72: FXJS (Jitter Buffer Size for IP FAX [1-30]×10 ms) ex. Enter "15" for 150 ms.

CDN73: FXPT = 1 (Payload Type for IP FAX is G.711)

CDN74: FXPS (Payload Size for IP FAX [1-4]×10 ms) ex. Enter "2" for 20 ms.

Note: To change the data for existing route (RT), initialize or reboot the MG-SIP.

STEP 10: **ATRK** - Specify trunk data on Virtual Speech Channel and Virtual Signaling Channel.

RT: Specify Route Number [1-255]

TK: Specify Trunk Number [1-255]

LENS: Specify Accommodated Location (Six digits)

Register routes/trunks/channels to use for each path.

Path for MG-SIP	Number of Channels	Route Number	LENS	
			Highway 1	Highway 2
Path A	8	10	020040 - 020047	-
Path B	32	20	020050 - 020077	020160 - 020167
Path C	2	30	-	020170 - 020171
Total	42	-	-	-

Note: When using this command, consider the following:

- Virtual speech channels for each path must be assigned to continuous channels from the first channel of a highway group. When all the channels cannot fit within one highway and need to be split across multiple highway groups, assign remaining channels from the beginning of the next group.
- Only MG-SIP virtual communication channel can be registered for HW (4 group) with at least one MG-SIP virtual communication channel already registered.
- Do not assign the LENS that belongs to Even-numbered module group, Unit=0 and Group=0 to a trunk, because they are not supported by MG-SIP.

STEP 11: **ADPC** - Assign Point Code to Virtual Speech Channel. On the office data, a vacant number is used for a Point Code. For more details, see the Command Manual.

RT: Specify an external route number.

PC: Specify a vacant Point Code (Point Code used by Common Line/ISDN cannot be used).

Note: Assign the same Point Code to all the routes registered by the ATRK command.

STEP 12: **ACSC** - Assign the first channel of 32 channels (a highway) LENS to the even-numbered CSC group, and LENS of Virtual Signaling Channels to the odd-numbered CSC group.

Note: Assigning LENS of Virtual Signaling Channels to odd-numbered CSC group makes any setting with the ACIC2 command unnecessary.

CSCG: Specify the CSC Group Number (even numbers from 130 to 255 and odd number continuous with them)

GROUP: Specify the CIC Group Number
CCH: Specify the first channel of 32-channel LENS for even-numbered CSCG and Virtual Signaling LENS for odd-numbered CSCG.

<Example of Data Settings>

CSC Group Number (CSCG)	CIC Group Number (CICG)	LENS of Signaling Channel (CCH)			Remarks
		MG 00-07	U 0-3	G 00-23	
130	0	02	0	4	Path A/B
	1	02	0	16	Path B/C
	2	02	0	4	**Note 2**
	3	02	0	4	**Note 2**
	4	02	0	4	**Note 2**
	5	02	0	4	**Note 2**
	6	02	0	4	**Note 2**
	7	02	0	4	**Note 2**

Note 2: Apply the same settings as CICG=0 to the remaining CIC Group Numbers (CICG).

In the next step, assign LENS of Virtual Signaling Channel to the next CSCG (greater by one than the last CSCG).

Note: When your system contains a single MG-SIP, assign the same LENS to all the numbers.

CSC Group Number (CSCG)	CIC Group Number (CICG)	LENS of Signaling Channel (CCH)			Remarks
		MG 00-07	U 0-3	G 00-23	
131	0	02	0	4	
	1	02	0	4	
	2	02	0	4	
	3	02	0	4	
	4	02	0	4	
	5	02	0	4	
	6	02	0	4	
	7	02	0	4	

STEP 13: **ACIC1** - Specify LENS of Virtual Signaling Channel on Point Code as follows:
PC: Enter Point Code specified by ADPC.
CSCG: Enter CSC group specified by ACSC.

STEP 14: **MBTK** - Cancel Make Busy of Virtual Speech Channel.

STEP 15: **AMGIL** - Assign the data regarding MG-SIP as follows:

MG-ID: Enter MAC Address (six bytes) of MG-SIP. * Enter the Telephony Server side (LAN side) MAC address.

FUNCTION = Setting

KIND = SIP-MG

LINE = 0 (fixed)

CH = 0 (fixed)

LENS: Enter the accommodated location of Virtual Signaling Channel (the first number of HW).

Retry = 0

Setting Type of Signaling Packet Service: Specify QoS settings of Signaling that is used when Internal PHI-BRI connects MG-SIP.

Usable CH Number: Enter the number of channels to be used per MG-SIP (1-128).

Note: Be sure to set 128ch (fixed) when this feature (MG-SIP TIE Line Path Splitting) is used.

Service of Authentication Exists: Unchecked

Setting of Path Data

Note: When the Set button of the command is pressed while this check box is OFF, all the registered Path data is cleared.

Note: Assign the following data as you have created with the ATRK command. Assign the counter IP address and port number according to the settings of counter gateway.

List of Path Data

INDEX: Index of Path data (1-32)

Select: Choose the check box for the path to be registered.

Note: When the Set button of the command is pressed while the check box for the already registered path is OFF, the path data is cleared.

Path ID: Identification Number of Path (1-255)

RT: External Route Number (1-255)

* Enter the route number (route number of Virtual Speech Channel for MG-SIP assigned by the ATRK command) to be assigned to this path.

LGRT: Logical Route Number (display only)

TK: Trunk Number (1-255)

* Enter the first trunk number of the trunks (trunk number of Virtual Speech Channel for MG-SIP assigned by the ATRK command) to be assigned to this path.

CH: Enter the number of MG-SIP channels to be assigned to this path.

IP Ver: IP Address Version = 1 (IPv4)

GW IP: Enter an IP address of the counter gateway (0.0.0.1-255.255.255.254).

GW PORT: Enter a port number of the counter gateway (1024-65535).

FLT: Gateway Fault Detection Capability (0/1)

* To enable MG-SIP Fault Detection per Path, set this parameter to 1 (Enable).

<Example of AMGIL Command Settings>

Path of MG-SIP	Select	Path ID	RT	LGRT	TK	CH	IP Ver	GW IP	GW PORT	FLT
Path A	ON	1	10	10	1	8	IPv4	172.16.30.1	5060	Enable
Path B	ON	2	20	20	1	32	IPv4	172.16.40.20	5060	Enable
Path C	ON	3	30	30	1	2	IPv4	172.16.50.32	5060	Enable

STEP 16: **AMGVL**- Register the voice control data when connecting to MG-SIP. If this command data is not assigned, the values assigned by the AIVCL command will be set automatically.
A RT: Specify an external route number of MG-SIP Virtual Speech Channel Route Number.
TYPE: Select “Route Specified” or “Location ID Specified”.
B RT: Specify an external route number of MG-SIP to be connected.
B LOC-ID: Specify a Location ID to which IP stations to be connected belong (Normal setting).
* When IPPAD is accommodated in your system, assign LOC-ID=0.
Setting Type of Voice Packet Service: Specify QoS settings of Voice Speech
TYPE: Displayed when the check box of “Setting of Payload” is ON. Specify Payload type. *G.711 and 20ms must be set for public SIP network.
SIZE: Displayed when the check box of “Setting of Payload” is ON. Specify Payload size.

STEP 17: **ANPD/ANPDL/ANPDN**- Assign the minimum number of digits needed to determine the service required to the first digit received.
TN: Specify a Tenant Number.
1stDC: Specify the first digit of service access code.
CI: Choose a Connection Status Index.
NND: Specify the minimum necessary number of digits for each Access Code.

STEP 18: **ASPA/ASPAL/ASPAN**- Define the Access Codes for each counter office.
TN: Specify a Tenant Number
ACC: Specify an Access Code. [Maximum 6 digits]
CI: Choose a Connection Status Index.
SRV = LCR (Least Cost Routing)
RT or LGRT: Specify a Dummy Route Number.

STEP 19: **AFRS/AFRSL/AFRSN**- Assign a Number Pattern Code (NPC) and Outgoing Route Selection Number (OPR) to the Dummy Route.
TN: Specify a Tenant Number.
RT or LGRT: Specify a Dummy Route Number.
NPC: Define a Number Pattern Code. (maximum 16 digits)
OPR: Define an Outgoing Route Selection Number. (0-4000)

STEP 20: **AOPR/AOPRL/AOPRN**- Assign Virtual Speech Channels to each Outgoing Route Selection Pattern Routing (OPR) number.
TDPTN: Specify a Time of Day Change Pattern Number. (0-7)

OPR: Specify an Outgoing Route Selection Number. (0-4000)
RA: Specify the Route Advancing Order. (0-7)
E: Specify Route Advancing End Indication. (0/1)
RT or LGRT: Specify an Outside Route Number. (1-255)
SKIP: Specify the number of digits to be skipped from dialed numbers. (0-24) **Note 3**
PNL: Specify a Pattern Number Location. (0-999)

Note 3: If an Access Code is to be removed from the dialed number before the call is sent out, specify the number of digits of the Access Code for SKIP. Otherwise, incorrect number will be passed to the MG-SIP. When the Multi-Point connection is enabled on the MG-SIP, change the setting on the MG-SIP, accordingly.

STEP 21: **AMND/AMNDL/AMNDN**- AMND/AMNDL/AMNDN - Specify the number of maximum necessary digits for a Destination Code (DC).
TN: Specify a Tenant Number
DC: Specify a Destination Code. (Maximum 10 digits)
MND: Specify maximum necessary digits for the DC. (1-24)

STEP 22: **ARNP/ARNPL/ARNPN**- Assign an Access Code (ACC) to be used for Self Office Identification Code and each individual routing.
RT or LGRT: Specify a Route Number. (0-255)
ACC: Specify an Access Code. (Maximum 4 digits)

Note: For an Open Numbering network, the following data should be assigned: RT = 0 ACC: The Office Code of the self office (including ACC for the originating call)

Note: All Offices within an Open Numbering network must have an Office Code with the same number of digits.

STEP 23: **ACNAL**- Assign the Calling Number Pattern (CNP) on an Access Code basis. *When this command is used, assign ARTD command CDN66 to DC = 15.

- Settings for a Call Originated to SIP Network
  Specify Calling Number Pattern for Access Code of the gateway to connect to by adding information about the route for which Caller ID Notification is required.

  IC/OG = OG (Outgoing Call)
  RT: Enter External Route Number for which Caller ID Notification is required per Access Code.
  ACC: Enter Access Code (up to 16 digits).
  CNP: Enter Calling Number Pattern (a number from 1 to 255).

  Note: To modify CNP set by the ACNAL command, use the ACND command; ACNDN command cannot be used.

- Settings for a Call Arriving from SIP Network
  Specify Dialed Number Confirmation for Access Code by adding information about the route for inbound DID calls.

  IC/OG = IC (Incoming Call)
  RT: Enter External Route Number to be arrived.
  ACC: Enter Access Code (up to 16 digits).

DC: Dialed Number Confirmation (1-8)

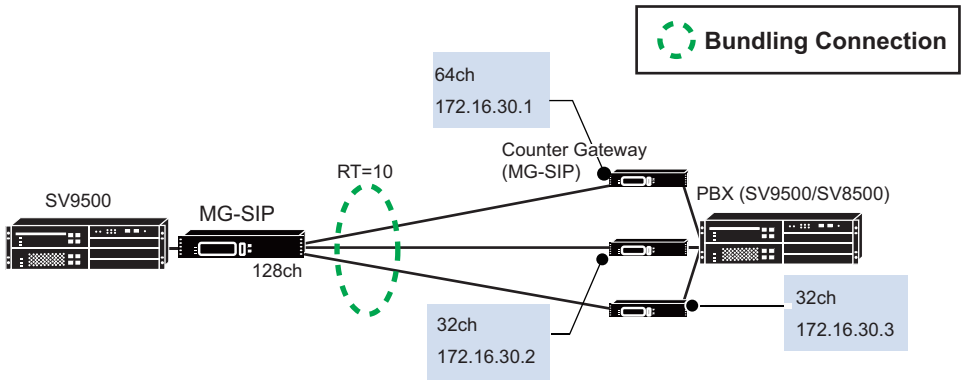
STEP 24: **ACND**- Assign the data for number change to CNP that has been specified by the ACNAL/ACNP command.
CNP: Calling Number Pattern
SKIP: Number of Skip Digits
ADD: Number of Digits of the Additional Number
DC: Additional Digit Information

STEP 25: **ACDD**- Change Digit Code according to DID Addressing.
DAY/NIGHT: Day/Night Mode
RT: Enter the B-Channel Route Number.
DC: Digit string in the dialed number to be replaced with a string specified in CDC. **Note 4**
CDC: Enter Change Digit Code (Maximum 24 digits)
AD: Enter Additional Dialing
XFR: Enable/Disable Transfer Service

Note 4: Enter the same value (number of digits) as that of DC of the ACNAL command. When the ACNAL command is not used, the value assigned to the DC of the ARTD command is applied.

Bundling Connection

Office data settings for Bundling Connection are as follows:



* The following shows only the differences from the office data settings for Route Splitting.

STEP 1: **ATRK**- Assign trunk data on Virtual Speech Channel.
RT: Specify Route Number (1-255).
TK: Specify Trunk Number (1-255).
LENS: Specify Accommodated Location (six digits).

Register routes/trunks/channels to use for each path.

Path for MG-SIP	Number of Channels	Route Number	LENS			
			Highway 1	Highway 2	Highway 3	Highway 4
Path A	64	10	021000 - 021037	021080-021117	-	-
Path B	32	10	-	-	022000-022037	-
Path C	32	10	-	-	-	023200-023237
Total	128	-	-	-	-	-

Note: When using this command, consider the following:

- Virtual speech channels for each path must be assigned to continuous channels from the first channel of a highway group. When all the channels cannot fit within one highway and need to be split across multiple highway groups, assign remaining channels from the beginning of the next group.
- Only than MG-SIP virtual communication channel can be registered for HW (4 group) with at least one MG-SIP virtual communication channel already registered.
- Do not assign the LENS that belongs to Even-numbered module group, Unit=0 and Group=0 to a trunk, because they are not supported by MG-SIP.

STEP 2: **ACSC** - Assign the first channel of 32 channels (a highway) LENS to the even-numbered CSC group, and LENS of Virtual Signaling Channel to the odd-numbered CSC group.
CSCG: Specify the CSC Group Number (even numbers from 130 to 255 and odd number continuous with them)

GROUP: Specify the CIC Group Number
CCH: Specify the first channel of 32-channel LENS for even-numbered CSCG and Virtual Signaling LENS for odd-numbered CSCG.

<Example of Data Settings>
Even-numbered CSCG for MG-SIP: the first LENS of 32-channels
The top of CICG for MG-SIP: LENS of Virtual Signaling Channel

CSC Group Number (CSCG)	CIC Group Number (CICG)	LENS of Signaling Channel (CCH)			Remarks
		MG 00-07	U 0-3	G 00-23	
130	0	02	1	0	Path A
	1	02	1	8	
	2	02	2	0	Path B
	3	02	3	20	Path C
	4	02	1	0	**Note 5**
	5	02	1	0	**Note 5**
	6	02	1	0	**Note 5**
	7	02	1	0	**Note 5**

Note 5: Apply the same settings as CICG=0 to the remaining CIC Group Numbers (CICG).

In the next step, assign LENS of Virtual Signaling Channel to the next CSCG (greater by one than the last CSCG).

Note: When your system contains a single MG-SIP, assign the same LENS to all the numbers.

CSC Group Number (CSCG)	CIC Group Number (CICG)	LENS of Signaling Channel (CCH)			Remarks
		MG 00-07	U 0-3	G 00-23	
131	0	02	1	0	
	1	02	1	0	
	2	02	1	0	
	3	02	1	0	
	4	02	1	0	
	5	02	1	0	
	6	02	1	0	
	7	02	1	0	

STEP 3: **AMGIL** - Assign the data regarding MG-SIP as follows:

MG-ID: Enter MAC Address (six bytes) of MG-SIP. * Enter the Telephony Server side (LAN side) MAC address.

FUNCTION = Setting

KIND = SIP-MG

LINE = 0 (fixed)

CH = 0 (fixed)

LENS: Enter the accommodated location of Virtual Signaling Channel (the first number of HW).

Retry = 0

Setting Type of Signaling Packet Service: Specify QoS settings of Signaling that is used when Internal PHI-BRI connects MG-SIP.

Usable CH Number: Enter the number of channels to be used per MG-SIP (1-128).

Note: Be sure to set 128ch (fixed) when this feature (MG-SIP TIE Line Path Splitting) is used.

Service of Authentication Exists: Unchecked

Setting of Path Data

Note: When the Set button of the command is pressed is pressed while this check box is OFF, all the registered Path data is cleared.

Note: Assign the following data as you have created with the ATRK command. Assign the counter IP address and port number according to the settings of counter gateway.

List of Path Data

INDEX: Index of Path data (1-32)

Select: Choose the check box for the path to be registered.

Note: When the Set button of the command is pressed while the check box for the already registered path is OFF, the path data is cleared.

Path ID: Identification Number of Path (1-255)

RT: External Route Number (1-255)

* Enter the route number (route number of Virtual Speech Channel for MG-SIP assigned by the ATRK command) to be assigned to this path.

LGRT: Logical Route Number (display only)

TK: Trunk Number (1-255)

* Enter the first trunk number of the trunks (trunk number of Virtual Speech Channel for MG-SIP assigned by the ATRK command) to be assigned to this path.

CH: Enter the number of MG-SIP channels to be assigned to this path.

IP Ver: IP Address Version = 1 (IPv4)

GW IP: Enter an IP address of the counter gateway (0.0.0.1-255.255.255.254).

GW PORT: Enter a port number of the counter gateway (1024-65535).

FLT: Gateway Fault Detection Capability (0/1)

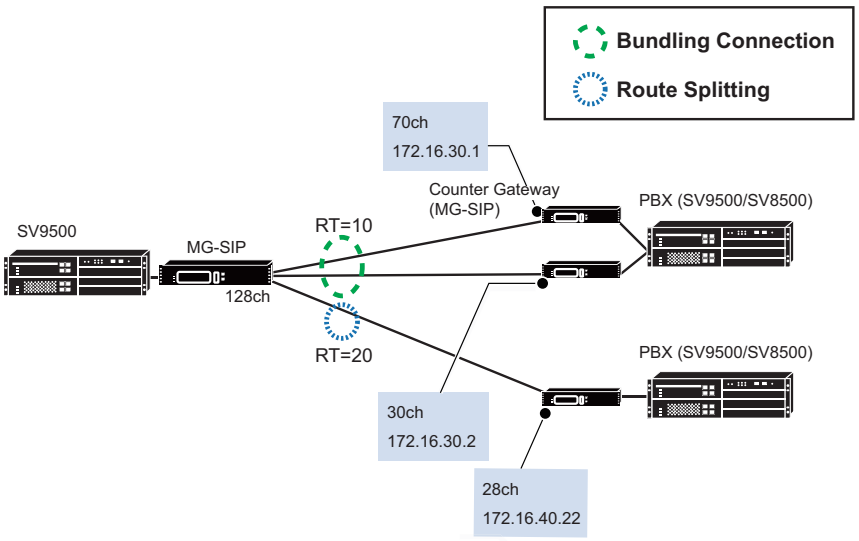
* To enable MG-SIP Fault Detection per Path, set this parameter to 1 (Enable).

<Example of AMGIL Command Settings>

Path of MG-SIP	Select	Path ID	RT	LGRT	TK	CH	IP Ver	GW IP	GW PORT	FLT
Path A	ON	1	10	10	1	64	IPv4	172.16.30.1	5060	Enable
Path B	ON	2	10	10	65	32	IPv4	172.16.30.2	5060	Enable
Path C	ON	3	10	10	97	32	IPv4	172.16.30.3	5060	Enable

Combination of Route Splitting and Bundling Connection

Office data settings for Combination of Route Splitting and Bundling Connection are as follows:



* The following shows only the differences from the office data settings for Route Splitting.

STEP 1: **ATRK**- Assign trunk data on Virtual Speech Channel.
RT: Specify Route Number (1-255).
TK: Specify Trunk Number (1-255).
LENS: Specify Accommodated Location (six digits).

Register routes/trunks/channels to use for each path.

Path for MG-SIP	Number of Channels	Route Number	LENS			
			Highway 1	Highway 2	Highway 3	Highway 4
Path A	70	10	020040- 020077	020160-020197	020080-020085	-
Path B	30	10	-	-	020086-020117	020200-020203
Path C	28	20	-	-	-	020204-020237
Total	128	-	-	-	-	-

Note: When using this command, consider the following:
- Virtual speech channels for each path must be assigned to continuous channels from the first channel of a highway group. When all the channels cannot fit within one highway and need to be split across multiple highway groups, assign remaining channels from the beginning of the next group.
- Only MG-SIP virtual communication channel can be registered for HW (4 group) with at least one MG-SIP virtual communication channel already registered.
- Do not assign the LENS that belongs to Even-numbered module group, Unit=0 and Group=0 to a trunk, because they are not supported by MG-SIP.

STEP 2: **ADPC** - Assign Point Code to Virtual Speech Channel. On the office data, a vacant number is used for a Point Code. For more details, see the Command Manual.

RT: Specify an external route number.

PC: Specify a vacant Point Code (Point Code used by CCIS/ISDN cannot be used).

Note: Assign the same Point Code to all the routes registered by the ATRK command.

STEP 3: **ACSC** - Assign the first channel of 32 channels (a highway) LENS to the even-numbered CSC group, and LENS of Virtual Signaling Channel to the odd-numbered CSC group.

CSCG: Specify the CSC Group Number (even numbers from 130 to 255 and odd number continuous with them)

GROUP: Specify the CIC Group Number

CCH: Specify the first channel of 32-channel LENS for even-numbered CSCG and Virtual Signaling LENS for odd-numbered CSCG.

<Example of Data Settings>

Even-numbered CSCG for MG-SIP: the first LENS of 32-channels

The top of CICG for MG-SIP: LENS of Virtual Signaling Channel

CSC Group Number (CSCG)	CIC Group Number (CICG)	LENS of Signaling Channel (CCH)			Remarks
		MG 00-07	U 0-3	G 00-23	
130	0	02	0	4	Path A
	1	02	0	16	
	2	02	0	8	Path A/B
	3	02	0	20	Path B/C
	4	02	0	4	**Note 6**
	5	02	0	4	**Note 6**
	6	02	0	4	**Note 6**
	7	02	0	4	**Note 6**

Note 6: Apply the same settings as CICG=0 to the remaining CIC Group Numbers (CICG).

In the next step, assign LENS of Virtual Signaling Channel to the next CSCG (greater by one than the last CSCG).

Note: When your system contains a single MG-SIP, assign the same LENS to all the numbers.

CSC Group Number (CSCG)	CIC Group Number (CICG)	LENS of Signaling Channel (CCH)			Remarks
		MG 00-07	U 0-3	G 00-23	
131	0	02	0	4	
	1	02	0	4	
	2	02	0	4	
	3	02	0	4	
	4	02	0	4	
	5	02	0	4	
	6	02	0	4	
	7	02	0	4	

STEP 4: **AMGIL** - Assign the data regarding to MG-SIP as follows:

MG-ID: Enter MAC Address (six bytes) of MG-SIP. *Enter the Telephony Server side (LAN side) MAC address.

FUNCTION = Setting

KIND = SIP-MG

LINE = 0 (fixed)

CH = 0 (fixed)

LENS: Enter the accommodated location of Virtual Signaling Channel (the first number of HW).

Retry = 0

Setting Type of Signaling Packet Service: Specify QoS settings of Signaling that is used when Internal PHI-BRI connects MG-SIP.

Usable CH Number: Enter the number of channels to be used per MG-SIP (1-128).

Note: Be sure to set 128ch (fixed) when this feature (MG-SIP TIE Line Path Splitting) is used.

Service of Authentication Exists: Unchecked

Setting of Path Data

Note: When the Set button of the command is pressed while this check box is OFF, all the registered Path data is cleared.

Note: Assign the following data as you have created with the ATRK command. Assign the counter IP address and port number according to the settings of counter gateway.

List of Path Data

INDEX: Index of Path data (1-32)

Select: Choose the check box for the path to be registered.

Note: When the Set button of the command is pressed while the check box for the already registered path is OFF, the path data is cleared.

Path ID: Identification Number of Path (1-255)

RT: External Route Number (1-255)

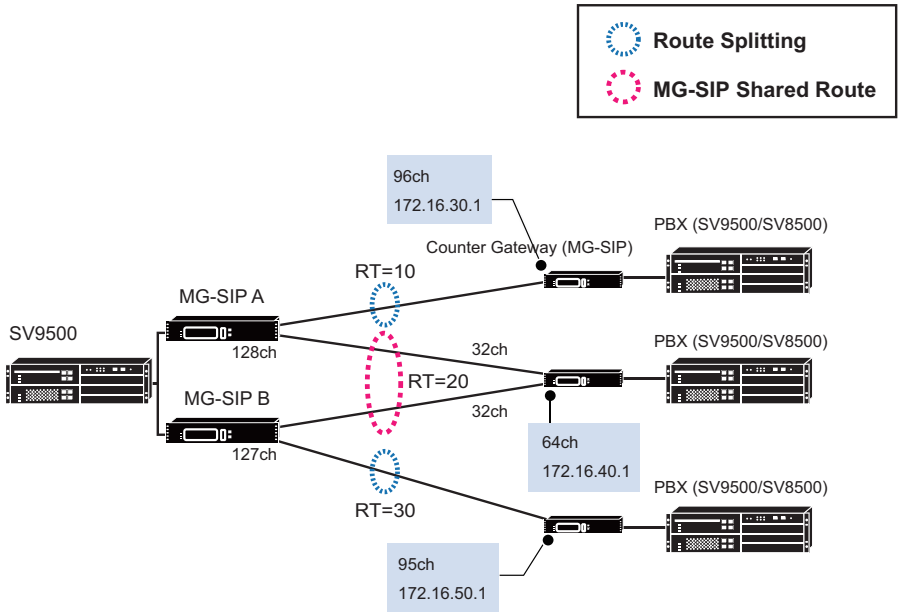
* Enter the route number (route number of Virtual Speech Channel for MG-SIP assigned by the ATRK command) to be assigned to this path.
LGRT: Logical Route Number (display only)
TK: Trunk Number (1-255)
* Enter the first trunk number of the trunks (trunk number of Virtual Speech Channel for MG-SIP assigned by the ATRK command) to be assigned to this path.
CH: Enter the number of MG-SIP channels to be assigned to this path.
IP Ver: IP Address Version = 1 (IPv4)
GW IP: Enter an IP address of the counter gateway (0.0.0.1-255.255.255.254).
GW PORT: Enter a port number of the counter gateway (1024-65535).
FLT: Gateway Fault Detection Capability (0/1)
* To enable MG-SIP Fault Detection per Path, set this parameter to 1 (Enable).

<Example of AMGIL Command Settings>

Path of MG-SIP	Select	Path ID	RT	LGRT	TK	CH	IP Ver	GW IP	GW PORT	FLT
Path A	ON	1	10	10	1	70	IPv4	172.16.30.1	5060	Enable
Path B	ON	2	10	10	71	30	IPv4	172.16.30.2	5060	Enable
Path C	ON	3	20	20	1	28	IPv4	172.16.40.22	5060	Enable

Combination of MG-SIP Shared Route and Route Splitting

Office data settings for Combination of MG-SIP Shared Route and Route Splitting are as follows:



* The following shows only the differences from the office data settings for Route Splitting.

STEP 1: **ATRK**- Assign trunk data on Virtual Speech Channel.
RT: Specify Route Number (1-255).
TK: Specify Trunk Number (1-255).
LENS: Specify Accommodated Location (six digits).

Register routes/trunks/channels to use for each path.

	Path for MG-SIP	Number of Channels	Route Number	LENS			
				Highway 1	Highway 2	Highway 3	Highway 4
MG-SIP A	Path A	96	10	030000-030037	030040-030077	030080-030117	-
	Path B	32	20	-	-	-	021000-021037
MG-SIP B	Path B	32	20	021040-021077	-	-	-
	Path C	95	30	-	022000-022037	022040-022077	022080-022116

Note: When using this command, consider the following:

- Virtual speech channels for each path must be assigned to continuous channels from the first channel of a highway group. When all the channels cannot fit within one highway and need to be split across multiple highway groups, assign remaining channels from the beginning of the next group.
- Only MG-SIP virtual communication channel can be registered for HW (4 group) with at least one MG-SIP virtual communication channel already registered.
- Do not assign the LENS that belongs to Even-numbered module group, Unit=0 and Group=0 to a trunk, because they are not supported by MG-SIP.

STEP 2: **ACSC** - Assign the first channel of 32 channels (a highway) LENS to the even-numbered CSC group, and LENS of Virtual Signaling Channel to the odd-numbered CSC group.

CSCG: Specify the CSC Group Number (even numbers from 130 to 255 and odd number continuous with them)

GROUP: Specify the CIC Group Number

CCH: Specify the first channel of 32-channel LENS for even-numbered CSCG and Virtual Signaling LENS for odd-numbered CSCG.

<Example of Data Settings>

Even-numbered CSCG for MG-SIP: the first LENS of 32-channels

The top of CICG for MG-SIP: LENS of Virtual Signaling Channel

CSC Group Number (CSCG)	CIC Group Number (CICG)	LENS of Signaling Channel (CCH)			Remarks
		MG 00-07	U 0-3	G 00-23	
130	0	03	0	0	Path A
	1	03	0	4	
	2	03	0	8	
	3	02	1	0	Path B
	4	02	1	4	
	5	02	2	0	Path C
	6	02	2	4	
	7	02	2	8	

In the next step, assign LENS of Virtual Signaling Channel to the next CSCG (greater by one than the last CSCG).

CSC Group Number (CSCG)	CIC Group Number (CICG)	LENS of Signaling Channel (CCH)			Remarks
		MG 00-07	U 0-3	G 00-23	
131	0	03	0	0	MG-SIP A
	1	03	0	0	
	2	03	0	0	
	3	03	0	0	
	4	02	1	4	MG-SIP B
	5	02	1	4	
	6	02	1	4	
	7	02	1	4	

STEP 3: **AMGIL** - Assign the data regarding MG-SIP A and MG-SIP B respectively as follows:

MG-ID: Enter MAC Address (six bytes) of MG-SIP. * Enter the Telephony Server side (LAN side) MAC address.

FUNCTION = Setting

KIND = SIP-MG

LINE = 0 (fixed)

CH = 0 (fixed)

LENS: Enter the accommodated location of Virtual Signaling Channel (the first number of HW).

Retry = 0

Setting Type of Signaling Packet Service: Specify QoS settings of Signaling that is used when Internal PHI-BRI connects MG-SIP.

Usable CH Number: Enter the number of channels to be used per MG-SIP (1-128).

Note: Be sure to set 128ch (fixed) when this feature (MG-SIP TIE Line Path Splitting) is used.

Service of Authentication Exists: Unchecked

Setting of Path Data

Note: When the Set button of the command is pressed while this check box is OFF, all the registered Path data is cleared.

Note: Assign the following data as you have created with the ATRK command. Assign the counter IP address and port number according to the settings of counter gateway.

List of Path Data

INDEX: Index of Path data (1-32)

Select: Choose the check box for the path to be registered.

Note: When the Set button of the command is pressed while the check box for the already registered path is OFF, the path data is cleared.

Path ID: Identification Number of Path (1-255)

RT: External Route Number (1-255)

* Enter the route number (route number of Virtual Speech Channel for MG-SIP assigned by the ATRK command) to be assigned to this path.
LGRT: Logical Route Number (display only)
TK: Trunk Number (1-255)
* Enter the first trunk number of the trunks (trunk number of Virtual Speech Channel for MG-SIP assigned by the ATRK command) to be assigned to this path.
CH: Enter the number of MG-SIP channels to be assigned to this path.
IP Ver: IP Address Version = 1 (IPv4)
GW IP: Enter an IP address of the counter gateway (0.0.0.1-255.255.255.254).
GW PORT: Enter a port number of the counter gateway (1024-65535).
FLT: Gateway Fault Detection Capability (0/1)
* To enable MG-SIP Fault Detection per Path, set this parameter to 1 (Enable).

<Example of AMGIL Command Settings for **MG-SIP A**>

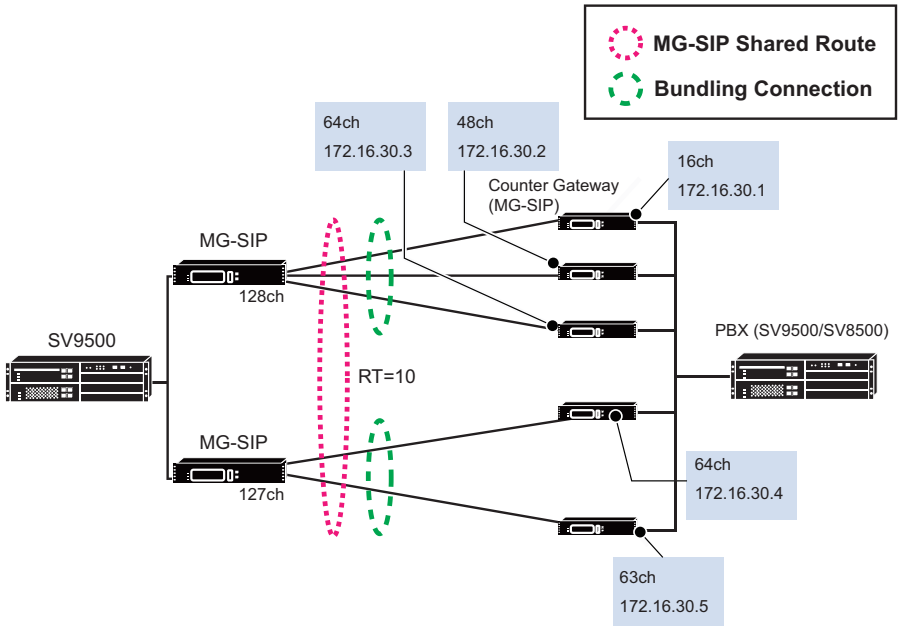
Path of MG-SIP	Select	Path ID	RT	LGRT	TK	CH	IP Ver	GW IP	GW PORT	FLT
Path A	ON	1	10	10	1	96	IPv4	172.16.30.1	5060	Enable
Path B	ON	2	20	20	1	32	IPv4	172.16.40.1	5060	Enable

<Example of AMGIL Command Settings for **MG-SIP B**>

Path of MG-SIP	Select	Path ID	RT	LGRT	TK	CH	IP Ver	GW IP	GW PORT	FLT
Path B	ON	1	20	20	33	32	IPv4	172.16.40.1	5060	Enable
Path C	ON	2	30	30	1	95	IPv4	172.16.50.1	5060	Enable

Combination of MG-SIP Shared Route and Bundling Connection

Office data settings for Combination of MG-SIP Shared Route and Bundling Connection are as follows:



* The following shows only the differences from the office data settings for Route Splitting.

STEP 1: **ATRK**- Assign trunk data on Virtual Speech Channel.
RT: Specify Route Number (1-255).
TK: Specify Trunk Number (1-255).
LENS: Specify Accommodated Location (six digits).

Register routes/trunks/channels to use for each path.

	Path for MG-SIP	Number of Channels	Route Number	LENS			
				Highway 1	Highway 2	Highway 3	Highway 4
MG-SIP A	Path A	16	10	020040-020057	-	-	-
	Path B	48	10	020060-020077	020080-020117	-	-
	Path C	64	10	-	-	020120-020157	020160-020197
MG-SIP B	Path D	64	10	021000-021037	021040-021077	-	-
	Path E	63	10	-	-	021080-021117	021120-021156

Note: When using this command, consider the following:

- Virtual speech channels for each path must be assigned to continuous channels from the first channel of a highway group. When all the channels cannot fit within one highway and need to be split across multiple highway groups, assign remaining channels from the beginning of the next group.
- Only MG-SIP virtual communication channel can be registered for HW (4 group) with at least one MG-SIP virtual communication channel already registered.
- Do not assign the LENS that belongs to Even-numbered module group, Unit=0 and Group=0 to a trunk, because they are not supported by MG-SIP.

STEP 2: **ACSC** - Assign the first channel of 32 channels (a highway) LENS to the even-numbered CSC group, and LENS of Virtual Signaling Channel to the odd-numbered CSC group.

CSCG: Specify the CSC Group Number (even numbers from 130 to 255 and odd number continuous with them)

GROUP: Specify the CIC Group Number

CCH: Specify the first channel of 32-channel LENS for even-numbered CSCG and Virtual Signaling LENS for odd-numbered CSCG.

<Example of Data Settings>

Even-numbered CSCG for MG-SIP: the first LENS of 32-channels

The top of CICG for MG-SIP: LENS of Virtual Signaling Channel

CSC Group Number (CSCG)	CIC Group Number (CICG)	LENS of Signaling Channel (CCH)			Remarks
		MG 00-07	U 0-3	G 00-23	
130	0	02	0	4	Path A/B
	1	02	0	8	Path B
	2	02	0	12	Path C
	3	02	0	16	
	4	02	1	0	Path D
	5	02	1	4	
	6	02	1	8	Path E
	7	02	1	12	

In the next step, assign LENS of Virtual Signaling Channel to the next CSCG (greater by one than the last CSCG).

CSC Group Number (CSCG)	CIC Group Number (CICG)	LENS of Signaling Channel (CCH)			Remarks
		MG 00-07	U 0-3	G 00-23	
131	0	02	0	4	MG-SIP A
	1	02	0	4	
	2	02	0	4	
	3	02	0	4	
	4	02	1	0	MG-SIP B
	5	02	1	0	
	6	02	1	0	
	7	02	1	0	

STEP 3: **AMGIL** - Assign the data regarding MG-SIP A and MG-SIP B respectively as follows:

MG-ID: Enter MAC Address (six bytes) of MG-SIP. * Enter the Telephony Server side (LAN side) MAC address.

FUNCTION = Setting

KIND = SIP-MG

LINE = 0 (fixed)

CH = 0 (fixed)

LENS: Enter the accommodated location of Virtual Signaling Channel (the first number of HW).

Retry = 0

Setting Type of Signaling Packet Service: Specify QoS settings of Signaling that is used when Internal PHI-BRI connects MG-SIP.

Usable CH Number: Enter the number of channels to be used per MG-SIP (1-128).

Note: Be sure to set 128ch (fixed) when this feature (MG-SIP TIE Line Path Splitting) is used.

Service of Authentication Exists: Unchecked

Setting of Path Data

Note: When the Set button of the command is pressed while this check box is OFF, all the registered Path data is cleared.

Note: Assign the following data as you have created with the ATRK command. Assign the counter IP address and port number according to the settings of counter gateway.

List of Path Data

INDEX: Index of Path data (1-32)

Select: Choose the check box for the path to be registered.

Note: When the Set button of the command is pressed while the check box for the already registered path is OFF, the path data is cleared.

Path ID: Identification Number of Path (1-255)

RT: External Route Number (1-255)

* Enter the route number (route number of Virtual Speech Channel for MG-SIP assigned by the ATRK command) to be assigned to this path.
LGRT: Logical Route Number (display only)
TK: Trunk Number (1-255)
* Enter the first trunk number of the trunks (trunk number of Virtual Speech Channel for MG-SIP assigned by the ATRK command) to be assigned to this path.
CH: Enter the number of MG-SIP channels to be assigned to this path.
IP Ver: IP Address Version = 1 (IPv4)
GW IP: Enter an IP address of the counter gateway (0.0.0.1-255.255.255.254).
GW PORT: Enter a port number of the counter gateway (1024-65535).
FLT: Gateway Fault Detection Capability (0/1)
* To enable MG-SIP Fault Detection per Path, set this parameter to 1 (Enable).

<Example of AMGIL Command Settings for **MG-SIP A**>

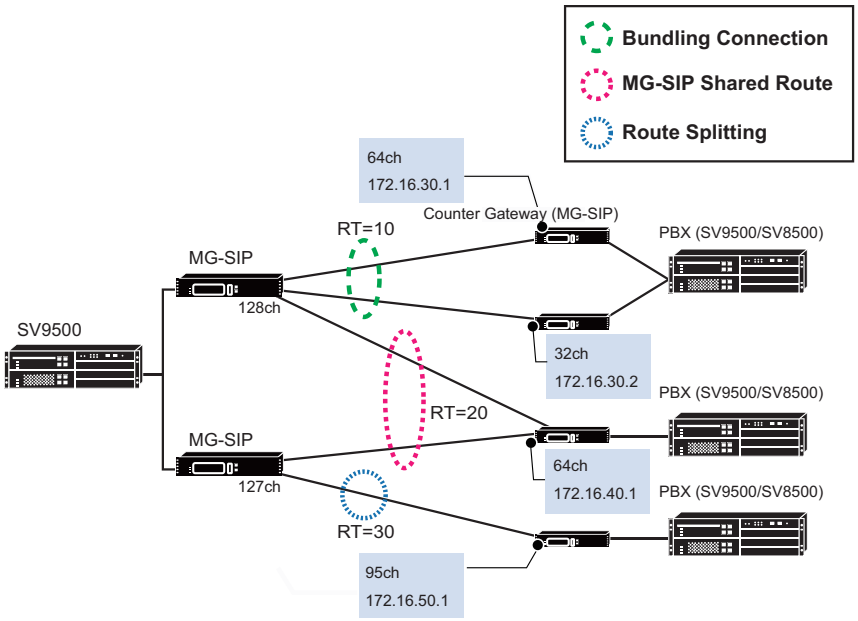
Path of MG-SIP	Select	Path ID	RT	LGRT	TK	CH	IP Ver	GW IP	GW PORT	FLT
Path A	ON	1	10	10	1	16	IPv4	172.16.30.1	5060	Enable
Path B	ON	2	10	10	17	48	IPv4	172.16.30.2	5060	Enable
Path C	ON	3	10	10	65	64	IPv4	172.16.30.3	5060	Enable

<Example of AMGIL Command Settings for **MG-SIP B**>

Path of MG-SIP	Select	Path ID	RT	LGRT	TK	CH	IP Ver	GW IP	GW PORT	FLT
Path D	ON	1	10	10	129	64	IPv4	172.16.30.4	5060	Enable
Path E	ON	2	10	10	193	63	IPv4	172.16.30.5	5060	Enable

Combination of MG-SIP Shared Route, Route Splitting and Bundling Connection

Office data settings for Combination of MG-SIP Shared Route, Route Splitting and Bundling Connection are as follows:



* The following shows only the differences from the office data settings for Route Splitting.

STEP 1: **ATRK**- Assign trunk data on Virtual Speech Channel.
RT: Specify Route Number (1-255).
TK: Specify Trunk Number (1-255).
LENS: Specify Accommodated Location (six digits).

Register routes/trunks/channels to use for each path.

	Path for MG-SIP	Number of Channels	Route Number	LENS			
				Highway 1	Highway 2	Highway 3	Highway 4
MG-SIP A	Path A	64	10	020040-020077	020080-020117	-	-
	Path B	32	10	-	-	020120-020157	-
	Path C	32	20	-	-	-	020160-020197
MG-SIP B	Path D	32	20	021000-021037	-	-	-
	Path E	95	30	-	021040-021077	021080-021117	021120-021156

Note: When using this command, consider the following:

- Virtual speech channels for each path must be assigned to continuous channels from the first channel of a highway group. When all the channels cannot fit within one highway and need to be split across multiple highway groups, assign remaining channels from the beginning of the next group.
- Only MG-SIP virtual communication channel can be registered for HW (4 group) with at least one MG-SIP virtual communication channel already registered.
- Do not assign the LENS that belongs to Even-numbered module group, Unit=0 and Group=0 to a trunk, because they are not supported by MG-SIP.

STEP 2: **ACSC** - Assign the first channel of 32 channels (a highway) LENS to the even-numbered CSC group, and LENS of Virtual Signaling Channel to the odd-numbered CSC group.

CSCG: Specify the CSC Group Number (even numbers from 130 to 255 and odd number continuous with them)

GROUP: Specify the CIC Group Number

CCH: Specify the first channel of 32-channel LENS for even-numbered CSCG and Virtual Signaling LENS for odd-numbered CSCG.

<Example of Data Settings>

Even-numbered CSCG for MG-SIP: the first LENS of 32-channels

The top of CICG for MG-SIP: LENS of Virtual Signaling Channel

CSC Group Number (CSCG)	CIC Group Number (CICG)	LENS of Signaling Channel (CCH)			Remarks
		MG 00-07	U 0-3	G 00-23	
130	0	02	0	4	Path A
	1	02	0	8	
	2	02	0	12	Path B
	3	02	0	16	Path C
	4	02	1	0	Path D
	5	02	1	4	Path E
	6	02	1	8	
	7	02	1	12	

In the next step, assign LENS of Virtual Signaling Channel to the next CSCG (greater by one than the last CSCG).

CSC Group Number (CSCG)	CIC Group Number (CICG)	LENS of Signaling Channel (CCH)			Remarks
		MG 00-07	U 0-3	G 00-23	
131	0	02	0	4	MG-SIP A
	1	02	0	4	
	2	02	0	4	
	3	02	0	4	
	4	02	1	0	MG-SIP B
	5	02	1	0	
	6	02	1	0	
	7	02	1	0	

STEP 3: **AMGIL** - Assign the data regarding MG-SIP A and MG-SIP B respectively as follows:

MG-ID: Enter MAC Address (six bytes) of MG-SIP. * Enter the Telephony Server side (LAN side) MAC address.

FUNCTION = Setting

KIND = SIP-MG

LINE = 0 (fixed)

CH = 0 (fixed)

LENS: Enter the accommodated location of Virtual Signaling Channel (the first number of HW).

Retry = 0

Setting Type of Signaling Packet Service: Specify QoS settings of Signaling that is used when Internal PHI-BRI connects MG-SIP.

Usable CH Number: Enter the number of channels to be used per MG-SIP (1-128).

Note: Be sure to set 128ch (fixed) when this feature (MG-SIP TIE Line Path Splitting) is used.

Service of Authentication Exists: Unchecked

Setting of Path Data

Note: When the Set button of the command is pressed while this check box is OFF, all the registered Path data is cleared.

Note: Assign the following data as you have created with the ATRK command. Assign the counter IP address and port number according to the settings of counter gateway.

List of Path Data

INDEX: Index of Path data (1-32)

Select: Choose the check box for the path to be registered.

Note: When the Set button of the command is pressed while the check box for the already registered path is OFF, the path data is cleared.

Path ID: Identification Number of Path (1-255)

RT: External Route Number (1-255)

* Enter the route number (route number of Virtual Speech Channel for MG-SIP assigned by the ATRK command) to be assigned to this path.
LGRT: Logical Route Number (display only)
TK: Trunk Number (1-255)
* Enter the first trunk number of the trunks (trunk number of Virtual Speech Channel for MG-SIP assigned by the ATRK command) to be assigned to this path.
CH: Enter the number of MG-SIP channels to be assigned to this path.
IP Ver: IP Address Version = 1 (IPv4)
GW IP: Enter an IP address of the counter gateway (0.0.0.1-255.255.255.254).
GW PORT: Enter a port number of the counter gateway (1024-65535).
FLT: Gateway Fault Detection Capability (0/1)
* To enable MG-SIP Fault Detection per Path, set this parameter to 1 (Enable).

<Example of AMGIL Command Settings for **MG-SIP A**>

Path of MG-SIP	Select	Path ID	RT	LGRT	TK	CH	IP Ver	GW IP	GW PORT	FLT
Path A	ON	1	10	10	1	64	IPv4	172.16.30.1	5060	Enable
Path B	ON	2	10	10	65	32	IPv4	172.16.30.2	5060	Enable
Path C	ON	3	20	20	1	32	IPv4	172.16.40.1	5060	Enable

<Example of AMGIL Command Settings for **MG-SIP B**>

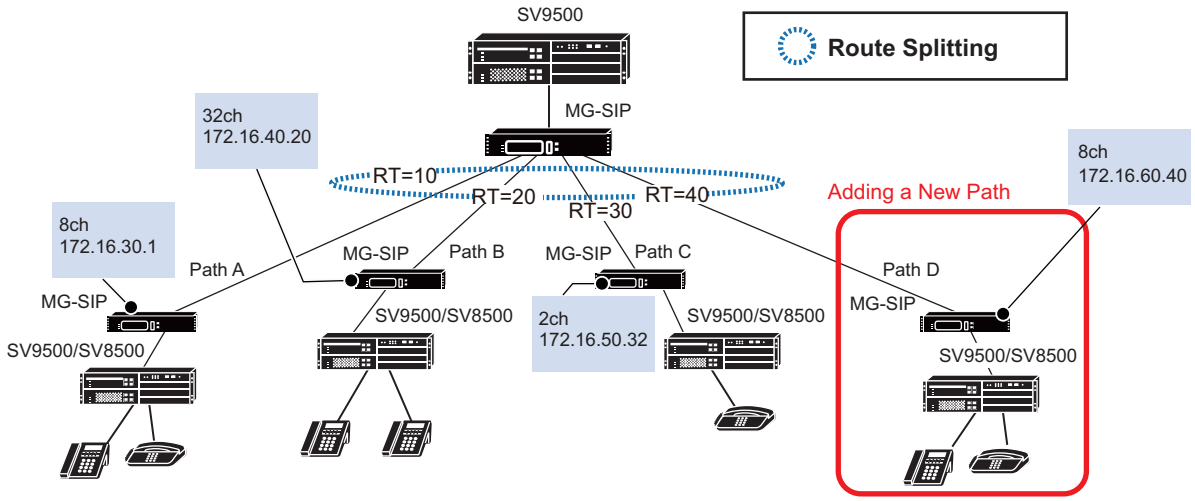
Path of MG-SIP	Select	Path ID	RT	LGRT	TK	CH	IP Ver	GW IP	GW PORT	FLT
Path D	ON	1	20	20	33	32	IPv4	172.16.40.1	5060	Enable
Path E	ON	2	30	30	1	95	IPv4	172.16.50.1	5060	Enable

Example of Data Settings for MG-SIP TIE Line Path Splitting

The following shows some examples of the office data settings when registering or deleting path information during an operation.

Example 1: Adding New Path and Trunk to Registered Highway

- In the case that a reset of MG-SIP is not required:



STEP 1: **ARTD/ARTDN**- Assign a new route (RT=40) for Path D.

STEP 2: **ARTI/ARTIN**- Assign a new route (RT=40) for Path D.

STEP 3: **ATRK**- Assign a new trunk for Path D.

Path for MG-SIP	Route Number	Trunk Number	Number of Channels	LENS			
				Highway 1	Highway 2	Highway 3	Highway 4
Path A	10	1-8	8	020040-020047	-	-	-
Path B	20	1-32	32	020050-020077	020160-020167	-	-
Path C	30	1-2	2	-	020170-020171	-	-
Path D	40	1-8	8	-	020180-020187	-	-

STEP 4: **ACSC** - The setting of this command data is not required to be changed because the trunk of Path D can be accommodated within the LENS of CIC Group 1.

CSC Group Number (CSCG)	CIC Group Number (CICG)	LENS of Signaling Channel (CCH)			Remarks
		MG 00-07	U 0-3	G 00-23	
130	0	02	0	4	used for Path A/B
	1	02	0	16	used for Path B/C -> used for Path B/C/D
	2	02	0	4	Note 7
	3	02	0	4	
	4	02	0	4	
	5	02	0	4	
	6	02	0	4	
	7	02	0	4	

Note 7: Apply the same settings as CICG=0 to the remaining CIC Group Numbers (CICG).

STEP 5: **ADPC**- Assign the same PC as RT=10/20/30 for RT=40.

STEP 6: **MBTK**- Cancel Make-busy for RT=40/TK=1-8.

STEP 7: **AMGIL**- Assign path information data of Path D.

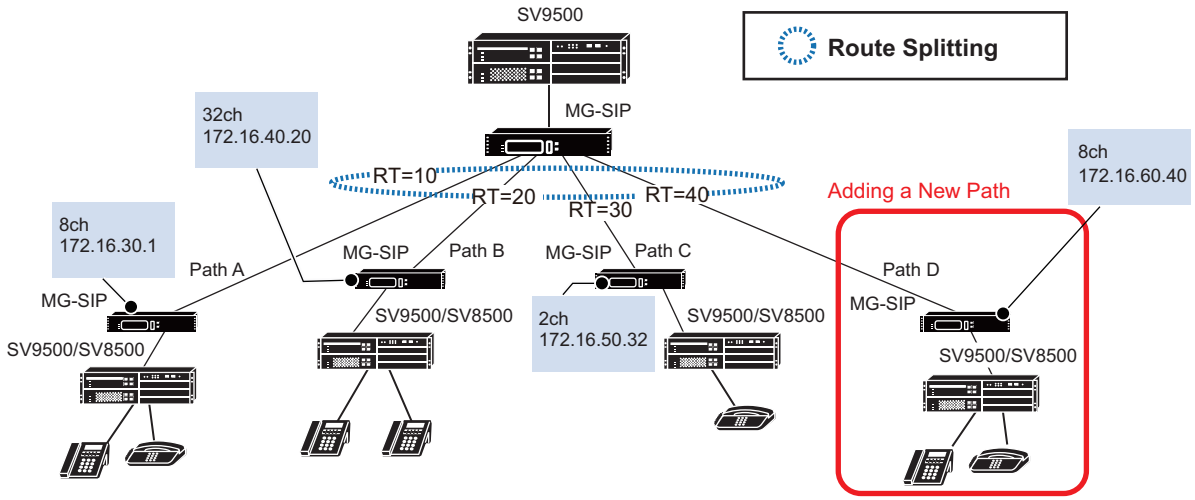
Path of MG-SIP	Select	Path ID	RT	LGRT	TK	CH	IP Ver	GW IP	GW PORT	FLT
Path A	ON	1	10	10	1	8	IPv4	172.16.30.1	5060	Enable
Path B	ON	2	20	20	1	32	IPv4	172.16.40.20	5060	Enable
Path C	ON	3	30	30	1	2	IPv4	172.16.50.32	5060	Enable
Path D	ON	4	40	40	1	8	IPv4	172.16.60.40	5060	Enable

Note: The reset of MG-SIP is not required.

STEP 8: **AMGIL**- Reset Path D.
MG-ID: MAC address of MG-SIP (six bytes)
* Enter MAC address of Telephony Server side (LAN side).
FUNCTION: Reset
Reset Type: PATH (Reset Automatically Select)
The check box for Path D is automatically selected.

Example 2: Adding New Path and Trunk to Unregistered Highway

- In the case that a reset of MG-SIP is required:



STEP 1: **ARTD/ARTDN**- Assign a new route (RT = 40) for Path D.

STEP 2: **ARTI/ARTIN**- Assign a new route (RT = 40) for Path D.

STEP 3: **ATRK**- Assign a new trunk for Path D.

Path for MG-SIP	Route Number	Trunk Number	Number of Channels	LENS			
				Highway 1	Highway 2	Highway 3	Highway 4
Path A	10	1-8	8	020040-020047	-	-	-
Path B	20	1-32	32	020050-020077	020160-020167	-	-
Path C	30	1-2	2	-	020170-020171	-	-
Path D	40	1-8	8	-	-	020080-020088	-

STEP 4: **MBTK**- For each trunk of RT = 10, RT = 20 and RT = 30, select "2: Make Busy (Bothway)" from the MB drop-down list box.

Note: Make sure that all the channels are placed in the idle state, then go to the next step.

STEP 5: **AMGIL**- Delete the registration of MG-SIP.

Note: When the registration of MG-SIP is deleted, the associated path information will also be deleted.

Note: After the registration of MG-SIP is deleted, MG-SIP will be reset.

STEP 6: **ACSC** - The setting of this command data is not required to be changed because the trunk of Path D can be accommodated within the LENS of CIC Group 2.

<Before the change>

CSC Group Number (CSCG)	CIC Group Number (CICG)	LENS of Signaling Channel (CCH)			Remarks
		MG 00-07	U 0-3	G 00-23	
130	0	02	0	4	used for Path A/B
	1	02	0	16	used for Path B/C
	2	02	0	4	Note 8
	3	02	0	4	
	4	02	0	4	
	5	02	0	4	
	6	02	0	4	
	7	02	0	4	

<After the change>

CSC Group Number (CSCG)	CIC Group Number (CICG)	LENS of Signaling Channel (CCH)			Remarks
		MG 00-07	U 0-3	G 00-23	
130	0	02	0	4	used for Path A/B
	1	02	0	16	used for Path B/C
	2	02	0	8	used for Path D
	3	02	0	4	Note 8
	4	02	0	4	
	5	02	0	4	
	6	02	0	4	
	7	02	0	4	

Note 8: Apply the same settings as CICG=0 to the remaining CIC Group Numbers (CICG).

STEP 7: **MBTK**- Cancel Make-busy for all the trunks of RT=10/20/30.

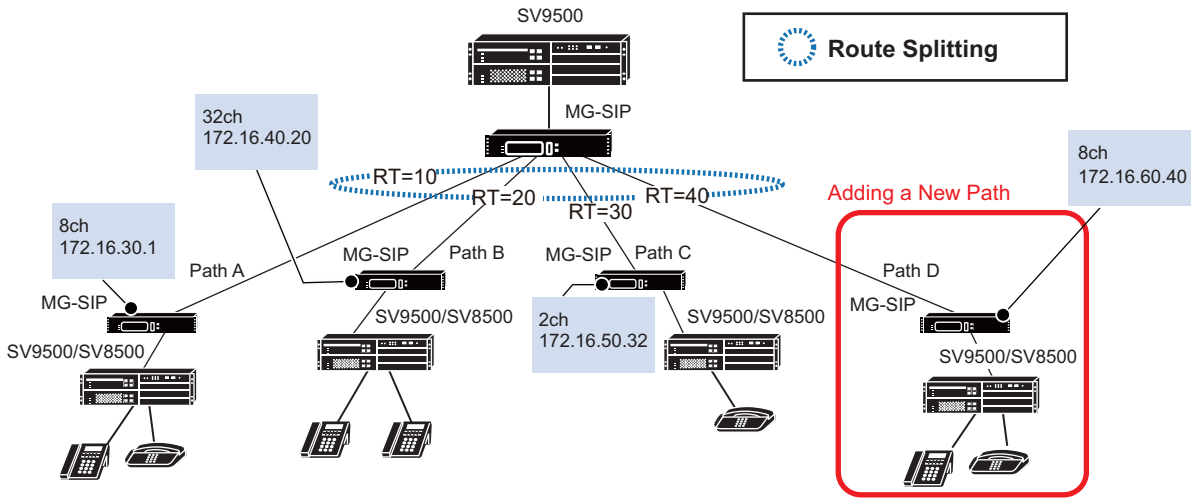
STEP 8: **ADPC**- Assign the same PC as RT=10/20/30 for RT=40.

STEP 9: **MBTK**- Cancel Make-busy for RT=40/TK=1-8.

STEP 10: **AMGIL**- Enter MAC address of MG-SIP and then enter LENS of MG-SIP Virtual Signaling Channel (LENS=020000).
Assign path data of Path D in addition to Path A, Path B and Path C (the same data as it was before the deletion).

Path of MG-SIP	Select	Path ID	RT	LGRT	TK	CH	IP Ver	GW IP	GW PORT	FLT
Path A	ON	1	10	10	1	8	IPv4	172.16.30.1	5060	Enable
Path B	ON	2	20	20	1	32	IPv4	172.16.40.20	5060	Enable
Path C	ON	3	30	30	1	2	IPv4	172.16.50.32	5060	Enable
Path D	ON	4	40	40	1	8	IPv4	172.16.60.40	5060	Enable

Example 3: When Avoiding Reset of MG-SIP in Example 2



<Preconditions>

- LENS needs to be assigned through the ACSC command.
- When a new path or trunk is added to a new highway, the data changing of the ACSC command can be omitted by preliminarily assigning LENS.

CSC Group Number (CSCG)	CIC Group Number (CICG)	LENS of Signaling Channel (CCH)			Remarks
		MG 00-07	U 0-3	G 00-23	
130	0	02	0	4	used for Path A/B
	1	02	0	16	used for Path B/C
	2	02	0	8	preliminarily assigned **Note 9**
	3	02	0	12	preliminarily assigned **Note 9**
	4	02	0	4	**Note 10**
	5	02	0	4	**Note 10**
	6	02	0	4	**Note 10**
	7	02	0	4	**Note 10**

Note 9: Assign only the virtual channels of the appropriate MG-SIP to the preliminarily assigned highway (no terminal other than the MG-SIP can be assigned to the highway).

Note 10: Apply the same settings as CICG=0 to the remaining CIC Group Numbers (CICG).

STEP 1: **ARTD/ARTDN**- Assign a new route (RT=40) for Path D.

STEP 2: **ARTI/ARTIN**- Assign a new route (RT=40) for Path D.

STEP 3: **ATRK**- Assign a new trunk for Path D.

Path for MG-SIP	Route Number	Trunk Number	Number of Channels	LENS			
				Highway 1	Highway 2	Highway 3	Highway 4
Path A	10	1-8	8	020040-020047	-	-	-
Path B	20	1-32	32	020050-020077	020160-020167	-	-
Path C	30	1-2	2	-	020170-020171	-	-
Path D	40	1-8	8	-	-	020080-020087	-

STEP 4: **ACSC**- The setting of this command data is not required to be changed because the trunk of Path D can be accommodated within the LENS of CIC Group 2.

CSC Group Number (CSCG)	CIC Group Number (CICG)	LENS of Signaling Channel (CCH)			Remarks
		MG 00-07	U 0-3	G 00-23	
130	0	02	0	4	used for Path A/B
	1	02	0	16	used for Path B/C
	2	02	0	8	preliminarily assigned-> used for Path D
	3	02	0	12	preliminarily assigned
	4	02	0	4	**Note 11**
	5	02	0	4	**Note 11**
	6	02	0	4	**Note 11**
	7	02	0	4	**Note 11**

Note 11: Apply the same settings as CICG=0 to the remaining CIC Group Numbers (CICG).

STEP 5: **ADPC**- Assign the same PC as RT=10/20/30 for RT=40.

STEP 6: **MBTK**- Cancel Make-busy for RT=40/TK=1-8.

STEP 7: **AMGIL**- Assign path information data of Path D.

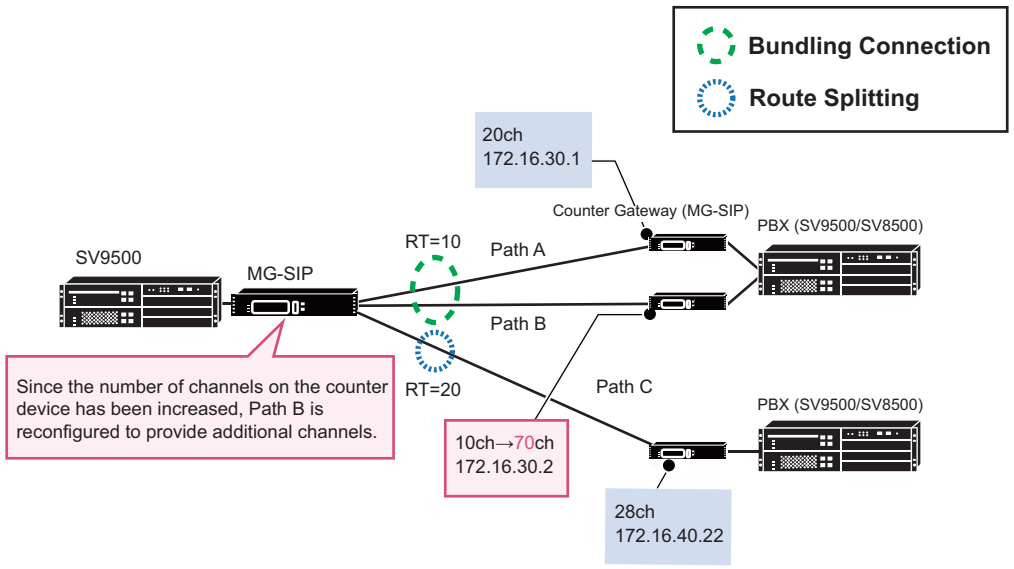
Path of MG-SIP	Select	Path ID	RT	LGRT	TK	CH	IP Ver	GW IP	GW PORT	FLT
Path A	ON	1	10	10	1	8	IPv4	172.16.30.1	5060	Enable
Path B	ON	2	20	20	1	32	IPv4	172.16.40.20	5060	Enable
Path C	ON	3	30	30	1	2	IPv4	172.16.50.32	5060	Enable
Path D	ON	4	40	40	1	8	IPv4	172.16.60.40	5060	Enable

Note: The reset of MG-SIP is not required.

STEP 8: **AMGIL**- Reset Path D.
MG-ID: MAC address of MG-SIP (six bytes)
* Enter MAC address of Telephony Server side (LAN side).
FUNCTION: Reset
Reset Type: PATH (Reset Automatically Select)
The check box for Path D is automatically selected.

Example 4: Changing the Number of Channels for a Path

- In the case that a reset of MG-SIP is required:



STEP 1: **MBTK**- For each trunk of RT = 10 and RT = 20, select "2: Make Busy (Bothway)" from the MB drop-down list box.

Note: Make sure that all the channels are placed in the idle state, then go to the next step.

STEP 2: **AMGIL**- Delete the registration of MG-SIP.

Note: When the registration of MG-SIP is deleted, the associated path information will also be deleted.

Note: After the registration of MG-SIP is deleted, MG-SIP will be reset.

STEP 3: **ATRK**

Delete the trunk for Path B (RT=10, TK=21-30).

Path for MG-SIP	Route Number	Trunk Number	Number of Channels	LENS			
				Highway 1	Highway 2	Highway 3	Highway 4
Path A	10	1-20	20	020040-020063	-	-	-
Path B	10	21-30	10	020064-020075	-	-	-
Path C	20	1-28	28	020076-020077	020080-020111	-	-

Note: Since a deleted speech channel changes to an unused state, the channel can be used as the new path if the channel is within a range of the appropriate speech channels. For more details, see Example 8: Adding New Path and Trunk to Speech Channel deleted in Examples 4 and 5.

Assign a new trunk for Path B (RT=10, TK=31-100).

Path for MG-SIP	Route Number	Trunk Number	Number of Channels	LENS			
				Highway 1	Highway 2	Highway 3	Highway 4
Path A	10	1-20	20	020040-020063	-	-	-
Path C	20	1-28	28	020076-020077	020080-020111	-	-
Path B	10	31-100	70	-	020112-020117	020120-020157	020160-020197

STEP 4: **ACSC** - Delete the already registered CSC Group information and assign LENS of the highway used for the trunk of Path D to unused CIC Groups 2 and 3.

<Before the change>

CSC Group Number (CSCG)	CIC Group Number (CICG)	LENS of Signaling Channel (CCH)			Remarks
		MG 00-07	U 0-3	G 00-23	
130	0	02	0	4	used for Path A/B/C
	1	02	0	8	used for Path C
	2	02	0	4	Note 12
	3	02	0	4	
	4	02	0	4	
	5	02	0	4	
	6	02	0	4	
	7	02	0	4	

<After the change>

CSC Group Number (CSCG)	CIC Group Number (CICG)	LENS of Signaling Channel (CCH)			Remarks
		MG 00-07	U 0-3	G 00-23	
130	0	02	0	4	used for Path A/B/C -> used for Path A/C
	1	02	0	8	used for Path C -> used for Path B/C
	2	02	0	12	used for Path B
	3	02	0	16	used for Path B
	4	02	0	4	Note 12
	5	02	0	4	
	6	02	0	4	
	7	02	0	4	

Note 12: Apply the same settings as CICG=0 to the remaining CIC Group Numbers (CICG).

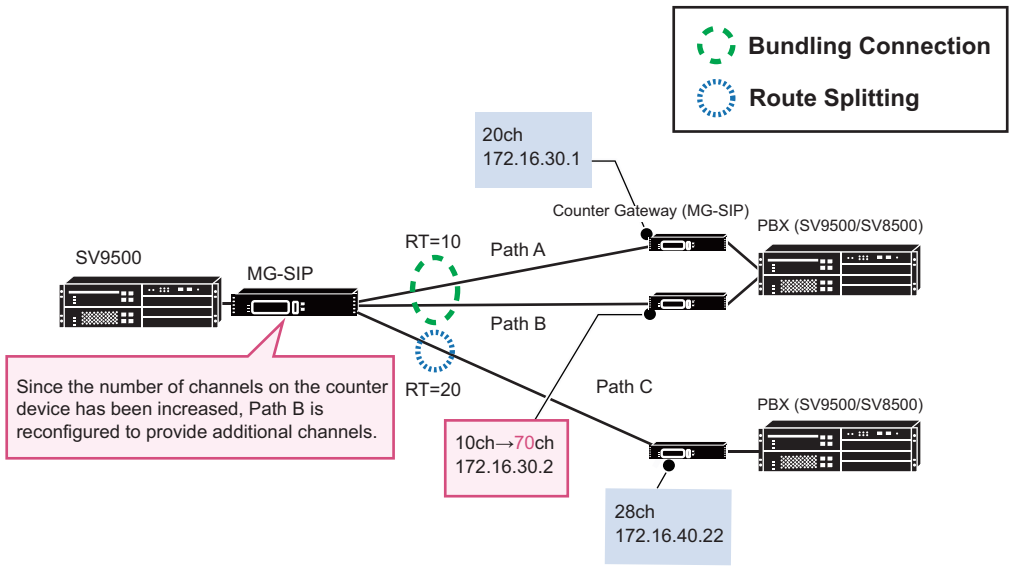
STEP 5: **MBTK**- Cancel Make-busy for all the trunks of RT = 10 and RT = 20.

STEP 6: **AMGIL**- Enter MAC address of MG-SIP and then enter LENS of MG-SIP Virtual Signaling Channel (LENS=020040).
Assign path data of Path B in addition to Path A and Path C (the same data as it was before the deletion).

Path of MG-SIP	Select	Path ID	RT	LGRT	TK	CH	IP Ver	GW IP	GW PORT	FLT
Path A	ON	1	10	10	1	20	IPv4	172.16.30.1	5060	Enable
Path B	ON	2	10	10	31	70	IPv4	172.16.30.2	5060	Enable
Path C	ON	3	20	20	1	28	IPv4	172.16.40.22	5060	Enable

Example 5: Changing the Number of Channels for a Path, Avoiding Reset of MG-SIP

- In the case that a reset of MG-SIP is not required:



<Preconditions>

- LENS needs to be assigned through the ACSC command.
- When adding a new path or trunk to a new highway, the data changing of the ACSC command can be omitted by preliminarily assigning LENS.

CSC Group Number (CSCG)	CIC Group Number (CICG)	LENS of Signaling Channel (CCH)			Remarks
		MG 00-07	U 0-3	G 00-23	
130	0	02	0	4	used for Path A/B/C
	1	02	0	8	used for Path C
	2	02	0	12	preliminarily assigned **Note 13**
	3	02	0	16	preliminarily assigned **Note 13**
	4	02	0	4	**Note 14**
	5	02	0	4	**Note 14**
	6	02	0	4	**Note 14**
	7	02	0	4	**Note 14**

Note 13: Assign only the virtual channels of the appropriate MG-SIP to the preliminarily assigned highway (no other than the MG-SIP can be assigned to the highway).

Note 14: Apply the same settings as CICG=0 to the remaining CIC Group Numbers (CICG).

STEP 1: **MBTK**- For each trunk of RT = 10, select "2: Make Busy (Bothway)" from the MB drop-down list box.

Note: Make sure that all the channels are placed in the idle state, then go to the next step.

STEP 2: **AMGIL**- Delete the path information of Path B.

Delete the path information of Path B.

Clear the Select check box of Path B to be deleted. Path B data will be deleted by clicking the SET button while the Select check boxes of the other paths (Path A and Path C) are ON.

Path of MG-SIP	Select	Path ID	RT	TK	CH	GW IP	GW PORT
Path A	ON	1	10	1	20	172.16.30.1	5060
Path B	OFF	2	10	21	10	172.16.30.2	5060
Path C	ON	3	20	1	28	172.16.40.22	5060

STEP 3: **ATRK**

Delete the trunk for Path B (RT=10, TK=21-30).

Path for MG-SIP	Route Number	Trunk Number	Number of Channels	LENS			
				Highway 1	Highway 2	Highway 3	Highway 4
Path A	10	1-20	20	020040-020063	-	-	-
Path B	10	21-30	10	020064-020075	-	-	-
Path C	20	1-28	28	020076-020077	020080-020111	-	-

Note: Since a deleted speech channel changes to a unused state, the channel can be used as the new path if the channel is within a range of the appropriate speech channels. For more details, see Example 8: Adding New Path and Trunk to Speech Channel deleted in Examples 4 and 5.

Assign a new trunk for Path B (RT=10, TK=31-100).

Path for MG-SIP	Route Number	Trunk Number	Number of Channels	LENS			
				Highway 1	Highway 2	Highway 3	Highway 4
Path A	10	1-20	20	020040-020063	-	-	-
Path C	20	1-28	28	020076-020077	020080-020111	-	-
Path B	10	31-100	70	-	020112-020117	020120-020157	020160-020197

STEP 4: **ACSC**- The setting of this command data is not required to be changed because the trunk of Path B can be accommodated within the LENS of CIC Groups 1, 2 and 3.

CSC Group Number (CSCG)	CIC Group Number (CICG)	LENS of Signaling Channel (CCH)			Remarks
		MG 00-07	U 0-3	G 00-23	
130	0	02	0	4	used for Path A/B/C -> used for Path A/C
	1	02	0	8	used for Path C -> used for Path B/C
	2	02	0	12	preliminarily assigned -> used for Path B
	3	02	0	16	preliminarily assigned -> used for Path B
	4	02	0	4	**Note 15**
	5	02	0	4	**Note 15**
	6	02	0	4	**Note 15**
	7	02	0	4	**Note 15**

Note 15: Apply the same settings as CICG=0 to the remaining CIC Group Numbers (CICG).

STEP 5: **MBTK**- Cancel Make-busy for all the trunks of RT = 10.

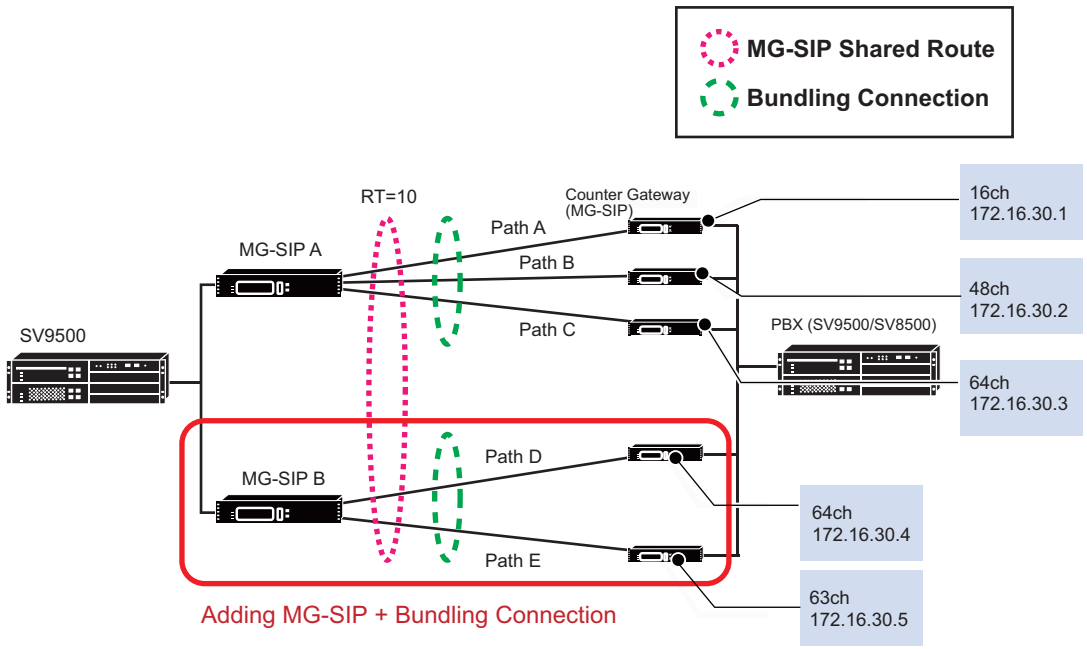
STEP 6: **AMGIL**- Assign the path information for Path B.

Path of MG-SIP	Select	Path ID	RT	LGRT	TK	CH	IP Ver	GW IP	GW PORT	FLT
Path A	ON	1	10	10	1	20	IPv4	172.16.30.1	5060	Enable
Path C	ON	3	20	20	1	28	IPv4	172.16.40.22	5060	Enable
Path B	ON	2	10	10	31	70	IPv4	172.16.30.2	5060	Enable

Note: The reset of MG-SIP is not required.

STEP 7: **AMGIL**- Reset Path B.
MG-ID: MAC address of MG-SIP (six bytes)
* Enter MAC address of Telephony Server side (LAN side).
FUNCTION: Reset
Reset Type: PATH (Reset Automatically Select)
The check box for Path B is automatically selected.

Example 6: Adding an MG-SIP



STEP 1: **ATRK**

Assign the trunk for Path D and Path E (RT=10).

MG-SIP	Path for MG-SIP	Number of Channels	LENS			
			Highway 1	Highway 2	Highway 3	Highway 4
MG-SIP A	Path A	16	020040 - 020057	-	-	-
	Path B	48	020060- 020077	020080- 020117	-	-
	Path C	64	-	-	020120 - 020157	020160 - 020197
MG-SIP B	Path D	64	021000 - 021037	021040 - 021077	-	-
	Path E	63	-	-	021080 - 021117	021120 - 021156

STEP 2: **MBTK**- For each trunk of RT = 10, select "2: Make Busy (Bothway)" from the MB drop-down list box.

Note: Make sure that all the channels are placed in the idle state, then go to the next step.

STEP 3: **AMGIL**- Delete the registration data of MG-SIP A.

Note: When the registration of MG-SIP is deleted, the associated path information will also be deleted.

Note: After the registration of MG-SIP is deleted, MG-SIP will be reset.

STEP 4: **ACSC** - Delete the already registered CSC Group information and assign LENS of the highway used for the trunk of Path D and Path E to unused CIC Groups 4 to 7.

<Before the change>

CSC Group Number (CSCG)	CIC Group Number (CICG)	LENS of Signaling Channel (CCH)			Remarks
		MG 00-07	U 0-3	G 00-23	
130	0	02	0	4	MG-SIP A
	1	02	0	8	
	2	02	0	12	
	3	02	0	16	
	4	02	0	4	Note 16
	5	02	0	4	Note 16
	6	02	0	4	Note 16
	7	02	0	4	Note 16

<After the change>

CSC Group Number (CSCG)	CIC Group Number (CICG)	LENS of Signaling Channel (CCH)			Remarks
		MG 00-07	U 0-3	G 00-23	
130	0	02	0	4	MG-SIP A
	1	02	0	8	
	2	02	0	12	
	3	02	0	16	
	4	02	1	0	MG-SIP B
	5	02	1	4	
	6	02	1	8	
	7	02	1	12	

In the next step, delete the next CSCG (greater by one than the last CSCG).
Then assign LENS of MG-SIP B Virtual Signaling Channel to the same CIC Group Number as that changed in the even-numbered CSC Group (CSCG).

<Before the change>

CSC Group Number (CSCG)	CIC Group Number (CICG)	LENS of Signaling Channel (CCH)			Remarks
		MG 00-07	U 0-3	G 00-23	
131	0	02	0	4	MG-SIP A
	1	02	0	4	
	2	02	0	4	
	3	02	0	4	
	4	02	0	4	Note 16
	5	02	0	4	
	6	02	0	4	
	7	02	0	4	

<After the change>

CSC Group Number (CSCG)	CIC Group Number (CICG)	LENS of Signaling Channel (CCH)			Remarks
		MG 00-07	U 0-3	G 00-23	
131	0	02	0	4	MG-SIP A
	1	02	0	4	
	2	02	0	4	
	3	02	0	4	
	4	02	1	0	MG-SIP B
	5	02	1	0	
	6	02	1	0	
	7	02	1	0	

Note 16: Apply the same settings as CICG=0 to the remaining CIC Group Numbers (CICG).

STEP 5: **MBTK**- Cancel Make-busy for all the trunks of RT = 10.

STEP 6: **AMGIL**

Enter MAC address of MG-SIP A and then enter LENS of MG-SIP Virtual Signaling Channel (LENS=020040).

Assign path data of Path A/B/C for MG-SIP A (the same data as it was before the deletion).

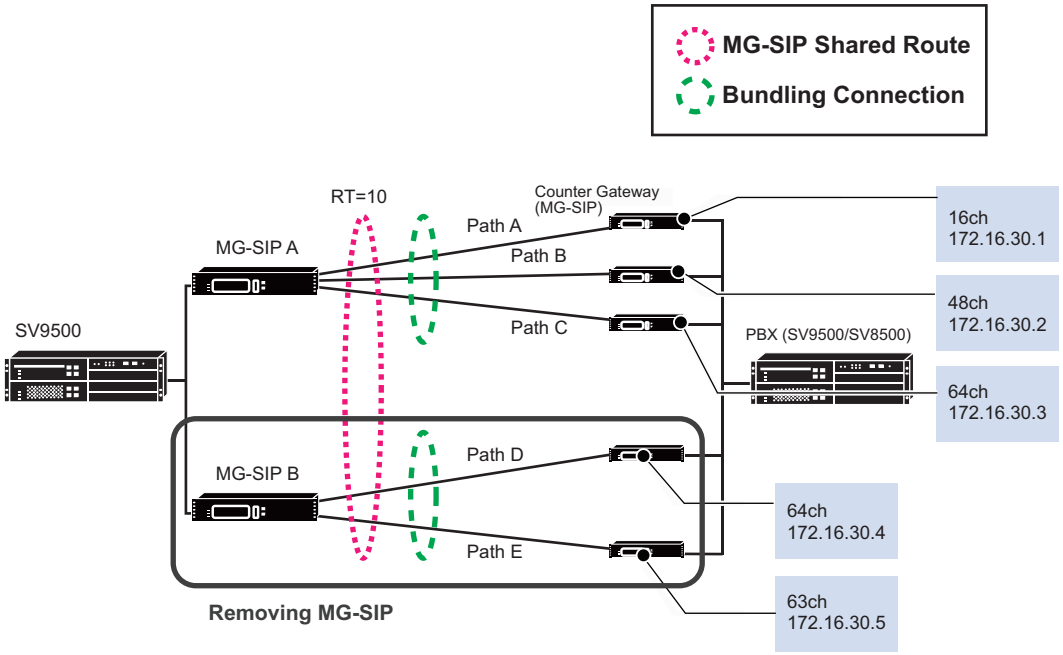
Path of MG-SIP	Select	Path ID	RT	LGRT	TK	CH	IP Ver	GW IP	GW PORT	FLT
Path A	ON	1	10	10	1	16	IPv4	172.16.30.1	5060	Enable
Path B	ON	2	10	10	17	48	IPv4	172.16.30.2	5060	Enable
Path C	ON	3	10	10	65	64	IPv4	172.16.30.3	5060	Enable

Then, enter MAC address of MG-SIP B and then enter LENS of MG-SIP Virtual Signaling Channel (LENS=021000).

Assign path data of Path D/E for MG-SIP B.

Path of MG-SIP	Select	Path ID	RT	LGRT	TK	CH	IP Ver	GW IP	GW PORT	FLT
Path D	ON	1	10	10	129	64	IPv4	172.16.30.4	5060	Enable
Path E	ON	2	10	10	193	63	IPv4	172.16.30.5	5060	Enable

Example 7: Remove an MG-SIP



STEP 1: **MBTK**- For each trunk of RT = 10, select "2: Make Busy (Bothway)" from the MB drop-down list box.

Note: Make sure that all the channels are placed in the idle state, then go to the next step.

STEP 2: **AMGIL**- Delete the registration data of MG-SIP A and MG-SIP B.

Note: When the registration of MG-SIP is deleted, the associated path information will also be deleted.

Note: After the registration of MG-SIP is deleted, MG-SIP will be reset.

STEP 3: **ACSC** - Delete the already registered CSC Group information (even-numbered CSCG) and assign the LENS (same as that of CIC Group Number 0) of CIC Group Numbers 4 to 7.

<Before the change>

CSC Group Number (CSCG)	CIC Group Number (CICG)	LENS of Signaling Channel (CCH)			Remarks
		MG 00-07	U 0-3	G 00-23	
130	0	02	0	4	MG-SIP A
	1	02	0	8	
	2	02	0	12	
	3	02	0	16	
	4	02	1	0	MG-SIP B
	5	02	1	4	
	6	02	1	8	
	7	02	1	12	

<After the change>

CSC Group Number (CSCG)	CIC Group Number (CICG)	LENS of Signaling Channel (CCH)			Remarks
		MG 00-07	U 0-3	G 00-23	
130	0	02	0	4	MG-SIP A
	1	02	0	8	
	2	02	0	12	
	3	02	0	16	
	4	02	0	4	Note 17
	5	02	0	4	
	6	02	0	4	
	7	02	0	4	

In the next step, delete the next CSCG (greater by one than the last CSCG).
Assign LENS of MG-SIP B Virtual Signaling Channel to the same CIC Group Number as that changed in the even-numbered CSC Group (CSCG).

<Before the change>

CSC Group Number (CSCG)	CIC Group Number (CICG)	LENS of Signaling Channel (CCH)			Remarks
		MG 00-07	U 0-3	G 00-23	
131	0	02	0	4	MG-SIP A
	1	02	0	4	
	2	02	0	4	
	3	02	0	4	
	4	02	1	0	MG-SIP B
	5	02	1	0	
	6	02	1	0	
	7	02	1	0	

<After the change>

CSC Group Number (CSCG)	CIC Group Number (CICG)	LENS of Signaling Channel (CCH)			Remarks
		MG 00-07	U 0-3	G 00-23	
131	0	02	0	4	MG-SIP A
	1	02	0	4	
	2	02	0	4	
	3	02	0	4	
	4	02	0	4	Note 17
	5	02	0	4	
	6	02	0	4	
	7	02	0	4	

Note 17: Apply the same settings as CICG=0 to the remaining CIC Group Numbers (CICG).

STEP 4: **MBTK**- Cancel Make-busy for all the trunks of RT = 10.

STEP 5: **ATRK** - Delete the trunk for Path D and Path E.

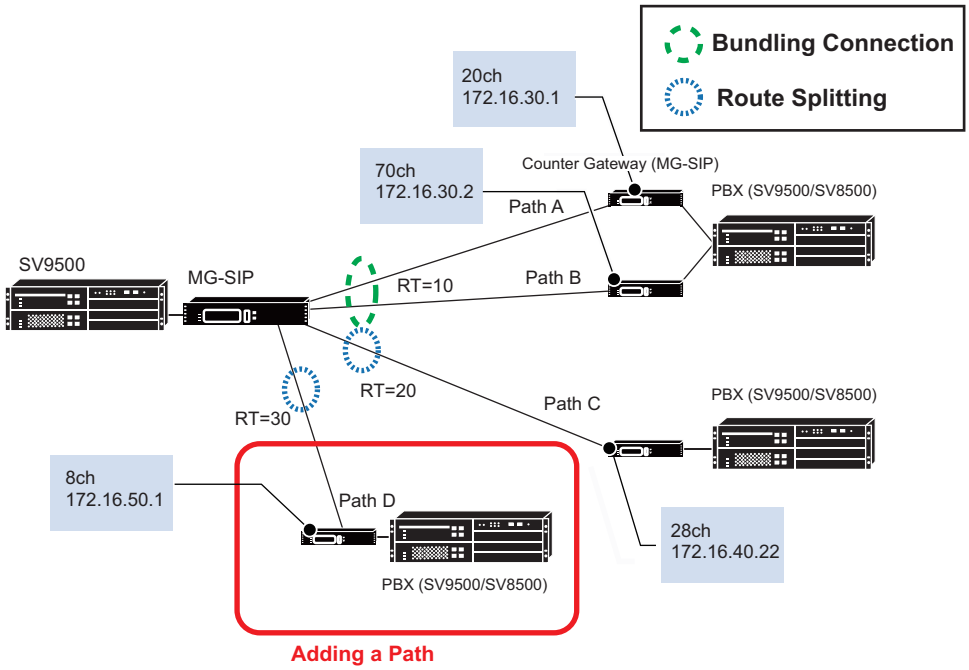
MG-SIP	Path for MG-SIP	Number of Channels	LENS			
			Highway 1	Highway 2	Highway 3	Highway 4
MG-SIP A	Path A	16	020040 - 020057	-	-	-
	Path B	48	020060- 020077	020080- 020117	-	-
	Path C	64	-	-	020120 - 020157	020160 - 020197
MG-SIP B	Path D	-	-	-	-	-
	Path E	-	-	-	-	-

STEP 6: **AMGIL** - Enter MAC address of MG-SIP A and then enter LENS of MG-SIP Virtual Signaling Channel (LENS=020040).

Assign path data of Path A/B/C for MG-SIP A (the same data as it was before the deletion).

Path of MG-SIP	Select	Path ID	RT	LGRT	TK	CH	IP Ver	GW IP	GW PORT	FLT
Path A	ON	1	10	10	1	16	IPv4	172.16.30.1	5060	Enable
Path B	ON	2	10	10	17	48	IPv4	172.16.30.2	5060	Enable
Path C	ON	3	10	10	65	64	IPv4	172.16.30.3	5060	Enable

Example 8: Adding New Path and Trunk to Speech Channel deleted in Examples 4 and 5



STEP 1: **ARTD/ARTDN**- Assign a new route for Path D.

STEP 2: **ARTI/ARTIN**- Assign a new route for Path D.

STEP 3: **ATRK**- Assign a new trunk for Path D.

Path for MG-SIP	Route Number	Trunk Number	Number of Channels	LENS			
				Highway 1	Highway 2	Highway 3	Highway 4
Path A	10	1-20	20	020040-020063	-	-	-
Path D	30	1-8	8	020064-020073	-	-	-
Path B	10	31-100	70	-	020112-020117	020120-020157	020160-020197
Path C	20	1-28	28	020076-020077	020080-020111	-	-

STEP 4: **ACSC**- The setting of this command data is not required to be changed because the trunk of Path D can be accommodated within the LENS of CIC Group 1.

CSC Group Number (CSCG)	CIC Group Number (CICG)	LENS of Signaling Channel (CCH)			Remarks
		MG 00-07	U 0-3	G 00-23	
130	0	02	0	4	used for Path A/C-> used for Path A/C/D
	1	02	0	8	used for Path B/C
	2	02	0	12	used for Path B
	3	02	0	16	used for Path B
	4	02	0	4	**Note 18**
	5	02	0	4	**Note 18**
	6	02	0	4	**Note 18**
	7	02	0	4	**Note 18**

Note 18: Apply the same settings as CICG=0 to the remaining CIC Group Numbers (CICG).

STEP 5: **ADPC**- Assign the same PC as RT=10/20 for RT=30.

STEP 6: **MBTK**- Cancel Make-busy for RT=30/TK=1-8.

STEP 7: **AMGIL**- Assign path information data of Path D.

Path of MG-SIP	Select	Path ID	RT	LGRT	TK	CH	IP Ver	GW IP	GW PORT	FLT
Path A	ON	1	10	10	1	20	IPv4	172.16.30.1	5060	Enable
Path B	ON	2	10	10	31	70	IPv4	172.16.30.2	5060	Enable
Path C	ON	3	20	20	1	2	IPv4	172.16.40.22	5060	Enable
Path D	ON	4	30	30	1	8	IPv4	172.16.50.1	5060	Enable

Note: The reset of MG-SIP is not required.

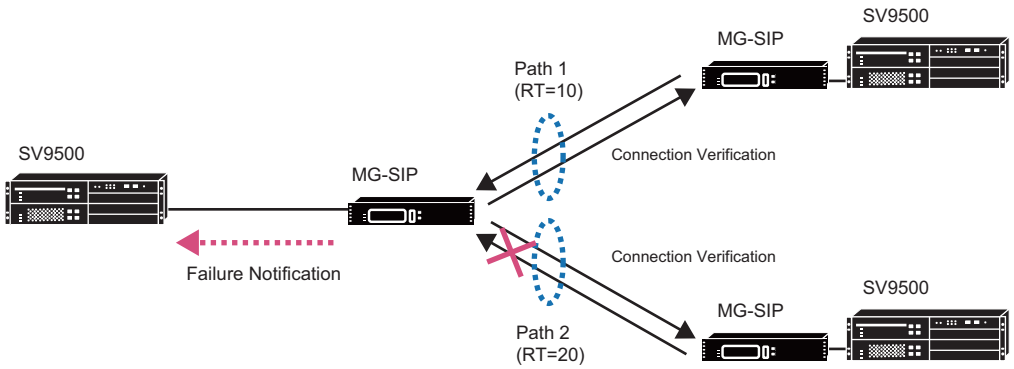
STEP 8: **AMGIL**- Reset Path D.
MG-ID: MAC address of MG-SIP (six bytes)
* Enter MAC address of Telephony Server side (LAN side).
FUNCTION: Reset
Reset Type: PATH (Reset Automatically Select)
The check box for Path D is automatically selected.

M-128 MG-SIP TIE Line Failure Detection Per Path

General Description

MG-SIP periodically sends keep-alive packets to each path to verify that the connection is valid. If MG-SIP detects a communication failure or recovery, it sends a failure notification to Telephony Server.
Telephony Server generates a system message after receiving a failure notification, then sets or cancels the make-busy status on the appropriate trunk.

Note: This feature is available since FP95-113 V3.



Operating Procedure

None.

Service Conditions

<Important Notices>

1. To use this feature, MG-SIP TIE LINE PATH SPLITTING [M-127] needs to be correctly set and to work properly.
2. When this feature is changed to valid or invalid, the corresponding paths become unavailable. After the change, reset the paths through the AMGIL command to make the paths available. In this case, the reset of MG-SIP is not required.
3. To use this feature, enable this feature with the set options command on MG-SIP and set health check interval.

<Interactions with Other Features>

None.

<Hardware/Software>

1. This service is available for MG-128SIPMGJ-B with firmware versions SP-4060 MGSIP PROG-L Issue 3 or later.

<Interactions with Networks>

None.

<Conditions Specific to This Feature>

1. Telephony Server generates a system message after receiving a failure notification from MG-SIP, then sets or cancel the make-busy status on the appropriate path (channel).
2. When a communication failure is detected, busy calls are released. When a call is originated to a path, where a failure is detected, in the period between the occurrence and detection of a failure, the call can be routed to an alternate route by PRI/PRA FAILSAFE ROUTING [P-49/50].
3. When Telephony Server was informed of communication failure in a path and has not yet been notified of the path's recovery, the Telephony Server blocks calls from the path.
4. Telephony Server generates the following system messages when receiving notification of a communication failure or recovery.

Time to Generate	System Message
Failure of Communication	35-L: IP EQUIPMENT NETWORK FAILURE WARNING (MEDIA GATEWAY)
Recovery of Communication	35-M: IP EQUIPMENT NETWORK RESTORE (MEDIA GATEWAY)

5. Telephony Server can simultaneously generate a maximum of 64 system messages. When Telephony Server needs to generate 65 or more system messages simultaneously, the 65th and following messages may not be generated.
   Example:
   When a failure occurs during communication between a gateway and an MG-SIP in an office where MG-SIP TIE LINE FAILURE DETECTION PER PATH [M-128] is enabled on 65 or more paths, MG-SIP detects the communication failures in all the paths and notifies Telephony Server about it. Due to the limit amount of simultaneous system message output, however, the Telephony Server can generate only 64 copies of the system message "35-L: IP EQUIPMENT NETWORK FAILURE WARNING (MEDIA GATEWAY)" at a time, leaving out the rest of the information about the failure.

<Other Conditions>

None.

Programming

* The following is explained under the presumption that the data settings for MG-SIP TIE LINE [M-122] have been completed.

STEP 1: **AMGIL** - Assign the data regarding to MG-SIP as follows:
MG-ID: Enter MAC Address (six bytes) of MG-SIP. * Enter the Telephony Server side (LAN side) MAC address.
FUNCTION = Setting
KIND = SIP-MG

LINE = 0 (fixed)
CH = 0 (fixed)
LENS: Enter the accommodated location of Virtual Signaling Channel (the first number of HW).
Retry = 0
Setting Type of Signaling Packet Service: Specify QoS settings of Signaling that is used when Internal PHI-BRI connects MG-SIP.
Usable CH Number: Enter the number of channels to be used per MG-SIP (1-128).

Note: Be sure to set 128ch (fixed) when this feature (MG-SIP TIE Line Path Splitting) is used.

Service of Authentication Exists: Unchecked
Setting of Path Data

Note: When the Set button of the command is pressed while this check box is OFF, all the registered Path data is cleared.

List of Path Data

INDEX: Index of Path data (1-32)
Select: Choose the check box for the path to be registered.

Note: When the Set button of the command is pressed while the check box for the already registered path is OFF, the path data is cleared.

Path ID: Identification Number of Path (1-255)
RT: External Route Number (1-255)
* Enter the route number (route number of Virtual Speech Channel for MG-SIP assigned by the ATRK command) to be assigned to this path.
LGRT: Logical Route Number (display only)
TK: Trunk Number (1-255)
* Enter the first trunk number of the trunks (trunk number of Virtual Speech Channel for MG-SIP assigned by the ATRK command) to be assigned to this path.
CH: Enter the number of MG-SIP channels to be assigned to this path.
IP Ver: IP Address Version = 1 (IPv4)
GW IP: Enter an IP address of the counter gateway (0.0.0.1-255.255.255.254).
GW PORT: Enter a port number of the counter gateway (1024-65535).
FLT: Gateway Fault Detection Capability (0/1)
* To enable MG-SIP Fault Detection per Path, set this parameter to 1 (Enable).

<Example of AMGIL Command Settings>

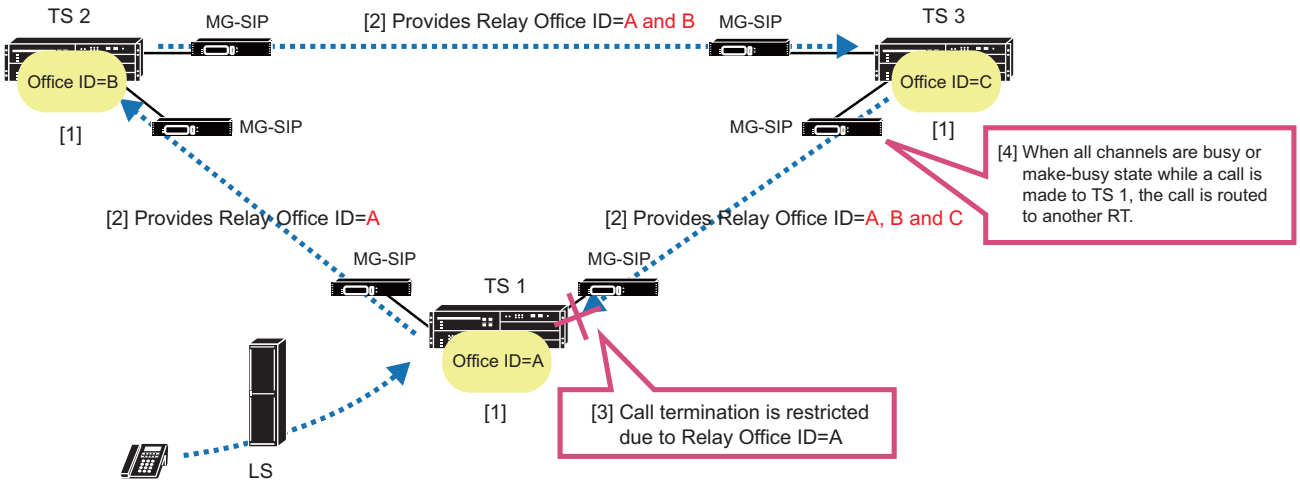
Path of MG-SIP	Select	Path ID	RT	LGRT	TK	CH	IP Ver	GW IP	GW PORT	FLT
Path A	ON	1	10	10	1	64	IPv4	172.16.30.1	5060	Enable
Path B	ON	2	10	10	65	32	IPv4	172.16.30.2	5060	Enable
Path C	ON	3	10	10	97	32	IPv4	172.16.40.1	5060	Disable

M-129 MG-SIP TIE Line Loop Detection

General Description

When a call routed via MG-SIP loops through nodes, the call termination is restricted.

Note: This feature is available since FP95-113 V3.



1. TS (Transit Switch) office checks the Relay Office ID when receiving a call from an office enabling Loop Detection.
2. When a call is originated from TS office to a tandem office, TS office notifies the tandem office about the Relay Office ID (received Relay Office ID + Self Office ID).
3. TS office checks the received Relay Office ID. When the TS's self Office ID is included in the Relay Office ID, the TS office restricts call originations.

   Note: Since the restriction of call terminations by Loop Detection is executed without analyzing incoming call numbers, station calls in the appropriate TS office and incoming calls routed to LS (Local Switch) stations via the TS office are restricted.
4. When a tandem call is restricted by Loop Detection, the call can be routed to an alternate route by PRI/PRA FAILSAFE ROUTING [P-49/50] (ASYDL/ASYDN SYS1, Index 1356, Bit 0 = 1).

Operating Procedure

None.

Service Conditions

<Important Notices>

None.

<Interactions with Other Features>

1. Call Transfer-All Calls / Blind Transfer / Call Forwarding - Don't Answer:
   When a station transfers an incoming call from an MG-SIP route enabling Loop Detection to another MG-SIP route also enabling Loop Detection, only the Office ID of the relay station is sent to the transfer destination office. The office of the relay station may not send the Relay Office ID received at the arrival of the incoming call to the destination office.
2. Call Forwarding - All Calls / Call Forwarding - Busy Line / Call Forwarding - Logout / Call Forwarding - PS Incoming Incomplete:
   When a call from an MG-SIP (the first MG-SIP) route enabling Loop Detection is forwarded to another MG-SIP (the second MG-SIP) route also enabling Loop Detection, the Relay Office ID received by the first MG-SIP office can be sent to the second MG-SIP office.
3. When a call is originated from MG-SIP through CCIS ALTERNATE ROUTING - BACKUP [C-178] or PRI/PRA FAILSAFE ROUTING [P-49/50], the Relay Office's information can be sent to the destination office.

<Hardware/Software>

1. This service is available for MG-128SIPMGJ-B with firmware versions SP-4060 MGSIP PROG-L Issue 3 or later.

<Interactions with Networks>

1. In an FCCS network, the same ID for Loop Detection must be set for all the node in the network (ASYDL/ASYDN, SYS1, Index 1355).

<Conditions Specific to This Feature>

1. Up to 32 Relay Office IDs can be registered.
2. This feature is only available for calls via routes with MG-SIP TIE Line Loop Detection enabled. If a call uses a route not supporting the feature, information of Relay offices will be reset.
3. When Loop Detection service is enabled (Self Office ID is set by the ASYDL/ASYDN. SYS1, Index 1355), if a call, after having passed through more than 31 relay offices, comes from an MG-SIP, Telephony Server blocks the call regardless of whether or not a loop is detected.
4. An error response (482 Loop Detected) is transmitted to the originating office of MG-SIP TIE Line call. When a call is routed to an alternate route by PRI/PRA FAILSAFE ROUTING [P-49/50], however, the other error response is issued according to the result of the alternating action.

<Other Conditions>

None.

Programming

* The following is required to be set for TS office.

* The following is explained under the presumption that the data settings for MG-SIP TIE LINE [M-122] have been completed.

STEP 1: **ASYDL/ASYDN**

SYS1, Index 1355: Assign an ID for Loop Detection (1-255). **Note 1**, **Note 2**

Note 1: When "0" is assigned, this feature becomes disabled.

Note 2: In an FCCS network, the same ID must be set for all the nodes in the network.

SYS1, Index 1356, Bit 0=1 (Alternate routing is available)

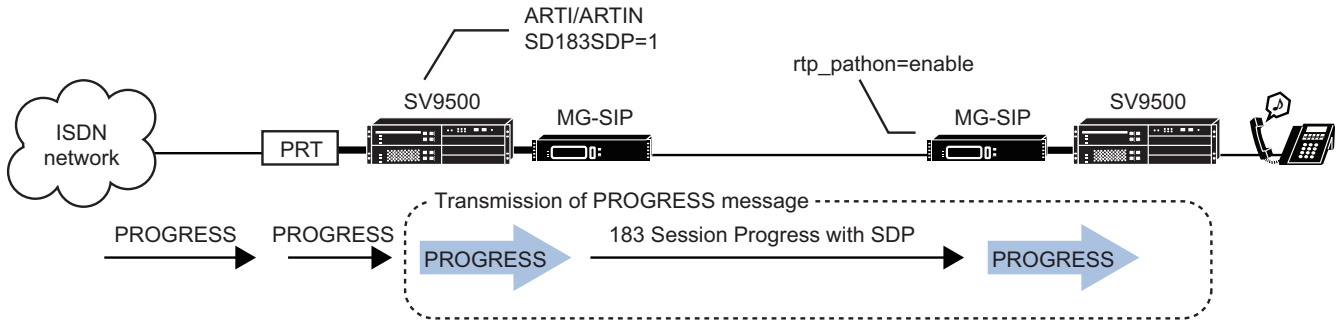
STEP 2: **ARTI/ARTIN** - Assign Trunk Application Data of MG-SIP for TIE Line.

Loop (CDN:107)=1 (Loop Detection is enabled)

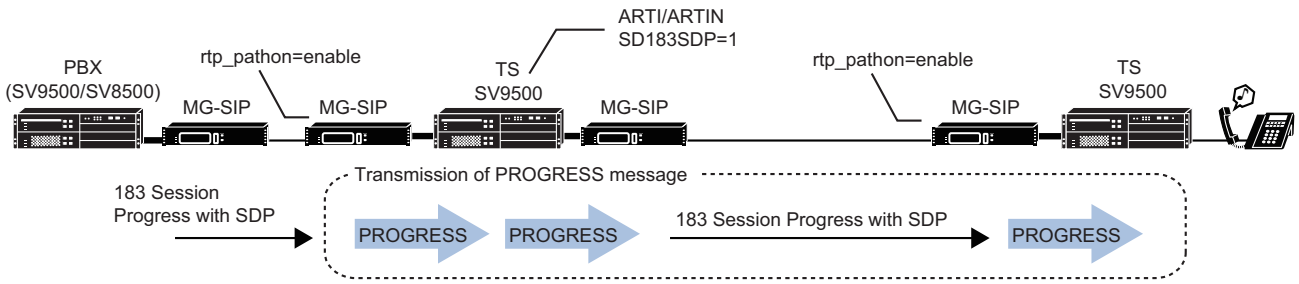
M-130 MG-SIP TIE Line In-band Tone

General Description

In a tandem connection from MG-SIP TIE Line to an external trunk (ISDN line, etc.), an in-band tone (counter office audible sound) such as announcement from the external trunk can be provided by transmitting the PROGRESS message received from the external trunk to MG-SIP. In-band tone can also be provided in a tandem connection to ACIS line (2 Mbps digital interface / ODT) by sending the PROGRESS message to MG-SIP.



Also when an MG-SIP receives 183 Session Progress with SDP from a counter SIP gateway, the MG-SIP sends the PROGRES message to Telephony Server and the PROGRESS message can be provided to another MG-SIP which has received the message.



Note: This feature is available since FP95-113 V3.

Operating Procedure

None.

Service Conditions

<Important Notices>

1. Both of the following settings are required to use this feature.
   a.) Set “enable” for the MG-SIP configuration command “set rtp_pathon”.

Note: This setting is required for an outlet-side MG-SIP which receives 183 Session Progress with SDP and sends the PROGRESS message.

b.) For a route of an inlet-side MG-SIP which hears an in-band tone, assign "1" to the SD183SDP (CDN: 108) parameter of the ARTI/ARTIN command.

Note: Note the following:

- When a counter MG-SIP/SIP gateway of the inlet-side MG-SIP does not support 183 Session Progress with SDP even if this setting is enabled, the inlet-side MG-SIP cannot send the announcement to the tandem destination.
- When a counter device of the inlet-side MG-SIP is an MG-SIP, set "enable" for the configuration command "set rtp_pathon" on the counter MG-SIP.

<Interactions with Other Features>

1. A calling party can hear an in-band tone also by setting ASYD, SYS1, Index 372, Bit 5=1 (Return message for a tandem call to ACIS line via MG-SIP=200 OK). However, when a station user attempts to transfer a call to an external trunk and then performs transfer switchback operation while hearing audible sound such as an in-band busy tone or announcement, the station user may continue to hear the audible sound even after the transfer switchback. For this reason, NEC recommends the following settings.

   ASYD, SYS1 Index 372, Bit 5=0

   ARTI/ARTIN SDN183SDP (CDN: 108)=1

<Hardware/Software>

1. MG-SIP supporting MG-SIP TIE Line is required.

<Interactions with Networks>

1. This feature is available only for tandem connections. The following trunks can be used as a destination trunk of MG-SIP.

   - ACIS line (2 Mbps digital interface/ODT)
   - PRT
   - BRT
   - MG-PRI/UG50-PRT/UG50-MG-PRT
   - MG-BRI
   - MG-SIP **Note 1**

   Note 1: MG-SIP is available only when the configuration command "set rtp_pathon" is enabled on it.

<Conditions Specific to This Feature>

1. Route Number of the MG-SIP to which you do not want to send an in-band tone should be set separately from the MG-SIP TIE Line route.

<Other Conditions>

None.

Programming

* The following is explained under presumption that the data settings for MG-SIP TIE LINE [M-122] have been completed.

STEP 1: **ARTI/ARTIN** - Set the following data to the virtual speech channel route (incoming route) of the MG-SIP which transmits the PROGRESS message on a tandem connection from a MG-SIP TIE Line to an external trunk.
SD183SDP (CDN:108) = 1 (MG-SIP TIE Line In-band Tone is enabled) **Note 2**

Note 2: This feature is available only for MG-SIP TIE Line.

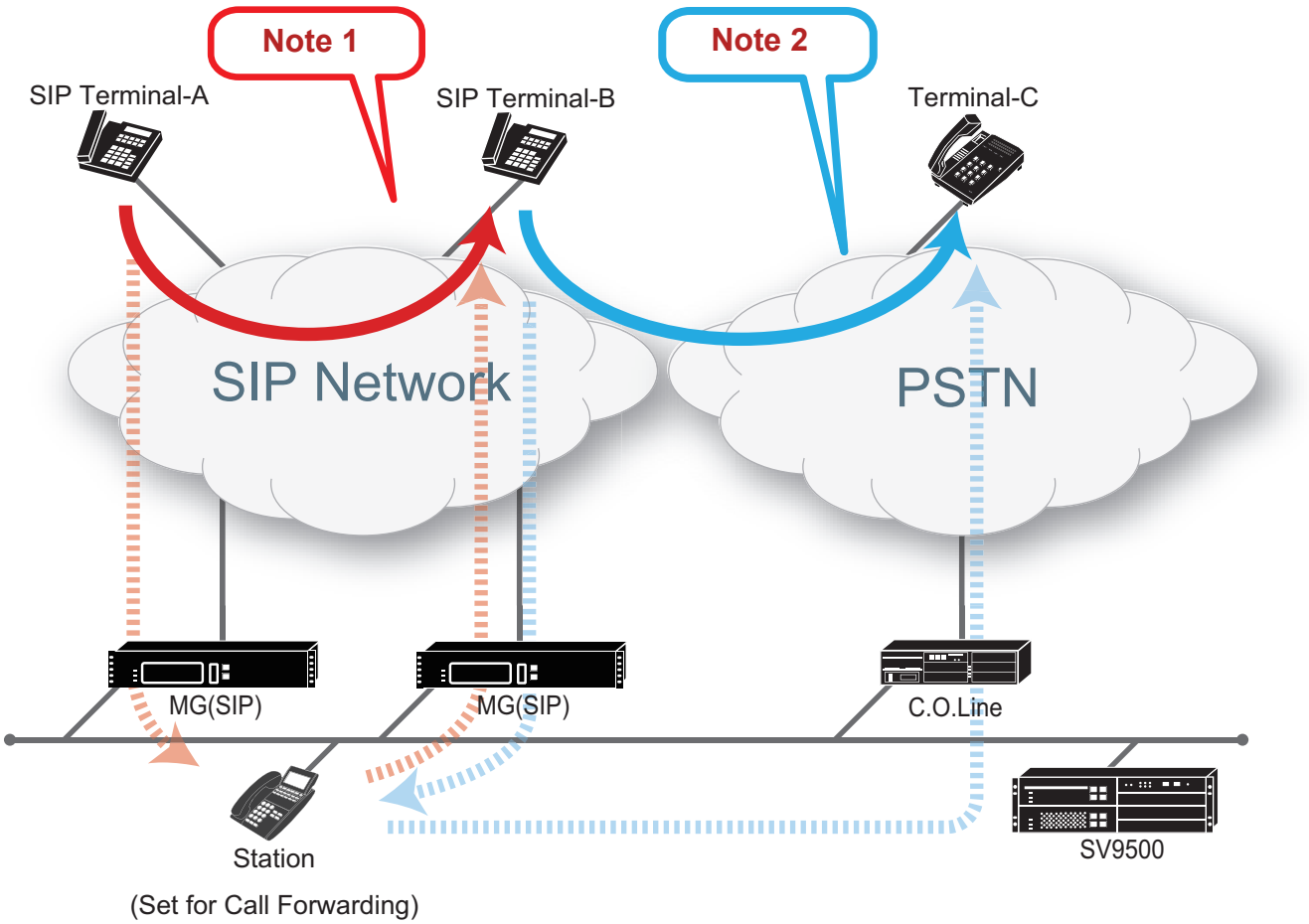
M-131 MG-SIP Call Forward - Moved Temporarily Redirect

General Description

When a SIP call comes through an MG-SIP to a station configured for Call Forwarding, this feature enables the MG-SIP to send a Call Forwarding destination with the 302 Moved Temporarily SIP response (compliant with RFC3261) back to the call originator.
While free from the need for control of the speech channel with the MG-SIP, the Telephony Server releases the channel, helping to make more efficient use of resources.

Note: Note the following:

- This feature is available since FP95-114 V4 and for New Zealand only.
- Terminals on the SIP Network side require connection tests of NEC.



Note 1: Redirecting SIP terminal-A to SIP terminal-B without MG-SIPs.

Note 2: Redirecting SIP terminal-B to Terminal-C without C.O.Line.

Operating Procedure

None.

Service Conditions

<Important Notice>

1. When an incoming call is from an MG-SIP connected to SIP networks in which call redirection using 302 Moved Temporarily is supported and the C.O. line number is assigned to the Call Forwarding destination, this feature is available.
2. This feature can be used in conjunction with the following services. If it is used with other services, trunks of MG-SIPs are not released.
    - CALL FORWARDING - ALL CALLS - OUTSIDE [C-28]
    - CALL FORWARDING - BUSY LINE - OUTSIDE [C-60]
    - CALL FORWARDING - DON'T ANSWER - OUTSIDE [C-74]
    - LOGGED OUT IP STATION - CALL DESTINATION [L-53]
    - CALL FORWARDING PS INCOMING CALL INCOMPLETE [C-155]
    - SPLIT CALL FORWARDING [S-99]
3. After the Telephony Server releases the speech path, the caller action depends on the SIP specification.

<Interactions with Other Features>

1. This feature does not work when a call is transferred via an MG-SIP after being processed by the Telephony Server.
2. When the forwarding destination is busy, the call is established on Multi-hop routing if they are assigned.
3. Peg Counts do not record event occurrences after call redirection with 302 Moved Temporarily SIP response. They count only event occurrences for calls that are forwarded through CALL FORWARDING - DON'T ANSWER.
4. The billing system does not provide any information about calls after being redirected with 302 Moved Temporarily SIP response. It generates information only about incoming calls forwarded through CALL FORWARDING - DON'T ANSWER, in which case these calls are recorded as incomplete calls.

<Hardware/Software>

1. This feature is only available for the following MG-SIPs.
    - MG-SIP16 [SCA-16SIPMG(US)-B/SCA-16SIPMG(US)]
      : SP-3988 MGSIP(16) PROG-G Issue 10 or later
    - MG-SIP128 [MG-128SIPMGL-A]
      : SP-4089 MGSIP(MM) PROG-A Issue 1 or later
    - Software-based MG-SIP
      : SP-4085 MGSIPVM PROG-E Issue 1 or later

<Interactions with Networks>

1. This feature can be used in conjunction with the FCCS services. However, it does not work with the CCIS services.

<Conditions Specific to This Feature>

1. For the destination route, the parameter TCL (CDN: 6) of ARTD/ARTDN commands as "1" (DDD Lines) is required.
2. The notification of the added/deleted Access Code number by using the following commands in the forwarding destination numbers is given to the call originator. Since the notified number is used by the call originator for redirecting, Access Code should be assigned the number of redirecting available.
   - AOPR/AOPRL/AOPRN
   - AADC/AADCL/AADCN
3. When all routes for forwarding destination are busy, this feature works. The following uses LCR Alternate Routing as an example:

	First Routing		Second Routing		Result
Ex.1	C.O. Line A	Idle	C.O. Line B	Idle	Redirect with Access Code of C.O. Line A
Ex.2	C.O. Line A	All Busy	C.O. Line B	Idle	
Ex.3	C.O. Line A	All Busy	TIE Line B	Idle	
Ex.4	TIE Line A	Idle	C.O. Line B	Idle	Call Forwarding to TIE Line A without Redirect
Ex.5	TIE Line A	All Busy	C.O. Line B	Idle	Redirect with Access Code of C.O. Line B (because the last destination is C.O. Line.) **Note 3**
Ex.6	TIE Line A	All Busy	C.O. Line B	All Busy	

Note 3: The second routing is routed via Public Network in case of Ex.5 and Ex.6.

4. When the forwarding destination is restricted by the office data, this feature does not work.
5. The call forwarding features provided by a telephone terminal (such as a Standard SIP terminal) are not supported.

Programming

ARTI/ARTIN - Enable MG-SIP CALL FORWARD - MOVED TEMPORARILY REDIRECT feature.
CDN111 : 302REDIRECT
0 : Out of service
1 : In service

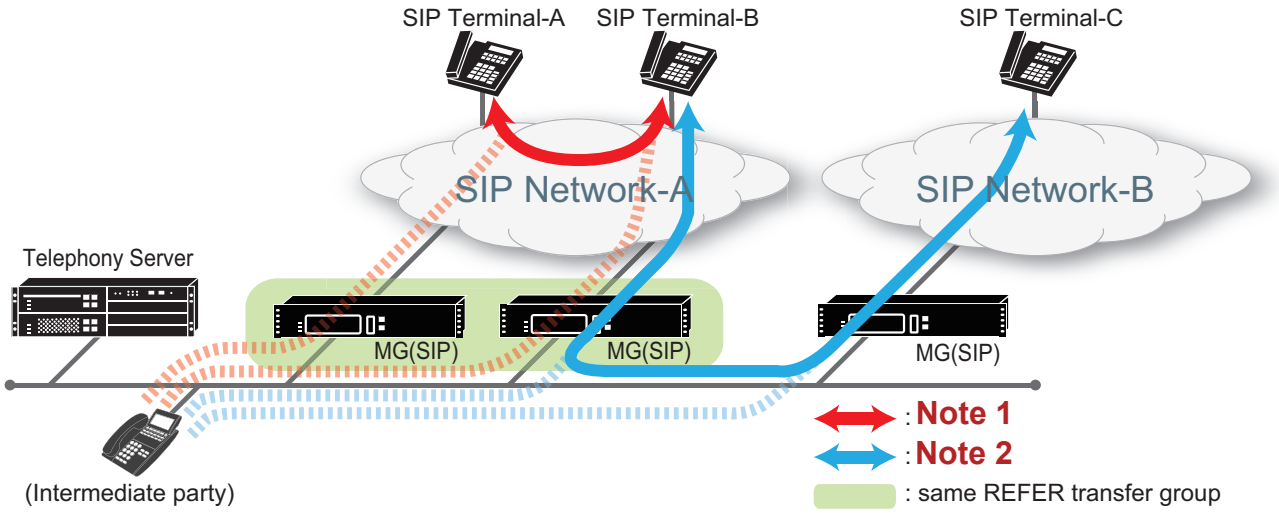
M-132 MG-SIP REFER Based Transfer

General Description

This feature uses the SIP REFER method based on RFC3515 to release the speech channel on a tandem call between MG-SIPs with the same REFER transfer group number, and thereby provides efficient handling of trunks of MG-SIP.

Note: Note the following:

- This feature is available since FP95-114 V4.
- This feature is available in North America, Australia and New Zealand.
- Terminals on the SIP Network side require connection tests of NEC.



Note 1: A call is established between SIP terminal-A and SIP terminal-B without MG-SIPs.

Note 2: A call is established between SIP terminal-B and SIP terminal-C with MG-SIPs because they are not in the same REFER transfer group.

Operating Procedure

None.

Service Conditions

<Important Notice>

1. This feature is available for following firmware version:

   - MG-128SIPMGL-A : SP-4089 MGSIP(MM) PROG-A
   - Software-based MG-SIP : SP-4085 MGSIPVM PROG-E

   Note: This feature requires the configuration of the MG-SIP. For setting details, see the following manuals.

MG-128SIPMGL-A	MG-SIP128 [MG-128SIPMGL-A]: CHAPTER 2 in Peripheral Equipment Description (IP Devices)
Software-based MGSIP	SOFTWARE-BASED MG-SIP: CHAPTER 2 in Software Model Peripheral Equipment Description

2. This feature works with a tandem connection between MG-SIPs with the same REFER transfer group number.
3. This feature is used when performing CALL TRANSFER - ALL CALLS [C-11], BLIND TRANSFER TO ATTENDANT [B-17], and BLIND TRANSFER TO STATION [B-18]. When performing other transfer, trunks of MG-SIPs are not released.
4. The SIP-compatible equipment in the SIP network side requires the following:

   - For call originator equipment (Receiving REFER request from MG-SIP)
     Processing the REFER and INVITE (Replaces) requests are required.
   - For transferred destination equipment (Receiving INVITE (Replaces) request from SIP network)
     Processing the INVITE (Replaces) request is required.

<Interactions with Other Features>

1. This feature does not work when an intermediate party terminal (a party that transfers the call) is as follows:
   - Attendant console
   - ISDN terminal
2. This feature does not work when a tandem call between MG-SIPs is established after CCIS link is reconnected.
3. When the Telephony Server receives a RELCOM message from an MG-SIP after switching a speech path, billing system and Peg Count terminate the counting.
4. Billing system does not count a call which has been completed by REFER transfer.
5. This feature does not work when an intermediate party drops out of the Three-Party Conference and the call backs to the two-way connection.

<Hardware/Software>

None.

<Interactions with Networks>

1. On the connection via FCCS network, only FCCS Networking via IP (Peer-to-Peer) is supported. FCCS Networking via DTI (DTI FCCS) is not supported.

<Conditions Specific to This Feature>

1. Enabling the REFER/NOTIFY function by using the `set sip_tel_service` command is required.
2. The speech channel is not released during the Ringing state after completion of Blind Transfer by REFER. The transferred call is switched only after the call is established (a called party answers) between a call originator and a called party.
3. The result of the path switching depends on the SIP specification (REFER/NOTIFY/INVITE (Replaces)). If the path switching fails, a tandem call continues without release of the channels.
4. The speech path is switched by using one or two MG-SIPs only when the path connects between MG-SIPs of forwarding and forwarding destination equipment.

Programming

ARTI/ARTIN - Assign the same REFER transfer group number to both MG-SIPs on tandem connection.
CDN109 : REF_GRP = 1 to 7 (REFER transfer group)
0 : REFER Transfer is out of service.
1 to 7 : REFER transfer group

N-1 Night Connection - Fixed

General Description

This feature routes calls normally directed to the ATTENDANT CONSOLE [A-3] to a preselected common Station within the system when the Night Mode has been entered.

Operating Procedure

To answer NIGHT CONNECTION Calls

1. The Incoming Calls to the ATTENDANT CONSOLE [A-3] in the night mode are automatically transferred to a preselected NIGHT CONNECTION - FIXED Station.
2. Calling Party hears Ringback Tone.
3. Ring signal is sent to the NIGHT CONNECTION - FIXED Station.
4. NIGHT CONNECTION - FIXED Station goes off-hook to answer the Incoming Call. If necessary, this Station can transfer the Incoming Call to another Station within the system, using CALL TRANSFER - ALL CALLS [C-11] feature.

Service Conditions

1. Calls that are being processed or are in the hold state when the night mode is entered, continue being processed or held after the mode changeover has been completed.
2. If the NIGHT CONNECTION - FIXED Station, to which an Incoming Call has been directed is busy, the Calling Party will hear Ringback Tone until the Station becomes idle.
3. This feature may be provided with NIGHT CONNECTION - FLEXIBLE [N-2] and TRUNK ANSWER FROM ANY Station [T-8] in the same system.
4. If the NIGHT CONNECTION - FIXED Station is set for CALL FORWARDING - ALL CALLS [C-5], Incoming Central Office Trunk Calls terminated to that Station will be forwarded to the designated Station.
5. Programmed NIGHT CONNECTION - FIXED Stations can use CALL HOLD [C-6], CALL TRANSFER [C-10, 11] and THREE-WAY CALLING [T-2] provided these features are programmed into its CLASS OF SERVICE - INDIVIDUAL [C-15].
6. Only Ringdown (COT) Incoming Calls will be routed to NIGHT CONNECTION - FIXED Stations.
7. If an Incoming Call has terminated to a NIGHT CONNECTION - FIXED, and if the Station is busy or in LINE LOCKOUT [L-3], the call will be forwarded to the TRUNK ANSWER FROM ANY STATION (TAS) [T-8] unit, if equipped.
8. When activating this feature within an FCCS Network, the different destination for Central Office Incoming Calls (Ringdown) cannot be set at each Tenant Group.

Interactions

1. The Night Station is assigned to each Trunk, and the same NIGHT CONNECTION - FIXED [N-1] can be assigned to several Trunks.
2. All Trunks to be terminated at the NIGHT CONNECTION - FIXED Station must be programmed in ACSI. Refer to DIRECT-IN TERMINATION [D-7] for more information on DITs.
3. The system must enter Night mode. Refer to ATTENDANT CONSOLE [A-3] and DAY/NIGHT CLASS OF SERVICE [D-15].
4. The Attendant Console must first be provided.

5. When the NIGHT CONNECTION - FLEXIBLE [N-2] feature is provided, assignments can be made by the ATTENDANT CONSOLE [A-3] or NIGHT CONNECTION - FIXED Station by use of the CALL FORWARDING - ALL CALLS [C-5] feature.
6. CALL FORWARDING - BUSY LINE/DON'T ANSWER [C-2, 3]: A Night Station user can activate the CALL FORWARDING - BUSY LINE/DON'T ANSWER [C-2, 3] feature if available to the user. A call ringing at an idle NIGHT CONNECTION - FIXED Station rings for 30 seconds before forwarding. A call forwarded from a busy NIGHT CONNECTION - FIXED Station rings normally at the designated Station. A Station-to-Attendant Call cannot be forwarded to a NIGHT CONNECTION - FIXED Station. An Incoming Trunk Call forwarded from an idle unanswered NIGHT CONNECTION - FIXED Station alerts the designated Station with distinctive ringing.
7. CALL WAITING [C-12, 13]: Call Waiting Tones are not directed to a busy NIGHT CONNECTION - FIXED Station. When the Night Station user goes on-hook, ringing is returned if a call is waiting.
8. CALL TRANSFER - ALL CALLS [C-11]: When the NIGHT CONNECTION feature is active, the CALL TRANSFER - ALL CALLS [C-11] feature is always enabled, regardless of the Class of Service of the Station on which this feature is enabled.
9. STATION MESSAGE DETAIL RECORDING (SMDR) [S-10]: The SMDR records contain Night Service Calls. The SMDR Call record includes the Station Number answering the call.
10. Incoming Restricted Line: When this feature or NIGHT CONNECTION -FLEXIBLE [N-2] is active, the line Class of Service restrictions can be automatically changed when the system is in the Night mode, to allow a Station normally restricted during the day to receive Incoming Calls.
11. NIGHT CONNECTION - FIXED [N-1]: Arrangements are provided to route calls normally directed to the Attendant, to a preselected Station Line within the system when Night service is activated. In addition, Incoming Calls from specific non-Direct Inward Dialing (DID) exchange network Trunks can be programmed to route to specific Station Lines. The routings are provided on a fixed basis via the PCPro. TRUNK ANSWER FROM ANY STATION (TAS) [T-8], when available, may also be provided for Night Calls not assigned to Night Stations. Assignments can be made per Incoming Trunk.
12. When CALL FORWARDING - ALL CALLS [C-5] or CALL FORWARDING - BUSY LINE/DON'T ANSWER [C-2, 3] is provided, and the Station is busy, Call Forwarding is implemented on the Station Line. If the Station is in a HUNT GROUP [S-7, 8, 9], Call Forwarding will take place prior to the Station hunting feature or UCD [U-1].
13. CALL QUEUING [C-9]: Incoming Trunk Calls to the Night Connection Station will queue on a First In/ First Out basis.
14. CLASS OF SERVICE - INDIVIDUAL [C-15]: Night Connection Station's class of service should be constructed to allow access to the routes and features required to complete the tasks of the Night Connection Station.
15. DIRECT INWARD DIALING [D-8]: Only Listed Directory Number DID Calls (as assigned in the ALDN command) will be directed to the Night Connection Station. All DID Trunks must be programmed to the Night Connection Station via the ACSI command.
16. When system is in the Night mode of operation, Incoming Calls are directed in the following order:

NIGHT CONNECTION - FLEXIBLE	[N-2]
NIGHT CONNECTION - FIXED	[N-1]
TAS	[T-8]

17. When this feature is enabled, the Day/Night switching feature of DAY-NIGHT MODE STATUS -D - II [D-164D - II] is not available.
    On the other hand, the Day/Night mode status display feature of DAY-NIGHT MODE STATUS -D - II [D-164D - II] is available.
18. This feature is not available when DAY-NIGHT MODE STATUS -D - II [D-164D - II] is enabled.

Programming

STEP 1: **ASYD** - System Data 1, Index 147. DIT supervisory timer for a busy Station. How often will the CPU look at the Idle/Busy status of the Station for rerouting the DIT Call to the destination? Always leave this Index as 00H.

For NIGHT CONNECTION-FIXED Only.

STEP 2: **ACSI** - Assign External Route Number, Trunk Number, Connection Service Index of "4" for NIGHT CONNECTION-FIXED, and the Tenant Number and Station Number to serve as the NIGHT CONNECTION-FIXED Station.

STEP 3: **ACFR** - Allow a Transfer Service Feature Index of 2, NIGHT CONNECTION-FIXED, for the types of Incoming Calls listed in Call Class Index (CCI).

STEP 4: **ALDN** - Assign the following data:

TN: TN number
LDN: LDN number
TYPE 1: "To Station"
TYPE 2: "To Outside
TYPE 3: "To ADC"
DC: Destination Code (used when Type = 2)
ADC: Abbreviated Digit Code (used when Type = 3)

STEP 5: **ASPD** - Necessary when using ADC.

TN: Tenant Number
ADC: Abbreviated Digit Code
CD: Called Number

FCCS Programming

This data setting describes only the additional or replacement commands used to activate FCCS Services. When used as an FCCS feature, these commands replace the non-network commands (for example, AAED is replaced by AAEDN). Note that the data setting for FCCS Service varies depending on the Node.

Note: When using ASYDL or ASYDN to program an FCCS feature, the basic programming of ASYD is required in addition to ASYDL or ASYDN.

Note: The following data should also be set in addition to the above-mentioned commands. The command to be set to the Node accommodating the COT: ASYD/ACFR.

ACSIL - This command must be assigned to the Node accommodating the COT. Assign External Route Number/Logical Route Number (RT/LGRT), Trunk Number (TK), and Connection Service Index (CSI) = 4 for DIT activated in Night mode. Then assign the User Group Number (UGN) and Telephone Number (TELN) to serve as the NIGHT CONNECTION - FIXED Station.

N-2 Night Connection - Flexible

General Description

This feature provides arrangements to Route Calls, usually directed to the ATTENDANT CONSOLE [A-3], instead to a preselected Station on a flexible, assignable basis within the system, when the Night Mode has been entered.

Operating Procedure

1. Answering the operation is the same as that for NIGHT CONNECTION - FIXED [N-1].
2. Each evening, before placing the ATTENDANT CONSOLE [A-3] into the Night Connection mode, the Attendant programs CALL FORWARDING - ALL CALLS [C-5] from the fixed NIGHT CONNECTION - FIXED Station to the desired Station.

Service Conditions

1. The NIGHT CONNECTION - FLEXIBLE Station may be programmed from either the ATTENDANT CONSOLE [A-3] or the NIGHT CONNECTION - FIXED [N-1] Station. Refer to CALL FORWARDING - ALL CALLS [C-5] for details.
2. NIGHT CONNECTION - FLEXIBLE: Arrangements are provided to Route Calls, normally directed to the Attendant, to a preselected Station Line within the system when Night service is activated. Additionally, Incoming Calls from specific non-DID exchange network Trunks can be arranged to route to the specific Station Line. The routings can be assigned on a flexible basis by the Attendant or NIGHT FIXED Station by use of the Call Forwarding feature and remain in effect until changed. For this feature to function, a NIGHT CONNECTION - FIXED Telephone Number must be assigned to the Trunk via the PCPro. In the event of a power failure, the assignments are not lost. The TAS [T-8] answering capability is provided for Night Calls not assigned to Night Stations.
3. When activating this feature within an FCCS Network, the different destination for Central Office Incoming Calls (Ringdown) cannot be set at each Tenant Group.

Interactions

1. When this feature is enabled, the Day/Night switching feature of DAY-NIGHT MODE STATUS -D - II [D-164D - II] is not available.
   On the other hand, the Day/Night mode status display feature of DAY-NIGHT MODE STATUS -D - II [D-164D - II] is available.
2. This feature is not available when DAY-NIGHT MODE STATUS -D - II [D-164D - II] is enabled.

Programming

STEP 1: **ACSI** - NIGHT CONNECTION-FLEXIBLE is a combination of CALL FORWARDING-ALL CALLS [C-5] and NIGHT CONNECTION-FIXED [N-1] features.

STEP 2: **ASFC** - CLASS OF SERVICE-INDIVIDUAL [C-15]: The Station chosen as the Night Connection-Flexible Station should be assigned a CLASS OF SERVICE-INDIVIDUAL [C-15] capable of performing the Night Station functions.

FCCS Programming

This data setting describes only the additional or replacement commands used to activate FCCS Services. When used as an FCCS feature, these commands replace the non-network commands (for example, AAED is replaced by AAEDN). Note that the data setting for FCCS Service varies depending on the Node.

ACSIL - This command must be assigned to the Node accommodating the COT. Assign External Route Number/Logical Route Number (RT/LGRT), Trunk Number (TK), and Connection Service Index (CSI) = 4 for DIT activated in Night mode. Then assign the User Group Number (UGN) and Telephone Number (TELN) to serve as the DIT Station.

N-3 Non-delay Operation

General Description

This feature allows the ATTENDANT CONSOLE [A-3] to place any Calling Party on hold, dialing the call, and connecting the Calling and Called Parties.

Operating Procedure

To operate

1. The Attendant answers an Operator Call by pressing the ANSWER or ATND key.
2. Press the START key.
3. Dial the exchange network Access Code.
4. Dial the desired Telephone Number.
5. Press the RELEASE key.
6. The parties are connected.

Service Conditions

1. The Attendant may terminate the Connection either before or after the Called Station answers.
2. There is no AUTOMATIC RECALL [A-8] for Station-to-Trunk and Trunk-to-Trunk Calls when the Called Party does not answer.
3. If the Station is restricted, the Attendant will receive a Reorder Tone when attempting to establish a Central Office Connection.
4. When the PASSING DIAL TONE [P-2] feature is provided, and the Attendant has accessed an Outgoing Trunk, the call may be released from the ATTENDANT CONSOLE [A-3], and the user is allowed to complete dialing.
5. This feature allows the Attendant to place an Outgoing Call for a Station user who reached the Attendant via a dial “0” call without requiring the Station user to hang up.
6. The Attendant can dial the Called Number for the Station user, or using the PASSING DIAL TONE [P-2] feature, allow the Station user to complete the dialing.
7. A call can be transferred to a Trunk in the Remote Node via FCCS link.
8. When an Attendant Console receives a release signal including inband information from C.O. line (ISDN trunk or MC&MG-COT/MG-COT), the release timing of the ISDN trunk (or MC&MG-COT/MG-COT) differs depending on whether the Calling Party is released from the connection or placed on hold.

    - When an Attendant Console receives a release signal from C.O. line after the Calling Party releases the connection, the Attendant Console hears inband information such as a Reorder Tone (ROT). Then, the ISDN trunk (or MC&MG-COT/MG-COT) is released when the Attendant Console presses the RELEASE key or receives a release signal indicating the end of inband information after 30 seconds.

    - When an Attendant Console receives the release signal from C.O. line before the Calling Party releases the connection, the ISDN trunk (or MC&MG-COT/MG-COT) is released upon the reception of the release signal.

Interactions

1. The Attendant Console can seize the Dial Tone and dial an outgoing number for any Station in the system. The Attendant Console cannot pass an unrestricted Dial Tone to any Station. See PASSING DIAL TONE [P-2] for more information.
2. The Attendant Console must first be provided.

Programming

STEP 1: **ARSC** - Construct a Route Restriction Class (RSC) that will allow outgoing access to various routes using the Route Restriction Index (RRI) 2 through the Attendant Console.

STEP 2: **ASDT** - Assign a Route Restriction Class (RSC) that allows outgoing access to the various routes via the Attendant Console to the Station.

FCCS Programming

This data setting describes only the additional or replacement commands used to activate FCCS Services. When used as an FCCS feature, these commands replace the non-network commands (for example, AAED is replaced by AAEDN). Note that the data setting for FCCS Service varies depending on the Node.

ARSCN - Construct a Route Restriction Class (RSC) that will allow outgoing access to various routes using the Route Restriction Index (RRI) 2 through the Attendant Console.

N-7D Non-exclusive Hold -D

General Description

This feature permits a Desktop terminal user to HOLD a call in progress. To retrieve a Held Call, press the flashing line key on a Desktop terminal.

Note: Executable from any Desktop terminal with the multi-line appearance of the station that put the call on hold.

Hold Fixation Mode is also available for retrieving a Held Call. Using this feature, a Held Call can be retrieved by simply lifting the handset or pressing the HOLD key on the Desktop terminal. When DT750 is operating in XML Fixed mode, use this feature.

Note: Executable only from the Desktop terminal that placed the call on hold.

Operating Procedure

To place a call on HOLD,

1. While a call is in progress with an office line, a tie line, or a station line, press the HOLD key.

To retrieve the Held Call,

1. Press the Line key (flashing) to retrieve a call.
2. Retrieve the Held Call.

- In Hold Fixation Mode;

1. Lift the handset if it is on hook, or press the HOLD key on a Desktop terminal if the handset is off-hook.
2. Retrieve the Held Call.

To answer the Automatic Recall [A-8],

1. When an Automatic Recall Timer expires, the Desktop terminal which put the call on hold is recalled.
2. Press the Line key (flashing) to retrieve a call.
3. Retrieve the Held Call.

- In Hold Fixation Mode;

1. When an Automatic Recall Timer expires, the Desktop terminal which put the call on hold is recalled.
2. Press the HOLD key on a Desktop terminal, or lift the handset.
3. Retrieve the Held Call.

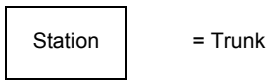
Service Conditions

1. Any Desktop terminal displaying the Held Line can seize the Held Call.
2. AUTOMATIC RECALL [A-8] is directed only to the Station that placed the call on Hold.
3. The Station initiating HOLD will receive a distinctive I-HOLD INDICATION [I-7D].
4. A two-party Hold is available when a Desktop terminal is engaged in a Three-party Conference with a Station and/or Trunk. An EXECUTIVE RIGHT-OF-WAY [E-1] Call cannot activate a two-party Hold.
5. When two parties are put on Hold, features such as EXECUTIVE RIGHT-OF-WAY [E-1], ATTENDANT OVERRIDE [A-7], CALL WAITING - ORIGINATING [C-31], CALL WAITING - TERMINATING [C-12], BUSY VERIFICATION [B-3] and ATTENDANT CAMP-ON [A-1], cannot be activated to the busy line.
6. The two held parties remain connected and can talk to each other. MUSIC ON HOLD [M-7] is not activated.

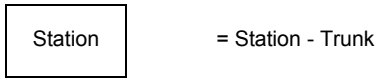
7. The two held parties cannot be placed on EXCLUSIVE HOLD [E-3D] by the Desktop terminal user.
8. If one of the two held parties is released from the Connection, MUSIC ON HOLD [M-7] will be activated on the remaining Connection. When the holding party returns to the Connection, a two-party Connection is established.
9. If the Sub Line Appearance placed in the NON-EXCLUSIVE HOLD condition is assigned to a Single Line Telephone, the Single Line Telephone can retrieve the Held Call by lifting the handset.
10. This feature operates when the Desktop terminal Station has been put in CONSULTATION HOLD by the other party.
11. This feature is available when a Station is in the following states, where "=" means a path and "-" means music on hold:
    a.) when a Station has a path to another Station:



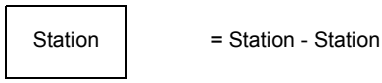
    b.) when a Station has a path to a Trunk:



    c.) when a Station has a path to a Station that has a Trunk held by MUSIC ON HOLD [M-7] for CONSULTATION HOLD - ALL CALLS [C-17]:



    d.) when a Station has a path to a Station that has a Station held by MUSIC ON HOLD [M-7] for CONSULTATION HOLD - ALL CALLS [C-17]:



    e.) when a Station has a path to two Trunks at one time for THREE-WAY CALLING [T-2]:



    f.) when a Station has a path to a Trunk and a Station at one time for THREE-WAY CALLING [T-2]:



12. This feature is not available when a Station is in the following states:
    a.) when a Station has a path to a Trunk and a Trunk held by MUSIC ON HOLD [M-7] for CONSULTATION HOLD - ALL CALLS [C-17]:



    b.) when a Station has a path to two Stations at one time for THREE-WAY CALLING [T-2]:



13. This service is available when a call with a station or a trunk is in progress; a call cannot be placed on HOLD while it is ringing, or talking with a console.
14. When both CALL PARK GROUP - D [C-176] and this service are assigned, CALL PARK GROUP - D has the higher priority.

15. When the held party is an IP terminal and the connection with the party is disconnected due to the accidents such as network failure or unplugging of cables, the lamp of the Line key on the Desktop terminal that held the call keeps flashing. If a user of this Desktop terminal tries to retrieve the call in this condition, dial tone is heard and the lamp of the Line key will stop flashing and turn on.

- Conditions for Hold Fixation Mode;

1. If a call terminates at a Sub Line while My Line is using this service, the call cannot be answered; likewise, if a call terminates at My Line while a Sub Line is using this service, the call cannot be answered.
2. When a station puts a call on HOLD using this service, it cannot perform any other operations than answering the Held Call (pressing the HOLD key on the Desktop terminal or lifting the handset).
3. This service is available with conference call (8/16/32 parties), but not with THREE-WAY CALLING [T-2].
4. It takes about a second from lifting the handset to seizing the call.
5. CALL HOLD [C-6] and this service can be used at the same time.
6. EXCLUSIVE HOLD-D [E-4D] and this service cannot be used at the same time.
7. When collaborating with softphones, this service is not available.
8. To assign this service while the system is in operation, perform the following operations after data setting.
    - IP Terminals: Re-registration
    - Legacy Terminals: Initialization of ELC circuit cards
9. This service is available at the following terminals.

IP Station		Digital Terminal
IP Terminal	SIP Terminal	
DtermIP (Proprietary Protocol)	DtermIP (SIP)	Dterm Series *i*
	DT800/DT700 Series	DT400/DT300 Series

10. One of the ELC cards listed below is required for this feature.

ELC Card	Firmware Requirements
CH-16ELCA	SP-4035 16ELCA PROG-A Version 1 or later.
PA-16ELCN	SP-3956 16ELCN PROG-A Version 1 or later.
PA-16ELCN-A	SP-4013 16ELCN-A PROG-A Version 1 or later.

Programming

STEP 1: **ASYD** - System Data 1, Index 144. Recall timer for NON-EXCLUSIVE HOLD. Assign data 00H for 30 seconds. ROM default data is 3FH.

System Data 2, Index 12, Bit 0. Is Automatic Recall Ringing for NON-EXCLUSIVE HOLD provided on a per Tenant basis? 0/1: Yes/No.

- In Hold Fixation Mode;

STEP 1: **ASFC** (Service Feature Restriction Class Data) - SFC: 0-15, SFI=254 (Hold Fixation Mode), RES=1 (Telephone call by the hold key during holding.)

STEP 2: **ASYD** - System Data 1, Index 64, Bit 7. Music-on-Hold Service. 0/1 = Out of service/In service.

System Data 1, Index 144. Recall timer for NON-EXCLUSIVE HOLD. Assign data 00H for 30 seconds. ROM default data is 3FH.

System Data 2, Index 12, Bit 0. Is Automatic Recall Ringing for NON-EXCLUSIVE HOLD provided on a per Tenant basis? 0/1: Yes/No.=1.

N-8D Non-square Line Assignment -D

General Description

This feature allows the specific keys on the Desktop terminal set to be freely assigned as Line keys or as Feature keys. These assignments are done on a per-Station basis.

Operating Procedure

Key designations are assigned via the PCPro.

Service Conditions

Programming one feature to more than one key will not provide enhanced service. For example, two MSG keys will not provide a Dterm set with eight message memories. See MESSAGE REMINDER [M-19].

Programming

STEP 1: **ASDT** - LENS equipped with a Desktop terminal Circuit Board must be assigned a Station Number and a Telephone Equipment Class (TEC) of 12.

STEP 2: **AKYD** - Assign a MY Line, a Prime Line, and various Sub Lines to the Dterms.

→TN: 1

→STN: Station Number

TP: Type of Dterm 0/1:

→RING: Assign ringing to the entire phone

0: No Ringing, Day or Night.
1: Ring Day.
2: Ring Night.
3: Not Used.
4: Ring Day & Night, 30 second delay.
5: Ring Day & Night, 20 second delay.
6: Ring Day & Night, 10 second delay.
7: Ring Day & Night, no delay.

PL TN STN:Prime Line Tenant and Station Number

KYN: Key Number (1-40)

KYI: Service Index

0: Key Not Used
1: Feature Key
→ 2: Multi Line Key

KD: Not assigned for Feature Key

TN STN:Tenant Number and Station Number of Multi Line

RG: Ringing Assignment per Multi Line Appearance
0: No Ringing, Day or Night.
1: Ring Day.
2: Ring Night.
3: Not Used.
4: Ring Day & Night, 30 second delay.
5: Ring Day & Night, 20 second delay.
6: Ring Day & Night, 10 second delay.
7: Ring Day & Night, no delay.

Any combination of available System Station Numbers may be assigned to the keys of the Desktop terminal instruments. There are no restrictions on order of Multi Line Appearance. Line Appearances may be arranged in any fashion, provided they are located within the same Module Group.

N-17 Night Connection Outside - System

General Description

This is a night transfer service on a system basis, enabling a ring-down or LDN Call to be transferred to a preselected Station Outside the System when night mode has been set.

Operating Procedure

1. The Attendant sets night mode. (Night key, PB, Handset Extraction, MB).
2. A ring-down or LDN Call terminates from a Trunk.
3. The call is transferred to the Outside Party in accordance with NIGHT CONNECTION OUTSIDE - System Data.

Service Conditions

1. The priority of this service is higher than TRUNK ANSWER FROM ANY Station (TAS) [T-8] service but lower than DIRECT-IN TERMINATION (DIT) [D-7] and NIGHT CONNECTION - FIXED [N-1] services. Thus, if DIT [D-7] and NIGHT CONNECTION - FIXED [N-1] are set, this service is not activated.
2. If the terminating side is a ring-down Trunk, a Release Signal cannot be received after communication is established. Therefore, the Trunk to which the call is transferred must be capable of sending back a Release Signal.
3. Tandem restriction is applied to this feature.
4. This service cannot be used in combination with CONSECUTIVE SPEED CALLING [C-62] service.
5. The Call Transfer Destination can be fixed on a per Tenant basis.

FCCS Conditions

1. This feature is available via FCCS. An Incoming Trunk Call (Ring-down or LDN Call) can be transferred to a preselected Outside Party via FCCS link.
2. This service is available for the Incoming Trunk Call (Ring-down or LDN Call).
3. The feature data via DM, LDM and NDM can coexist in a system. However, data registered on NDM has top priority. Data registered on LDM has precedence over data registered on DM. (Priority: NDM → LDM → DM)
4. When using this feature via FCCS link, only one Call Transfer destination can be registered, in a system.

Programming

STEP 1: **ASYD** - System Data 1, Index 77, Bit 0, DAY/NIGHT mode Changeover via handset jack 0/1: Not Required/Required.

STEP 2: **ALDN** - Assign LDN, allow night transfer, assign TYPE = 3, and the ADC number assigned in ASPD.

STEP 3: **ASPD** - Assign Abbreviated Digit Code (ADC) and Access Code to the dialed (000-999). This step is required only if the Access Code and Outside Number are greater than eight digits.

STEP 4: **ACFO** - Assign Call Forwarding Data, CF = 4: Night Connection, and the ADC assigned in ASPD.

STEP 5: **ARRC** - Allowing Incoming Route to outgoing route. Use Alternative Route Index (ARI-D-RES) for Direct Connection. Three possible restrictions (RES) may be assigned: data "0" = Connection is restricted; data "1" = Connection is allowed; or data "2" = toll restriction is required. For NIGHT CONNECTION-OUTSIDE-SYSTEM, toll restriction is not required.

STEP 6: **ACFR** - Assign Transfer Service Feature Index (TSFI) = 2: Direct In Termination Night Only, to be allowed CCI: 1 (LDN). This allows LDN to go to Night transfer destination. Assign TSFI: 5 (TAS) to be allowed CCI: 1 (LDN). This allows LDN to go to TRUNK ANSWER ANY STATION (TAS).

FCCS Programming

This feature is available via FCCS link. Data assignment shown below is required.

STEP 1: **ASYD** - System Data 1, Index 77, Bit 0, DAY/NIGHT mode Changeover via handset jack 0/1: Not Required/Required.

STEP 2: **ALDNN**- Assign LDN, allow night transfer, assign TYPE = 3, and the ADC number assigned in ASPD.

STEP 3: **ASPD** - Assign Abbreviated Digit Code (ADC) and Access Code to the dialed (000-999). This step is required only if the Access Code and Outside Number are greater than eight digits.

STEP 4: **ACFOL/ACFON** - User Group Number, UGN = 1 (Fixed), Call Forwarding Service, CF = 4 (Night Connection Outside), Abbreviated Digit Codes, ADC: Assign the abbreviated digit codes (ADC) and the speed dial number that are assigned by the ASPD command.

STEP 5: **ARRCN** - Allowing Incoming Route to outgoing route. Use Alternative Route Index (ARI-D-RES) for Direct Connection. Three possible restrictions (RES) may be assigned: data “0” = Connection is restricted; data “1” = Connection is allowed; or data “2” = toll restriction is required. For NIGHT CONNECTION-OUTSIDE-SYSTEM, toll restriction is not required.

STEP 6: **ACFR** - Assign Transfer Service Feature Index (TSFI) = 2: Direct In Termination Night Only, to be allowed CCI: 1 (LDN). This allows LDN to go to Night transfer destination. Assign TSFI: 5 (TAS) to be allowed CCI: 1 (LDN). This allows LDN to go to TRUNK ANSWER ANY STATION (TAS).

N-28 Name Display - System

General Description

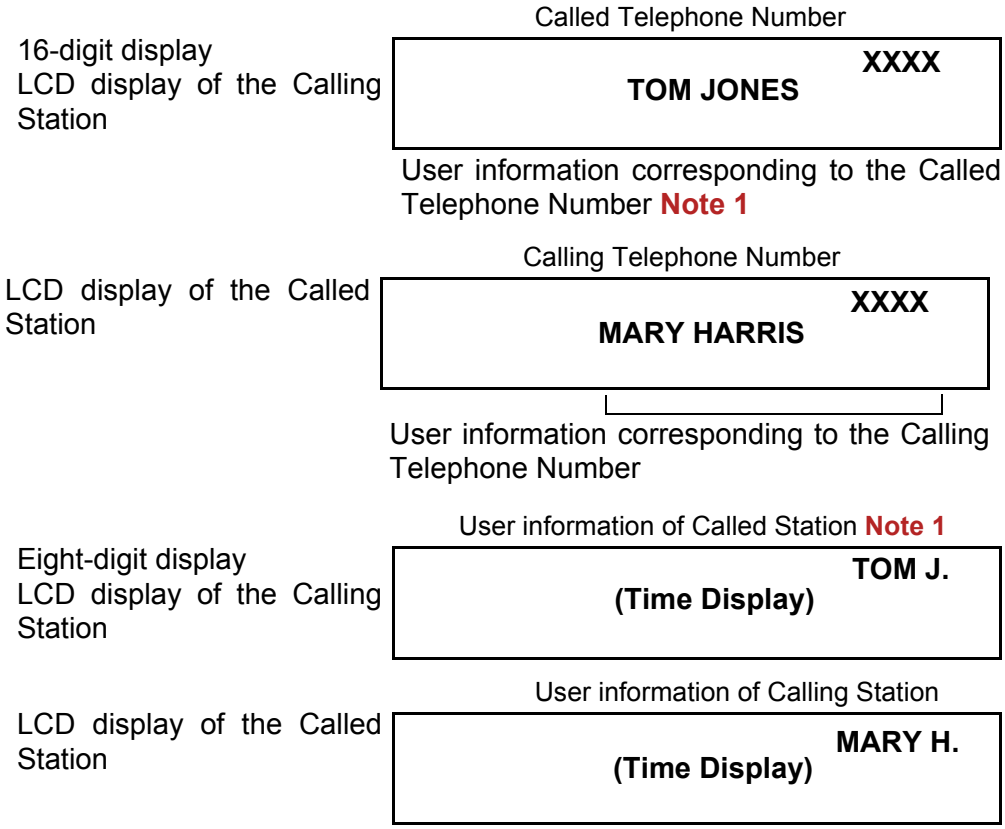
This feature allows a Desktop terminal with LCD display to indicate the user information corresponding to the Calling Telephone Number while engaged in STATION-TO-STATION CALLING [S-11].

Operating Procedure

Examples of LCD display are shown below.

Note: These examples below are only for Desktop Terminals and DESKCON.

1. When originating a Station-to-Station Call:

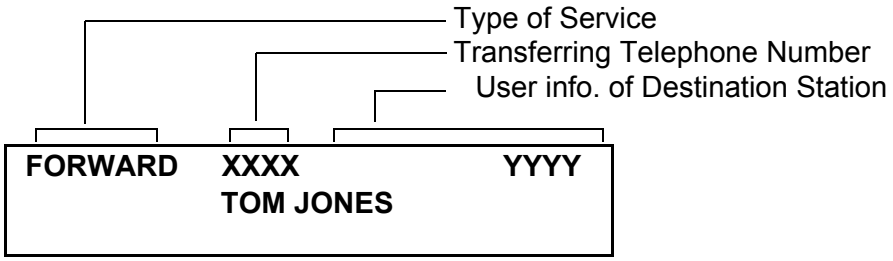


Note: The user information on the LCD flashes until the call is answered.

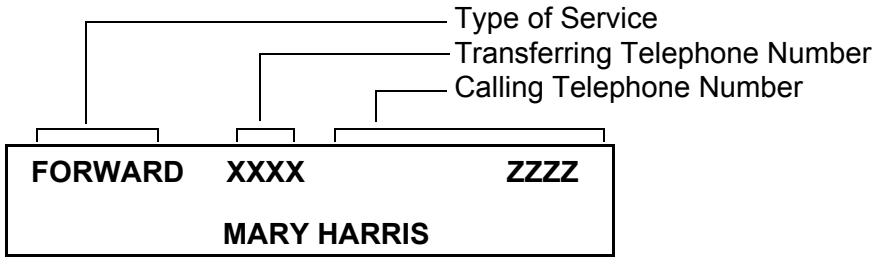
Note 1: When the called party is busy or restricting incoming calls, the called user name is generally not displayed on the calling party's LCD.

2. When a Station Call has been forwarded to another Station by CALL FORWARDING - ALL CALLS [C-5]/ BUSY LINE [C-2]/DON'T ANSWER [C-3]:

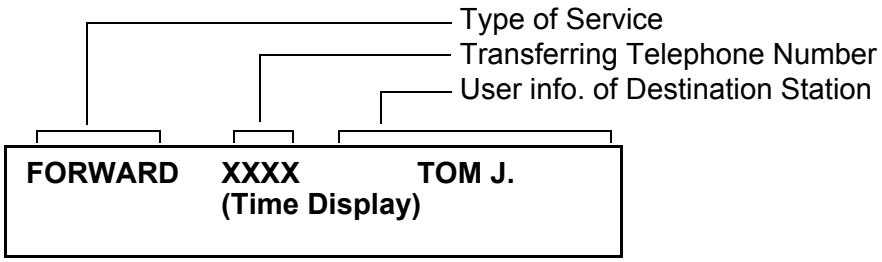
16-digit display
LCD display of the Calling Station



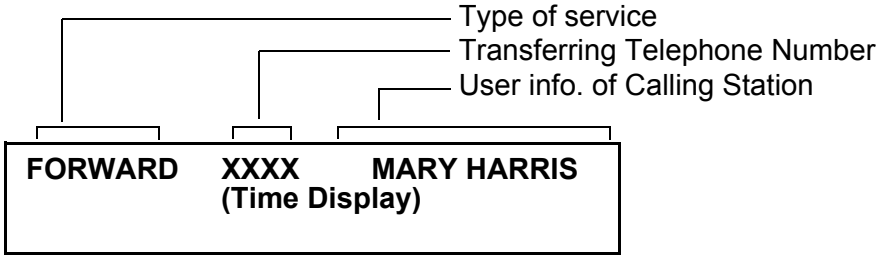
LCD display of the Destination Station



Eight-digit display
LCD display of the Calling Station



LCD display of the Destination Station



Note: The user information on the LCD flashes until the call is answered.

3. When both the Calling/Called Station are released from a Station-to-Station Call, the LCD of the Calling/ Called Station displays:



Service Conditions

1. This service is available when either of the Calling/Called Station is a Desktop terminal, with an LCD display.
2. By this service, ASCII character data can be displayed.
3. User information corresponding to the Telephone Number that can be displayed on the LCD is a maximum of 16 digits. The information displayed may consist of upper case letters, numeric characters, and most keyboard symbols.
4. Do not use both NAME DISPLAY assigned by the ANDD command and the one assigned by OAI Service (FLF) at same time. When NAME DISPLAY is assigned by the OAI Service (FLF), NAME DISPLAY using the ANDD command cannot be assigned.
5. Eight-digit display or 16-digit display is selectable by System Data, but cannot be used concurrently.
6. 16-digit display:
   a.) When ASYD, System Data 1, Index 390, Bit 2 is 0, the following occurs:

   - User information is displayed for My Line only.
   - User information cannot be displayed for Multi Line that specifies an identical Station Number for the Prime Line.

   b.) User information is displayed on the middle line of LCD, beginning on the 9th segment from the left. When the information is less than 16 digits, the unused segments are left blank. The information cannot be displayed on the upper line of LCD.
7. Eight-digit display of Desktop terminal:
   a.) User information is displayed on the upper line of LCD, beginning on the 17th segment from the left. When the information is less than eight digits, the unused segments are left blank.
8. If the Station has no name programmed for Name Display Service, the Telephone Number is repeated in the name field.
9. User information will not be guaranteed when 16-digit display service is changed to 8-digit display service.
10. When Telephone Number has been changed to a new one, the Telephone Number and user information to be displayed cannot be guaranteed. Therefore, System Data assignment must be properly managed.
11. User information may be assigned to all Sub Lines of a Desktop terminal.
12. The Name Display data stored on the hard disk is loaded to the memory at the time of system initialization (including power on) or when the Office Data load initialization is requested by PCPro command or the keys on the TOPU.
13. The Name Display data is backed up to the hard disk at the time of routine diagnosis.
14. If there is a difference between the contact name registered on the Phonebook of the Desktop terminal and the user name registered on the system by the ANDD command, the contact name on the Desktop terminal will be prioritized to display.
15. When the number of digits displayed in the Name Display Service is set to 16 digits by ASYD, System Data 1, Index 241, b1=1, the unused segments between the end of the name information to the 16th digit will be complemented with spaces if the name information registered on the system by the ANDD command is less than 16 digits.
16. When the number of digits displayed in the Name Display Service is set to 16 digits by ASYD, System Data 1, Index 241, b1=1 and the Calling number has priority for Caller ID display pattern on the LCD by ASYD, System Data 1, Index 390, b0=1, the name information with an asterisk (*) indicated on the 8th digit position will be displayed.

17. When Calling Name has priority to be displayed on the LCD (set by ASYD, System Data 1, Index 390, b0=0), and the Telephone Number has more than eight digits, the LCD displays Calling Number and intermediate number sequentially. To avoid this, set the data as follows;

    - Make the Telephone Number seven digits.
    - Do not display the intermediate number.

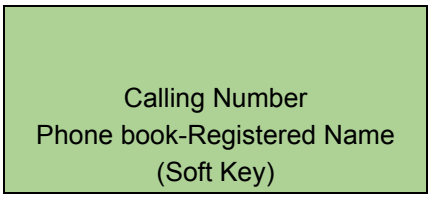
18. When the telephone number of outgoing call, incoming call, and ongoing call have been registered to the Phonebook of the Desktop terminal [DT820/DT750/DT830/DT730/DT830G/DT730G], the contact name registered to the Phonebook will be displayed. If the number has not been registered to the Phonebook of the Desktop terminal, or if the Desktop terminal does not have a Phonebook [DT710], the information about number and name provided by the system is displayed on the same line of the LCD where registered information is displayed.

    (a) The opposite party name can be displayed in the following four patterns by setting the system data [ASYDL - Sys 1, Index 673].

    Note: The DT820/DT710 that uses a three-lines in LCD including one line for the Soft key displays the name information in one line higher than that of the DT750/DT830/DT730/DT830G/DT730G.

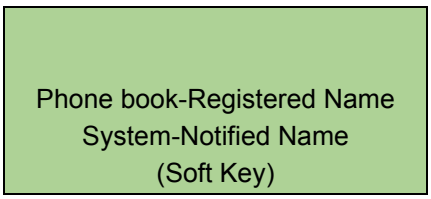
    - Pattern 1: ASYDL - Sys 1, Index 673, Bit 1=0, and Bit 0=0.

      The contact name registered to the Desktop terminal Phonebook is displayed on the third line on the LCD of the DT750/DT830/DT730/DT830G/DT730G and on the second line on the LCD of the DT820/DT710.

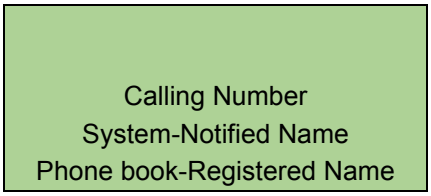
    - Pattern 2: ASYDL - Sys 1, Index 673, Bit 1=0 and Bit 0=1.

      The contact name registered to the Desktop terminal Phonebook is displayed on the second line on the LCD of the DT750/DT830/DT730/DT830G/DT730G and on the first-line on the LCD of the DT820/DT710.

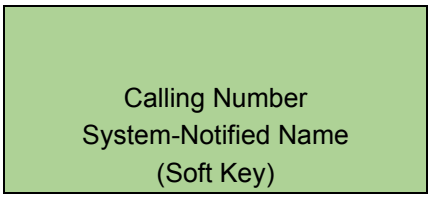
- Pattern 3: ASYDL - Sys 1, Index 673, Bit 1=1 and Bit 0=0.

  The contact name registered to the Desktop terminal Phonebook is displayed on the fourth line on the LCD of the DT750/DT830/DT730/DT830G/DT730G and on the third line on the LCD of the DT820/DT710.



- Pattern 4: ASYDL - Sys 1, Index 673, Bit 1=1 and Bit 0=1.

  The contact name registered to the Desktop terminal Phonebook is not displayed.



(b) Contact name registered to the Terminal Phonebook is preferentially displayed over the user name retrieved from the system. However, in rare cases, if there is a discrepancy between user name from the system and contact name in the Terminal Phonebook, the user name may be briefly shown prior to the display of the contact name.

(c) When the pattern 3 is set to display the name information, the soft key cannot be used regardless of the Desktop terminal state.

(d) The maximum digits to display the system-provided name information is as follows.

- Opposite Party Number: Maximum 24 digits

  Note: Depending on the setting of the system data [ASYD - Sys 1, Index 241, Bit 6], either the beginning or the end of the exceeding digits will be deleted.

- Opposite Party Name: Maximum 24 characters (single-byte)/12 characters (two-byte)

(e) When Subline is set as Prime Line, Calling Number may differ from Name (e.g. having an incoming call on Subline while receiving a call on My Line). Set My Line to the Prime Line.

(f) The opposite party information provided by the system is displayed on other lines than the Phonebook-registered name (contact name) described on (a) is displayed. As the system gives this information to My Line only, not to Subline, My Line and Subline display different information on LCD.

When a call arrives, the caller information is provided for My line. When a call is answered, the caller information is provided for the line that answers the call. Depending on the data setting, two kinds of name information may be displayed, or the number may not be displayed at all.

The following are the display patterns for each data setting for DT750/DT830/DT730/DT830G/DT730G.

Note: The system-provided name information means the [NAME] registered on the system Phonebook by the APBKN command, Name Display service by the ANDD command, and the [NAME] registered on the ACID command.

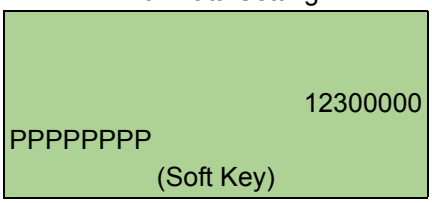
[Incoming Trunk Call]

- When the caller number and caller name are retrieved from the system.

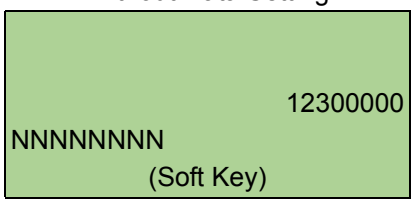
  PPPPPPPP: Contact name registered to the Terminal Phonebook
  NNNNNNNN: Data assigned with name display service (ANDD)

  Pattern 1: ASYDL - Sys 1, Index 673, Bit 1=0 and Bit 0=0. [The contact name of the Terminal Phonebook is displayed on the third line on the LCD of the DT750/DT830/DT730/DT830G/DT730G]

  With Data Setting

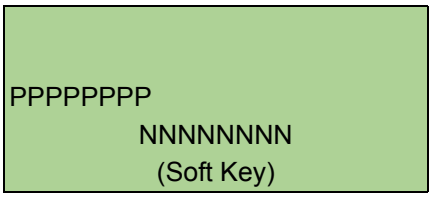
  

  Without Data Setting

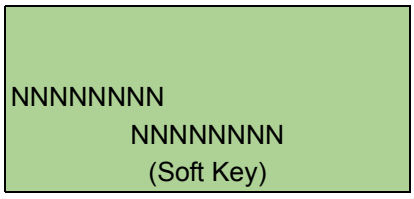
  

  Pattern 2: ASYDL - Sys 1, Index 673, Bit 1=0 and Bit 0=1. [The contact name of the Terminal Phonebook is displayed on the second line on the LCD of the DT750/DT830/DT730/DT830G/DT730G]

  With Data Setting

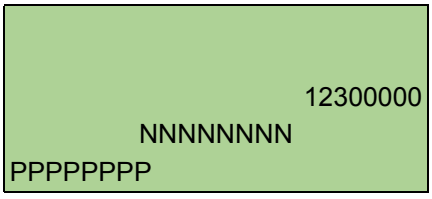
  

  Without Data Setting

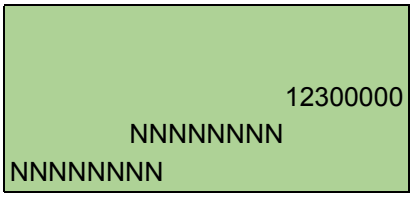
  

  Pattern 3: ASYDL - Sys 1, Index 673, Bit 1=1 and Bit 0=0. [The contact name of the Terminal Phonebook is displayed on the fourth line on the LCD of the DT750/DT830/DT730/DT830G/DT730G]

  With Data Setting

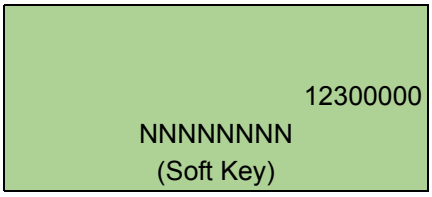
  

  Without Data Setting

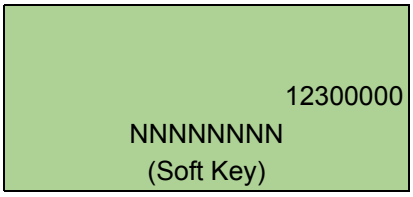
  

  Pattern 4: ASYDL - Sys 1, Index 673, Bit 1=1 and Bit 0=1. [The contact name of the Terminal Phonebook is not displayed.

With Data Setting



Without Data Setting

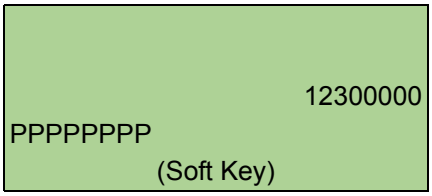


- When only the caller number is retrieved from the system.

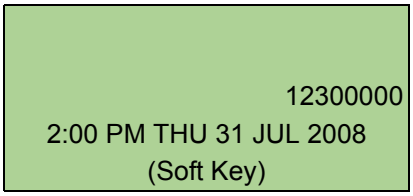
  PPPPPPPP: Contact name registered to the Terminal Phonebook
  NNNNNNNN: Data assigned with name display service (ANDD)

  Pattern 1: ASYDL - Sys 1, Index 673, Bit 1=0 and Bit 0=0. [The contact name of the Terminal Phonebook is displayed on the third line on the LCD of the DT750/DT830/DT730/ DT830G/DT730G]

With Data Setting

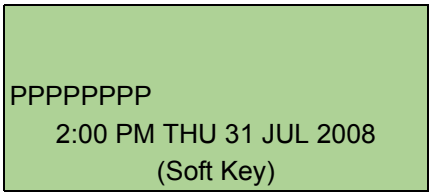


Without Data Setting

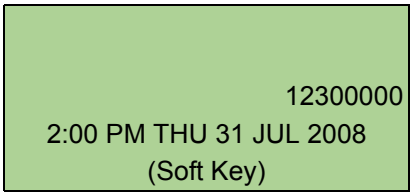


Pattern 2: ASYDL - Sys 1, Index 673, Bit 1=0 and Bit 0=1. [The contact name of the Terminal Phonebook is displayed on the second line on the LCD of the DT750/DT730/DT730G]

With Data Setting

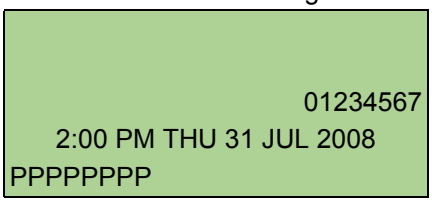


Without Data Setting

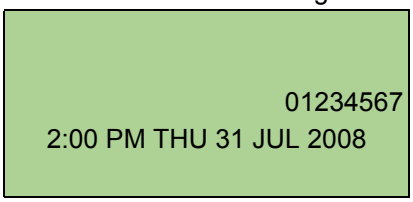


Pattern 3: ASYDL - Sys 1, Index 673, Bit 1=1 and Bit 0=0. [The contact name of the Terminal Phonebook is displayed on the fourth line on the LCD of the DT750/DT830/DT730/ DT830G/DT730G]

With Data Setting

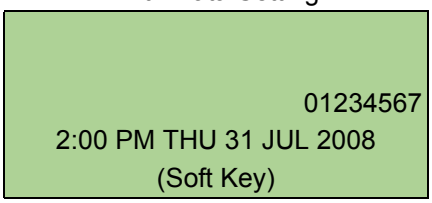


Without Data Setting

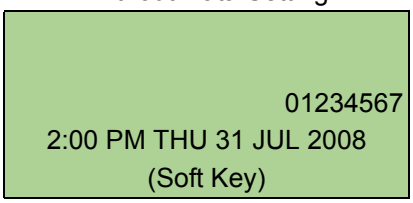


Pattern 4: ASYDL - Sys 1, Index 673, Bit 1=1 and Bit 0=1. [The contact name of the Terminal Phonebook is not displayed.]

With Data Setting



Without Data Setting



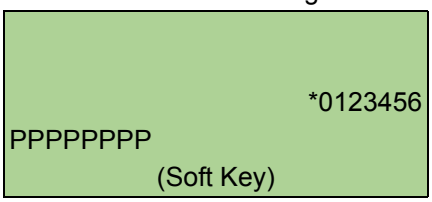
<Incoming Trunk Call from Outside>

- When the caller number and caller name are retrieved from the system with the system data [ASYD - Index 390, Bit 0=0].

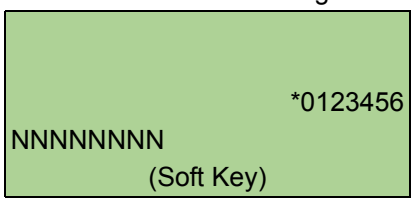
  PPPPPPPP: Contact name registered to the Terminal Phonebook
  NNNNNNNN: Caller ID (ACID)

  Pattern 1: ASYDL - Sys 1, Index 673, Bit 1=0 and Bit 0=0. [The contact name of the Terminal Phonebook is displayed on the third line on the LCD of the DT750/DT830/DT730/DT830G/DT730G]

With Data Setting

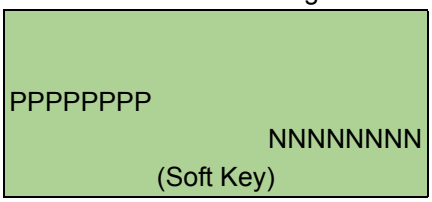


Without Data Setting

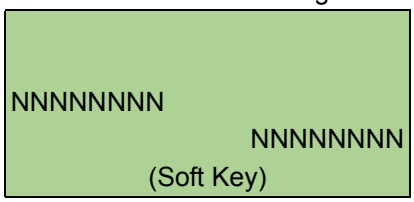


Pattern 2: ASYDL - Sys 1, Index 673, Bit 1=0 and Bit 0=1. [The contact name of the Terminal Phonebook is displayed on the second line on the LCD of the DT750/DT830/DT730/DT830G/DT730G]

With Data Setting

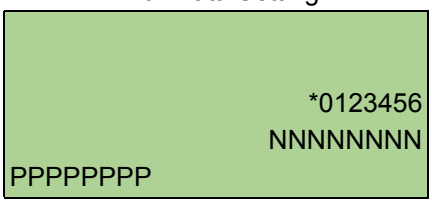


Without Data Setting

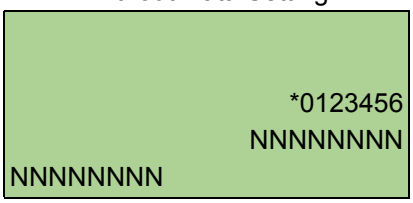


Pattern 3: ASYDL - Sys 1, Index 673, Bit 1=1 and Bit 0=0. [The contact name of the Terminal Phonebook is displayed on the fourth line on the LCD of the DT750/DT830/DT730/DT830G/DT730G]

With Data Setting

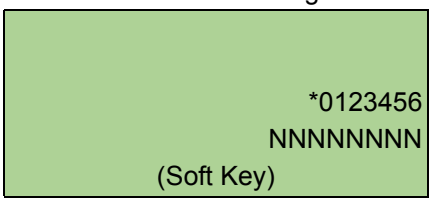


Without Data Setting

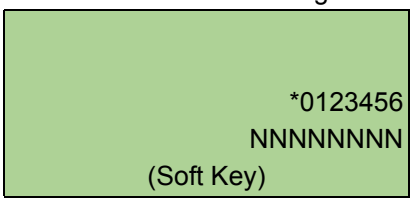


Pattern 4: ASYDL - Sys 1, Index 673, Bit 1=1 and Bit 0=1. [The contact name of the Terminal Phonebook is not displayed.]

With Data Setting



Without Data Setting

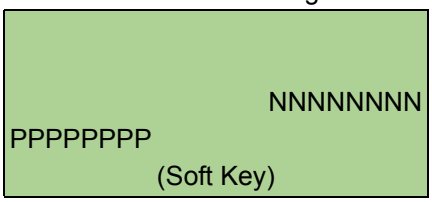


- When the caller number and caller name are retrieved from the system with the system data [ASYD - Index 390, Bit 0=1].

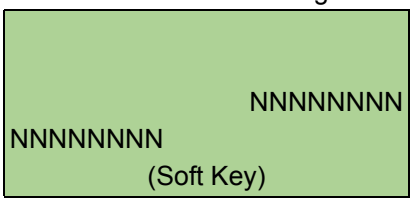
PPPPPPPP: Contact name registered on the Terminal Phonebook
NNNNNNNN: Caller ID (ACID)

Pattern 1: ASYDL - Sys 1, Index 673, Bit 1=0 and Bit 0=0. [The contact name of the Terminal Phonebook is displayed on the third line on the LCD of the DT750/DT830/DT730/DT830G/DT730G]

With Data Setting

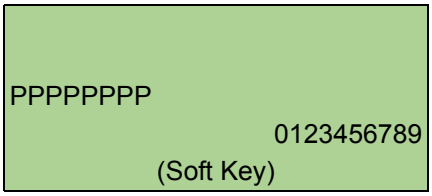


Without Data Setting

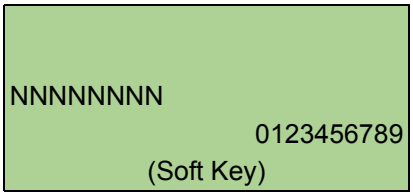


Pattern 2: ASYDL - Sys 1, Index 673, Bit 1=0 and Bit 0=1. [The contact name of the Terminal Phonebook is displayed on the second line on the LCD of the DT750/DT830/DT730/DT830G/DT730G]

With Data Setting

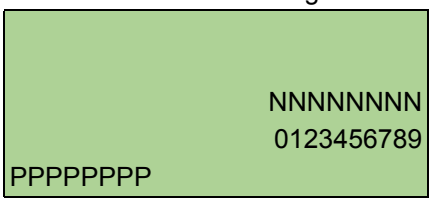


Without Data Setting

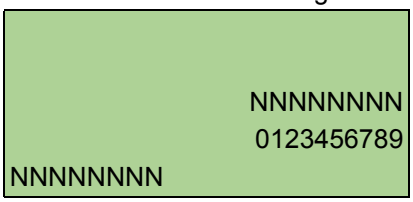


Pattern 3: ASYDL - Sys 1, Index 673, Bit 1=1 and Bit 0=0. [The contact name of the Terminal Phonebook is displayed on the fourth line on the LCD of the DT750/DT830/DT730/DT830G/DT730G]

With Data Setting

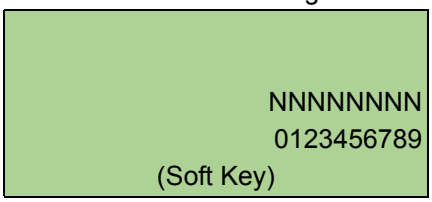


Without Data Setting

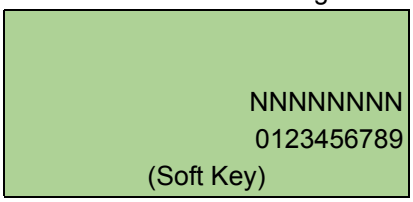


Pattern 4: ASYDL - Sys 1, Index 673, Bit 1=1 and Bit 0=1. [The contact name of the Terminal Phonebook is not displayed.]

With Data Setting



Without Data Setting

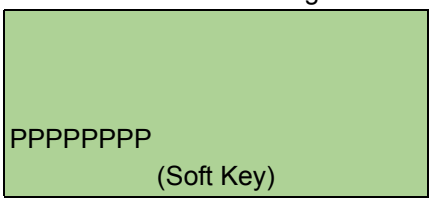


- When only the caller number is retrieved from the system with the system data [ASYD - Index 390, Bit 0=0/1].

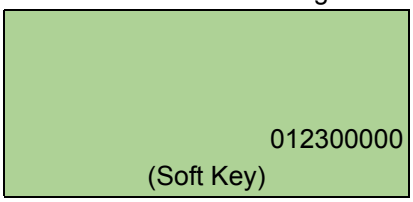
PPPPPPPP: Contact name registered on the Terminal Phonebook
NNNNNNNN: Caller ID (ACID)

Pattern 1: ASYDL - Sys 1, Index 673, Bit 1=0 and Bit 0=0. [The contact name of the Terminal Phonebook is displayed on the third line on the LCD of the DT750/DT830/DT730/DT830G/DT730G]

With Data Setting

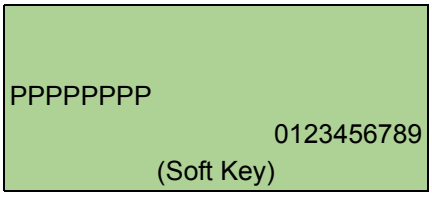


Without Data Setting

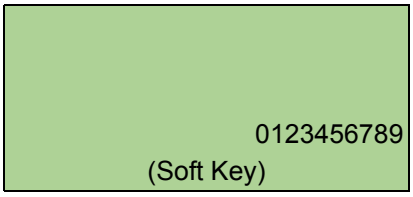


Pattern 2: ASYDL - Sys 1, Index 673, Bit 1=0 and Bit 0=1. [The contact name of the Terminal Phonebook is displayed on the second line on the LCD of the DT750/DT830/DT730/DT830G/DT730G]

With Data Setting

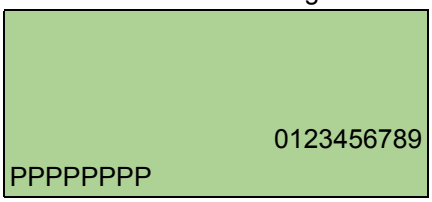


Without Data Setting

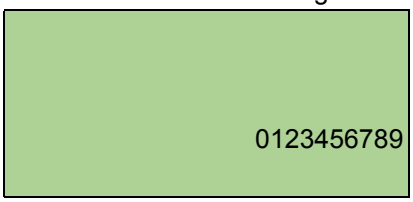


Pattern 3: ASYDL - Sys 1, Index 673, Bit 1=1 and Bit 0=0. [The contact name of the Terminal Phonebook is displayed on the fourth line on the LCD of the DT750/DT830/DT730/DT830G/DT730G]

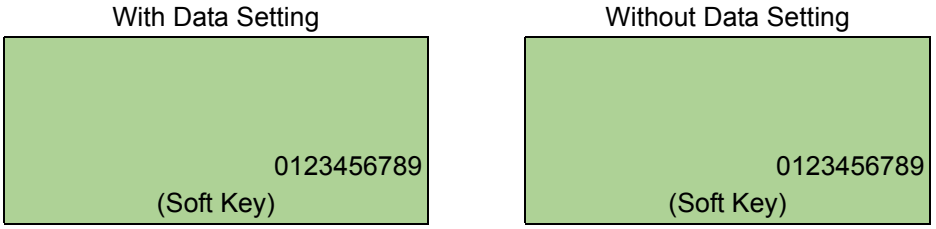
With Data Setting



Without Data Setting



Pattern 4: ASYDL - Sys 1, Index 673, Bit 1=1 and Bit 0=1. [The contact name of the Terminal Phonebook is not displayed.]



19. If a destination number is registered in the Terminal Phonebook, the registered name corresponding to the number is displayed while placing outgoing calls (hears Ringback Tone), receiving incoming calls (hears ring tone), or talking. When the status of a call while talking changes to any other status such as a call on hold, the name display is no longer available. However, in case the name display information sent from the system has been given at this time, the name is displayed.
20. When a station call coming directly from the calling party (not transferred) encounters a busy, no-answer condition due to call restriction, or lockout, the called party's name is generally NOT displayed on the calling party's LCD (**Note 2**); when a calling party manually transfers a station/trunk call, the name of the destination is displayed on its LCD.

Note 2: Despite this, some terminals (calling party) may display the called party's name.

21. When a call arrives at a SIP Handler Controlled SIP terminal which supports RFC3325 and RFC5876, Calling Name can be displayed on the SIP Handler Controlled SIP terminal (in the INVITE request message, Name Display information is given in the display-name field of the From header). However, when the connected party is changed by the transferring operation or shifting from a three-way call into a two-way call due to the disconnect of one of the three parties, the display of the connected party information depends on the standard SIP terminal. For the conditions of the display, refer to SIP HANDLER - SIP TERMINAL [S-167] of Data Programming Manual - Business.
22. Name information assigned with the NAME parameter of the ANDD command is displayed on the SIP Handler Controlled SIP terminal. Only Name Display information containing symbols and alphanumeric characters can be notified.

Programming

[For using Name Display Service]

STEP 1: **ASYD** - System Data 1, Index 78, Bit 3. Station Number with Name Display 0/1=No/Yes (only used with eight-digit display)
System Data 1, Index 78, Bit 5. Name Display Service
0/1: Out/In Service. Assign 1 (In Service).
System Data 1, Index 79, Bit 5. Name Display Enhance Service
0/1: Shows the name of the calling person on the answered phone/Shows time and date on the answered phone
System Data 1, Index 79, Bit 6, OAI Service 0/1: In Service/Out. Assign 0 (OAI is in service).
System Data 1, Index 241, Bit 1, The Number of digits displayed in Name Display Service 0/1: eight digits/16 digits
System Data 1, Index 390, Bit 2, Caller ID display on the bottom line of Desktop terminal LCD for Multiple Line
0/1: display on the bottom line (Multiple Line is not used) / display on the bottom line (Multiple Line is used)

Note: ASYD - System Data 1, Index 31. CM capacity is 04 or 06 (hexadecimal).

STEP 2: **ANDD** - Assign user's information to Station Number.

[For displaying the contact name registered to the Phonebook of DT800 Series/DT700 Series]

STEP 1: **ASYDL** - System Data 1, Index 673, Bit 0-1 (Display Location of User Information).

- In the case of DT750/DT830/DT730 Series

Bit 1	Bit 0	Description
0	0	Third Line (Recommended)
0	1	Second Line
1	0	Fourth Line
1	1	Not to display

- In the case of DT820/DT710

Bit 1	Bit 0	Description
0	0	Second Line (Recommended)
0	1	First Line
1	0	Third Line

Bit 1	Bit 0	Description
1	1	Not to display

[For using Name Display Service on the Soft Phone]

STEP 1: **ASYD** - System Data 1, Index 390, Bit 0=1 (Calling Number has priority on the Soft Phone Display)
STEP 2: **ASYDL** - System Data 1, Index 673, Bit 0-1 (Display Location of User Information)

Bit 1	Bit 0	Description
1	0	Third Line
1	1	Not to display

Note: Note the following to use Name Display Service on the Soft Phone.
- Name Display information is not stored in the call history.
- Only Number Display information is stored in the call history. If a calling number registered in the Soft Phone directory is received, calling name corresponding to the received number appears on the display.
- Name Display information consists of only numerics,"*" and "#" is not supported.
- When a calling number sent from the system as a display information exceeds 16 digits, the 17th or later characters are not stored in the incoming call history.

N-41 Network Night Switching

General Description

This feature allows the Attendant Console in a Main node to operate not only station calls from the intra - offices but also calls from all nodes belonging to Main node as a Centralized Attendant.
The Attendant Console in a Hub node is normally invalid (Night mode). When backup is required because of line cutting, etc., it may be valid (Day mode) and may operate all incoming calls terminating to Attendant Consoles within the Hub node including its own office as a Centralized Attendant there.

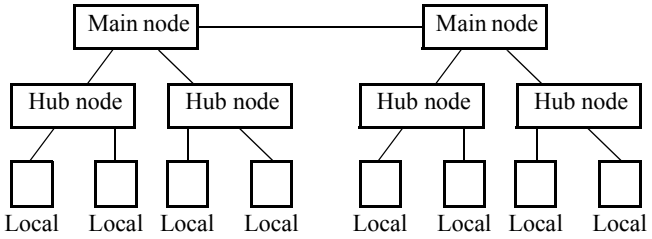
Note: This feature is available in Australia only.

Operating Procedure

Refer to "How to Operate Attendant Console".

Service Conditions

1. Network configuration is shown below:



2. Working modes of each node are as below:
   a.) Main Day mode
   b.) Hub Night/Day mode
   c.) Local Night mode
3. Usually all incoming calls like ATND/LDN/CF calls, to Attendant Consoles in each node are handled by the Attendant Console in a Main node. However, when the Attendant Console in a Hub node is in Day mode, intra-office ATND/LDN/CF calls and incoming calls from Local Nodes are handled by the Attendant Console not in the Main node but in the Hub node.

Node / Call	Hub node (Night mode)			Hub node (Day mode)			Incoming call
	Main	Hub	Local	Main	Hub	Local	
ATND	Main ATT	Main ATT	Main ATT	Main ATT	Hub ATT	Hub ATT	ATND
LDN	Main ATT	Main ATT	Main ATT	Main ATT	Hub ATT	Hub ATT	LDN
CF-ATT	Main ATT	Main ATT	Main ATT	Main ATT	Hub ATT	Hub ATT	ATND

4. All incoming calls to the Attendant Console in a Local and Hub node are usually provided with above conditions.

Interactions

When the emergency mode is set, incoming calls are transacted according to EMERGENCY SYSTEM CONTROL [E-22] service. For further details, refer to EMERGENCY SYSTEM CONTROL.

O-1 On-line Maintenance

General Description

This feature permits maintenance routines to be performed on the System without affecting normal system operation.

Operating Procedure

Refer to the “Command Manual” and “Operations and Maintenance Manual”.

Service Conditions

1. The following maintenance functions may be performed while the system is ON-LINE:
   a.) Generic Program and Operating Memory testing
   b.) Fault Contents Display and Diagnosis
   c.) System/Office Data Entry and Change
   d.) Traffic Measurement
   e.) Testing facilities of Outside Trunks is provided as an option
   f.) PEG COUNT [P-7] display

Interactions

1. PCPro must first be provided. For details of PCPro, refer to PCPro Setup Guide.
2. Verify all cable Connections.

Programming

Refer to the “Command Manual” and “Operations and Maintenance Manual”.

O-2 Outgoing Trunk Queuing

General Description

This feature allows a station user to dial a specified Access Code or press Line/Feature key and enter a first-in, first-out queue, when encountering a Trunk busy signal. As soon as an OUTGOING TRUNK becomes available, stations in the queue will be called back on a first-come, first-served basis.

Operating Procedure

When LEAST COST ROUTING - 3/6-DIGIT [L-5] is not provided:

1. Dial the Trunk Access Code and receive Busy Tone.
2. Press the switch hook and receive Special Dial Tone.
3. Dial the OUTGOING TRUNK QUEUING (OGQ) Access Code and receive Service Set Tone.
4. Replace the handset.

When LEAST COST ROUTING - 3/6-DIGIT [L-5] is provided:

1. Dial the Trunk Access Code and receive the Dial Tone.
2. Dial the desired number and receive Busy Tone.
3. Press the switch hook and receive Special Dial Tone.
4. Dial the OGQ Access Code and receive Service Set Tone.
5. Replace the handset.

<The following procedures are applied for Desktop terminals only.>

When LEAST COST ROUTING - 3/6-DIGIT [L-5] is not provided:

1. Dial the Trunk Access Code. Receive Busy Tone.
2. Press the CALL BACK key. Receive Service Set Tone. The LCD displays:



3. Replace the handset.
4. When the Trunk becomes idle, the station will be rung.

When LEAST COST ROUTING - 3/6-DIGIT [L-5] is provided:

1. Dial the Trunk Access Code. Receive the Dial Tone.
2. Dial the desired number. Receive Busy Tone. The LCD displays:



3. Press the CALL BACK key. Receive Service Set Tone. The LCD displays:



4. Replace the handset.

5. When the Trunk becomes idle, the station will be rung. Once connected to the Trunk, the system automatically redials the number.

Service Conditions

1. The following table shows connection availability of this feature.

X: Service available N: Service not available

Calling Party	Outgoing Trunk (OGT)			Remarks
	COT Trunk	CCIS Trunk	ISDN Trunk	
Analog Terminal	X	X	X	
Attendant Console	N	N	N	
ISDN Terminal	N	N	X	
Desktop Terminal **Note 1**	X	X	X	

Note 1: This service is not available for Outgoing Calls from Sub Line.

2. After an OUTGOING TRUNK becomes available, the user's station will ring for 30 seconds. If not answered within that time, the station will automatically be dropped from the queue.
3. When this feature is used in conjunction with SPEED CALLING [S-3, 21, 23], or LEAST COST ROUTING [L-5], where a Sender is involved, the Sender will automatically dial out the Called Subscriber Number when the handset is lifted.
4. If the user wishes to remove himself from the queue prior to being rung, the specified Cancellation Code must be dialed.
5. Individual stations may only initiate one Outgoing Trunk queue at a time. Subsequent attempts will result in a Reorder Tone.
6. Stations may be restricted from using this feature in CLASS OF SERVICE - INDIVIDUAL [C-15].

Note: The Class Of Service for OFF-HOOK QUEUING [O-7] is different from the Class Of Service for OUTGOING TRUNK QUEUING [O-2]. Both services are mutually exclusive.

7. This feature is not available for an ATTENDANT CONSOLE [A-3].
8. The maximum number of simultaneous OGQs per system:

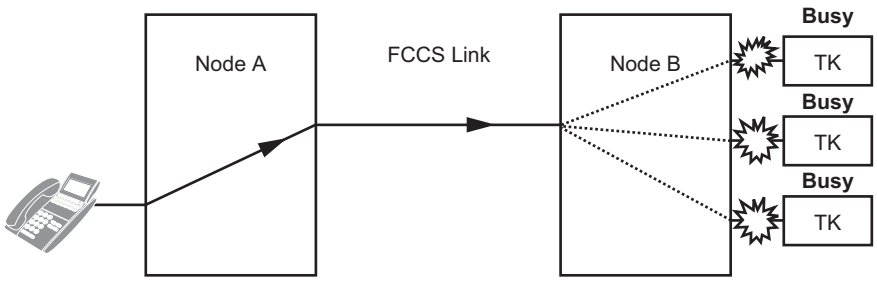
   1 IMG: 64 calls
   4 IMG: 256 calls

   The queue slots are used commonly by OFF-HOOK QUEUING [O-7], INTER-OFFICE OFF-HOOK QUEUING [I-11], and OGQ [O-2, 16].
9. CALL PICKUP - GROUP [C-7] cannot be used to answer a call directed to another station using OGQ.
10. OGQ is common to all Central Office Trunk Groups (COT, TIE, FX, WATS, etc.) in the system.
11. OGQ Call back will return to the originating station, not the CALL FORWARDING [C-2, 3, 5] terminating station.
12. OGQ is denied to a station Line when the CALL BACK [C-1] or CALL HOLD [C-6] feature is already activated.
13. The Attendant cannot CAMP-ON [A-1] a call to a station that has activated OGQ.
14. If a station user goes on-hook within six to eight seconds after encountering all Trunks busy, OUTGOING TRUNK QUEUING can be set automatically to the station.
15. The station is set for OGQ for a minimum of 30 seconds to one hour, maximum. This time is adjustable through the System Data. After this time, OGQ will be canceled automatically.

16. ACCOUNT CODE [A-18] and FORCED ACCOUNT CODE [F-7] information can be recorded on STATION MESSAGE DETAIL RECORDING (SMDR) [S-10], when used in conjunction with OGQ.
17. This feature can be activated when using a Trunk in the Remote Node via FCCS link.

Example: *Node A and Node B are in an FCCS network.*

- *Although a station belonging to Node A tried to call outside using a trunk belonging to Node B, all the trunks are busy. The call can be in the queue using this feature.*



18. The maximum number of the OGQs stations is 1,024 within the FCCS Network (NCN can manage a maximum of 1,024 calls).
19. OGQ data is transmitted to and stored in the Network Control Node (NCN). Therefore, all the OGQ data will be cleared by the system initialization of the NCN.
20. When the access to the NCN is impossible (system down or cable disconnection or any trouble has happened), the user cannot set or cancel OGQ data. (After dialing the Access Code or pressing the Feature keys for set/cancel OGQ, the user hears the Reorder Tone.)
21. When the Trunk becomes idle status after the NCN is recovered, the call in the queue will be called back, if data has not been cleared. If data has been cleared, the call is canceled automatically.

<The following conditions are applied for Desktop terminals only.>

1. To cancel the OUTGOING TRUNK QUEUING from a Desktop terminal, go off-hook and press CALL BACK key. The LCD displays:

OGQ CANCEL
(Time Display)

The user will hear Service Set Tone.

2. OUTGOING TRUNK QUEUING and CALL BACK [C-1] are mutually exclusive.
3. OUTGOING TRUNK QUEUING is only available from the MY Line of a Desktop terminal.

Interactions

1. Dial pad operation for Dterm sets may be used. A FLASH BUTTON [F-3D] must be programmed to a programmable Line/Feature key.
2. OGQ-AUTOMATIC may operate with either same or separate Access Codes. If the Automatic option is chosen, Service Set Tone will be sent when encountering an all Trunks busy situation. No dial pad operation will be required.
3. When testing OGQ, at least one Trunk of the Trunk Group being tested must be engaged in a station-to-Trunk or Trunk-to-Trunk Connection. Use of the MB key only on the Trunk circuit card will not allow OGQ to test and operate correctly.
4. Do not disable continuous Service Set Tone by changing System Data 1, Index 42, Bit 7 to data “1”. This changes “Continuous” Service Set Tone to “burst” Special Dial Tone, and Special Dial Tone to Burst Special Dial Tone, followed by Continuous Dial Tone. This will also disable TANDEM OFF-HOOK QUEUING.
5. This feature is not available for forwarding calls with Call Forwarding-Don't Answer-Outside [C-74].
6. This feature is not available for forwarding calls with Split Call Forwarding - All Calls (for trunk), Split Call Forwarding - Busy Line (for trunk), or Split Call Forwarding - Don't Answer (for trunk).
7. To operate this feature from a SIP Handler Controlled SIP terminal, perform the one digit dial hooking operation while hearing Busy Tone (BT) and dial the access code for this feature. To hear Busy Tone (BT), enable IP-DTG with the AUACL/AUACN command.

Programming

STEP 1: **ASYD** - System Data 1, Index 68, Bit 3. Enables automatic cancel timer, as set in System Data 1, Index 159. 0/1: -/Enable.

System Data 1, Index 159. OGQ release timer. Standard data of 00H for 30 minutes may be assigned. (RAM Data of 3C (hexadecimal) provides for 30 minutes).

System Data 2, Index 4, Bit 0. Should the Access Code for CALL BACK and OGQ be the same? 0/1: No/Yes. Usually assign data "1". Assign on a per Tenant basis.

System Data 2, Index 4, Bit 1. Enable Automatic OGQ? 0/1: No/Yes. Assign on a per Tenant basis. Automatic operation eliminates the need to access via dial pad.

System Data 2, Index 4, Bit 2. Enable OFF-HOOK OGQ for the Attendant Console? 0/1: No/Yes. Assign on a per Tenant basis. (See OUTGOING TRUNK QUEUING-ATTENDANT [O-14].)

System Data 1, Index 139, timer for the recall ringing.

STEP 2: **ANPD** - Reserve a Number Level for Service Feature Access and cancel. Select Normal Service (N), Hooking Service (H) or Busy Service (B) from the combo box for Connection Indexes. (applicable to any feature assigned to this Level). Assign NND in accordance with a predetermined Numbering Plan.

STEP 3: **ASPA** - If System Data 2, Index 4, Bit 0 is assigned for separate Access Codes, assign an Access Code to OGQ, SRV = "SSC: Service Code", SID = 19, and to OGQ cancel, SRV = "SSC: Service Code", SID = 20. Assign SID = 19 for a Connection Index of "B: Busy" and SID = 20 for a Connection Index of "N: Normal". If the Access Codes are the same, either the Access Codes for OGQ or the Access Codes for CALL BACK may be assigned.

STEP 4: **ASFC** - If Access Codes are separate, assign a Service Feature Restriction Class that allows SFI 1 (Outgoing Trunk Queuing) to stations that will have OGQ. If Access Codes are the same, assign an SFC that allows SFI 1, or SFI 1 and SFI 2.

STEP 5: **ARTD** - Routes (including the dummy route) to be used for OGQ must be flagged in CDN 9:TQ, Trunk Queuing Service. Always allow queuing to Route 31, the LCR (S) flag route.

Note: To change the data for existing route (RT), initialize or reboot the trunk device such as circuit card or MG.

STEP 6: **AKYD** - Desktop terminal - set the following data to assign CALL BACK key and a FLASH button.

KYI = 1: Function Key.
FKY = 5: For a CALL BACK key
FKY = 21: For a FLASH Button

STEP 7: **ADSL** - SN = 6 (Trunk is Busy) and FKY = 5 (Call Back; CB).

This command is used for assigning the function to each Soft key.

FCCS Programming

This data setting describes only the additional or replacement commands used to activate FCCS Services. When used as an FCCS feature, these commands replace the non-network commands (for example, AAED is replaced by AAEDN). Note that the data setting for FCCS Service varies depending on the Node.

Note: When using ASYDL or ASYDN to program an FCCS feature, the basic programming of ASYD is required in addition to ASYDL or ASYDN.

STEP 1: **ALRTN** - Assign this command to connect the Logical Route to the Physical Route.

STEP 2: **ASYDN** - These data must be assigned to the NCN.

System Data 1, Index 41:Designate the Trunk Queuing Override Timer

Index 68, Bit 3: Enables automatic cancel timer

0/1: -/Automatic cancel timer is Available

Index 159: Designate Automatic Cancel Timer

STEP 3: **ASPAN** - These data must be assigned to the NCN.

TN: Tenant Number

ACC: Access number

CI: "B: Busy" (for Entry), "N: Normal" (for Cancel)

SRV: "SSC: Service Code"

SID: 19 = OG Queuing-Entry

CI: "N: Normal"

SID: 20 = OG Queuing-Cancel

STEP 4: **ANPDN** - These data must be assigned to the NCN.

TN: Tenant

1stDC: 1st Digit

CI: Select Normal Service (N), Hooking Service (H), or Busy Service (B) from the combo box for Connection Indexes.

NND: Number of Necessary Digits

BLF: Busy Lamp Field Display Activated on ATT

STEP 5: **ARTDN** - These data must be assigned to the NCN.

LGRT: Logical Route Number

Assign CDN 9:TQ, Trunk Queuing Service = 1

Note: The current data should also be set in addition to the above-mentioned commands.
The command to be set to the Node that accommodates the Trunk: ARTD
The commands to be set to all the Nodes: ASFC, AKYD
For Desktop terminals, the command to be set to all the Nodes are ASFC, AKYD, ASPA.

Note: To change the data for existing route (RT), initialize or reboot the trunk device such as circuit card or MG.

O-6 Off-Hook Alarm

General Description

This feature allows a station user to call the Attendant or a predetermined station by simply staying OFF-HOOK. The Calling Number is automatically displayed at the ATTENDANT CONSOLE [A-3] or a Desktop terminal with LCD display.

Operating Procedure

To operate

1. Lift the handset and stay off-hook.
2. After a predetermined time elapses (2 to 30 sec.), the call will terminate at the ATTENDANT CONSOLE [A-3] or the predetermined station.
3. The calling Telephone Number will be displayed at the ATTENDANT CONSOLE [A-3] or Desktop terminal when answered.

Service Conditions

1. Predetermined timing interval is programmable through the PCPro (2 to 30 seconds).
2. In the night mode, an OFF-HOOK ALARM call directed to the ATTENDANT CONSOLE [A-3] will not terminate at the ATTENDANT NIGHT TRANSFER [A-19] predetermined night station. The caller will receive a Reorder Tone.
3. A station will be assigned as a member of an OFF-HOOK ALARM Group and will be connected to the predetermined stations that are assigned for each Off-Hook Group.
4. When the OFF-HOOK ALARM call designating station uses Telephone Number (TELN):
   Maximum number of OFF-HOOK ALARM Group is 255 per Tenant Group.
   When the OFF-HOOK ALARM call designating station uses Physical Station Number (STN):
   Maximum number of OFF-HOOK ALARM Group is 7 per Tenant Group.

   Note: There is no limit on the number of stations which belong to an OFF-HOOK ALARM Group.

5. A designating station for OFF-HOOK ALARM can be assigned to each OFF-HOOK ALARM Group. A Group Number must be assigned to all the member stations to be included in the Group.
6. The station assigned as a terminating station of each OFF-HOOK ALARM Group can be a member of a HUNTING GROUP [S-7, 8, 9].
7. This service is programmed via System Data. Stations may be restricted from using this feature in CLASS OF SERVICE - INDIVIDUAL [C-15].
8. The ATTENDANT CONSOLE [A-3] can answer by pressing the EMG key, which must be assigned by the PCPro. Using the ANSWER key will not give priority to the OFF-HOOK ALARM call.
9. The predetermined target station can set CALL FORWARDING - ALL CALLS [C-5] and CALL FORWARDING - BUSY LINE [C-2].
10. When the predetermined station is busy, the OFF-HOOK ALARM call receives Ringback Tone until the station becomes idle. When the predetermined designating station is in the LINE LOCKOUT [L-3] status or make busy status, the OFF-HOOK ALARM call receives the Reorder Tone (ROT).
11. This feature cannot be applied while in CONSULTATION HOLD - ALL CALLS [C-17] or CALL HOLD [C-6] mode.

12. When an OFF-HOOK ALARM call is directed to a Pilot station of a UCD [U-1] Group, queuing is not available when all the stations within the group are busy.
13. The designating station for OFF-HOOK ALARM must be the Desktop terminal. When the designating station has set CALL FORWARDING-ALL CALLS [C-5], the Call Forwarding target station must also be the Desktop terminal. When the designating station is a member of a HUNTING GROUP [S-7, 8, 9], all stations within the Hunt Group must also be the Desktop terminal.
14. An OFF-HOOK ALARM call cannot be originated from the ISDN terminal. The ISDN terminal and PS cannot be assigned as a designating station of the OFF-HOOK ALARM call.
15. The predetermined target station for ATTENDANT NIGHT TRANSFER [A-19] and PRIORITY CALL [P-9] can be assigned as the designating station for OFF-HOOK ALARM [O-6].
16. The ringing tone provided when the designating station receives an OFF-HOOK ALARM call is the same as that of station call.
17. When the designating station receives an OFF-HOOK ALARM call, "ALARM", the designating station displays the number of the calling station with the handset lifted on its LCD.

[For North America only]

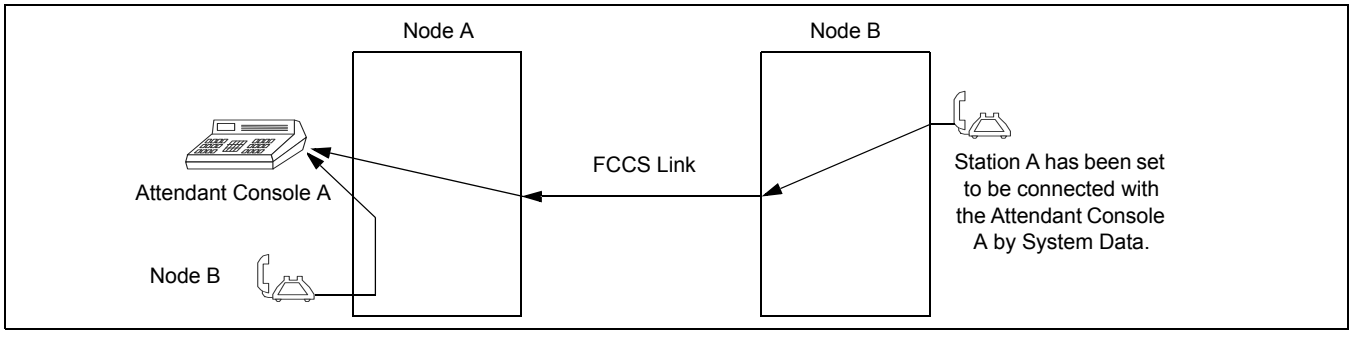
18. This feature allows a station user to call the Attendant or a predetermined station by simply not completing a call and staying off-hook after dialing some digits. The following conditions must be considered.
    a.) All numbers (0-9, *, #) must be defined in ANPD/ANPDL/ANPDN for this feature to work correctly.
    b.) This feature is mutually exclusive with Automatic return to Idle Desktop terminal.

Example: A station user goes off-hook and dials only a partial number.
1. Station 1000 goes off hook and dials 10.
2. 1 is defined in ANPD as four digits and is in ASPA as STN.
3. After Inter digit timer expires (ASYD, System Data 1, Index 129 Bit 0-3) the call is routed to the proper off hook alarm destination. **Note 1**

Note 1: This destination is determined by the programming shown later in this section.

FCCS Conditions

1. This feature can also be activated when a station and the called party belong to different Nodes.
2. If a station belongs to the Node that has an Attendant Console, the station is connected with the Attendant Console. If a station belongs to the Node that does not have an Attendant Console, the station is connected with the specified Attendant Console in the Remote Node according to the System Data assigned. See the figure below.



3. The OFF-HOOK ALARM call can be terminated at the predetermined station/Attendant Console in the different node via FCCS link.
4. The feature data via DM, LDM and NDM can coexist in a system. However, data registered on NDM has top priority. Data registered on LDM has precedence over data registered on DM. (Priority: NDM → LDM → DM)

Interactions

1. A total of seven OFF-HOOK ALARM Groups per Tenant may be assigned to the system, if the termination point is a station. Any station not assigned in ASGD will go to Reorder Tone after the Register Prepause timer (ASYD, System Data 1, Index 128) expires. If the termination point is the Attendant Console, all stations in the Tenant will terminate to the Attendant Console.
2. ASAT cannot be used to construct different OFF-HOOK ALARM Groups terminating to specific Attendant Consoles. If the Tenant is equipped with more than one Attendant Console, OFF-HOOK ALARM calls will be directed to all Consoles with the OFF-HOOK ALARM keys programmed in AAKP.
3. The following features will function on the terminating calls to a predetermined station:

CALL PICKUP	[C-7]
CALL TRANSFER	[C-11]
CALL HOLD	[C-6]
CALL FORWARDING	[C-5A]

4. No tone is provided on CALL WAITING-TERMINATING [C-12].
5. This feature works also when a SIP Handler Controlled SIP terminal is a destination of a station in off-hook state.
6. This feature is not activated on the SIP Handler Controlled SIP terminal even after a predetermined time elapses.

Programming

STEP 1: **ASYD** - System Data 1, Index 128, Bits 0-3. Register Prepause timer. After the timer expires, the station assigned to an OFF-HOOK ALARM Group will ring at the terminating Attendant Console or station. Leave Index 128 as 00H for normal operation.

System Data 1, Index 168, Bit 4. Enable OFF-HOOK ALARM? 0/1: No/Yes.

System Data 1, Index 168, Bit 5. Is the terminating point for OFF-HOOK ALARM an Attendant Console or Station? 0/1: Attendant Console/Station. If Attendant Console is chosen, go to AAKP command. If station is chosen, go to the ASID and ASGD commands.

STEP 2: **ASID** - This command defines the purpose of the Special Incoming Group, and is used in conjunction with ASGD. Assign the following:

TN: Tenant Number
F: Function Number (0-7)
0: OFF-HOOK ALARM
GN: Group Number (1-7).
S/A: Special Incoming Equipment
S: Station
A: Attendant Console
SSTN: Special Station Number

STEP 3: **ASGD** - This command defines what stations belong to the Special Incoming Group. Assign the following:

TN: Tenant Number
STN: Station Number
F: Function Number (0-7)
0: Off-Hook Alarm
GN: Group Number (1-7)

STEP 4: **ASFC** - Assign a Service Feature Restriction Class that allows SFI 30 to stations that will have OFF-HOOK ALARM.

STEP 5: **AAKP** - Assign the Attendant Console Key Pattern. The ATN: Attendant Number (TRK Number for RTE 901); KYN = 1-6; and FUNC = 2. (The Console must be in Night mode to assign this command.)

[For North America only]

This feature allows a station user to call the Attendant or a predetermined station by simply not completing a call and staying off-hook after dialing some digits. The following data programming is required in addition to the above.

STEP 1: **ASFC** - Assign Service Feature Restriction Class of the station.

SFI = 203 (OFF-HOOK ALARM for Inter-digit time-out)
RES = 1 (Allowed)

STEP 2: **ANPD/ANPDL/ANPDN**- Define all number groups (0-9, *, #).

FCCS Programming

This data setting describes only the additional or replacement commands used to activate FCCS Services. When used as an FCCS feature, these commands replace the non-network commands (for example, AAED is replaced by AAEDN). Note that the data setting for FCCS Service varies depending on the Node.

Note: When using ASYDL or ASYDN to program an FCCS feature, the basic programming of ASYD is required in addition to ASYDL or ASYDN.

ASYDL - This data must be set to all the Nodes.
System Data 1, Index 640, assign the Fusion Point Code (FPC) of the Node providing the Attendant Console. **Note 2**

Note 2: Assign the Self FPC at the Terminating Node.
The following data should also be set in addition to the above-mentioned commands.
The commands to be set to the Node that accommodates the Attendant Console:
AAKP, ACFR (when the destination is the Night ATT: ATNR)
The commands to be set to all the Nodes: ASYD, ASFC

-The following data setting is required when the OFF-HOOK ALARM call terminates to a station in the Remote Node via FCCS.

(When entering data on NDM)

STEP 1: **ASYD** - The same data must be assigned to all Nodes.
System Data 1, Index 128, Bits 0-3. Register Prepause timer. After the time expires, the station assigned to an OFF-HOOK ALARM Group will ring at the terminating Attendant Console or station. Leave Index 128 as 00H for normal operation.
System Data 1, Index 168, Bit 4. Enable OFF-HOOK ALARM? 0/1: No/Yes.
System Data 1, Index 168, Bit 5. Assign data "1" (station is the designation of OFF-HOOK ALARM call).

STEP 2: **ASIDN** - This command defines the purpose of the Special Incoming Group and is used in conjunction with the ASGDN command. Assign the following:
TN: Tenant Number
F: Function Number. Assign "0 (OFF-Hook Alarm)".
GN: Group Number (1-255)
S/A: Special Incoming Equipment. Assign "S" (Telephone Number (of the station)).
STELN: Telephone Number of Special Station Number

STEP 3: **ASGDN** - This command defines the stations belonging to the Special Incoming Group. Assign the following:
UGN: User Group Number
STELN: Telephone Number of the station
F: Function Number. Assign "0 (OFF-Hook Alarm)".
GN: Group Number (1-255)

STEP 4: **ASFC** - The same data must be assigned to all Nodes.
Assign a Service Feature Restriction Class that allows SFI 30 to stations that will have OFF-HOOK ALARM.

(When entering data on LDM)

STEP 1: **ASYD** - The same data must be assigned to all Nodes.

System Data 1, Index 128, Bits 0-3. Register Prepause timer. After the time expires, the station assigned to an OFF-HOOK ALARM Group will ring at the terminating Attendant Console or station. Leave Index 128 as 00H for normal operation.

System Data 1, Index 168, Bit 4. Enable OFF-HOOK ALARM? 0/1: No/Yes.

System Data 1, Index 168, Bit 5. Assign data "1" (station is the designation of OFF-HOOK ALARM call).

STEP 2: **ASIDL** - This command defines the purpose of the Special Incoming Group and is used in conjunction with the ASGDL command. Assign the following:

TN: Tenant Number

F: Function Number. Assign "0 (OFF-Hook Alarm)".

GN: Group Number (1-255)

S/A: Special Incoming Equipment. Assign "S" (Telephone Number (of the station)).

STELN: Telephone Number of Special Station Number

STEP 3: **ASGDL** - This command defines the stations belonging to the Special Incoming Group. Assign the following:

UGN: User Group Number

STELN: Telephone Number of the station

F: Function Number. Assign "0 (OFF-Hook Alarm)".

GN: Group Number (1-255)

STEP 4: **ASFC** - The same data must be assigned to all Nodes.

Assign a Service Feature Restriction Class that allows SFI 30 to stations that will have OFF-HOOK ALARM.

O-7 Off-Hook Queuing

General Description

This service allows a Station user, upon encountering a Trunk-busy condition, to remain off-hook and automatically enter a first-in, first-out queue. As soon as an Outgoing Trunk becomes available, the switch connects the next call to this Trunk.

Operating Procedure

To operate

1. Lift the handset. Receive a Dial Tone.
2. Dial the Outgoing Call Access Code and the desired number (LCR access).
3. The System determines that no Trunks are available and places the call in an available queue slot.
4. The caller receives Service Set Tone. Remains off-hook and waits.
5. When a Trunk becomes available, the Dialed Number is automatically sent forward.

Service Conditions

1. The following table shows connection availability of this feature.

X: Service available N: Service not available

Calling Party	Outgoing Trunk (OGT)			Remarks
	COT Trunk	CCIS Trunk	ISDN Trunk	
Analog Terminal	X	X	X	
Attendant Console	X	X	X	
Attendant Console (from Station)	X	X	X	
Attendant Console (from Trunk)	X	X	X	
ISDN Terminal	N	N	X	
Desktop Terminal **Note 1**	X	X	X	

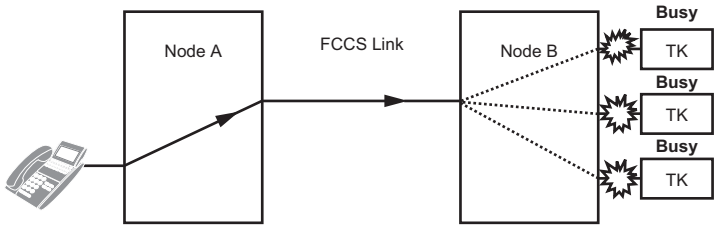
Note 1: This service is not available for Outgoing Calls from Sub Line.

2. This feature is usually used in conjunction with LEAST COST ROUTING - 3/6-DIGIT [L-5].
3. The maximum number of simultaneous OFF-HOOK QUEUEs per system:
   1 IMG: 64 calls
   4 IMG: 256 calls
4. The queue slots are used commonly by OFF-HOOK QUEUING, OUTGOING TRUNK QUEUING [O-2] and INTER-OFFICE OFF-HOOK QUEUING [I-11].
5. If no queue slots are available, the Station user will receive Busy Tone.
6. Route Class of the LCR can give the choice of the OFF-HOOK QUEUING.
7. CLASS OF SERVICE - INDIVIDUAL [C-15] is applied for each Station. This class of service is different from the class of service for OUTGOING TRUNK QUEUING [O-2]. Both services are mutually exclusive.
8. When automatic outgoing queuing (on-hook queuing) is in service (Tenant basis), OFF-HOOK QUEUING service is not effective.

9. ACCOUNT CODE [A-18] and FORCED ACCOUNT CODE [F-7] information can be recorded on STATION MESSAGE DETAIL RECORDING (SMDR) [S-10], when used in conjunction with OFF-HOOK QUEUING.
10. This feature can also be activated when using a Trunk in the Remote Node via FCCS link.

Example: *Node A and Node B are in an FCCS network.*

- *Although a station belonging to Node A tried to call outside using a trunk belonging to Node B, all the trunks are busy. The call can be in the queue using this feature.*



11. The maximum of number of the OGQ Stations is 1,024 within the FCCS Network (NCN can manage a maximum of 1,024 calls).
12. OGQ data is transmitted to and stored in the Network Control Node (NCN). Therefore, all the OGQ data will be cleared by the system initialization of the NCN.
13. When the access to the NCN is impossible (system down, cable disconnection or trouble has occurred), the user cannot set or cancel OGQ data. (After dialing the Access Code or pressing the Feature keys for set/cancel OGQ, the user hears the Reorder Tone.)
14. When the Trunk becomes idle status after the NCN is recovered, the call in the queue will be called back, if data has not been cleared. If data has been cleared, the call is canceled automatically.

Interactions

1. UNIVERSAL SENDER [U-3] must first be provided.
2. The System attempts Trunk seizure every four seconds until the Connection has been completed.
3. LEAST-COST ROUTING-3/6 DIGIT [L-5] information will be applied to these attempts. Therefore, OUTGOING TRUNK QUEUING will be set against all routes in the Outgoing Routing Pattern (OPR).
4. If all queuing memory slots are in use, the Station user will receive a Trunk Busy Signal.
5. The time a call may remain in queue is unlimited.
6. The following features cannot be activated toward the busy Station: CALL WAITING - ORIGINATING [C-31], CALL WAITING - TERMINATING [C-12], ATTENDANT CAMP-ON WITH TONE INDICATION [A-1], BUSY VERIFICATION [B-3] and EXECUTIVE RIGHT-OF-WAY [E-1].
7. Do not disable constant Service Set Tone by changing System Data 1, Index 42, Bit 7 to data “1”. This changes constant Service Set Tone to burst Special Dial Tone, and Special Dial Tone to burst Special Dial Tone, followed by constant Dial Tone. This will also disable TANDEM OFF-HOOK QUEUING.
8. When testing OGQ, at least one Trunk of the Trunk Group being tested must be engaged in a Station-to-Trunk or Trunk-to-Trunk Connection. Use of the MB key on the circuit card will not allow OGQ to test and operate properly.
9. This feature is not available for forwarding calls with Call Forwarding-Don't Answer-Outside [C-74].

10. This feature is not available for forwarding calls with Call Forwarding - All Calls/Split Call Forwarding - All Calls (for trunk), Call Forwarding - Busy Line/Split Call Forwarding - Busy Line (for trunk), or Call Forwarding - Don't Answer/Split Call Forwarding - Don't Answer (for trunk).

Programming

STEP 1: **ASFC** - Assign a Service Feature Restriction Class that allows SFI 21 to Stations that will have OFF-HOOK QUEUING.

STEP 2: **ARTD** - A dummy route must be flagged in CDN 9:TQ. Also allow OUTGOING TRUNK QUEUING to the LCR(s) flag route.

Note: To change the data for existing route (RT), initialize or reboot the trunk device such as circuit card or MG.

FCCS Programming

This data setting describes only the additional or replacement commands used to activate FCCS Services. When used as an FCCS feature, these commands replace the non-network commands (for example, AAED is replaced by AAEDN). Note that the data setting for FCCS Service varies depending on the Node.

Note: When using ASYDL or ASYDN to program an FCCS feature, the basic programming of ASYD is required in addition to ASYDL or ASYDN.

STEP 1: **ALRTN** - Assign this command to connect the Logical Route to the Physical Route.

Note: The following data should also be set in addition to the above-mentioned commands.
The command to be set to the Node accommodating the Trunk: ARTD
The commands to be set to all the Nodes: ASYD, ASFC

STEP 2: **ASYDN** - This data must be assigned to the NCN. System Data 1, Index 41: Designate the Trunk Queuing Override Timer.

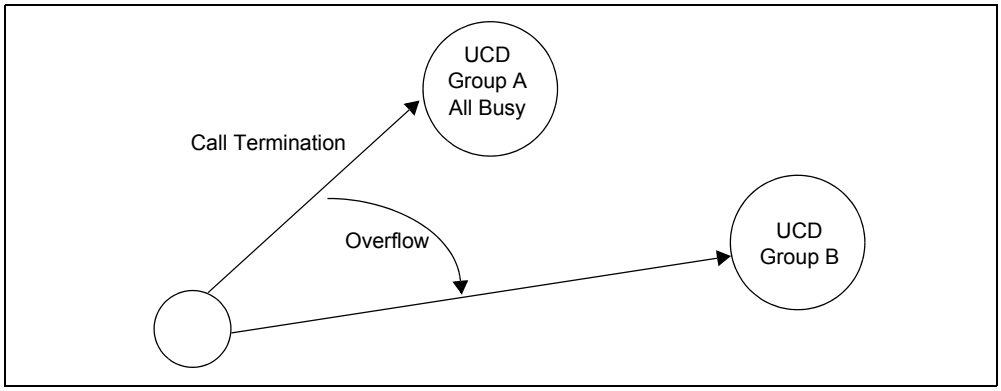
STEP 3: **ASPAN** - This data must be assigned to the NCN.

TN: Tenant Number
ACC: Access Number
CI: “B: Busy” (for Entry), “N: Normal” (for Cancel)
SRV: “SSC: Service Code”
SID: 19 - OG Queuing - Entry
CI: “N: Normal”
SID: 20 = OG Queuing - Cancel

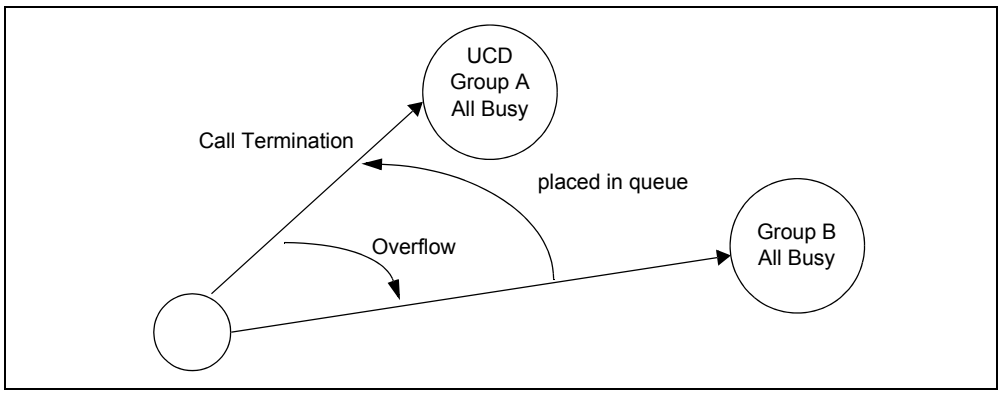
O-13 Overflow - UCD

General Description

1. When a call has terminated to UCD [U-1] Group A, and the Incoming Call has encountered all Stations busy in Group A, the call is transferred to UCD [U-1] Group B, if Group B is registered as the OVERFLOW-UCD destination.



2. If all Stations are busy in Group B, then the call is placed in queue for the originally called UCD [U-1] Group (Group A).



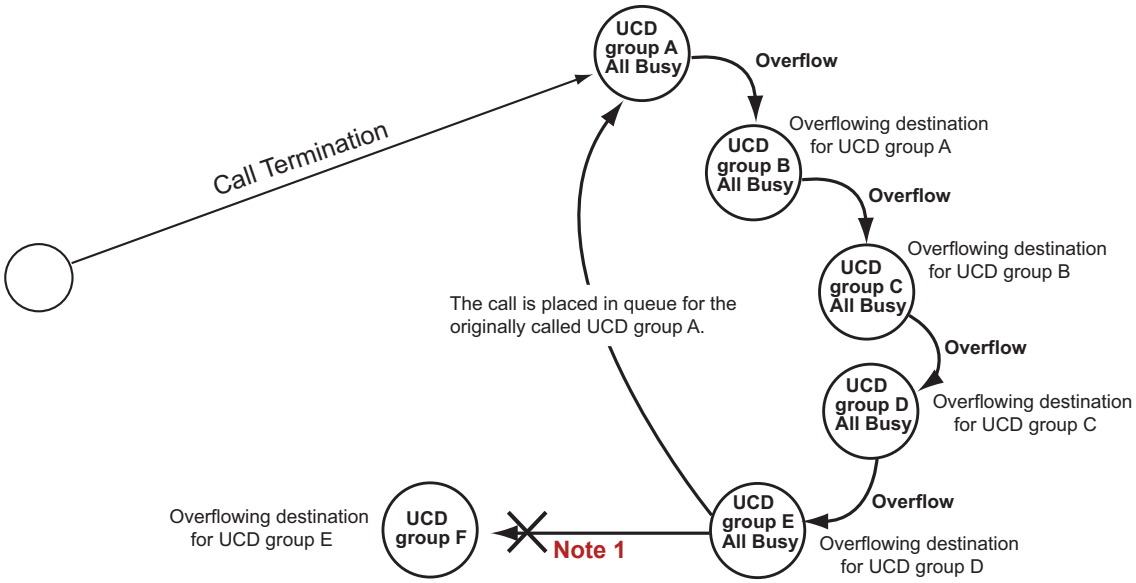
Operating Procedure

No manual operation is required.

Service Conditions

1. An OVERFLOW Group must be a UCD [U-1] Group.
2. One OVERFLOW Group can be provided for one UCD [U-1] Group.
3. An Originating UCD Group will only Overflow to its Overflow UCD Group one time. If the Overflow UCD Group is busy and the call is placed into the Originating UCD Group's queue, the Originating UCD Group will not attempt an Overflow a second time.
4. An OVERFLOW Group cannot be registered into a queue.

5. A call can overflow a maximum of four times at the maximum by using this feature. If overflowing destinations are all busy, the call is queued into original group (depending on System Data).
For details, refer to the figure below.



Note 1: The call cannot be overflowed more than 5 times.

6. UCD OVERFLOW data can be programmed in Local Data Memory (LDM).
7. The overflowing destination which can be assigned to UCD Group is as follows:

UCD Group (that is busy)		Overflowing Destination
• UCD Group assigned by ASHU command	→	UCD Group designated by AUOG command
• UCD Group assigned by ASHUL command	→	UCD Group designated by AUOGL command
• UCD Group assigned by ASHUN command	→	UCD Group designated by AUOGN command

8. This feature can be activated via FCCS.
9. When using the FCCS Network, the number type (Telephone Number or Physical Station Number) for the Station assignment must be the same between the originally Called Group and the Forwarded Group.
10. After PS has been assigned to the destination UCD group, if a call from the PS cannot connected to the group due to busy status, the call will receive Busy Tone.
11. When using Location Diversity [L-55], the Telephone Number of User Profile Data must be assigned to the station by using AUOGN command.

Interactions

1. ALL UNIFORM CALL DISTRIBUTION [U-1] interactions apply to the UCD Overflow Group. The Incoming Overflow Call is handled in the same way as a regular Incoming Call. If both UCD Groups are busy, the call will queue at Group A.
2. Do not set ASFC, SFI 252=1 (Multi-Boss/Multi-Secretary is in service) to stations belonging to UCD groups.

Programming

STEP 1: **ASYD** - System Data 1, Index 60, Bit 3: UCD Queuing in service. 0/1 = Yes/No

STEP 2: **ASHU/ASHUL** - Assign the required data for the UCD Group:

TN/UGN: Tenant Number/User Group Number
STN/TELN: Station Number/Telephone Number
[Station List in Group]
STN/TELN: Station Number/Telephone Number

STEP 3: **ANPD** - Reserve a Number Level for Feature Access. Select Normal Service (N), Hooking Service (H) or Busy Service (B) from the combo box for Connection Indexes. (if applicable to any feature in this Level). Assign NND in accordance with a predetermined Numbering Plan.

STEP 4: **ASPA** - Assign an Access Code to UNIFORM CALL DISTRIBUTION [U-1], SRV = "SSCA: Service Code Appendix", SIDA = 50 Busy Set, SRV = "SSCA: Service Code Appendix", SIDA = 51 Busy Cancel. Connection Status CI = "N: Normal".

STEP 5: **AUOG/AUOGL** - Assign the UCD OVERFLOW data via this command. Assign the following:

TN-A, STN-A/UGN-A, TELN-A: Tenant Number and Station Number/User Group Number and Telephone Number that belong to a UCD Group.

TN-B, STN-B/UGN-B, TELN-B: Tenant Number and Station Number/User Group Number and Telephone Number of a member Station in a UCD Group to be hunted in the case where the UCD Group designated by TN-A and STN-A happens to be busy.

TRANSLATION: When UCD Group A is busy, calls will overflow to UCD Group B.

STEP 6: **AKYD** - To a programmable Line/Feature key on the UCD Stations (when the Stations are Dterms), assign FKY = 19-UCD Busy Out.

STEP 7: **ADSL** - Assign UCDBO key; Assign SN = 0 (Idle), 3 (Called Station Busy) and FKY = 19 (UCD Busy Out).

This command is required for assigning a function to each Soft key.

FCCS Programming

This data setting describes only the additional or replacement commands used to activate FCCS Services. When used as an FCCS feature, these commands replace the non-network commands (for example, AAED is replaced by AAEDN). Note that the data setting for FCCS Service varies depending on the Node.

Note: When using ASYDL or ASYDN to program an FCCS feature, the basic programming of ASYD is required in addition to ASYDL or ASYDN.

STEP 1: **ASYDN** - This data must be assigned to the NCN. System Data 1, Index 60, Bit 3: 0/1 = UCD Queuing is used/not used

STEP 2: **AUOGN** - This data must be assigned to the NCN. Assign the UCD OVERFLOW data via this command.

UGN-A: User Group Number belonging to a UCD Group
TELN-A: Telephone Number belonging to a UCD Group
UGN-B: User Group Number of Overflow Destination
TELN-B: Telephone Number of Overflow Destination

O-14 Operator Monitoring

General Description

When an exchange or a TIE Line Incoming Call has terminated to the Attendant position and the Attendant has connected the caller to the requested Station, this feature allows the Attendant to monitor the call by depressing the "MON" (Optional) key on the Console.

Note: This feature is only available in Australia.

Operating Procedure

1. The Attendant at the Attendant position answers an exchange or a TIE Line Incoming Call.
2. The Attendant dials the requested Station Number.
3. The Called Station answers.
4. The Attendant depressing the "MON" (Optional) key on the Console.
5. The Attendant enters the monitor state.

Service Conditions

1. This feature is available in Australia only.
2. This service can be activated for Connections established between an external Calling Party and an internal Called Station via the Attendant position.
3. The Attendant in the monitor state is not allowed to reenter the Connection.
4. The Attendant may end monitoring by depressing the RELEASE key.
5. While the call is being monitored by the Attendant, no hooking services can be performed at the Called Station.
6. The Attendant cannot perform any other services for the Connection during monitoring.
7. This feature cannot be activated for Inter-Office CCIS.

Programming

STEP 1: **ASYD**

- SYSTEM DATA1

< Index6 >

Bit 4-6 (ATT CON Key Arrangement Pattern): Assign 2 Hexadecimal (for Australia).

STEP 2: **ASFC** - Assign Service Feature Restriction Class of the Station.

SFI = 49 (Operator Monitoring)

RES = 1 (Allowed)

O-15 Overlapping

General Description

This feature allows the Attendant at an Attendant position to originate a call to a Station or to an external line (exchange or TIE) and hold it in the monitor mode before or after the Called Party answers, thus allowing another call to be handled.

Note: This feature is only available in Australia.

Operating Procedure

1. Overlapping while the Called Party is being rung
   a.) The Attendant originates a call to a Station or external line (exchange or TIE).
   b.) Ringback Tone is sent to the Attendant.
   c.) When the Attendant depresses the "OL" (Optional) key, the call is held in monitor mode.
   d.) By depressing a Loop key before or after the Called Party answers, the Attendant reenters the Connection.
2. Overlapping while speech is in progress
   a.) The Attendant originates a call to a Station or external line (exchange or TIE), and the Called Party answers.
   b.) When the Attendant depresses the "OL" key, the call is held in monitor mode.
   c.) The Attendant reenters the Connection by depressing a Loop key.
   d.) If the Called Party is released while being held, the Loop is opened and enters the idle state.

Service Conditions

1. This feature is available in Australia only.
2. This feature is effective only for Two-party Connections established between an Attendant and a Station or external line party.
3. Regardless of whether the Called Party answers or not, the Attendant can reenter the Connection or set the service again.
4. An Attendant position can set this feature to only one call at a time.
5. No MUSIC ON HOLD [M-7] is sent to the party who is placed on hold while call is in progress.
6. The Attendant cannot reenter the Held Call while handling another call.
7. This feature is assignable on a Tenant basis.
8. When the call in progress is held by the Attendant using this service, SHF operations at the held side become ineffective, and no further services can be performed for the Held Call.
9. This feature will not be counted as SERVICE FEATURE PEG COUNT [S-27].
10. In the OVERLAPPING state, MB lamp of the Trunk for OVERLAPPING will be illuminated.
    (The MB lamp will remain illuminated thereafter.)
    To turn off the lamp:
    1) On/Off of "MB" key of the Trunk
    2) System initialization
    3) LP initialization
11. When mounting Trunks for OVERLAPPING (8COTBF), the mounting slot is fixed in the conventional unit.

12. There will be no output of message for the Message Center Interface (MCI).
13. The billing for a call answered while OVERLAPPING will be charged to the ATTENDANT CONSOLE [A-3].
14. This feature is not available for Inter-Office CCIS.

Programming

STEP 1: **ASYD**

- SYSTEM DATA1

< Index 6 >

Bit 6 (ATT CON Key Arrangement Pattern) : $\frac{b6}{0}\ \frac{b5}{1}\ \frac{b4}{0}$ (For Australia)

- SYSTEM DATA 2

< Index 15 >

Bit 0 (Overlapping Service): 0/1 = Out/In service

Assign 1.

STEP 2: **ATRK** - Assign Overlapping Trunk (PA-8COT BF) data.

RT (Route Number): xxx

TK (Trunk Number): xxx

LENS (Line Equipment Numbers) : MG U G LV

xx x xx 0
xx x xx 1 } **Note 1**

Note 1: As an Overlapping Trunk, No.0 and No.1 circuits of the PA-8COT BF circuit card can be used.

STEP 3: **MBTK** - Assign Make Idle status to the Overlapping Trunk.

RT: xxx

TK: xxx

MB: 0 (= idle)

STEP 4: **ATRK** - Delete Trunk data of the Attendant Console which is to be equipped with Overlapping feature.

If the Attendant Console is to be newly installed, skip this STEP4. **Note 2**

RT: 901 (= Attendant Console)

TK: xxx

LENS: xxx200/xxx201

TN: xx

Note 2: Place both circuits of the ATI Card to Busy status before deleting data of the Attendant Console.

STEP 5: **ATRK** - Assign Trunk data of the Attendant Console data.

RT: 901 (= Attendant Console)

TK: xxx

LENS: xxx200/xxx201

TN: xx

STEP 6: **MBTK** - Assign Make Idle status to the newly assigned Trunk in the ATRK command.

O-16 Outgoing Trunk Queuing - Attendant

General Description

This feature allows an ATTENDANT CONSOLE [A-3], upon encountering a Trunk busy condition, to remain Off-Hook and automatically enter a first-in, first-out queue. As soon as an Outgoing Trunk becomes available, the switch connects the next call to that Trunk.

Operating Procedure

To operate

1. Dial the Access Code and the desired number.
2. When no Trunks are available, the call automatically enters an available queue slot.
3. The Attendant receives Service Set Tone; remains off-hook and waits.
4. When a Trunk becomes available, the Dialed Number is automatically sent forward.

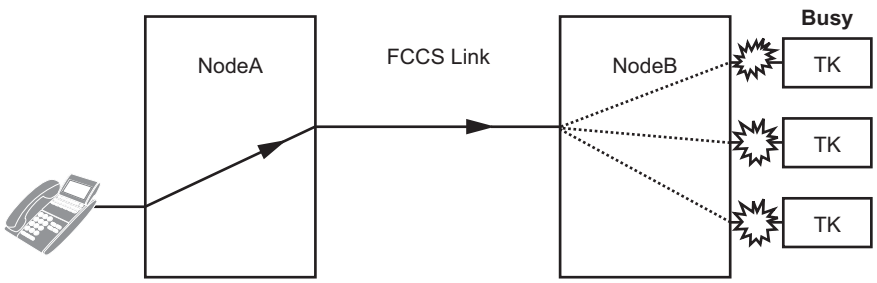
Service Conditions

1. This feature is used with LEAST COST ROUTING - 3/6-DIGIT [L-5].
2. The maximum number of Stations or ATTENDANT CONSOLEs [A-3] that may use OFF-HOOK QUEUING [O-7] simultaneously is 256 per system.
3. The queue slots are shared by OFF-HOOK QUEUING [O-7], OUTGOING TRUNK QUEUING (OGQ) [O-2, 12D, 16] and INTER-OFFICE OFF-HOOK QUEUING [I-11].
4. If no queue slots are available, the ATTENDANT CONSOLE [A-3] will receive Busy Tone.
5. CLASS OF SERVICE - INDIVIDUAL [C-15] is applied to each ATTENDANT CONSOLE [A-3]. (RSC 0 and SFC 0 are reserved for ATTENDANT CONSOLE [A-3].)
6. OFF-HOOK QUEUING is utilized for this feature.

<FCCS Conditions>

1. This feature can also be activated when using a Trunk in the Remote Node via FCCS Link.

Example:
Node A and Node B are in a FCCS Network.
Although a station belonging to Node A tried to call outside using a trunk belonging to Node B, all the trunks are busy.
The call can be in the queue using this feature.



2. The maximum of number for the OGQ Station is 1,024 within the FCCS Network (NCN can manage a maximum of 1,024 calls).

3. OGQ data is transmitted to and stored in the Network Control Node (NCN). Therefore, all the OGQ data will be cleared by the system initialization of the NCN.
4. When the access to the NCN is impossible (system down or the cable disconnection or any trouble has been happened), the user can not set or cancel OGQ data. (After dialing the Access Code or pressing the Function keys for set/cancel OGQ, the user hears the Reorder Tone.)
5. If data has not been cleared after the NCN is recovered, the call in the queue will be called back when a Trunk becomes idle status. If data has been cleared, the call is canceled automatically.

Programming

STEP 1: **ASYD** - System Data 2, Index 4, Bit 2: Off-Hook OGQ for Attendant 0/1: Out/In Service.

STEP 2: **ASFC** - Assign 1 (= Allowed) in SFI21 (= Off-Hook OGQ) for SFC0.

FCCS Programming

This data setting is required when FCCS Service data is added to Non-FCCS Service data. Note that the data setting for FCCS Service varies depending on the Node.

STEP 1: **ASYDN** - This data must be assigned to the NCN.
System Data 1, Index 41: Designate the Trunk Queuing Override Timer

STEP 2: **ASPAN** - These data must be assigned to the NCN.

TN: Tenant Number
ACC: Access number
CI: "B: Busy" (for Entry), "N: Normal" (for Cancel)
SRV: "SSC: Service Code"
SID: 19 = OG Queuing-Entry
CI: "N: Normal"
SID: 20 = OG Queuing-Cancel

Note: The following data should also be set in addition to the above-mentioned commands.
- The command to be set to the Node accommodating the Trunk
  ARTD
- The commands to be set to all the Nodes
  ASYD, ASFC

O-21 Outgoing Trunk Busy Announcements

General Description

This feature permits a Station-originated call, upon encountering a Trunk busy signal, to be automatically routed to a recorded announcement informing the caller that all Outgoing Trunks are busy.

Note: When using this feature with VS32, see VS32 Conference Server [V-26].

Operating Procedure

No manual operation is required after this feature has been programmed in System Data.

Service Conditions

1. This feature is available when a Station-originated call or tandem call encounters a busy condition.
2. MISCELLANEOUS TRUNK RESTRICTION [M-2] is also available for this feature.
3. The Announcement Machine can be provided on a per Tenant basis.
4. The availability of this feature cannot be determined for each Outgoing Trunk Route.
5. The other services activated by an Outgoing Trunk-busy condition are executed prior to this service, for example:

OFF-HOOK QUEUING	[O-7]
OUTGOING TRUNK QUEUING	[O-2]
ROUTE ADVANCE	[R-6]

6. This feature is not available when a Call Origination is impossible because of a Sender-busy condition, even though there are idle Trunks.
7. This feature is available when a call using STATION-TO-STATION CALLING [S-11] is transferred by CALL FORWARDING - ALL CALLS - OUTSIDE [C-28] or CALL FORWARDING - BUSY LINE - OUTSIDE [C-60] and encounters busy Outgoing Trunks.
8. This feature is available when a Station Calling on a DIRECT INWARD DIALING [D-8] or a TIE Line is transferred by CALL FORWARDING - ALL CALLS - OUTSIDE [C-28] or CALL FORWARDING - BUSY LINE - OUTSIDE [C-60] and encounters busy Outgoing Trunks.
9. This feature is not available when a call is transferred by CALL FORWARDING - DON'T ANSWER - OUTSIDE [C-74] and encounters busy Outgoing Trunks.
10. This feature requires one of the following hardware alternatives:
    a.) Central Office Trunk (COT) or 2W E&M Trunk (TLT) and an Announcement Machine
    b.) Digital Announcement Trunk (DAT).
11. This feature is not available for DIT CALL FORWARDING-OUTSIDE.
12. This feature is only available to a Trunk in the same Node within the FCCS Network.
13. This feature is not available for Incoming Routes of CCIS.

Programming

STEP 1: **AAED** - Assign Announcement Equipment Number = 4, and Multiple Connection = 0 are programmed.

STEP 2: **ARSC** - Set Route Restriction Class to allow RES = 1 for Announcement Trunk.

STEP 3: **ARRC** - Assign Alternative Route Restriction to allow or disallow the DID or TIE Line Route to the Announcement Route.

O-37D Off-Hook Line Number Display -D

General Description

This feature permits the Dterm LCD to display the Calling Telephone Number/Trunk Number when the Dterm user originates a call by pressing a Line key or using TRUNK LINE APPEARANCE [T-23] service.

Operating Procedure

1. When a call is originated by pressing the Line key:
   - In the on-hook status; the LCD displays:



   In the idle status and no display.
   - In the off-hook status; the LCD displays:



   Selected line number is displayed.
   - When dialing the one digit ("3"); the LCD displays:



   Selected line number is cleared and Dialed Number is displayed.
2. When a call is originated by using TRUNK LINE APPEARANCE service; the LCD displays:
   - In the on-hook status:



   In the idle status and no display.
   - In the off-hook status:



   Selected Trunk Number is displayed.
   - When dialing the one digit ("0"); the LCD displays:



   Selected Trunk Number is cleared and Dialed Number is displayed.

Service Conditions

1. This service is provided when off-hook. In the preselect status, this is not provided.
2. This service is not provided when a call is originated by using HOTLINE [H-1] service.
3. In case a call is originated by using TRUNK LINE APPEARANCE service, the connected Trunk Number is displayed.
4. If the Trunk Number is not assigned, the LCD doesn't display.
5. The maximum digits of the displayed Trunk Number is 4.
6. The display position of the selected line/Trunk Number may be changed by System Data as below:

a.)



b.)



c.)



Programming

ASYD - System Data 1, Index 389, Bits 5 and 6. Location of Off-Hook Line Number Display on the Dterm.

00: Right side (Standard Setting)
01: Center
10: Left side
11: -

System Data 1, Index 389, Bit 7. Off-Hook Line Number Display -D. 0/1 = Out/In Service. Assign data "1".

O-41D Off-Hook Line Number Display -D (US)

General Description

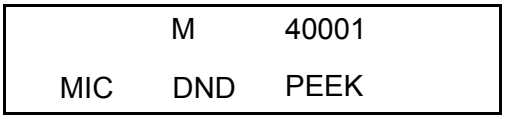
This is an enhancement to OFF-HOOK LINE NUMBER DISPLAY -D [O-37D]. In addition to the display of the Station Number, this feature will insert a letter to designate the My Line or Prime Line.

Note: This feature is available in North America only.

Operating Procedure

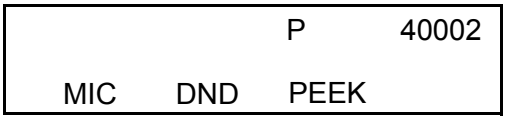
- My Line key selected with line number display in center

1. A Desktop terminal station presses the SPEAKER key or handset off hook.
2. LCD displays the My Line or Prime Line number and the letter “M”.



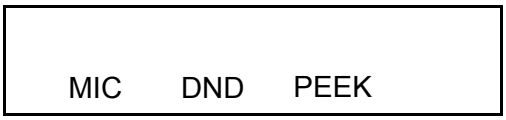
- Prime Line key selected with line number right display

1. A Dterm Station presses the SPEAKER key or handset off hook.
2. Prime Line is selected
3. LCD displays the Prime Line number and the letter “P”.

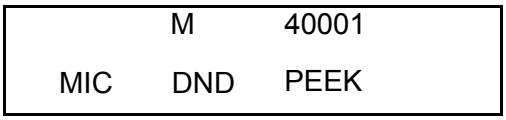


- Idle Condition

1. In the idle status no display.

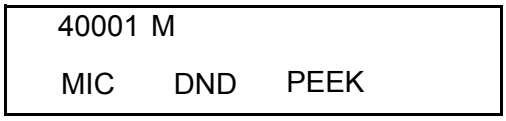


- My Line and Prime Line not split with line number display in center



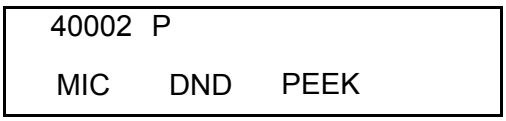
- My Line key selected with line number left display

1. A Desktop terminal station presses the SPEAKER key or handset off hook.
2. LCD displays the My Line or Prime Line number and the letter “M”.

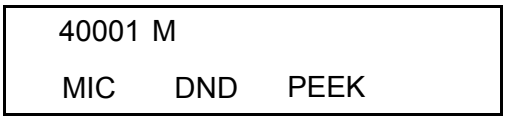


- Prime Line key selected with line number left display

1. A Desktop terminal station presses the SPEAKER key or handset off hook.
2. Prime Line is selected.
3. LCD displays the Prime Line number and the letter “P”.



- My Line and Prime Line not split with line number left display



Service Conditions

1. OFF-HOOK LINE NUMBER DISPLAY -D [O-37D] must be allowed.
2. Sub Lines, other than the Prime Line, will not be designated by a letter.
3. The maximum Station Number length that can be used with My Line Designator is five digits. If the maximum length is exceeded the designation letter will not be displayed.
4. This feature is only available for Dterm.
5. Wireless terminals are not supported for this feature.
6. The letter designator, “M/P” is not assignable.
7. Letter designation is not provided for SOFTWARE MULTIPLE MY LINE APPEARANCE [S-143].
8. This service is not provided for Hotel systems.

Interactions

1. This service is provided when off-hook. In the pre-select status, this is not provided.
2. This service is not provided when a call is originated by using HOTLINE [H-1].
3. The display position of the selected line may be changed by System Data as below.

Programming

ASYD - System Data 1, Index 389, Bits 5 and 6. Location of Off-Hook Line Number Display on the Dterm.

00: Right side (Standard Setting)

01: Center

10: Left side

11: -

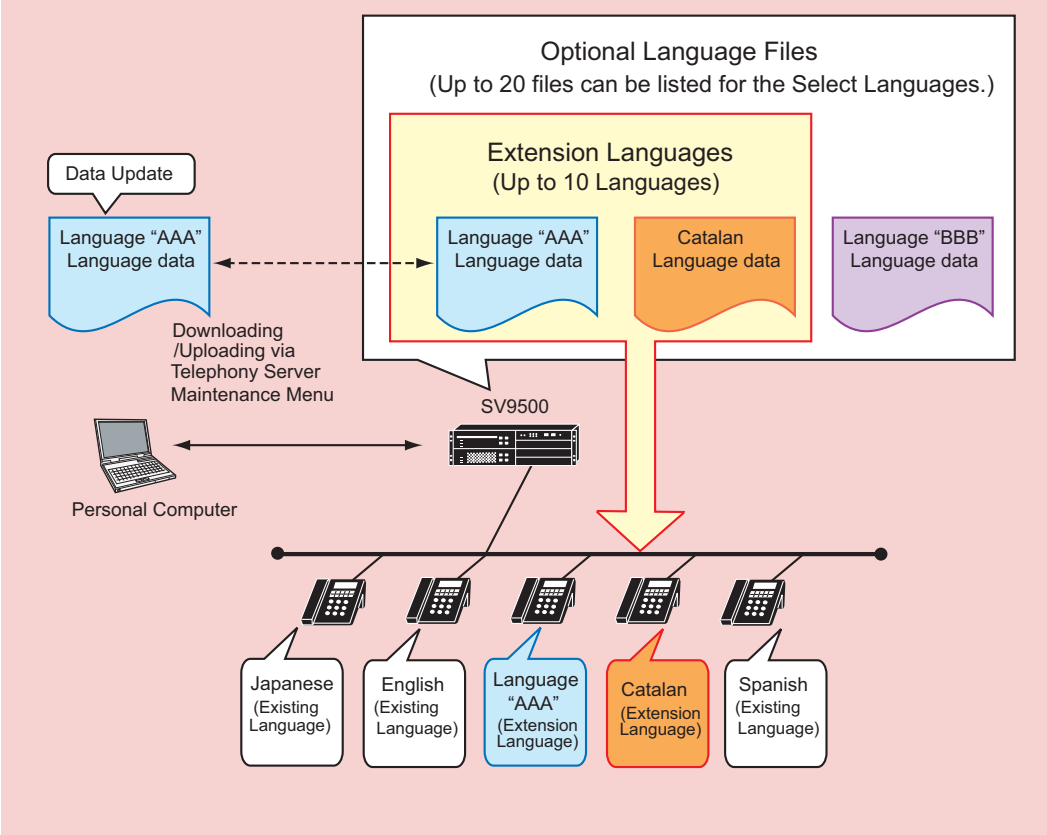
System Data 1, Index 389, Bit 7. Off-Hook Line Number Display -D. 0/1 = Out/In Service. Assign data "1".

ASFC - For Stations to use OFF-HOOK LINE NUMBER DISPLAY -D (US), assign 1 to a Class of Service that allows SFI 235 (My Line Designator).

O-42 Optional Language Pack Loading

General Description

Optional Language Pack Loading offers a quick and easy way to provide additional language options for the telephony network. A maximum of 20 language files can be included in the Select Language list of the system, from which you can select up to 10 to configure the Extension Languages for daily use.
Languages specified for the Extension Languages, as well as the existing languages used in the MULTILINGUAL DISPLAY [M-117] feature, are available in the entire system or on the individual Desktop terminals. You can also easily add or modify the data of the Optional Language files from a personal computer via the Telephony Server Maintenance Menu. To add a new option, simply create and upload the desired language file to SV9500; to modify the data, download and edit the target data then upload it to SV9500.



Operating Procedure

No manual operation is required.

Service Conditions

1. For hardware requirements for the Optional Language Pack Loading feature, refer to MULTILINGUAL DISPLAY [M-117]. The requirements specific to Extension Languages are as follows:
   a.) In the case of DT700 Series, terminals with firmware version 3.0.0.0 or later are supported. (Downloading language files is required.).
   b.) Desktop terminals on which Extension Languages are used must support the corresponding character code. If not, some characters cannot be displayed correctly.
   c.) To support Chinese on Digital Terminal, the following firmware is required.

Equipment		Firmware	Remarks
Terminal	DT330 (DTL-32D-1P)	V2.50 or later	Font version of the terminal must be V2.00 or later.
Circuit Card	CH-16ELCA	Any version	

   d.) The available terminals for using Azerbaijani/Turkish are as follows:
   - DT830G (with firmware version 5.1.1.0 or later)
   - DT830 (with firmware version 5.1.1.0 or later)
   - DT820 (with firmware version 3.1.1.0 or later)

   Note: When using other than the terminals above, some characters cannot be displayed correctly.

2. To use this feature, system data (ASYDL, SYS1, Index 679, Bit 2=1) must be assigned.
3. A maximum of 10 languages can be specified as the Extension Languages.
4. By using this feature, Optional Language files can be newly created. For details of creating or editing Optional Language files, refer to Management in this service.
5. To use a language specified as Extension Language, data assignment with the ASLG command or configuration on a Desktop terminal is required.
6. Some texts are presented in NEC English or the First Language instead of the specified Extension Language when:

   - The Optional Language Pack Loading feature is disabled while the Extension Language is being used.
   - The language data in use is deleted.
   - There are flaws in the data of the Extension Language.

   In this case, specify another language by using the ASLG command or assigning Office data to the Desktop terminal.

7. Data specified on the [EXTENSION] tab of the ASLG command screen is saved in SV9500 in a format different from that of Office Data. Even if Office Data is not saved or another Office data is uploaded, the data of the Extension Languages is retained.
8. Regardless of the settings of system data (ASYDL, SYS1, Index 679, Bit 2=1), if there are data of Extension Languages assigned, when the Telephony Server is started, the data is loaded (read from the [EXTENSION] tab of the ASLG command). However, if the system data is set to out of service (ASYDL, SYS1, Index 679, Bit 2=0), the Extension Language number from the [EXTENSION] tab of the ASLG command cannot be set.

9. Any changes such as adding, editing or removing data in the Extension Languages require restarting the SV9500. When these changes are made using a Desktop terminal, initialization of ELC card is needed.
10. The language to be displayed on the configuration menu of a DT800 Series/DT700 Series terminal is basically a language specified for <Details of Data Class 1-4 Menu Language Number>. However, the use of some languages requires menu download. In this case, when the specified language is not loaded to the Desktop terminal, English is displayed instead. To use a language other than English, select one through "menu" → "Setting" → "User Setting" → "Display" → "Language."

    Note: For languages available on DT800 Series/DT700 Series terminals, refer to <Details of Data Class 1-4 Menu Language Number>.

11. The LCD of IP terminal (Proprietary Protocol) displays date/weekday using a character set specified for the Menu Language Number (Kind 1: Language Data, Data No.4: Menu Language Number), based on the user's Desktop terminal date/weekday (not SV9500 date/weekday). If the character set specified from the SV9500 is not available on the Desktop terminal, date/weekday is displayed in English instead.
12. Extension Languages with Terminal Character Codes other than nec-code (Kana and alphanumeric characters) or ascii (ASCII character set) cannot be specified for the First Language, but must be specified for the Second Language. These character codes include sjis (Shift JIS), nec-code-rus (Cyrillic), euc-cn (Simplified Chinese), and big5 (Traditional Chinese).
13. The conditions for using Azerbaijani and Turkish are as follows:

    Note: These languages are available:
    - from FP95-114 V4 Issue 4.0
    - only for EMEA

    a.) Azerbaijani/Turkish must be used solely for the whole system, it cannot be used in conjunction with other languages including all nodes.

    b.) Assign data with PCPro version 7.1.0 or later. Also, PCPro requires a setting before assigning data. In the PCPro main menu, select "Settings" → "Accept Extensional Characters for"→ "Turkish" (restarting PCPro is required after performing this setting).
    Azerbaijani/Turkish can be used by the following commands. To see the available characters, refer to the Main Menu section of PCPro Window Layout chapter of PCPro Setup Manual.
    - ANDD
    - ANDD_T
    - APBKN
    - AUPDN

    c.) If Azerbaijani/Turkish is used in NAME DISPLAY - SYSTEM [N-28] and PHONEBOOK [P-83] (System Phonebook), assign ASYDL/ASYDN, SYS1, Index 686, Bit 7=1.

    d.) Azerbaijani/Turkish requires in the ASLG command to set Russian as first language and Azerbaijani/Turkish as second language.

    e.) To display on a terminal LCD in Azerbaijani/Turkish, the following steps on the terminal side are also required.
        1. Open the [7.Character Code] menu ([3.Character Code] in case of DT820) by selecting [0.Config] → [3.Maintenance Settings] → [4.Adjust].
        2. Select [1: Eastern Europe].
        3. Save it and exit the menu.

Interactions

1. When using system or features that employ multiple nodes in the network, set all the nodes with the same Extension Language data. Desktop terminals follow the settings of Extension Language implemented in the node to which they register.
2. Changes made in language data with the ASLG command (including the addition or deletion of Extension Language files) are not automatically reflected in SR-MGC. You must manually update the SR-MGC by following the steps outlined below. For detailed procedure, see 1. Download/Upload/Deletion of Language Files.

   Step 1: Download the following data from the SV9500:
   - All language data in the SV9500
     (.txt files in the pattern of "<language name>_<file version>.txt" - for example, "Catalan_0001.txt")
   - Information file of Extension Languages
     (File name: exlang_info)

   Step 2: Upload the data you downloaded in the previous step to the SR-MGC.

   Step 3: In the SINZ command, select **System Initialize** for **Process Kind Select**, and initialize the SR-MGC.

   Note: Data assignment of Extension Languages with the ASLG command is not available on the SR-MGC side.

Programming

For the data assignment to use Optional Language Pack Loading, refer to "Programming" of MULTILINGUAL DISPLAY [M-117].

Management

1. Download/Upload/Deletion of Language Files

This section explains how to download/upload language files (.txt files) between SV9500 and a personal computer by using the Telephony Server Maintenance Menu. Relevant information regarding these procedures is also provided.

1.1 Understanding Optional Language File

Sample files of Optional Language are installed in SV9500 by a setup program on the installation DVD/CD. To update the Optional Language files, use the Telephony Server Maintenance Menu or to download the desired language file from SV9500 for editing the data with a text editor (including Microsoft® Office Excel®), and then upload back.

Note: For details on editing of Optional Language files, see "2. Creation/Editing of Optional Language Data."

1.2 Downloading

The following data can be obtained by downloading from SV9500.

- All Optional Language files stored in the SV9500

  (“<language name>_<file version>.txt” - for example, “Catalan_0001.txt”)

- Information file of Extension Languages **Note 1**

  (File name: exlang_info)

- Sample files of Optional Language in Excel format (for example, “Exlang_0001.xls”)

  (An Optional Language file can be created by saving an Excel spreadsheet in Tab Separated Values (TSV) format.)

Note 1: Note the following conditions for Information file of Extension Languages.
- This file exists only when it is used by the ASLG command and the Optional Language Pack Loading feature is already in use.
- The data of Optional Language file can be backed up by downloading the related language files and Information file to the maintenance console. The saved Optional Language files can be restored by uploading them with the Information file back to the SV9500. To make the change effective, restart the SV9500 (or access the SINZ command and execute [Program Load & Office Data Load & System Initialize] on PCPro) or reconfigure it with the ASLG command.

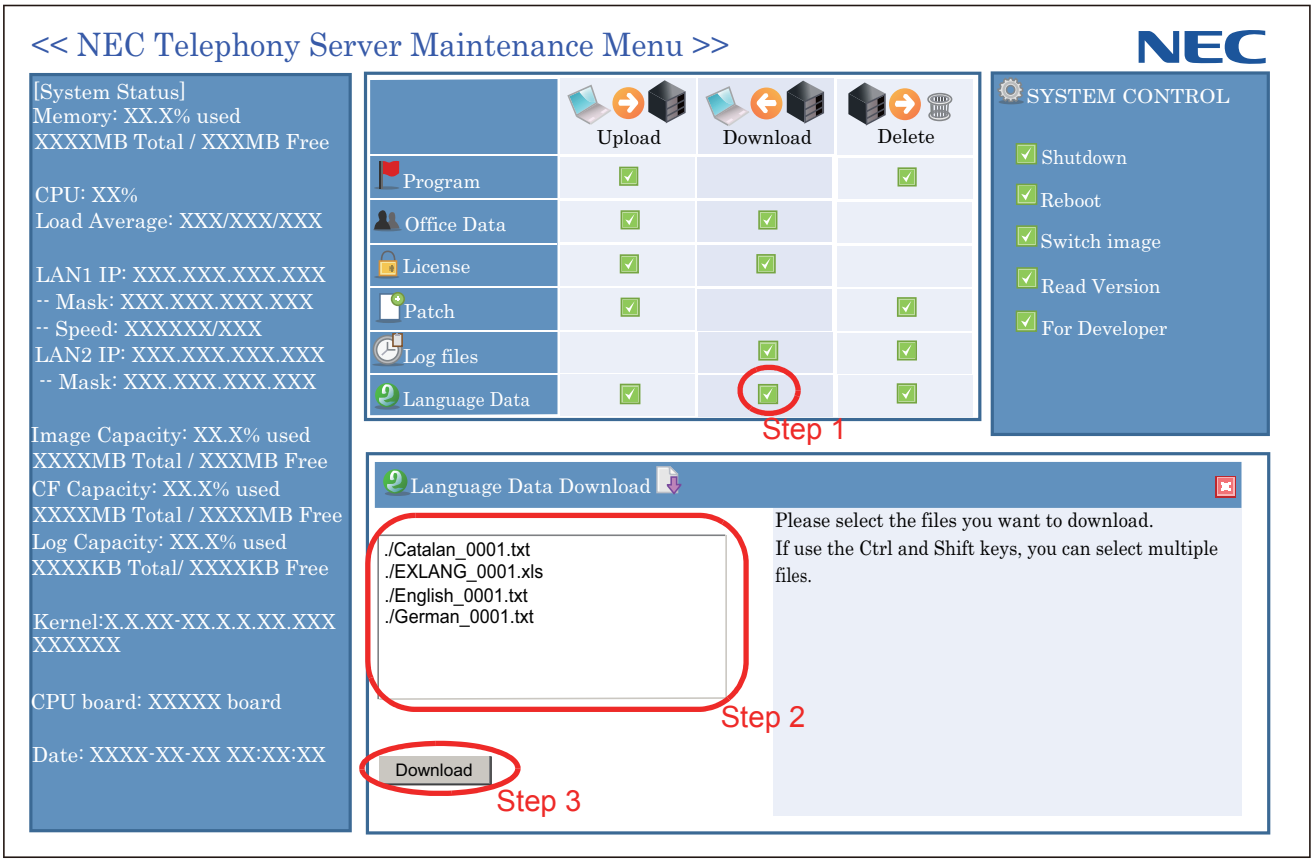
1.3 Working with Optional Language File via Telephony Server Maintenance Menu

Telephony Server Maintenance Menu provides the following functionality:

Functionality	Description
Download of language files	Downloads language files from SV9500 to the personal computer.
Upload of language files	Uploads language files from the personal computer to SV9500.
Deletion of language files from SV9500	Removes language files from SV9500.

<Download of Language Files>

The following is the procedure for downloading language files from SV9500.



STEP 1: On Telephony Server Maintenance Menu, select the check box in the [Download] column of the [Language Data] row.
The [Language Data Download] window opens.

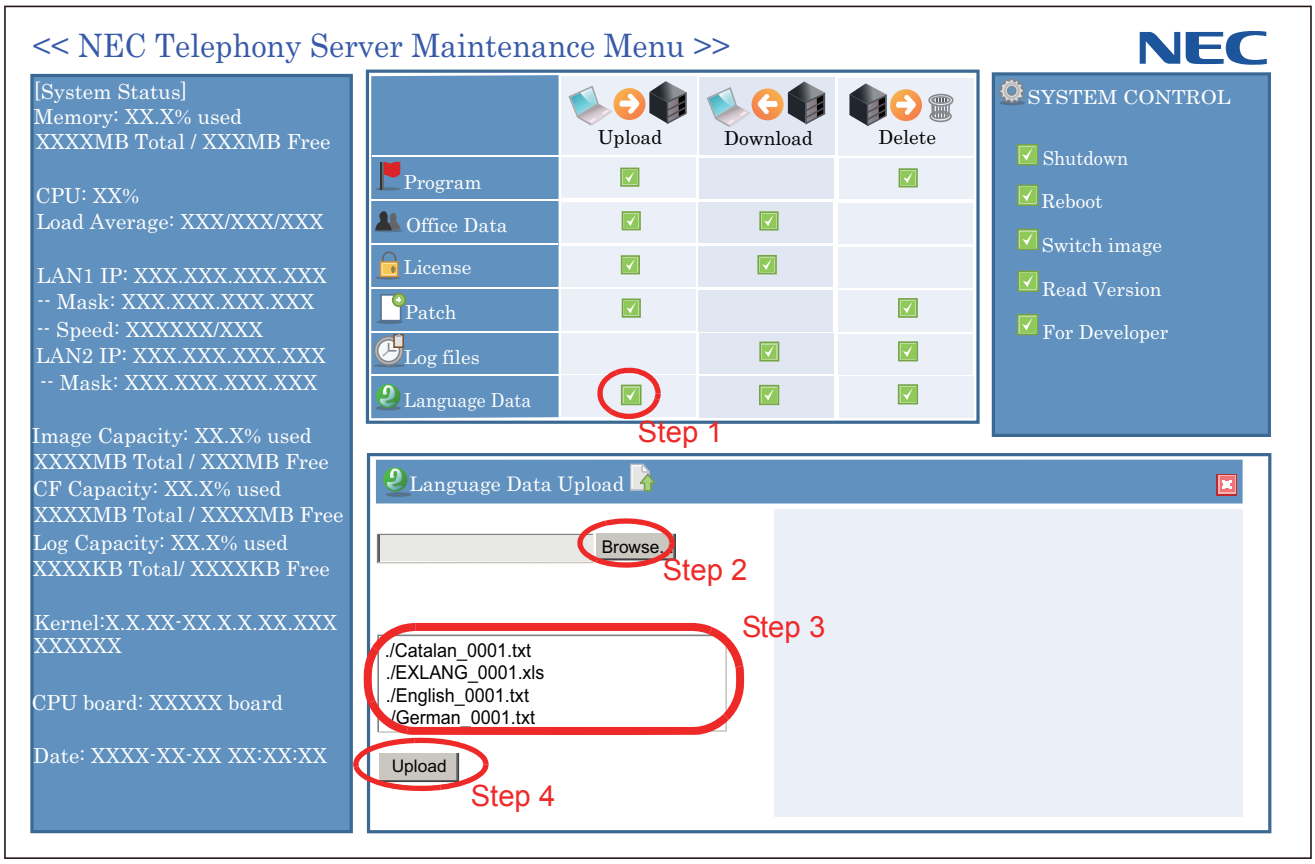
STEP 2: In the [Language Data Download] window, click the desired language file.

STEP 3: Click the [Download] button.
A dialog box appears.

STEP 4: Choose [Save As] in the open dialog box and save the language file in the appropriate folder.

<Upload of Language Files>

The following is the procedure for uploading language files to SV9500.



STEP 1: On Telephony Server Maintenance Menu, select the check box in the [Upload] column of the [Language Data] row.
The [Language Data Upload] window opens.

STEP 2: In the [Language Data Upload] window, click the [Browse] button.
A dialog box appears.

STEP 3: In the open dialog box, select the desired language file and choose [Open].

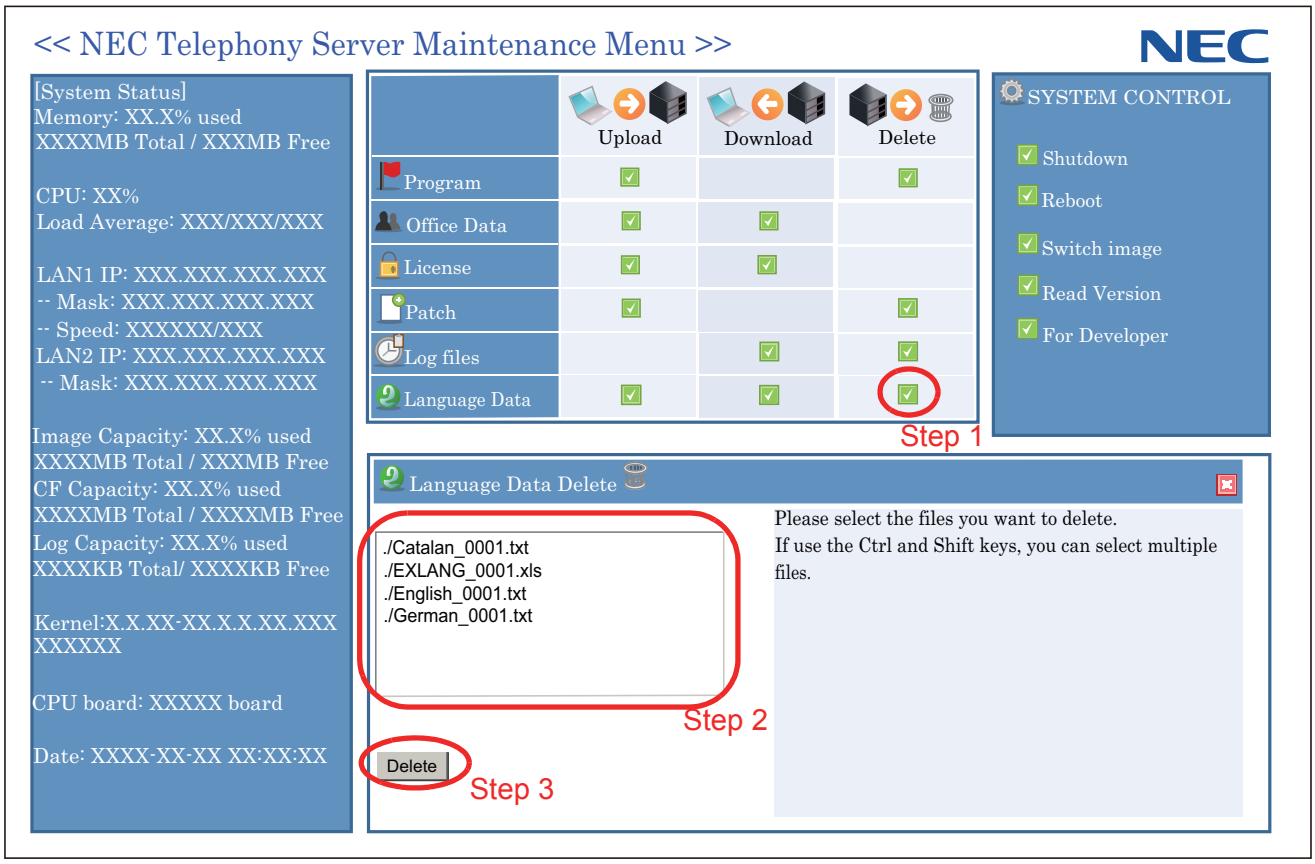
STEP 4: Click the [Upload] button.
When the upload process is completed, a message appears, showing "Operation successfully completed."

STEP 5: Click [OK].

Note: Uploading Optional Language files to SV9500 requires the setting on the [EXTENSION] tab of the ASLG command. Any changes made in an Optional Language file will not take effect until the process is complete.

<Deletion of Language Files from SV9500>

The following is the procedure for deleting language files from SV9500.



STEP 1: On Telephony Server Maintenance Menu, select the check box in the [Delete] column of the [Language Data] row.
The [Language Data Delete] window opens.

STEP 2: In the [Language Data Delete] window, click the desired file.

STEP 3: Click the [Delete] button.
A confirmation message appears, asking “Are you sure you want to delete?”

STEP 4: Click [OK].
A message appears, showing “Operation successfully completed.”

STEP 5: Click [OK].

Note: If all language files are deleted, the data of the Optional Language Pack Loading return to their original state after installation.

2. Creation/Editing of Optional Language Data

This section explains how to create and edit language files used for the Optional Language Pack Loading feature.

2.1 Creating/Editing Optional Language Data

The following is the outline of the entire procedure:

STEP 1: Obtaining language files

When creating Optional Language files, download Sample files of Optional Language in Excel format (for example, Exlang_0001.xls), following the instructions given in "1. Download/Upload/Deletion of Language Files."

When editing the existing Optional Language files, download the desired.txt files.

STEP 2: Creating/Editing Optional Language files

Create or edit Optional Language files by referring to "2.2 Conditions of Optional Language Data" and "2.3 Information about Optional Language Data."

STEP 3: Saving the created/changed Optional Language files

Save the created or changed Optional Language files referring to "2.4 Creating Optional Language Data from Excel."

STEP 4: Uploading language files to SV9500

Upload the language files to the SV9500, following the instructions given in "1. Download/Upload/Deletion of Language Files."

2.2 Conditions of Optional Language Data

1. The Optional Language files must be in Tab Separated Values (TSV) format. (The file extension is '.txt'.) To create a new Optional Language file, use a Sample file of Optional Language, which has been installed in SV9500 by a setup program on the installation DVD/CD.

   Note: When the Optional Language file is updated, increase the version number of the file.

2. Text in an Optional Language file is represented in ASCII (including nec-code and nec-code-rus) or sjis based on the Terminal Character Code in language data. If text contains characters not representable in the specified character set, the characters are displayed as question marks on the LCD of a Desktop terminal.

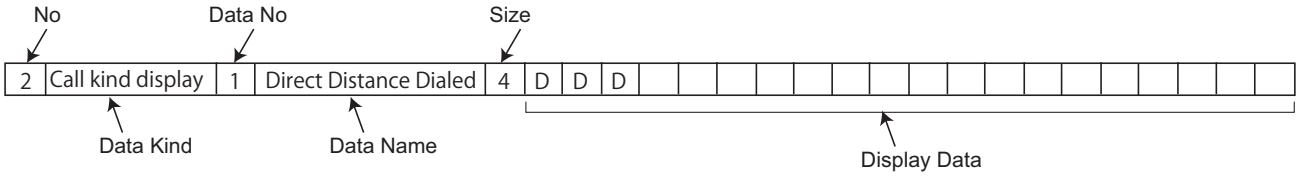
   Note: For available character codes, refer to <Details of Data Class 1-4 Menu Language Number>.

3. Do not use blank spaces in the file name of Optional Language file.
   (Example)
   Unusable: Exlang 0001.txt
   Usable: Exlang_0001.txt
4. In this feature, character set in language data is automatically encoded. However, a character set consisting of very few characters may not be encoded correctly. In this case, enter the characters of the source character set in the comment field of the language data.
5. Multiple character codes cannot be specified for the same Optional Language data.

2.3 Information about Optional Language Data

<Data in Optional Language File>

Optional Language files are created in Tab Separated Values (TSV) format, which can be edited with Microsoft® Office Excel® (hereinafter referred to as Excel) or text editor. The following is an example of Excel file format.



Field	Editing	Description
No	N	Data class number. e.g. 1 = Language_Data, 2 = Call_kind_display, 3 = Display_Data_4
Data Kind	X	Data class name. e.g. Language Data, Call_kind display, Display Data
Data No	N	Serial number within a data class.
Data Name	X	Data name. e.g. Direct Distance Dialed, Foreign Exchange
Size	N	Maximum data size.
Display Data	X	Text to be displayed. • Data to be interpreted as binary values should be entered in hexadecimal notation by adding 0x as the prefix of the number. • Text of multi-byte characters must be left-aligned.

X: Available N: Not Available

<Example of Optional Language File>

The following are examples of language files (of Catalan) in Excel format and in Tab Separated Values (TSV) format, respectively:

- Excel file

Kind	Kind Name	Data Number	Data Name	Size	Display Data										
1	Language Data	0	Language Number	4	0	0	0	2							
1	Language Data	1	Language Name **Note 1**	16	C	a	t	a	l	a	n				
1	Language Data	2	Language Version	4	0	0	0	1							
1	Language Data	3	Character Code	1	0x00										
1	Language Data	4	Menu Language Number	1	0x13										
2	Call kind display	0	Nop	4											
2	Call kind display	1	Direct Distance Dialed	4	D	D	D								
2	Call kind display	2	Foreign Exchange	4	F	X									

Note 1: Use only alphanumeric characters.

- Text file in TSV format

Kind (t) Kind Name (t) Data Number (t) Data Name (t) Size (t) Display Data

1 (t) Language Information (t) 0 (t) Language_Number (t) 4 (t) 0 (t) 0 (t) 0 (t) 2

1 (t) Language Information (t) 1 (t) Language_Name (t)16 (t) C (t) a (t) t (t) a (t) l (t) a (t) n

1 (t) Language Information (t) 2 (t) Language_Version (t) 4 (t) 0 (t) 0 (t) 0 (t) 1

1 (t) Language Information (t) 3 (t) Character_Code (t) 1 (t) 0x01

1 (t) Language Information (t) 4 (t) Menu Language Number1 (t) 1 (t) 0x12

2 (t) Call_kind_display (t) 0 (t) Nop (t) 4

2 (t) Call_kind_display (t) 1 (t) Direct_Distance_Dialed (t) 4 (t) D (t) D (t) D (t)

2 (t) Call_kind_display (t) 2 (t) Foreign_Exchange (t) 4 (t) F (t) X

Note: "(t)" in the above image means a tab separating each field.

<Details of Data Class 1-4 Menu Language Number>

Data class "1: Language Data" of the language file contains information on Optional Language settings. The details of each parameter are presented below:

No.	Data No.	Description
1	0	Language Number [0000 - 9999] Identifier (ID) of Optional Languages. Specify the number corresponding to the target Optional Language. If the desired language is not listed, select an available number from 1000 through 9999.
		0000 = - (Not available.) 0001 = English 0002 = Catalan 0003 - 000E = Reserved. 000F = Azerbaijani 0010 - 0999 = Reserved. 1000 - 9999 = User-defined.
1	1	Language Name (the name of Optional Language.) Specify the Optional Language name, which is displayed in the ASLG command window.
1	2	Language Version [0000 - 9999] When the Optional Language file is updated, increase the version number of the file, which is also displayed in the ASLG command window.
1	3	Terminal Character Code [0x00 - 0xFF] Specify the character code, using the following binary values (in hexadecimal format by adding 0x as the prefix of the number.) 0x00 = nec-code (Kana and alphanumeric characters) 0x01 = nec-code-rus (Cyrillic and alphanumeric characters) **Note 1** 0x02 = ascii (ASCII character set) 0x03 = sjis (Shift JIS character set) 0x04 = unicode (Reserved for Unicode) **Note 2** 0x05 = euc-cn (GB2312) (Simplified Chinese characters) 0x06 = big5 (Traditional Chinese characters) 0x07 - 0xFF = Reserved. **Note:1** Assign this data for Azerbaijani/Turkish. **Note:2** Not available.

<table>
<tr><th>No.</th><th>Data No.</th><th colspan="2">Description</th></tr>
<tr><td rowspan="2">1</td><td rowspan="2">4</td><td colspan="2">Menu Language Number [0x00 - 0xFF]
Specify the menu language number, using the following binary values (in hexadecimal format by adding 0x as the prefix of the number.)

Note: Specify the number corresponding to the languages displayed on the menu screen of DT800/DT700 Series. If the applicable language is not listed, choose “0x00 = English.”</td></tr>
<tr><td>0x00 = English
0x01 = Portuguese
0x02 = Spanish
0x03 = Canadian French
0x04 = German
0x05 = Italian
0x06 = Dutch
0x07 = Norwegian
0x08 = Danish
0x09 = Swedish
0x0A = Greek</td><td>0x0B = NEC English
0x0C = NEC Portuguese
0x0D = NEC Spanish
0x0E = French (for NECP)
0x0F = Russian
0x10 = Turkish
0x11 = Romanian
0x12 = Polish
0x13 = Catalan
0x14 = Azerbaijani
0x15 - 0x00FE = Reserved.
0xFF = Japanese</td></tr>
</table>

<Call History Display for Answered/Unanswered Call>

The following is the procedure for setting Call History to be displayed on LCD. Call History Display data for answered calls and unanswered calls can be set separately. Specify “52: Call History” as data class of the language file when you set Call History Display data. The procedure for setting the display data differs according to whether “52: Call History” is specified or not.

- When data class “52: Call History” is specified

  Set characters to be displayed on LCD for Data Number “0: Answered” and “1: Missed”.

- When data class “52: Call History” is not specified

  Set the following for Data Kind“52”.

  [LCD display for answered call]

  Kind 52
  Kind Name: Call History
  Data Number: 0
  Data Name: Answered
  Size: 8
  Display Data: Text to be displayed (“Answered” in the example below)

[LCD display for unanswered call]

Kind: 52
Kind Name: Call History
Data Number: 1
Data Name: Missed
Size: 8
Display Data: Text to be displayed (“Missed” in the example below)

The following are examples of language files in Excel format and in Tab Separated Values (TSV) format, respectively:

- Excel file

Kind	Kind Name	Data Number	Data Name	Size	Display Data										
:	:	:	:												
51	Softkey Display for Call History (Default)	0	Softkey(<<<)	6	<	<	<								
51	Softkey Display for Call History (Default)	1	Softkey(<<<)	6	>	>	>								
51	Softkey Display for Call History (Default)	2	Softkey(DELETE)	6	E	S	B	O	R	R					
51	Softkey Display for Call History (Default)	3	Softkey(END)	6		F	0x13								
52	Call History	0	Answered	8	A	n	s	w	e	r	e	d			
52	Call History	1	Missed	8	M	i	s	s	e	d					

Note: Add data class “Call History” to Kind “52” of HELP Data Kind worksheet. Some cells of sample files of Optional Language in Excel format (for example, Exlang_0001.xls) may be protected. When creating a language file, cancellation of protection for worksheet is necessary before editing the language file. To cancel the protection, click on Tools-Protection-Protect Sheet. After editing the file, protect the worksheet again.

- Text file in TSV format

```
Kind (t) Kind Name (t) Data Number (t) Data Name (t) Size (t) Display Data
:
51 (t) Softkey Display for Call History (Default) (t) 0 (t) Softkey (<<<) (t) 6 (t)  (t) < (t) < (t) <
51 (t) Softkey Display for Call History (Default) (t) 1 (t) Softkey (>>>) (t) 6 (t)  (t) > (t) > (t) >
51 (t) Softkey Display for Call History (Default) (t) 2 (t) Softkey (DELETE) (t) 6 (t) E (t) S (t) B (t) O (t) R (t) R
```

51 (t) Softkey Display for Call History (Default) (t) 3 (t) Softkey (END) (t) 6 (t) F (t) 0x13

52 (t) Call History (t) 0 (t) Answered (t) 8 (t)

52 (t) Call History (t) 1 (t) Missed (t) 8 (t)

Note: "(t)" in the above image means a tab separating each field.

2.4 Creating Optional Language Data from Excel

An Optional Language file can be created by saving an excel spreadsheet of Optional Language Sample file in Tab Separated Values (TSV) format (*.txt) or Unicode format (*.txt).

<Optional Language File Creation>

- From the Excel menu, click [File] → [Save As] to display [Save As] dialog box, and choose a file type, either tab-separated file format or Unicode format.
- The file name must be 16 alphanumeric characters (including underscore(_)) or less.

  For example, <language name>_<version>.txt (Catalan_0001.txt)

- Rename the file by changing the file name or version number.

Note: If you update an Optional Language file without changing its name, the new information may not be reflected.

3. Example of Procedures for Multilingual Display with Extension Languages

This section describes the steps for:

- Adding Extension Language Files
- Updating Extension Language File (Customized Language File)
- Updating Extension Language File (Sample Language File)

3.1 Adding Extension Language Files

The following is an example of procedure to add two new Extension Language files, Catalan and English2. In this example, assume that English2 is a customized file from a sample language file (English).

Note: The following steps (STEP 1: to STEP 3:) are required only when a existing system is upgraded. You can begin with STEP 4:, If the system version remains the same.

STEP 1: Initialization of a Compact Flash (CF) card
Initialize a CF card, referring to Appliance Model Installation Manual.

Note: Back up the data of CF card if necessary.

STEP 2: Installation of the program through Telephony Server Maintenance Menu
Install the program on Telephony Server Maintenance Menu, referring to Appliance Model Installation Manual.

STEP 3: Restarting the system from Telephony Server Maintenance Menu
[1] Access Telephony Server Maintenance Menu.
[2] From [System Control], select [Shutdown] or [Reboot].
If you select [Shutdown], turn on the power of the system after shutdown.

STEP 4: Transferring language data from the CF card to a personal computer
Use Telephony Server Maintenance Menu to transfer language data from CF card to the personal computer.
Fore more information, refer to <Download of Language Files> in "1.3 Working with Optional Language File via Telephony Server Maintenance Menu."

STEP 5: Editing the language data
Use Excel or text editor to edit language data.
Fore more information, refer to "2. Creation/Editing of Optional Language Data".
- Catalan: Catalan_0002.txt
- English2: English2_001.txt **Note 1**

Note 1: If you add English2 (English2_001.txt) to Extension Language, text is presented in mixed case. However, on terminals under IPG/UG50, month and day do not appear in mixed case.

STEP 6: Transferring the language data from the personal computer to the CF card

Use Telephony Server Maintenance Menu to transfer language data from the personal computer to CF card.

Fore more information, refer to <Upload of Language Files> in "1.3 Working with Optional Language File via Telephony Server Maintenance Menu."

STEP 7: Assignment of the following Office Data

[1] ASYDL/ASYDN SYS1 Index 679 Bit 2=1.

[2] Click the [EXTENSION] tab on the ASLG command and click the [Get] button.
Specify an Extension Language and click the [Set] button.

[3] Click the [FIRST] tab and specify a first language to be used in the system.

Note: Extension Languages with Terminal Character Codes other than nec-code (Kana and alphanumeric characters) or ascii (ASCII character set) cannot be specified for the First Language, but must be specified for the Second Language. These character codes include sjis (Shift JIS), nec-code-rus (Cyrillic), euc-cn (Simplified Chinese), and big5 (Traditional Chinese).

[4] Click the [SECOND] tab and specify the setting.
Second Language field: Select "Select Terminal".
LANG NO: Specify a language number for second language.

[5] Click the [TERMINAL] tab and [Unit] tab and specify each setting.

STEP 8: Backing up the Office Data

Back up the Data Memory with the MEM_HDD/HDD_MEM_N command.
Direction Select: MEM->HDD
Data Type Select: Data Memory

STEP 9: Initialization of SV9500 to apply the changes

Select "System Initialize" on the [TYPE] tab of the SINZ/SINZN command.

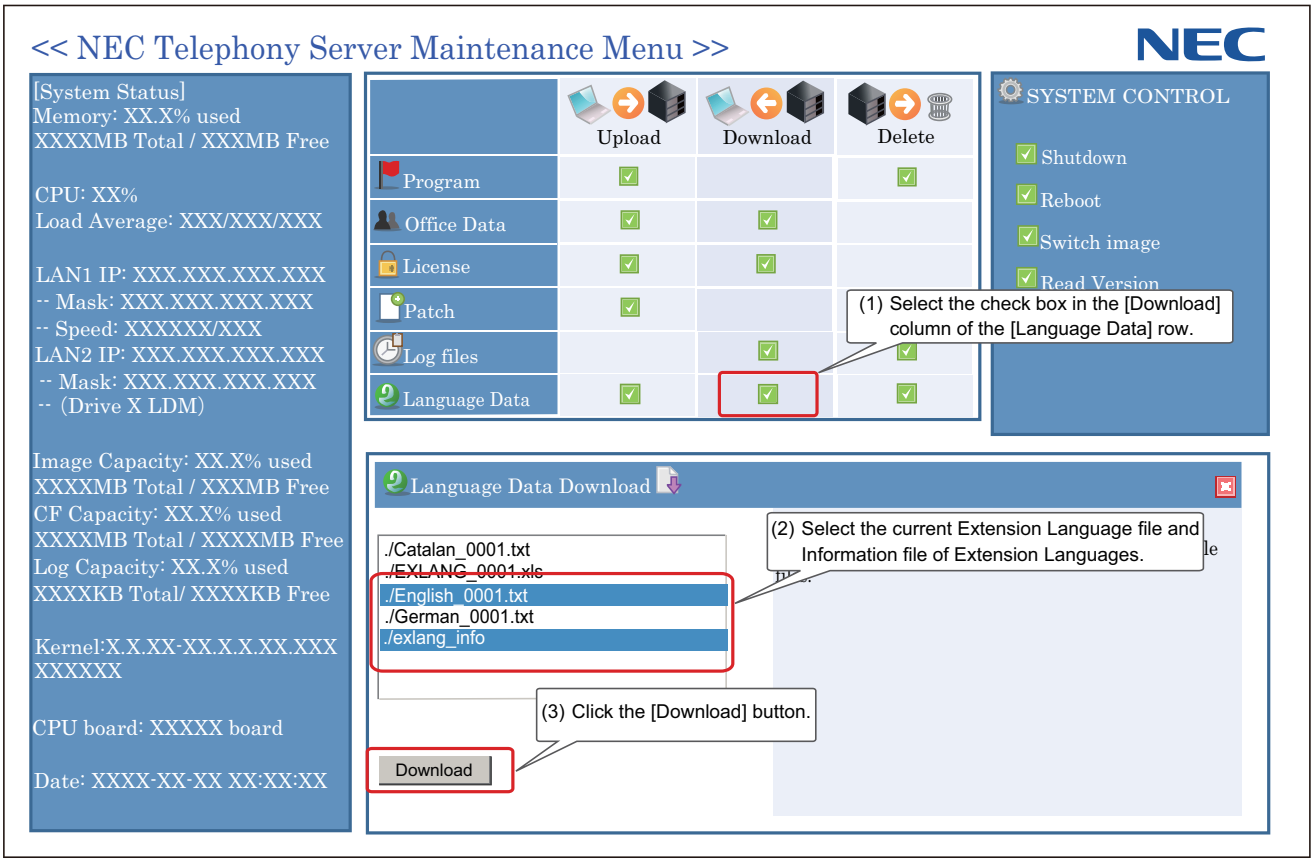
3.2 Updating Extension Language File (Customized Language File)

The following is an example of procedure to update a customized language file with the latest data. In this example, assume that English2 is modified from a Sample Language file (English).

Note: If you want to update a sample language file, see "3.3 Updating Extension Language File (Sample Language File)."

STEP 1: Creating a current language data backup prior to program update

By referring to "1. Download/Upload/Deletion of Language Files," download the current Extension Language file (English_0001.txt) and Information file of Extension Languages (exlang_info).

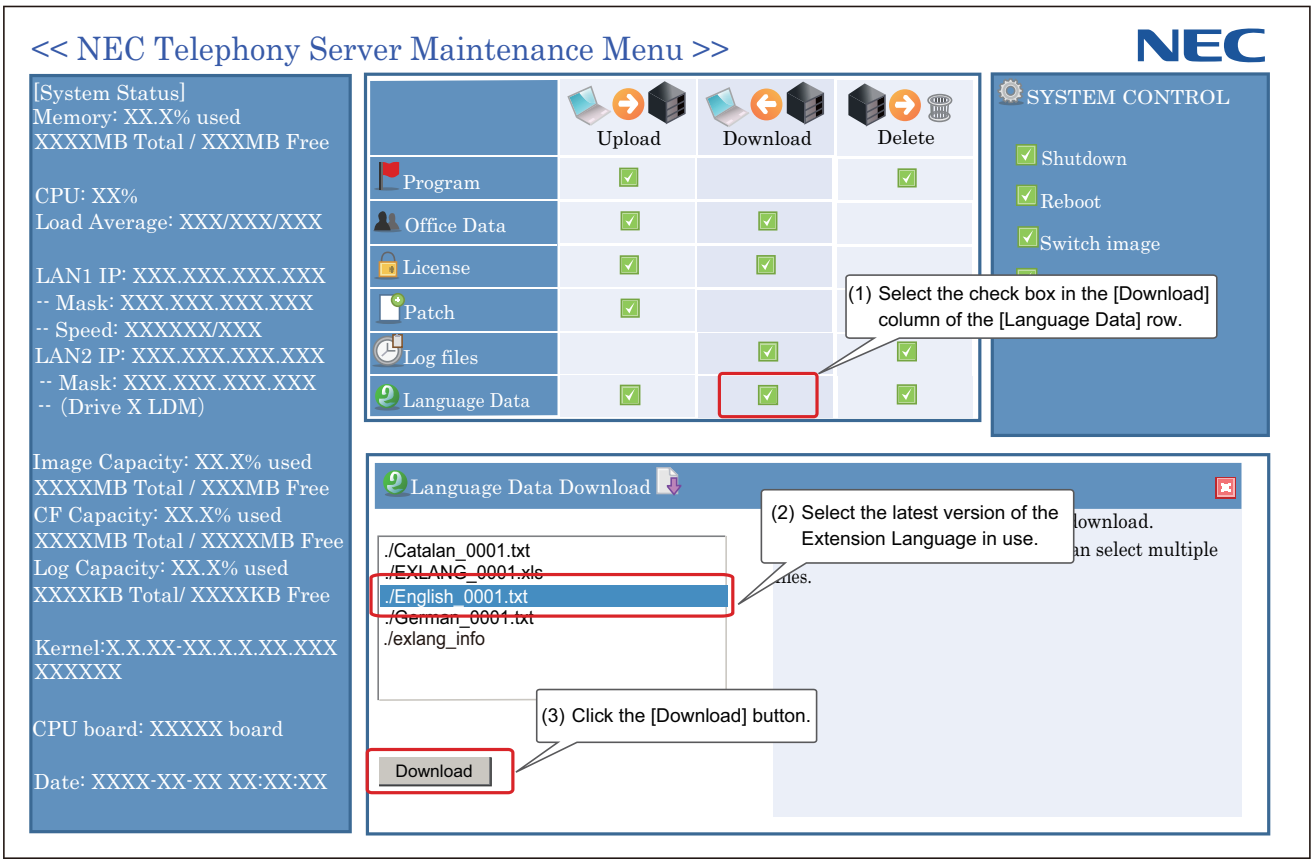


STEP 2: Downloading the latest Extension Language file after the program update

After updating the program, download the latest version of the Extension Language in use (English_0001.txt).

If necessary, download a Sample file of Optional Language in Excel format (EXLANG_0001.xls), and create new language data by referring to "2.4 Creating Optional Language Data from Excel."

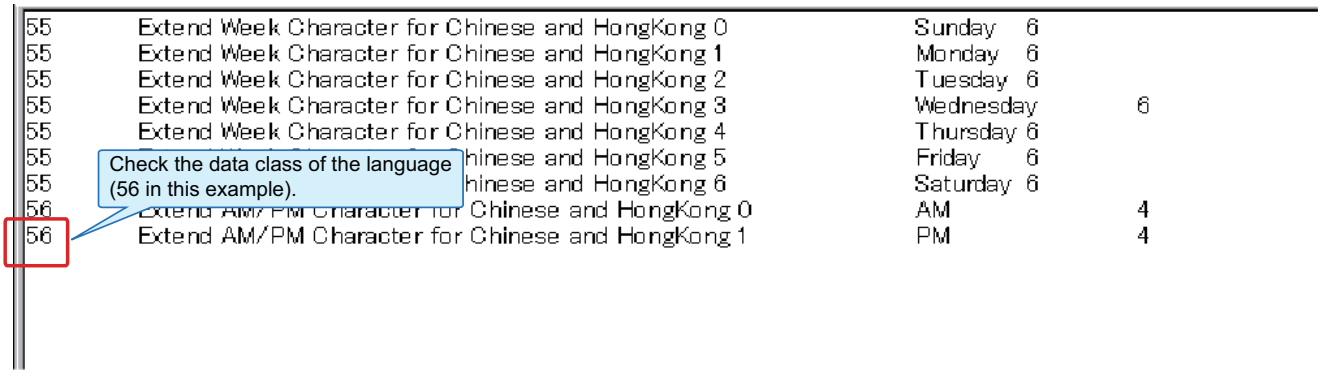
Note: Be careful not to overwrite the backup of the language file created in Step 1.



STEP 3: Checking the data class of the language in the last line of the current Extension Language file

[1] Open the backup of the current Extension Language file with a text editor, and scroll down to the last line.

[2] Write down or memorize the data class number of the last line (in the leftmost column of the line).

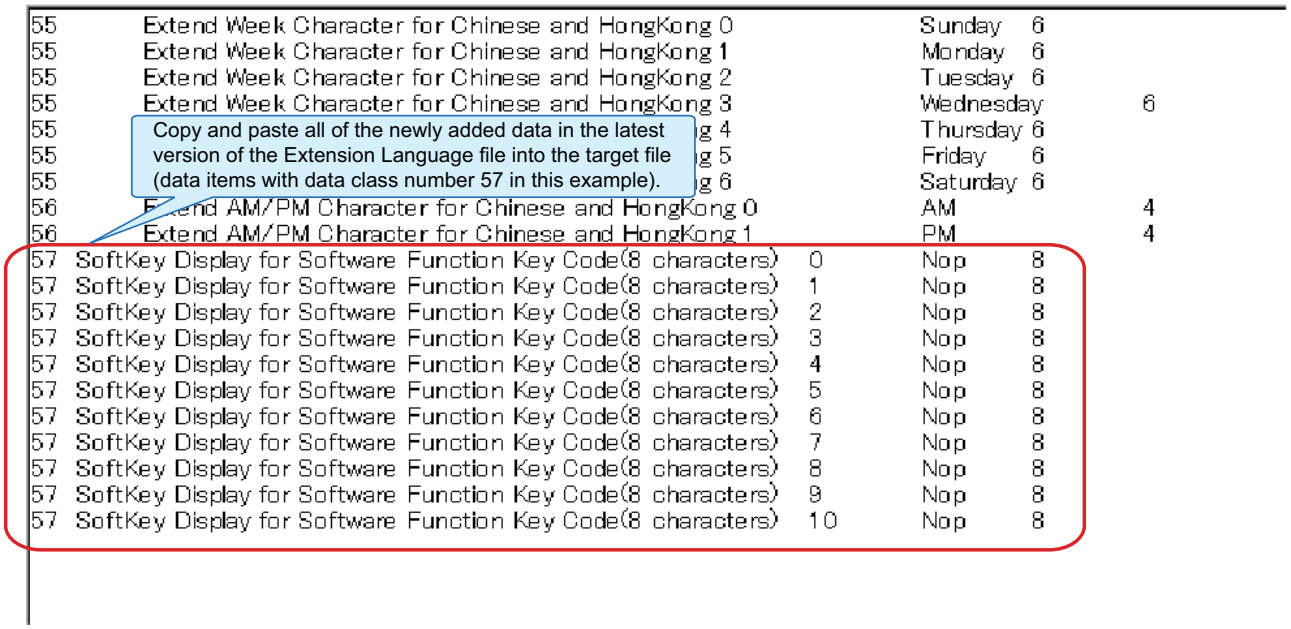


STEP 4: Updating the current Extension Language file with new information

[1] Duplicate the backup of the current Extension Language file and save it in a different folder. The duplicated file is the target file to which the update is to be applied.

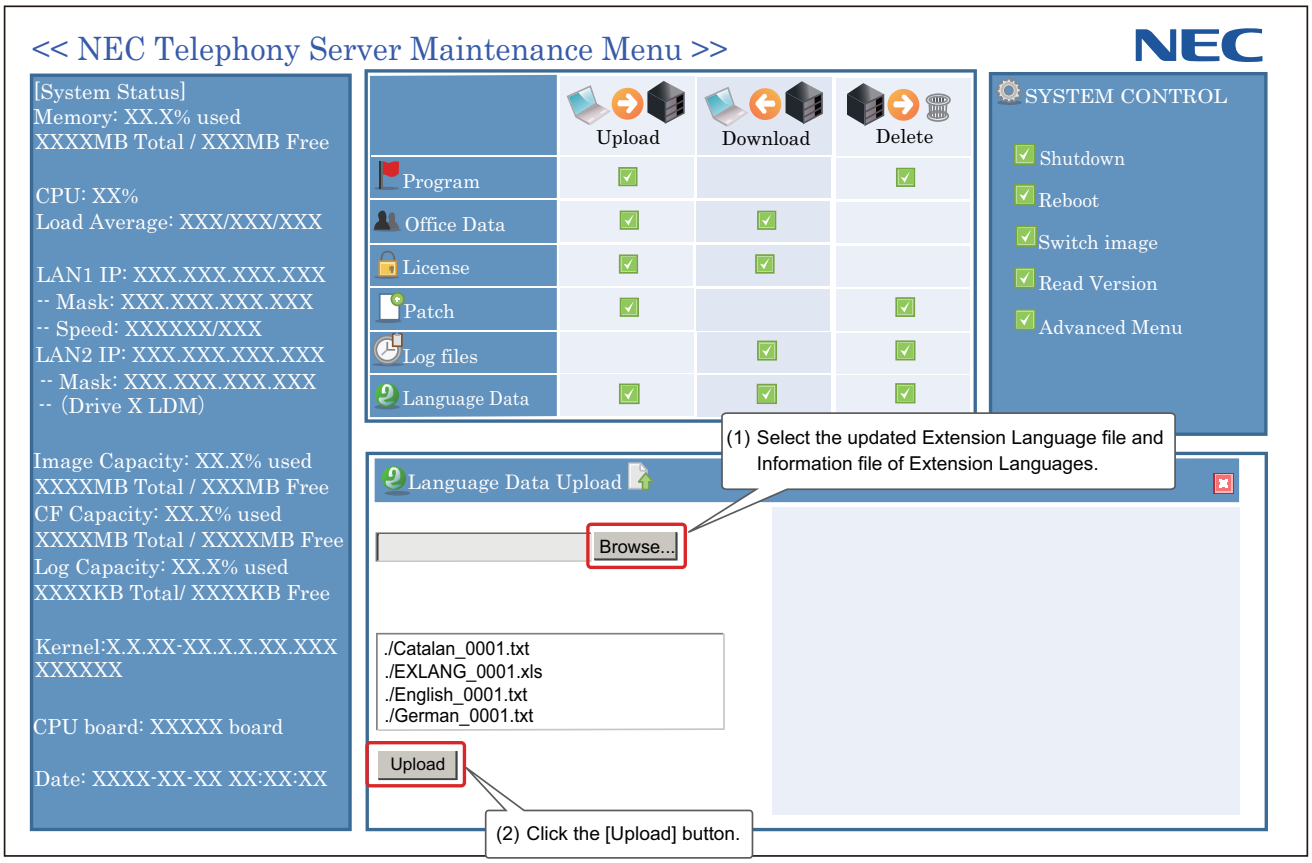
[2] Open the latest Extension Language file (English_0001.txt) with a text editor.

[3] Copy and paste all of the newly added data in the latest version of the Extension Language file into the target file, then save it.



STEP 5: Uploading the updated Extension Language file

Upload the updated Extension Language file (target file) and the Information file of Extension Languages (exlang_info) to the SV9500.



STEP 6: Verifying the integrity of the Extension Language settings

[1] Log in to the SV9500 through PCPro.

[2] Go to the ASLG command, then select the [EXTENSION] tab.

[3] Click the [Get] button, and verify that no errors exist.

Note: System initialization is required to make changes to Extension Language data effective. The changes do not take effect when the modified data is uploaded to SV9500.

STEP 7: Verifying the validity of the update

[1] From PCPro, initialize the SV9500 (System Initialize) with the SINZ command to reflect the changes.

[2] After the system initialization, log in to the SV9500 through PCPro.

[3] Repeat the process in Step 6, in order to verify that the update has been applied successfully.

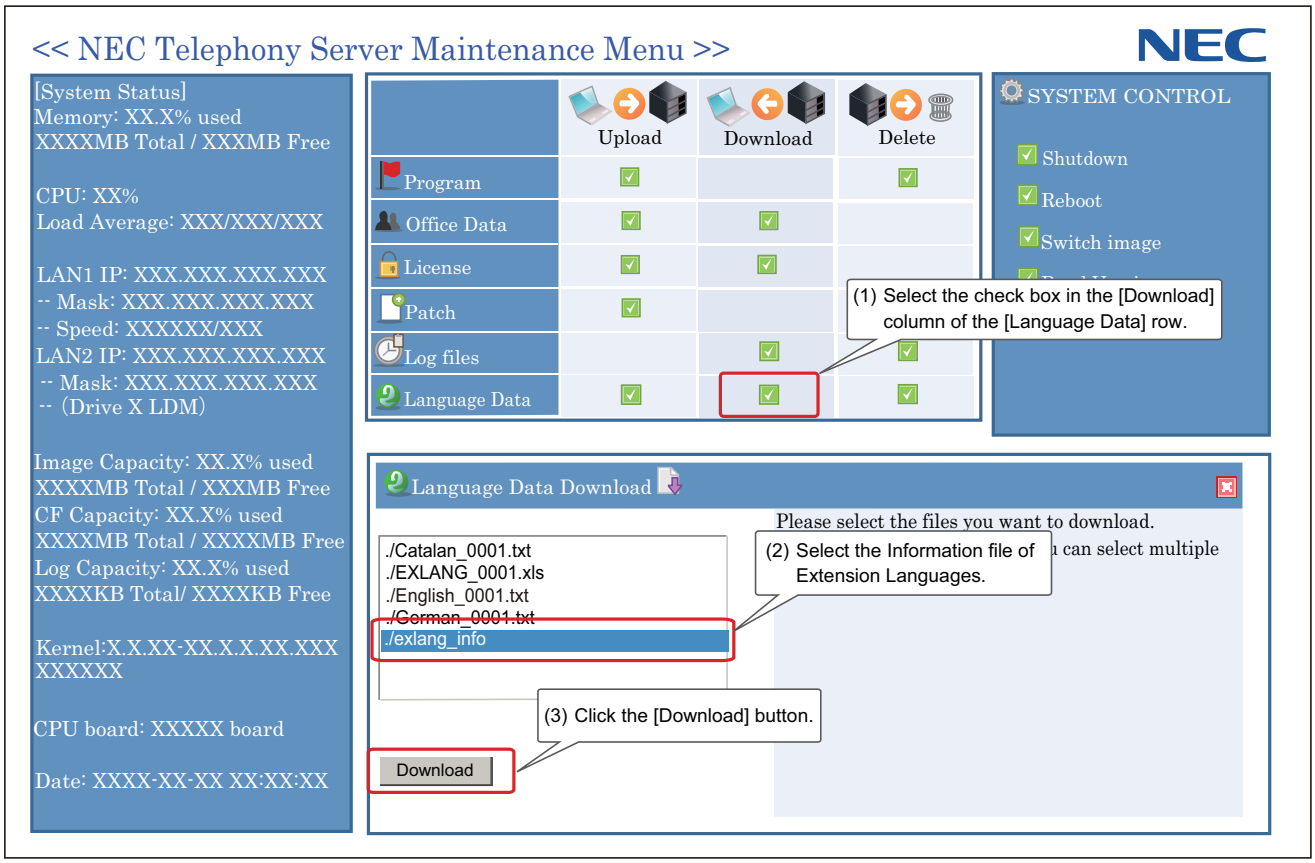
3.3 Updating Extension Language File (Sample Language File)

The following is an example of procedure to update a Sample Language file with the latest data.

Note: If you want to update a customized language file, see "3.2 Updating Extension Language File (Customized Language File)."

STEP 1: Creating a current language data backup prior to program update

By referring to "1. Download/Upload/Deletion of Language Files," download the current Information file of Extension Languages (exlang_info).



STEP 2: Uploading the Information file of Extension Languages after the program update

After updating the program, upload the Information file of Extension Languages (exlang_info) to the SV9500.

STEP 3: Verifying the integrity of the Extension Language settings

[1] Log in to the SV9500 through PCPro.

[2] Go to the ASLG command, then select the [EXTENSION] tab.

[3] Click the [Get] button, and verify that no errors exist.

Note: System initialization is required to make changes to Extension Language data effective. The changes do not take effect when the modified data is uploaded to SV9500.

STEP 4: Verifying the validity of the update

[1] From PCPro, initialize the SV9500 (System Initialize) with the SINZ command to reflect the changes.

[2] After the system initialization, log in to the SV9500 through PCPro.

[3] Repeat the process in Step 3, in order to verify that the update has been applied successfully.

P-1 Paging Access

General Description

This feature provides both ATTENDANT CONSOLE [A-3] and Station users with dial access to PAGING equipment so that they can page anyone over the speaker. PAGING equipment is configured as the Paging Trunk and connected using an interface.

Operating Procedure

To operate

1. Go off-hook. Receive Dial Tone.
2. Dial the Paging Access Code. Receive Ringback Tone for several seconds.
3. Connection to Paging equipment is established.
4. Page anyone over the speaker.
   - Non-delay operation → Wait with holding the handset
   - Delay operation → Hang up and wait

To answer

1. The called someone lifts the handset of any station, and receives Dial Tone.
2. Dial the Access Code of Paging Answer.
   - Non-delay operation → Immediately connected to the calling party.
   - Delay operation → Connected to the calling party after hearing Ringback Tone.

Service Conditions

1. Amplifiers and speakers must be customer provided.
2. One Trunk Circuit is required for each Paging Access.
3. Where TENANT SERVICE [T-12] is provided, separate Access Codes and hardware should be provided for each Tenant.
4. Maximum Paging Access Code is six digits.
5. ALL ZONE PAGING [A-53]: If the system provides Multi Zone Paging with a separate direct Access Code assigned for each area, Stations or Attendants can use All Zone Paging by dialing a special Access Code common for all Paging systems.
6. PRIORITY PAGING [P-30]: A predetermined Station or the Attendant can preempt a Paging Trunk that is connected to any Station.
7. Loop signaling is the recommended signaling type.
8. The following can be used as a paging trunk interface:

   - COT Circuit Cards
   - TLT Circuit Cards
   - MC&MG-COT (4LC2COT)
   - MG-COT (6COT)
   - UG50 (GCD-PGTA) **Note 1 Note 2**

   Note 1: UG50 (GCD-4COTA/B/C) cannot be used as a paging trunk interface.

Note 2: Available from UG50 software version 2.00.

9. PAGING ACCESS [P-1], PAGING TRANSFER [P-10] and RADIO PAGING [R-16] are provided to the paging Station and the paged Station in FCCS Nodes.
10. When PAGING ACCESS [P-1]/PAGING TRANSFER [P-10]/RADIO PAGING [R-16] service is set to a Station in the Remote Node or the PAGING ACCESS [P-1]/PAGING TRANSFER [P-10]/RADIO PAGING [R-16] call is answered from the Remote Node via FCCS link:
    a.) the Logical Route Number should be applied to each physical route that provides the service.
    b.) the restriction of Route-to-Route Connection between FCCS Nodes should be released.
11. The same paging route cannot be used by another Station while the paged Station does not answer the page activated using the route. However, the paging activation is automatically cleared when the call is not answered within a predetermined time.
12. When using a FCCS network:
    a.) In case of setting the office data with LDM, a maximum of six routes per node can be provided with paging trunks, however, this service cannot be used between nodes.
    b.) In case of setting the office data with NDM, a maximum of six routes per FCCS network can be provided with paging trunks. Also, in a FCCS network, the maximum number of concurrent paging for PAGING ACCESS, RADIO PAGING [R-16] and PAGING TRANSFER [P-10] is 20.
13. A user can page one Station at a time.
14. Follow the conditions below to use this feature with MC&MG-COT.
    a.) Radio Paging feature is not supported.
    b.) One Paging Trunk is used per MC&MG-COT equipment.
    c.) MG (COT) side accommodates speech signal to LINE0/MG0, and activation signal (loop activation) to LINE1/MG1.
    d.) The power for activation signal is not supplied from MC&MG-COT.
    e.) C.O. line interface cannot be accommodated when MC&MG-COT is used by Paging features. However, MC function can be used to connect analog telephone to TEL2/MC2 and TEL3/MC3. Do not use TEL0/MC0 and TEL1/MC1. When PFT is activated, TEL0/MC0 and TEL1/MC1 are connected to LINE0/MG0 and LINE1/MG1.
15. Follow the conditions below to use this feature with MG-COT [SCA-6COTA]/MG-COT [SCA-6COTB]/ MG-COT [SCA-6COTC].
    a.) Radio Paging feature is not supported.
    b.) One Paging Trunk is used per MCMG/MG-COT.
    c.) MCMG/MG-COT accommodates speech signal to MG0, and activation signal (loop activation) to MG1.
    d.) The power for activation signal is not supplied from MCMG/MG-COT.
    e.) MCMG's MG2-MG6 connectors and MG-COT's MG2-MG5 connectors can be used for ordinary analog C.O. line interface even when the MCMG/MG-COT is used as a Paging Trunk. Assign the Paging Trunk data by ATRK command different from the C.O. trunk data.
16. MEET-ME PAGING [M-1, 11] and PAGING ACCESS [P-1] can be provided simultaneously in the same system.
17. To directly answer PAGING ACCESS [P-1] by the Attendant Console, the Paged party should use the INDIVIDUAL ATTENDANT ACCESS [I-4] feature.
18. Paging is limited to Stations or Attendant Consoles. Other parties (Central Office Trunks Calls) must have the Attendant Console or Station user perform the Paging operation.

19. BUSY VERIFICATION [B-3], EXECUTIVE RIGHT OF WAY [E-1]: These features will be denied if the Paging route has been assigned for Privacy. Verify the data in the Assignment of Route Data command, ARTD, CDN 44:PRV. If CDN 6:TCL is assigned as data "8", CDN 44:PRV is not required.
20. For simple PAGING ACCESS [P-1], assignments in ASFC are not required. Refer directly to the other Paging features, such as PAGING TRANSFER [P-10].
21. The Attendant can dial the PRIORITY PAGING [P-30] Access Code and release a Station.
22. When disconnecting from a Paging Trunk, the Trunk becomes idle immediately.
23. Paging is possible from a CONSULTATION HOLD [C-17]. Station A places Party B on soft hold and dials the Access Code to the Paging Trunk. A momentary pressing of the switch hook disconnects the Paging Trunk immediately and reconnects Party B to Station A.
24. CALL HOLD [C-6]: A Paging Trunk cannot be placed on Call hold.
25. BUSY is displayed on the Dterm LCD display when the Paging Trunk is busy. Reorder Tone is returned.
26. If PRIORITY PAGING [P-30] is used with PAGING ACCESS [P-1]; ARTD, CDN 6 must be assigned as data "8", Page with Ringback Tone. System Data 1, Index 73 affects the operation of Page routes assigned Page with Ringback Tone. Therefore, if PRIORITY PAGING [P-30] is used with PAGING ACCESS [P-1], then MEET-ME PAGE [M-1, 11] and PAGING TRANSFER [P-10] are not available to the system.
27. PRIORITY PAGE [P-30] access will not disconnect an Attendant Console engaged in a Paging Trunk Connection.
28. This feature cannot be used with LOCATION DIVERSITY [L-55].

Programming

Note: For UG50 programming, see Peripheral Equipment Description (UG50) manual.

<When COT Circuit Card is used for Paging Trunk>

STEP 1: **ASYD** - System Data 1, Index 73, Bits 0 and 1. For PAGING function only, assign data "00".

STEP 2: **ARTD** - Assign the Paging route as shown below:

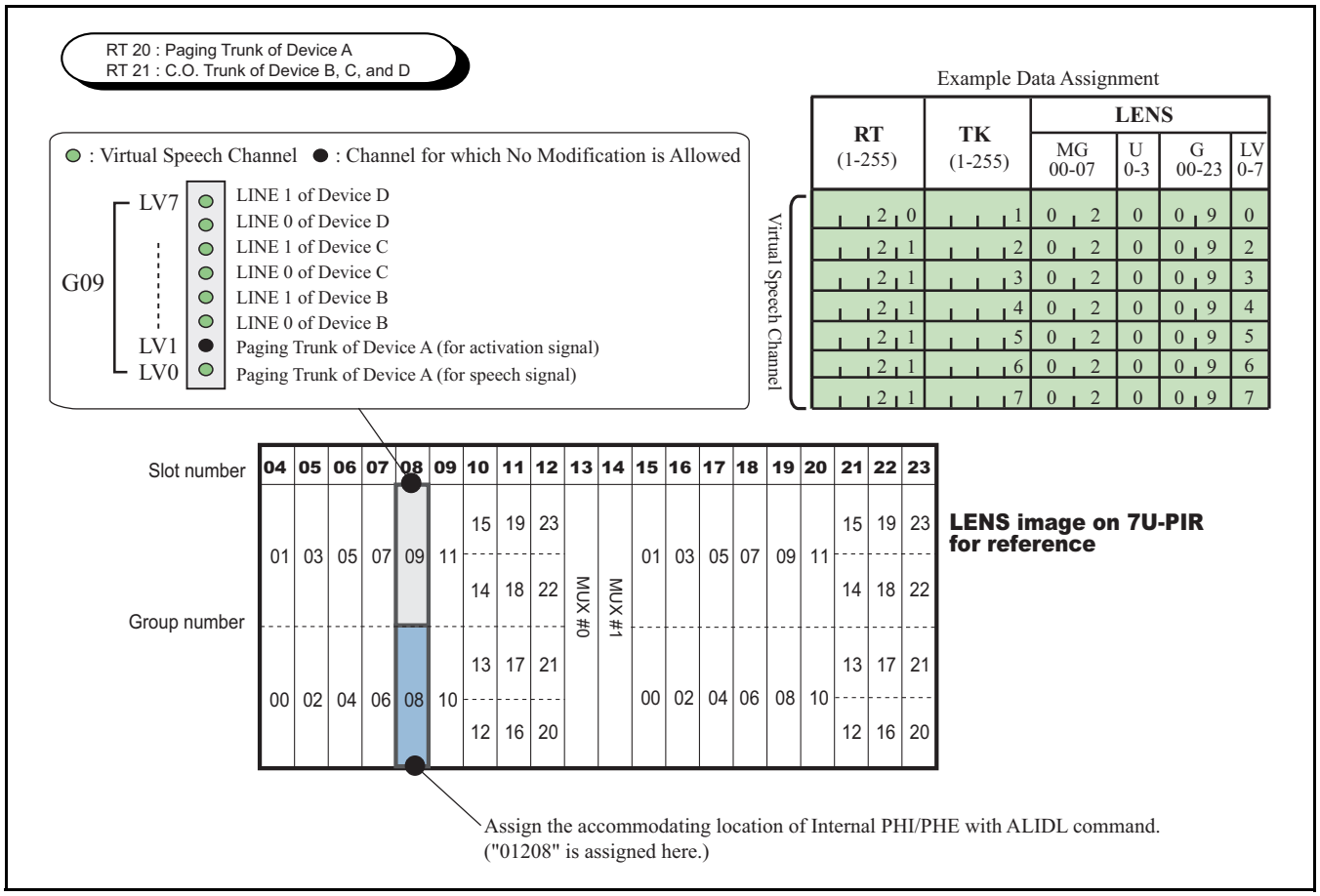
RT: 1

1-OSGS	:2	2-ONSG	:3	4-INSG	:3	5-TF	:1	6-TCL	:8
7-L/T	:1	8-RLP	:2	15-LSG	:0/1	44-PRV	:1		

Note: To change the data for existing route (RT), initialize the circuit card.

STEP 3: **ATRK** - Assign the LENS, Paging Trunk Route Number, Trunk Number, and Tenant Number.

Example: ATRK data assignment when COT card is used for Paging Trunk



STEP 4: **MBTK** - Assign the Make Idle status to the Paging Trunks.

STEP 5: **ANPD** - Assign a Level for Trunk Access. Select Normal Service (N) or Hooking Service (H) from the combo box for Connection Indexes. Assign the Number of Necessary Digits according to a predetermined Numbering Plan. Busy Lamp Field is not activated.

STEP 6: **ASPA** - Assign the PAGE ACCESS Code. Type of Service, SRV = "OGC: Outgoing Call". Assign the correct Route Number associated with this Access Code. Assign Connection Indexes, CI = "N: Normal", "H: Hooking".

STEP 7: **ARSC** - Assign to the Station that is allowed access to Paging Trunks a Route Restriction Class (RSC) that is allowed these routes in Route Restriction Indexes (RRI) 2 and 3.

STEP 8: **ASFC** - Assign Station to receive PAGING TRANSFER a Service Feature Restriction Class that allows SFI = 38. This allows a Station to dial PAGE answer codes.

<When MC&MG-COT is used for Paging Trunk>

The following steps assume that the basic setup has already been done on the MC&MG-COT.

STEP 1: **ASYD** - System Data 1

Index 73, Bit 1 and Bit 0: 00 = No answer, 01 = No delay, 10 = Delay, 11 = Delay with Paging Transfer **Note 3**

Index 74, Bit 0-3: Cancel Timer for unanswered Paging

Note 3: Delay with Paging Transfer is applied for the Paging Access on Attendant Console when 01, 10, or 11 is assigned in Index 73, Bit 0 & 1.

Note: Be sure to assign LENS to each LMG accommodates virtual speech/signal channel in 16-IMG system.

STEP 2: **ASTD** - State Translation Data

STM = 2, STS = 2 (Outgoing call to C.O. line or Tie line after holding COT) is set to ST = 1.

STM = 2, STS = 3 (Outgoing call to C.O. line after holding Station or Tie line) is set to ST = 1 (allowed).

STEP 3: **ANPD/ANPDL** - Numbering Plan Data. Assign the minimum number of digits to the first digit of Access Code for Paging Access and Paging Answer.

STEP 4: **ASPA/ASPAL** - Special Access Code.

SRV = "OGC: Outgoing Call," assign the route number to the Access Code of Paging Access.

SRV = "PAGA: Paging Answer," assign the route number to the Access Code of Paging Answer.

SRV = "PAGC: Paging Cancel," assign the route number to the Access Code of Paging Cancel.

STEP 5: **ARSC** - Route Restriction Class. Release the restriction of Paging Trunk access on stations, RRI-2 and RRI-3 to be allowed.

STEP 6: **ARRC** - Alternate Route Restriction Data. Release the restriction of Paging Trunk access on incoming trunk.

STEP 7: **ASFC** - Service Feature Restriction Class. Assign the Service Feature Restriction Class of Paging Answer. SFI 38 (Meet-Me Paging) is allowed.

STEP 8: **ARTD** - Route Class Data. Assign the following route class data to the Paging Trunk route.

CDN 1: OSGS (Signal Interface for Outgoing) = 2 (Second Dial Tone)

CDN 6: TCL (Trunk Class) = 8 (Paging)

In addition to the basic data of Paging Trunk, the following data assignment is required:

CDN 60: TC/EC = 1 (Echo Canceller is available)

CDN 114: DCH = 1 (Channel integration is available)

CDN 125: MGCOT = 1 (MG-COT is available)

Note: To change the data for existing route (RT), reboot the MG.

STEP 9: **ARTI** - Trunk Application Data. Assign the application data to Paging Trunk route.
CDN 68: VIR = 2 (MG connection)

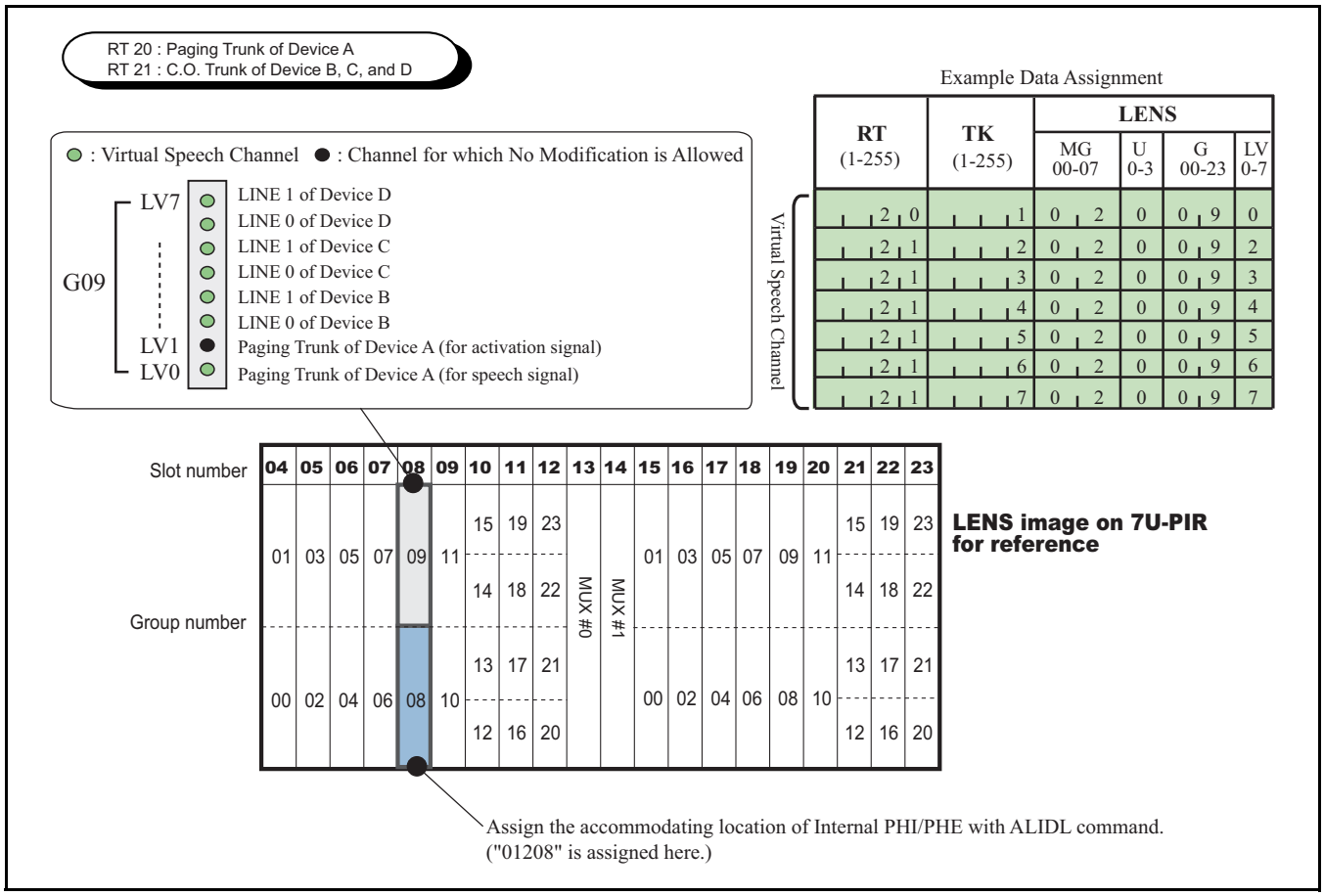
Note: To change the data for existing route (RT), reboot the MG.

STEP 10: **ATRK** - Trunk Data. Assign the Paging Trunk. Be sure to assign the LENS of Paging Trunk to even numbered LV.

Note: Please note the following important points:

- Be sure to assign the LENS of Paging Trunk on LV0.
- Do not assign the Paging Trunk data to even-numbered Module Group, Unit=0, and Group=0. MC&MG-COT cannot use this LENS.
- Do not assign the trunk data of other MG devices.
- Do not assign anything to LV1, LV6, and LV7.
- Make sure not to assign the Route Number (RT) duplicating the MG-COT route.
- The first two channels of an MC&MG-COT card can be used for a Paging Trunk. In this case, these channels cannot be connected to C.O. lines.

Example: ATRK data assignment when MC&MG-COT is used for Paging Trunk



STEP 11: **MBTK** - Trunk Make Busy. Release the make-busy state of the Paging Trunk, specifying the route number and the trunk number.

STEP 12: **AMGIL** - Media Gateway Information. Assign the MC&MG-COT related data.

MG-ID: MAC address of MC&MG-COT

KIND: MC-MG(COT)

UG50 Multi-Slot Mode: Select this check box if registering UG50 in Multi-slot mode.

LINE: Fixed to 0. **Note 4**

CH: Fixed to 0.

LENS: Mounting location of Paging Trunk (even numbered LV). **Note 5**

Setting Type of Signaling Packet Service: QoS setting for the signal route of MG-COT (ToS value for signal control)

Music On Hold Exists: No setting

Country Code: Select the country name

Law of A/D Conversion: μ-law/A-law **Note 6**

PB Receive Complete Timing of Dial-In [0-15]: Fixed to 0.

False Answer Timing: Fixed to 0.

Note 4: Make sure that the ATRK data has already been assigned before assigning the LENS data by AMGIL command. (Be sure to assign the trunk data in the order of ATRK → AMGIL.)

Note 5: Please note the following important points:
- LENS:000000 is not available for the users.
- Make sure that the LENS list in this command includes the data that you specified with the ATRK command in the previous step.
- Of the first two channels for the Paging Trunk, do not assign C.O. trunk data to the one with the larger channel number (CH=1) with the AMGIL command.

Note 6: Recommended Law of A/D conversion type is displayed according to the selected country code.)

<When MG-COT [SCA-6COTA]/MG-COT [SCA-6COTB]/MG-COT [SCA-6COTC] is used for Paging Trunk>

The following steps assume that the basic setup has already been done on the MG-COT [SCA-6COTA]/MG-COT [SCA-6COTB]/MG-COT [SCA-6COTC].

STEP 1: **ASYD** - System Data 1

Index 73, Bit 1 and Bit 0: 00 = No answer, 01 = No delay, 10 = Delay, 11 = Delay with Paging Transfer **Note 7**

Index 74, Bit 0-3: Cancel Timer for unanswered Paging

Note 7: Delay with Paging Transfer is applied for the Paging Access on Attendant Console when 01, 10, or 11 is assigned in Index 73, Bit 0 & 1.

Note: Be sure to assign LENS to each (LMG accommodates virtual speech/signal) channel in 16-IMG system.

STEP 2: **ASTD** - State Translation Data

STM = 2, STS = 2 (Outgoing call to C.O. line or Tie line after holding COT) is set to ST = 1.

STM = 2, STS = 3 (Outgoing call to C.O. line after holding Station or Tie line) is set to ST = 1 (allowed).

STEP 3: **ANPD/ANPDL** - Numbering Plan Data. Assign the minimum number of digits to the first digit of Access Code for Paging Access and Paging Answer.

STEP 4: **ASPA/ASPAL** - Special Access Code.
SRV = "OGC: Outgoing Call," assign the route number to the Access Code of Paging Access.
SRV = "PAGA: Paging Answer," assign the route number to the Access Code of Paging Answer.
SRV = "PAGC: Paging Cancel," assign the route number to the Access Code of Paging Cancel.

STEP 5: **ARSC** - Route Restriction Class. Release the restriction of Paging Trunk access on stations, RRI-2 and RRI-3 to be allowed.

STEP 6: **ARRC** - Alternate Route Restriction Data. Release the restriction of Paging Trunk access on incoming trunk.

STEP 7: **ASFC** - Service Feature Restriction Class. Assign the Service Feature Restriction Class of Paging Answer. SFI 38 (Meet-Me Paging) is allowed.

STEP 8: **ARTD** - Route Class Data. Assign the following route class data to the Paging Trunk route.
CDN 1: OSGS (Signal Interface for Outgoing) = 2 (Second Dial Tone)
CDN 6: TCL (Trunk Class) = 8 (Paging)

In addition to the basic data of Paging Trunk, the following data assignment is required:
CDN 60: TC/EC = 1 (Echo Canceller is available)
CDN 114: DCH = 1 (Channel integration is available)
CDN 125: MGCOT = 1 (MG-COT is available)

Note: To change the data for existing route (RT), reboot the MG.

STEP 9: **ARTI** - Trunk Application Data. Assign the application data to Paging Trunk route.
CDN 68: VIR = 2 (MG connection)

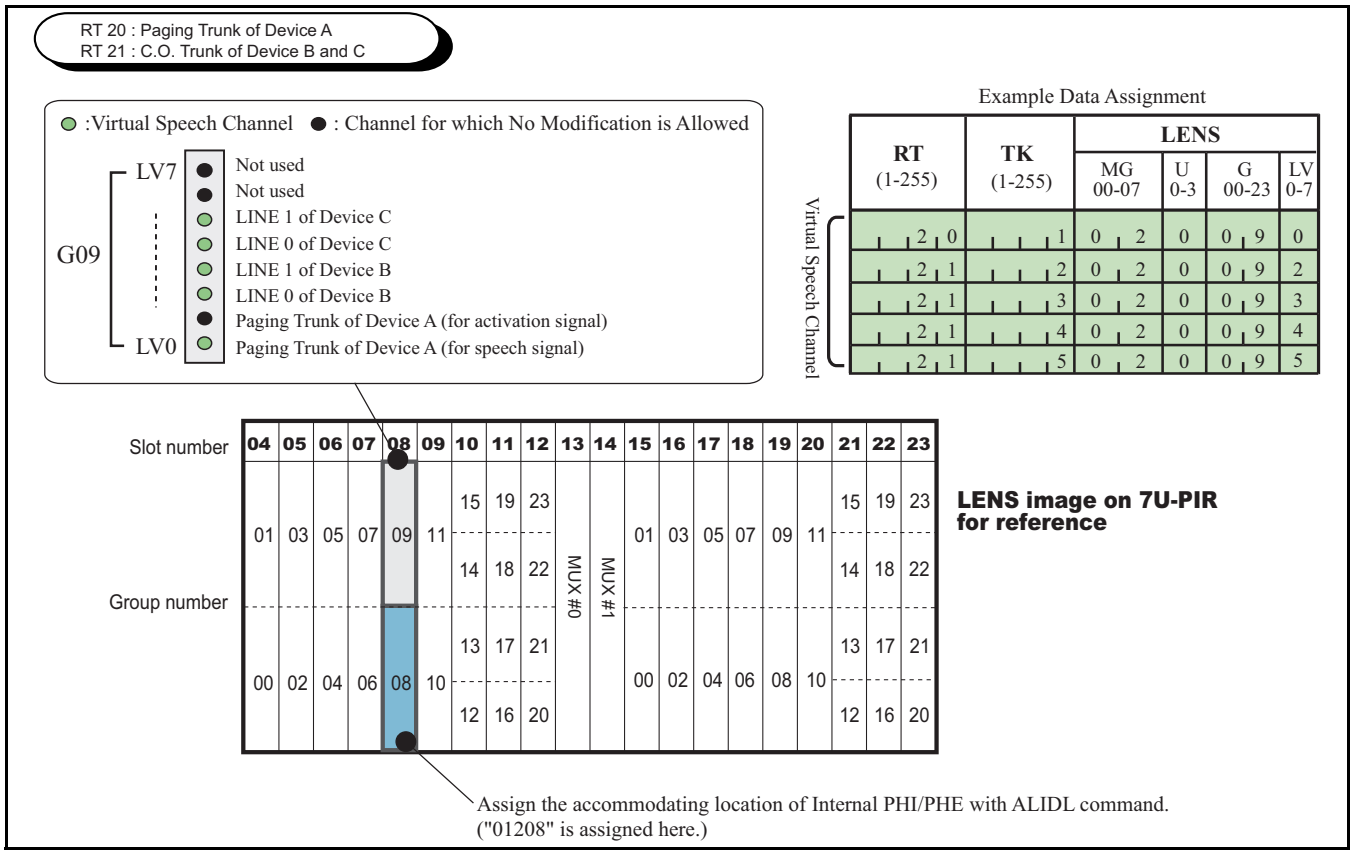
Note: To change the data for existing route (RT), reboot the MG.

STEP 10: **ATRK** - Trunk Data. Assign the Paging Trunk. Be sure to assign the LENS of Paging Trunk to even numbered LV.

Note: Please note the following important points:

- Be sure to assign the LENS of Paging Trunk on LV0.
- Do not assign the Paging Trunk data to even-numbered Module Group, Unit=0, and Group=0. MG-COT cannot use this LENS.
- Do not assign the trunk data of other MG devices.
- Do not assign anything to LV1, LV6, and LV7.
- Make sure not to assign the Route Number (RT) duplicating the MG-COT route.
- The first two channels of an MG-COT card can be used for a Paging Trunk. In this case, these channels cannot be connected to C.O. lines.

Example: ATRK data assignment when MG-COT is used for Paging Trunk



STEP 11: **MBTK** - Trunk Make Busy. Release the make-busy state of the Paging Trunk, specifying the route number and the trunk number.

STEP 12: **AMGIL** - Media Gateway Information. Assign the MCMG/MG-COT related data.

MG-ID: MAC address of MCMG/MG-COT

KIND: MG(6COT)/MG(6COT) [SIP]

UG50 Multi-Slot Mode: Select this check box if registering UG50 in Multi-slot mode.

LINE: 0 (Click the check box of "LINE0" to ON)

CH: Fixed to 0.

LENS: Mounting location of Paging Trunk (5 digits). **Note 8**

Setting Type of Signaling Packet Service: QoS setting for the signal route of MCMG/MG-COT (ToS value for signal control)

Music On Hold Exists: No setting

Country Code: Select the country name **Note 9**

Law of A/D Conversion: μ-law/A-law **Note 10**, **Note 11**

PB Receive Complete Timing of Dial-In [0-15]: Fixed to 0.

False Answer Timing: Fixed to 0.

Note 8: Please note the following important points:
- LENS:000000 is not available for the users.
- Make sure that the LENS list in this command includes the data that you specified with the ATRK command in the previous step.
- Of the first two channels for the Paging Trunk, do not assign C.O. trunk data to the one with the larger channel number (CH=1) with the AMGIL command.

Note 9: For MG-COT [SCA-6COTB], assign "U.K".

Note 10: Recommended Law of A/D conversion type is displayed according to the selected country code.

Note 11: For MG-COT [SCA-6COTB], assign "A-law".

FCCS Programming

This data setting describes only the additional or replacement commands used to activate FCCS Services. When used as an FCCS feature, these commands replace the non-network commands (for example, AAED is replaced by AAEDN). Note that the data setting for FCCS Service varies depending on the Node.

Note: When using ASYDL or ASYDN to program an FCCS feature, the basic programming of ASYD is required in addition to ASYDL or ASYDN.

STEP 1: **ALRTN** - Assign this command to connect the Logical Route to the Physical Route.

STEP 2: **ASYDN** - System Data 1, Index 73, Bits 0 and 1. For PAGING function only, assign data "00".

STEP 3: **ARTDN** - Assign the Paging Route as shown below:

RT: 1

1-OSGS	:2	2-ONSG	:3	4-INSG	:3	5-TF	:1	6-TCL	:8
7-L/T	:1	8-RLP	:2	15-LSG	:0/1	44-PRV	:1		

Note: To change the data for existing route (RT), initialize the circuit card.

STEP 4: **ATRKN** - Assign the LENS, Paging Trunk Route Number, Trunk Number, and Tenant Number.

STEP 5: **MBTKN** - Assign the Make Idle status to the Paging Trunks.

STEP 6: **ANPDN** - Assign a Level for Trunk Access. Select Normal Service (N) or Hooking Service (H) from the combo box for Connection Indexes. Assign the Number of Necessary Digits according to a predetermined Numbering Plan. Busy Lamp Field is not activated.

STEP 7: **ASPAN** - Assign the PAGE ACCESS Code. Type of Service, SRV = "OGC: Outgoing Call". Assign the correct Route Number associated with this Access Code. Assign Connection Indexes, CI = "N: Normal", "H: Hooking".

STEP 8: **ARSCN** - Assign to the Station that is allowed access to Paging Trunks a Route Restriction Class (RSC) that is allowed these Routes in Route Restriction Indexes (RRI) 2 and 3.

When using ASYDL or ASYDN to program an FCCS feature, the basic programming of ASYD is required in addition to ASYDL or ASYDN.

P-2 Passing Dial Tone

General Description

This feature allows the Attendant to PASS DIAL TONE to a restricted Station user, enabling that user to place a call that would normally be restricted.

Operating Procedure

1. Restricted user dials Attendant.
2. Attendant presses the ANSWER or ATND key. Tenant Number, Trunk Route Restriction Class and Telephone Number are displayed.
3. Obtain call details.
4. Dial central office Access Code. Tenant Number, and Trunk Identification Number are displayed.
5. Press the RELEASE key. User hears Dial Tone.
6. User places the call.

Service Conditions

1. This feature permits PASSING DIAL TONE to restricted and unrestricted Stations only by the Attendant.
2. When LEAST COST ROUTING - 3/6-DIGIT [L-5] is provided in the system, the Attendant cannot pass Dial Tone. The call must be completed via the ATTENDANT CONSOLE [A-3].
3. TOLL RESTRICTION [T-7] is activated via system programming for the PASSING DIAL TONE feature.
4. The PASSING DIAL TONE feature is limited by the number of Trunks that can be accessed by the Attendant and the Class of Service restrictions of the Station.
5. Restriction from Outgoing Calls: This feature can be overridden by the use of the PASSING DIAL TONE feature.
6. The Attendant is allowed to pass Dial Tone to any Station that has a class of service that allows normal direct dial access capabilities to the Trunk Group seized by the Attendant.
7. Any non-restricted Station can dial through over the selected Outgoing Trunk. The type of address signals sent to the called location are determined by the Trunk Route Class Data.
8. Where STATION MESSAGE DETAIL RECORDING (SMDR) [S-10] is provided, through-dialed calls are charged to the Calling Station Line.
9. This feature is available when the Numbering Plan for Call Origination is Outgoing Call (OGC) or Outgoing Call Alternative (OGCA), and the answer-back signal is the Second Dial Tone.
10. When placing Outside Calls, activation of the TOLL RESTRICTION - 3/6-DIGIT [T-7] is determined by System Data of the Calling Station on a Tenant basis.
11. If TOLL RESTRICTION- 3/6-DIGIT [T-7] is not activated via System Data, neither the three-digit Area Code nor the six-digit Area and Office Code are restricted.
12. If TOLL RESTRICTION- 3/6-DIGIT [T-7] is activated via System Data, the call will be restricted when the Calling Party dials the restricted Area Code.
13. This service can not be used for Peer-to-Peer FCCS.
14. After dialing central office Access Code such as 9, the operator must release the line before DEST lamp comes on.

Interactions

1. Any Station that is restricted from a route may receive access to that route by the Attendant Console. A Station that is toll restricted will NOT be able to place calls to destinations normally restricted by having the Attendant Console pass Dial Tone. The same ATDP tables will apply to that Station. The Attendant Console must dial the number for the Station when that Station is toll restricted. See NON-DELAY OPERATION [N-3].
2. The associated programming that should be examined is ATTENDANT CONSOLE [A-3], TOLL DENIAL/TOLL DIVERSION [T-7], TOLL RESTRICTION-3/6 DIGIT [T-8], LEAST-COST ROUTING-3/6 DIGIT [L-5], CLASS OF SERVICE-INDIVIDUAL [C-17], and the various Trunk Access features.
3. NIGHT CONNECTION FIXED/FLEXIBLE [N-1, 2]: The NIGHT CONNECTION FIXED/FLEXIBLE Stations cannot pass Dial Tone to another Station.
4. ATTENDANT NIGHT TRANSFER [A-19]: The ATTENDANT NIGHT TRANSFER Station cannot pass Dial Tone to another Station.
5. When LEAST-COST ROUTING-3/6 DIGIT [L-5] is programmed, the ATDP tables will be affected. During feature operation, the Attendant will dial the route Access Code and then release Dial Tone to the Station. Because that route is assigned in ARNP tables against the LCR (S) Access Codes, the ATDP tables must also use the LCR (S) Access Code. For example, the Attendant dials 81 and releases. The Station dials 1-516-753-7526. ATDP must be programmed as DC: 9151 and TDI: 1, allowed; not as DC: 81151. (ARNP for the route in this example is assigned with the Access Code = 9.)

Programming

STEP 1: **ASYD** - System Data 2, Index 15, Bit 3. Toll restriction on an Outgoing Call placed by PASSING DIAL TONE. 0/1: Valid/Invalid.

STEP 2: **ARSC** - Construct a Route Restriction Class (RSC) that will allow outgoing access to the various routes via Route Restriction Index (RRI) 2, via the Attendant Console. Be sure to assign RRI 3 as data "0 or 2", Connection restricted.

STEP 3: **ASDT** - Assign to the Station a Route Restriction Class (RSC) that allows outgoing access to the various routes via the Attendant Console.

STEP 4: **ATDP** - For those Route Restriction Classes that will receive Dial Tone from the Attendant Console, construct allow tables. The system does not pass the RSC of the Attendant Console to the restricted Station. It bypasses the RRI 3 restriction and instructs the system to look at the ATDP tables for that RSC and govern Connections accordingly.

FCCS Programming

This data setting describes only the additional or replacement commands used to activate FCCS Services. When used as an FCCS feature, these commands replace the non-network commands (for example, AAED is replaced by AAEDN). Note that the data setting for FCCS Service varies depending on the Node.

Note: When using ASYDL or ASYDN to program an FCCS feature, the basic programming of ASYD is required in addition to ASYDL or ASYDN.

STEP 1: **ALRTN** - Assign this command to connect the Logical Route to the Physical Route.

STEP 2: **ASYDN** - System Data 2, Index 15, Bit 3. Toll Restriction on an Outgoing Call placed by PASSING DIAL TONE. 0/1: Valid/Invalid.

STEP 3: **ARSCN** - Construct a Route Restriction Class (RSC) that will allow outgoing access to the various Routes via Route Restriction Index (RRI) 2, via the Attendant Console. Be sure to assign RRI 3 as data "0 or 2", Connection restricted.

STEP 4: **ASDTN** - Assign to the Station a Route Restriction Class (RSC) that allows outgoing access to the various Routes via the Attendant Console.

STEP 5: **ATDPN** - For those Route Restriction Classes that will receive Dial Tone from the Attendant Console, construct allow Tables. The system does not pass the RSC of the Attendant Console to the restricted Station. It bypasses the RRI 3 restriction and instructs the system to look at the ATDP Tables for that RSC and govern Connections accordingly.

P-3 Power Failure Transfer

General Description

This feature provides for certain specified Trunks to be automatically connected to designated Stations in the event of a loss of AC power. It is normally used when the system is not equipped with reserve power.

Station Application

Operation is automatic if there is loss of input power to the system.

Service Conditions

1. If the Central Office is wired Ground Start, Stations must be equipped with ground buttons to complete calls over the exchange network.
2. Power failure Station Lines should not be wired to common bells. These require Outside power sources and will not operate.
3. Type of telephone set (DP or DTMF) connected to Central Office Line via POWER FAILURE TRANSFER shall correspond to the signaling system of the Central Office.
4. The maximum number of POWER FAILURE TRANSFER Circuits per PIR is 24 (12 circuits per package).
5. The Power Failure Trunk (PFT) and associated Central Office Trunk should be in the same PIR.
6. When the POWER FAILURE TRANSFER feature is activated, telephone service is limited to Incoming Calls and/or Outgoing Calls via the serving Central Office. DID Calls cannot be received.
7. This feature is available with the combination of COT and Analog Telephone/Desktop terminal with PFT adapter.
8. All calls in progress and/or established calls are lost when a POWER FAILURE TRANSFER occurs. When commercial power is restored, the System is automatically reinitialized. Back-up service is provided via the HDD unit to retain the following features:

CALL FORWARDING	[C-2, C-3, C-5]
SPEED CALLING - GROUP	[S-23]
SPEED CALLING - STATION	[S-21]
NAME DISPLAY DATA	[N-28]

Interactions

1. A PFT circuit card is required.
2. The POWER FAILURE TRANSFER Station and Central Office Trunk must be on the same Module Group.
3. When power is lost and the POWER FAILURE TRANSFER (PFT) feature is activated, telephone service is limited to Incoming and Outgoing Calls via the predesignated PFT Stations.
4. DID service will be lost, as this service is dependent on the operational status of Incoming Register Trunks (IRTs).
5. When power is lost, all calls (Station-to-Station and Station-to-Trunk) are lost. When power is restored, PFT Calls will stay up until manually disconnected by the participants. Service to the PFT Stations is then restored to normal.

Programming

STEP 1: **ASYD** - System Data 1, Index 77, Bits 4 and 5. Specify a module in which PFT card is mounted.

STEP 2: **AUNT** - Specify the information on the PFT circuit card accommodated in. System initialization is required to make it effective.
MG: Specify the MG number of PFT.
TYPE=2 (PFT Circuit Card)
PKG=1 (Mounted)

To delete

The deletion procedure is the reverse order of assigning the feature.

P-4 Pushbutton Calling

General Description

This feature permits Station users to originate calls using Pushbutton Telephones and also allows Dterm Stations the ability to control external devices requiring DTMF Signals to initiate or perform functions (e.g., Code-A-Phone, Conference Unit, etc.).

Operating Procedure

1. The Station users press pushbuttons corresponding to the desired numbers.
   OR
2. After a Connection is established between a Dterm Station and a Station or Trunk port with an external device attached, the Dterm user may press the keypad or SPEED CALLING BUTTONS - ONE TOUCH -D [S-26D] key to transmit DTMF Signals to control the external device.

Service Conditions

1. If DTMF lines from the Central Office are not available, then the PUSHBUTTON-TO-ROTARY CONVERSION [P-6] feature must be provided.
2. Rotary and Pushbutton instruments may be used simultaneously with PUSHBUTTON CALLING.
3. Timing for the DTMF Signals can be set from 0.128 to 2.048 seconds via system commands on a system-wide basis.
4. DTMF frequency combinations are as follows:

Low - High	1,209Hz	1,336Hz	1,477Hz
697Hz	1	2	3
770Hz	4	5	6
852Hz	7	8	9
941Hz	*	0	#

Interactions

1. Each 8RST circuit card contains eight Register and eight Sender Circuits. Each Register Circuit can accept either Pushbutton or rotary dial pulses. The senders can transmit Pushbutton or dial pulse signals and are also used with the ATTENDANT CONSOLE [A-3], LEAST-COST ROUTING-3/6 DIGIT [L-5], SPEED CALLING-SYSTEM, STATION, GROUP [S-3, 21, 23], OUTGOING TRUNK QUEUING [O-2], OFF-HOOK QUEUING [O-7], and PUSHBUTTON-TO-ROTARY CONVERSION [P-6].
2. Register Circuits are located on the first four Levels (0-3) and Sender Circuits are located on the last four Levels (4-7) of the 8RST circuit card.

Programming

STEP 1: **ASYD** - System Data 1, Index 128. Register Prepause Timer. How long the Register waits for the first digit. Normally assign data 00H for 12 seconds.

System Data 1, Index 129. Register Inter-digit Timer. How long the Register waits between digits. Normally assign data 00H for six seconds.

System Data 2, Index 2, Bit 0. Type of Sender Signal for Station? 0/1: DP/PB. Assign as appropriate. (This data will only be used if the ARTD Data of a specific route is assigned as DP/PB.)

System Data 3, Index 11. Normally assign data 00H for 128 milliseconds.

STEP 2: **ASDT** - Assign Dterm, Single Line Stations, a Telephone Equipment Class (TEC) of 2; Pushbutton (PB), TEC = 3, or Dial-Pulse/ Pushbutton (DP/PB).

STEP 3: **ATRK** - Assign Originating Register Trunks (ORT) to the system in a quantity calculated from expected traffic. Registers are located on Levels 0-3 of any 8RST circuit card.

Also, senders must be assigned to the system in a quantity calculated from expected traffic. Senders are located on Levels 4-7 of any 8RST circuit card.

RT: Route Number

Trunk Route Number

Intra-office Route Number (901-931)

901-ATT 912-ORT for ATT
902-ORT 913-TCFT for ATT
903-IRT 915- Night ATT
905-Sender 916-MFCR
909-DCFT 917-MFCS
919 to 926-Modem

TK: Trunk Number
TN: Tenant Number
RSC: Route Restriction Class
SFC: Service Feature Restriction Class

STEP 4: **ARTD** - Routes must be assigned as either PB, DP, or DP/PB. CDN 2:ONSG, Type of Outgoing Signal Selection (ONSG) is assigned to each route as data “1”, DP; data “2”, PB; or data “3”, DP/PB.

Note: To change the data for existing route (RT), initialize or reboot the trunk device such as a circuit card or MG.

P-5 Pushbutton Calling - Attendant Only

General Description

This feature permits an Attendant to place all calls over DTMF Signaling lines from a Pushbutton keypad on the ATTENDANT CONSOLE [A-3].

Operating Procedure

Attendant presses the Pushbutton keypad to dial.

Service Conditions

1. This feature requires that all Central Office Trunks and TIE Trunk terminals accept Pushbutton signaling (DTMF).
2. PUSHBUTTON CALLING - ATTENDANT ONLY may be added to the system without providing PUSHBUTTON CALLING [P-4] capability to other Stations.

Interactions

The Attendant Console must first be provided.

Programming

STEP 1: **ASYD** - System Data 1, Index 128. Register Prepause Timer. How long the Register waits for the first digit. Normally assign data 00H for 12 seconds.

System Data 1, Index 134. Register Inter-digit Timer for the Attendant Console. How long the Register waits between digits. Normally assign data 00H for four seconds.

System Data 2, Index 2, Bit 1. Sender signal for Attendant Console is DP/PB? 0/1: DP/PB. Assign as appropriate.

STEP 2: **ATRK** - Assign Originating Register Trunks (ORT) to the Attendant Console (optional). Registers are located on Levels 0-3 of any 8RST circuit card.

Also, senders must be assigned to the system in a quantity calculated from expected traffic.

RT: Route Number

Trunk Route Number

Intra-office Route Number (901-931)

901-ATT	912-ORT for ATT
902-ORT	913-TCFT for ATT
903-IRT	915- Night ATT
905-Sender	916-MFCR
909-DCFT	917-MFCS
	919 to 926-Modem

TK: Trunk Number

TN: Tenant Number

RSC: Route Restriction Class

SFC: Service Feature Restriction Class

STEP 3: **ARTD** - Routes must be assigned as either PB, DP, or DP/PB. CDN 2:ONSG, Type of Outgoing Signal Selection, is assigned to each route as data “1”, DP; data “2”, PB; or data “3”, DP/PB.

Note: To change the data for existing route (RT), initialize or reboot the trunk device such as a circuit card or MG.

P-6 Pushbutton to Rotary Conversion

General Description

This feature allows DTMF telephones to be used when DTMF Signaling is not provided or are not available from the Central Office and/or TIE Line.

Operating Procedure

Station users dial the desired Telephone Numbers.

Service Conditions

1. PUSHBUTTON TO ROTARY CONVERSION is software controlled on a per Trunk Route basis.
2. See UNIVERSAL SENDER [U-3] for reference.
3. When the following combination of data settings are assigned, the PB Station cannot originate a call:

   ASDT - TEC = 3 (DP/PB) Station data for the Calling Station
   ASPA - SSC = "OGC: Outgoing Call"/"OGCA: Outgoing Call with route advance"
   ARTD - CDN1: OSGS = 2 (Second Dial Tone)
   CDN2: ONSG = 2 (PB, 60 msec. Interruption or CCIS No.7)

Interactions

1. Each 8RST circuit card contains four Register and four Sender Circuits. Each Register Circuit can accept either pushbutton or rotary dial pulses. The senders can outpulse pushbutton or dial pulse signals and are used with the ATTENDANT CONSOLE [A-3], LEAST-COST ROUTING-3/6 DIGIT [L-5], SPEED CALLING-SYSTEM, STATION, GROUP [S-3, 21, 23], OUTGOING TRUNK QUEUING [O-2], OFF-HOOK QUEUING [O-7], and PUSHBUTTON-TO-ROTARY CONVERSION.
2. Register Circuits are located on the first four Levels (0-3), and Sender Circuits are located on the last four Levels (4-7) of the 8RST circuit card.
3. Each Register may be assigned as either an outgoing or an Incoming Register. Incoming Calls have priority over Outgoing Calls. Thus, a Register programmed as outgoing may serve as an Incoming Register when none are available. However, an Incoming Register will never serve as an Outgoing Register.

Programming

STEP 1: **ASYD** - System Data 2, Index 2, Bit 0. Sender signal for Station is DP/PB? 0/1: DP/PB. Always assign as data "0". Assign on a per Tenant basis. This Bit takes priority when ARTD, CDN 2:ONSG is assigned as data "3", DP/PB.

STEP 2: **ASDT** - Assign Single Line Stations a Telephone Equipment Class (TEC) of Pushbutton (PB) TEC = 2 or dial pulse/pushbutton (DP/PB), TEC = 3.

STEP 3: **ATRK** - Assign Originating Register Trunks (ORT) to the system in a quantity calculated from expected traffic. Registers are located on Levels 0-3 of any 8RST circuit card.

Also, senders must be assigned to the system in a quantity calculated from expected traffic. Senders are located on Levels 4-7 of any 8RST circuit card.

RT: Route Number

Trunk Route Number

Intra-office Route Number (901-931)

901-ATT
902-ORT 913-TCFT for ATT
903-IRT 915- Night ATT
905-Sender 916-MFCR
909-DCFT 917-MFCS
919 to 926-Modem

TK: Trunk Number
TN: Tenant Number
RSC: Route Restriction Class
SFC: Service Feature Restriction Class

STEP 4: **ARTD** - Outgoing routes must be assigned as DP or DP/PB. Flag CDN 2:ONSG as data "1", DP or data "3", DP/PB.

Note: To change the data for existing route (RT), initialize or reboot the trunk device such as a circuit card or MG.

FCCS Programming

This data setting describes only the additional or replacement commands used to activate FCCS Services. When used as an FCCS feature, these commands replace the non-network commands (for example, AAED is replaced by AAEDN). Note that the data setting for FCCS Service varies depending on the Node.

Note: When using ASYDL or ASYDN to program an FCCS feature, the basic programming of ASYD is required in addition to ASYDL or ASYDN.

ARTDN - Outgoing routes must be assigned as DP or DP/PB. Flag CDN 2:ONSG as data "1", DP or data "3", DP/PB.

Note: To change the data for existing route (RT), initialize or reboot the trunk device such as a circuit card or MG.

P-7 Peg Count

General Description

This feature permits traffic studies and traffic analysis information to be accessed from the PCPro and to be printed out.

Operating Procedure

Please refer to TRAFFIC MANAGEMENT of CHAPTER 2 in the Operation and Maintenance Manual.

Service Conditions

1. This feature can be used for FCCS Network. For details, refer to the Operations and Maintenance Guide.
2. All information required for this feature is registered at the PCPro.
3. The following information is obtainable:
    - STATION PEG COUNT
    - ATT PEG COUNT
    - ROUTE PEG COUNT
    - SERVICE PEG COUNT
    - UCD ROUTE PEG COUNT
    - UCD GROUP PEG COUNT
    - UCD STATION PEG COUNT
    - ATT ANSWERING PEG COUNT
    - CS PEG COUNT
    - CONNECTION ROUTE PEG COUNT
    - IP SERVICE PEG COUNT
    - SIP TERMINAL PEG COUNT
4. The above information may be registered and canceled as required.
5. The measurement information output (to hard disk of the maintenance console used for PCPro) cycle can be any period of time ranging from 30 minutes to 120 minutes (in 10-minute increments), as required.
6. Terminal traffic is not available for PS, WLAN Handset and Standard SIP terminal.
7. Traffic measurement duration time is designated for more than one hour. If the traffic measurement duration time is not designated, the measurement is performed indefinitely until a stop command is entered on the PCPro.

Peg Count	Telephony Server
Station Peg Count	TN = 1-255
Route Peg Count	RT = 1-255
ATT Peg Count - Incoming	TN = 1-255
ATT Peg Count - Answer	ATT = 1-16

8. The following shows the condition of SIP Terminal Peg Count.
   a.) This service is available for SIP terminals supported by SP CONTROLLED SIP TERMINAL [S-168] only. For the details, refer to SP CONTROLLED SIP TERMINAL [S-168].
   b.) When SIP Wireless Terminal Backup [S-154] is used in the system, Traffic Measurement is performed for the present active side SP. Traffic Measurement for the standby side SP cannot be performed. When the situation of active side/standby side is changed, the Traffic Measurement target is also changed to the SP switched as active side.
   c.) When packet loss is occurred, the traffic data is not counted.
   d.) When the active side SP is switched, the traffic data is not counted even if sending and receiving the Keep Alive messages are failed between the SP Controlled SIP terminal and SP.
   e.) The system takes over peg count data during the system initialization (Non Load Initialization). However, system does not take over the peg count data to SR-MGC.
   f.) When traffic measurements of SIP terminals are performed with other traffic measurements, all traffics are measured in the same time. For the details, refer to the following table.

×: Available –: Not available

Type of Traffic Measurement	Measurement target items	SP Controlled Standard SIP Terminal	WLAN Handset	PS	Digital Terminal
Type 1 Terminal Traffic Data	Traffic data for LENS	× Note 1	× Note 1	× Note 2	×
Type 3 Station Peg Count	Originator	×	×	×	×
	Sta to Sta Outgoing	×	×	×	×
	Outgoing	×	×	×	×
	Other Outgoing	×	×	×	×
	Sta to Sta Incoming	×	×	×	×
	Incoming	×	×	×	×
	Ringing	×	×	×	×
	No Answer 30 Seconds	×	×	×	×
	Answer	×	×	×	×
	No Answer	×	×	×	×
	Other Incoming	×	×	×	×

Type of Traffic Measurement	Measurement target items	SP Controlled Standard SIP Terminal	WLAN Handset	PS	Digital Terminal
Type 6 Service Peg Count	Call Back	–	×	×	×
	Call Back Cancel	–	×	×	×
	Erow	–	×	×	×
	Call Waiting Originate	–	×	×	×
	Call Waiting Terminate	–	×	×	×
	Call Pick Up	×	×	×	×
	Direct Call Pick Up	×	×	×	×
	CFA Entry	×	×	×	×
	CFA Cancel	×	×	×	×
	CFB Entry	×	×	×	×
	CFB Cancel	×	×	×	×
	CFDA Entry	×	×	×	×
	CFDA Cancel	×	×	×	×
	System Speed Call Access	×	×	×	×
	Consult Hold	×	×	×	×
	Call Transfer	×	×	×	×
	Add On Conference	–	×	×	×

Type of Traffic Measurement	Measurement target items	SP Controlled Standard SIP Terminal	WLAN Handset	PS	Digital Terminal
Type 17 CS Peg Count	Outgoing Calls	–	–	×	–
	Completed Outgoing Calls	–	–	×	–
	Terminating Calls	–	–	×	–
	Completed Terminating Calls	–	–	×	–
	No Answer of Ringing Calls	–	–	×	–
	Re-Connected Channel Changeover	–	–	×	–
	Completed Re-Connected Channel Changeover	–	–	×	–
	Position Assignment Request Count	–	–	×	–
	Bch All Busy Count for CS/ZT	–	–	×	–
	Abnormal Access Count	–	–	×	–
	Channel Changeover for CS/ZT	–	–	×	–
Type 22 SIP Terminal Peg Count	Sending frequency	×	×	–	–
	Sending completion call frequency	×	×	–	–
	Arrival frequency	×	×	–	–
	Arrival no response frequency	×	×	–	–
	Handover demand frequency	–	×	–	–
	Handover completion call frequency	–	×	–	–
	SIP extension terminal register demand frequency	×	×	–	–
	SIP extension terminal digest attestation success	×	×	–	–
	SIP extension terminal digest attestation failure	×	×	–	–
	Talk band all speaking frequency	×	×	–	–
	SIP extension terminal cut keep alive failure	×	×	–	–

Note 1: Counted in the virtual speech channel of SIP server.

Note 2: Counted in the LENS of CS/ZT.

9. For more information, refer to TRAFFIC MANAGEMENT of CHAPTER 2 in the Operation and Maintenance Manual.

Interactions

1. If the system is reset, traffic measurement requests are no longer in service.
2. Do not set ASFC, SFI 252=1 (Multi-Boss/Multi-Secretary is in service) to stations belonging to UCD groups when using UCD GROUP/Station/Route PEG COUNT.

Programming

Please refer to TRAFFIC MANAGEMENT of CHAPTER 2 in the Operation and Maintenance Manual.

P-9 Priority Call

General Description

This feature allows the ATTENDANT CONSOLE [A-3] to answer various types of calls in the order of their priority. A special type of call can be handled prior to a regular call, at the Attendant's discretion.

Operating Procedure

To operate

1. Lift the handset and receive Dial Tone.
2. Dial the PRIORITY CALL number, then wait for the Attendant to answer.

Note: PRIORITY CALL will terminate on a designated key on the ATTENDANT CONSOLE [A-3].

Service Conditions

1. The PRIORITY CALL must be answered via a designated key in order to gain a priority capability. The keys must be assigned by the PCPro.
2. A PRIORITY CALL will not be picked up first, by pressing the ANSWER key, if other calls are waiting.
3. A PRIORITY CALL cannot be initiated while in CONSULTATION HOLD - ALL CALLS [C-17] or CALL HOLD [C-6].
4. In the night mode, a PRIORITY CALL will terminate at the predetermined Station.
5. The Calling Station obtains this feature via CLASS OF SERVICE - INDIVIDUAL [C-15].
6. A Station, instead of the ATTENDANT CONSOLE [A-3], can be the destination of a PRIORITY CALL in the day mode.
7. If the PRIORITY CALL encounters the Destination Station busy, the call will wait for the Station to become idle. Ringback Tone will be heard while the caller is waiting.
8. Three Levels of PRIORITY CALLs can be assigned.
9. When the destination is a Station, the originating Stations should be assigned as a member of a PRIORITY CALL Group.
10. The terminating Station of each group can be a member of a HUNTING GROUP [S-7, 8, 9].
11. The terminating Station can set CALL FORWARDING - ALL CALLS [C-5] and CALL FORWARDING - BUSY LINE [C-2].
12. The terminating Station for ATTENDANT NIGHT TRANSFER [A-19] and/or OFF-HOOK ALARM [O-6] can be assigned as the terminating Station of a PRIORITY CALL.
13. If the destination Dterm is busy (and not the pilot number of a UCD Group), when a Priority Call is placed, Ringback Tone is returned to the Calling Party rather than Busy Tone, and the Calling Party is placed in queue.
14. When a UCD [U-1] Station is assigned as the destination of PRIORITY CALL, queuing function of UCD is not available when all the Stations within the UCD Group are busy.
15. The Destination Station for the Priority Call must be the Dterm. When the Destination Station is set CALL FORWARDING-ALL CALLS [C-5], the Call Forwarding target Station must also be the Dterm. When the Destination Station is a member of a HUNTING GROUP [S-7, 8, 9], all Stations within the Hunt Group must also be the Dterm.

16. The maximum number of PRIORITY CALL Groups is 255 per the Level of Priority Call(s) (Priority Call 1, 2, 3).
17. An ISDN terminal user cannot use this feature.

FCCS Conditions

1. The Fusion Point Code (FPC) of the Node which accommodates the Called Attendant Console must be assigned after System Data assignment.
2. The Priority Call can be terminated at a predetermined Station in the Remote Node via FCCS link.
3. When the Access Code which is assigned by the ASPAN command is dialed, the data programmed in the ASGDN and the ASIDN command is applied. When the Access Code which is assigned by the ASPAL command is dialed, the data programmed in the ASGDL and the ASIDL commands is applied. When the Access Code which is assigned by the ASPA command is dialed, the data programmed in the ASGD and the ASID commands is applied.

Interactions

1. If the destination Dterm is busy when a Priority Call is placed, Ringback Tone is returned to the Calling Party rather than a Busy Tone, and the Calling Party is placed in queue.
2. A Warning Tone is not provided at the Called Dterm while a Priority Call is in progress.
3. If the destination of a Priority Call is the Attendant Console, the Priority Call does not transfer to the ATTENDANT NIGHT TRANSFER [A-19] Station when the system goes into Night mode.
4. A PRIORITY CALL will not be forwarded to Outside.

Programming

STEP 1: **ASYD** - System Data 1, Index 168, Bit 6. Where will Priority Call terminate? 0/1: Attendant Console/ Station. If Attendant Console (0), go to Step 2, AAKP command.

System Data 2, Indexes 8 and 9. If the call will terminate to the Attendant Console, assign if the call will be pegged for CALL WAITING LAMP [C-13].

STEP 2: **AAKP** - Assignment of Attendant Console Key Pattern. The ATN: Attendant Number (TRK Number for RTE 901); KYN = 1-6, and FUNC = 3, Priority Call 1; FUNC = 4, Priority Call 2; FUNC = 5, Priority Call 3. This command is not required if the Priority Call terminates at a Station.

STEP 3: **ANPD** - Reserve a Numbering Plan Level for Feature Access. Select Normal Service (N) from the combo box for Connection Indexes. Assign NND in accordance with the predetermined Numbering Plan.

STEP 4: **ASPA** - For SRV = "SSC: Service Code", SID = 37, Priority Call 1; SID = 38, Priority Call 2; SID = 39, Priority Call 3; assign the Access Code. Assign for CI of "N: Normal".

STEP 5: **ASID** - This command defines the purpose of the Special Incoming Group and is used in conjunction with the ASGD command. Assign the following:

TN: Tenant Number
F: Function Number (0-7)
4: Priority Call 1
5: Priority Call 2
6: Priority Call 3
GN: Group Number (1-7),
S/A: Special Incoming Equipment
S: Station
A: Attendant Console
SSTN: Special Station Number

STEP 6: **ASGD** - This command defines the Stations belonging to the Special Incoming Group. When the point of termination is a Station, assign the following:

TN: Tenant Number
STN: Station Number
F: Function Number (0-7)
4: Priority Call 1
5: Priority Call 2
6: Priority Call 3
GN: Group Number (1-7)

STEP 7: **ASFC** - Service Feature Index (SFI) 54-Priority Call 1. SFI 55-Priority Call 2. SFI 56-Priority Call 3. Assign a Service Feature Restriction Class to Stations that allow one or all of the SFIs above, depending on system configuration.

FCCS Programming

This data setting describes only the additional or replacement commands used to activate FCCS Services. When used as an FCCS feature, these commands replace the non-network commands (for example, AAED is replaced by AAEDN). Note that the data setting for FCCS Service varies depending on the Node.

Note: When using ASYDL or ASYDN to program an FCCS feature, the basic programming of ASYD is required in addition to ASYDL or ASYDN.

STEP 1: **ASPAL** - This data must be set to the Node calling the Attendant.
Assign the FPC of the Node which accommodates the Called Attendant Console. This must be assigned after designating SID = 37, 38, 39.

The following data setting is required when a Priority Call is routed to a predetermined Station in the Remote Node via FCCS links.

(When entering data on NDM)
STEP 1: **ASYD** - The same data must be assigned to all Nodes.
System Data 1, Index 168, Bit 6. Assign data "1" (Station is the destination of Priority Call in the day mode.)
STEP 2: **ANPDN** - Reserve a Numbering Plan Level for Feature Access. Select Normal Service (N) from the combo box for Connection Indexes. Assign NND in accordance with the predetermined Numbering Plan.
STEP 3: **ASPAN** - For SRV = "SSC: Service Code", SID = 37, Priority Call 1; SID = 38, Priority Call 2; SID = 39, Priority Call 3; assign the Access Code. Assign for CI of "N: Normal".
STEP 4: **ASIDN** - This command defines the purpose of the Special Incoming Group and is used in conjunction with the ASGDN command. Assign the following:

TN:	Tenant Number
F:	Function Number (0-7)
4:	Priority Call 1
5:	Priority Call 2
6:	Priority Call 3
GN:	Group Number (1-255)
S/A:	Special Incoming Equipment. Assign "S" (Telephone Number (of the Station)).
STELN:	Telephone Number of the Special Station

STEP 5: **ASGDN** - This command defines the Stations belonging to the Special Incoming Group. When the point of termination is a Station, assign the following:

UGN:	User Group Number
STELN:	Telephone Number of the Station
F:	Function Number (0-7)
4:	Priority Call 1
5:	Priority Call 2
6:	Priority Call 3
GN:	Group Number (1-255)

STEP 6: **ASFC** - The same data must be assigned to all Nodes.
Service Feature Index (SFI) 54-Priority Call 1. SFI 55-Priority Call 2. SFI 56-Priority Call 3. Assign a Service Feature Restriction Class to Stations that allow one or all of the SFIs above, depending on system configuration.

(When entering data on LDM)
STEP 1: **ASYD** - The same data must be assigned to all Nodes.
System Data 1, Index 168, Bit 6. Assign data "1" (Station is the destination of Priority Call in the day

mode.)

STEP 2: **ANPDL** - Reserve a Numbering Plan Level for Feature Access. Select Normal Service (N) from the combo box for Connection Indexes. Assign NND in accordance with the predetermined Numbering Plan.

STEP 3: **ASPAL** - For SRV = "SSC: Service Code", SID = 37, Priority Call 1; SID = 38, Priority Call 2; SID = 39, Priority Call 3; assign the Access Code. Assign for CI of "N: Normal".

STEP 4: **ASIDL** - This command defines the purpose of the Special Incoming Group and is used in conjunction with the ASGDL command. Assign the following:

TN: Tenant Number
F: Function Number (0-7)
4: Priority Call 1
5: Priority Call 2
6: Priority Call 3
GN: Group Number (1-255)
S/A: Special Incoming Equipment. Assign "S" (Telephone Number (of the Station)).
STELN: Telephone Number of the Special Station

STEP 5: **ASGDL** - This command defines the Stations belonging to the Special Incoming Group. When the point of termination is a Station, assign the following:

UGN: User Group Number
STELN: Telephone Number of the Station
F: Function Number (0-7)
4: Priority Call 1
5: Priority Call 2
6: Priority Call 3
GN: Group Number (1-255)

STEP 6: **ASFC** - The same data must be assigned to all Nodes.

Service Feature Index (SFI) 54-Priority Call 1. SFI 55-Priority Call 2. SFI 56-Priority Call 3. Assign a Service Feature Restriction Class to Stations that allow one or all of the SFIs above, depending on system configuration.

P-10 Paging Transfer

General Description

Station users can page anyone over the speaker after holding a station/trunk call to transfer. PAGING equipment is configured as the Paging Trunk and connected using an interface.

Operating Procedure

While a Station user is engaged in a call

1. Switch hook flash or press the TRANSFER key (on Desktop terminal). Receive Special Dial Tone.
2. Dial the Paging Access Code. Receive continuous Ringback Tone for approximately one second.
3. Page the desired party.

There are three paging options

1. NON-DELAY SYSTEM: The party paged is immediately connected to the paging Station when the paged party dials the Paging Answer Access Code.
2. DELAY SYSTEM: The paging Station user replaces the handset after the page; the paged party dials the Paging Answer Access Code. The paging Station rings, and the paged Station receives Ringback Tone. After the paging Station user lifts the handset, the paging and paged Stations are connected.
3. PAGING TRANSFER SUPERVISION: The paging Station user replaces the handset after the page. By dialing the Paging Answer Access Code, the paged party is directly connected to the party on hold.

For Non-Delay and Delay systems

1. When a paging Station replaces the handset after talking with the paged party, the Held Party is directly connected to the paged party.

Service Conditions

1. PAGING TRANSFER service can be selected according to System Data assignment:
   a.) Non-Delay system
   b.) Delay and Non-Delay system
   c.) PAGING TRANSFER Supervision
2. Paging Trunks can be provided for a maximum of six routes.
3. When a paging route has been accessed and is not answered within a predetermined time, the paging activation is automatically cleared.
4. The following can be used as a paging trunk interface:

   - COT Circuit Cards
   - TLT Circuit Cards
   - MC&MG-COT (4LC2COT)
   - MG-COT (6COT)
   - UG50 (GCD-PGTA) **Note 1 Note 2**

   Note 1: UG50 (GCD-4COTA/B/C) cannot be used as a paging trunk interface.

   Note 2: Available from UG50 software version 2.00.

5. Loop signaling is the recommended signaling type.
6. When the Attendant Console activates the service, the conditions are as follows:
   a.) An Attendant Console user can connect the paged party with the Calling Party the same way as PAGING TRANSFER SUPERVISION only (see Operating Procedure).
   b.) If the paged party answers the page before the Attendant Console releases the Loop, the Attendant Console is called from the paged party as an Operator Call. In this case, the Attendant Console can connect the paged party with the Held Party (Calling Party) in the procedure below:
      - Press the CANCEL key to release the Loop to be used for the page.
      - Answer the Operator Call and press the HOLD key to hold the Loop.
      - Press the Loop key holding the Calling Party to be back to the original Connection and perform the normal operation for Attendant Console Loop-to-Loop Connection (see MEET-ME PAGING [M-3] for details) to connect the two parties.
7. When using this feature via the FCCS network, PAGING ACCESS [P-1], PAGING TRANSFER [P-10] and RADIO PAGING [R-16] are provided to the paging Station and the paged Station in FCCS Nodes.
8. When using this feature via the FCCS Network, when PAGING ACCESS [P-1]/PAGING TRANSFER [P-10]/RADIO PAGING [R-16] service is set to a Station in the Remote Node or the PAGING ACCESS [P-1]/PAGING TRANSFER [P-10]/RADIO PAGING [R-16] Call is answered from the Remote Node via FCCS link:
   a.) the Logical Route Number should be applied to each physical route that is provided the service.
   b.) the restriction of Route-to-Route Connection between FCCS Nodes should be released.
9. When using this feature via the FCCS Network, the same paging route cannot be used by another Station while the paged Station does not answer the page activated using the route. However, the paging activation is automatically cleared when the call is not answered within a predetermined time.
10. When using this feature via the FCCS Network, a user can page one Station at a time.
11. When using this feature via the FCCS network:
   a.) In case of setting the office data with LDM, a maximum of six routes per node can be provided with paging trunks, however, this service cannot be used between nodes.
   b.) In case of setting the office data with NDM, a maximum of six routes per FCCS network can be provided with paging trunks. Also, in a FCCS network, the maximum number of concurrent paging for PAGING ACCESS[P-1], RADIO PAGING [R-16] and PAGING TRANSFER [P-10] is 20.
12. Follow the conditions below to use this feature by MC&MG-COT equipment.
   a.) Radio Paging feature is not supported.
   b.) One Paging Trunk is used per MC&MG-COT equipment.
   c.) MG (COT) side accommodates speech signal to LINE0/MG0, and activation signal (loop activation) to LINE1/MG1.
   d.) The power for activation signal is not supplied from MC&MG-COT.
   e.) C.O. line interface cannot be accommodated when MC&MG-COT is used by Paging features. However, MC function can be used to connect analog telephone to TEL2/MC2 and TEL3/MC3. Do not use TEL0/MC0 and TEL1/MC1. When PFT is activated, TEL0/MC0 and TEL1/MC1 are connected to LINE0/MG0 and LINE1/MG1.
13. Follow the conditions below to use this feature by MG-COT [SCA-6COTA]/MG-COT [SCA-6COTB]/ MG-COT [SCA-6COTC].
   a.) Radio Paging feature is not supported.
   b.) One Paging Trunk is used per MCMG/MG-COT.
   c.) MCMG/MG-COT accommodates speech signal to MG0, and activation signal (loop activation) to MG1.

d.) The power for activation signal is not supplied from MCMG/MG-COT.
e.) MCMG's MG2~MG6 connectors and MG-COT's MG2~MG5 connectors can be used for ordinary analog C.O. line interface even when the MCMG/MG-COT is used as a Paging Trunk. Assign the Paging Trunk data by ATRK command different from the C.O. trunk data.

Interactions

1. PAGING TRANSFER (MEET-ME PAGING) and PAGING ACCESS [P-1] can be provided simultaneously in the same system. Separate Access Codes for each service is required. A MEET-ME PAGING Answer Code is required for each MEET-ME PAGING zone.
2. Paging is limited to Stations or Attendant Consoles. Other parties (Central Office Trunk Calls) must have the Attendant Console or Station user perform the paging operation.
3. The Attendant cannot dial the MEET-ME PAGING answer codes.
4. BUSY VERIFICATION [B-3], EXECUTIVE RIGHT OF WAY [E-1]: These features will be denied to Paging routes assigned in CDN 6:TCL as data "8", Page with Ringback Tone.
5. PAGING TRANSFER OPERATION: Station A is connected to Party B. Station A momentarily presses the switch hook or presses the TRANSFER key (if Dterm phone) and then dials the PAGING TRANSFER (MEET-ME PAGE) route Access Code. When connected to the page, Station A requests Station C to dial the MEET-ME PAGE Answer Code. Station A hangs up. When Station C dials the Answer Code, Party B is connected to Station C. Station A is removed from the Connection.
6. If Station C does not call the MEET-ME PAGE Answer Code after 30 seconds, Party B rings back to Station A. This timer can be changed by ASYD System Data 1 Index 74 Bit 0-3.
7. Stations allowed Priority Page, SFI = 57, will barge into the paging route, except when the Attendant Console is paging.
8. To answer PAGING TRANSFER while connected to another party, CALL HOLD [C-6] must be used. THREE-WAY CALLING [T-2] may not be established.
9. A Station must be assigned a Route Restriction Class that allows access to the routes involved in the PAGING TRANSFER.

Programming

Note: For UG50 programming, see Peripheral Equipment Description (UG50) manual.

<When COT Circuit Card is used for Paging Trunk>

STEP 1: **ASYD** - System Data 1, Index 73, Bits 0 and 1:

For MEET-ME PAGE, Non-Delay service, Bit 0=1, Bit 1=0.

For MEET-ME PAGE, Delay service, Bit 0=0, Bit 1=1.

For MEET-ME PAGE, Delay service with Call Transfer, Bit 0=1, Bit 1=1.

System Data 1, Index 73, Bit 3. Are PAGE Cancel Codes provided for each route? 0/1: Per route/ Common to all routes.

System Data 1, Index 74, Bits 0 and 3. Assign the cancel timer for unanswered page. Assign data 00H for 30 seconds.

(0-F + 1) x 30 = Cancel Timer.

System Data 2, Index 1, Bits 0 and 1. Consultation Hold allowed or denied. 00: Denied; 01: Originating and Terminating Calls allowed; Tandem Call Denied; 11: All calls allowed.

STEP 2: **ARTD** - Assign the paging route as shown below:

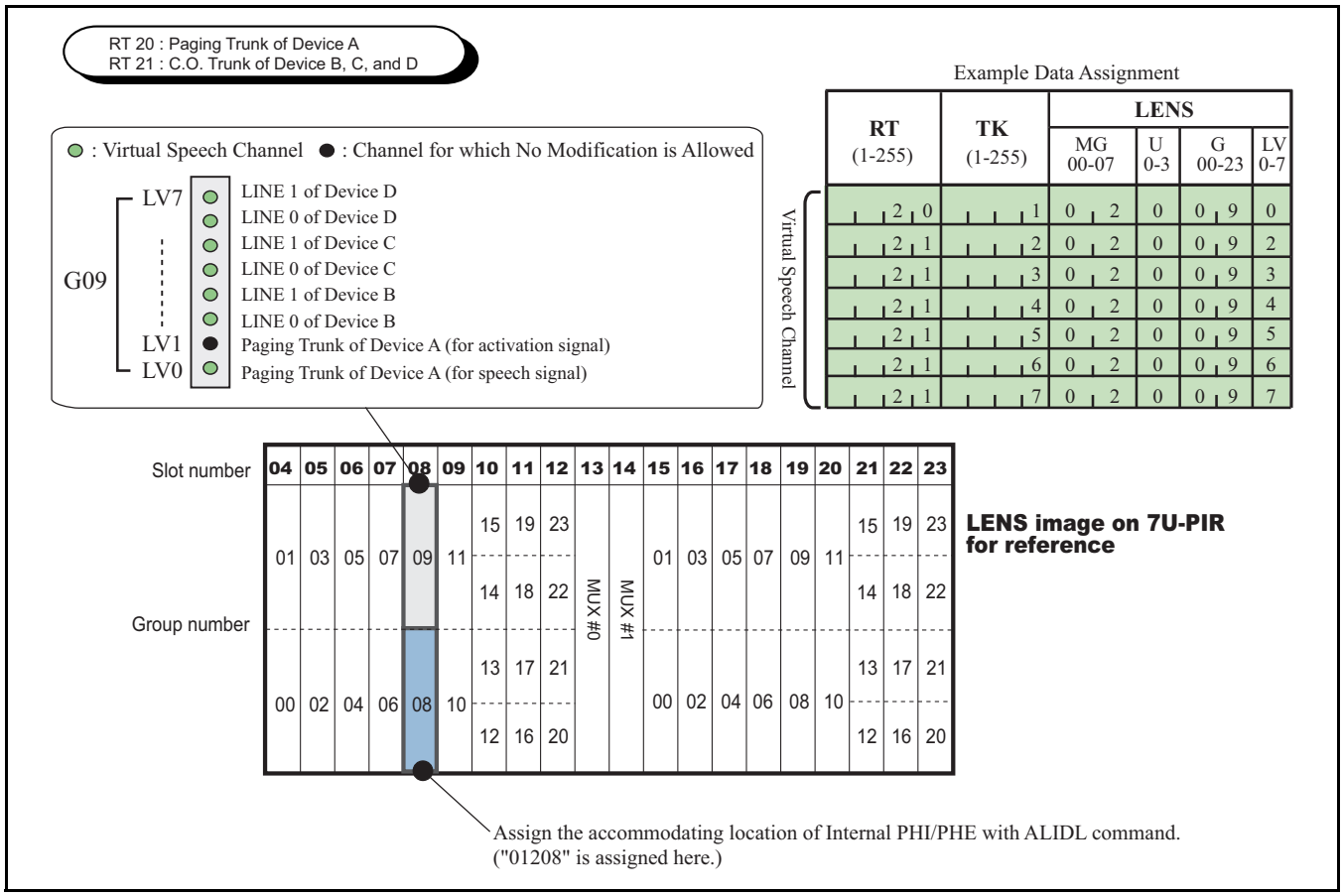
RT: 1

1-OSGS	:2	2-ONSG	:3	5-TF	:1	6-TCL	:8	7-L/T	:1
8-RLP	:2	15-LSG	:1						

Note: To change the data for existing route (RT), initialize the circuit card.

STEP 3: **ATRK** - Assign the LENS, Paging Trunk Route Number, Trunk Number, and Tenant Number.

Example: ATRK data assignment when COT card is used for Paging Trunk



STEP 4: **MBTK** - Assign the Make Idle status to the Paging Trunks.

STEP 5: **ANPD** - Reserve Levels for Trunk Access and for PAGE Answer and Cancel Codes. Select the Trunk Access Level for a Connection Index (CI) of Normal Service (N) or Hooking Service (H). Number of Necessary Digits is usually one or two (maximum of three digits). Busy Lamp Field is not activated.

If the PAGE Answer and Cancel Codes reside in a Level reserved for features, then assign according to a predetermined Numbering Plan. If the PAGE Answer and Cancel Codes reside in a separate Level, then select a CI of Normal Service (N) only. The number of Necessary Digits is usually two, and the Busy Lamp Field is OFF.

STEP 6: **ASPA** - Assign Paging Route Access Code. Type of Service, SRV = “OGC: Outgoing Call”. Assign the proper Route Number that is associated with this Paging Trunk Access Code. Assign for CI = “N: Normal” and “H: Hooking”.

Assign the PAGE answer and cancel Access Code. Type of Service, SRV = "PAGA: Paging Answer" and SRV = "PAGC: Paging Cancel". Assign the proper Route Number that is associated with this Access Code. Assign for CI = "N: Normal".

STEP 7: **ARSC** - Allow RRI: 2 and 3 access to this route for the RSCs assigned to Stations given access to the paging routes. Allow RRI: 0 and 1 on the Incoming Route to be transferred to the RSCs given access to the page answer.

STEP 8: **ASFC** - Assign Station to receive PAGING TRANSFER a Service Feature Restriction Class that allows SFI = 38. This allows a Station to dial PAGE answer codes.

<When MC&MG-COT is used for Paging Trunk>

The following steps assume that the basic setup has already been done on the MC&MG-COT.

STEP 1: **ASYD** - System Data 1

Index73, Bit 1 and Bit 0: 00=No answer, 01=No delay, 10=Delay, 11=Delay with Paging Transfer **Note 3**

Index74, Bit 0-3: Cancel Timer for unanswered Paging

Note 3: Delay with Paging Transfer is applied for the Paging Access on Attendant Console when 01, 10, or 11 is assigned in Index 73, Bit 0 & 1.

Note: Be sure to assign LENS to each LMG accommodates virtual speech/signal channel in 16-IMG system.

STEP 2: **ASTD** - State Translation Data

STM=2, STS=2 (Outgoing call to C.O. line or Tie line after holding COT) is set to ST=1.

STM=2, STS=3 (Outgoing call to C.O. line after holding Station or Tie line) is set to ST=1 (allowed).

STEP 3: **ANPD/ANPDL** - Numbering Plan Data. Assign the minimum number of digits to the first digit of Access Code for Paging Access and Paging Answer.

STEP 4: **ASPA/ASPAL** - Special Access Code.

SRV="OGC: Outgoing Call", assign the route number to the Access Code of Paging Access.

SRV="PAGA: Paging Answer", assign the route number to the Access Code of Paging Answer.

SRV="PAGC: Paging Cancel", assign the route number to the Access Code of Paging Cancel.

STEP 5: **ARSC** - Route Restriction Class. Release the restriction of Paging Trunk access on stations, RRI-2 and RRI-3 to be allowed.

STEP 6: **ARRC** - Alternate Route Restriction Data. Release the restriction of Paging Trunk access on incoming trunk.

STEP 7: **ASFC** - Service Feature Restriction Class. Assign the Service Feature Restriction Class of Paging Answer. SFI 38 (Meet-Me Paging) is allowed.

STEP 8: **ARTD** - Route Class Data. Assign the following route class data to the Paging Trunk route.

CDN 1: OSGS (Signal Interface for Outgoing) = 2 (Second Dial Tone)

CDN 6: TCL (Trunk Class) = 8 (Paging)

In addition to the basic data of Paging Trunk, the following data assignment is required:

CDN 60: TC/EC=1 (Echo Canceller is available)

CDN 114: DCH=1 (Channel integration is available)

CDN 125: MGCOT=1 (MG-COT is available)

Note: To change the data for existing route (RT), reboot the MG.

STEP 9: **ARTI** - Trunk Application Data. Assign the application data to Paging Trunk route.
CDN 68: VIR=2 (MG connection)

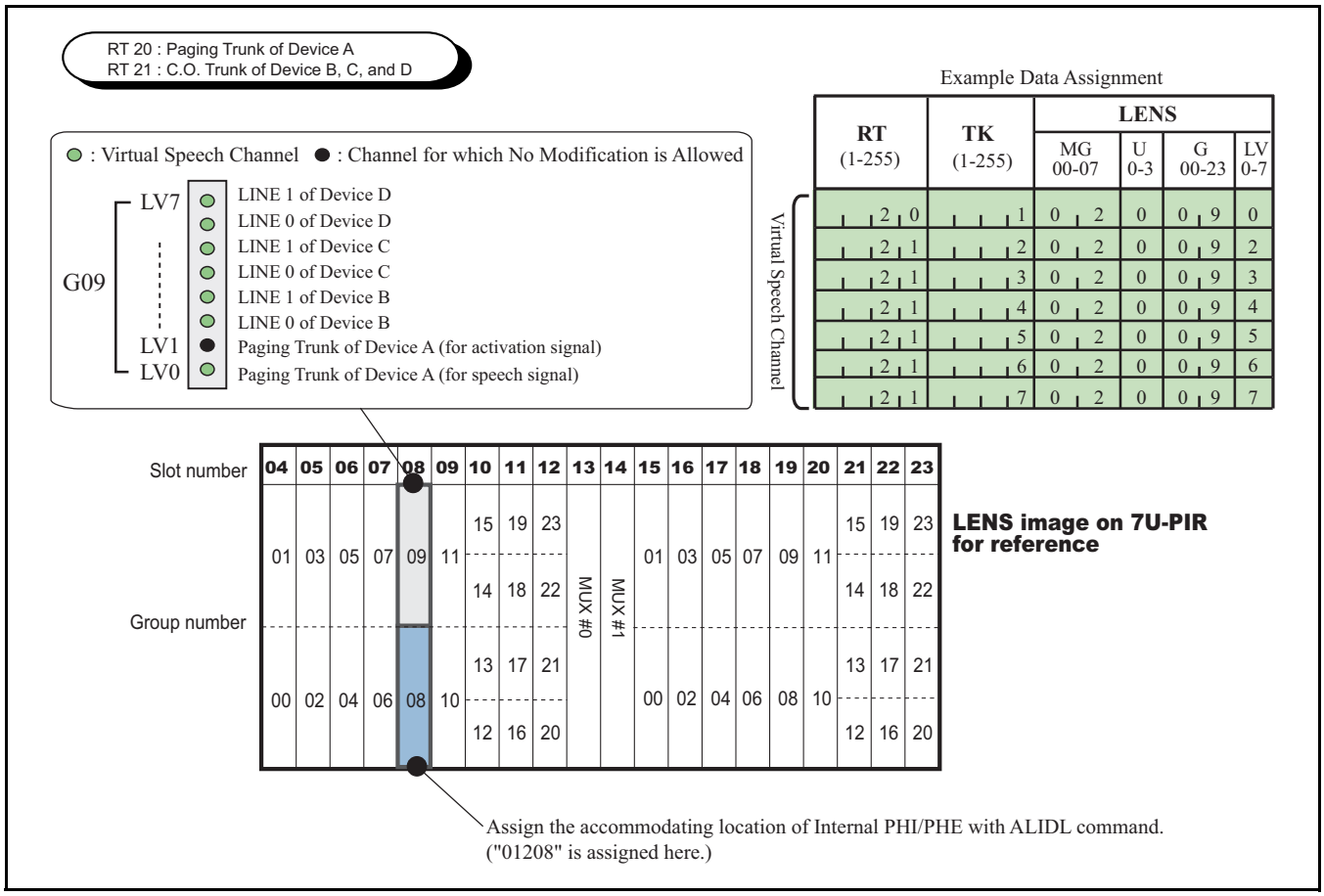
Note: To change the data for existing route (RT), reboot the MG.

STEP 10: **ATRK** - Trunk Data. Assign the Paging Trunk. Be sure to assign the LENS of Paging Trunk to even numbered LV.

Note: Please note the following important points:

- Be sure to assign the LENS of Paging Trunk on LV0.
- Do not assign the Paging Trunk data to even-numbered Module Group, Unit=0, and Group=0. MC&MG-COT cannot use this LENS.
- Do not assign the trunk data of other MG devices.
- Do not assign anything to LV1, LV6, and LV7.
- Make sure not to assign the Route Number (RT) duplicating the MG-COT route.
- The first two channels of an MC&MG-COT card can be used for a Paging Trunk. In this case, these channels cannot be connected to C.O. lines.

Example: ATRK data assignment when MC&MG-COT is used for Paging Trunk



STEP 11: **MBTK** - Trunk Make Busy. Release the make-busy state of the Paging Trunk, specifying the route number and the trunk number.

STEP 12: **AMGIL** - Media Gateway Information. Assign the MC&MG-COT related data.

MG-ID: MAC address of MC&MG-COT

KIND: MC-MG(COT)

UG50 Multi-Slot Mode: Select this check box if registering UG50 in Multi-slot mode.

LINE: Fixed to 0. **Note 4**

CH: Fixed to 0.

LENS: Mounting location of Paging Trunk (even numbered LV). **Note 5**

Setting Type of Signaling Packet Service: QoS setting for the signal route of MG-COTMG-COT (ToS value for signal control)

Music On Hold Exists: No setting

Country Code: Select the country name

Law of A/D Conversion: μ-law/A-law **Note 6**

PB Receive Complete Timing of Dial-In [0-15]: Fixed to 0.

False Answer Timing: Fixed to 0.

Note 4: Make sure that the ATRK data has already been assigned before assigning the LENS data by AMGIL command. (Be sure to assign the trunk data in the order of ATRK → AMGIL.)

Note 5: Please note the following important points:
- LENS:000000 is not available for the users.
- Make sure that the LENS list in this command includes the data that you specified with the ATRK command in the previous step.
- Of the first two channels for the Paging Trunk, do not assign C.O. trunk data to the one with the larger channel number (CH=1) with the AMGIL command.

Note 6: Recommended Law of A/D conversion type is displayed according to the selected country code.

<When MG-COT [SCA-6COTA]/MG-COT [SCA-6COTB]/MG-COT [SCA-6COTC] is used for Paging Trunk>

The following steps assume that the basic setup has already been done on the MG-COT [SCA-6COTA]/MG-COT [SCA-6COTB]/MG-COT [SCA-6COTC].

STEP 1: **ASYD** - System Data 1
Index73, Bit 1 and Bit 0: 00=No answer, 01=No delay, 10=Delay, 11=Delay with Paging Transfer **Note 7**
Index74, Bit 0-3: Cancel Timer for unanswered Paging

Note 7: Delay with Paging Transfer is applied for the Paging Access on Attendant Console when 01, 10, or 11 is assigned in Index 73, Bit 0 & 1.

Note: Be sure to assign LENS to each (LMG accommodates virtual speech/signal) channel in 16-IMG system.

STEP 2: **ASTD** - State Translation Data
STM=2, STS=2 (Outgoing call to C.O. line or Tie line after holding COT) is set to ST=1.
STM=2, STS=3 (Outgoing call to C.O. line after holding Station or Tie line) is set to ST=1 (allowed).

STEP 3: **ANPD/ANPDL** - Numbering Plan Data. Assign the minimum number of digits to the first digit of Access Code for Paging Access and Paging Answer.

STEP 4: **ASPA/ASPAL** - Special Access Code.
SRV="OGC: Outgoing Call", assign the route number to the Access Code of Paging Access.
SRV="PAGA: Paging Answer", assign the route number to the Access Code of Paging Answer.
SRV="PAGC: Paging Cancel", assign the route number to the Access Code of Paging Cancel.

STEP 5: **ARSC** - Route Restriction Class. Allow RRI-2 and RRI-3 by releasing the restriction of Paging Trunk access on stations.

STEP 6: **ARRC** - Alternate Route Restriction Data. Release the restriction of Paging Trunk access on incoming trunk.

STEP 7: **ASFC** - Service Feature Restriction Class. Assign the Service Feature Restriction Class of Paging Answer. SFI 38 (Meet-Me Paging) is allowed.

STEP 8: **ARTD** - Route Class Data. Assign the following route class data to the Paging Trunk route.
CDN 1: OSGS (Signal Interface for Outgoing) = 2 (Second Dial Tone)
CDN 6: TCL (Trunk Class) = 8 (Paging)

In addition to the basic data of Paging Trunk, the following data assignment is required:
CDN 60: TC/EC=1 (Echo Canceller is available)
CDN 114: DCH=1 (Channel integration is available)
CDN 125: MGCOT=1 (MG-COT is available)

Note: To change the data for existing route (RT), reboot the MG.

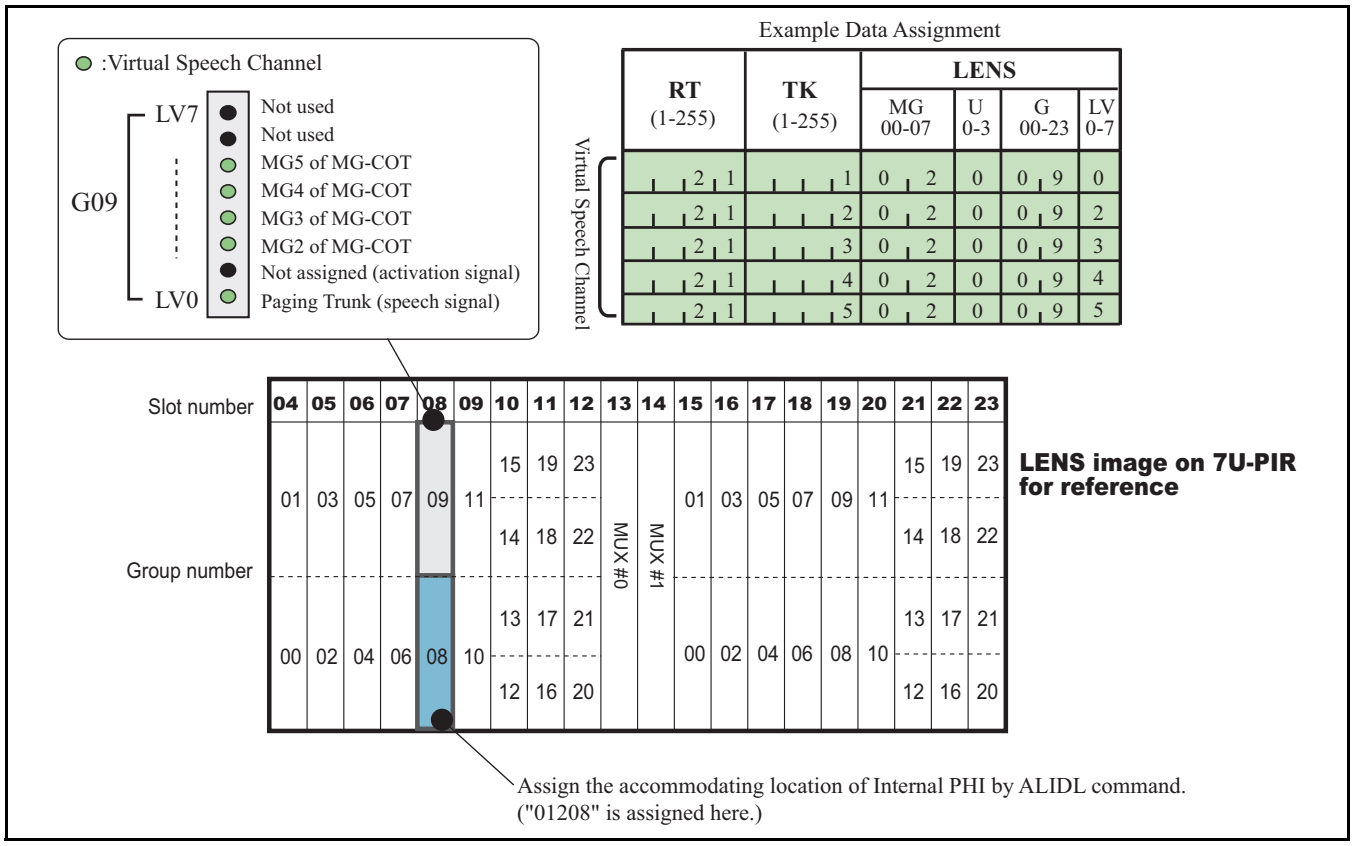
STEP 9: **ARTI** - Trunk Application Data. Assign the application data to Paging Trunk route.
CDN 68: VIR=2 (MG connection)

Note: To change the data for existing route (RT), reboot the MG.

STEP 10: **ATRK** - Trunk Data. Assign the Paging Trunk.

Note: Please note the following important points:
- Be sure to assign the LENS of Paging Trunk on LV0.
- Do not assign the Paging Trunk data to even-numbered Module Group, Unit=0, and Group=0. MG-COT cannot use this LENS.
- Do not assign the trunk data of other MG devices.
- Do not assign anything to LV1, LV6, and LV7.
- Make sure not to assign the Route Number (RT) duplicating the MG-COT route.

Example of ATRK Data Assignment (when MG-COT is used for Paging Trunk)



STEP 11: **MBTK** - Trunk Make Busy. Release the make-busy state of the Paging Trunk, specifying the route number and the trunk number.

STEP 12: **AMGIL** - Media Gateway Information. Assign the MCMG/MG-COT related data.

MG-ID: MAC address of MCMG/MG-COT

KIND: MG(6COT)/MG(6COT) [SIP]

UG50 Multi-Slot Mode: Select this check box if registering UG50 in Multi-slot mode.

LINE: 0 (Click the check box of "LINE0" to ON.)

CH: Fixed to 0.

LENS: Mounting location of Paging Trunk (five digits). *Make sure the LENS data has already been assigned by ATRK command.

Setting Type of Signaling Packet Service: QoS setting for the signal route of MCMG/MG-COT (ToS value for signal control)

Music On Hold Exists: No setting

Country Code: Select the country name

Law of A/D Conversion: μ-law/A-law **Note 8**

PB Receive Complete Timing of Dial-In [0-15]: Fixed to 0.

False Answer Timing: Fixed to 0.

Note 8: Recommended Law of A/D conversion type is displayed according to the selected country code.)

FCCS Programming

This data setting describes only the additional or replacement commands used to activate FCCS Services. When used as an FCCS feature, these commands replace the non-network commands (for example, AAED is replaced by AAEDN). Note that the data setting for FCCS Service varies depending on the Node.

Note: When using ASYDL or ASYDN to program an FCCS feature, the basic programming of ASYD is required in addition to ASYDL or ASYDN.

Note: The current command data shown is set to all Nodes using the service: ASTD, ARTD, ATRK, ASFC

STEP 1: **ALRTN** - This data must be assigned to the NCN. Assign the Logical Route Number to the physical route.

STEP 2: **ANPDN** - This data must be assigned to the NCN. Reserve a Number Level for Service Feature Access and cancel.

STEP 3: **ASPAN** - This data must be assigned to the NCN. Assign the Access Code for Paging Transfer.

STEP 4: **ARSCN** - This data must be assigned to the NCN. Release the restriction between the Announcement Trunk and the Station.

STEP 5: **ARRCN** - This data must be assigned to the NCN. Release the restriction between the terminating Trunk and the Announcement Trunk.

P-11 Periodic Time Indication Tone

General Description

This feature provides a Tone every 180 seconds, if required, to the Station user who has made an Outgoing Call.

Operating Procedure

No manual operation is required.

Service Conditions

1. This feature is programmed on a per Trunk Route basis using the PCPro.
2. The PERIODIC TIME INDICATION TONE is 80 ms in duration and is sent every 180 seconds. See VARIABLE TIMING PARAMETERS [V-1] for other programming time periods.
3. This feature is only available for Station users who have placed Outgoing Calls to a Central Office line or to a TIE Line.
4. Incoming Calls or Outgoing Calls initiated by the Attendant will not have this feature.
5. When the Station user activates SPECIAL DIAL TONE after the Outgoing Call mode, this feature will be canceled automatically.
6. When the Station user activates the following features after the Outgoing Calls to a Central Office line or to a TIE Line, this feature will be canceled automatically:

ATTENDANT CAMP-ON WITH TONE INDICATION	[A-1]
ATTENDANT OVERRIDE	[A-7]
BOSS - SECRETARY OVERRIDE -D	[B-5D]
BUSY VERIFICATION	[B-3]
CALL HOLD	[C-6]
CALL TRANSFER	[C-10, C-11]
CALL WAITING - ORIGINATING	[C-31]
CALL WAITING - TERMINATING	[C-12]
CONSULTATION HOLD - ALL CALLS	[C-17]
DATA PRIVACY ON DEMAND	[D-17]
EXCLUSIVE HOLD -D	[E-4D]
EXECUTIVE RIGHT-OF-WAY	[E-1]
NON-EXCLUSIVE HOLD -D	[N-7D]
OFF-HOOK QUEUING	[O-7]
OUTGOING TRUNK QUEUING	[O-2]
PRIVACY RELEASE	[P-18]
THREE-WAY CALLING	[T-2]
VOICE CALL	[V-2]

Interactions

1. The timers set in System Data 1, Indexes 137 and 138 are approximate. Expect errors of ± 40 seconds. Keeping track of time for this service is a very low priority for the CPU.
2. PERIODIC TIME INDICATION TONE is used for Outgoing Connections only.

Programming

STEP 1: **ASYD** - System Data 1, Index 137. This timer sets the interval when the first tone will be sent to the Station. Assign 00H for 3 minutes.

System Data 1, Index 138. This timer sets the period between the first and second tones and all other subsequent tones. Assign 00H for 3 minutes.

STEP 2: **ARTD** - Assign CDN 29:TELP. Flag the routes with which the tone will operate.

Note: To change the data for existing route (RT), initialize or reboot the trunk device such as circuit card or MG.

STEP 3: **ASFC** - Assign to Stations that receive PERIODIC TIME INDICATION TONE a Service Feature Restriction Class that allows SFI 11 and SFI 33.

P-13D Prime Line Pickup -D

General Description

This feature allows a Dterm Station user to originate or answer a call from the line designated as the Primary Line by going off-hook. It is unnecessary to press the associated Line Key.

Operating Procedure

Lift the handset or press the SPEAKER Key. Receive Dial Tone.

Service Conditions

1. One Prime Line per Station is allowed.
2. Unless otherwise programmed, the MY Line is also the Prime Line.

Interactions

1. If no data is assigned via AKYD, the MY Line will serve as the default Prime Line. If any data is assigned in AKYD, a Prime Line must be assigned, even if the MY Line and Prime Line will be the same Telephone Number.
2. The Prime Line may be different than the MY Line. When they are different, the following features will not operate for the Prime Line: CALL BACK [C-1], CALL PARK [C-29], DO NOT DISTURB [D-21D], and OUTGOING TRUNK QUEUING [O-2]. CALL FORWARDING [C-2, 3, 5, 28] features may be set by the Attendant.
3. CLASS OF SERVICE-INDIVIDUAL [C-15]. The Prime Line will operate according to the CLASS OF SERVICE-INDIVIDUAL of the MY Line.
4. STATION MESSAGE DETAIL RECORDING [S-10]. All calls originated by the Prime Line will be recorded against the MY Line.
5. TOLL RESTRICTION-3/6 DIGIT [T-7]. The Prime Line will operate according to the restrictions of the MY Line.

Programming

AKYD - Assign the Station according to the following parameters:

TN: 1

STN: Station Number

TP: Type of Dterm (0-3)

PL ST, STN: Prime Line Tenant and Station

KYN: Key Number (1-40)

KYI: Service Index

- 0: Key Not Used
- 1: Feature Key
- 2: Multi Line Key

KD: 0

TN: Tenant

STN: Station Number

RG: Ringing Assignment per Multi Line Appearance

- 0: No Ringing.
- 1: Ringing in Day Mode Only.
- 2: Ringing in Night Mode Only.
- 3: Not Used
- 4: Ringing in Day and Night Mode - 30 second delay.
- 5: Ringing in Day and Night Mode - 20 second delay.
- 6: Ringing in Day and Night Mode - 10 second delay.
- 7: Ringing in Day and Night Mode.

P-14D Privacy -D

General Description

This feature allows a Desktop terminal to establish privacy so that no Station can interrupt a call via EXECUTIVE RIGHT-OF- WAY [E-1].

Operating Procedure

Note: Be careful not to set Do Not Disturb feature by pressing the DND key in on-hook status.

To Set:

(While a conversation is in progress)

1. While a conversation is in progress:
2. Press the DND key. The associated LED flashes red.
3. The user carries on the conversation. This feature prevents interruptions for the duration of the call.

(Before making a call)

1. Go off-hook, receive Dial Tone.
2. Press the DO NOT DISTURB (DND) key. The associated LED flashes red. The LCD displays:

PRIVACY SET **(Time Display)**

3. Dial the desired telephone number.
4. A call is established when a called party answers. This feature prevents interruptions for the duration of the call.

To Cancel:

1. While a conversation is in progress (The associated LED flashes red.):
2. Press the DND key. The associated LED goes out. **Note 1**
3. This feature is canceled.

Note 1: When the call is completed, this feature is automatically canceled by going on-hook.

Service Conditions

1. Privacy remains in effect while the Desktop terminal is in use with the feature activated. Override to either party is denied.
2. If the DND key is pressed before the conversation is concluded, PRIVACY -D is released. The LCD displays:

PRIVACY CANCEL **(Time Display)**

3. If the DND key is pressed with the Station on-hook, the DND -D [D-21D] feature has been activated. Calls can be made with DND [D-21D] activated. DND [D-21D] will not deny EXECUTIVE RIGHT- OF-WAY [E-1].
4. The PRIVACY -D feature may be activated or deactivated in any conversation as often as desired.
5. PRIVACY -D denies the following features:

ATTENDANT CAMP-ON WITH TONE INDICATION	[A-1]
ATTENDANT OVERRIDE	[A-7]
BOSS - SECRETARY OVERRIDE -D	[B-5D]
BUSY VERIFICATION	[B-3]
CALL WAITING - ORIGINATING	[C-31]
CALL WAITING - TERMINATING	[C-12]
EXECUTIVE RIGHT-OF-WAY	[E-1]

Programming

STEP 1: **AKYD** - The Desktop terminals are not equipped with a DND key. If required, assign the Station according to the following parameters.

TN: 1
STN: Station Number
TP: Type of Dterm (0-3)
KYN: Key Number (1-40)
KYI: Service Index
0: Key Not Used
1: Feature Key
2: Multi Line Key
KD: Not assigned for Feature Key
FKY: Feature Key Number
FKY = 50, DO NOT DISTURB

STEP 2: **ADSL** - Assign DND key; SN = 1, 2, 3, 4, 5, 6, 7, 15 and FKY = 50 (Do Not Disturb).
This command is required for assigning a function to each Soft key.

P-15D Privacy On All Lines -D

General Description

This feature restricts Dterm users from pressing a Busy Line Button and entering a conversation, which can occur with 1A2 key Telephones without Exclusion Circuits.

Operating Procedure

No manual operation is required.

Service Conditions

1. A Dterm Station user cannot seize the Prime Line if it is busy.
2. LINE PRESELECTION [L-9D] is denied if the selected line is busy.
3. PRIVACY RELEASE [P-18] overrides this feature.

Programming

No programming is required.

P-18 Privacy Release

General Description

This feature allows multiple Stations (maximum of six) accommodated in the same Multi Line Group to override a Dterm already engaged in communication. Also, by key operation of the Desktop terminal already engaged in communication (maximum of 8-Party Communication), overriding from other Stations in the same Multi Line Group can be restricted.

Operating Procedure

To activate PRIVACY RELEASE using the "P-RLS" key

Procedure 1

1. While engaged in a call with a station or trunk on line 201, Station B depresses the "P-RLS" key. The LED of the line 201 on other Dterms of the multi-line group flash. The LCD of Station B displays (for 5 seconds):



The P-RLS LED lights.

2. Station A may either:
   a.) depress the line 201 key, then lift the handset or depress the "SPEAKER" key.
   b.) lift the handset or depress the "SPEAKER" key, receive Dial Tone, then depress line 201 key.
3. THREE-WAY CALLING [T-2] is established. (Three parties receive the interrupting tone.)
4. Line 201 LED of Stations A and B flash, indicating I-USE [I-8], and that of other stations in the multi-line group light. The LCDs of Stations A, B, and the third party display:



The P-RLS LED on Station B goes off.

Procedure 2

1. While engaged in a call with a station or trunk on line 201, Station B depresses the "P-RLS" key. The LED of the line 201 on other Dterms of the multi-line group flash. The LCD of Station B displays (for 5 seconds):



The P-RLS LED lights.

2. Station A (single line) lifts the handset.
3. THREE-WAY CALLING [T-2] is established. (Three parties receive the interrupting tone.)
   Line 201 LED of Station B flashes, indicating I-USE [I-8], and that of other stations in the multi-line group light. The LCD of Station B displays:



The P-RLS LED goes off.

Procedure 3

1. While Station B is engaged in a call on line 201, Station A (Dterm) depresses line 201 key.
2. Station A lifts the handset or depresses the "SPEAKER" key.
3. If Station B has not depressed the "P-RLS" key yet, and if bridge call is available by depressing the "P-RLS" key, the LCD of Station A displays:



The LCD of Station B displays (for 5 seconds):



4. If Station B depresses the "P-RLS" key, THREE-WAY CALLING [T-2] is established. Three parties receive the interrupting tone, and the LCDs display:



The line 201 LED on Stations A and B flash.

Procedure 4

1. While Station B is engaged in a call on line 201, Station A (single line) lifts the handset; receives the Busy Tone.
2. If Station B has not depressed the "P-RLS" key yet, and if bridge call is available by depressing the "P-RLS" key, the LCD of Station B displays (for 5 seconds):



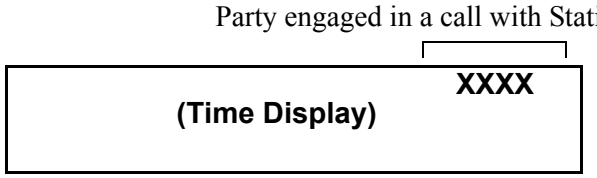
3. If Station B depresses the "P-RLS" key, THREE-WAY CALLING [T-2] is established. Three parties receive the interrupting tone, and the LCDs display:



To activate PRIVACY RELEASE using the "PRIVACY" key.

Procedure 1

1. While Station B is engaged in a call on line 201, Station A depresses line 201 key. The LCD of Station A displays (for 10 seconds):



2. Station A lifts the handset or depresses "SPEAKER" key. THREE-WAY CALLING [T-2] is established. Three parties receive interrupting tone, and the LCDs display:



The line 201 LEDs of Stations A and B flash, indicating I-USE [I-8].

3. By the same operation, a four-party call can be established. In this case, the fourth party receives the Ringback Tone after lifting the handset or depressing the "SPEAKER" key (step 2 above), and the LCD displays:



4. To establish a multiple-party call (max. 8), follow the steps 1 through 3 above. Note, that the last overriding party receives the Call Waiting Ringback Tone instead of the Ringback Tone (step 3).

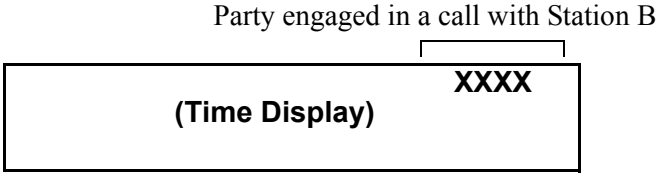
Procedure 2

1. While engaged in a call on line 201, Station B depresses the PRIVACY key. The LCD displays (for 5 seconds):



The PRIVACY LED lights.

2. Station A depresses the line 201 key. The LCD of Station A displays (for 10 seconds):



3. Station A lifts the handset or depresses the SPEAKER key. The LCD displays:



4. Station B depresses the PRIVACY key again. The LCDs of Stations A and B display:



The PRIVACY LED on Station B goes off.

5. By the same operation, a four-party call can be established. In this case, the fourth party receives the Ringback Tone after the PRIVACY key is depressed (step 4 above), and the LCD displays:



Then, a conference call is established.

6. To establish a multiple-party call (max. 8), follow the steps 1 through 5 above. Note, that the last overriding party receives the Call Waiting Ringback Tone instead of the Ringback Tone (step 5).

Procedure 3

1. While Station B is engaged in a call on line 201, Station A (single line) lifts the handset.
2. THREE-WAY CALLING [T-2] is established. Three parties receive the interrupting tone, and the LCD of Station B displays:

CONF **(Time Display)**

Procedure 4

1. While engaged in a call on line 201, Station B depresses the PRIVACY key. The LCD displays (for 5 seconds):

PRIVACY SET **(Time Display)**

The PRIVACY LED lights.

2. Station A (single line) lifts the handset; receives the Busy Tone.
3. Station B depresses the PRIVACY key.
4. THREE-WAY CALLING [T-2] is established. Three parties receive the interrupting tone, and the LCD of Station B displays:

CONF **(Time Display)**

The PRIVACY LED goes off.

Procedure 5

1. While Stations B to I are engaged in an eight-party call on lines 201 and 210, Station A depresses line 201 key. The LCD displays for 10 seconds:

CONF REST **(Time Display)**

Service Conditions

1. PRIVACY RELEASE is available for a Dterm connected to a Station or Trunk.
2. PRIVACY RELEASE is not available when one of the Stations in communication is a single line.
3. Single Line Stations cannot bridge into the busy Connection.
4. This feature is available when the PRIVACY key is pressed to establish PRIVACY RELEASE.
5. A Station has to be in one of the following states to set PRIVACY by pressing the PRIVACY key or the P-RLS key:
   a.) Connected to Station or Trunk,
   b.) Engaged in a maximum Seven-party Call.

   PRIVACY cannot be set when the overriding party has already received Ringback Tone.
6. PRIVACY RELEASE is available anytime by pressing the PRIVACY key or re-pressing P-RLS key. Also, PRIVACY will be released automatically when the Station that has set PRIVACY is released from the call.
7. PRIVACY RELEASE can be established even if the Dterm has activated the DO NOT DISTURB [D-21D] or DATA PRIVACY ON DEMAND [D-17] feature.
8. When the party (Station or Trunk) engaged in a call with the overridden party in PRIVACY RELEASE is released, and when multiple Stations are connected to the same line, PRIVACY RELEASE cannot be activated to any other Stations.
9. When a Multiple-party Call is established by an 8-Party Conference Trunk, the first overridden party can set/cancel Privacy to the Conference Trunk by key operation.
10. When a Station tries to bridge into a call and presses the Line Appearance, and when PRIVACY RELEASE is not available because of the lack of an 8-Party Conference Trunk, the LCD displays as follows, indicating that PRIVACY RELEASE is not available:

    CONF REST
    (Time Display)

11. PRIVACY RELEASE is not available when another Station is in the process of entering a Multiple-party Call (when receiving Ringback Tone). The Station will not receive any tone, and the LCD displays:

    CONF REST
    (Time Display)

12. When the Station engaged in communication with the overridden party is released in the process of entering a Multiple-party Call, the other Stations engaged in a call on the same line will not be released. In this case, the last overriding Station will be connected to an 8-Party Conference Trunk, but after that, EXECUTIVE RIGHT-OF-WAY [E-1] will not be available.

13. The overriding party's LCD display after pressing the Line Appearance of a busy Station is shown below:

Overriding party's terminal status	Overriding party's line status		
	Privacy Release available	Privacy Release not available	
		Conference Trunk Blocked	Not Engaged In Two-party Call
Idle		-	CONF REST
Call Waiting status	CONF	-	CONF RST
Busy	CONF	After disconnecting the busy Station, CONF REST is displayed and the status will be Call Waiting	CONF REST

Note: The Called Telephone Number is displayed even when Privacy is set.

14. PRIVACY RELEASE is available after any of the following operations is performed by the overriding Station:
    a.) The Station user presses the Line Appearance of a busy Station, goes off-hook or presses the SPEAKER key.
    b.) While connected to a Station/Attendant/OG Trunk/IC Trunk, the Station user presses the Line Appearance of a busy Station. (The call in progress will be disconnected.)
    c.) While engaged in THREE-WAY CALLING [T-2], the Station user presses the Line Appearance of a busy Station. The Stations, except the one that has pressed the Line Appearance, will be in a Two-party Connection.
    d.) The Station user presses the Line Appearance of a busy Station while hearing Dial Tone, Ringback Tone, Busy Tone, or Reorder Tone.
    e.) While answering a VOICE CALL [V-2], the Station user presses the Line Appearance of a busy Station. The call in progress will be disconnected.
    f.) While engaged in a Multiple-party Call, the Station user presses the Line Appearance of a busy Station.
    g.) While engaged in calling a Station by INTERCOM [I-9] Call or Intercom Bridge Call, the Station presses the Line Appearance of busy Station.
    h.) After placing a call on CONSULTATION HOLD [H-17], the Station originates a call to a Station or Trunk and then presses the Line Appearance of a busy Station.
15. The following features cannot be activated by any Station in a Multiple-party Call:

CALL TRANSFER-ALL CALLS	[C-11]
CALL HOLD	[C-6]
CONSULTATION HOLD-ALL CALLS	[C-17]
DATA PRIVACY ON DEMAND	[D-17]
NON-EXCLUSIVE HOLD	[N-7D]
EXCLUSIVE HOLD	[E-4D]

16. The following features are restricted to all Stations in a Multiple-party Call:

ATTENDANT CAMP-ON WITH TONE INDICATION	[A-1]
ATTENDANT OVERRIDE	[A-7]
BOSS - SECRETARY OVERRIDE -D	[B-5D]
CALL WAITING-ORIGINATING	[C-31]

CALL WAITING-TERMINATING	[C-12]
EXECUTIVE RIGHT-OF-WAY	[E-1]
MESSAGE REMINDER	[M-19]

Designated Trunk seizure by Attendant

17. 8-Party Conference Trunk assignments are as follows:
    a.) For all ports of the 8-Party Conference Trunk, any Telephone Number of up to 16 digits can be assigned. Also, a phantom number can be assigned to Port 0 (L = 0).
    b.) Ports 1 through 7 (L = 1-7) in the same Conference Trunk must be grouped in the same Circular Hunting Group.
    c.) When only one 8-Party Conference Trunk is available for a Tenant, the Telephone Number or phantom number of Port 0 (L = 0) should be assigned to System Data as the Trunk seizure number. When several Trunks (maximum of 20) are available, the Telephone Number or phantom number of Port 0 (L = 0) in one of the Trunks should be assigned as the Trunk seizure number.
18. 8-Party Conference Trunk seizure:
    a.) When only one Trunk is available for a Tenant, the Trunk cannot be seized unless all of the eight ports of the Trunk (where Trunk seizure number corresponding to the Tenant is assigned) are in idle status.
    b.) When multiple Trunks are available for a Tenant, the Trunk in which all of the eight ports in the Hunting Group (pilot Telephone Number = Trunk seizure number corresponding to the Tenant) are in an idle state, is hunted and seized.
    c.) If an 8-Party Conference Trunk seizure number corresponding to the Tenant is not assigned, the following method of Trunk seizure is also available:
        i.) The Trunk that contains Port 0, whose Telephone Number or phantom number is AB (A = number determined by System Data, B = the last number of the overridden Telephone Number), and all of the eight ports in an idle state are seized.
        ii.) If a Hunting Group (pilot Telephone Number = AB) is assigned among the first Ports of multiple Trunks, the Trunk in the Group where all of the eight ports are in an idle state is hunted and seized.
19. When ASYD, System Data 1, Index 384, Bit 0=0:
    Station B and C are engaged in a Two-party Connection. If Station A presses the Line Appearance for Station B or C, Station A display shows Station Number of B or C; however, this feature cannot be activated.
20. If Multiple Lines are programed beyond different Module Groups, Privacy Release is not available.

Programming

STEP 1: **ASYD** - System Data 1, Index 17, Bit 2. Is PRIVACY RELEASE service to be enabled? 0/1: No/Yes. Assign data 1.

System Data 1, Index 17, Bit 3. Is an Interrupt tone for PRIVACY RELEASE to be enabled? 0/1: No/Yes.

System Data 1, Index 17, Bit 6. Is Line Privacy Expansion service to be enabled? 0/1: No/Yes. Assign data 1.

System Data 1, Index 17, Bit 7, Bridge Call 0/1: Privacy/Non-Privacy Operation.

System Data 2, Index 7, Bits 4-7. Assign the first digit of the phantom Station Number that will be used in creating a STATION HUNTING-CIRCULAR [S-7] Group. The phantom Station Number is assigned to the first port of the 8-Party Conference card. (This first digit must also be designated as a Level used for Stations in ANPD and ASPA.)

System Data 1, Index 241, Bit 5=1 for single line terminals.

STEP 2: **ANPD** - Reserve two Number Levels for Station Access. One of these Number Levels will be for regular Station Numbers and the other will be for the Phantom Stations. This is the same number as assigned previously in System Data 2, Index 7, BITs 4-7.

STEP 3: **ASPA** - For the desired Station Number, assign SRV = STN (Station). Assign Connection Indexes (CI) of "N: Normal" and "H: Hooking".

STEP 4: **ASDT** - Assign Station data to each port of the CFT card.

STEP 5: **AKYD** - Assign FKI = 1, and Feature key Privacy Release [48] to a programmable Line/Feature key.

STEP 6: **ADSL** - Assign P-RLS key; Assign SN = 4 (Communication in Progress) and Feature key Privacy Release [48].

This command is used for assigning a function to each Soft key.

STEP 7: **APHN** - Assign Station Numbers to the CFT card Port 0 Station Numbers. A maximum of five phantom numbers can be assigned to a Port Station Number.

STEP 8: **ASHC** - Assign the Stations of the CFT card that are located on Levels 1-7 to a STATION HUNTING-CIRCULAR [S-7] Group. Assign the Stations associated with the two CFT cards, located on Level 0, to a STATION HUNTING-CIRCULAR [S-7] Group. (When it is necessary to proceed to Step 8 and farther, this data setting is not necessary.)

STEP 9: **ALPE** - Assign the Station Number or phantom number that is assigned to Port 0 of the CFT card.When two or more CFT cards are used in the same Tenant, assign the Station Number or phantom number that is assigned to Port 0 of any CFT cards.

STEP 10: **ASHP** or **ASHU** - When two or more CFT cards are used in the same Tenant, assign the Station Number or the phantom number that has been assigned to Port 0 of CFT card as STATION HUNTING-PILOT [S-9] Group or UCD [U-1] Group. In this case, the Station Number assigned by ALPE is used as the pilot number.

P-23 Periodic Warning Tone

General Description

This feature allows a Periodic Warning Tone to be sent to the station which is overridden, as one of the System audible tones.

Note: This feature is not available in North America and Australia.

Operating Procedure

No manual operation is required.

Service Conditions

1. Following origination of a Periodic Warning Tone is available:
   - Three bursts of 192 ms ON/OFF.
   - The repetition period is between 4 and 30 seconds and can be adjusted in two-seconds steps.
2. The periodic warning tone can be used for the following feature:
   - BUSY VERIFICATION [B-3]
   - ATTENDANT OVERRIDE [A-7]
   - EXECUTIVE RIGHT OF WAY [E-1]

Programming

STEP 1: **ASYD** - System Data 1, Index 59, Bit 5. Warning Tone Type: 0/1 = 2 Burst/3Burst
Bit 6: Sending Periodic Warning Tone for Override services (0/1 = Yes/No)
System Data 1, Index 158. Assigns the Periodic Warning Tone Timer. **Note 1**

Note 1: This data is common to Busy Verification, Execute Right of Way and Attendant Override.

P-24 Preset Conference

General Description

This service is one of the conference facilities that uses the Digital Conference Trunk of the System. This service enables the conference leader to simultaneously call conferees that have been designated beforehand, and to hold the conference by automatically dialing the Feature Access Code for a PRESET CONFERENCE up to 7 conferees for 8-Party Conference by PA-CFTW or 7/15/31 conferees for 8/16/32-Party Conference with PA-CFTV or CH-CFTA.

Note: This feature is not available in North America and Australia.

Operating Procedure

Calling a conference:

1. Dial the Feature Access Code (1-5 digits) for PRESET CONFERENCE.
2. Predesignated conferees (a maximum of 7, 15, or 31) will be called simultaneously and automatically. Receive Ringback Tone during that time.
3. When the first conferee answers, Ringback Tone stops and the conference call mode is entered. Ringing will continue to the conferees who do not answer.
4. The conferees who answer will enter the conference call one by one.

Specified two-party call with a conferee:

1. The conference leader can enter a two-party call mode with a conferee from the conference call mode by dialing the conferee's number (1 to 7/15/31).
2. Other conferees enter the conference call mode excluding the conference leader.

Forced release of a conferee:

1. The conference leader can make forced release of a conferee who is in two-party mode by dialing "#" from the two-party call mode.
2. After forced release, the conference leader returns to the conference if the remaining conferees are in conference mode. If all the remaining conferees have released, the conference leader releases.

Release of the conference call:

1. When one conferee who is in conference mode releases, the remaining conference leader will continue in the conference mode.
2. When the conference leader releases from the conference mode, all the conferees will be released.
3. When all the conferees release, the conference leader will be released.

Service Conditions

1. Calling a conference can be performed from a DTMF station of Dterm station.
2. Conferees need to be assigned from the PCPro. Conferees can register a maximum of seven lines from among DP lines, PB lines, Dterm lines, or outside lines (Central Office Lines and Tie Lines).
3. When a Central Office Line conferee participates in a conference, forced release by the conference leader is necessary because the conference trunk line does not release even if that conferee releases.
4. When a Central Office Line is registered as a conferee, it may happen that Ringback Tone or Busy Tone intrudes during the conference.
5. One Conference package is required for each PRESET CONFERENCE.

6. In case this features is utilized for the purpose of an emergency call (e.g. fire report) to some designated destinations, Central Office Lines are not recommended as the destinations (conferees) since the calling party may not know how to handle this function (i.e. forced release).
7. This feature is not available for external trunks which cannot detect an answering signal.
8. For external trunks which cannot detect a release signal, restrict the connection to conference port.
9. This feature is not available through CCIS/ISDN (including Q-SIG line and Dp Channel TIE line) trunks.

Programming

- When the called party is a station:

STEP 1: **ASYD** - Assign System Data, SYS1.
Index 20, Bit 6=0 (Line Circuit Reverse Relay Control when Reorder Tone is connected to a station.)

STEP 2: **ANPD** - Assigns the minimum necessary number of digits for the first number of the station number for Conference Port.

STEP 3: **ASPA** - Assigns the 1st Digit and NND of Conference Port.

STEP 4: **ASDT** - Assign the following data for Conference Ports.
STN: Station Number (Conference Port Number)
LENS: Line Equipment Numbers for Conference Port **Note 1**
TEC: Telephone Equipment Class
For Conference Leader. 1, 2, 3, 12
For Conference..... 14 (Hot Line)
SFC: Service Feature Restriction Class
RSC: Route Restriction Class

Note 1: Conference Leader's Level (LV) must be assigned "0".

STEP 5: **ASFC** - Assigns the SFC (Service Feature Restriction Class) of each Conference Port.
SFI 12 (Speed Calling - System): 0/1 = Out/In Service
SFI 14 (SMDS-Trunk Base): 0/1 = Out/In Service
SFI 19 (Speed Calling - Station/Group): 0/1 = Out/In Service
SFI 27 (Account Code/Authorization Code): 0/1 = Out/In Service
SFI 32 (Indicates Line Circuit Reverse Relay Control): This data must be assigned "1."
SFI 33 (Periodic Time Indication Tone): 0/1 = Out/In Service
SFI 50 (OFF Premise Extension): This data must be assigned "1".
SFI 58 (SMDS-Station to Station Call): 0/1 = Out/In Service
SFI 71 (Indicates Distinctive Dial Tone): This data must be assigned "1." **Note 2**
SFI 72 (Called Station Hooking Restriction): This data must be assigned "1".
SFI 80 (Conference Port): This data must be assigned "1".
Other SFIs must be assigned "0".

Note 2: This is assigned to the Conferee ports.

STEP 6: **ARSC** - Assigns RSC (Route Restriction Class) of each Conference Port.

Note: RRI 0 (IC via ATTCON) and RRI 1 (IC Directly) must be assigned "0".

STEP 7: **AHLS** - Assigns the called party to each Conferee Port.
HOT TN: Hotline Tenant Number (Conference Port Tenant Number)
HOT STN: Hotline Station Number (Conference Port Number)
TYPE: Connection Type **Note 3**
CON TN: Destination Tenant Number
CON STN: Destination Station Number

Note 3: This data must be assigned "1" (Station).

- When the called party is an outside party:

STEP 1: **ASYD** - Assign System Data, SYS1.
Index 20, Bit 6=0 (Line Circuit Reverse Relay Control when Reorder Tone is connected to a station.)
STEP 2: **ANPD** - Assigns the minimum necessary number of digits for the first number of the station number for Conference Port.
STEP 3: **ASPA** - Assigns the 1st Digit and NND of Conference Port.
STEP 4: **ASDT** - Assign the following data for Conference Ports.
STN: Station Number (Conference Port Number)
LENS: Line Equipment Numbers for Conference Port **Note 4**
TEC: Telephone Equipment Class
For Conference Leader.... 1, 2, 3, 12
For Conference..... 14 (Hotline)
SFC: Service Feature Restriction Class
RSC: Route Restriction Class

Note 4: Conference Leader's Level (LV) must be assigned "0."

STEP 5: **ASFC** - Assigns the SFC (Service Feature Restriction Class) of each Conference Port.
SFI 12 (Speed Calling - System): 0/1 = Out/In Service
SFI 14 (SMDS-Trunk Base): 0/1 = Out/In Service
SFI 19 (Speed Calling - Station/Group): 0/1 = Out/In Service
SFI 27 (Account Code/Authorization Code): 0/1 = Out/In Service
SFI 32 (Indicates Line Circuit Reverse Relay Control): This data must be assigned "1."
SFI 33 (Periodic Time Indication Tone): 0/1 = Out/In Service
SFI 50 (OFF Premise Extension): This data must be assigned "1".
SFI 58 (SMDS-Station to Station Call): 0/1 = Out/In Service
SFI 71 (Indicates Distinctive Dial Tone): This data must be assigned "1." **Note 5**
SFI 72 (Called Station Hooking Restriction): This data must be assigned "1".
SFI 80 (Conference Port): This data must be assigned "1".
Other SFIs must be assigned "0".

Note 5: This is assigned to the Conferee ports.

STEP 6: **ARSC** - Assigns RSC (Route Restriction Class) of each Conference Port.

Note: RRI 0 (IC via ATTCON) and RRI 1 (IC Directly) must be assigned “0”. And release the restriction between the Conference Port Outgoing Route (at RRI 3).

STEP 7: **ASPD** - Assigns the Abbreviated Digit Code to the called party.

STEP 8: **AHLS** - Assigns the called party to each Conferee Port.

HOT TN: Hotline Tenant Number (Conference Port Tenant Number)

HOT STN: Hotline Station Number (Conference Port Number)

TYPE: Connection Type

This data must be assigned “2” (Outside party (ADC)).

CON ADC: Speed Calling Code of Called Party.

P-25 Protection from Abnormal Dialing

General Description

This service offers the following protection features to prevent the Telephony Server station with Toll restriction from
originating a Central Office call by abnormal dialing:

1. Total digit count
2. Restart of Register (DP/DTMF Receiver)
3. Rejection of receiving DP signal from DTMF telephone

Note: This feature is not available in North America and Australia.

Operating Procedure

When abnormal dialing is performed, Reorder Tone will be sent to the station.

Service Conditions

The System has the following features for protection against abnormal dialing:

1. Total Digit Count
   Restriction of origination is imposed when the number of digits received exceeds the number predetermined by the system data. This prevents abnormal dialing by "Central Office Access Code + Hooking", "Central Office Access Code + X" etc. when there is a delay in receiving Dial Tone from the Central Office exchange, and when the length of the delay is irregular.
   When a station dials the digits exceeding the restricted number of digits, the System immediately generates Reorder Tone to the station.
2. Restart of Register
   When the Central Office exchange is slow to send Dial Tone, but the length of the delay is relatively stable and short, abnormal dialing by "Central Office Access Code + X" etc. is prevented by restarting register using time-out of seizure guard timer.
3. Rejection of receiving DP signal from DTMF telephone
   This method prevents improper origination by dialing the area code by DTMF after sending a DP signal from a DTMF station once the link has been established.
4. This feature is available for the station which holds the other station.

Programming

STEP 1: **ASYD** - System Data 1, Index 43, Bit 7. Restriction for Dialing more than the maximum of necessary digits (as assigned via command AMND) 0/1 = Not Required/Required

System Data 1, Index 135. Assigns the Seizure Guard Timer for Restart of Register.

STEP 2: **AMND** - Assign the Maximum Necessary Digits to Destination Code.

STEP 3: **ASDT** - Assigns the "TEC = 2 (PB Station)" to the PB Station.

P-26 Protection from C.O. Dial Tone Delay

General Description

This service allows extend the Register Prepose Timer up to 62 sec. in order to originate from the Telephony Server to an other exchange with a long Dial Tone delay.
This service also extend the Sender Prepose Timer up to 8 sec. to permit speed calling from a station to an other exchange with a long Dial Tone delay.

Note: This feature is not available in North America and Australia.

Operating Procedure

No manual operation is required.

Service Conditions

1. The Prepose Timer and Sender Prepose Timer are set by the system data.
2. The Sender Prepose Timer is effective only for the origination by sender connection such as speed calling on a trunk line using a Second Dial Tone.

P-30 Priority Paging

General Description

This feature allows a Station or an ATTENDANT CONSOLE [A-3] to make a PRIORITY PAGING [P-30] access when a Paging Call cannot normally be originated because another Paging Call is in progress or is waiting for an answer. ATTCON and Station can page anyone over the speaker connected to MC&MG-COT equipment, overriding other party's paging in progress.
PAGING equipment is configured as the Paging Trunk and connected using an interface.

Operating Procedure

Paging Access from a Station

1. The Paging Station goes off-hook. Hear Dial Tone.
2. The Station dials the PRIORITY PAGING [P-30] Access Code. Hear Dial Tone once again.
3. The Station dials the paging Access Code. Hear continuous Ringback Tone for several seconds.
4. The Station performs paging.

Paging Access from the Attendant Console

1. The Attendant presses the Loop key for an available Loop.
2. The Attendant dials the PRIORITY PAGING [P-30] Access Code.
3. The Attendant dials the paging Access Code; hears continuous Ringback Tone for several seconds.
4. The Attendant performs paging.

Service Conditions

1. A PRIORITY PAGING [P-30] access is allowed from a Station or from the ATTENDANT CONSOLE [A-3].
2. This feature is not priority radio paging for use with a Radio Paging Access service.
3. A PRIORITY PAGING access cannot be made from a Station or the ATTENDANT CONSOLE [A-3] which is waiting for an answer to another Paging Call.
4. PRIORITY PAGING Access Code + ALL ZONE PAGING [A-53] Access Code dialing is possible. However, even if this dialing method is used, the PRIORITY PAGING [P-30] function is not activated when the line is busy. It operates in the same way as ALL ZONE PAGING [A-53] Access Code dialing.
5. While a Paging Trunk forced release or Paging Memory clearing is being processed, the caller (Station operator) who activated the PRIORITY PAGING [P-30] access hears continuous Ringback Tone (for about 5 - 19 seconds).
6. If all Paging Trunks are busy, the Paging Trunk forcibly released is Trunk No. 1.
7. If Paging Trunk No. 1 is not equipped or the package is in Make Busy state, the caller (Station or operator) hears Reorder Tone. If Soft MB or Circuit MB is active, Paging Trunk No. 1 is seized.
8. If all Paging Trunks are busy and Paging Trunk No. 1 is being used at the ATTENDANT CONSOLE [A-3], Paging Trunk No. 1 is not forcibly released. In this case, the caller (Station or operator) hears Reorder Tone.
9. Multiple priority paging to the same route is possible.
10. If all Paging Memory being used by PRIORITY PAGING is in use, a PRIORITY PAGING Call to a different route is not forcibly cleared.
11. If all Paging Memory being used by PRIORITY PAGING Calls to a different route, the Paging Memory is not forcibly cleared. In this case, the caller (Station or Attendant) hears Reorder Tone.

12. Forced release of a Paging Trunk is processed so that the Station connected with that specific Paging Trunk is regarded as having released. Therefore, after the forced release of the Paging Trunk, the state of the Station is as follows:
    a.) if there is a call on hold: Dial Tone Connection
    b.) if there is no call on hold: Recall (Calling Only)
    (Answering - Non-Delay)
    Dial Tone Connection (Answering - Delay basis)
    (Paging Transfer)
13. Going on-hook by the Station after Dial Tone Connection will result in Recall. As the Paging Memory becomes free after this Recall, a PRIORITY PAGING Call can be originated.
14. Forced clearance of the Paging Memory is processed as a time-out of a Paging Call awaiting answer. Therefore, after the forced Paging Memory clearance, the state of the Station is as follows:
    a.) if there is no call on hold: Same as before clearance
    b.) if there is a call on hold: Recall **Note 1**

    Note 1: If a Recall cannot be made because of a busy line, etc., the Paging Memory is not cleared until the line is released and Recall is possible.

15. The following can be used as a paging trunk interface:

    - COT Circuit Cards
    - TLT Circuit Cards
    - MC&MG-COT (4LC2COT)
    - MG-COT (6COT)
    - UG50 (GCD-PGTA) **Note 2 Note 3**

    Note 2: UG50 (GCD-4COTA/B/C) cannot be used as a paging trunk interface.

    Note 3: Available from UG50 software version 2.00.

16. Loop signaling is the recommended signaling type.
17. Follow the conditions below to use this feature by MC&MG-COT equipment.
    a.) Radio Paging feature is not supported.
    b.) One Paging Trunk is used per MC&MG-COT equipment.
    c.) MG (COT) side accommodates speech signal to LINE0/MG0, and activation signal (loop activation) to LINE1/MG1.
    d.) The power for activation signal is not supplied from MC&MG-COT.
    e.) C.O. line interface cannot be accommodated when MC&MG-COT is used by Paging features. However, MC function can be used to connect analog telephone to TEL2/MC2 and TEL3/MC3. Do not use TEL0/MC0 and TEL1/MC1. When PFT is activated, TEL0/MC0 and TEL1/MC1 are connected to LINE0/MG0 and LINE1/MG1.
18. Follow the conditions below to use this feature by MG-COT [SCA-6COTA]/MG-COT [SCA-6COTB]/ MG-COT [SCA-6COTC].
    a.) Radio Paging feature is not supported.
    b.) One Paging Trunk is used per MCMG/MG-COT.

c.) MCMG/MG-COT accommodates speech signal to MG0, and activation signal (loop activation) to MG1.
d.) The power for activation signal is not supplied from MCMG/MG-COT.
e.) MCMG's MG2~MG6 connectors and MG-COT's MG2~MG5 connectors can be used for ordinary analog C.O. line interface even when the MCMG/MG-COT is used as a Paging Trunk. Assign the Paging Trunk data by ATRK command different from the C.O. trunk data.

Programming

Note: For UG50 programming, see Peripheral Equipment Description (UG50) manual.

<When COT Circuit Card is used for Paging Trunk>

STEP 1: **ASYD** - System Data 1, Index 73, Bits 0 and 1. For PAGING function only, assign data "00".

STEP 2: **ARTD** - Assign the Paging route as shown below:

RT: 1

1-OSGS	:2	2-ONSG	:3	4-INSG	:3	5-TF	:1	6-TCL	:8
7-L/T	:1	8-RLP	:2	15-LSG	:0/1	44-PRV	:1		

Note: To change the data for existing route (RT), initialize or reboot the trunk device such as a circuit card or MG.

STEP 3: **ATRK** - Assign the LENS, Paging Trunk Route Number, Trunk Number, and Tenant Number.

STEP 4: **MBTK** - Assign the Make Idle status to the Paging Trunks.

STEP 5: **ANPD** - Assign a Level for Trunk Access. Select Normal Service (N) or Hooking Service (H) from the combo box for Connection Indexes. Assign the Number of Necessary Digits according to a predetermined Numbering Plan. Busy Lamp Field is not activated.

STEP 6: **ASPA** - Assign the PAGE Access Code. Type of Service, SRV = "OGC: Outgoing Call". Assign the proper Route Number associated with this Access Code. Assign Connection Indexes, CI = "N: Normal", "H: Hooking".

STEP 7: **ARSC** - Assign to the Station that is allowed access to Paging Trunks a Route Restriction Class (RSC) that is allowed these routes in Route Restriction Indexes (RRI) 2 and 3.

STEP 8: **ASFC** - Allow SFI = 57, Priority Page, to the Service Feature Restriction Class of Stations allowed this feature.

STEP 9: **ASPA** - Assign an Access Code for SRV = "SSC: Service Code", SID = 40 (Priority Page).

<When MC&MG-COT is used for Paging Trunk>

The following steps assume that the basic setup has already been done on the MC&MG-COT.

STEP 1: **ASYD** - System Data 1

Index 186, Bit 6: 1 = MG in ISDN is available
Index 187: Fixed to 00.
Index 73, Bit 1 and Bit 0: Fixed to 00 for Priority Paging.
Index 74, Bit 0-3: Cancel Timer for unanswered Paging.

Note: Be sure to assign LENS to each LMG accommodates virtual speech/signal channel in 16-IMG system.

STEP 2: **ASTD** - State Translation Data
STM=2, STS=2 (Outgoing call to C.O. line or Tie line after holding COT) is set to ST=1.
STM=2, STS=3 (Outgoing call to C.O. line after holding Station or Tie line) is set to ST=1 (allowed).

STEP 3: **ANPD/ANPDL** - Numbering Plan Data. Assign the minimum number of digits to the first digit of Access Code for Paging Access and Paging Answer.

STEP 4: **ASPA/ASPAL** - Special Access Code.
SRV="OGC: Outgoing Call", assign the route number to the Access Code of Paging Access.
SRV="PAGA: Paging Answer", assign the route number to the Access Code of Paging Answer.
SRV="PAGC: Paging Cancel", assign the route number to the Access Code of Paging Cancel.
SRV="SSC: Service Code", SID=40 (Priority Paging)

STEP 5: **ARSC** - Route Restriction Class. Release the restriction of Paging Trunk access on stations, RRI-2 and RRI-3 to be allowed.

STEP 6: **ARRC** - Alternate Route Restriction Data. Release the restriction of Paging Trunk access on incoming trunk.

STEP 7: **ASFC** - Service Feature Restriction Class. Assign the Service Feature Restriction Class of Paging Answer. SFI 38 (Meet-Me Paging) is allowed.

STEP 8: **ARTD** - Route Class Data. Assign the following route class data to the Paging Trunk route.
CDN 1: OSGS (Signal Interface for Outgoing) = 2 (Second Dial Tone)
CDN 6: TCL (Trunk Class) = 8 (Paging)

In addition to the basic paging data shown on the former page.
CDN 44: PRV=1 (Line Trunk Privacy is available)
CDN 60: TC/EC=1 (Echo Canceller is available)
CDN 114: DCH=1 (Channel integration is available)
CDN 125: MGCOT=1 (MG-COT is available)

Note: To change the data for existing route (RT), reboot the MG.

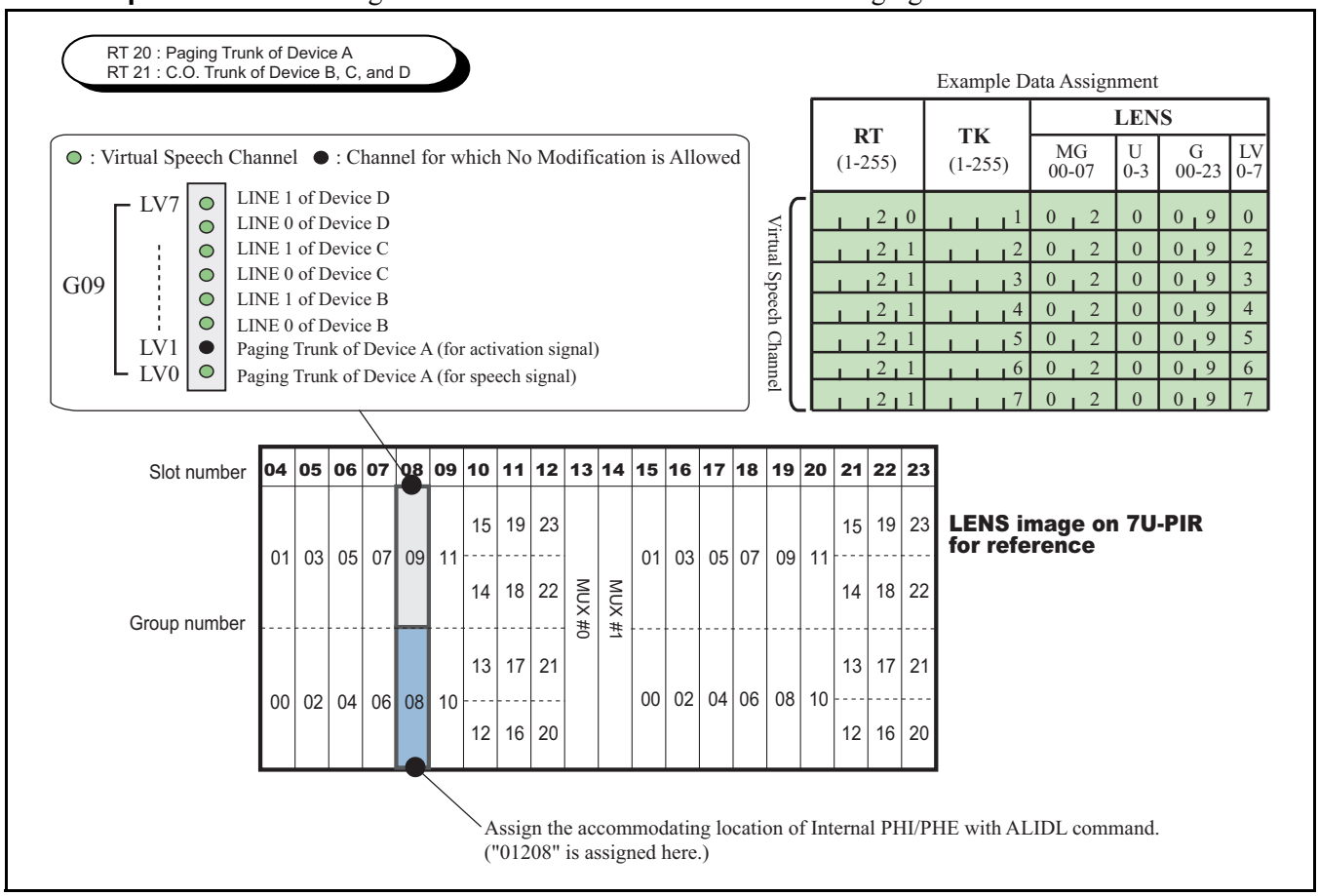
STEP 9: **ARTI** - Trunk Application Data. Assign the application data to Paging Trunk route.
CDN 68: VIR=2 (MG connection)

Note: To change the data for existing route (RT), reboot the MG.

STEP 10: **ATRK** - Trunk Data. Assign the Paging Trunk. Be sure to assign the LENS of Paging Trunk to even numbered LV.

Note: Please note the following important points:
- Be sure to assign the LENS of Paging Trunk on LV0.
- Do not assign the Paging Trunk data to even-numbered Module Group, Unit=0, and Group=0. MC&MG-COT cannot use this LENS.
- Do not assign the trunk data of other MG devices.
- Do not assign anything to LV1, LV6, and LV7.
- Make sure not to assign the Route Number (RT) duplicating the MG-COT route.
- The first two channels of an MC&MG-COT card can be used for a Paging Trunk. In this case, these channels cannot be connected to C.O. lines.

Example: ATRK data assignment when MC&MG-COT is used for Paging Trunk



STEP 11: **MBTK** - Trunk Make Busy. Release the make-busy state of the Paging Trunk, specifying the route number and the trunk number.

STEP 12: **AMGIL** - Media Gateway Information. Assign the MC&MG-COT related data.

MG-ID: MAC address of MC&MG-COT

KIND: MC-MG(COT)

UG50 Multi-Slot Mode: Select this check box if registering UG50 in Multi-slot mode.

LINE: Fixed to 0. **Note 4**

CH: Fixed to 0.

LENS: Mounting location of Paging Trunk (even numbered LV).

Setting Type of Signaling Packet Service: QoS setting for the signal route of MG-COT (ToS value for signal control)

Music On Hold Exists: No setting

Country Code: Select the country name

Law of A/D Conversion: μ-law/A-law **Note 5**

PB Receive Complete Timing of Dial-In [0-15]: Fixed to 0.

False Answer Timing: Fixed to 0.

Note 4: Make sure that the ATRK data has already been assigned before assigning the LENS data by AMGIL command. (Be sure to assign the trunk data in the order of ATRK → AMGIL.)

Note 5: Recommended Law of A/D conversion type is displayed according to the selected country code.

<When MG-COT [SCA-6COTA]/MG-COT [SCA-6COTB]/MG-COT [SCA-6COTC] is used for Paging Trunk>

The following steps assume that the basic setup has already been done on the MG-COT [SCA-6COTA]/MG-COT [SCA-6COTB]/MG-COT [SCA-6COTC].

STEP 1: **ASYD** - System Data 1
Index 73, Bit 1 and Bit 0: Fixed to 00 for Priority Paging.
Index 74, Bit 0-3: Cancel Timer for unanswered Paging

Note: Be sure to assign LENS to each (LMG accommodates virtual speech/signal) channel in 16-IMG system.

STEP 2: **ASTD** - State Translation Data
STM=2, STS=2 (Outgoing call to C.O. line or Tie line after holding COT) is set to ST=1.
STM=2, STS=3 (Outgoing call to C.O. line after holding Station or Tie line) is set to ST=1 (allowed).

STEP 3: **ANPD/ANPDL** - Numbering Plan Data. Assign the minimum number of digits to the first digit of Access Code for Paging Access and Paging Answer.

STEP 4: **ASPA/ASPAL** - Special Access Code.
SRV="OGC: Outgoing Call", assign the route number to the Access Code of Paging Access.
SRV="PAGA: Paging Answer", assign the route number to the Access Code of Paging Answer.
SRV="PAGC: Paging Cancel", assign the route number to the Access Code of Paging Cancel.
SRV="SSC: Service Code", SID=40 (Priority Paging)

STEP 5: **ARSC** - Route Restriction Class. Release the restriction of Paging Trunk access on stations, RRI-2 and RRI-3 to be allowed.

STEP 6: **ARRC** - Alternate Route Restriction Data. Release the restriction of Paging Trunk access on incoming trunk.

STEP 7: **ASFC** - Service Feature Restriction Class. Assign the Service Feature Restriction Class of Paging Answer and Priority Paging. SFI 38 (Meet-Me Paging) and SFI 57 (Priority Paging) is allowed.

STEP 8: **ARTD** - Route Class Data. Assign the following route class data to the Paging Trunk route.
CDN 1: OSGS (Signal Interface for Outgoing) = 2 (Second Dial Tone)
CDN 6: TCL (Trunk Class) = 8 (Paging)

In addition to the basic paging data shown on the former page.
CDN 44: PRV=1 (Line Trunk Privacy is available)
CDN 60: TC/EC=1 (Echo Canceller is available)
CDN 114: DCH=1 (Channel integration is available)
CDN 125: MGCOT=1 (MG-COT is available)

Note: To change the data for existing route (RT), initialize the circuit card.

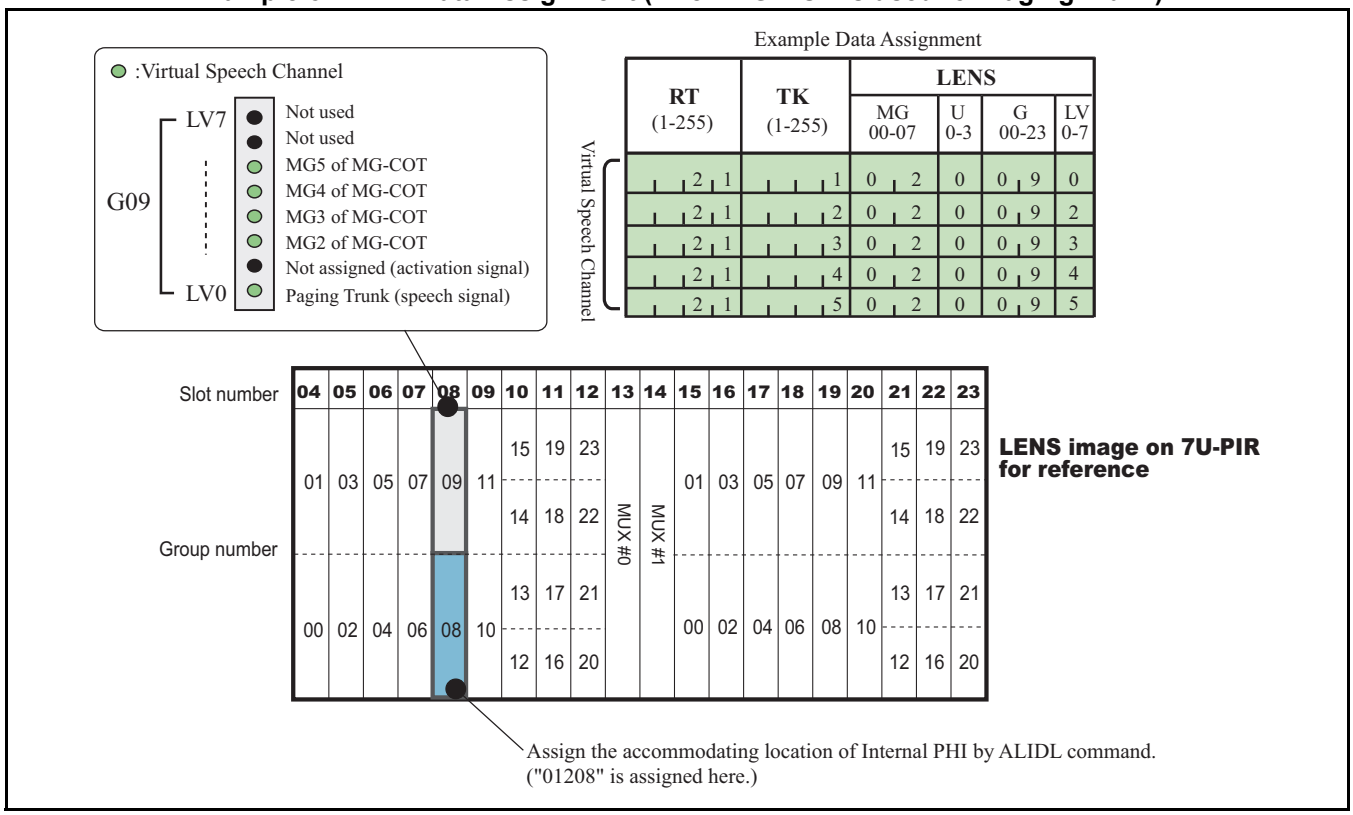
STEP 9: **ARTI** - Trunk Application Data. Assign the application data to Paging Trunk route.
CDN 68: VIR=2 (MG connection)

STEP 10: **ATRK** - Trunk Data. Assign the Paging Trunk.

Note: Please note the following important points:

- Be sure to assign the LENS of Paging Trunk on LV0.
- Do not assign the Paging Trunk data to even-numbered Module Group, Unit=0, and Group=0. MG-COT cannot use this LENS.
- Do not assign the trunk data of other MG devices.
- Do not assign anything to LV1, LV6, and LV7.
- Make sure not to assign the Route Number (RT) duplicating the MG-COT route.

Example of ATRK Data Assignment (when MG-COT is used for Paging Trunk)



STEP 11: **MBTK** - Trunk Make Busy. Release the make-busy state of the Paging Trunk, specifying the route number and the trunk number.

STEP 12: **AMGIL** - Media Gateway Information. Assign the MCMG/MG-COT related data.
MG-ID: MAC address of MCMG/MG-COT
KIND: MG(6COT)/MG(6COT) [SIP]
UG50 Multi-Slot Mode: Select this check box if registering UG50 in Multi-slot mode.
LINE: Fixed to 0.
CH: Fixed to 0.
LENS: Mounting location of Paging Trunk (even numbered LV).
Setting Type of Signaling Packet Service: QoS setting for the signal route of MCMG/MG-COT (ToS value for signal control)
Music On Hold Exists: No setting
Country Code: Select the country name
Law of A/D Conversion: μ-law/A-law **Note 6**
PB Receive Complete Timing of Dial-In: Fixed to 0.
False Answer Timing: Fixed to 0.

Note 6: Recommended Law of A/D conversion type is displayed according to the selected country code.

P-57 Pad Lock

General Description

This feature temporarily restricts telephones from making unauthorized calls by dialing special Access Code when Station users are away from their seats. In addition, the Station user can assign the desired Route Restriction Class (RSC) from the Station.

(only for the Countries except North America)
The Station user can assign his/her own PAD LOCK ID Code, in place of Authorization Code.

Note: This feature is not available in Australia.

Operating Procedure

<For North America>
To set from the Station

1. Lift the handset and receive DT (Dial Tone).
2. Dial PAD LOCK Access Code receive SST (Service Set Tone) or no tone.
3. Dial Authorization Code and receive DT (Dial Tone). PAD Lock is set.
4. Replace the handset.

To cancel from the Station

1. Lift the handset and receive DT (Dial Tone).
2. Dial PAD LOCK Cancel Code and receive SPDT (Special Dial Tone).
3. Dial Authorization Code and receive SST (Service Set Tone).
4. Replace the handset.

To assign the desired RSC from the Station (only for LEAST COST ROUTING [L-5] Calls)

1. Lift the handset and receive DT (Dial Tone).
2. Enter the number to be called.
3. If an Authorization Code is required, the caller will receive SST (Service Set Tone).
4. Enter the Authorization Code. PAD Lock is set.

To restore the RSC form the Station

1. Lift the handset and receive DT (Dial Tone).
2. Dial PAD LOCK Cancel Code and receive SPDT (Special Dial Tone).
3. Dial Authorization Code and receive SST (Service Set Tone).
4. Replace the handset.

Note: For security purposes, the PAD LOCK ID codes (PAD Lock Access Code and Unlock Access Code) can be eliminated by setting ASFC: SFI 137 (Authorization Code Display Elimination) = 1. For details, see Authorization Code Display Elimination [A-99].

<For the Countries except North America>
To set from the Station

1. Lift the handset and receive DT (Dial Tone).
2. Dial PAD LOCK Access Code and receive SST (Service Set Tone) or no tone.
3. Dial Authorization Code and receive DT (Dial Tone) again. PAD LOCK is set.
4. Replace the handset.

To cancel from Station

1. Lift the handset and receive DT (Dial Tone).
2. Dial UNLOCK Access Code and receive SPDT (Special Dial Tone).
3. Dial Authorization Code and receive SST (Service Set Tone).
4. Replace the handset.

To assign the desired RSC from the Station (only for LEAST COST ROUTING [L-5] calls)

1. Lift the handset and receive DT (Dial Tone).
2. Dial the number to be called.
3. If an Authorization Code is required, the caller will receive SST (Service Set Tone).
4. Dial the Authorization Code. PAD Lock is set.

To restore the RSC from the Station

1. Lift the handset and receive DT (Dial Tone).
2. Dial UNLOCK Access Code and receive SPDT (Special Dial Tone).
3. Dial Authorization Code and receive SST (Service Set Tone).
4. Replace the handset.

Setting PAD LOCK ID Code

1. Lift the handset and receive DT (Dial Tone).
2. Dial the Access Code for setting PAD LOCK ID Code and receive SPDT (Special Dial Tone).
3. Dial the desired PAD LOCK ID Code and receive DT (Dial Tone).
4. Dial the PAD LOCK ID Code again and receive SST (Service Set Tone).
5. Dial the desired Route Restriction Class (01-15 / 00) and receive SST (Service Set Tone).

Canceling PAD LOCK ID Code

1. Lift the handset and receive DT (Dial Tone).
2. Dial the Access Code for canceling PAD LOCK ID Code and receive SPDT (Special Dial Tone).
3. Dial the PAD LOCK ID Code and receive SPDT (Special Dial Tone).
4. Dial the PAD LOCK ID Code again and receive SST (Service Set Tone).

Changing PAD LOCK ID Code

1. Lift the handset and receive DT (Dial Tone).
2. Dial the Access Code for setting PAD LOCK ID Code and receive SPDT (Special Dial Tone).
3. Dial the old PAD LOCK ID Code and receive SPDT (Special Dial Tone).
4. Dial the desired PAD LOCK ID Code and receive DT (Dial Tone).
5. Dial the PAD LOCK ID Code again and receive SST (Service Set Tone).
6. Dial the desired Route Restriction Class (01-15 / 00) and receive SST (Service Set Tone).

Setting PAD LOCK

1. Lift the handset and receive DT (Dial Tone).
2. Dial the Access Code for setting PAD LOCK and receive SST (Service Set Tone) or no tone.
3. Dial your own ID Code and receive DT (Dial Tone). PAD Lock is set.

Canceling PAD LOCK

1. Lift the handset and receive DT (Dial Tone).
2. Dial the Access Code for canceling PAD LOCK and receive SST (Service Set Tone) or no tone.
3. Dial your own ID Code and receive SST (Service Set Tone).

To assign the desired RSC from the Station by using the PAD Lock ID Code (only for LEAST COST ROUTING [L-5] Calls)

1. Lift the handset and receive DT (Dial Tone).

2. Enter the number to be called.
3. If an Authorization Code is required, the caller will receive SST (Service Set Tone).
4. Enter your own ID Code instead of an Authorization Code. PAD LOCK is set.

To restore the RSC from the Station by using the PAD Lock ID Code (only for LEAST COST ROUTING [L-5] Calls)

1. Lift the handset and receive DT (Dial Tone).
2. Dial UNLOCK Access Code and receive SPDT (Special Dial Tone).
3. Dial your own ID Code and receive SST (Service Set Tone).
4. Replace the handset.

Note: For security purposes, the PAD LOCK ID codes (PAD Lock Access Code and Unlock Access Code) can be eliminated by setting ASFC: SFI 137 (Authorization Code Display Elimination) = 1. For details, see Authorization Code Display Elimination [A-99].

Service Conditions

1. Stations in Pad Lock status are able to place Station-to-Station Calls and Outgoing Calls using Attendant assisted Calling.
2. If the Station under Pad Lock status dials a Central Office line Access Code, the Station is rerouted to Reorder Tone, the Attendant Console or a recorded announcement depending on to RSC.

   [Example]

   - When Toll Calls are restricted, Toll Denial/Toll Diversion [T-6] can be in service.
   - When Call Originations are restricted, Route Restriction - Announcement [R-27] can be in service.

3. In PAD LOCK status, when trunk calls are received, stations will receive Reorder Tone or, on a Tenant basis, can be assigned to transfer to the Attendant Console.
   When the Attendant Console receives calls transferred from Stations in PAD LOCK status, Call Forwarding - Intercept / Announcement [C-25] can be in service.
4. PAD LOCK status will continue unless it is canceled.
5. Stations are assigned this feature via Class of Service.
6. This feature is set/canceled from the Station only. The Attendant Console cannot set or cancel this feature.
7. If a Dterm is in PAD LOCK status, the My Line and Sub Line are restricted.
8. When the Station in PAD LOCK status attempts to set PAD LOCK again, it is connected to Reorder Tone (RST/REST is displayed on the Dterm).
9. Conversion of Service Feature Restriction Class is not available.
10. SFC class for PAD LOCK is set to My Line even when the SFC was set against Sub Line.
11. The Authorization Code [A-20] feature cannot be applied to the Station, that is already placed in PAD LOCK status. If an Authorization Code is dialed despite this, ROT is returned to the Station (and “RST” is displayed on the LCD, if the Station is Dterm).

(only for the countries except North America)

12. PAD LOCK ID Code is available only for Stations (Single Line Telephone, Dterm, ISDN terminal).
13. PAD LOCK ID Code can be set on a Station basis.
14. The list of PAD LOCK ID codes cannot be output.
15. PAD LOCK ID Code data is saved by the PCPro command or the routine diagnosis program for saving Call Forwarding data.

16. PAD LOCK ID codes cannot be output to SMDR.
17. When PAD LOCK using PAD LOCK ID Code is in service, if the PAD LOCK ID Code is not assigned to a Station, PAD LOCK using Authorization Code is available for the Station.
18. PAD LOCK ID Code assigned from the Station can be deleted by using RPID command.

Interactions

PAD LOCK ID codes are shared with passwords for "DtermIP Login Function". For more details, please refer to Interactions of [D-159] DtermIP.

Programming

STEP 1: **ASYD** - System Data 1, Index 4, Bit 4. Temporary Class Conversion. Assign as data **"**1.**"** Assign the data associated with Authorization Code service.

<To Send No Tone after dialing the Access Code for PAD LOCK>

System Data 1, Index 43, Bit 2. Assign as data **"**0**"** (Out of Service).

<To Send SST after dialing the Access Code for PAD LOCK>

System Data 1, Index 43, Bit 2. Assign as data **"**1**"** (In Service). Index 42, Bit 7. Assign as data **"**0**"** (Standard [SST: Continuous, SPDT: Burst]).

<To Send EPN Tone after dialing the Access Code for PAD LOCK> For North America Only

System Data 1, Index 43, Bit 2. Assign as data "1" (In Service). Index 42, Bit 7. Assign as data "1" (EPN Tone).

System Data 1, Index 93, Bit 6. Separate or Common Tenant Data Table development for AMND command. 0/1 = Separate/Common. Assign 0.

STEP 2: **ASPA** - For SRV = "SSC: Service Code", SID = 42, assign the Access Code for Dial Access to lock. For SRV = "SSCA: Service Code Appendix", SIDA = 85, assign the Access Code for Dial Access to unlock.

(only for the countries except North America)

For SRV = "SSCA: Service Code Appendix", SIDA = 104, assign the Access Code for setting PAD LOCK ID Code. For SRV = "SSCA: Service Code Appendix", SIDA = 105, assign the Access Code for canceling PAD LOCK ID Code.

STEP 3: **ASFC** - For Stations allowed Authorization Code (PAD LOCK), assign a Class of Service that allows SFI 27 (Authorization Code), and assign a Class of Service that allows SFI 125 (PAD LOCK).

(only for the countries except North America)

For Stations allowed PAD LOCK using PAD LOCK ID Code, assign a Class of Service that allows SFI 154.

STEP 4: **AMND** - For Tenant 0, assign the Destination Code (DC) as the first digit of Authorization Code and Maximum Necessary Digits (MND) of Authorization Code.

STEP 5: **AATC** - Assign the TN, Digit Code (DC), the Authorization Code Restriction (ACR) as data **"**1**"**, and Service Feature Restriction Class (SFC) and Route Restriction Class (RSC).

(only for the countries except North America)

For PAD LOCK using PAD LOCK ID Code, this data assignment is not required.

(only for the countries except North America)

This command is used to delete the assigned PAD LOCK ID Code.

RPID - Delete the assigned PAD LOCK ID Code.

TN: Tenant Number

STN: Station Number

P-59 Personal Ringer

General Description

This feature provides another ringer pattern for the Call Terminating to the personal number and allows the station user to distinguish the Called Number upon hearing ringing.
Ringer patterns are as follows:

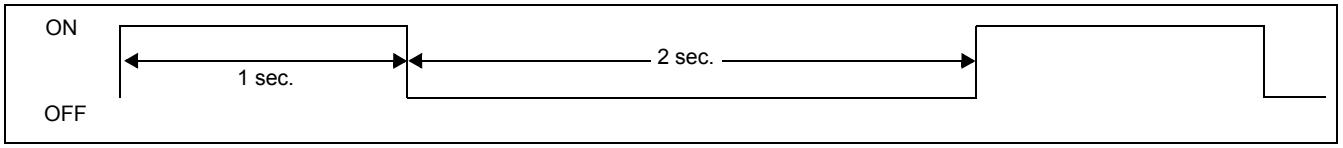
Note: For the definition of terms related to ringer patterns, see DISTINCTIVE RINGING [D-10].

(1) Ringer Pattern 0: Used for an incoming call from a central office line

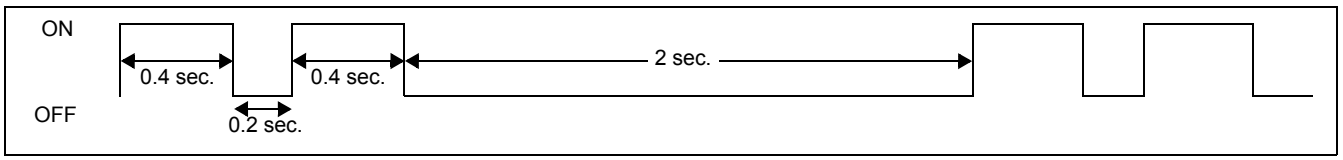
Recommended Setting of Ringer Pattern 0

System Data	Description	Recommended Value
ASYD, System data 3, Index 0	Duration of the Ring Cycle (timer value for ringer used for incoming trunk calls)	21 (Hex)
ASYD, System data 3, Index 3, Bit 0	Multiple Ringings per Ringing Period for Ringer Pattern 0	Bit 0 = 0 (Disabled)

<Recommended Ring Cycle>



<Ring Cycle When Multiple Ringings per Ringing Period is Enabled
(ASYD, System Data 3, Index 3, Bit 0 = 1)>

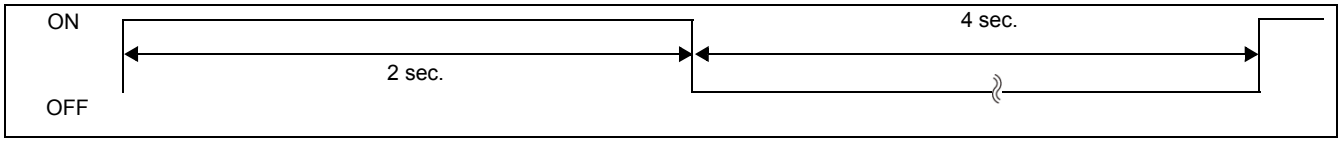


(2) Ringer Pattern 1: Used for an incoming call from a station or TIE Line Trunk to the pilot number.

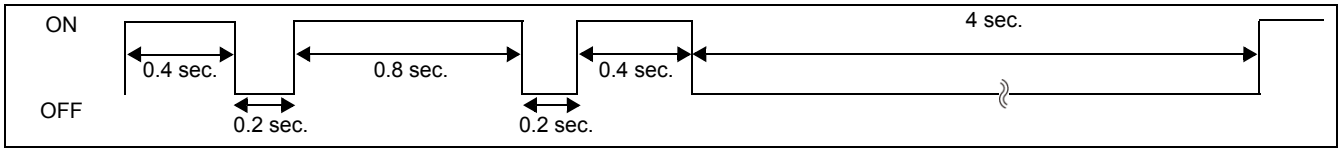
Recommended Setting of Ringer Pattern 1

System Data	Description	Recommended Value
ASYD, System data 3, Index 1	Duration of the Ring Cycle (timer value for ringer used for station calls)	42 (Hex)
ASYD, System data 3, Index 3, Bit 1	Multiple Ringings per Ringing Period for Ringer Pattern 1	Bit 1 = 0 (Disabled)

<Recommended Ring Cycle>



<Ring Cycle When Multiple Ringings per Ringing Period is Enabled
(ASYD, System Data 3, Index 3, Bit 1 = 1)>

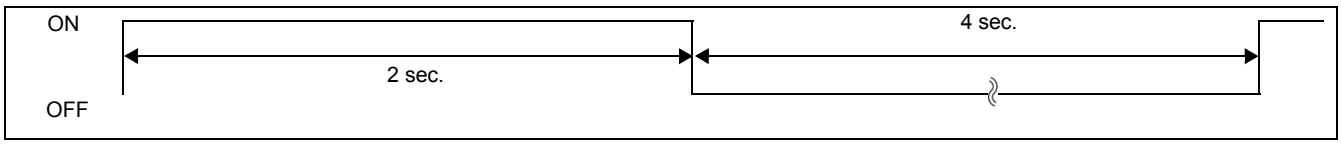


(3) Ringer Pattern 5: Used for an incoming call from a station or TIE line trunk to the Personal Number, CALL BACK [C-1], OUTGOING TRUNK QUEUING [O-2], Priority Ringing in CALL WAITING [C-12], AUTOMATIC/MANUAL INTERCOM [A-30], and DIAL INTERCOM [D-87].

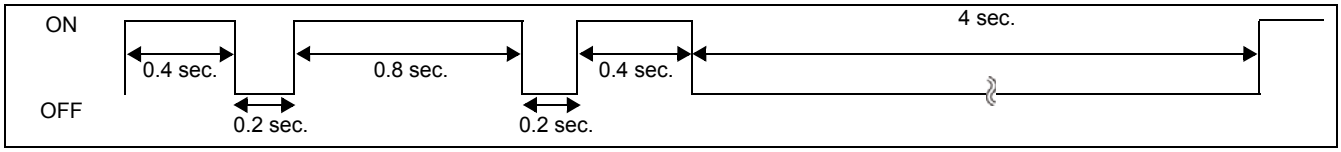
Recommended Setting of Ringer Pattern 5

System Data	Description	Recommended Value
ASYD, System data 3, Index 7	Duration of the Ring Cycle (timer value for ringer for Call Back, OG Trunk Queuing, Priority Ringing in Call Waiting, Auto/Manual Intercom, and Dial Intercom)	42 (Hex)
ASYD, System data 3, Index 3, Bit 5	Multiple Ringings per Ringing Period for Ringer Pattern 5	Bit 5 = 0 (Disabled)

<Recommended Ring Cycle>



<Ring Cycle When Multiple Ringings per Ringing Period is Enabled
(ASYD, System Data 3, Index 3, Bit 5 = 1)>



Operating Procedure

No manual operation is required.

Service Conditions

1. When a station/TIE line incoming call is forwarded with an operator/station assistance, ringer pattern is determined according to Service Feature Restriction Class of the forwarded station.
2. When a My Line of another Desktop terminal is assigned to a Sub Line, Ringer Pattern 5 is used for station/ TIE Line Incoming Calls to the Sub Line. Accordingly, ringer of the Sub Line must be set to OFF with the AKYD command. (Assign 0 to the parameter RG.)
3. This feature is available for Desktop terminals (i.e. not available for an analog station or Virtual Line).
4. Ringer Pattern 0 is used for central office line incoming calls regardless of this feature.
5. SFI-148 (Personal Ringer) has precedence over the DT700/DT800 Ringing Tone configured on the terminal.

Interactions

1. Even if a station/TIE line incoming call is forwarded by STATION HUNTING - TERMINAL [S-9]/ STATION HUNTING - CIRCULAR [S-7] or UNIFORM CALL DISTRIBUTION [U-1], ringer pattern is determined according to Service Feature Restriction Class of the forwarded station.
2. The ringer pattern options provided by this feature are not available on telephones connected to UG50, where those patterns are translated into corresponding ringer patterns. For details, see Peripheral Equipment Description (UG50), Chapter 1, “Limitations on Voice and Tone.”

Programming

STEP 1: **ASYD** - System Data 1, Index 417, Bit 0: Recall Ringer Pattern for CALL BACK [C-1], CALL WAITING - ORIGINATING [C-31], and OUTGOING TRUNK QUEUING [O-2]. 0/1 = Ringer Pattern 5/Ringer Pattern 1.

System Data 3, Index 0. Duration of the Ring Cycle (Timer Value) used for Ringer Pattern 0.

System Data 3, Index 1. Duration of the Ring Cycle (Timer Value) used for Ringer Pattern 1.

System Data 3, Index 3. Multiple Ringings per Ringing Period

Bit 0: 2 Ringings (0.4 seconds-ON, 0.2 seconds-OFF, 0.4 seconds-ON) per Ringing Period for Ringer Pattern 0. 0/1 = Disabled/Enabled.

Bit 1: 3 Ringings (0.4 seconds-ON, 0.2 seconds-OFF, 0.8 seconds-ON, 0.2 seconds-OFF, 0.4 seconds-ON) per Ringing Period for Ringer Pattern 1. 0/1 = Disabled/Enabled.

Bit 5: 3 Ringings (0.4 seconds-ON, 0.2 seconds-OFF, 0.8 seconds-ON, 0.2 seconds-OFF, 0.4 seconds-ON) per Ringing Period for Ringer Pattern 5. 0/1 = Disabled/Enabled.

System Data 3, Index 7. Duration of the Ring Cycle (Timer Value) used for Ringer Pattern 5.

Note: For information on how to customize the Ringer Pattern, refer to Command Manual.

Note: When the System Data listed above is changed, circuit card (ELC) initialization is required.

STEP 2: **ASDT** - Assign station data of the Desktop terminal using this service.

STEP 3: **ASFC** - Assign Service Feature Restriction Class (SFC) that allows SFI = 148 (Personal Ringer) to the Desktop terminals using this service.

STEP 4: **AKYD** - When assigning a My Line of another Desktop terminal to a Sub Line on your Desktop terminal, turn off the ringer of the Sub Line by setting the parameter RG to 0.

P-60 Privacy Release - Single Line

General Description

When a single line station user is engaged in a call (station or trunk), and when the same single line is assigned as a Sub Line of a Desktop terminal, this feature allows the Dterm user to enter the connection by depressing the Sub Line key, resulting in the three-party call.

Note: This feature is not available in North America and Australia.

Operating Procedure

To activate this feature:

1. The single line station user is engaged in a call.
2. The Dterm user lifts the handset or depresses the SPEAKER key; receives Dial Tone.
3. The Dterm user depresses the sub-line key where the single line is assigned. Camp-On Tone is sent to the single line and the station/trunk engaged in a call.
4. Three-party call is established.

Service Conditions

1. Only up to three-party call can be established.
2. This feature is available depending on the station service class of the single line station.
3. This feature is not available when the single line station user is engaged in a call with the Attendant.

Programming

STEP 1: **ASYD** - System Data 1, Index 17, Bit 2. 0/1 = Privacy Release is not provided/provided.
System Data 1, Index 17, Bit 3. Does Privacy Release Warning Tone send? 0/1 = No/Yes.
System Data 1, Index 17, Bit 7. Kind of key assigned to a Line/Feature key or Soft key on the Desktop terminal. **Note 1**
0 = P-RLS key (When this key is pressed, Privacy Release is activated.)
1 = DND key (When this key is pressed, Privacy is activated.)

Note 1: This bit is valid when assigning System Data 1, Index 17, Bit 2 = 1 (Privacy Release is provided).

STEP 2: **AKYD** - Assign Privacy Release key or DND key to a Line/Feature key on the Desktop terminal.
KYI = 1, FKY = 48 (Privacy Release) or FKY = 50 [Do not Disturb (DND)]

STEP 3: **ASFC** - Assign Service Feature Restriction Class that allows SFI = 158 (Privacy Release - Single Line) to a single line telephone using this feature.

P-75 Peer to Peer Bandwidth Control

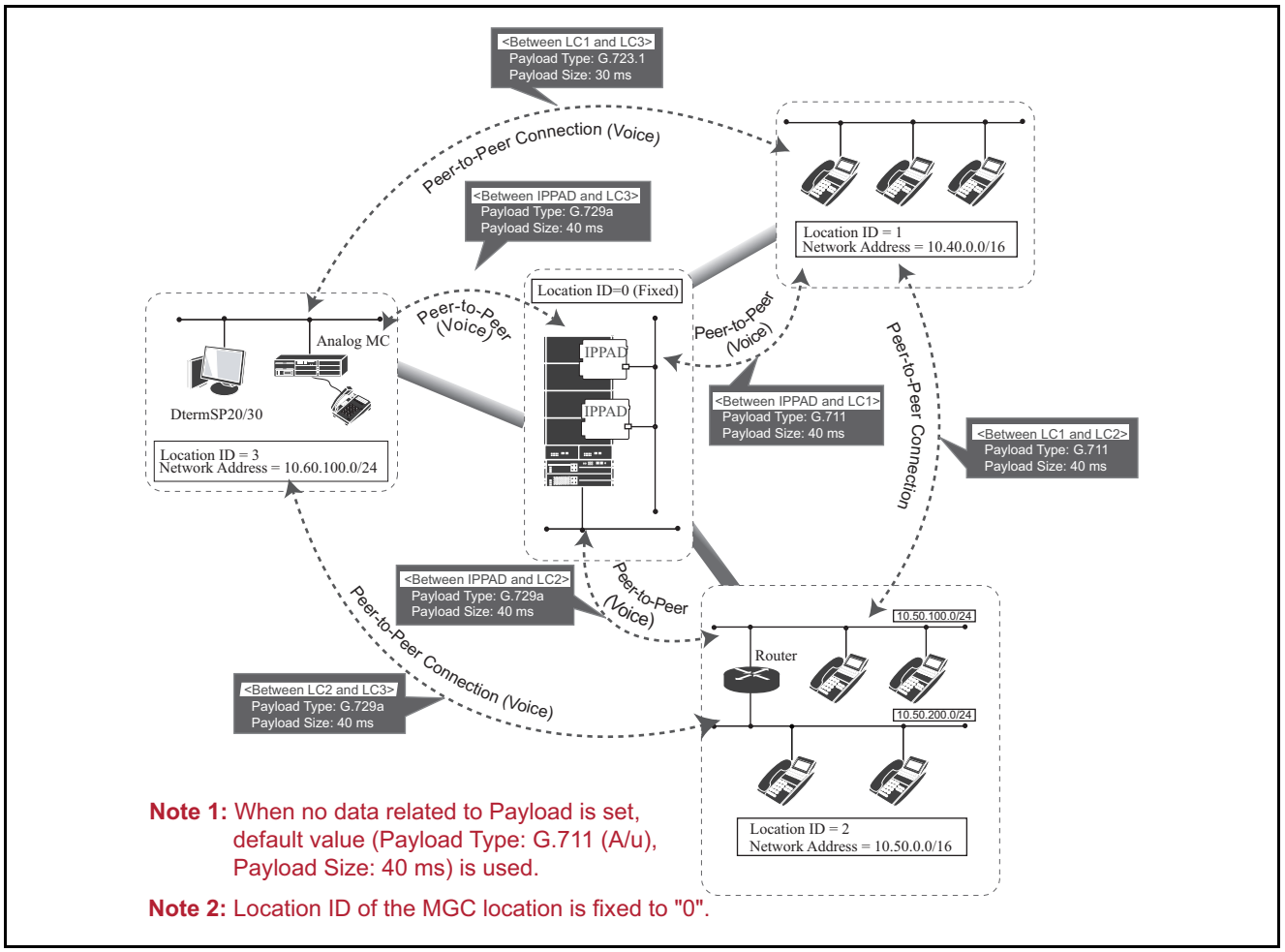
General Description

This feature controls the bandwidth for speech signal to prevent network congestion and decline in voice quality. Also, this feature allows the System to manage the bandwidth value in use statistically. When a call is originated or terminated, the system automatically determines whether sufficient bandwidth can be reserved or not.

Bandwidth Control for Virtual Line is available. This enhancement enables the specific Station in the location to use a different bandwidth by assigning a Location ID to the Station Number individually. (Hereinafter called "Virtual Line Bandwidth Control")

Note: Location is a group of equipment accommodated to Telephony Server, which is divided by network address unit. This is a basic unit for performing LAN management (ToS control, Bandwidth control, VLAN). Each location requires an unique ID, which is called Location ID (LOC-ID).

Bandwidth Control



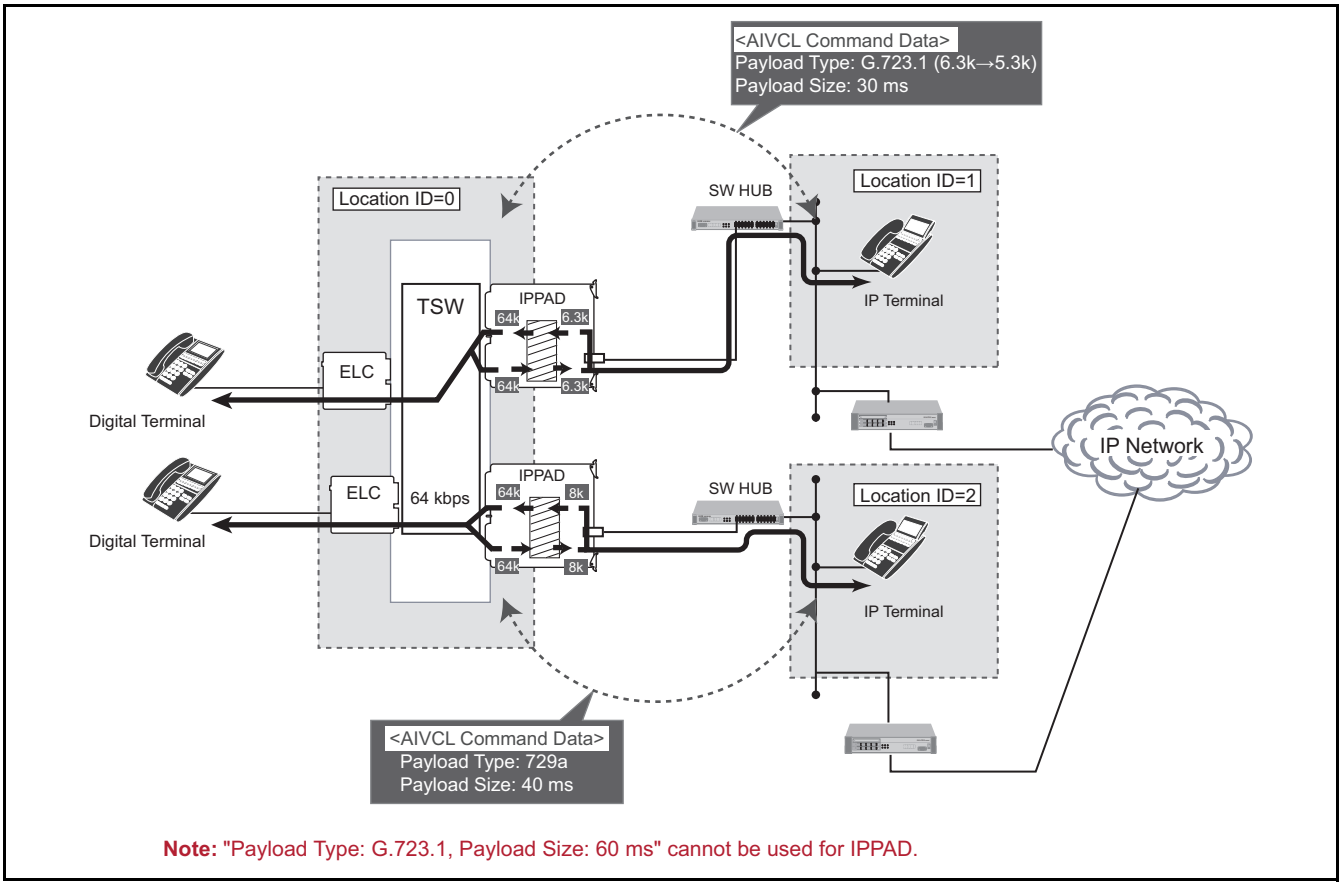
Note 1: When no data related to Payload is set, default value (Payload Type: G.711 (A/u), Payload Size: 40 ms) is used.

Note 2: Location ID of the MGC location is fixed to "0".

Bandwidth Control for IPPAD

Voice compression function is performed between the IPPAD card and IP terminals. IPPAD receives “uncompressed” PCM signal from the System, and compresses the received signal on the card. The compressed signal is sent to the IP terminals. (See the figure below)

Bandwidth Control for IPPAD



Note: "Payload Type: G.723.1, Payload Size: 60 ms" cannot be used for IPPAD.

The system can manage the bandwidth in use according to the Office Data setting. This will allow the system to determine whether sufficient bandwidth can be reserved or not for a Call Origination/Termination. If there is not enough bandwidth for the call, the system restricts the Call Origination/Termination. By restricting the bandwidth used for voice over IP, the system can prevent congestions and deterioration of the voice quality. Also, the system can maintain the bandwidth for other data communications and minimize influence to network.

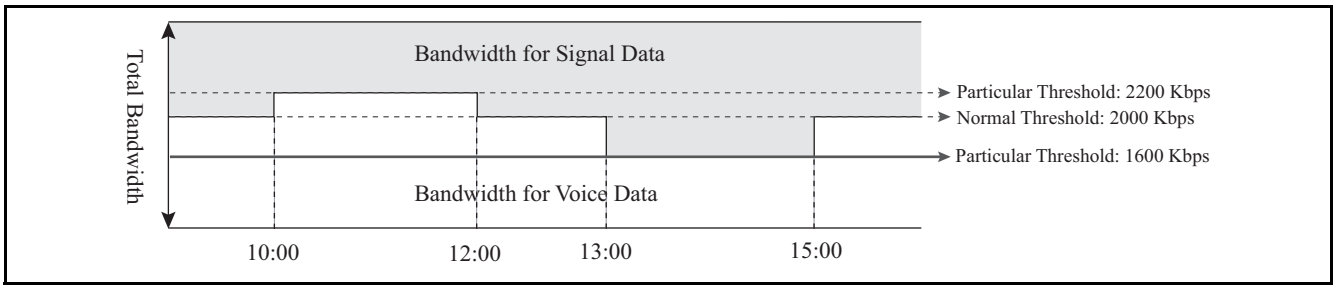
Bandwidth Threshold Value

Bandwidth threshold value is determined with consideration given to the bandwidth of network and the load on signal data. The load on signal data is increased/decreased depending on the time. The threshold value that is applied only for specified time zone should be set, in addition to the Normal threshold value.

Example: Normal Bandwidth: 2000 Kbps

Particular Bandwidth during 10:00 - 12:00: 2200 Kbps

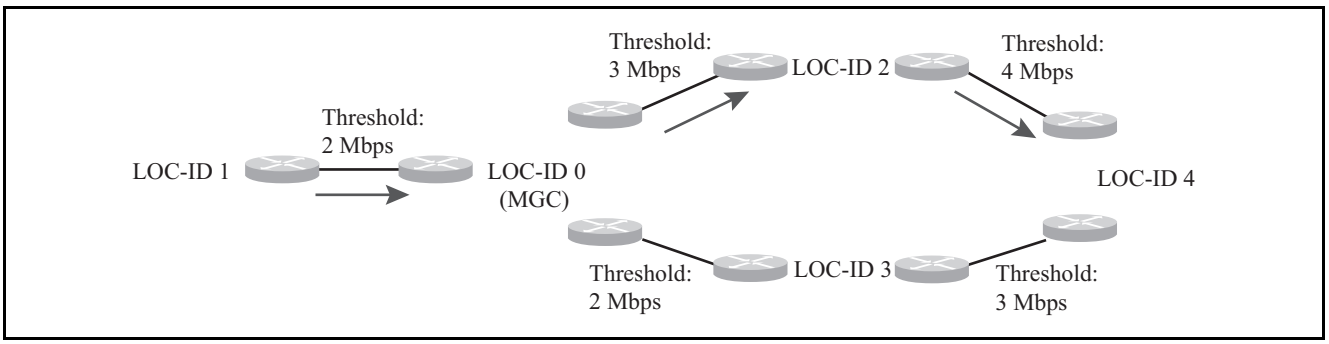
Particular Bandwidth during 13:00 - 15:00: 1600 Kbps



Bandwidth-determination

When a call is originated or answered, the system automatically determines whether sufficient bandwidth can be reserved or not. If there is not enough bandwidth for the call, the system restricts the Call Origination/Termination (treated as busy condition).

Example: Call between LOC-ID 1 and LOC-ID 4



Example of Threshold Programming

A LOC-ID	B LOC-ID	Threshold Value
LOC-ID = 0	LOC-ID = 1	2 Mbps
LOC-ID = 0	LOC-ID = 2	3 Mbps
LOC-ID = 0	LOC-ID = 3	2 Mbps
LOC-ID = 2	LOC-ID = 4	4 Mbps
LOC-ID = 3	LOC-ID = 4	3 Mbps

Example of Routing Table

	B LOC-ID = 0	B LOC-ID = 1	B LOC-ID = 2	B LOC-ID = 3	B LOC-ID = 4
B LOC-ID = 0		(Adjacent)	(Adjacent)	(Adjacent)	LOC-ID = 2
B LOC-ID = 1			LOC-ID = 0	LOC-ID = 0	LOC-ID = 0, 2
B LOC-ID = 2				LOC-ID = 4	(Adjacent)
B LOC-ID = 3					(Adjacent)
B LOC-ID = 4					

How to determine:

1. Reads the relay Locations between Location 1 and 4 from the routing table.
   (LOC-ID = 1) - (LOC-ID = 0) - (LOC-ID = 2) - (LOC-ID = 4)
2. Reads all the threshold values assigned between each Locations.
3. Compares the threshold values with the actual used bandwidth between each Locations, and then determine whether sufficient bandwidth can be reserved or not.

	Actual Used Bandwidth	Threshold Value		
between LOC-ID 1 and LOC-ID 0:	2 Mbps	2 Mbps	→	Not Enough
between LOC-ID 0 and LOC-ID 2:	2 Mbps	3 Mbps	→	Enough
between LOC-ID 2 and LOC ID 4:	1.5 Mbps	4 Mbps	→	Enough

4. The call is restricted because there is not enough bandwidth between LOC-ID 1 and LOC-ID 0. (When there is not enough bandwidth even at a Location, the call is restricted.)

Operating Procedure

None

Service Conditions

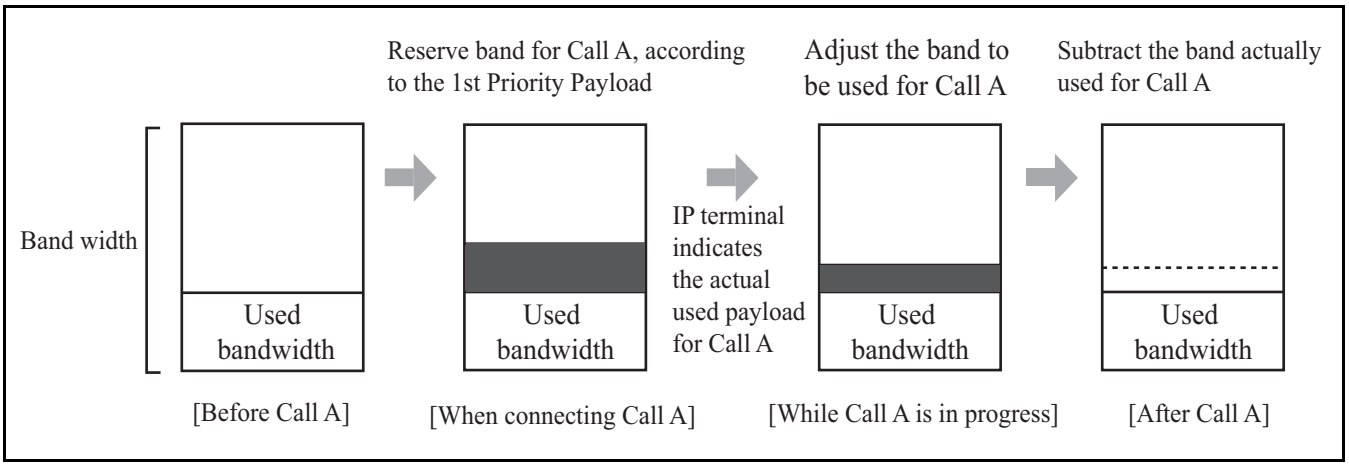
1. This feature is applied for DtermIP, Analog MC, Soft Phone, IPPAD, MG, etc.
2. When Bandwidth Control is applied for IPPAD, be sure to use all the IPPAD cards with voice compression function in the system.

   Note: In FCCS via IP and CCIS via IP network, mount IPPAD card corresponding to voice compression function in every node.

3. This feature is only applied to the IP terminals which belong to the Location.
4. This feature is applied to static routing network only. If a route other than that has been assigned by office data is used, accurate bandwidth control cannot be made. For this reason, bandwidth control cannot be applied to dynamic routing network, where routing data is dynamically changed.
5. This feature is applied to IP terminals accommodated in an MGC. It is not applied to IP terminals accommodated in different MGCs, where two terminals are directly connected via LAN (PHC).
6. This feature is not applied between IP terminals that belong to the same Location ID.
7. Bandwidth Control by this feature is not available when:
   - A call is answered (available only when a call is originated or terminated)
   - The originating terminal is assigned with Preclude From Bandwidth Restriction (SFI = 199 in ASFC command)
   - A call is established without originating. (For example, Blind Transfer to Station [B-17], Blind Transfer to Attendant [B-18], Call Transfer - All Calls [C-11], Three-Way Calling [T-2] and when getting back from Call Hold)
8. In the following cases, the bandwidth may exceed the threshold value.
   - When answering a call after the bandwidth is used up to the threshold value while ringing
   - When the bandwidth is used up to the threshold value and the bandwidth in active use became larger than the reserved value in consequence of negotiation
   - If the threshold value shifts to a smaller value than the actual bandwidth value in a certain time zone assigned for Particular bandwidth
   - When the threshold value assigned is smaller than the actual bandwidth value by office data

- When connected without restriction while the bandwidth is used up to the threshold value

9. The number of hops (from one Location ID to another) is up to 16 steps.
10. The number of adjacent locations is up to 1024 patterns.
11. When the bandwidth exceeds the threshold value, call origination and termination are restricted.
12. Assign a Service Feature Restriction Class that allows SFI = 199 to the Stations that are granted the ability to originate/terminate the call even when the bandwidth restriction is applied.
13. If bandwidth is not enough, a call origination is restricted.
14. Checking for available bandwidth is performed only for My Line. The Sub Line is not checked. The call can reach the destination even if the bandwidth of the Sub Line is not enough.
15. If there is shortage of the bandwidth when connecting external Music-On-Hold, the built-in Music-On-Hold of the terminal is used.
16. This feature is not applied to bandwidth used by IP Trunk (Not restricted).
17. The following explains how to control the bandwidth to be used:

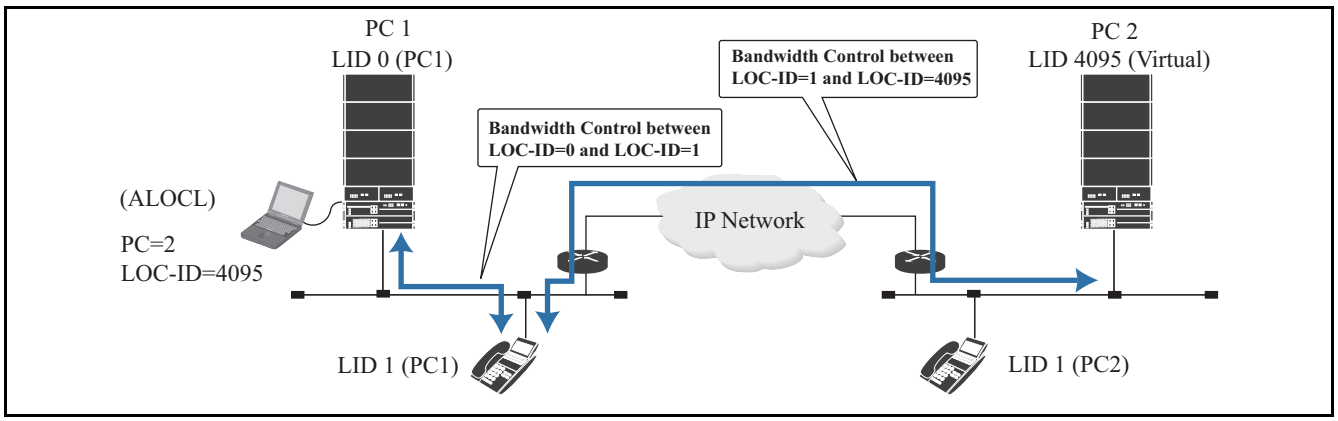


18. The following values are used for the add-subtract method of bandwidth:
[Amount of bandwidth to be used for a call per compression method, packer transmission cycle (full duplex communication: Ether, IP, UDP RTP header)]

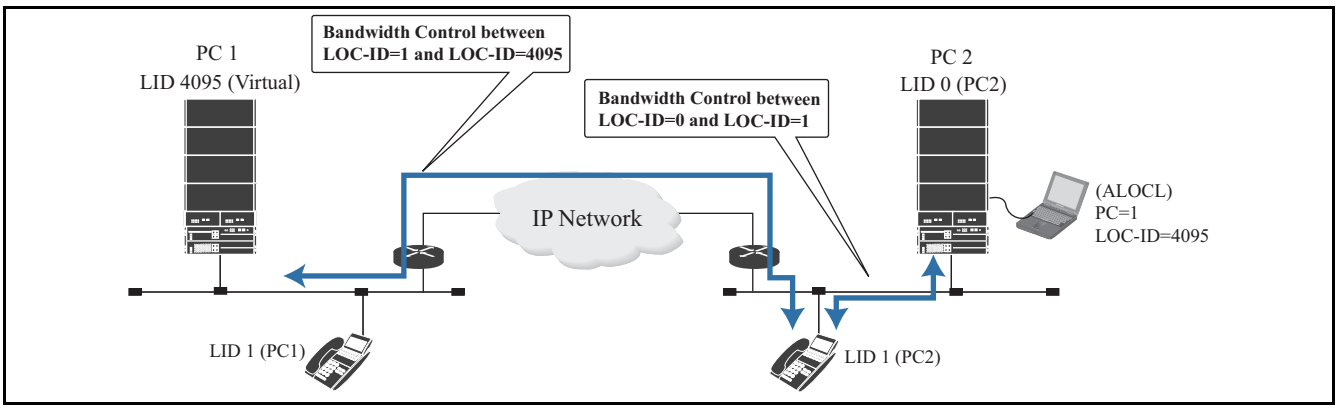
Compressing Method (Payload Type)	Packet Transmission Cycle				
	10 ms	20 ms	30 ms	40 ms	60 ms
G.711 (64Kbps)	112.0 (Kbps)	88.0 (Kbps)	80.0 (Kbps)	76.0 (Kbps)	-
G.729a (8Kbps)	56.0 (Kbps)	32.0 (Kbps)	24.0 (Kbps)	20.0 (Kbps)	-
G.723.1 (5.3Kbps→6.3Kbps)	-	-	21.3 (Kbps)	-	13.3 (Kbps)
G.723.1 (6.3Kbps→5.3Kbps)	-	-	22.3 (Kbps)	-	14.3 (Kbps)
G.711 (FAX)	112.0 (Kbps)	88.0 (Kbps)	80.0 (Kbps)	76.0 (Kbps)	-
G.726 (FAX)	80.0 (Kbps)	56.0 (Kbps)	48.0 (Kbps)	44.0 (Kbps)	-
G.722	-	88.0 (Kbps)	-	76.0 (Kbps)	-
AMR-NB	-	37.2 (Kbps)	-	25.0 (Kbps)	-
AMR-WB	-	48.8 (Kbps)	-	36.6 (Kbps)	-
G.711 μ law (FEC)	-	163.2 (Kbps)	-	145.6 (Kbps)	-
G.711 A law (FEC)	-	163.2 (Kbps)	-	145.6 (Kbps)	-

G.729a (FEC)	-	51.2 (Kbps)	-	33.6 (Kbps)	-
G.722 (FEC)	-	163.2 (Kbps)	-	145.6 (Kbps)	-
AMR-NB (FEC)	-	61.6 (Kbps)	-	43.6 (Kbps)	-
AMR-WB (FEC)	-	84.8 (Kbps)	-	66.8 (Kbps)	-
T.30 (FAX)	13.6 (Kbps) (Packet Transmission Cycle is not specified.)				

19. Some types of PS save the incoming call history data even when the call termination to the PS is restricted due to the shortage of bandwidth.
20. By allocating Location ID to the destination PC (Point Code), the system can control bandwidth for the call established via PHC. For details, refer to the figure below.
    a.) When allocating LID 4095 to PC 2 (destination PC) in the PC 1 Office data, PC 1 Office can manage the bandwidth in the following area.



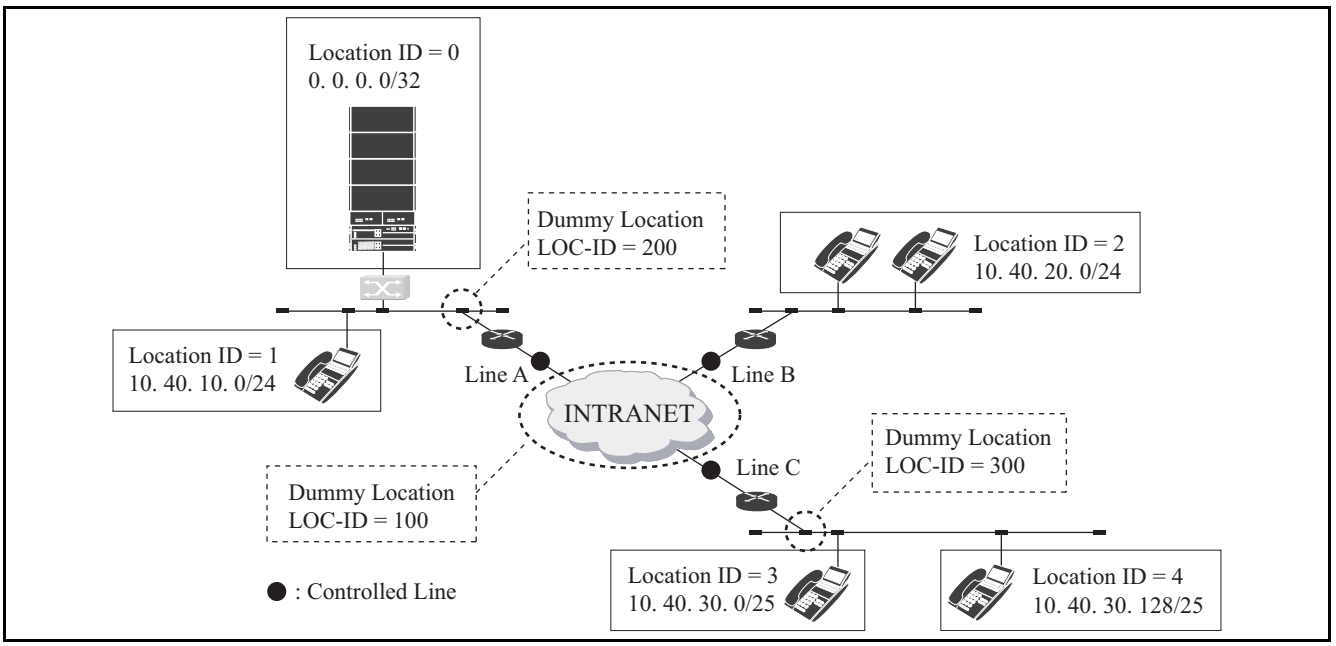
    b.) When allocating LOC-ID=4095 to PC 1 (destination PC) in the PC 2 Office data, PC 2 Office can manage the bandwidth in the following area.



- PC 2 office manages the bandwidth between LID 0 (PC2) and LID 1 (PC2), LID 1 (PC2) and LID 4095 (PC1).
  If there is not enough bandwidth between these LIDs, PC 2 office sends the “Busy” signal to PC 1 office.
- PC 1 office manages the bandwidth between LID 0 (PC1) and LID 1 (PC1), LID 1 (PC1) and LID 4095 (PC2).
  If there is not enough bandwidth between these LIDs, the call origination is restricted.

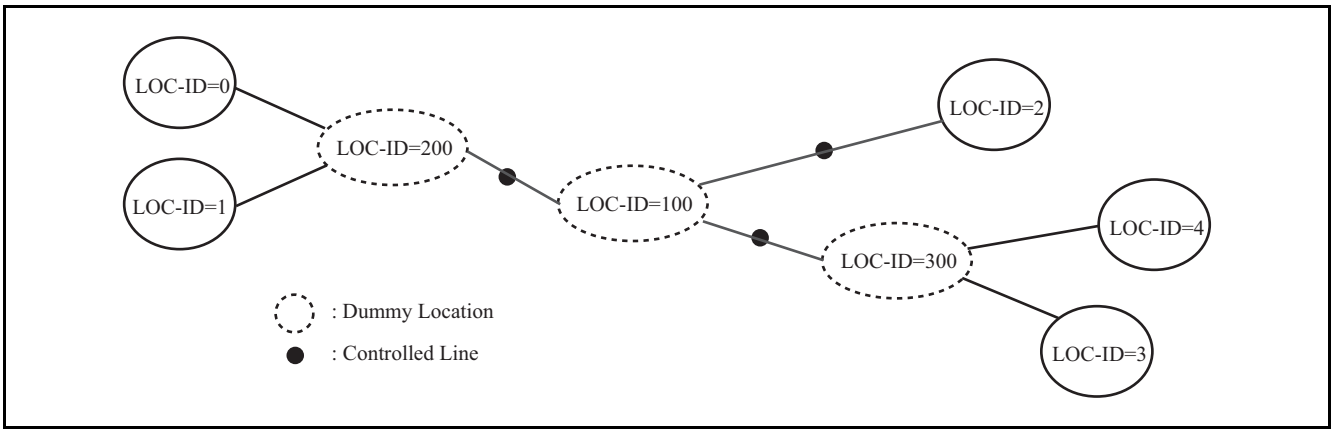
If a call is originated to the terminal belonging to LID 0 in PC 2 from the terminal belonging to LID 0 in PC1.

21. (**Dummy Location**) When network routes branched off in multiple directions, it is available to combine these routes into one by assigning Dummy Location as a branch point. Also, when multiple Locations exist in a single network, these Locations can be combined into one by allocating a Dummy Location between the network and another one.



- LOC-ID = 0 and LOC-ID 1 can be combined into one as LOC-ID = 200 (Dummy Location) because they exist in the same network.
- LOC-ID = 3 and LOC-ID 4 also can be combined into one as LOC-ID = 300 (Dummy Location) because they exist in the same network.
- LOC-ID = 100 (Dummy Location) can be allocated to a branch point between LOC-ID = 200, LOC-ID = 300 and LOC-ID = 2.

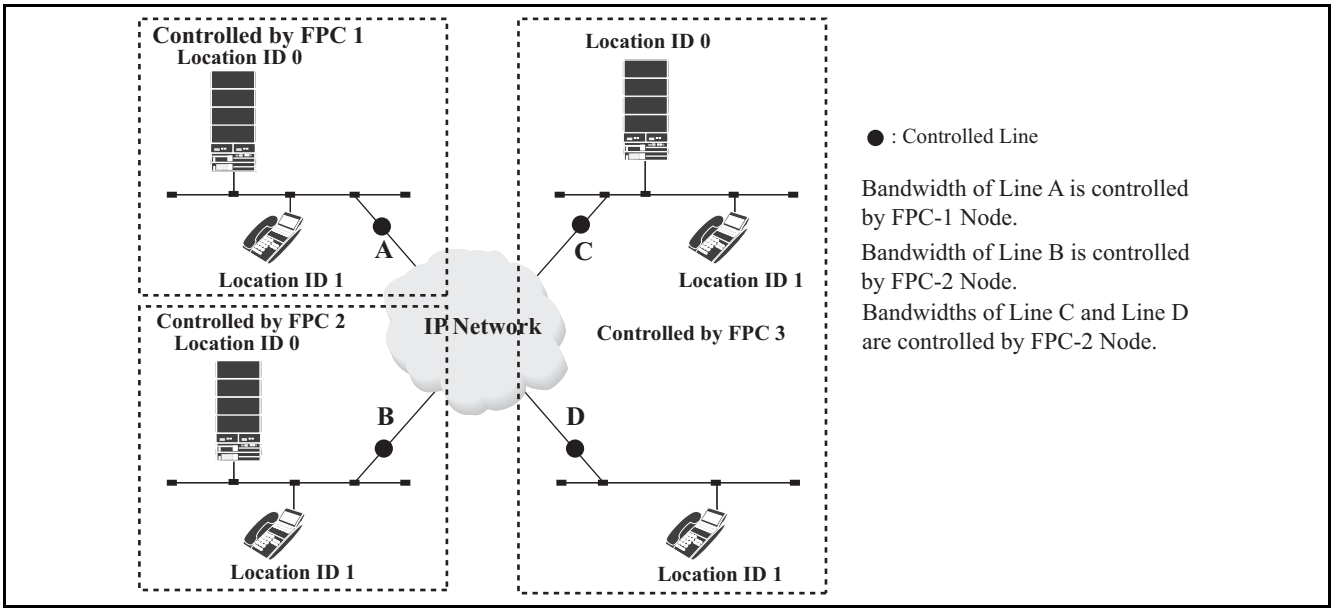
Therefore, the configuration of Location IDs is as follows:



Note: For the example data programming of Dummy Location, refer to the Programming section.

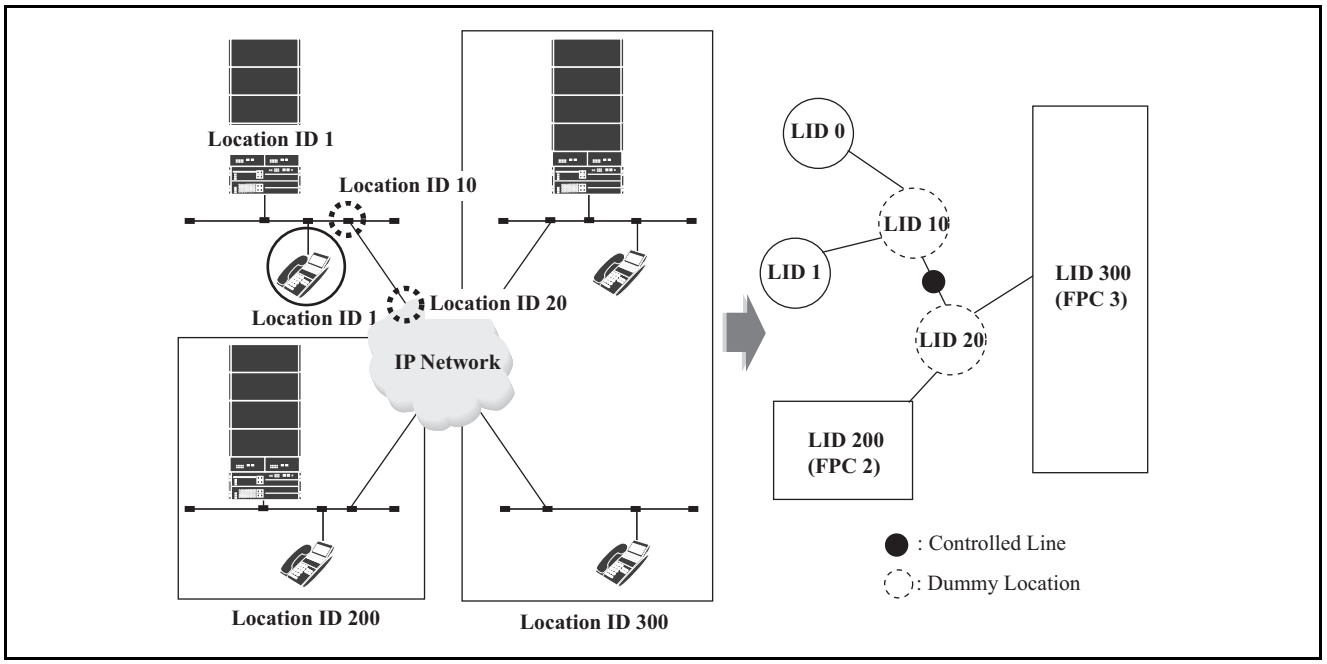
22. A single Location ID can be allocated to multiple network addresses. In the following cases, the data programming (Location configuration) can be simplified.
    a.) When using different payloads in a single network:
    Example:
    172.16.253.1 /124: Location ID 1 [Saving bandwidth → Assigns G.729a, 40ms (20 Kbps)]
    172.16.253.128/197: Location ID 2 [Improving sound quality → Assigns G.711, 40ms (80 Kbps)]
    172.16.253.198 /254: Location ID 1 [Saving bandwidth → Assigns G.729a, 40ms (20 Kbps)]

23. Bandwidth control in a FCCS network is available by allocating Location ID to Fusion Point Code (FPC). The following figure shows the case where controlling bandwidths of the lines between the IP network and each node in a star-topology network.



Note: By allocating a Location ID to each FPC (Fusion Point Code), each node can control the bandwidth individually.

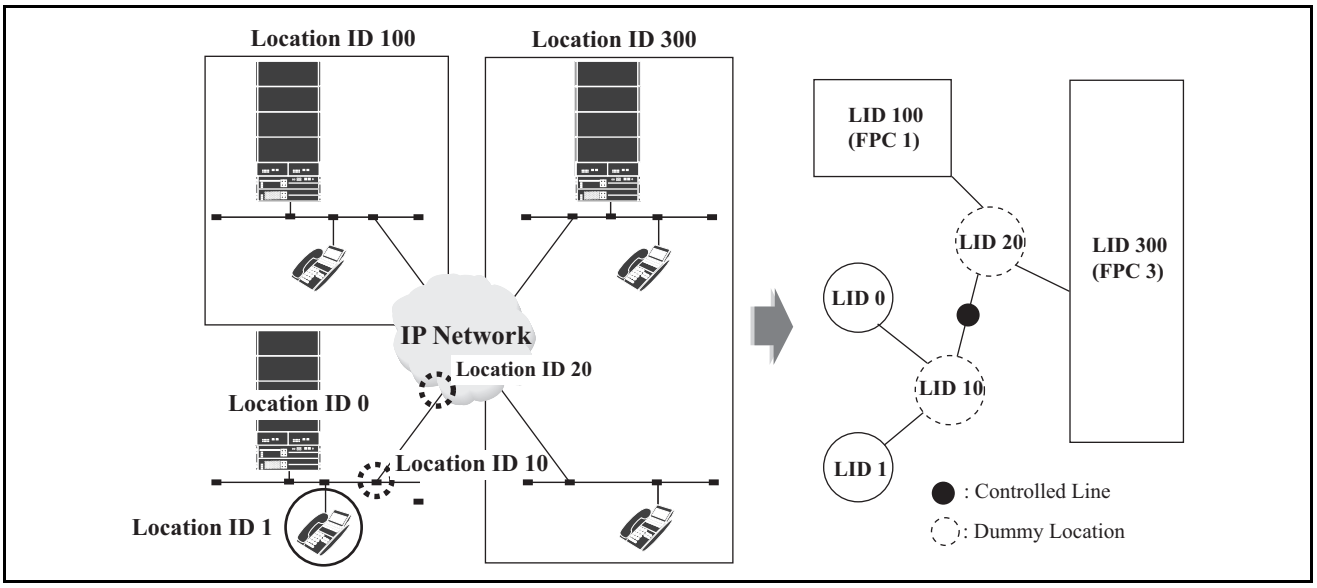
<Location IDs assigned in FPC 1>



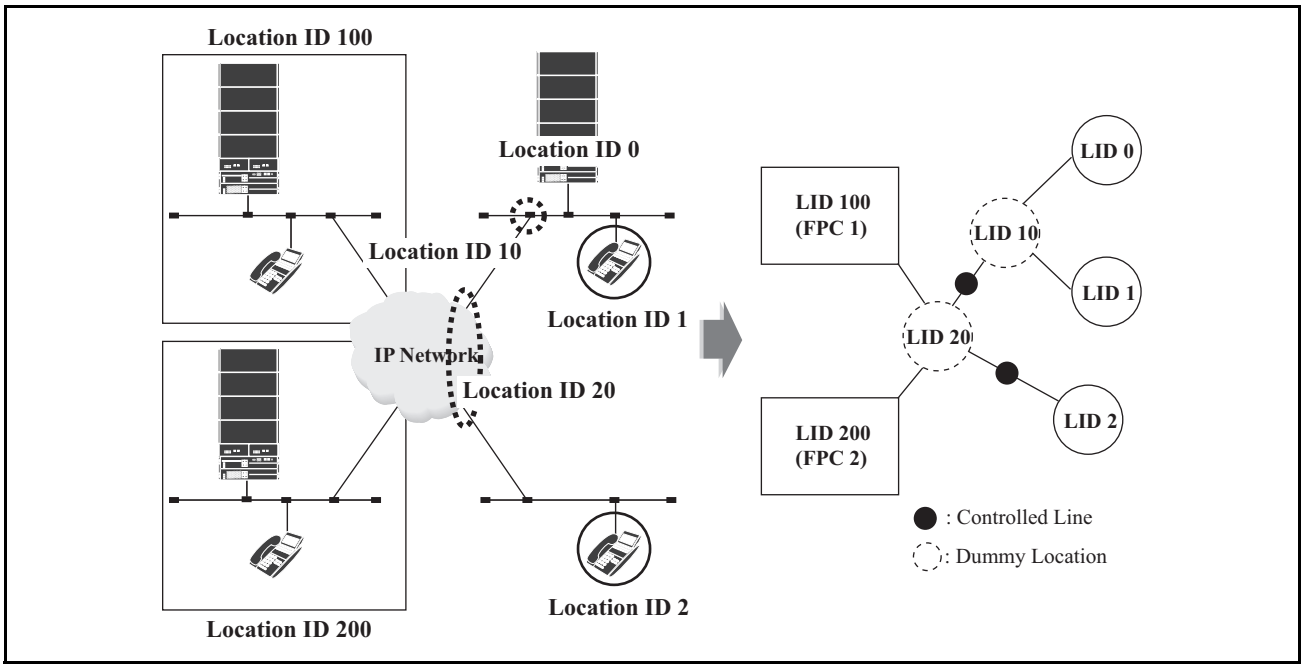
- Location ID 200 is assigned to FPC 2.
- Location 300 is assigned to FPC 3.
- Dummy Locations (Location IDs 10 and 20) are assigned as branch points.

By configuring the locations and routing tables as shown in the above figure, all calls from FPC 1 to FPC 2 and 3 are pass through the line between Dummy Locations 10 and 20. Thus, the threshold value can be assigned to the line between Nodes A and B to control the bandwidth of the line.

<Location IDs assigned in FPC 2>



<**Location IDs assigned in FPC 3**>



Bandwidth Check at the time of call origination is performed by the following procedure.

Example: In the case where Terminal A in LID 1 of FPC 2 originates a call to Terminal B in LID 3 of FPC 3.

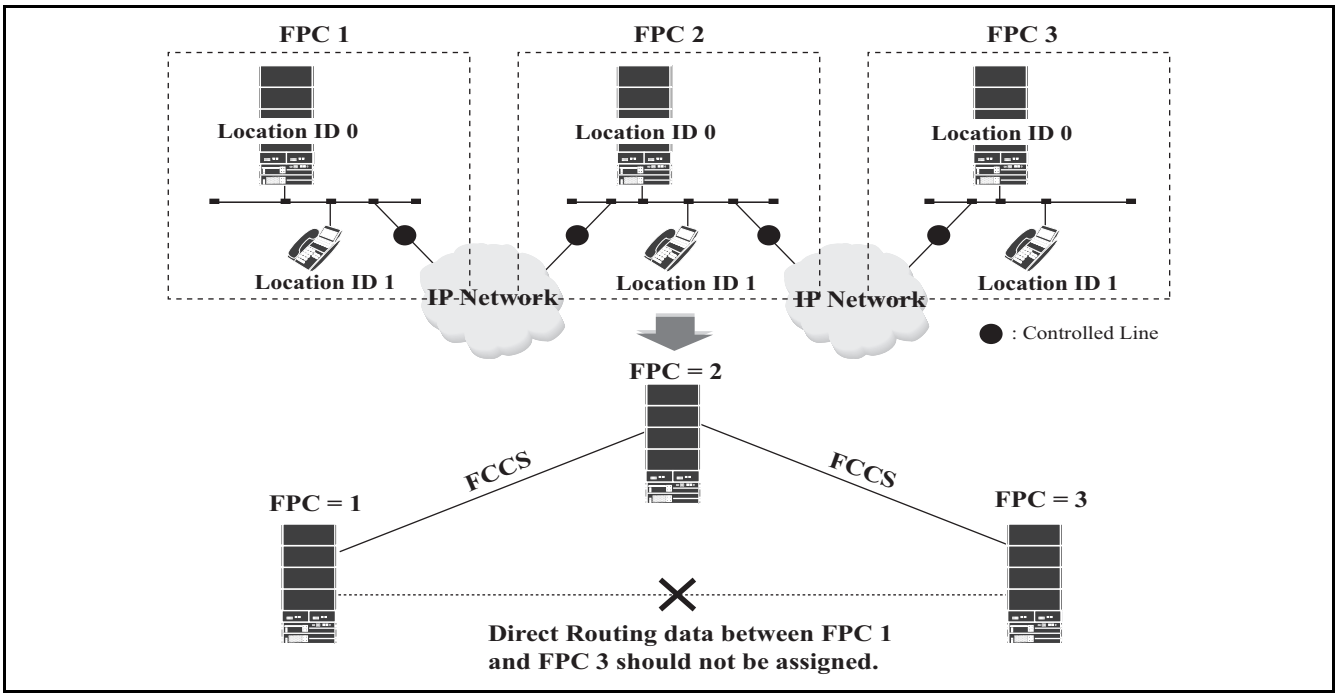
- FPC 2 (Originating Node) retrieves the route from LID 1 to LID 300.
  result: LID 1 → LID 10 → LID 20 → LID 30
- FPC 2 checks the bandwidth of each line between LID 1 and LID 10, LID 10 and LID 20, LID 20 and LID 30.
  If adequate bandwidth is not available, the call origination is restricted.
  If adequate bandwidth is available, FPC 2 sends call setup message to FPC 3.
- FPC 3 (Terminating Node) retrieves the route from LID 200 (FPC 2) to LID 2.
  result: LID 200 → LID 20 → LID 2
- FPC 3 checks the bandwidth of each line between LID 200 and LID 20, LID 20 and LID 2.
  If adequate bandwidth is not available, the call origination is restricted.
  If adequate bandwidth is available, the call origination begins.

- **Conditions**

When using this feature via FCCS network, the following conditions must be considered:

a.) The alternate routing due to bandwidth shortage is counted as one of the 8 allowed alternate routing.
b.) When multiple Locations use the same line, the data programming of the bandwidth control can be simplified by using the Dummy Location.
c.) As shown in the following figure, if multiple IP networks exist between FPC 1 and FPC 3, the connection route data between FPC 1 and FPC 3 should be assigned as Tandem Connection.

Example 1: Network configuration to be assigned as Tandem Connection



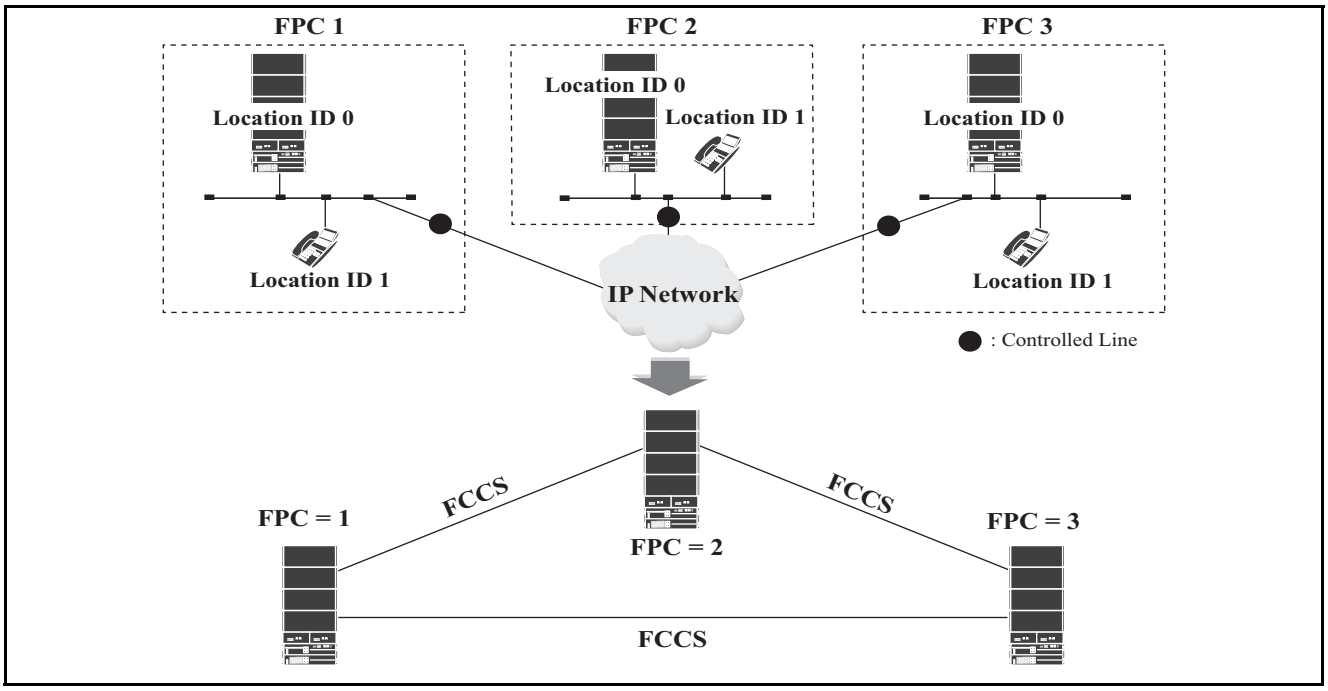
The following bandwidth control is performed for a call between Terminal A in LID 1 of FPC 1 and Terminal B in LID 1 of FPC 3.

- FPC 1: Bandwidth control of Line A.
- FPC 2: Bandwidth control of Lines B and C.
- FPC 3: Bandwidth control of Line D.

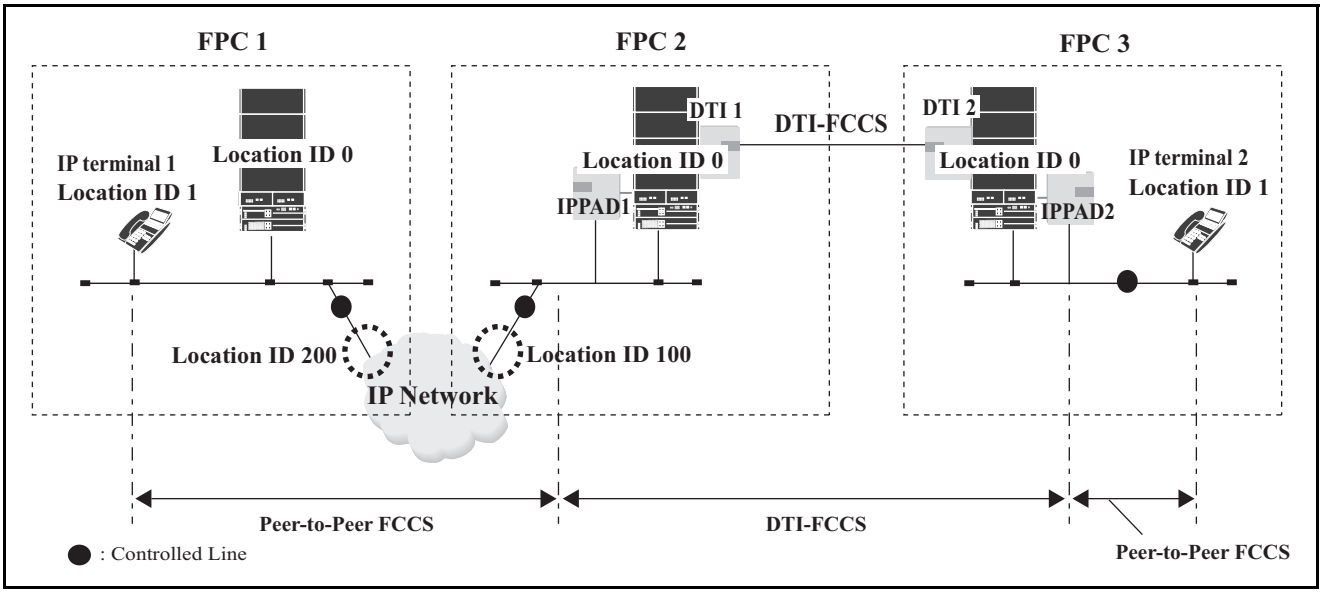
In the above network configuration, even if the direct FCCS routing data between FPC 1 and FPC 3 is assigned, the bandwidth control is not available for Lines B and C controlled by FPC 2. To control the bandwidth of Lines B and C, the connection data between FPC 1 and FPC 3 must be assigned as Tandem Connection (A connection between FPC 1 and FPC 3 is established via FPC 2).

On the other hand, as shown in the following figure, if FPC 1 and FPC 3 are connected via a single IP network, the connection route data between FPC 1 and FPC 3 should not be assigned as Tandem Connection.

Example 2: Network configuration not to be assigned as Tandem Connection



d.) Even if DTI-FCCS and Peer-to-Peer FCCS coexist in a network, no particular consideration is required. The following is an example of the network configuration including both DTI-FCCS and Peer-to-Peer FCCS.



FPC	Switch Control	Connection Method	Bandwidth Control
1	IP terminal 1- IPPAD 1	Peer-to-Peer connection	Location ID 1 - 200

FPC	Switch Control	Connection Method	Bandwidth Control
2	IP terminal 1- IPPAD 1	Peer-to-Peer connection	Location ID 100 - 0
	IPPAD 1 - DTI 1	DTI connection	None
3	IPPAD 1 - DTI 1	DTI connection	None
	IPPAD 2 - IP terminal 2	Peer-to-Peer connection	Location ID 1 - 1

<Conditions for Virtual Line Bandwidth Control>

1. This feature is only available for Desktop terminals and Virtual Lines (ASDT/AISTL TEC = 12/18).
2. The conditions when this feature is enabled for the Station Number of Virtual Line or Sub Line are listed below.
   a.) The bandwidth between the Location ID assigned for the Called Station Number and the Location ID assigned for the Calling Party is controlled when a call is terminated to Virtual Line or Sub Line. The Location ID of the Calling Party is assigned to the following.
   - The Calling Station which originated a call to the Virtual Line or Sub Line
   - Incoming Trunk **Note 1**
   - PC (Point Code) of the connected party when the incoming call is established via PHC
   - FPC (Fusion Point Code) of the connected party when the incoming call is established via Peer-to-Peer FCCS

   b.) The Location ID assigned for the Desktop terminal which operated a call origination is used for Virtual Line Bandwidth Control when a call is originated from Virtual Line or Sub Line. However, when a call is originated via PHC or via Peer-to-Peer FCCS, the Location ID assigned for each Node is used as is conventionally done.

   c.) For the Bandwidth Control when an incoming call to a Virtual Line or Sub Line is answered, the Location ID of the Desktop terminal which answered the incoming call is used.

   Note 1: When using a conventional trunk, Location ID = 0 (for the System) is assigned. When using MG, the Location ID assigned for MG is used.

3. When this feature is enabled for My Line, the Location ID assigned for the Station Number takes the priority when conflicted with the Location ID assignment for Network Address.

Programming

STEP 1: **ALOCL** - Assign the Location ID data.
TYPE: Select either "Network Address", "PC", "FPC" or "STN".

Note: When using this feature via Peer-to-Peer FCCS, allocate Location ID to all Node FPCs (except Self Node FPC) in the network.

- When selecting "Network Address":
  IP ADDRESS: Assign the IP Address. For the Telephony Server (i.e. IPPAD), assign "0.0.0.0."
  MASK BIT: Assign the MASK BIT in the range 1-31 except when IP ADDRESS 0.0.0.0 is determined. When IP ADDRESS = 0.0.0.0, MASK BIT = 32 is automatically assigned.
  LOC-ID: Assign any number in the range 0-4095, associated with the IP address assigned in the IP ADDRESS parameter. When 0.0.0.0 is assigned in the IP ADDRESS parameter, LOC-ID is automatically set to "0" (the value cannot be changed).
- When selecting "PC":
  PC: Assign the Destination Point Code.
  LOC-ID: Assign any number in the range 1-4095.
- When selecting "FPC":
  FPC: Assign Fusion Point Code (1-253).
  LOC-ID: Assign any number in the range 1-4095.
- When selecting "STN":
  TN: Assign Tenant Number (1-63).
  STN: Assign Station Number (maximum of five digits for Business)
  LOC-ID: Assign any number in the range 1-4095.

Note: LOC-ID = 0 is predetermined to the location ID of the Telephony Server.

STEP 2: **AIVCL/AIVCN** - Assign the IP service Voice Control data. **When using this feature via Peer-to-Peer FCCS, assign AIVCN data to the NCN. AIVCN data has precedence over AIVCL data.**
for AIVCL:
A LOC-ID: Location ID of the self-Location. (0-4095)
B LOC-ID: Location ID of the target Location. (0-4095)
for AIVCN:
A FPC: Fusion Point Code of the self-Location. (1-253)
A LOC-ID: Location ID of the self-Location. (0-4095)
B FPC: Fusion Point Code of the target Location. (1-253)
B: LOC-ID: Location ID of the target Location. (0-4095)

Note: Only the Location ID corresponding to the network address can be assigned on the AIVCL/AIVCN. The Location IDs corresponding to PC or FPC cannot be assigned.

[Setting of Jitter Buffer]

Select the "Setting of Jitter Buffer" check box to assign the following Jitter Buffer data.

Note: If the check box is not checked, the default settings (JB MIN = 1 (10ms), JB MAX = 30 (300ms)) will be provided.

JB MIN: Jitter Buffer minimum value

JB MAX: Jitter Buffer maximum value

[Setting of Payload]

Select the "Setting of Payload" check box to assign the following payload data.

PRI1-PRI4: Select a payload type (TYPE) and payload size (SIZE) in order of precedence (PRI1(high) → PRI4 (low))

TYPE (Payload Type):

- G.711
- G.729a
- G.723.1 (5.3K → 6.3K)
- G.723.1 (6.3K → 5.3K)
- G.722 **Note 2**, **Note 3**
- AMR-NB **Note 2**
- AMR-WB **Note 2**
- G.711 μ law (FEC) **Note 2**, **Note 4**
- G.711 A law (FEC) **Note 2**, **Note 4**
- G.729a (FEC) **Note 2**, **Note 4**
- G.722 (FEC) **Note 2**, **Note 4**
- AMR-NB (FEC) **Note 2**, **Note 4**
- AMR-WB (FEC) **Note 2**, **Note 4**

Note 2: These payload types are available for DtermSP30 Version 15. These data are effective only for the connection between DtermSP30s Version 15.

Note 3: Payload type G.722 is effective for the following connections in addition to the connection between DtermSP30s Version 15:

- The connection between DT700 Series (DT730/DT730G/DT750)
- The connection between DtermSP30 Version 15 and DT700 Series (DT730/DT730G/DT750)

Note 4: When a payload type with "FEC" is selected, the data that is not transmitted correctly because of packet loss or bit error can be corrected according to the redundant data by the method of Forward Error Correction (FEC).

SIZE (Payload Size):

- 10ms
- 20ms
- 30ms
- 40ms
- 60ms

Note: For available payload type/payload size, refer to the Payload Setting Available for each IP Equipment section in Chapter 3 of System Description.

Note: When no payload type and size are specified, the values assigned via System data (ASYDL, System Data 1, Index 819, Bits 0 - 3 and 4 - 7) are automatically applied.

Note: If payload type G.711 is specified to more than two priorities in PRI 1 through PRI 4, assign different payload size to each priority.
- When payload types G.723.1 (5.3K → 6.3K) and G.723.1 (6.3K → 5.3K) are to be assigned on the same terminal, specify the same value to the payload size of both types.
- It is not necessary to assign the payload type/size to all priorities. Specifying the first priority only is allowed.

The following are required to activate Bandwidth Control feature.
The following data setting are required when allocating Location ID to the destination PC to control the bandwidth for the call via PHC.

STEP 3: **ASFC** - Assign the following SFI to grant an exception to the bandwidth restriction.
SFI = 199 (Bandwidth Restriction)
RES = 1 (Restriction is not applied)

STEP 4: **ABNDL** - Assign bandwidth control data such as routing table, bandwidth threshold value.

a.) Enter the Location ID.
START LOC-ID: Enter LOC-ID of Calling Party (0-4095)
END LOC-ID: Enter LOC-ID of Called Party (0-4095)

<Routing table> Field

b.) Press the Get button to display the routing table between the specified locations.
c.) Enter the data in the routing table.
1st - 15th LOC-ID: Enter LOC-ID which relays START LOC-ID and END LOC-ID.
When START LOC and END LOC are adjacent to each other, leave the routing table blank.
d.) The routing table is set by pressing "Set."
e.) By pressing "Disp", Bandwidth border data on a location-location basis can be displayed in Bandwidth border data field (on the right of the window).
Data: Display the existence information of bandwidth border data on a location-location basis.
O/X (Bandwidth border data is existing/not existing)

<Bandwidth border data> Field

f.) Assign the bandwidth border values.
LOC-ID: Indicates locations whose bandwidth border values are displayed below.

<Normal bandwidth data> -This value is applied for 24 hours -
Assign the following data related to Normal Bandwidth
Limit value (Kbps): Assign maximum value of Normal bandwidth. (1-16777215)
Warning value (Kbps): Assign warning value of Normal bandwidth. (1-16777215) At this value, system message "36-G/36-H" is issued as a warning.
Alarm: System message "36-I/36-J" output 0/1 = Provided/not provided

<Particular bandwidth data> -*This value is applied only for specified time zone. A maximum of 10* time zone patterns can be assigned -
Assign the following data related to Particular Bandwidth
Time zone: Time zone to which the specified bandwidth value is applied Assign the value 00:00~23:59 format.

Note: Do not assign duplicate time zone.

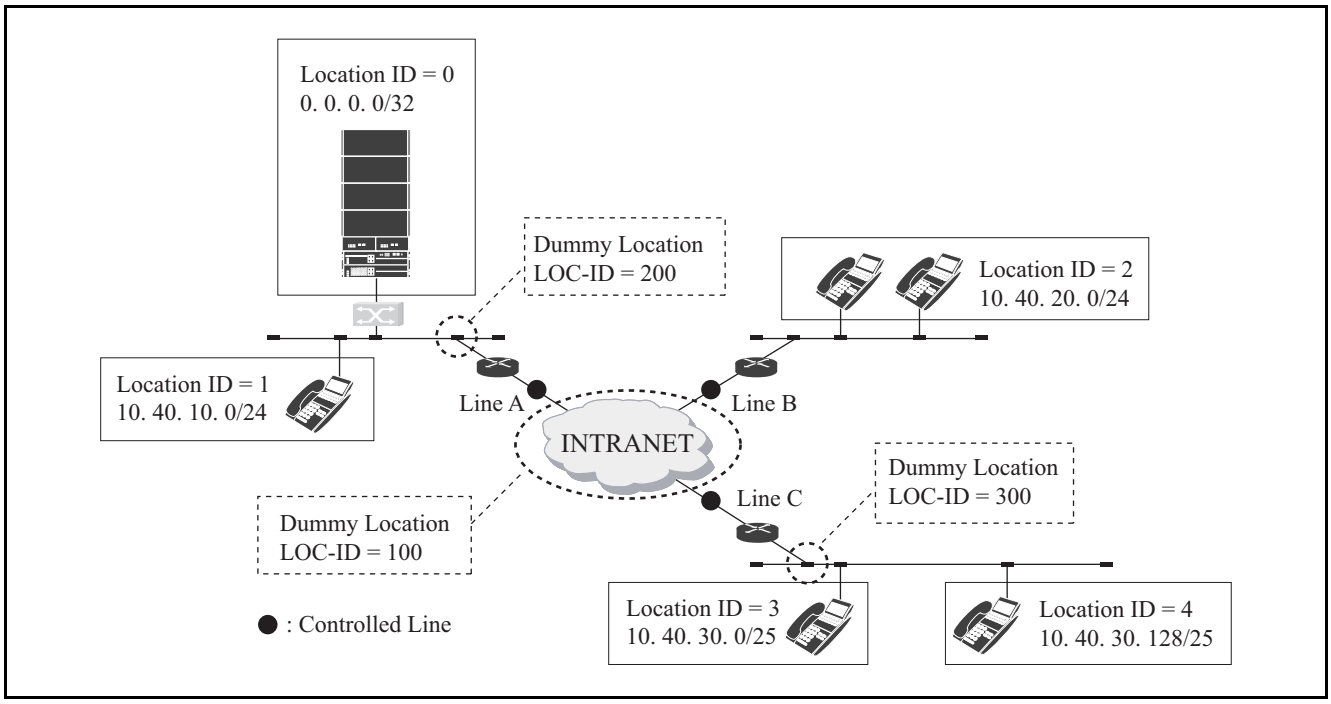
Note: Do not assign time zone across 00:00. (Ex. 23:00~01:00 is not allowed.)

Limit value (Kbps): Assign maximum value of Particular bandwidth. (1-16777215)
Warning value (Kbps): Assign warning value of Particular bandwidth. (1-16777215) At this value, system message "36-G/36-H" is issued as a warning.
Alarm: System message "36-I/36-J" output 0/1 = Provided/not provided

g.) By pressing the Set button, the values are set.

h.) By pressing the Disp button between locations whose bandwidth border data has not been assigned, assign the bandwidth border data one by one. If the bandwidth border data is not required, this procedure is not necessary.

<Example Data Programming 1: Star-topology Network (Stand-alone)>



1. ALOCL

	IP Address	LOC-ID
1	0. 0. 0. 0/32	0
2	10. 40. 10. 0/24	1
3	10. 40. 20. 0/24	2
4	10. 40. 30. 0/25	3
5	10. 40. 30. 128/25	4

2. AIVCL

	LOC-ID A	LOC-ID B	PAYLOAD TYPE/SIZE
1	0	1	G. 711, 40ms
2	0	2	G. 729a, 20ms
3	0	3	G. 729a, 20ms
4	0	4	G. 729a, 20ms
5	1	2	G. 729a, 20ms
6	1	3	G. 729a, 20ms
7	1	4	G. 729a, 20ms
8	2	3	G. 729a, 20ms
9	2	4	G. 729a, 20ms
10	3	4	G. 711, 40ms

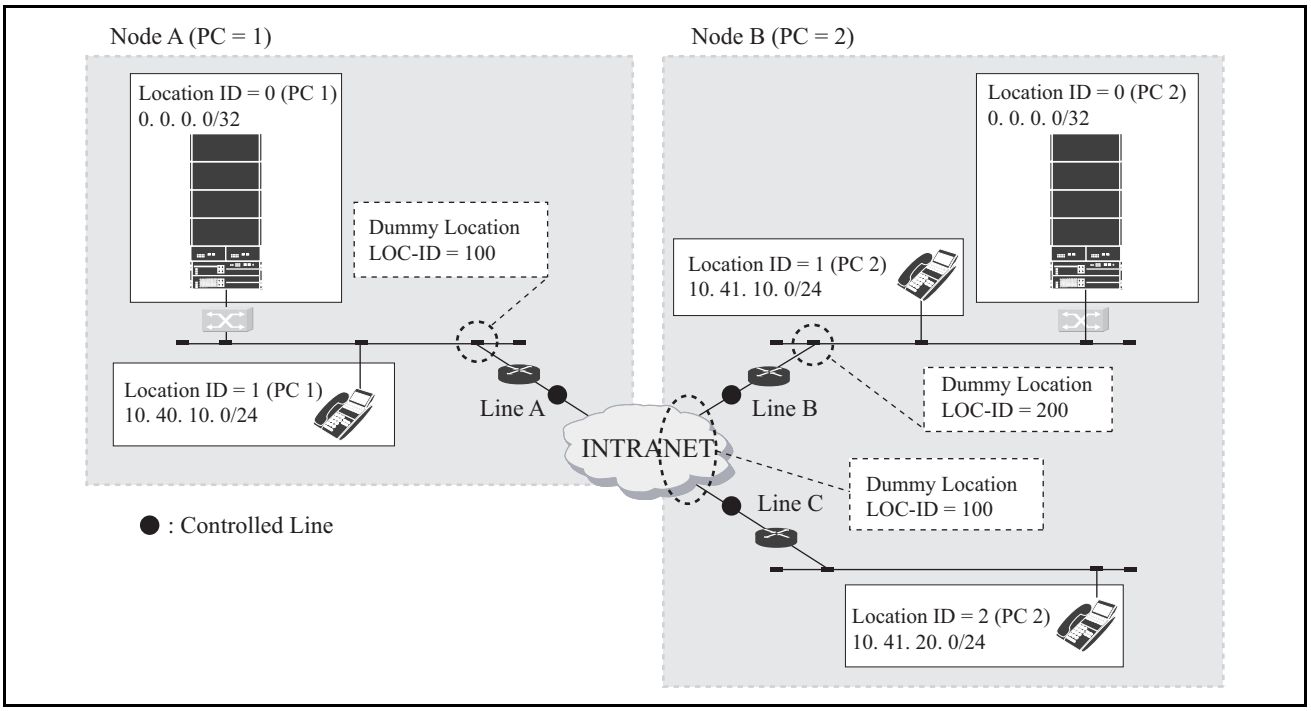
3. ABNDL (Routing Table)

	START LOC-ID	1st - 15th LOC-ID	END LOC-ID
1	0	200 - 100	2
2	0	200 - 100 - 300	3
3	0	200 - 100 - 300	4
4	1	200 - 100	2
5	1	200 - 100 - 300	3
6	1	200 - 100 - 300	4
7	2	100 - 300	3
8	2	100 - 300	4

Bandwidth Threshold Value

	LOC-ID A	LOC-ID B	Threshold Value	Remarks
1	100	200	1520 Kbps	Line A
2	2	100	760 Kbps	Line B
3	100	300	1140 Kbps	Line C

<Example Data Programming 2: Star-topology Network (Peer-to-Peer CCIS)>



Data Programming in Node A (PC=1)

1. ALOCL

	IP Address	LOC-ID
1	0. 0. 0. 0/32	0
2	10. 40. 10. 0/24	1
3	10. 41. 10. 0/24	11
4	10. 41. 20. 0/24	12
	PC (Point Code)	LOC-ID
5	2	20

2. AIVCL

	LOC-ID A	LOC-ID B	PAYLOAD TYPE/SIZE
1	0	1	G. 711, 40ms
2	0	11	G. 729a, 20ms
3	0	12	G. 729a, 20ms
4	1	11	G. 729a, 20ms
5	1	12	G. 729a, 20ms

3. ABNDL (Routing Table)

	START LOC-ID	1st - 15th LOC-ID	END LOC-ID
1	0	100	20
2	1	100	30

Bandwidth Threshold Value

	LOC-ID A	LOC-ID B	Threshold Value	Remarks
1	20	100	1520 Kbps	Line A

Data Programming in Node B (PC=2)

1. ALOCL

	IP Address	LOC-ID
1	0. 0. 0. 0/32	0
2	10. 41. 10. 0/24	1
3	10. 41. 20. 0/24	2
4	10. 40. 10. 0/24	11
	PC (Point Code)	LOC-ID
5	1	10

2. AIVCL

	LOC-ID A	LOC-ID B	PAYLOAD TYPE/SIZE
1	0	1	G. 711, 40ms
2	0	2	G. 729a, 20ms
3	0	11	G. 729a, 20ms
4	1	2	G. 729a, 20ms
5	1	11	G. 729a, 20ms
6	2	11	G. 729a, 20ms

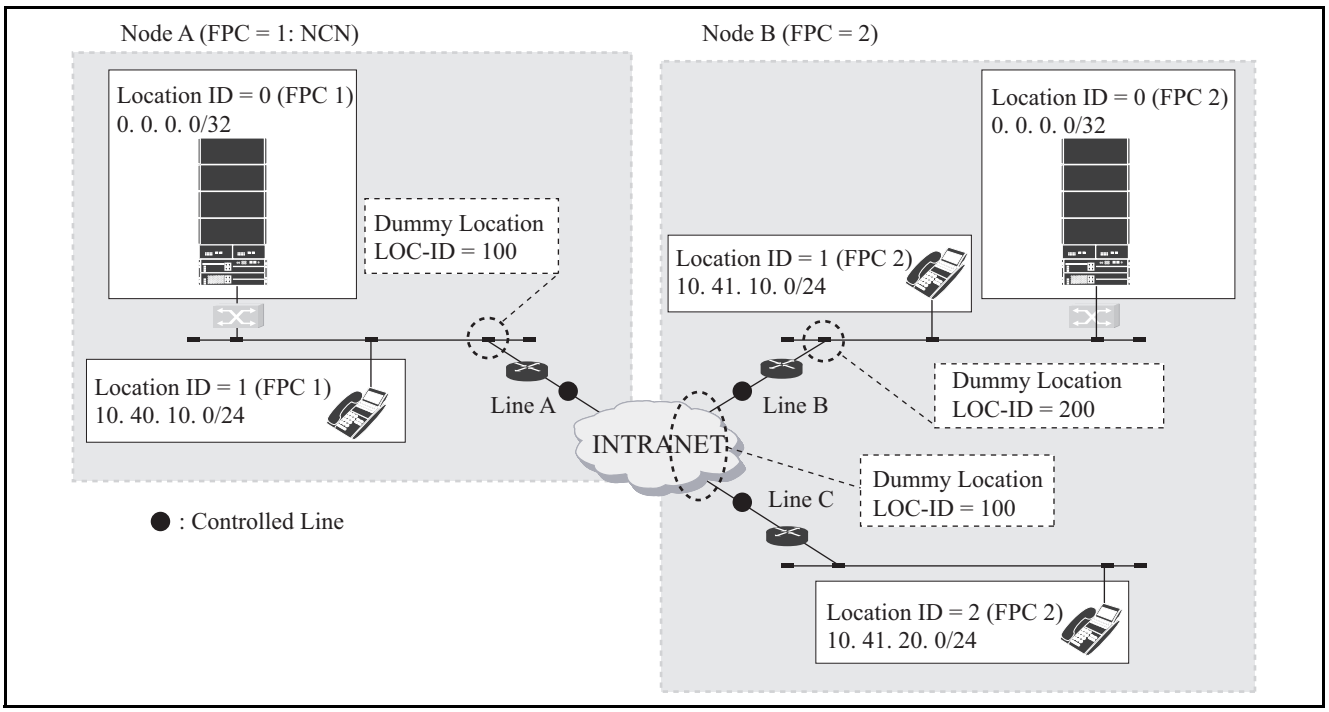
3. ABNDL (Routing Table)

	START LOC-ID	1st - 15th LOC-ID	END LOC-ID
1	0	200 - 100	2
2	0	200 - 100	10
3	1	200 - 100	2
4	1	200 - 100	10
5	2	100	10

Bandwidth Threshold Value

	LOC-ID A	LOC-ID B	Threshold Value	Remarks
1	100	200	1140 Kbps	Line B
2	2	100	760 Kbps	Line C

<Example Data Programming 3: Star-topology Network (Peer-to-Peer FCCS)>



Data Programming in Node A (FPC=1 NCN)

1. ALOCL

	IP Address	LOC-ID
1	0. 0. 0. 0/32	0
2	10. 40. 10. 0/24	1
	FPC (Fusion Point Code)	LOC-ID
3	2	20

2. AIVCN

	FPC A	LOC-ID A	FPC B	LOC-ID B	PAYLOAD TYPE/ SIZE
1	1	0	1	1	G. 711, 40ms
2	1	0	2	0	G. 729a, 20ms
3	1	0	2	1	G. 729a, 20ms
4	1	0	2	2	G. 729a, 20ms
5	1	1	2	0	G. 729a, 20ms
6	1	1	2	1	G. 729a, 20ms
7	1	1	2	2	G. 729a, 20ms
8	2	0	2	1	G. 711, 40ms
9	2	0	2	2	G. 729a, 20ms
10	2	1	2	2	G. 729a, 20ms

3. ABNDL (Routing Table)

	START LOC-ID	1st - 15th LOC-ID	END LOC-ID
1	0	100	20
2	1	100	20

Bandwidth Threshold Value

	LOC-ID A	LOC-ID B	Threshold Value	Remarks
1	20	100	1520 Kbps	Line A

Data Programming in Node B (FPC=2)

1. ALOCL

	IP Address	LOC-ID
1	0. 0. 0. 0/32	0
2	10. 41. 10. 0/24	1
3	10. 41. 20. 0/24	2
	FPC (Fusion Point Code)	LOC-ID
4	1	10

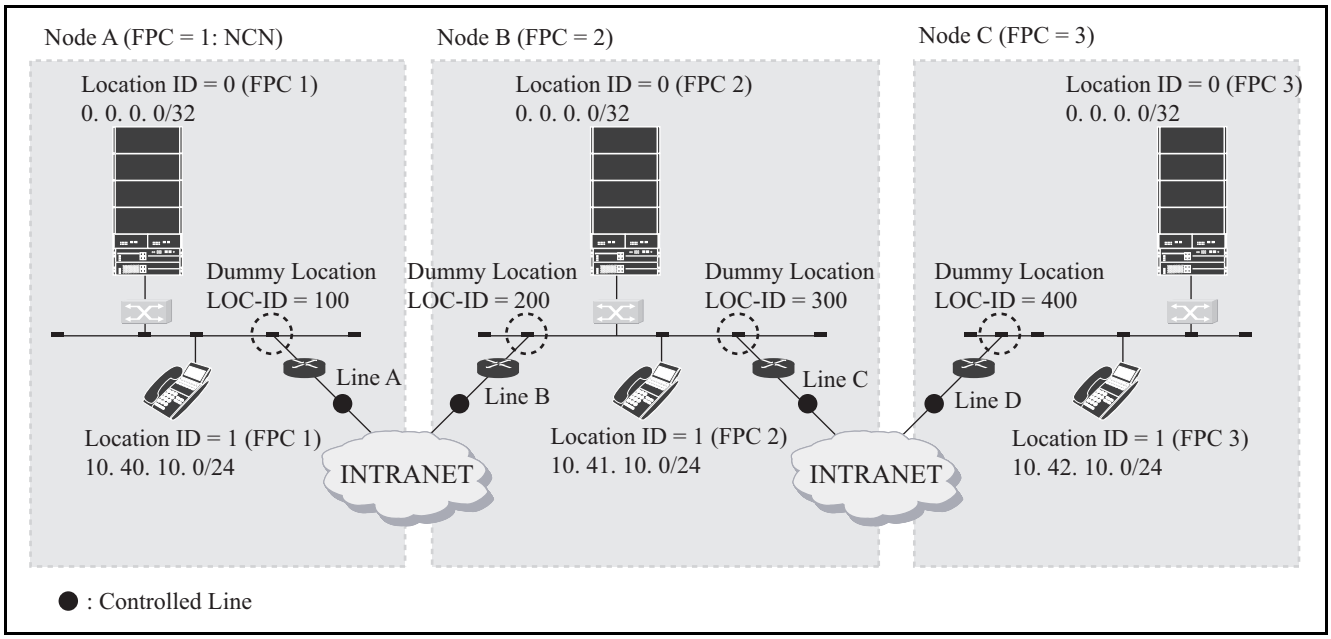
2. ABNDL (Routing Table)

	START LOC-ID	1st - 15th LOC-ID	END LOC-ID
1	0	200 - 100	2
2	0	200 - 100	10
3	1	200 - 100	2
4	1	200 - 100	10
5	2	100	10

Bandwidth Threshold Value

	LOC-ID A	LOC-ID B	Threshold Value	Remarks
1	100	200	1140 Kbps	Line B
2	2	100	760 Kbps	Line C

<Example Data Programming 4: Tandem Connection Network (Peer-to-Peer FCCS)>



Data Programming in Node A (FPC=1 NCN)

1. ALOCL

	IP Address	LOC-ID
1	0. 0. 0. 0/32	0
2	10. 40. 10. 0/24	1
	FPC (Fusion Point Code)	LOC-ID
3	2	20

2. AIVCN

	FPC A	LOC-ID A	FPC B	LOC-ID B	PAYLOAD TYPE/ SIZE
1	1	0	1	1	G. 711, 40ms
2	1	0	2	0	G. 729a, 20ms
3	1	0	2	1	G. 729a, 20ms
4	1	0	3	0	G. 729a, 20ms
5	1	0	3	1	G. 729a, 20ms
6	1	1	2	0	G. 729a, 20ms
7	1	1	2	1	G. 729a, 20ms
8	1	1	3	0	G. 729a, 20ms
9	1	1	3	1	G. 729a, 20ms
10	2	0	2	1	G. 711, 40ms
11	2	0	3	0	G. 729a, 20ms
12	2	0	3	1	G. 729a, 20ms
13	2	1	3	0	G. 729a, 20ms
14	2	1	3	1	G. 729a, 20ms
15	2	0	3	1	G. 711, 40ms

3. ABNDL (Routing Table)

	START LOC-ID	1st - 15th LOC-ID	END LOC-ID
1	0	100	20
2	1	100	20

Bandwidth Threshold Value

	LOC-ID A	LOC-ID B	Threshold Value	Remarks
1	20	100	1520 Kbps	Line A

Data Programming in Node B (FPC=2)

1. ALOCL

	IP Address	LOC-ID
1	0. 0. 0. 0/32	0
2	10. 41. 10. 0/24	1
	FPC (Fusion Point Code)	LOC-ID
3	1	10
4	3	30

2. ABNDL (Routing Table)

	START LOC-ID	1st - 15th LOC-ID	END LOC-ID
1	0	200	10
2	0	300	30
3	1	200	10
4	1	300	30
5	10	200 - 300	30

Bandwidth Threshold Value

	LOC-ID A	LOC-ID B	Threshold Value	Remarks
1	10	200	1140 Kbps	Line B
2	30	100	760 Kbps	Line C

Data Programming in Node C (FPC=3)

1. ALOCL

	IP Address	LOC-ID
1	0. 0. 0. 0/32	0
2	10. 42. 10. 0/24	1
	FPC (Fusion Point Code)	LOC-ID
3	20	20

2. ABNDL (Routing Table)

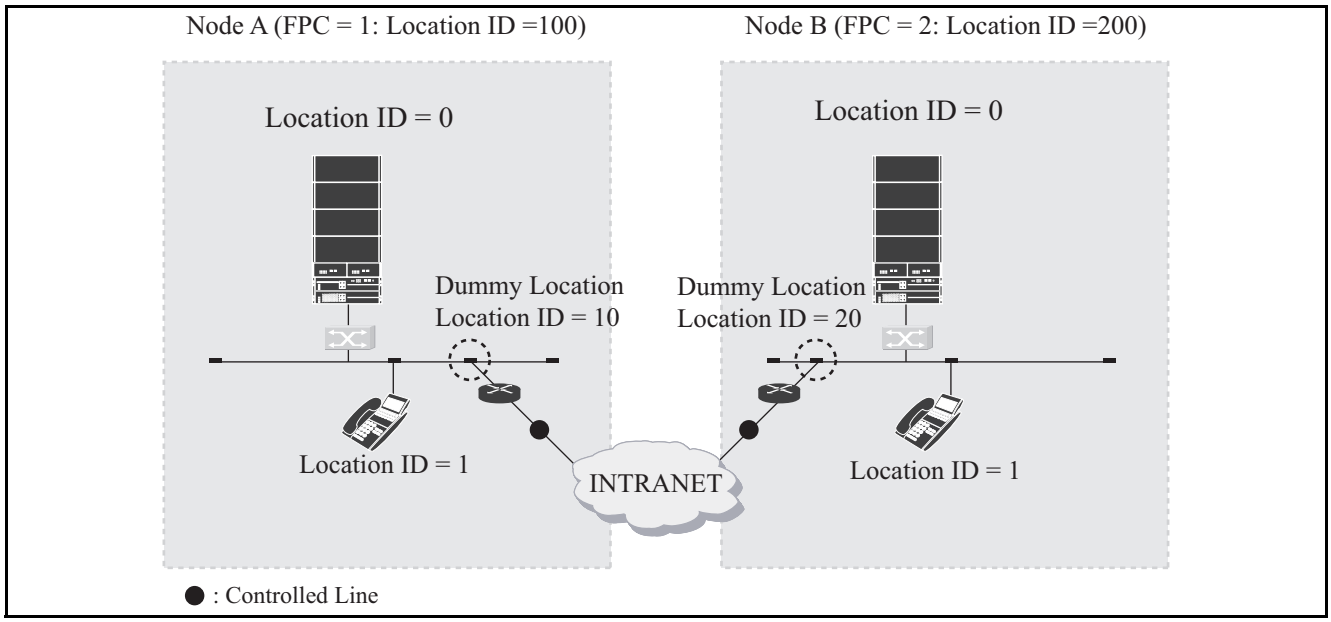
	START LOC-ID	1st - 15th LOC-ID	END LOC-ID
1	0	400	20
2	1	400	20

Bandwidth Threshold Value

	LOC-ID A	LOC-ID B	Threshold Value	Remarks
1	20	400	1140 Kbps	Line B

<Example Payload Data Programming>

It is necessary to harmonize the IP voice control data such as payload from end to end. For this reason, the payload data among Peer-to-Peer FCCS network must be harmonized from end to end (not according to the relay routes).



[NDM Data Assignment (AIVCN)]

- FPC = 1, LOC-ID = 0 - FPC = 2, LOC-ID = 0
- FPC = 1, LOC-ID = 0 - FPC = 2, LOC-ID = 1
- FPC = 1, LOC-ID = 1 - FPC = 2, LOC-ID = 0
- FPC = 1, LOC-ID = 1 - FPC = 2, LOC-ID = 1

[ABNDL Data Programming in Node A]

Routing Table:

- LOC-ID = 0 → LOC-ID = 10 → LOC-ID = 200
- LOC-ID = 1 → LOC-ID = 10 → LOC-ID = 200

Threshold:

- LOC-ID = 10-200: xx (Kbps)

[ABNDL Data Programming in Node B]

Routing Table:

- LOC-ID = 0 → LOC-ID = 20 → LOC-ID = 100
- LOC-ID = 1 → LOC-ID = 20 → LOC-ID = 100

Threshold:

- LOC-ID = 20-200: xx (Kbps)

P-77 Phantom Number Distinctive Ringing - Single Line
(Series 8800 software feature)

General Description

This feature allows Incoming Calls to a Phantom Number to ring with a Distinctive Ringing Pattern, allowing a user to determine if the call is for them by the Ringing Pattern.

Note: This feature is available in North America only.

Operating Procedure

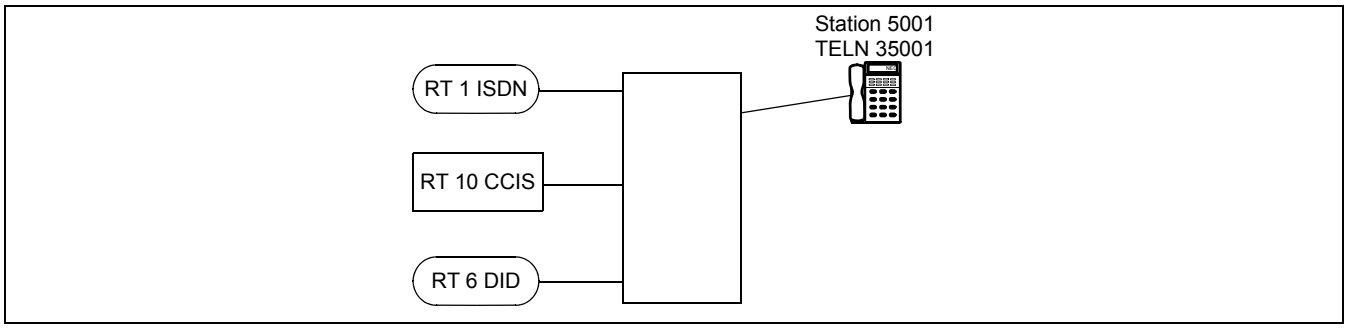
There will be no change in the Operating Procedure for the user.

Environment

- APHN has Station 5001.
  1 - 5088: Ringer Pattern 0
  2 - 5089: Ringer Pattern 6
  3 - 5090: Ringer Pattern 7
  4 - 5091: Ringer Pattern 1
  5 - 5092: Ringer Pattern 1
- APHNN TELN has 35001.
  1 - 35088: Ringer Pattern 0
  2 - 35089: Ringer Pattern 6
  3 - 35090: Ringer Pattern 7
  4 - 35091: Ringer Pattern 1
  5 - 35092: Ringer Pattern 1
  6 - 35093: Ringer Pattern 1
  7 - 35094: Ringer Pattern 1
  8 - 35095: Ringer Pattern 1
  9 - 35096: Ringer Pattern 1
  10 - 35097: Ringer Pattern 1
  11 - 35098: Ringer Pattern 1
  12 - 35099: Ringer Pattern 1
  13 - 35100: Ringer Pattern 1
  14 - 35101: Ringer Pattern 1
  15 - 35102: Ringer Pattern 1
  16 - 35103: Ringer Pattern 1

These Phantom Numbers can be either four digits or five digits for ISDN, CCIS and DID. They can also be a maximum of six digits for APHN, and a maximum of 16 digits for APHNN.

Example of operation



ISDN Case

- Route 1 dials 972-233-5001.
  This call rings with Pattern 1.
- Route 1 dials 972-233-5088.
  This call rings with Pattern 0.
- Route 1 dials 972-233-5089.
  This call rings with Pattern 6.
- Route 1 dials 972-233-5091.
  This call rings with Pattern 7.
- Route 1 dials 972-233-5092 ~ 972-233-5103.
  This call rings with Pattern 1.

CCIS Case

- Route 10 dials 35001.
  This call rings with Pattern 1.
- Route 10 dials 35088.
  This call rings with Pattern 0.
- Route 10 dials 35089.
  This call rings with Pattern 6.
- Route 10 dials 35090.
  This call rings with Pattern 7.
- Route 10 dials 35091 ~ 35103.
  This call rings with Pattern 1.

DID case

- Route 1 dials 972-233-5001.
  This call rings with Pattern 1.
- Route 1 dials 972-233-5088.
  This call rings with Pattern 0.
- Route 1 dials 972-233-5089.
  This call rings with Pattern 6.
- Route 1 dials 972-233-5091.
  This call rings with Pattern 7.
- Route 1 dials 972-233-5092 ~ 972-233-5103.
  This call rings with Pattern 1.

Service Conditions

1. These patterns cannot be changed from the data that follows:
   a.) Phantom Number 1 in the Phantom List: Ringer Pattern 0
   b.) Phantom Number 2 in the Phantom List: Ringer Pattern 6
   c.) Phantom Number 3 in the Phantom List: Ringer Pattern 7
   d.) Phantom Number 4-5 (for Station Dial) or 4-16 (NDM) in the Phantom List: Ringer Pattern 1
2. Calls to the main station Number will ring with Pattern 1.
3. This feature is mutually exclusive with Personal Ringer (SFI 148).
4. If both features are on, Personal Ringer will have higher priority.
5. This feature is mutually exclusive with Distinctive Ringing - Caller ID.
6. This feature is only available for Incoming Trunks, such as ISDN, CCIS, DID.
7. The Phantom Number in Station Dial Phantom Call is not available for a FCCS Call and Internal Call.
8. This feature is only available for a Single Line (Analog Phone).
9. This feature is only for Station Dial (the maximum amount of Phantom Numbers for one station is five) and NDM (the maximum amount of Phantom Numbers for one station is 16). This feature is not available for LDM.
10. This feature cannot be used with Distinctive Ringing or Distinctive Ringing for Toll Calls.
11. This feature cannot be used with Distinctive Ringing Priority List.
12. After the Ringer Timers are changed in System Data 3, the line package must be reset before the new Ringer Timers will go in to effect.
13. Phantom Numbers 4 and 5 may ring as Pattern 1.
14. The Ringer Patterns 0, 6, and 7 may begin with one burst of Caller ID ring.
15. This feature works with Call Transfer - Terminating, Call Transfer - Attendant and Three-way Calling.
16. This feature does not work with Call Forwarding.
17. If Phantom Number Distinctive Ringing - Single Line, Distinctive Ringing List and Distinctive Ringing for Toll Call are active at same time, and the Called Number is a Phantom Number, the highest priority is Phantom Number Distinctive Ringing - Single Line, then Distinctive Ringing List, then Distinctive Ringing for Toll Call.

Programming

To assign

STEP 1: **ASYD** - Assignment of System Data. See the Command Manual for details on Distinctive Ringing Patterns assigned in System Data 3 Indexes.

STEP 2: **ASFC** - Assign of Service Feature Restriction Class.

TN: X
SFC: X
SFI: 193 must be set to "1" for this service to be in effect.

STEP 3: **APHN/N** - Assign the Phantom Numbers to the main station. For APHN, five Phantom Numbers can be assigned. For example, 5088, 5089, 5090, 5091, 5092.
For APHNN, 16 Phantom Numbers can be assigned. For example, 35088, 35089, 35090, 35091, 35092, 35093, 35094, 35095, 35096, 35097, 35098, 35099, 35100, 35101, 35102, 35103.

To delete

STEP 1: **APHN/N** - Delete the Phantom Numbers of the main station. For APHN, five Phantom Numbers can be deleted. For example, 5088, 5089, 5090, 5091, 5092.
For APHNN, 16 Phantom Numbers can be deleted. For example, 35088, 35089, 35090, 35091, 35092, 35093, 35094, 35095, 35096, 35097, 35098, 35099, 35100, 35101, 35102, 35103.

STEP 2: **ASFC** - Assignment of Service Feature Restriction Class.

TN: X

SFC: X

SFI: 193 must be set to "0" for this service to be removed.

STEP 3: **ASYD** - Assignment of System Data. See the Command Manual for details on Distinctive Ringing Patterns assigned in System Data 3 Indexes.

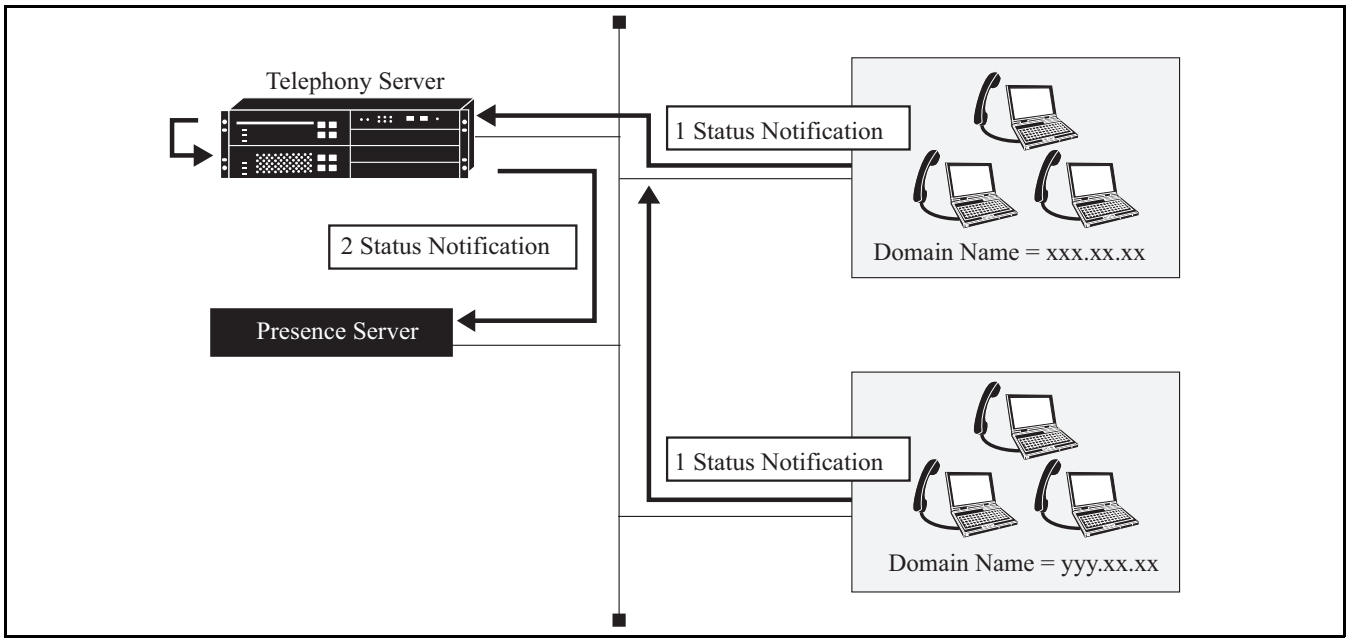
P-78 Presence

General Description

This feature allows SIP terminals to display the current status of the target party controlled by Presence Server on their LCDs or screens. By knowing the current status of the target party before making a call, the SIP terminal's user can choose the most suitable communication tool for each target party.

Telephony Server supports the following services:

- For Soft Phone (SIP):
  Controls the domain name, IP address and port number of Presence Server that notifies the status of Soft Phone (SIP), on a terminal basis.
- For WLAN Handset (MH Series):
  Controls the IP address and port number of Presence Server that notifies the status of WLAN Handset (MH Series), on an SP basis.
- Sends login (when Registration process is completed) and logout (when the terminal is logged out or Health Check NG is detected) information for each SIP terminal to Presence Server specified by TP.



Operating Procedure

No manual operation is required.

Service Conditions

1. This feature is available for the following SIP terminals supporting Presence service.

   - Soft Phone (SIP) **Note 1**
   - WLAN Handset (MH Series)

   Note 1: When using this feature with Soft Phone (Proprietary Protocol), the status display for other terminals cannot be updated automatically. Therefore, it is necessary to update the status display manually. Also, Presence Server must support the Proprietary Protocol (no particular data programming is required). For details, see the manual for Presence Server.

2. This feature requires a Presence Server.
3. A maximum of 128 characters (single-byte alphanumeric) can be used as a domain name.
4. One domain name cannot be shared among multiple Presence Servers.
5. In FCCS network, a maximum of 1020 domain names can be used.
6. If any data described in the Programming section is changed, applicable terminals need to be reset.

<For Soft Phone (SIP) only>

1. It is necessary to assign the Telephone Number (TELN) of Soft Phone (SIP).
2. IP address and port number of the SIP Presence Server are set per domain name. Also, one Presence Server can be specified by multiple domain names. In this case, SIP Presence Server must support multiple domain names.
3. When the existing data assigned on the INDEX parameter of ADNIN command is changed, the applicable Soft Phone needs to be reset to enable the new data.
4. Microsoft® XML Version 3.0 or later is required for the Soft Phone's PC to communicate with Presence Server. It has already been installed if Internet Explorer 6.0 is installed or Windows® XP is used on the PC. If it is necessary to install it, use Microsoft® XML Version 3.0 application software supplied with Soft Phone install CD-ROM or download it from the official home page of Microsoft®.
5. Soft Phone uses its Prime Line Number to access to a Presence Server. Thus, the same Prime Line Number cannot be assigned to multiple Soft Phones.
6. This feature is available for Soft Phones using Soft Phone Wired Phone Handset [S-147] or Soft Phone WLAN Handset [S-159]. Soft Phones in logout status are treated as "absence."
7. It is necessary to reboot the applicable SIP terminals when changing the Presence Server's IP address.
8. When replacing hardware for the Presence Server, Maintenance Personnel must perform operations in the following order: **Note 2**
   a. Startup the new Presence Server.
   b. Verify that the new Presence Server is connected to the network.
   c. Perform data assignment to the new Presence Server.
   d. Shutdown the old Presence Server.

   Note 2: If the Maintenance Personnel does not perform operations in the above order, it may become necessary to reboot the Telephony Server and the Presence Server.

<For WLAN Handset (MH Series) only>

IP address and port number of Presence Server are assigned on a SP basis. Multiple Presence Servers cannot be used for a SP.

Programming

Note: If any of the following data is changed, applicable SIP terminals needs to be reset.

Note: For detailed information on each command, refer to Command Manuals.

<Programming for Soft Phone (SIP)>

STEP 1: **ASDIN** - Assign IP address and port Number of the Presence Server corresponding to the assigned domain name.

STEP 2: **ALGSN** - Assign the Telephone Number of Soft Phone (SIP).

STEP 3: **ADNIN** - Allocate the Index of domain name assigned on ASDIN to UGN and TELN assigned on ALGSN.

<Programming for WLAN Handset (MH Series)>

ASSDL/ASSDN - Assign IP address and port Number of the Presence Server on the PRESENCE SERVER tab.

P-83 Phonebook

General Description

This feature offers the phonebook function used on a station basis or a system basis. During the call origination using Phonebook, name and dial number registered in Phonebook which corresponds to the number dialed by users are displayed. During the call termination, name and dial number registered in Phonebook which corresponds to the calling party number are displayed. There are four types of Phonebook: Local Phonebook, System Phonebook, phonebook on telephone terminal (hereinafter called Terminal Phonebook) and Corporate Phonebook.

- **Local Phonebook**

  Local Phonebook is used on a station basis. Data is managed per station number and can be modified or registered as a new entry by operation from telephone terminal. Up to 100 registrations are available per station. Up to 32 digits can be registered for each dial number. Up to 24 characters can be registered for each name. For stand alone system, Local Phonebooks of 4000 stations can be used. For FCCS system, Local Phonebooks of 4000 stations can be used per Node.

- **System Phonebook**

  System Phonebook is used on a system basis. Data can be modified or registered as a new entry with Maintenance Console (PCPro). Up to 12000 registrations (**Note 1**) are available per FCCS system or per stand alone system.

  Note 1: For FP95-112 V2 or later, up to 60000 registrations are available by System Data (ASYDL/ASYDN, Index 683, Bit 4=1).

  Up to 32 digits can be registered for each dial number. Up to 24 characters can be registered for each name. For FP95-112 V2 or later, the data can be registered separately per UGN with System Data (ASYDL/ASYDN, Index 1136, Bit 0).

  Note: If you change the system data while using the terminal, delete the phonebook data and register it again after changing the system data.

<For DT820/DT710, the following features are not available.>

- **Terminal Phonebook (indicated as Personal Directory on the terminal LCD)**

  Terminal Phonebook is installed with telephone terminal.This phonebook is available in DT830G/DT830/DT700 Series. Up to 12000 registrations are available per FCCS system or per stand alone system. Up to 32 digits can be registered for each dial number. Up to 24 characters can be registered for each name.

- **Corporate Phonebook**

  Corporate Phonebook is created with XML application and controlled by the external XML server. This phonebook is shared in the entire system. Please ask the system integrator about details of the Corporate Phonebook application on external XML server.

Operating Procedure

For operating procedure, refer to the related User's Guide.

Service Conditions

<Important Notices>

1. The following Phonebook types can be specified by ASFC/ASFCN command.

Kind of Phonebook	ASFC/N SFI Data
Local Phonebook	ASFC/N SFI:245 = 1
System Phonebook	ASFC/N SFI:246 = 1
Terminal Phonebook **Note 2**	ASFC/N SFI:247 = 1

Note 2: When SFI 247 (Terminal Phonebook) setting is changed;
DT830G/DT830/DT700 Series terminals require to request re-registration. (Make-busy and make-busy cancel operation by MBST command is also available.)
Data change does not turn effective without re-registration.

2. Each terminal can use the following phonebooks in combination;

	Local Phonebook	System Phonebook
Terminal Phonebook	Not supported	Supported

Note: Terminal Phonebook can be used in combination with Local Phonebook through the entire system; however, Terminal Phonebook cannot be used in combination with Local Phonebook in one terminal.
When SFI 247=1 (Terminal Phonebook is allowed) and SFI 245=1 (Local Phonebook is allowed) are assigned, SFI 247=1 takes priority and Terminal Phonebook is effective.

Note: The following shows the priorities to display the name when a called party receives a call.
Terminal Phonebook > Local Phonebook > System Phonebook

3. As Terminal Phonebook is not supported in DT820/DT400 Series/DT300 Series and DT710, only the Local and System Phonebook are available for those terminals.
When SFI 247=1 (Terminal Phonebook) is selected on DT820/DT400 Series/DT300 Series and DT710, only the System Phonebook can be used.
Available Phonebook type depends on the SFI 247 (Terminal Phonebook) setting;

SFI 247	DT710 (2-button type)/ DT410 (2-button type)/ DT310 (2-button/12-button type)	DT820/ DT710 (6-button type)/ DT430/DT330/ DT410 (6-button type)/ DT310 (6-button type)/	DT830G/DT730G/ DT830/DT730/ DT750
0	Not supported	Local Phonebook/System Phonebook	Local Phonebook/System Phonebook
1	Not supported	System Phonebook	Terminal Phonebook/System Phonebook

*This table shows the conditions when SFI 245=1 (Local Phonebook is allowed) and SFI 246=1 (System Phonebook is allowed) are assigned.

4. When the same dial numbers or the same names are registered to Phonebook, the following conditions are applied.

	Same Name, Different Dial Number Name: abc, Number: 123 Name: abc, Number: 345	**Different Name, Same Dial Number** Name: abc, Number: 123 Name: def, Number: 123	**Same Name, Same Dial Number** Name: abc, Number: 123 Name: abc, Number: 123
Local Phonebook	Supported	Supported **Note 4**	Supported
System Phonebook	Supported **Note 3**	Not supported	Not supported

Note 3: Up to 20 different dial numbers can be registered with the same name.

Note 4: For the following situation, there may be difference in the displayed name between “A call is originated using Phonebook” and “the originating station returns to the former line.” It is because the system searches Local Phonebook database for the number again when the originating station returns to the former line.

A call is originated using Phonebook -> the line is engaged in the call -> the originating station performs operation of Call Forwarding and Call Hold -> the originating station returns to the former line

5. When terminals other than DT830G/DT830/DT700 Series/DT400 Series/DT300 Series are used, they work in Dterm Series i mode. So SFI 247 (Terminal Phonebook) setting takes no effect.
6. When the name search is executed by entering a few characters in the beginning of the name on the Desktop terminal LCD, the matched name with entered characters is displayed at first.
Then the display shows other names in the Phonebook database by pressing UP and DOWN key.

<Example> Searching the database for the name includes **“**TOU**”** when the following four names are in the database.

ABIKO
Shinbashi
→TOUHOKU
chiba

- “TOUHOKU” shown above with the arrow appears on the display as the search results of “TOU”.
- When DOWN key is pressed once while “TOUHOKU” is on the display, “chiba” appears.
When DOWN key is pressed twice while “TOUHOKU” is on the display, “ABIKO”, which is at the top of the database, appears.

- When UP key is pressed while “TOUHOKU” is on the display, “Shinbashi” appears.
  When UP key is pressed three times while “TOUHOKU” is on the display, “chiba”, which is at the last of the database, appears.

  Note: The order of names in the database changes depending on whether distinguishing between upper case and lower case characters. The above explains the example when ASYDL System data 1 Index 677 Bit 7=0 (Distinction between upper case and lower case characters is in service) is assigned.

7. Searching database method whether this feature distinguishes between upper case and lower case characters while searching database for a name can be specified by system data (ASYDL System data 1 Index 677 Bit 7). The searching database method depends on the setting of ASYDL System data 1 Index 677 Bit 7 at the time APBKN data assignment is finished. If APBKN data is assigned and the Administrator wants to change the searching method by changing ASYDL System data 1 Index 677 Bit 7, the APBKN entries must first be deleted before changing ASYDL System data 1 Index 677 Bit 7. APBKN entries must then be reentered to follow the new search method.

   - ASYDL System data 1 Index 677 Bit 7=0 (Distinction between upper case and lower case characters is in service.)

     <Example> ABIKO
     Shinbashi
     →TOUHOKU
     chiba

     * Upper case and lower case are distinguished as different character strings. Upper case has priority over lower case.

   - ASYDL System data 1 Index 677 Bit 7=1 (Distinction between upper case and lower case characters is out of service.)

     <Example> ABIKO
     chiba
     Shinbashi
     TOUHOKU

     * Upper case and lower case are not distinguished as different character strings.

8. Hooking or a pause cannot be registered in a dial number.
9. When Abbreviated Call Codes using Speed Calling - Station, Group, or System features are registered in the Phonebook, a call cannot be originated by dialing these codes.
10. When a call rings on a PS line of a DT800/DT700/DT400/DT300 Series where the PS line is accommodated as a Sub Line, the calling party name registered in the Phonebook is not displayed until the call is answered.
11. If you register 60000 phonebook entries, sufficient memory is required. Therefore, execute DDMO command to check the free size area of the data memory in NDM, and change the System Data (ASYDL/ASYDN SYS1 Index 514/Index 957) if required. **Note 5**, **Note 6**

Note 5: If the NDM is insufficient, registration error occurs.

Note 6: To register 10000 phonebook entries, 0.7MB (713KB) of free disk space is required. When the data memory in NDM is about 2.0 MB (2048KB), you can register up to 28700 (2048 divided by 713 equals 2.87) entries.

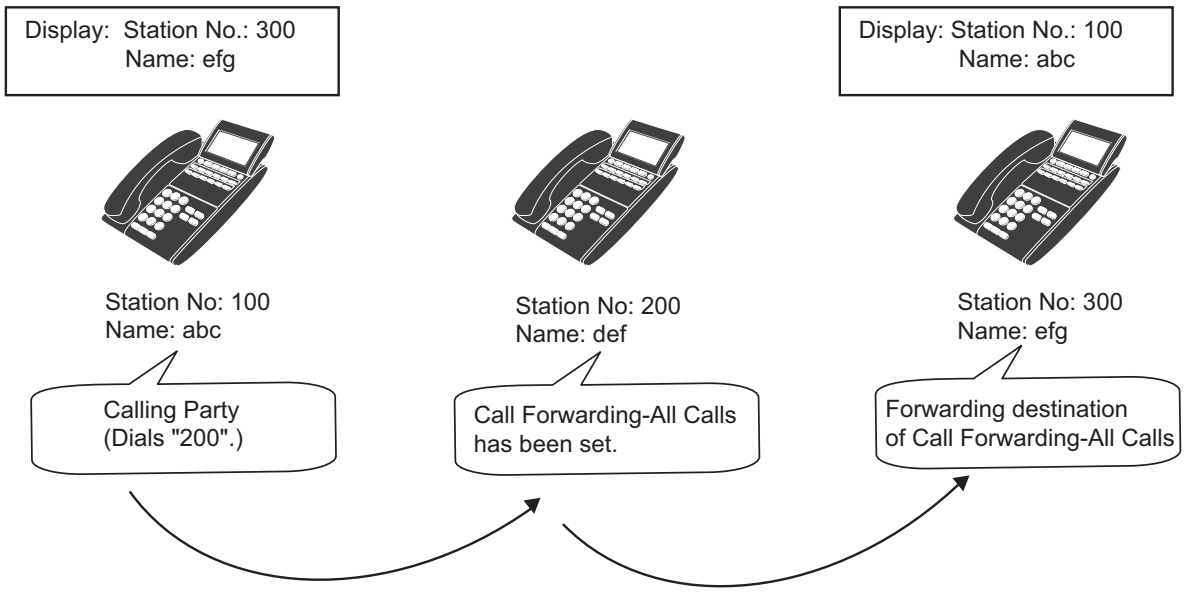
<Interactions with Other Features>

1. Backup of Local Phonebook and System Phonebook data is available by executing maintenance commands and periodical diagnosis.
2. PHONEBOOK [P-83] cannot be used when SOFT PHONE WLAN HANDSET [S-159] or SOFT PHONE WIRED-PHONE HANDSET [S-147] is activated, key operation is not effective.
3. The following shows whether Phonebook is supported by SR-MGC and FCCS Cluster.

	SR-MGC	FCCS Cluster
Local Phonebook	Not supported	Not supported
System Phonebook	Supported	Supported **Note 7**

Note 7: FCCS Cluster is supported when data is assigned to Backup node in advance.

4. When Call Forwarding feature is set to the originally called station, calling number is searched from Phonebook of forwarding destination.



5. Name display by Phonebook is not supported when the forwarding destination of the following Call Forwarding features is out side of Self Office.
   - Call Forwarding - All Calls [C-5]
   - Call Forwarding - Busy Line [C-2]
   - Call Forwarding - Don't Answer [C-3]
   - Call Forwarding - Logout
   - Call Forwarding - PS Incoming Incomplete [C-155]

6. Searching Phonebook database for the name is not available if called party number is not received from counter office when the call is originated to out side of Self Office by HOTLINE [H-1].
7. When the same dial numbers are registered to both NAME DISPLAY-SYSTEM [N-28] (ANDD command) and Phonebook, the data registered to Phonebook takes priority.
8. It is possible to change the indication location of the name and number information notified from the system on DT700 Series LCD. For the detailed information, refer to NAME DISPLAY-SYSTEM [N-28].
9. Display of Phonebook takes priority over CALLING PARTY NUMBER-NAME ASSIGNMENT AND DISPLAY [C-157].
10. For the information about Phonebook for Hotel System, refer to PHONEBOOK FOR HOTEL [P-85].
11. When the number of digits of the dialed number is less than the maximum necessary digits (values assigned to MND parameter of AMND/AMNDL/AMNDN), the call is originated after a predetermined prepause timer assigned by ASYD, SYS 1, Index 128 Bits 0-3 (default: 12 seconds) expires.
12. This feature does not support OAI or ACD system.

<Hardware/Software>

1. Desktop terminals (except for Softphone) equipped with Soft key function are required.
2. To activate PHONEBOOK [P-83] on DT800 Series/DT700 Series/DT400 Series/DT300 Series equipped with the cursor key or Menu key, Phonebook can be activated by pressing the cursor key, Menu key, Feature key or Soft key.

   Note: In the cases of DT820/DT710 6-button type and DT410/DT310 6-button type, cursor key, Feature key or Soft key is available to activate the feature. This feature cannot be activated by Menu key on DT820/DT710 6-button type and DT410/DT310 6-button type.

3. To use this feature, one of the following cards is required:

   - ELC card mounted in 7U-PIR
   - PA-16ELCN-A in DT300 Series mode
   - PA-16ELCN in DT300 Series mode

4. The key operation for Speed Calling-One Touch takes no effect while the Phonebook is displayed on a LCD of a terminal.

<Interactions with Networks>

1. Display pattern of calling name and calling number for incoming Trunk Calls can be changed by system data (ASYD, System Data 1, Index 390, Bit 0). For incoming Station Calls, the pattern cannot be changed.

   - Example when name display takes priority; the beginning of number is displayed:

     Name “Suzuki Taro (Japan-Chiba-Abiko)”, Number “047156789012345678901234567 8”

     0471567*
     Suzuki Taro (Japan-Chiba*

- Example when number display takes priority; the beginning of number is displayed:

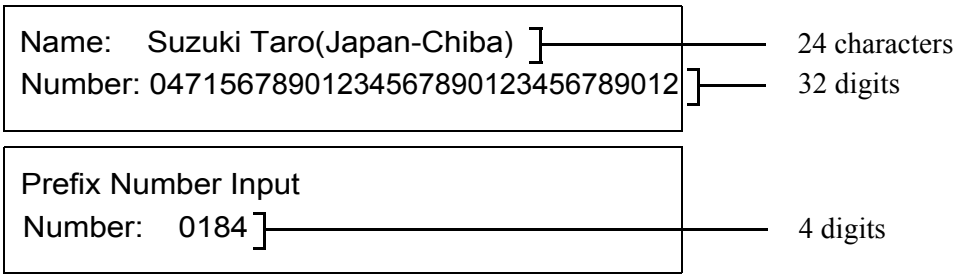
  Name "Suzuki Taro (Japan-Chiba-Abiko)", Number "047156789012345678901234567 8"

  Suzuki *
  047156789012345678901 23*

2. The prefix (the access number of terminated route) can be included to the dial number registered in Phonebook data. When the prefix is included, be sure to match the prefix with the number assigned in "to add the access code of incoming route" by ACNP and ACND command. For the data assignment, refer to Programming.
3. Adding prefix operation using "Prfi" key in Phonebook data display is available only once. To retry the operation, cancel the assigned prefix by pressing "Cnl" key then restart adding prefix operation.
4. Up to 10 digits can be added as a prefix.
5. When the prefix is added, up to 32 digits including the prefix can be sent as dial number of Phonebook data.

- Example of display before prefix is added:

  Name "Suzuki Taro(Japan-Chiba)", Number "047156789012345678901234567890 12"



- Example of display after prefix is added:

  Name "Suzuki Taro(Japan-Chiba)", Number "047156789012345678901234567890 12"



<Conditions Specific to This Feature>

1. Display of character types are small (lower case) and large (upper case).
2. "0-9", "*" and "#" can be used for numeric input.
3. A blank space is also distinguished while searching database for a name.

4. Even if there is a blank space (0×20) at the end of input name for the name registration, the name is registered without the blank space. For the name registration of System Phonebook assigned by APBKN command, the blank space is not deleted when there is a blank space at the end of input name.
5. When a call is terminated to a line of a terminal being engaged in Phonebook operation, the operation is continued. If the terminal answers the call, Phonebook operation is canceled and the terminal is engaged in the call. When the call is finished, the terminal becomes the state Phonebook operation is canceled.
6. Phonebook operation is available only from the terminal in idle state. Phonebook operation is not effective while the terminal in a call or in the Headset Mode.
7. Phonebook operation will be exited in midstream by pressing “Exit” key and “Pbook” key. When “Cnl” key is pressed, the display returns to previous display.
8. If a calling party number of the terminated call is registered to both of System Phonebook or Local Phonebook, registered data to Local Phonebook takes priority.
9. When the incoming call is terminated to Sub line, if a station set the terminated line as Prime line, the data for name display and number display will be searched from Phonebook of a station which the terminated line is set as My line. However, if the call is answered, the data for name display and number display will be searched from Phonebook of the answering station.
10. If there are two stations (Station A and B) both with the same line appearance when Multiple Line Operation -D [M-20D] is activated, on Station A the terminated line being its Prime Line and on Station B the terminated line being its My Line, the name display on the stations will be as shown below:
    * In this case, Station B uses Local Phonebook only.

    - Station A (the incoming call terminates to Prime Line of this station):
      While ringing state, the station displays the name as found in the phonebook of Station B.
      After the station goes off-hook, the name then is updated to display the name from the phonebook of Station A.

    - Station B (the incoming call terminates to My Line of this station):
      The station displays the name searched from the phonebook of Station B.

11. The first entry of Phonebook data is displayed when the call is originated/terminated by dialing the number.
12. When the REDIAL key is pressed while displaying Phonebook data, the dial number assigned to redial key is displayed for 10 seconds or so.
13. When “SP (blank space)” is input during name registration, edit or search by telephone terminal key operation, “_ (underscore)” is displayed though it is recognized as “SP (blank space)” in data programming. Because of that, “_ (underscore)” is not displayed if “SP (blank space)” is input by APBKN command.
14. When digit key is pressed during phonebook operation, the pressed number is not displayed.
15. Pressing the SPEED CALLING ONE-TOUCH key while displaying phonebook data, the key operation for Speed Calling-One Touch takes no effect on DT800/DT700/DT400/DT300 Series.
16. Even though System Phonebook and Terminal Phonebook are set as out of service (ASFC SFI: 246=0/ SFI: 247=0), the parameters under [Directory] menu will be displayed on the terminal’s configuration menu regardless of SFI: 246/SFI: 247 data setting.
17. Phonebook information can be displayed in 4-line with a DT830G/DT830/DT700 Series terminal that has 4-line (except the line for icon display) LCD. By displaying Phonebook information in 4-line, you can see the name and the number in the search result display or the deletion confirmation display, whereas in 3-line, you can only see the name.

If you wish to display Phonebook in 3-line as before, assign the system data (ASYDL, SYS1, Index 681, Bit 1=1).

Note: The search result with 4-line display is only available for Local Phonebook and System Phonebook. Deletion confirmation with 4-line display is only available for Local Phonebook.

<Conditions Specific to UMGi/RNIP System>

1. Local Phone Book is not available for an RN in survival mode.

<Other Conditions>

1. Conditions specific to Local Phonebook are as follows.
   a.) Local Phonebook data is stored on a LENS basis.
   b.) For available input characters, numbers, and marks of Local Phonebook data, refer to the character code <Character Code> on page 1912.
   c.) Input characters, numbers, and marks cannot be modified from the middle of the string array. they need to be deleted from the ending and modified by inputting again.
   d.) When this feature is activated, be sure to execute the routine diagnosis program and back up the Local Phonebook data periodically by ASYD System Data 1 Index 306 data assignment.
   e.) Data registration to Local Phonebook is not available during Local Phonebook data backup. The display shows “Entry NG (Data Backup).”
   f.) Local Phonebook data can be saved and verified between flash card and PCPro using Telephony Server Maintenance Menu. For more information, refer to Operations and Maintenance Manual.
   g.) Local Phonebook data can be copied and verified between Drive C and Drive B by specifying “Office Data” in Data Type Select field by the HDD_HDD/HDD_HDD_N command. For more information, refer to Operations and Maintenance Manual.
2. Conditions specific to System Phonebook are as follows.
   a.) System Phonebook data can be edited by Maintenance command. The data cannot be edited from the terminal.
   b.) The following characters, numbers, and marks can be used for System Phonebook.

   A-Z, a-z, 0-9, !, ?, -, +, %, &, /, (,), =, ., ,, :, ", ', ;, *, #, @, [,], SP (blank space)

   À, Á, Â, Ã, Ä, Å, Æ, Ç, É, Ê, Ñ, Ö, Ü, ß, à, á, â, ä, å, æ, ç, è, é ,ê, ë,
   ì, í, î, ï, ñ, ò, ó, ô, õ, ö, ù, ú, û, ü, ¿, ¡

   Note: õ is displayed in capital letter. There is no small letter for this character available.

   c.) Special characters such as characters having an umlaut can be input in NAME field of APBKN/CPBKN command on PCPro by changing input language with the function of Operating Systems. To change input language, refer to the manuals provided by each OS.

<Conditions Specific to combination of Call History and Phonebook>

Available: × Not available: N Not supported: –

Kind of Phonebook	DT400/DT300 Series Note 9			DT800/DT700 Series					DtermIP/ DtermSIP with LCD
	DT430 / DT330	DT410/DT310		DT750	DT830/ DT730/ DT830G/ DT730G	DT820	DT710		
		6-button type	2-button/ 12-button type				6-button type	2-button type	
Local Phonebook (ASFC/N SFI:245)	×	×	–	×	×	×	×	–	×
System Phonebook (ASFC/N SFI:246)	×	×	–	×	×	×	×	–	×
Terminal Phonebook (ASFC/N SFI:247) Note 8	N	N	N	×	×	–	N	N	N
Corporate Phonebook	N	N	N	×	×	–	N	N	N
Terminal Call History (outgoing call history/incoming call history)	×	N	N	×	×	–	N	N	N
Call History	×	×	–	×	×	×	×	–	N

Note: Phonebook can be used in combination with Call History on DT830G/DT830/DT700 Series/ DT400 Series/DT300 Series terminals only.

Note 8: When SFI 247 (Terminal Phonebook) setting is changed; DT830G/DT830/DT700 Series telephone terminals require to request re-registration. (Make-busy and make-busy cancel operation by MBST command is also available.) Data change does not turn effective without re-registration.

Note 9: The condition in this table applies to the terminals working in DT400 Series/DT300 Series Mode. Phonebook does not support the terminals working in Dterm Series i Mode.

1. Terminated numbers can be registered from Call History to Local Phonebook on DT830G/DT830/DT700 Series.

   Note: On telephone terminals before DT830G/DT830/DT700 Series/DT400 Series/DT300 Series, Call History and Local Phonebook are not interlocked, so incoming numbers cannot be registered from Call History to Local Phonebook.

   Note: For the information of phonebook entry to Terminal Phonebook using Terminal Call History, refer to the related manuals of DT830G/DT830/DT700 Series.
2. When an incoming call terminates on the accommodating line while registering from Call History to Local Phonebook, the station can keep phonebook operation while ringing. When the incoming call is answered, the connection is established and phonebook operation is canceled. After releasing the call, the LCD shows empty display (not back to incoming call history display).
3. Call History can store up to 32 digits of dial number and 16 characters of name. On the other hand, Local Phonebook can store up to 32 digits of dial number and 24 characters of name. Therefore, all the digits of numbers stored in Call History can be saved to Local Phonebook.
4. The entry from Call History to Local Phonebook is every time dealt with a new data addition. So the existing data in Local Phonebook is not edited even when the name is listed in the Call History.
5. As DT400 Series/DT300 Series (DT400/DT310) 2-button/12-button type and DT700 Series (DT710) 2-button type (without LCD) cannot use Call History [C-195], phonebook entry from Call History to Local Phonebook is not available.
6. For DT800/DT700 Series, you can add a prefix before the displayed number when originating a call by Terminal Phonebook. For DT400 Series/DT300 Series, prefix function is not supported.

   Note: For using prefix function, register the prefix with user setting on the terminal beforehand. For details, refer to the related manuals of DT800 Series/DT700 Series.

7. The following types of data entry are not supported:

   - From Call History to Terminal Phonebook
   - From Terminal Call History to Local Phonebook

8. In registering operation from Call History to Local Phonebook, the LCD cannot go back to Call History indication from phonebook entry by the CNL key.
9. When the combination of Call History and Terminal Phonebook is used, “Pbook” Soft key does not appear on Call History display.
10. The priority order to display the name on incoming call termination and call history is as follows.

High Priority



Low Priority

- Registered name on the selected phonebook (Local Phonebook/Terminal Phonebook)
- Registered name on System Phonebook
- Registered name assigned by Name Display feature (ANDD) and Caller ID feature (ACID)

Note: In the following cases, contact names are not displayed in the above priority order.
- The combination of Terminal Phonebook and Call History is used:
- Contact name of priority order I is not displayed. Contact name of priority order II or priority order III is displayed.
- The combination of Local Phonebook and Terminal Call History when ASYDL, System data 1, Index 673 Bit 4=0 is assigned (inform only Calling Number to call history):
- Contact name is not displayed as the call history.
- The combination of Terminal Phonebook and Terminal Call History when ASYDL, System data 1, Index 673 Bit 4=0 is assigned (inform only Calling Number to call history):
- Contact name of the priority order II and III are not displayed as the call history.

Programming

STEP 1: **ASYD** - Assign the system data.

<Assignment of periodical Local Phonebook data backup>

System Data 1
- Index 86, Bit 0 = 1 (System Message [7-0] is registered when no errors are detected by the Routine Diagnosis Program)
- Index 86, Bit 1 = 1 (System Message [7-P] is registered when an error is detected by the Routine Diagnosis Program)
- Index 87 (Routine Diagnosis Program Start Time [Hour data])
- Index 88 (Routine Diagnosis Program Start Time [Minute data])
- Index 90, Bit 1 = 1 (Backup Call Forwarding and Individual Speed Data to HDD)
- Index 306, Bit 0 = 1 (Local Phonebook data is saved) **Note 10**

<Assignment of prior data for telephone terminal display>

System Data 1
- Index 241, Bit 6 (Assign the data when the length of Caller ID exceeds the limit.) 0/1: Only the former/latter part of data appears on the LCD.
- Index 390, Bit 0 (Assign the prior data to display.) 0/1: Assigned Name/Calling Number

<Assignment of Caller ID display on the bottom line of telephone terminal>

System Data 1
- Index 390, Bit 2 = 1 (Caller ID display on the bottom line of telephone terminal LCD is in service.)

Note 10: ASYD, System Data 1, Index 306 is activated when System Data 1, Index 90, Bit 1=1 is set.

STEP 2: **ASYDL** - System Data 1, Index 683, Bit 4 = 0/1 (the maximum number of registration data for System Phonebook (12000 entries/60000 entries)).

Note: For an FCCS network, assign this system data with the ASYDN command.

Assign the system data to set Display location of contact name registered to the Desktop terminal Phonebook.

System Data 1
- Index 673, Bit 0-1

- In the case of DT750/DT830/DT730 Series

Bit 1	Bit 0	Description
0	0	Third Line
0	1	Second Line
1	0	Fourth Line
1	1	Not to display

- In the case of DT820/DT710

Bit 1	Bit 0	Description
0	0	Second Line
0	1	First Line
1	0	Third Line
1	1	Not to display

Note: To set similar indication location to DT400 Series/DT300 Series, set "Not to display."

Assign the system data for the distinction between upper case and lower case characters while searching the database for the name.
System Data 1
- Index 677, Bit 7 = 0/1 (Distinction between upper case and lower case characters is in service/out of service.) **Note 11**

Note 11: If APBKN data is assigned and the Administrator wants to change the searching method by changing ASYDL System data 1 Index 677 Bit 7, the APBKN entries must first be deleted before changing ASYDL System data 1 Index 677 Bit 7. APBKN entries must then be reentered to follow the new search method.

Assign the system data to choose the number of lines you wish to display on an LCD for Phonebook.
System Data 1
- Index 681, Bit 1 = 0/1 (4 lines/3 lines)

Note: For an FCCS network, assign this system data with the ASYDN command.

Assign the system data to set UGN data separation.
System Data 1
- Index 1136, Bit 0 = 0/1 (Individual/Common)

Note: For an FCCS network, assign this system data with the ASYDN command.

STEP 3: **ASFC** - For Stations allowed Phonebook, assign a Class of Service that allows the following.
SFI 245 (Local Phonebook) **Note 12**
SFI 246 (System Phonebook)
SFI 247 (Terminal Phonebook) **Note 13**

Note 12: When SFI 247=1 (Terminal Phonebook is allowed) and SFI 245=1 (Local Phonebook is allowed) are assigned, SFI 247 takes priority and Terminal Phonebook is effective.

Note 13: Re-registration of terminal, circuit card initialization and system initialization are required after changing data assignment.

STEP 4: **ADSL** - To activate Phonebook by the Soft key, assign Soft key as Phonebook key.

SN: 0 (Idle)
FKY: 180 (Phonebook)

STEP 5: **ADKS** - Register each Soft key pattern data to the desired telephone terminal. (This command data is necessary when using a Soft key as Phonebook key.)
TN: Tenant Number
STN: Station Number
SKP: 1-15 (Soft Key Pattern assigned by ADSL)
LKP: 1-3 (Line Key Pattern)
PG: 0/1 (Page Scroll key allocation)

STEP 6: **AKYD** - To activate Phonebook by the programmable Line/Feature key, assign the programmable Line/Feature key as Phonebook key.
TN: Tenant Number
STN: Station Number
KYN: Key Number (1-40)
KYI: 1 (Function Key)
FKY: 180 (Phonebook)

STEP 7: **ACNP** - Assign Calling Number Pattern to add the access code of incoming route.
IC/OG = IC: Incoming Call (Calling Party Recognition)
RT: External Route Number of incoming trunk.
CNP: Calling Number Pattern.
Enter a number within the range between 1 and 255.

STEP 8: **ACND** - Assign Calling Number Data to add the access code of incoming route.
CNP: Calling Number Pattern assigned in ACNP command [1-255]
SKIP: Number of Skip Digits
ADD: Number of Digits of the Additional Number [0-24]
DC: Additional Digit Information [Maximum 24 digits]

<Local Phonebook data deletion/copy/move>

CLPB - Control of Local Phone Book
Select Process Type and Parameter Type, and read out by pressing Get button or input the appropriate data. Then execute the process by pressing Execute button.

Process Type:
Local Phonebook data delete
Local Phonebook data copy **Note 14**
Local Phonebook data move **Note 14**, **Note 15**
Parameter Type:
STN: Select to delete/copy/move the data for specified Physical Station Number.
TELN(LDM): Select to delete/copy/move the data for specified Telephone Number (LDM).

TELN(NDM): Select to delete/copy/move the data for specified Telephone Number (NDM).
LENS: Select to delete/copy/move the data for specified Line Equipment Numbers.

Readout or input data:
TN: Tenant Number
STN: Physical Station Number (0-9, #, * [Maximum 6 digits])
UGN: User Group Number (1-255)
TELN(LDM): Telephone Number (0-9, #, * [Maximum 16 digits])
TELN(NDM): Telephone Number (0-9, #, * [Maximum 16 digits])
LENS: Line Equipment Numbers
Phonebook Data Num: Number of Local Phonebook Data (0-100) is displayed.

Note 14: "Phonebook data copy" and "Phonebook data move" are not available if Phonebook data exists in the destination.

Note 15: If "Local Phonebook data move" is executed, all of originally moved data will be cleared.

<System Phonebook editing>

APBKN - Assign System Phonebook data.
<To register the name and the dial number as a new entry>
TYPE: SPECIFY
TN: Tenant Number
UGN: User Group Number
NAME: Name to register [Maximum 24 characters] **Note 16**, **Note 17**
Click Get button to read out the existing information.
INPUT DC: Dial number to register [Maximum 32 digits] **Note 17**, **Note 18**, **Note 19**
After inputting the all data, click Set button.

<To edit the dial number of the existing name>
TYPE: SEARCH
TN: Tenant Number
UGN: User Group Number
SEARCH NAME: Name [Maximum 24 characters] **Note 20**, **Note 21**
After inputting the name, click Get button.

NAME: Existing name searched by SEARCH NAME [Maximum 24 characters]
INPUT DC: Dial number to register [Maximum 32 digits] **Note 17**, **Note 18**, **Note 19**
After inputting the all data, click Set button.

Note 16: Do not leave a blank space at the end of name.

Note 17: Do not register a name or a dial number if the same name or the dial number has been already registered.

Note 18: Up to 20 dial numbers (DC) can be registered in each name.

Note 19: After inputting the dial number in INPUT DC field, press “List Add” button. The number will be added to the list of DC.

Note 20: Execute SEARCH NAME in order to find the existing name from the System Phonebook database. Name search is available by a few characters in the beginning of the name if it is matched. Also the first entry of database is displayed if name search is executed without data input in this field.

Note 21: Do not leave a blank space at the 24th character position.

<Batch Assignment of System Phonebook data>

System Phone Book data can be assigned/deleted in a lump by CPBKN command with drawing up text format database. For the details, refer to Operations and Maintenance Manual.

FCCS Programming

This data setting describes only the additional or replacement commands used to activate FCCS Services. When used as an FCCS feature, this command replace the non-network command (for example, AAED is replaced by AAEDN). Note that the data setting for FCCS Service varies depending on the Node.

STEP 1: **ADSLN** - Assign Phonebook Soft key.
SN: 0 (Idle)
FKY: 180 (Phonebook)

STEP 2: **ASFCN** - For Stations allowed Phonebook, assign a Class of Service that allows the following.
SFI 245 (Local Phonebook) **Note 22**
SFI 246 (System Phonebook)
SFI 247 (Terminal Phonebook) **Note 22**

Note 22: When SFI 247=1 (Terminal Phonebook is allowed) and SFI 245=1 (Local Phonebook is allowed) are assigned, SFI 247 takes priority and Terminal Phonebook is effective.

<Character Code for Local Phonebook Display>

Available input characters, numbers, and marks for Local Phonebook are shown in the following tables.

Character type: Capital

Digit Code	1	2	3	4	5	6	7	8	9	0	*	#
Characters	.	A	D	G	J	M	P	T	W	SP	*	#
	,	B	E	H	K	N	Q	U	X	0		
	:	C	F	I	L	O	R	V	Y			
	!	À	É	4	5	Ñ	S	Ü	Z			
	?	Á	Ê			Ö	ß	8	9			
	-	Â	3			6	7					
	+	Ã										
	%	Ä										
	&	Å										
	/	Æ										
	(	Ç										
	)	2										
	=											
	¿											
	¡											
	1											

SP: Blank space

Character type: Small

Digit Code	1	2	3	4	5	6	7	8	9	0	*	#
Characters	.	a	d	g	j	m	p	t	w	SP	*	#
	,	b	e	h	k	n	q	u	x	0		
	:	c	f	i	l	o	r	v	y			
	!	à	è	ì	5	ñ	s	ù	z			
	?	á	é	í		ò	ß	ú	9			
	-	â	ê	î		ó	7	û				
	+	ä	ë	ï		ô		ü				
	%	å	3	4		Õ		8				
	&	æ				ö						
	/	ç				6						
	(	2										
	)											
	=											
	¿											
	¡											
	1											

SP: Blank space

Note: õ is displayed in capital letter. There is no small letter for this character available.

P-84 Preventative Maintenance Notification

General Description

This feature notifies the date of the periodical parts which should be replaced in the System Message.

Operating Procedure

Refer to Operations and Maintenance Manual.

Service Conditions

Refer to Operations and Maintenance Manual.

Programming

Refer to Operations and Maintenance Manual.

P-87 Physical Station Number Calling Restriction

General Description

Making a call to a Physical Station Number can be restricted by system data when a Telephone Number is assigned to a destination besides a Physical Station Number. This restriction prevents a calling party from establishing a connection with a wrong destination in an office where Physical Station Numbers and Telephone Numbers coexist.

<Example of Connection Establishment with Wrong Destination>

If a calling party dials an unregistered Telephone Number "2000" when the following data are set, a call arrives at Station A whose Physical Station Number is "200." As a result, the calling party establishes a connection with a wrong called party.

- Station A

  AISTL: STN = 200

  ALGSN: TELN = 2001

- Numbering Plan Data

  ANPD: 1stDC = 2, NND = 3

  ASPA: ACC = 2, SRV = STN

  ANPDN: 1stDC = 2, NND = 4

  ASPAN: ACC = 2, SRV = TELN

  * Telephone Number "2000" is unregistered.

Operating Procedure

No manual operation is required.

Service Conditions

<Important Notices>

1. To activate this service, Telephone Numbers are required for all stations.
2. Numbering Plan data for Physical Station Numbers assigned by such as the ANPD and ASPA commands are required even when this feature is enabled.
3. When a Physical Station Number and a Telephone Number are identical, the Physical Station Number is not restricted.

<Interactions with Other Features>

1. Calls which are not typical two-way calls are also restricted when they are dialed with Physical Station Numbers.

   Example:

   - Dialing a forwarding destination number during operations of Blind Transfer

- Ringing third party to establish a three-way call
- Dialing a forwarding destination number when Call Forwarding features such as Call Forwarding - All Calls [C-5], Call Forwarding - Busy Line [C-2], Call Forwarding - Don't Answer [C-3], Call Forwarding - PS Incoming Incomplete [C-155], and Logged Out IP Station - Call Destination [L-53] are active.
- Dialing a destination number using Automated Attendant [A-82]/Remote Access To System [R-2]
- Answering a held call on a station which belongs to Call Park Group
- Picking up a call directed to another station

When a Physical Station Number is dialed, it is equivalent to dialing an unassigned station number. So, when you use intercept features such as Call Forwarding - Intercept/Announcement [C-25] and Call Forwarding - Intercept - CCIS [C-48], those features are activated.

2. When Physical Station Numbers are assigned to the SPEED CALLING ONE-TOUCH keys, re-assignments of Telephone Numbers are required to those keys.
3. When destination numbers for Speed Calling - Station [S-21] are Physical Station Numbers, re-assignments of Telephone Numbers are required.
4. When Physical Station Numbers are registered to Terminal Phone Book or System Phone Book, re-assignments of Telephone Numbers are required.
5. This feature is also applied to Physical Station Numbers specified for sending Instant Messages.
6. When this feature is in service, Monitor Numbers assigned by the AMNO command cannot be used. The Monitor Numbers need to be assigned by the AMNOL/AMNON commands.
7. To output the billing information with SMDR, specify Telephone Numbers as numbers to be output. (ASYDL, System Data 1, Index 641, Bit 0=1)
8. To use MCI, specify Telephone Numbers as numbers to be output. (ASYDL, System Data 1, Index 641, Bit 1=1)
9. Assignments of Telephone Numbers are required for the following numbers.
   - Individual Attendant Number
   - Station Number used for Conference port for Multiple-Party Conference (PRESET, MEET-ME, PROGRESSIVE, or Add-On) using VS32
   - Station Number used for Conference port for Preset Conference and Add-On Conference using CFT card
10. When this feature is in service, using the ACSIL/ACSIN command is recommended to set the data for Direct-In Termination (DIT) [D-7]. An incoming call can arrive at a destination if the destination number is a Physical Station Number even when it is set by using the ACSI command.
11. When this feature is in service, using the AHLSN command is recommended to set the data for Hotline [H-1]. An incoming call can arrive at a destination if the destination number is a Physical Station Number even when it is set by using the AHLS command.
12. When this feature is in service, using the AHLSN command is recommended to set the data for Brokerage - Hotline [B-6]/Brokerage Hotline - Outside [B-12]. An incoming call can arrive at a destination if the destination number is a Physical Station Number even when it is set using the AHLS command.

13. Even if this feature is in service, Group Number set using the ADGN command can receive a call when a calling party dials the Group Number.
14. This feature cannot be used in the Hotel system.

<Hardware/Software>

1. When this feature is in service, you need to use a Telephone Number to log in to an IP terminal, a SIP Multiline terminal and a Soft Phone (Proprietary Protocol mode/SIP mode). Login will be failed when you use a Physical Station Number.

<Interactions with Networks>

None.

<Conditions Specific to This Feature>

1. Telephone Numbers are required for all stations although you can still set/get Physical Station Numbers through PCPro commands. So be sure to set Telephone Numbers for all stations to activate this feature.

   Examples of Assignment Requiring Telephone Numbers:

   - Assignment of Station Data by the ASDT command
   - Assignment of Key Data for Desktop terminal by the AKYD command
   - Assignment of Station Hunting Group - Circular by the ASHC command
   - Assignment of Call Forwarding Data by the ACFO command

2. You need to use Telephone Numbers for some parameters of the following commands.

   - Assignment of Call Forwarding Station Data by the ACFS/ACFS_T command

     Set a Call Forwarding Destination Number (CFD) with a Telephone Number.

   - Assignment of Change Digit Code for Dial in Service by the ACDD/ACDDL/ACDDN command

     Set a Change Digit Code (CDC) with a Telephone Number when a station number is set as a destination of incoming Direct Inward Dialing (DID) call.

   - Assignment of MAC address of IP Enabled Dterm for LDM by the AMAC command

     Select TELN for TYPE and set a Telephone Number (TELN).

3. When this feature is in service, a MAC address cannot be set by specifying a Physical Station Number (STN) of an IP terminal. Set the MAC address by specifying a Telephone Number (TELN).

<Other Conditions>

None.

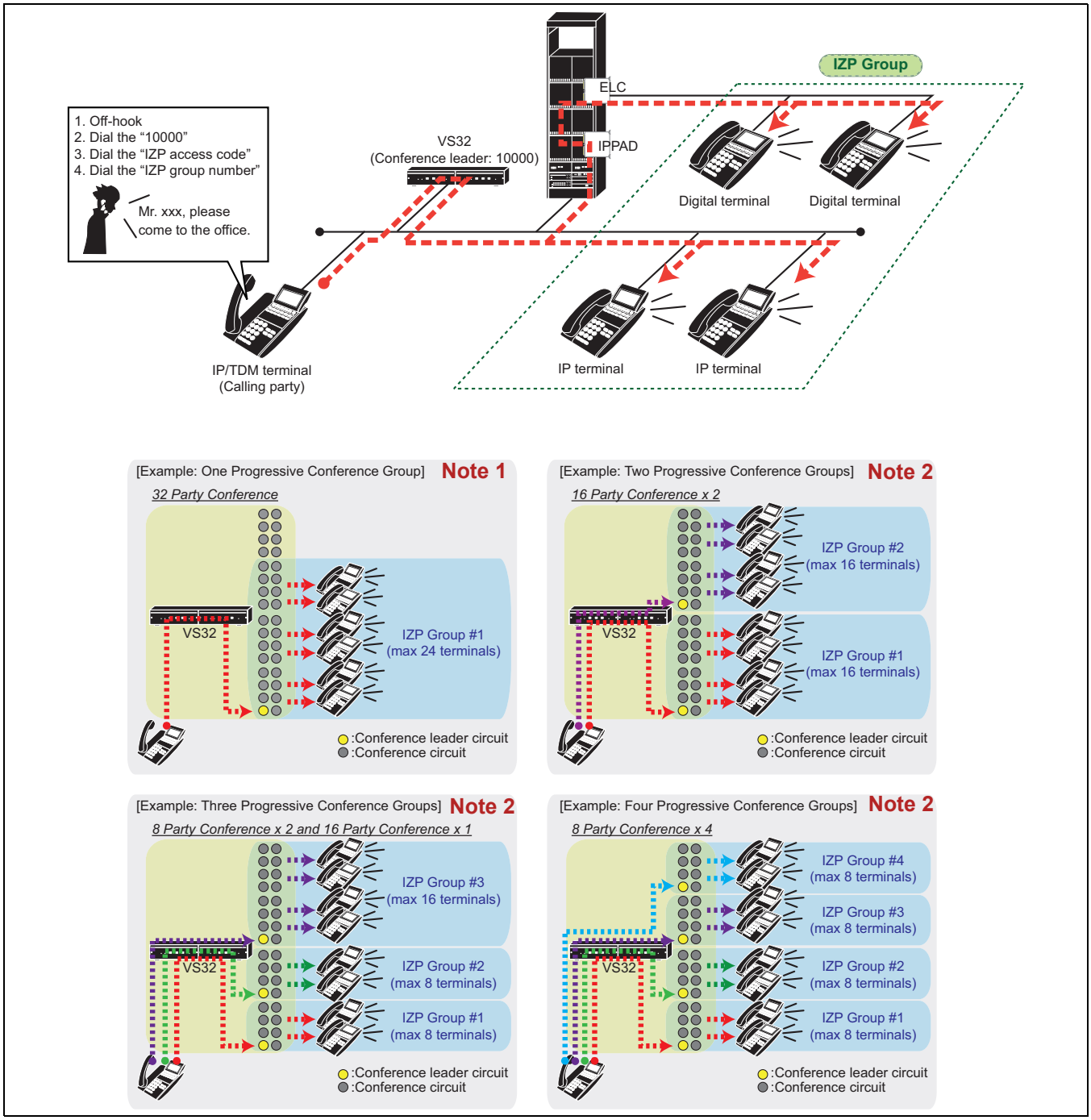
Programming

STEP 1: **ASYDL/ASYDN** - System Data 1, Index 1144, Bit 0=1 (Physical Station Number Calling Restriction is in service)

P-89 Progressive Paging - VS32

General Description

This feature uses INTERNAL ZONE PAGING [I-21] in conjunction with the progressive conference feature of VS32 CONFERENCE SERVER [V-26] to provide up to 8, 16 or 24 Desktop terminals in a paging group.



Note 1: When using 32 Party Progressive Conference group with this feature, you are still limited to a maximum of 24 simultaneous voice sessions, regardless if all 32 channels are assigned or not.

Note 2: If you are using multiple VS32 Progressive Conference groups for IZP, the Conference Leader (Pilot Number) of each conference can be assigned in a Hunt Group (separate from the Conferee ports). This allows the calling party to call one Pilot Number and have the PBX hunt for an idle VS32 Progressive Conference; otherwise the calling party must call the Pilot Numbers of each Progressive Conference individually until an idle group is found. Generally the hunt group would consist of Conference Groups of the same size.

Operating Procedure

To originate

1. Lift the handset. Receive Dial Tone.
2. Initiate a call to the Progressive Conference pilot number on the VS32.
3. Dial the IZP Access Code and the IZP Group Number.
4. Announce the call over the built-in speakers on the Desktop terminals in the PROGRESSIVE PAGING - VS32 Group.

To answer

1. Press the Answer Key or the Line Key.
2. If the Line Key was pressed, go off-hook on the handset or press the Speaker key.

Note: The Answer Key can be used if Line Preference Service via AKYD is in service (ASYD Command, System Data 1, Index 79, Bit 0= 1.)

<Conditions for this Feature>

1. Up to a maximum of 24 stations (IP and TDM) are supported in a group.
2. A VS32 can have multiple Progressive Conference groups. Depending on assignment, the maximum number of paged stations may differ (still cannot exceed 24 in one IZP Group). IZP member assignments by the AIZP command may need to be limited to 8/16/24 parties.
3. A two-way call is established on the VS32 after a paging call is answered. All conference ports are occupied until the 2 way call is released and the Conference Leader's extension is released.
4. When Progressive Conference is used, * and # cannot be used as an Access Code for Internal Zone Paging.
5. There must be an IP-PAD channel available for every Digital terminal assigned in an IZP Group or the Conference Leader will hear a busy tone after dialing the IZP Group Number (even if the IP Stations are idle).
6. The VS32 Conference Ports must be assigned to a higher numbered LENS than any of the IP Stations that use this feature. Also, the LENS must be consecutive and normally begins with Level-0 of the Line Group.
7. The My Line and Prime Line must be the same Station Number in AKYD for this feature to operate correctly (for paged stations). Therefore, a TEC-18 or a TEC-3 is not supported as a Prime Line with this feature.

<Interactions with Other Features>

1. All Service Conditions of the progressive conference feature of VS32 CONFERENCE SERVER [V-26] and INTERNAL ZONE PAGING [I-21] are applicable.

2. Do not assign INTERNAL ZONE PAGING – MULTI-ZONE [I-49] when using this feature.
3. This feature is not supported with Off-Hook Suppression (set in the AKYD command, parameter S=1).
4. Headset Mode is not compatible with this feature.
5. Due to the Prime Line and the Off-Hook Suppression limitations, ACD Positions are not supported.
6. Using one VS32 Progressive Conference to add stations belonging to separate Zone Intercom Groups is not supported.
7. Conference Leader port can be a member of a Pilot Hunt Group.
8. The IZP Key from AKYD (FKY85) cannot be used with this feature.

<Hardware/Software>

1. This feature is supported on the VS32 in Proprietary Protocol Mode only.

<Programming Related Conditions>

1. When IP stations are used, assign System Data 1, Index 679, Bit 1= 1 (Multiple Session for SIP Multiple Line Terminals: In Service). After the setting, a re-registration must be performed.
2. The answer method of a paging call is determined by ASYD System Data 1, Index 388, Bit 4.
3. Voice codec of each paging channel must be the same single codec, so paged terminals should accept common codec. Location ID/Codec setting should be considered.
4. Stations belonging to IZP Groups that use this feature must be allowed SFI 145 (Internal Zone Paging Answer) to avoid potential phone lock-ups.
5. The STN registered with the AIVSL command registers a dummy station with a unused extension number to invite the paged extension.

Interactions

When a Station answers a Voice Call using this feature, the call becomes Station-to-Station Connection Status. Accordingly, various hooking services are available.

Programming

Before starting this programming, assign the data for INTERNAL ZONE PAGING [I-21] and the progressive conference feature of VS32 CONFERENCE SERVER [V-26].

STEP 1: **ASYD** - System Data 1, Index 27, Bit 0-3: Tone Kind

Note that the following data is available when SFI 71 = 1 in the ASFC command.

DATA	TONE	DATA	TONE
0	Dial Tone (DT)	8	-
1	Special Dial Tone (SPDT)	9	Sender Transmitting Tone (SDTT)
2	Ringing Back Tone (RBT)	A	Call Waiting Ringing Back Tone (CWRBT)
3	Consecutive Ringing Back Tone (CRBT)	B	-
4	Busy Tone (BT)	C	-
5	Reorder Tone (ROT)	D	-
6	Service Set Tone (SST)	E	-
7	Second Dial Tone (SDT)	F	-

STEP 2: **ASYD** - System Data 1, Index 79, Bit 0: 0/1 = Line Preference Service via AKYD Out of service/In service.
System Data 1, Index 321, Bit 7: 0/1 = Voice Call Chime 4 chime/1 chime.
System Data 1, Index 388, Bit 4: 0/1= Answer the INTERNAL ZONE PAGING call by pressing Prime Line key after going off-hook/by going off-hook.

STEP 3: **ASYDL** - System Data 1, Index 679, Bit 1: 1 (Multiple Session for SIP Multiple Line Terminals: In Service). **Note 3**
System Data 1, Index 804, Bit 3 (Designation of the maximum digit number of INTERNAL ZONE PAGING Group Number) 0/1 = two/three digits.

STEP 4: **ASFC** - Assign the following SFI to the SFC for the VS32 Progressive Conferee Ports:
SFI 32 = 1: Line Circuit Reverse Relay Control
SFI 71 = 1: Distinctive Dial Tone
SFI 72 = 1: Called Station Switch Hook Flash Restriction
SFI 221 = 1: Stop MOH during 8/16/32 Party Conference Calls
SFI 134: IZP Calling (for stations originating an IZP)

Assign the following SFI to the SFC for stations that are to be called (those stations belonging to the IZP Groups):
SFI 145: IZP Answer (for stations answering an IZP via VS32 Progressive Conference).

Assign the following SFI to the SFC for stations that allowed to initiate a VS32 Progressive Conference and also to SFC for the Pilot Number(s) used for the VS32 Progressive Conference:
SFI 134: IZP Calling (for stations originating an IZP).

STEP 5: **AIVSL** - Assign VS32 Conference Server Related Data. **Note 4**

VS-ID: Assign the MAC Address of VS32.
CH: Assign the VS32 Conference Server channel number **Note 5**
KIND: Choose 8-party conference, 16-party conference, or 32-party conference.
CONF KIND = PROGRESSIVE
TN: Tenant number
STN: Station number
LENS: Line equipment numbers
TEC = 2 (indicates PB)
RSC: Assign Route Restriction Class other than ordinary extensions use.
SFC: Assign Service Feature Restriction Class other than ordinary extensions use.

STEP 6: Set the type of Station Hunting for Conference Leader port by using either of the following commands:
ASHP - Set the type of Station Hunting to Pilot.
ASHC - Set the type of Station Hunting to Circular.

STEP 7: **ANPD** - Reserve a Number Level for Feature Access. Select Normal Service (N) or Hooking Service (H) from the combo box for Connection Indexes. Assign NND according to the predetermined Numbering Plan.

STEP 8: **ASPA** - Assign the Access Code for INTERNAL ZONE PAGING, SRV = "SSCA: Service Code Appendix", SIDA = 98. Connection Status Indexes for "N: Normal", and "H: Hooking" services.

STEP 9: **AIZP** - Assign INTERNAL ZONE PAGING Group Data.
IZPG: Assign IZP Group from 1 - 99. **Note 6**
IZPSTN: Enter the TDM Station Numbers and/or IP Station Numbers that are to be called via the VS32 Progressive Conference.

Note 3: Only when IP stations are used. After the setting, a re-registration must be performed.

Note 4: Because SIP mode is not supported, make sure that the box to select SIP mode (Use SIP Data) is unmarked.

Note 5: In the case of 8-party conference, assign the extension number to the VS32 Conference Server channel listed below:
CH#0 (Conference leader port) and CH#1 through #7 (Conferee port).
CH#8 (Conference leader port) and CH#9 through #15 (Conferee port).
CH#16 (Conference leader port) and CH#17 through #23 (Conferee port).
CH#24 (Conference leader port) and CH#25 through #31 (Conferee port).
In the case of 16-party conference, assign the extension number to the VS32 Conference Server channel listed below:
CH#0 (Conference leader port) and CH#1 through #15 (Conferee port).
CH#16 (Conference leader port) and CH#17 through #31 (Conferee port).
In the case of 32-party conference, assign the extension number to the VS32 Conference Server channel listed below:
CH#0 (Conference leader port) and CH#1 through #31 (Conferee port).

Note 6: Generally do not assign Groups1 to 4 because those are used for Multi-Zone Intercom; which is not supported with this feature.

P-90 Preference Setting - Multi-Line Telephone with PCPro

General Description

This feature adds another method for setting preferences on Desktop terminals other than using a combination of the Feature key and some other keys such as the digit keys, the Answer key, and the Originate key on each terminal. Previously, preferences on a Desktop terminal could only be set terminal by terminal, but with this feature, the settings can also be changed/deleted effectively from Telephony Server.

Note: This feature is available since FP95-114 V4 Issue 2.0.

Operating Procedure

None.

Service Conditions

<Important Notices>

1. When preference settings on an operating Desktop terminal are changed or deleted with the ATIFD command, the settings that were assigned to the terminal by a user with a combination of the Feature key and some other keys are overwritten, and the terminal is reset automatically.
2. After setting Desktop terminal preferences with the ATIFD command, execute the MEM_HDD/ MEM_HDD_N command to save the settings as One Touch Speed Calling for DtermIP.

<Interactions with Other Features>

1. To use Dynamic Dial Pad or Preset Dial, System Data assignment is required. For details, see SELECTIVE DIALING [S-162].
2. LOCATION DIVERSITY [L-55] is not supported.

<Hardware/Software>

1. This feature is available on the following Desktop terminals:
   - IP terminals
   - SIP Multiple Line terminals
2. Terminals from DT800/DT700 Series connected to DSS Console should operate under the following conditions:
   a.) The preference settings of a terminal that is paired with DSS Console become effective. Therefore, the settings of this feature to a station number of DSS Console are not used.

Note: When a terminal logs in to a station number of DSS Console, the preference settings of the DSS Console become effective. Therefore, the settings of this feature to a station number of a terminal are not used.

b.) When preference settings assigned to a station number of DSS Console are changed with the ATIFD command, the terminal that is paired with the DSS Console would be reset. LINE BUSY is displayed on the terminal.

<Interactions with Networks>

1. In the FCCS network, it is not possible to list up the data from all nodes at once. The Listup command (LTIFD) operates in the following ways:
   a.) On a Network Control Node, only the data of designated FPC is listed.
   b.) On a Local Node, only the data of self-node is listed.
2. In the FCCS network, login to a Network Control Node is required to register/delete data with Telephone Number (NDM) to a terminal in other node.

<Conditions Specific to This Feature>

1. After preference settings of a terminal are changed with the ATIFD command, if its user adds further changes to the terminal with a combination of the Feature key and some other keys, the settings assigned with the ATIFD command will be overwritten.
2. If preference settings of a Desktop terminal are deleted with the ATIFD command, the terminal goes back to its default state.
3. Any changes made with the ATIFD command are reflected after the resetting of the terminal.
4. Terminals are automatically reset after changing the preference settings with the ATIFD command. Conditions for terminal reset are the same with those for the AKYD command.

   Note: If the ATIFD command is executed to a Desktop terminal which is being used at that moment (originating, speaking, or receiving a My Line call), the terminal is reset after it becomes NOT BUSY.
   Terminals that are being switched to a headset (lamp is ON) are not automatically reset. After assigning the data, reset the terminal manually or use the DISD command.

5. The Feature key + the digit key 4 can change both sending and receiving volume settings of a handset at a time. With the ATIFD command, these volumes must be set separately.
6. Some of the values that can be assigned with the ATIFD command become invalid on some terminals. See the following table for the correspondence of functions and terminals.

X: Valid N: Invalid

Function	Terminal Operation	IP Terminal	SIP Multiple Line Terminal
Sidetone	Feature key + digit key 0	X	X
Microphone	Feature key + digit key 1	X	X
Handset Receiving Volume	Feature key + digit key 2/4	X	X

X: Valid N: Invalid

Function	Terminal Operation	IP Terminal	SIP Multiple Line Terminal
Handset Sending Volume	Feature key + digit key 4	X	X
Ringing Tone	Feature key + digit key 3	X	N
Hands-Free	Feature key + digit key 5/6	X	X
Call Indicator Lamp	Feature key + digit key 7	X	X
Dynamic Dial Pad	Feature key + digit key 8	X	X
Preset Dial	Feature key + digit key 8	X	X
Answer Preset	Feature key + Answer key	X	X
Originate Preset	Feature key + Originate key	X	X

7. If you need to program more than one Desktop terminal, use the QuickPro.

   Note: For details, see 5. Procedures to Create/Register Data, Chapter 7 IP QuickPro of PC-Pro Setup Manual.

8. When a terminal is taken over by a backup node (Geo Redundant Node/Survivable Remote Node/SR-MGC) in a Geographic Redundancy system, the terminal operates with the data that was reflected through office data copy. **Note 1 Note 2**

   Note 1: For details about office data copy, see “Office data copy” in Chapter 7, Operation and maintenance in Geographic Redundancy Configuration Manual.

   Note 2: It is possible to change the settings with the ATIFD command from a PCPro that logged in to a backup node.

<Conditions Specific to UMGi System>

There are the following conditions for UMGi system.

1. Operate from a PCPro that logged in to CCN. This feature is not available from a PCPro that logged in to an RN.
2. CCN manages preference settings that are changed with the ATIFD command or with operations conducted from a terminal that is accommodated in an RN in the normal mode.
   To retain the settings, execute the MEM_HDD command from a PCPro that logged in to CCN, and select One Touch Speed Calling for DtermIP and save.
3. Using the ATIFD command to change preference settings of terminals that are accommodated in RNs is available only when they are in the normal mode. **Note 1**

   Note 1: CCN can set preference settings on terminals accommodated in an RN that is in the survival mode, but the settings are not reflected to the terminals until they register to the RN operating in the normal mode.

4. It is possible to change preference settings from a terminal accommodated in an RN that is in the survival mode. However if the RN changes from the survival mode to the normal mode, and the terminal re-registers to the system, the preference settings go back to the state that CCN retains (the state before RN changed to the survival mode, or the state that was set with the ATIFD command whilst in the survival mode.)

<Other Conditions>

None.

Programming

STEP 1: **ATIFD** - Assignment of Individual Function Data for Terminal
Assign individual function data to a terminal.

(a) In the case where a station number is to be selected:

Station Number Type: STN (Station Number)
TN: Tenant Number (1-63)
STN: Station Number (Up to 6 digits)

[Function List]
Sidetone: 0/1=ON (Default)/OFF
Microphone: 0/1=OFF (Default)/ON
Handset Receiving Volume: 0/1=Small (Default)/Large
Handset Sending Volume: 0/1=Small (Default)/Large
Ringing Tone: 1-10=Ringing Tone 1(Default), Ringing Tone 2-10
Hands-Free: 0/1=OFF (Default)/ON **Note 1**
Call Indicator Lamp: 0/1=ON (Default)/OFF
Dynamic Dial Pad: 0/1=OFF (Default)/ON **Note 2**
Preset Dial: 0/1=OFF (Default)/ON **Note 3**
Answer Preset: 0/1=OFF (Default)/ON
Originate Preset: 0/1=OFF (Default)/ON

(b) In the case where a telephone number (LDM) is to be selected:

Station Number Type: TELN (LDM)
UGN: User Group Number
TELN: Telephone Number (16 digits maximum)
TN: Tenant Number (Display only)
STN: Station Number (Display only)

[Function List]

Sidetone: 0/1=ON (Default)/OFF
Microphone: 0/1=OFF (Default)/ON
Handset Receiving Volume: 0/1=Small (Default)/Large
Handset Sending Volume: 0/1=Small (Default)/Large
Ringing Tone: 1-10=Ringing Tone 1(Default), Ringing Tone 2-10
Hands-Free: 0/1=OFF/ON **Note 1**
Call Indicator Lamp: 0/1=ON (Default)/OFF
Dynamic Dial Pad: 0/1=OFF (Default)/ON **Note 2**
Preset Dial: 0/1=OFF (Default)/ON **Note 3**
Answer Preset: 0/1=OFF (Default)/ON
Originate Preset: 0/1=OFF (Default)/ON

(c) In the case where a telephone number (NDM) is to be selected:

Station Number Type:	TELN (NDM)
UGN:	User Group Number
TELN:	Telephone Number (16 digits maximum)
FPC:	Fusion Point Code (Display only)
TN:	Tenant Number (Display only)
STN:	Station Number (Display only)

[Function List]

Sidetone: 0/1=ON (Default)/OFF
Microphone: 0/1=OFF (Default)/ON
Handset Receiving Volume: 0/1=Small (Default)/Large
Handset Sending Volume: 0/1=Small (Default)/Large
Ringing Tone: 1-10=Ringing Tone 1(Default), Ringing Tone 2-10
Hands-Free: 0/1=OFF (Default)/ON **Note 1**
Call Indicator Lamp: 0/1=ON (Default)/OFF
Dynamic Dial Pad: 0/1=OFF (Default)/ON **Note 2**
Preset Dial: 0/1=OFF (Default)/ON **Note 3**
Answer Preset: 0/1=OFF (Default)/ON
Originate Preset: 0/1=OFF (Default)/ON

Note 1: To use Hands-Free, set Microphone to 1 (ON).

Note 2: To enable Dynamic Dial Pad, assign System Data (ASYD, SYS1, Index 384, Bit 6=1), and set Preset Dial to 0 (OFF).

Note 3: To enable Preset Dial, assign System Data (ASYD, SYS1, Index 1256, Bit 1=1), and set Dynamic Dial Pad to 0 (OFF).

STEP 2: **MEM_HDD/MEM_HDD_N** - Data Control Between Memory and HDD/Data Control Between Memory and HDD for Fusion Network
Back up the data assigned with the ATIFD command to the hard disk.
Direction Select: Memory to Hard Disk
Data Type Select: One Touch Speed Calling for DtermIP

<To delete the setting>

STEP 1: **ATIFD** - Assignment of Individual Function Data for Terminal
Delete individual function data of a terminal.
STEP 2: **MEM_HDD/MEM_HDD_N** - Data Control Between Memory and HDD/Data Control Between Memory and HDD for Fusion Network
Back up the data assigned with the ATIFD command to the hard disk.
Direction Select: Memory to Hard Disk
Data Type Select: One Touch Speed Calling for DtermIP

Q-9 Quick Transfer VMM
(Series 8800 software feature)

General Description

This feature permits a Desktop terminal or Attendant Console to quickly transfers a call to another station's call Forwarding - Don't Answer destination.

Note: This feature is available in North America only.

Operating Procedure

To quickly transfer a call to another Station's Call Forwarding - Don't Answer destination from a Dterm:

1. Receive a call and answer.
2. Press the TRANSFER key. Dial the second telephone number.
3. Receive Ringback Tone and press the CALL REDIRECT key.
4. The second telephone stops ringing and the call is forwarded to its Call Forwarding - Don't Answer destination. If the destination is VMM, the call is forwarded to its VMM.

To quickly originate a call to another Station's Call Forwarding - Don't Answer destination from a Dterm:

1. Dial the Station Number.
2. Receive Ringback Tone and press the CALL REDIRECT key.
3. The second telephone stops ringing and the call is forwarded to its Call Forwarding - Don't Answer destination. If the destination is VMM, the call is forwarded to its VMM.

To quickly transfer a call to a Station's Call Forwarding - Don't Answer destination from an Attendant Console:

1. Receive a call and answer.
2. Dial the second telephone number.
3. Receive Ringback Tone and press the START key.
4. The second telephone stops ringing and the call is forwarded to its Call Forwarding - Don't Answer destination. If the destination is VMM, the call is forwarded to its VMM.

Service Conditions

1. Quick transfer VMM can be performed by a Dterm or a Attendant Console.
2. Refer to the service conditions and limitations of CALL REDIRECT [C-173].
3. This feature also works for multiple hop Call Forwarding - Don't Answers.
   a.) The Quick Transfer VMM key must be pressed for each multiple forward.
4. Quick Transfer VMM does not support a direct call from an Attendant Console to a Station and then using Quick Transfer VMM.
   a.) For the Attendant Console to use Quick Transfer VMM it must have a held party.
5. This feature uses the normal Call Forwarding - Don't Answer call processing for a call when the Function key is pressed.
   a.) If the call would not forward after the timer was completed it will not forward for Quick Transfer VMM either.
   b.) Refer to CALL FORWARDING - DON'T ANSWER [C-3], CALL FORWARDING DON'T ANSWER - OUTSIDE [C-74] for service conditions and limitations.

Programming

STEP 1: **ASYDL** - System Data 1, Index 803, Bit 5: 0/1 = CFD with hold party is not available for Attendant Console/CFD with hold party is available for Attendant Console

STEP 2: **AKYD** - Assign Call Redirect key.
KYI: 1 FKY: 132

STEP 3: **ADSL** - Assign Call Redirect Soft key.
SKP: 0 SN: 2 FKY: 132

R-2 Remote Access to System

General Description

This feature allows a user Outside the System to access the system via an exchange Network Connection without Attendant or Station assistance. The Outside user may originate calls over any or all of facilities, such as WATS, FX, TIE Line or CCSA.

When a wrong Code is received from a Station/Trunk, detailed information on the unauthorized user is output as System Message [26-M] Toll Fraud Report.

Operating Procedure

To operate, using AUTHORIZATION CODE [A-20]/FORCED ACCOUNT CODE [F-7]

1. Lift the handset. Receive Dial Tone.
2. Dial the Remote Access number to connect to the System. Receive Ringback Tone through public network.
3. If there is an Announcement Service, an announcement will be heard. Receive Dial Tone.
4. Dial the AUTHORIZATION CODE/FORCED ACCOUNT CODE. Receive Dial Tone.
5. Dial the desired Telephone Number. Hear Ringback Tone.
6. The Called Party answers.

To operate without using AUTHORIZATION CODE [A-20] / FORCED ACCOUNT CODE [F-7]

1. Lift the handset. Receive Dial Tone.
2. Dial the Remote Access number to connect to the System. Receive Ringback Tone through the public network.
3. Receive Dial Tone.
4. Dial the desired Telephone Number. Hear Ringback Tone.
5. The Called Party answers.

Service Conditions

1. When a call terminates from a TIE Line, this feature is available by dialing a specific Telephone Number.
2. After a call terminates to AUTOMATED ATTENDANT [A-82], AUTHORIZATION CODE [A-20] and FORCED ACCOUNT CODE [F-7] services are available.
3. AUTHORIZATION CODE [A-20] is predetermined on a system basis. Refer to AUTHORIZATION CODE [A-20] for more information.
4. Announcement Service is not provided when AUTHORIZATION CODE [A-20] service is not available.
5. Until the call is connected to an Announcement Trunk, Ringback Tone is heard.
6. Announcement Trunks are to be used for Single Connections only. Announcement Service is not available when the Announcement Trunks are all busy or are not provided. (The caller receives Dial Tone.)
7. When a call terminates to the Called Station, the following services are available:

CALL FORWARDING - ALL CALLS	[C-5]
CALL FORWARDING - BUSY LINE	[C-2]
UCD	[U-1]
CALL WAITING	

8. The second announcement is not provided to the ISDN Trunk.
9. The announcement is provided for a maximum of 30 seconds. Therefore, the message recording should be completed within 30 seconds.

Interactions

1. When using the Authorization Code in conjunction with REMOTE ACCESS TO SYSTEM [R-2], consider the following points:

   - Authorization Codes will not be recorded in SMDR. System Data 2, Index 3, Bit 5 is the only difference between Forced Account Code and Authorization Code. This Bit determines where the dialed Code will appear in SMDR. When assigned as Authorization Code, those digits will not appear in SMDR for this feature.

   - The FORCED ACCOUNT CODE [F-7] feature is available even when some type of access is restricted by the Authorization Code Restriction option.

   - If you want to set a tandem connection restriction, do not use the ARRC command but use the AATC and ARSC commands, as follows:

     - In the AATC command, define Authorization Code Restriction by setting the ACR option to "Restricted" and specifying Route Restriction Class (RSC).

     - In the ARSC command, assign route restriction information to the Route Restriction Class you specified above.

   - When you set the ACR option to "Allowed," an Authorization Code removes access restrictions on the user, regardless of the setting of the ARSC or ARRC command.

2. The additional step of the ARAC command, for DID Trunks does not eliminate the need for Forced Account Code assignments. In addition, Remote Access to System [R-2] via DID requires the use of Forced Account Codes [F-7].
3. The System recognizes a momentary pressing of the switch hook as a disconnect by the remote party. Therefore, all features activated via a Connection Index of Hooking (H) are not available to the REMOTE ACCESS TO SYSTEM feature user. A remote access user may participate in a feature that requires a switch hook flash generated by an internal Station.
4. STATION MESSAGE DETAIL RECORDING (SMDR) [S-10]-REMOTE ACCESS TO SYSTEM Outgoing Calls are recorded in SMDR. The field in the SMDR report that lists the Calling Station will list the route and Trunk Number.
5. SPEED CALLING SYSTEM [S-3] and SPEED CALLING-STATION/INDIVIDUAL [S-21, 23] may not be accessed via REMOTE ACCESS TO SYSTEM [R-2].
6. If the REMOTE ACCESS TO SYSTEM feature is used to Call Stations internal to the system, allow in ARSC the Incoming Remote Access route to the RSCs of the Station that may be called.
7. See AUTHORIZATION CODES [A-20] for Check/Sum method assignments.

Programming

STEP 1: **ASYD** - System Data 1, Index 43, Bit 0. Is Authorization Code required with REMOTE ACCESS TO SYSTEM? 0/1: Yes/No. If this is assigned as data "1", then no assignment in AATC and AMND is required.

System Data 1, Index 243. Timer Counter of the forced release for a Tandem Connection to the Trunk not providing Answer signal. (Data 00H = 64 seconds.)

System Data 2, Index 3, Bit 5 must be assigned as data "1", FORCED ACCOUNT CODE [F-7].

System Data 1, Index 91, Bits 2 and 3 must be assigned as data "0" (all system messages are output).

System Data 1, Index 435, Bit 4: Enable Toll Fraud Report. 0/1: Out of Service/In Service. Assign data as "1" if 26-M system Message are required.

STEP 2: **AATC** - Define an Authorization Code for a Tenant and determine what type of access is allowed or restricted by the code:

TN: Tenant Number

DC: Authorization Code

ACR: Authorization Code Restriction

Choose "Restricted" or "Allowed" based on application. If you want to keep some functionality access restricted, choose "Restricted" and specify the following parameters:

RSC: Route Restriction Class **Note 1**

SFC: Service Feature Class **Note 1**

The remote access line will be route restricted accordingly. The SFC parameter is not operational.

Note 1: Only available when "Restricted" is chosen for the ACR parameter.

STEP 3: **ARTD** - Assign the Remote Access route as shown below:

Central Office TRUNK:

RT: 1

3-ISGS	:1	4-INSG	:2/3	5-TF	:2	6-TCL	:1	7-L/T	:1
8-RLP	:2	10-SMDR	:0/1	38-FA	:1				

Note: To change the data for existing route (RT), initialize or reboot the trunk device such as a circuit card or MG.

DIRECT INWARD DIAL (WINK START):

RT: 1

3-ISGS	:7	4-INSG	:2/3	5-TF	:2	6-TCL	:1/4	7-L/T	:1
8-RLP	:2	10-SMDR	:0/1	15-LSG	:8	38-FA	:1		

Note: To change the data for existing route (RT), initialize or reboot the trunk device such as a circuit card or MG.

STEP 4: **AMND** - For Tenant 0, assign the Destination Code (DC) as the first digit of all Forced Account Codes and the Maximum Necessary Digits (MND) of the Authorization Code. The MND value will include the digits added by the Check Sum method. (Only required if AATC is assigned.)

STEP 5: **ATRK** - Assign the LENS, Remote Access Trunk Route Number, Trunk Number, and Tenant Number.

STEP 6: **MBTK** - Assign the Make Idle status to the REMOTE ACCESS TO SYSTEM Trunks.

STEP 7: **ACSI** - Assign External Route Number, Trunk Number, and Connection Service Index (CSI) = 5, Remote Access to Node, Incoming DDD Ringdown Trunks are automatically connected as a Remote Access Trunk. They do not use ARAC Access Codes.

STEP 8: **ARAC** - This command is used when a TIE Line or DID line will serve as a Remote Access to the Telephony Server line. A DID Call will pulse into the system with digits that resemble a Station Number. That Station Number is assigned as the REMOTE ACCESS TO SYSTEM Access Code.

STEP 9: **ARRC** - Allow Trunk-to-Trunk Connection between the Remote Access route and to those outgoing routes this feature will access. Assign using ARI-D-RES, Direct Connection.

STEP 10: **ACFR** - Allow the types of Incoming Calls listed in the Call Class Index (CCI) LDN, CCSA, that will be connected to Remote Access. REMOTE ACCESS TO SYSTEM is Transfer Service Feature Index (TSFI) number 7.

STEP 11: **ASTD** - Assign the state translation data: STM = 4, SYS = 0, ST = 1 (Remote Access to System is in Service).

FCCS Programming

This data setting describes only the additional or replacement commands used to activate FCCS Services. When used as an FCCS feature, these commands replace the non-network commands (for example, AAED is replaced by AAEDN). Note that the data setting for FCCS Service varies depending on the Node.

STEP 1: **ARTDN** - Assign the Remote Access route as shown below:

Central Office TRUNK:

RT: 1

3-ISGS	:1	4-INSG	:2/3	5-TF	:2	6-TCL	:1	7-L/T	:1
8-RLP	:2	10-SMDR	:0/1	38-FA	:1				

DIRECT INWARD DIAL (WINK START):

RT: 1

3-ISGS	:7	4-INSG	:2/3	5-TF	:2	6-TCL	:1/4	7-L/T	:1
8-RLP	:2	10-SMDR	:0/1	15-LSG	:8	38-FA	:1		

STEP 2: **ARRCN** - Allow Trunk-to-Trunk Connection between the Remote Access route and to those outgoing routes this feature will access. Assign using ARI-D-RES, Direct Connection.

Note: When using ASYDL or ASYDN to program an FCCS feature, the basic programming of ASYD is required in addition to ASYDL or ASYDN.

R-3 Reserve Power

General Description

This feature provides a system with a backup power supply, functioning from a battery source, in the event of a commercial power failure.

Operating Procedure

Power is automatically supplied to the System in the event of a commercial power failure.

Service Conditions

1. Batteries should be installed separate from the System.
2. Batteries and chargers are not NEC supplied.
3. No interruption of the System operation will occur during switch-over from commercial to battery power.
4. Duration of battery operation is a direct function of the capacity of batteries equipped.

Programming

No programming is required.

R-4 Restriction from Outgoing Calls

General Description

This feature automatically denies preselected Station Lines within the System the ability to place Outgoing Calls and/or certain miscellaneous Trunk Calls without Attendant assistance.

Operating Procedure

Operation is fully automatic that the restricted Stations will receive Reorder Tone upon attempting to place a restricted call.

Service Conditions

1. This feature is programmable via System Data programming from the PCPro.
2. Non-restricted Stations within the system can CALL TRANSFER - ALL CALLS [C-11] Outgoing Calls to the Stations restricted by this feature.
3. Only the Attendant can permit the restricted Stations to place Outgoing Calls via either of the following features:

   PASSING DIAL TONE [P-2]
   NON-DELAY OPERATION [N-3]
4. Refer to CLASS OF SERVICE - INDIVIDUAL [C-15] and DAY/NIGHT CLASS OF SERVICE [D-15] for further details.
5. Station Line access to a given Trunk Group is controlled by assigning a Route Restriction Class (RSC) to each line. There can be a maximum of 15 different RSCs (RSC1 - 15) per system, each of which can be programmed to restrict an assigned Station from none, one, or more than one Trunk Group.
6. Two outgoing Trunk Group restriction tables are provided per system. The Direct Dial Access table is indexed when a Station user dials the Trunk Group Access Code and the ATTENDANT CONSOLE [A-3] table is indexed on Attendant completed calls. The restriction options per RSC can differ per table, thus, allowing Attendant completion of Outgoing Calls for the restricted Stations.
7. CALL TRANSFER - ALL CALLS [C-11]: If a restricted Station user calls another Station, the Called Station cannot use the Call Transfer feature to connect the Calling Station to a Trunk Group to which the Calling Station is restricted from Outgoing Calls. If transfer is attempted, the controlling Station is recalled.
8. TIE LINE ACCESS [T-3]: If the Station user accesses a TIE Trunk to a distant Office, access to the public exchange network can only be controlled by the distant Office.
9. THREE-WAY CALLING [T-2]: If a restricted Station user calls another Station, the Called Station cannot add-on an Outside exchange party via a Trunk Group in which the Calling Station is restricted from Outgoing Calls. An attempt to add-on (SHF) will be ignored.
10. NIGHT SERVICE: Restrictions can be changed for a given Station when the system is switched to the night mode of operation.

Interactions

1. Transferring restrictions may not be desired for a restricted Station.
2. CALL TRANSFER-ALL CALLS [C-11]: Station A is allowed access to outgoing on Route 1. Station B is restricted access for outgoing on Route 1 via RRIs 2 and 3. The unrestricted Station A makes an Outgoing Call on route 1 and cannot use the CALL TRANSFER-ALL CALLS feature to connect Station B to Route 1. If CALL TRANSFER-ALL CALLS is attempted, Station A is immediately recalled.
3. CONSULTATION HOLD [C-17]: Station A may consult with Station B when Route 1 is on Soft Hold.
4. THREE-WAY CALLING [T-2]: Station A cannot include restricted Station B in a Tree-Way Calling Connection with Route 1.
5. DAY/NIGHT CLASS OF SERVICE [D-15]: Restricted Station B may be allowed access to Route 1 when the system is placed in the NIGHT mode. See DAY/NIGHT CLASS OF SERVICE [D-15].
6. RESTRICTION FROM OUTGOING CALLS are constructed on a per-route restriction basis. Two Route Restriction Indexes are constructed: RRI 2, Outgoing Calls via the Attendant Console and RRI 3, Outgoing Calls via Direct Dial Access.

Programming

STEP 1: **ASYD** - System Data 1, Index 75. Should the following commands be developed for Day/Night? 0/1: No/Yes.

Bit 0- Develop ARSC command for Day/Night?

Bit 3- Develop ATDP command for Day/Night?

System Data 2, Index 1, Bits 4 and 5. Should a Toll-Restricted Station be routed to the Attendant Console or receive Reorder Tone?

Reorder Tone: Bit 4 = “0”, Bit 5 = “0”.

Attendant Console: Bit 4 = “0”, Bit 5 = “0”.

STEP 2: **ARSC** - Sixteen different Route Restriction Classes may be constructed to either allow, deny, or toll restrict access to the various routes of the system. Assign the RSC using Route Restriction Indexes 2 (outgoing via the Attendant Console) and 3 (Outgoing Direct).

STEP 3: **ASDT** - Assign Tenant Number, Station Number, LENS, and the Telephone Equipment Class (TEC) to the Station. This is used in regular Business systems. The Service Feature Restriction Class (SFC) and Route Restriction Class (RSC) are also assigned to the telephone.

TN: Tenant Number
STN: Station Number
LENS: Line Equipment Numbers (6 digits)
TEC: Telephone Class (1-31)
1: DP (10 PPS)
2: PB
3: DP/PB
12: Dterm
13: Data terminal Via Dterm
14: Hotline
15: CAS Line
16: Data Terminal Via Data Module
18: Virtual Circuit
23: ISDN Terminal
27: Eight Conference Equipment
RSC: Route Restriction Class (0-15) for assignment of RSC, ARSC command.
SFC: Service Feature Restriction Class (0-15) for assignment of SFC, ASFC command.

FCCS Programming

This data setting describes only the additional or replacement commands used to activate FCCS Services. When used as an FCCS feature, these commands replace the non-network commands (for example, AAED is replaced by AAEDN). Note that the data setting for FCCS Service varies depending on the Node.

Note: When using ASYDL or ASYDN to program an FCCS feature, the basic programming of ASYD is required in addition to ASYDL or ASYDN.

ARSCN - Sixteen different Route Restriction Classes may be constructed to either allow, deny, or toll restrict access to the various routes of the system. Assign the RSC using Route Restriction Indexes 2 (outgoing via the Attendant Console) and 3 (Outgoing Direct).

R-5 Rotary Dial Calling

General Description

This feature permits Station users to originate calls over rotary or pushbutton lines using Rotary Dial telephones.

Operating Procedure

Users dial rotary telephones in the normal fashion.

Service Conditions

1. Rotary Dial telephones installed should be Industry Standard 10 PPS instruments.
2. Pushbutton instruments may be used in conjunction with ROTARY DIAL CALLING.
3. It is possible to provide only the ATTENDANT CONSOLE [A-3] with Pushbutton Calling, while all other Stations are assigned ROTARY DIAL CALLING. Refer to PUSHBUTTON CALLING - ATTENDANT ONLY [P-5] for further details.
4. If ROTARY DIAL CALLING is assigned to a line, only Rotary Dial sets can be used on that line.
5. POWER FAILURE TRANSFER [P-3]: Rotary Dial telephones must be used as POWER FAILURE TRANSFER [P-3] Stations if the serving central office facility is not arranged for Pushbutton Calling. A Ground Start button on the Station set is required if the central office facility uses Ground Start signaling.

Interactions

1. Rotary-to-Pushbutton conversion may be accomplished by assigning data “2” for CDN 2:ONSG in ARTD; or assigning data “1” in ASYD, System Data 2, Index 1, Bit 0, and assigning CDN 2:ONSG as data “3”, DP/PB.
2. See UNIVERSAL SENDER [U-3] for Sender programming.
3. ROTARY DIAL CALLING may be accomplished via an extension of PUSH-BUTTON CALLING [P-4] if a Telephone Equipment Class of 3 has been programmed.
4. The * or # symbols cannot be used as feature activation codes with ROTARY DIAL CALLING.
5. The Originating Registers Trunks (Route 902) can accept both rotary and Pushbutton Signals (a separate Register Group is not required).

Programming

STEP 1: **ASYD** - System Data 2, Index 2, Bit 0. Sender Signal for Station is DP only/PB? 0/1: DP only/PB only. This Bit is operative only when ARTD CDN 2:ONSG is assigned as data "3", DP/PB. Assign data "0".

STEP 2: **ASDT** - Assign Single Line Stations a Telephone Equipment Class (TEC) as rotary (DP) TEC = 1; or dial pulse/pushbutton (DP/PB), TEC = 3. Dterm sets are always pushbutton.

STEP 3: **ATRK** - Assign Originating Register Trunks (ORT) to the system in a quantity calculated from expected traffic. Registers are located on Levels 0-3 of any 8RST circuit card.

Also, senders must be assigned to the system in a quantity calculated from expected traffic. Senders are located on Levels 4-7 of any 8RST circuit card.

RT: Route Number

Trunk Route Number

Intra-office Route Number (901-931)

901 - ATT	912 - ORT for ATT
902 - ORT	913 - TCFT for ATT
903 - IRT	915 - Night ATT
905 - Sender	916 - MFCR
909 - DCFT	917 - MFCS
	919 to 926 - Modem

TK: Trunk Number

TN: Tenant Number

RSC: Route Restriction Class

SFC: Service Feature Restriction Class

STEP 4: **ARTD** - Routes must be assigned as either PB, DP, or DP/PB. Flag CDN 2:ONSG as data "1", DP, or data "3", DP/PB.

Note: To change the data for existing route (RT), initialize or reboot the trunk device such as a circuit card or MG.

R-6 Route Advance

General Description

This feature automatically routes Outgoing Calls over alternate facilities when the first-choice Trunk Group is busy. User selects the first-choice route by dialing the corresponding Access Code, and the equipment then advances through Alternate Trunk Groups only if the first choice is busy.

Operating Procedure

No manual operation is required.

Service Conditions

1. No indication is provided to a Station user as to whether the call is routed over first-choice or subsequent choice facilities.
2. STATION MESSAGE DETAIL RECORDING SYSTEM [S-10] may be used in conjunction with this feature.
3. Only Trunks of the same type may be programmed for ROUTE ADVANCE.
4. Careful consideration should be given to the use of FX Trunks as an alternate facility. These lines require outpulsing of digits for long distance which the user may not dial, because he will not know he is connected to such a facility.
5. The maximum number of routes to be included in a single Route Advance Group is 15.
6. The total number of routes that can be contained in all ROUTE ADVANCE Groups is 31.
7. The same route can be included in two or more Alternative Groups.
8. The maximum number of Trunks to be included in a route is 255.
9. No Code conversion capability is provided with ROUTE ADVANCE; therefore, the exact digits (after the Access Code) the user dials will be sent over the selected Trunk, regardless of the Trunk Group used. The user will not know what Trunk Group is selected; therefore, only those Trunk Groups that accept the same dialing format may be assigned to a given Route Advance Group.
10. ROUTE ADVANCE occurs only when the dialed Code accesses the first-choice Trunk Group in the ROUTE ADVANCE list.
11. In most cases, FX Trunk Groups to a Foreign Numbering Plan Area do not require the dialing of the FNPA Area Code. Therefore, these FX Trunks may not be used in the same ROUTE ADVANCE list with local exchange or WATS Trunks.
12. TIE Lines cannot be assigned to a ROUTE ADVANCE list that includes Central Office, FX, or WATS Trunk Groups.
13. The dialing party may be either a Station Line, ATTENDANT CONSOLE [A-3], or a TIE Trunk Line.
14. When using the FCCS Network, the Outgoing Call using a Trunk in the Remote Node is available. Therefore, Route Restriction class data must be identical in the FCCS Network. For more detailed information, see the Data Programming Manual - FCCS.
15. MG (COT) is not available for OGC/OGCA. For MG (COT), use LCR Access Code.
16. This feature can be activated via MG-SIP128/96/16. When using this feature with MG-SIP128/96/16, see service condition listed in MEDIA GATEWAY SIP [M-111] in this manual and PRI/PRA FAILSAFE ROUTING [P-49/50] in Data Programming Manual - ISDN.

Interactions

1. When OUTGOING TRUNK QUEUING [O-2] is assigned with the ROUTE ADVANCE feature, if all Trunk Groups in one ROUTE ADVANCE list are busy, the call queues on the first-choice Trunk Group. Each time the system checks for an available Trunk, only the first-choice Trunk Group is scanned; if a Trunk is not available, the call remains in queue.
2. TOLL RESTRICTION - 3/6 DIGIT [T-6] may be used in conjunction with ROUTE ADVANCE. Only routes of the same type may be used in ROUTE ADVANCE. Two differently labeled Central Office Trunks, such as, DDD and WATS may be in the same group.
3. There is no digit translation available for this feature. The dialed digits must be usable by whatever route is available.
4. This feature is available for a Station activating CONSULTATION HOLD [C-17]. However, in the following cases, ROUTE ADVANCE is not activated, and the Station user receives Reorder Tone:
   a.) when either the first-choice Trunk Group or Alternate Trunk Group is restricted by TOLL DENIAL [T-6].
   b.) when both the first-choice Trunk Group and Alternate Trunk Group are busy or restricted by TOLL DENIAL [T-6].
5. ROUTE ADVANCE is not activated on PAGING TRANSFER [P-10] or RADIO PAGING [R-16] routes. Therefore, the caller receives Reorder Tone when the first Trunk Group is busy or restricted by TOLL DENIAL [T-6].

Programming

STEP 1: **ASYD** - System Data 1, Index 65, Assign the number of routes that will be used in the system.

STEP 2: **ANPD** - Reserve a Number Level for Trunk Access. Select Normal Service (N) or Hooking Service (H) from the combo box for Connection Indexes. The Number of Necessary Digits should be one or two, depending on Numbering Plan. BLF should be Off.

STEP 3: **ASPA** - Assign the Service SRV = “OGCA: Outgoing Call with route advance”. Assign the Access Code to the lead Route Number to be seized. In the desired order, program the routes to be selected when the first choice route is busy. Count = the total number of routes in the Route Advance Group, including the first-choice route.

STEP 4: **ARSC** - Prepare a Route Restriction Class (RSC) that will allow access to all of the desired routes in the Route Advance Group for RRI 2 and RRI 3. Assign the RSC to the Stations that are allowed to access the Route Advance Group.

FCCS Programming

This data setting describes only the additional or replacement commands used to activate FCCS Services. When used as an FCCS feature, these commands replace the non-network commands (for example, AAED is replaced by AAEDN). Note that the data setting for FCCS Service varies depending on the Node.

Note: When using ASYDL or ASYDN to program an FCCS feature, the basic programming of ASYD is required in addition to ASYDL or ASYDN.

Note: For more detailed information, see the Data Programming Manual - FCCS.

STEP 1: **ASYDL** - System Data 1, Index 800. Assign the Tenant table development.

STEP 2: **ANPDL** - Reserve a Number Level for Trunk Access.

STEP 3: **ASPAL** - Assign SRV = “OGCA: Outgoing Call with route advance” and the Access Code to the lead Logical Route Number to be seized.

STEP 4: **ARSCN** - Assign the RSC to the Stations that are allowed to access the Route Advance Group.

R-16 Radio Paging

General Description

This feature allows Station users dialing access to customer-owned RADIO PAGING equipment and to selectively tone or voice/tone alert individuals carrying pocket RADIO PAGING receivers. The paged party may be connected to the paging party by going to the nearest telephone and dialing a unique Answer-back Code.

Operating Procedure

To page

1. Paging Station user dials the RADIO PAGING Access Code. Receives an acknowledgment tone from the radio equipment.
2. Paging Station user dials the Radio Number of desired party. Receives Ringback Tone from the radio equipment.
3. Paging Station user hangs up and waits for an answer signal.

To answer

- Using OGC Access Code, a paging Station accesses Radio Paging equipment whether or not the automatic answer function is provided. (System Data 1, Index 73, Bit 4 = 1 or 0.)

  1. Paged party dials the RADIO PAGING Answer Code.
  2. Paged party receives Special Dial Tone.
  3. Paged party dials his own individual Radio (beeper) Number.
  4. Paged Station is rung. The paged party receives Ringback Tone.

     Note: This operation is not available when assigning System Data 1, Index 73, Bit 4 = 1. Only Non-Delay operation is available.

  5. Paging Station is connected to the paged party.

- Using LCR/LCRS Access Code, a paging Station accesses Radio Paging equipment providing the automatic answer function. (System Data 1, Index 73, Bit 4 = 1.)

  1. Paged party dials the Paging Station Number/Individual Attendant Number which is displayed on the radio pager (beeper).
  2. Paging Station/Attendant is rung; paged party receives Ringback Tone.
  3. Paging Station/Attendant is connected to paged party.

Service Conditions

1. RADIO PAGING equipment is customer provided; refer to the manufacturer's description for the following specifications:
   a.) types of tones
   b.) capability of receiving individual Radio Numbers
   c.) capability to call the addressed Radio Terminal
   d.) maximum number of Stations that can be assigned individual radio access numbers.
2. A maximum of three digits can be assigned as access and answer codes.
3. The maximum number of digits assigned as individual Radio Numbers is three.
4. ATTENDANT CONSOLE [A-3] Attendants can activate this feature.

5. A Station that has the CALL HOLD [C-6] feature activated cannot use RADIO PAGING.
6. The maximum number of RADIO PAGING answer zones is six.
7. The maximum number of routes that can be assigned RADIO PAGING is six.
8. If the paging Station user has failed to hang up within a predetermined, programmable time interval (after activation), the Paging Connection is forcibly released.
9. If the paged party does not answer within 30 seconds, the System will automatically page that party again. If the paged party does not answer the second page within 30 seconds, the paging request will be canceled from memory. (See VARIABLE TIMING PARAMETERS [V-1]).
10. The following can be used as a paging trunk interface:
    - COT Circuit Cards
    - TLT Circuit Cards
11. Loop signaling is the recommended signaling type.
12. PAGING ACCESS [P-1], PAGING TRANSFER [P-10] and RADIO PAGING [R-16] are provided to the paging Station and the paged Station in FCCS Nodes.
13. When PAGING ACCESS [P-1]/PAGING TRANSFER [P-10]/RADIO PAGING [R-16] service is set to a Station in the Remote Node or the PAGING ACCESS [P-1]/PAGING TRANSFER [P-10]/RADIO PAGING [R-16] Call is answered from the Remote Node via FCCS link:
    a.) the Logical Route Number should be applied to each physical route that is provided the service.
    b.) the restriction of Route-to-Route Connection between FCCS Nodes should be released.
14. The same paging route cannot be used by another Station while the paged Station does not answer the page activated using the route. However, the paging activation is automatically cleared when the call is not answered within a predetermined time.
15. A user can page one Station at a time.
16. When using a FCCS network:
    a.) In case of setting the office data with LDM, a maximum of six routes per node can be provided with paging trunks, however, this service cannot be used between nodes.
    b.) In case of setting the office data with NDM, a maximum of six routes per FCCS network can be provided with paging trunks. Also, in a FCCS network, the maximum number of concurrent paging for PAGING ACCESS [P-1], PAGING TRANSFER [P-10] and RADIO PAGING [R-16] is 20.

Interactions

1. If the range of the radio page transmitter exceeds the physical limits of the System, an Incoming Central Office Call acting as a response to a Radio Page cannot be connected in Loop-to-Loop procedures. The paging party must disconnect, and the Attendant must extend the paged party's Central Office Call.
2. RADIO PAGING and PAGING ACCESS [P-1] can be provided simultaneously in the same system. Separate Access Codes for each feature are required. A RADIO PAGING Answer Code is required for each Radio Paging zone.
3. RADIO PAGING is limited to Stations or Attendant Consoles. Other parties (Central Office Trunk Calls) must have the Attendant or Station user perform the paging operation.
4. The Attendant cannot dial the RADIO PAGING answer codes.
5. BUSY VERIFICATION [B-3], EXECUTIVE RIGHT OF WAY [E-1]: These features will be denied when connected to the Radio Paging route.
6. Radio Page Answer will not follow any hunting on forwarding that the paging Station has set.

Programming

STEP 1: **ASYD** - System Data 1, Index 73, Bit 2. Is the beeper number transmitted via the RADIO PAGING equipment a two or three-digit number? 0/1: two digits/three digits.

System Data 1, Index 73, Bit 4. Radio Paging - Station Number/Individual ATT Number is automatically sent? 0/1: No/Yes.

System Data 1, Index 73, Bit 5. If the paged party doesn't respond, should the system resend the beeper number? 0/1: Yes/No. This also doubles the effective timing in System Data 1, Index 74, Bits 4-7.

System Data 1, Index 74, Bits 4-7. Recall timer for radio page. If the paged party does not respond by the first time out, the paged party is called again and the time out is repeated.

(N + 1) x 30 = Time Out

N = a hexadecimal value set in Bits 4-7.

If the paged party doesn't respond by the same period of time after the second page, the page is canceled.

System Data 1, Index 244, Bit 0. Answer by Radio Paging Answer Code. 0/1: Out/In Service. Note that this Bit is valid when System Data 1, Index 73, Bit 4 is 1 (Station Number/Individual ATT Number is automatically sent).

STEP 2: **ARTD** - Assign the Radio Paging route as shown below:

RT: 1

1-OSGS	: 2	2-ONSG	: 3	5-TF	: 1	6-TCL	: 12	7-L/T	: 1
8-RLP	: 2	9-TQ	: 1	15-LSG	: 0/1				

Note: To change the data for existing route (RT), initialize the circuit card.

STEP 3: **ATRK** - Assign the LENS, Paging Trunk Route Number, Trunk Number, and Tenant Number.

STEP 4: **MBTK** - Assign the Make Idle status to the Paging Trunks.

STEP 5: **ANPD** - Reserve a Level for Trunk Access. Select Normal Service (N) or Hooking Service (H) from the combo box for Connection Indexes. Number of Necessary Digits is usually one or two. Busy Lamp Field is not activated. Reserve a Level for RADIO PAGE Answer and Cancel Codes. Assign for Normal Service (N). Number of Necessary Digits is usually 2. Busy Lamp Field is not activated.

STEP 6: **ASPA** - Assign the RADIO PAGE Access Code. Type of Service, SRV = "OGC: Outgoing Call" or SRV = "LCR: Least Cost Routing" / "LCRS: Register Sender LCR" (available only when ASYD, System Data 1, Index 73, Bit 4 is "1"). Assign the proper Route Number that is associated with this Access Code. Assign for "N: Normal" and "H: Hooking".

Assign the Access Code in the Level reserved in ANPD above. Type of Service, SRV = PAGA (Paging Answer) and SRV = PAGC (Paging Cancel). Assign the proper Route Number that is associated with this route. Assign for a Connection Index of Normal (N).

STEP 7: **AMND** - Assign necessary maximum number of digits for Radio Paging Access and Answer. Note that MND includes the digits of Radio Number. (MND = digits of RADIO PAGING Access/Answer Code + digits of Radio Number)

STEP 8: **ARSC** - Allow RRI: 2 and 3 access to this route for the RSCs assigned to the Stations that will be given access to the paging routes.

STEP 9: **ASFC** - Assign a Service Feature Restriction Class that allows SFI 37 to Stations that will have RADIO PAGING answer capability.

STEP 10: **ASAT** - Assign the Individual Attendant Number, if necessary.

FCCS Programming

This data setting describes only the additional or replacement commands used to activate FCCS Services. When used as an FCCS feature, these commands replace the non-network commands (for example, AAED is replaced by AAEDN). Note that the data setting for FCCS Service varies depending on the Node.

Note: When using ASYDL or ASYDN to program an FCCS feature, the basic programming of ASYD is required in addition to ASYDL or ASYDN.

Note: The current command data shown is set to all Nodes using the service: ASTD, ARTD, ATRK, ASFC

STEP 1: **ALRTN** - This data must be assigned to the NCN. Assign the Logical Route Number to the physical route.

STEP 2: **ANPDN** - This data must be assigned to the NCN. Reserve a Number Level for Service Feature Access and cancel.

STEP 3: **ASPAN** - This data must be assigned to the NCN. Assign the Access Code for Paging Transfer.

STEP 4: **ARSCN** - This data must be assigned to the NCN. Release the restriction between the Announcement Trunk and the Station.

STEP 5: **ARRCN** - This data must be assigned to the NCN. Release the restriction between the terminating Trunk and the Announcement Trunk.

R-20 Restriction from Outgoing Calls - Attendant

General Description

This service permits Attendant Consoles to disable key operation by setting the Night key of each console. When this service is implemented, all origination and termination operations at the relevant Attendant Console will be disabled regardless of whether the system is in Day Mode or Night Mode.

Note: This feature is not available in North America and Australia.

Operating Procedure

To disable key operation of an Attendant Console:

1. Set the Night key of the Attendant Console to ON.

To effectuate key operation of an Attendant Console:

1. Set the Night key of the Attendant Console to OFF.

Service Conditions

1. Implementation of this feature can be selected for each system by the system data.
   When this service is not used, key operation is effective with the Night key set to ON.
2. When this service is implemented, an Attendant Console with the Night key set to ON cannot originate or answer the call even if the system is in Day Mode.

R-23 Reverted Call Metering

General Description

When the Attendant has placed a call requested by a Station to an external (exchange or TIE) line, a display of the Station Number and billing information (Metering Count) is provided on the Console after the call has ended.

Note: This feature is available in Australia only.

Operating Procedure

1. **Reverted Call Service Setting**
   a.) Non-Delay Operation
   - The Attendant answers a call from a Station.
   - The Attendant places a call as requested by the Station user.
   - The Attendant depresses the "RC" (Optional) key after the called distant office answers.

   b.) Delay Operation
   - The Attendant answers a call from a Station.
   - The Station user is released.
   - The Attendant places a call as requested by the Station user.
   - The Attendant calls the Station user after the called distant office answers.
   - The Attendant depresses the "RC" key either before or after the Station user answers.

2. **Answering Reverted Call Recall**
   a.) A reverted call ends and the distant office is released.
   b.) The Attendant depresses either the "SC/RC" (SERIAL CALL RECALL) or the "ANSWER" key when "RC" incoming is indicated.
   c.) When the Attendant answers, the Station Number is indicated in the display area as well as the SRC indication.
   d.) When the Attendant depresses the "DEST" key, the indication changes from "SRC" to "DEST" and the metering count value is displayed in the display area.
   e.) By operating the "SRC/DEST" key, the Attendant has a choice of display between the Station Number and the metering count value.
   f.) The Attendant may use either the "RELEASE" or the "CANCEL" key to be released.

Service Conditions

1. This feature is available in Australia only.
2. This feature is effective only for Outgoing Calls placed to external lines via the Attendant. Note, however, that it cannot be set for a call that is terminated to the Attendant by means of CALL FORWARDING - INTERCEPT/DON'T ANSWER/BUSY LINE [C-18, 3, 2] or RECALL.
3. This service is not effective for Outgoing Calls placed to CCIS line via the Attendant.
4. The feature is actually set when the Attendant presses the "SC/RC" key, that is, after the called distant office answers in the case of NON-DELAY OPERATION [N-3] or the Calling Station answers upon being called back by the Attendant in the case of DELAY OPERATION [D-20].
5. As soon as this feature is set for the call, the established Connection is released from the Loop and all hooking services become effective at the Station.
6. Once this feature is set, it cannot be canceled.

7. In a NON-DELAY OPERATION [N-3], if the Attendant presses the “SC/RC” key before the called distant office answers, RBT is sent to the Calling Station and the ATTENDANT CONSOLE [A-3] enters the idle state.
8. When this feature is set in the manner described in 5 above, if the Calling Station party is released before the Called Party answers, no recall will be made to the Attendant.
9. In a DELAY OPERATION [D-20], if the Attendant presses the “SC/RC” key before the Station answers, RBT is sent to the Called Party at the distant office and the ATTENDANT CONSOLE [A-3] enters the idle state.
10. In a DELAY OPERATION [D-20] performing Non-delay mode service setting, the Attendant is not released from the Loop until the Calling Station answers, provided that the associated System Data is set to “Loop Release” (System Data 2 Index 2 Bit 5 = 1). In this case, the Attendant cannot reenter the Connection before the Station party answers, though the service once set remains.
11. CALL PICK-UP service cannot be used upon answering a call for which the REVERTED CALL METERING [R-23] service is set in the Non-Delay setting mode.
12. CALL FORWARDING-DON'T ANSWER [C-3] cannot be set for any call to which REVERTED CALL METERING [R-23] is set in the Non-Delay setting mode.
13. Recall Incoming Connection is indicated to the Attendant by means of an “SC/RC” lamp and buzzer indication after the distant office is released.
14. Either the “SC/RC” answer key or the “ANSWER” key is used by the Attendant to answer a REVERTED CALL METERING Recall Incoming.
15. REVERTED CALL METERING recall service is performed in the normal way in the event that the system switches from Day mode to Night mode after the service is set.
16. The metering count information is indicated as a number of up to four digits. Any count value exceeding 9999 will be displayed in the overflow indication mode.
17. The metering count value indicates “0” in any of the following circumstances:
    a.) The Calling Station is not granted this billing facility.
    b.) The called office is not granted this billing facility.
    c.) Connection is made over a line to which metering pulses are not sent out.
18. Even if the call is transferred to another Station by the Station party who originally requested the call, the same Station Number remains displayed regardless of the billing system (Total/Separate).
19. In the case of the separate billing system also, the metering count value is the total number of pulses counted.

Programming

STEP 1: **ASYD**

- SYSTEM DATA1

< Index6 >

Bit 6 (ATT CON Key Arrangement Pattern): Assign 2 Hexadecimal.

< Index186 >

Bit 4 (Serial Call-Loop Release): 0/1 = Out/In Service.

- SYSTEM DATA2

< Index15 >

Bit 6-7 (Reverted Call Metering Service): 00/01 = Out/In Service.

Assign 01.

R-27 Route Restriction - Announcement

General Description

This feature permits a Station-originated Call, dialed to a restricted outgoing number, to be automatically routed to a recorded announcement informing the caller that the Dialed Number is restricted. (TOLL RESTRICTION [T-7])

Operating Procedure

No manual operation is required after this feature has been programmed in System Data.

Service Conditions

1. This feature is available when a Station-originated Call encounters a restriction.
2. Access to the announcement of this feature is restricted by MISCELLANEOUS TRUNK RESTRICTION [M-3].
3. The Announcement Machine can be provided on a per Tenant basis.
4. The availability of this service cannot be determined for each Originating Route.
5. TOLL DIVERSION [T-6] takes precedence over this feature.
6. Digital Announcement Trunk (DAT) and the Calling Station must be accommodated in the same Node within the FCCS Network.

Programming

STEP 1: **ASYD** - System Data 2, Index 1, Bit 4, 5. Toll Restriction (disposition of Restricted Call) Assign Bits 4 and 5 as "0".

STEP 2: **AAED** - Equipment type 7 = Outgoing Route Restriction Announcement

Also see ANNOUNCEMENT SERVICE [A-59] for programming of announcement and MISCELLANEOUS TRUNK RESTRICTION [M-3] for Trunk restriction programming.

R-37 RS-464A PAD

General Description

This service is for changing the current PAD control system for the lines and external Trunks to a PAD control system which conforms to the RS-464A specification.

Note: This feature is available in North America only.

The following table outlines the comparison between the current PAD control and the RS-464A PAD control.

Calling Terminal	Called Terminal	RS-464A PAD Control	The Current PAD Control
ONS	ONS	6dB PAD ON	5dB PAD ON
ONS	OPS	3dB PAD ON	PAD OFF (0dB)
OPS	ONS	3dB PAD ON	PAD OFF (0dB)
OPS	OPS	PAD OFF (0dB)	PAD OFF (0dB)
ONS	ATT	6dB PAD ON	PAD OFF (0dB)
OPS	ATT	3dB PAD ON	PAD OFF (0dB)
ONS	Trunk	Data assigned to ARTD, PAD	Data assigned to ARTD, PAD
OPS	Trunk	Data assigned to ARTD, PAD2	Data assigned to ARTD, PAD
ATT	Trunk	Data assigned to ARTD, PAD	Data assigned to ARTD, PAD

Note: ONS: Line interface to On-Premises Line (General Telephone)
OPS: Line interface to Off-Premises Line (Long Line Telephone)

When using the 24CCT/24PRT, APAD and ARTD, PAD 2 are ignored when switch settings are set to ARTD fixed, and PAD patterns 1 and 2 are selectable by switch settings.

Note: In the following tables, -db indicates a decrease in volume Level; +db indicates an increase in volume.

PAD PATTERN 2			PAD PATTERN 2			PAD PATTERN 2		
16 COTBE with SFI 50 = 0			16 COTBE with SFI 50 = 1			16 COTBE with APAD Setting		
ARTD PAD	Transmit PAD	Receive PAD	ARTD PAD 2	Transmit PAD	Receive PAD	APAD Setting	Transmit PAD	Receive PAD
0	0db	0db	0	ARTD	PAD	1	0db	0db
1	0db	0db	1	0db	0db	2	-3db	-3db
2	-3db	-3db	2	-3db	-3db	3	-6db	-6db
3	-6db	-6db	3	-6db	-6db	4	0db	+6db
4	0db	+6db	4	0db	+6db	5	Unused	Unused
5	0db	0db	5	0db	0db	15	Pad Off	Pad Off
6	Unused	Unused	6	Unused	Unused			
7	0db	0db	7	0db	0db			

TLT (DID) with SFI 50 = 0			TLT (DID) with SFI 50 = 1			TLT (DID) with APAD Setting		
ARTD PAD	Transmit PAD	Receive PAD	ARTD PAD 2	Transmit PAD	Receive PAD	APAD Setting	Transmit PAD	Receive PAD
0	0db	0db	0	ARTD	PAD	1	-3db	-3db
1	-3db	-3db	1	-3db	-3db	2	-6db	-6db
2	-6db	-6db	2	-6db	-6db	3	-6db	+6db
3	-6db	+6db	3	-6db	+6db	4	0db	+6db
4	0db	+6db	4	0db	+6db	5	Unused	Unused
5	0db	0db	5	0db	0db	15	Pad Off	Pad Off
6	Unused	Unused	6	Unused	Unused			
7	0db	0db	7	0db	0db			

In 4 Wire E+M Mode, the TLT can set PAD values using switch settings and ARTD and APAD values.

In the following table, P0 and P1 are assigned by setting ARTD to a value of 0, CDN 30:PAD and setting values by using Switch 1 (P0) and Switch 2 (P1). P2 and P3 are assigned by setting ARTD/APAD to a value of 5, and PAD values are set by using Switch 3 positions 1, 2, 3 and 4 (P2) and positions 5, 6, 7 and 8 (P3).

TLT (2W E+M) with SFI 50 = 0			TLT (2W E+M) with SFI 50 = 1			TLT (2W E+M) with APAD Setting		
ARTD PAD	Transmit PAD	Receive PAD	ARTD PAD 2	Transmit PAD	Receive PAD	APAD Setting	Transmit PAD	Receive PAD
0	0db	0db	0	ARTD	PAD	1	-3db	-3db
1	-3db	-3db	1	-3db	-3db	2	-6db	-6db
2	-6db	-6db	2	-6db	-6db	3	-6db	+6db
3	-6db	+6db	3	-6db	+6db	4	0db	+6db
4	0db	+6db	4	0db	+6db	5	Unused	Unused
5	0db	0db	5	0db	0db	15	Pad Off	Pad Off
6	Unused	Unused	6	Unused	Unused			
7	0db	0db	7	0db	0db			

TLT (4W E+M) with SFI 50 = 0			TLT (4W E+M) with SFI 50 = 1			TLT (4W E+M) with APAD Setting		
ARTD PAD	Transmit PAD	Receive PAD	ARTD PAD 2	Transmit PAD	Receive PAD	APAD Setting	Transmit PAD	Receive PAD
0	P0	P1	0	ARTD	PAD	1	-3db	-3db
1	-3db	-3db	1	-3db	-3db	2	-6db	-6db
2	-6db	-6db	2	-6db	-6db	3	-12db	-11db
3	-12db	-11db	3	-12db	-11db	4	-16db	-11db
4	-16db	-11db	4	-16db	-11db	5	P2	P3
5	P2	P3	5	P2	P3	15	Pad Off	Pad Off
6	Unused	Unused	6	Unused	Unused			
7	0db	0db	7	0db	0db			

Operating Procedure

The following Connections are performed:

STATION-TO-STATION CONNECTION
STATION-TO-ATT CONNECTION
CONNECTION VIA ATT
OUTGOING Central Office LINE CONNECTION FROM STATION
OUTGOING Central Office LINE CONNECTION FROM ATT

Service Conditions

1. RS-464A PAD control is performed for lines and external Trunks.
2. ONS (General Telephone) and OPS (Long Line Telephone) are identified by the SFC of the Station concerned.
3. An Attendant Console is regarded as an ONS. The Attendant Console will follow the PAD values assigned in ARTD.
4. If PAD data is assigned by APAD command, this data overrides other PAD data assigned by ARTD.
5. PAD settings for the TLT can be set using ARTD/APAD or a combination of switch settings and ARTD/APAD.
6. This service is used for each of the following Connections:
   STATION-TO-STATION CONNECTION
   STATION-TO-ATT CONNECTION
   CONNECTION VIA ATT
   OUTGOING Central Office LINE CONNECTION FROM STATION
   OUTGOING Central Office LINE CONNECTION FROM ATT
7. If the impedance of the Trunk does not match the impedance setting on the Trunk, the PAD values will not match the values in the previous tables.

Programming

Assignment of Telephone:

STEP 1: **ASDT** - Assign the Station data.

STEP 2: **ASFC** - Assign either ONS or OPS to the SFC of the Station.
SFI = 50, RES: 0/1 = ONS (General Telephone) /OPS (Long Line Telephone)
Stations that have SFI 50 in ASFC set to a 0, will follow the PAD setting in PAD 2 of ARTD.
Stations that have SFI 50 in ASFC set to a 1 will follow the PAD set in PAD 2. PAD 2 settings override the values set in PAD.

STEP 3: **ASYD** - System 1, Index 4, Bit 6 affects Station-to-Station Calls only. (0 = 6dB, 1 = 0dB)

Assignment of External Trunk:

STEP 1: **ARTD** - Assign Route Class Data for the Route Number of the External Trunk.
When assigning Route Class Data, assign PAD control data between ONS and External Trunk to CDN 30:PAD or CDN 32:PAD2.
Assign expansion Route Class Data for the Route Number of the External Trunk.
When assigning expansion Route Class Data, assign PAD control data between OPS and External Trunk to PAD2 (CDN 32).
When PAD 2 is not assigned, the value assigned in PAD will be used.

Note: To change the data for existing route (RT), initialize the circuit card.

STEP 2: **ATRK** - Assign Trunk data of External Trunk.

STEP 3: **MBTK** - Cancel the make-busy of the External Trunk assigning by ATRK.

STEP 4: **APAD** - Assign the PAD control data for Tandem Connection, Station-to-Trunk Connections, and Outgoing/Incoming Connection. When assigning APAD for Station-to-Station Connections, the OGPAD parameter has no effect when assigned (only the ICPAD parameter is used).

R-49 Ringer Mute

General Description

This feature allows a Station user to stop ringing by pressing the associated Feature key on the Desktop terminal.

Note: This feature is not available in Australia.

Operating Procedure

To operate

1. The Desktop terminal user presses the RINGER MUTE key. The ringing is muted and the LED of the key lights. RING OFF is displayed for three seconds.
2. The Desktop terminal user presses the RINGER MUTE key again. The ringing is recovered and the LED of the key is turned off. RING ON is displayed for three seconds.

Service Conditions

1. The Feature key for Ringer Mute must be assigned to the Desktop terminal.
2. Ringer Mute which has been set becomes ineffective upon system initialization, individual initialization of LP or initialization of PM.
3. While this service is activated, ringing for Call Back is not provided as well.
4. This service has no effect on the status of DO NOT DISTURB [D-11] service. For example:
   a.) When Do Not Disturb is set, the Station is in the Do Not Disturb status.
   b.) When Ringer Mute is set, the Station is in the Do Not Disturb and Ringer Mute status.
   c.) When Do Not Disturb is reset, the Station is in the Ringer Mute status.
5. When this service is activated, ringing for any call is muted.
6. If a Station user presses the RINGER MUTE key while a call is terminating, ringer for this call will be muted.
7. (Condition for DtermIP) Ringer Mute needs to be set each time the registration of DtermIP is performed. By assigning "1" to ASYDL, SYS 1, Index 824, Bit 0, DtermIP can maintain the Ringer Mute setting even after the Registration.
8. While this service is activated on SIP Multiple Line terminals, a particular tone is generated instead of the ringing for Call Hold and Call Back, etc.
9. The setting of ASYD, SYS1, Index 448, Bit 3 is also available for non-ACD-agent-position Desktop terminals in the Do Not Disturb or Ringer mute mode when receiving a line call.

Programming

STEP 1: **AKYD** - For Desktop terminal sets, RINGER MUTE may be assigned to a programmable Line/Feature key. Assign Type:Function:RING MUTE [117].

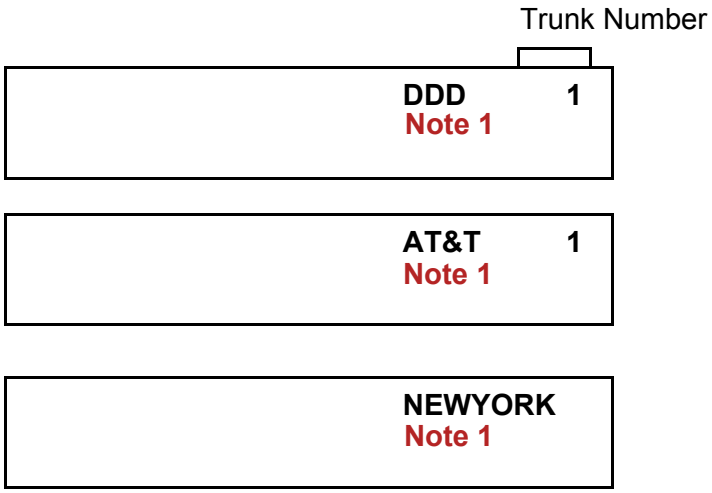
STEP 2: **ADSL** - Assign SN = 0 (Idle) and FKY = 117 (Ringer Mute: RG-OF).
This command is used for assigning the function to each Soft key.

R-51 Route Name Display

General Description

This feature allows a user to assign a desired Trunk type for a route by the System Data. The assigned Trunk type (maximum of 4 or 8 digits) is displayed on the upper line of Dterm LCD.

- Four-digit display



- Eight-digit display

Note 1: Trunk type can be changed by System Data.

Operating Procedure

No manual operation is required.

Service Conditions

1. Trunk types are assigned a maximum of 4 or 8 digits (depending on System Data) using ASCII characters.
2. The Trunk type assigned by this feature is not displayed when a call terminates from the ISDN Trunk or when a Station is connected to the call from/to the ISDN Trunk.
3. The Trunk type assigned by this feature is displayed only when a call terminates to a Station or when a Station is connected to the call. In case of the call through a CCIS line, the assigned Trunk type is not displayed. If a Station is connected to the Trunk user through the CCIS, ROM data corresponding to the Trunk Class of the route is displayed. If a Station is connected to a Station belonging to a different office within the CCIS network, the Telephone Number of the opposite Station is displayed.
4. The Trunk type assigned by this feature cannot be displayed at the I-terminal and PCS.
5. This feature is not applicable to the Attendant Console.
6. The current Caller ID Services and Calling Party Number - Name Assignment and Display [C-157] service have priority over this feature.
7. When a call terminates from the Trunk which has been assigned the Central Office line number, the LCD display is as shown below:

- Four-digit display



ABCD: Trunk type assigned by this feature.
123: Trunk Number preassigned.

- Eight-digit display



ABCDEFGH: Trunk type assigned by this feature.

8. This feature is available when the Dterm user presses the call terminating Sub Line. This feature is also available when the user presses the held Sub Line after holding a call using the HOLD Key.
9. For data assignment to each of NDM (Network Data Memory) and LDM (Local Data Memory), different commands can be used. However, the data in NDM has a priority. The logical route must be assigned to NDM, and the physical route must be assigned to LDM.

Programming / FCCS Programming

This data setting describes only the additional or replacement commands used to activate FCCS Services. When used as an FCCS feature, these commands replace the non-network commands (for example, AAED is replaced by AAEDN). Note that the data setting for FCCS Service varies depending on the Node.

Note: When using ASYDL or ASYDN to program an FCCS feature, the basic programming of ASYD is required in addition to ASYDL or ASYDN.

STEP 1: **ASYDL** - System Data 1, Index 512, FPC of the Self Node.
System Data 1, Index 513, LDM usage.
System Data 1, Index 514, NDM usage.

Note: This data is required for the FCCS Network.

STEP 2: **ADRTN/ADRTL** - Assign the Route Name data on a route basis.
LGRT/RT: Logical Route or Physical Route Number
TYPE: A maximum digit for the Route Name data
1/2 = four digits / eight digits
DISP: Displayed Data

Note: When the system is used within an FCCS Network, assign the Logical Route Number via LGRT parameter of ADRTN. When the system is not used within an FCCS Network, assign the Physical Route Number via RT parameter of ADRTL. ADRTN command data has priority over the ADRTL command data. Route Name data programmed by ADRTN is displayed at all FCCS Nodes in the network.

R-54 Remote Call Forwarding Control

General Description

This feature allows a Station user to set/cancel the following forwarding features from the target Station by dialing the identification (ID) Code preset for the forwarding Station.

- CALL FORWARDING - ALL CALLS [C-5]
- CALL FORWARDING - BUSY LINE [C-2]
- CALL FORWARDING - DON'T ANSWER [C-3]
- SPLIT CALL FORWARDING [S-99]

These forwarding features can also be set/canceled from an Outside Station via Central Office/TIE Line.

Operating Procedure

To set Station ID Code of the Forwarding Station

1. Lift the handset. Receive Dial Tone.
2. Dial the Access Code for setting Station ID Code. Receive Special Dial Tone.
3. Dial the desired Station ID Code (a maximum of ten digits). Receive Service Set Tone. **Note 1**, **Note 2**

Note 1: Dial a desired Station ID Code (in the range of 1-10 digits). If the number of digits for Station ID Code is less than 10, dial "#" after the Station ID Code. Dialing "#" implies the completion of Station ID Code.
"#" or "*" can be used as a character which indicate the completion of Station ID Code (according to the setting of system data).

Note 2: When Inter digit timer expires, the dialed number becomes invalid (the user hears ROT).

To cancel Station ID Code of the Forwarding Station

1. Lift the handset. Receive Dial Tone.
2. Dial the Access Code for setting Station ID Code. Receive Special Dial Tone.
3. Dial "#" or "*". Receive Service Set Tone. **Note 3**

Note 3: The system data (ASYDL/ASYDN, Index 895, Bit 4) determines whether to use "#" or "*" as a character which indicates the completion of Station ID Codes.

To set from the Target Station

1. Lift the handset. Receive Dial Tone.
2. Dial the Remote Call Forwarding Control Access Code.

The following shows examples of how Remote Call Forwarding Control Access Code is dialed.

1234: Remote Call Forwarding Control Access Code
5678: Automated Attendant Number/Remote Access Number

- Dial Remote Call Forwarding Control Access Code to forward a call from a party in the same node.

  Example: 1234

- Dial Trunk Access Code for accessing the forwarding node + office code of the forwarding node + Remote Call Forwarding Control Access Code to forward a call from a party in a different node via TIE line

  Example: 8-26-1234

- Access the forwarding system using REMOTE ACCESS TO SYSTEM [R-2]/AUTOMATED ATTENDANT [A-82] and then dial Remote Call Forwarding Control Access Code to forward a call from a party via an analog C.O. line or ISDN line.

  Example 1 (with announcement): 0471-85-5678 → <Announcement (RBT → 2nd DT)> → 1234
  Example 2 (without announcement): 0471-85-5678 → (RBT) → (2nd DT) → 1234

- Access the forwarding system using REMOTE ACCESS TO SYSTEM [R-2]/AUTOMATED ATTENDANT [A-82] and then dial Remote Call Forwarding Control Access Code to forward a call from a party via Dp Channel TIE line, Q-SIG line, or SIP network.

  Example: 8-26-5678 → (RBT) → (2nd DT) → 1234

3. Receive Ringback Tone. About six seconds after, receive Special Dial Tone.
4. Dial the forwarding Station ID Code. Receive Special Dial Tone.
5. Dial the Forwarding Station Number. Receive Dial Tone.
6. Dial the Access Code or press the Feature key for setting the desired forwarding feature. Receive Special Dial Tone.
7. Dial the Target Station Number (A maximum of 12 digits). Receive Service Set Tone.

To cancel from the Forwarding Station

1. Lift the handset. Receive Dial Tone.
2. Dial the Cancel Code or press the Feature key for canceling the forwarding feature. Receive Service Set Tone.

To cancel from the Target Station

(This procedure can also be applied when canceling the forwarding feature which has set at the Forwarding Station.)

1. Lift the handset. Receive Dial Tone.
2. Dial the Remote Call Forwarding Control Access Code.

   The following shows examples of how Remote Call Forwarding Control Access Code is dialed.

   1234: Remote Call Forwarding Control Access Code
   5678: Automated Attendant Number/Remote Access Number

   - Dial Remote Call Forwarding Control Access Code to forward a call from a party in the same node.

     Example: 1234

   - Dial Trunk Access Code for accessing the forwarding node + office code of the forwarding node + Remote Call Forwarding Control Access Code to forward a call from a party in a different node via TIE line

     Example: 8-26-1234

- Access the forwarding system using REMOTE ACCESS TO SYSTEM [R-2]/AUTOMATED ATTENDANT [A-82] and then dial Remote Call Forwarding Control Access Code to forward a call from a party via an analog C.O. line or ISDN line.

  Example 1 (with announcement): 0471-85-5678 → <Announcement (RBT → 2nd DT)> → 1234
  Example 2 (without announcement): 0471-85-5678 → (RBT) → (2nd DT) → 1234

- Access the forwarding system using REMOTE ACCESS TO SYSTEM [R-2]/AUTOMATED ATTENDANT [A-82] and then dial Remote Call Forwarding Control Access Code to forward a call from a party via Dp Channel TIE line, Q-SIG line, or SIP network.

  Example: 8-26-5678 → (RBT) → (2nd DT) → 1234

3. Receive Ringback Tone. About six seconds after, receive Special Dial Tone.
4. Dial the Forwarding Station ID Code. Receive Special Dial Tone.
5. Dial the Forwarding Station Number. Receive Dial Tone.
6. Dial the Cancel Code or press the Feature key for canceling the forwarding feature. Receive Service Set Tone.

Service Conditions

1. ID Code entry is restricted when the button "#" is not pressed as the last digit.
2. When setting a new ID Code, the previous ID Code is automatically erased.
3. The list of ID codes cannot be output.
4. ID Code data is saved by the PCPro command or the routine diagnosis program for saving Call Forwarding data.
5. Each target Station must be a PB (DTMF) telephone.
6. This feature is restricted when:
   - a wrong ID Code is dialed from the Target Station.
   - the forwarding features from the designated Station are restricted by Service Feature Restriction Class.
7. When changing a target Station Number, the former setting must be canceled in advance.
8. When the forwarding Station is in one of the following conditions, this service cannot be activated.
   - Busy
   - Lockout
   - Make Busy
   - Not assigned
   - Logout (IP Station)
9. The target Station must not have any other call placed on hold when setting this feature.
10. When this feature is set from an Outside Station, DID [D-7] Connection is required.
11. This feature can be activated via REMOTE ACCESS TO SYSTEM [R-2] or AUTOMATED ATTENDANT [A-82].

    Note: To forward a call via SIP network, analog C.O.line Dp Channel TIE line, or Q-SIG line, use REMOTE ACCESS TO SYSTEM [R-2]/AUTOMATED ATTENDANT [A-82] to access the system accommodating the forwarding station.
12. This feature can be set from IP Station to IP Station. However the following are not available:
    - from non-IP Station to IP Station
    - from Trunk to IP Station

13. The following shows service conditions concerning Station ID Code.
    a.) Up to 10 digits (0-9) can be dialed as a Station ID Code.
    b.) If the number of digits for Station ID Code is less than 10, dial "#" or "*" after the Station ID Code. Dialing "#" implies the completion of Station ID Code. (**Note 4**) If Inter digit timer expires before dialing "#" or "*", this feature is restricted.

    Note 4: "#" or "*" can be used as a character which indicate the completion of Station ID Code (according to the setting of system data).

    c.) Station ID Codes cannot be displayed or listed by PCPro. Therefore, user needs to manage their Station ID Codes by themselves.
    d.) While CALL FORWARDING I'M HERE is being set, the dialed numbers are not displayed on the terminal (displayed as "****"). However, while Station ID Codes are being set, the user can choose to hide or show them by setting of SFI 137 (ASFC/ASFCN).
    e.) Station ID Codes are shared between this feature and REMOTE CALL FORWARDING CONTROL [R-54]
    f.) Even if User Authentication is enabled (ASFC/ASFCN SFI 243=1), this feature can be set without dialing Station ID Code if no Station ID Code is set in advance.

Programming

STEP 1: **ASYD** - System Data 1, Index 79, Bit 2. Split Call Forwarding.
0/1 = Out/In Service. Assign data "1".

STEP 2: **ASYDL** - System Data 1, Index 895, Bit 4. Determines whether to use **"#"** or **"*"** as a character which indicates the completion of Station ID Codes.
0/1 = "#" is used as a character which indicates the completion of Station ID Codes/"*" is used as a character which indicates the completion of Station ID Codes.

STEP 3: **ARTD** - Assign Route Class Data of the Trunk for SPLIT CALL FORWARDING.
CDN: 68 (SCF) = 1

Note: To change the data for existing route (RT), initialize or reboot the trunk device such as circuit card or MG.

STEP 4: **ANPD** - Reserve Number Levels for each forwarding feature, ID Number Entry Access Code, and REMOTE CALL FORWARDING CONTROL Access Codes. Select Normal Service for Connection Index to each Number Level. Assign NND according to the predetermined Numbering Plan.

STEP 5: **ASPA/ASPAL** -Assign the Access Codes for the following features.

TN: Tenant Number
ACC: Access Code
SRV: "STN: Station Number" (MANUAL ROAMING)
SRV: "SSC: Service Code"
SID: 8 (Split Call Forwarding - All Calls set)
9 (Split Call Forwarding - All Calls cancel)
10 (Split Call Forwarding - Busy Line set)
11 (Split Call Forwarding - Busy Line cancel)

12 (Split Call Forwarding - Don't Answer set)
13 (Split Call Forwarding - Don't Answer cancel)

SRV: "SSCA: Service Code Appendix"

SIDA: 86 (Call Forwarding - All Calls set)
87 (Call Forwarding - Busy Line set)
88 (Call Forwarding - Don't Answer set)
89 (Call Forwarding - All Calls cancel)
90 (Call Forwarding - Busy Line cancel)
91 (Call Forwarding - Don't Answer cancel)
100 (ID Number Entry)

STEP 6: **ARAC** -Assign the REMOTE CALL FORWARDING CONTROL Access Code.

ICTN: Tenant Number of Incoming Trunk
ACC: Access Code (Max. 5 digits)
R/M: M (MANUAL ROAMING)

STEP 7: **ASFC** -Assign a Service Feature Restriction Class that allows the following SFIs to the Stations that will activate REMOTE CALL FORWARDING CONTROL.

SFI: 7 (Call Forwarding - All Calls/Split Call Forwarding - All Calls)
SFI: 8 (Call Forwarding - Busy Line/Split Call Forwarding - Busy Line)
SFI: 9 (Call Forwarding - Don't Answer/Split Call Forwarding - Don't Answer)
SFI: 137 (Authorization Code Display Elimination)
SFI: 143 (ID Number Entry)

R-55 Remote Call Hold

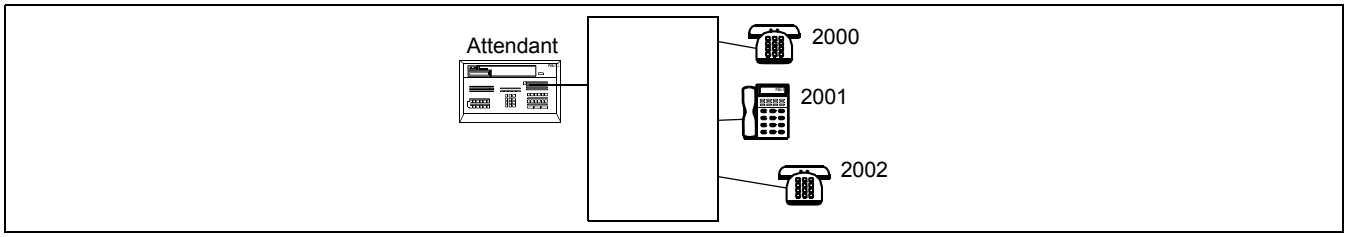
(Series 8400 software feature)

General Description

This feature provides the ability for a Desktop terminal or the Attendant Console to place a call on hold (Remote Hold), at another Desktop terminal or Virtual Extension. The original call that is placed on Remote Hold can be from an Internal Station or Trunk.

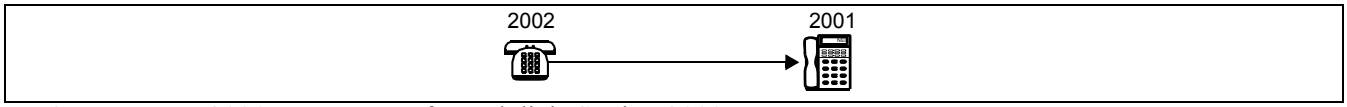
Note: This feature is available in North America only.

Operating Procedure

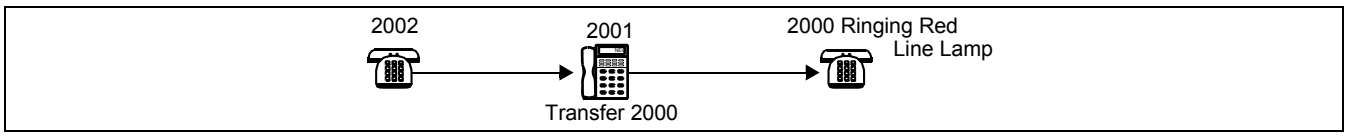


Desktop Terminal Procedure

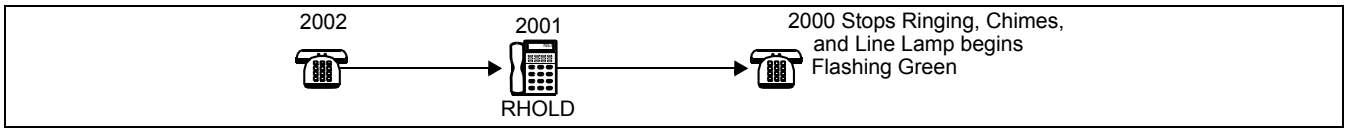
1. Station 2001 receives a call from Station 2002.



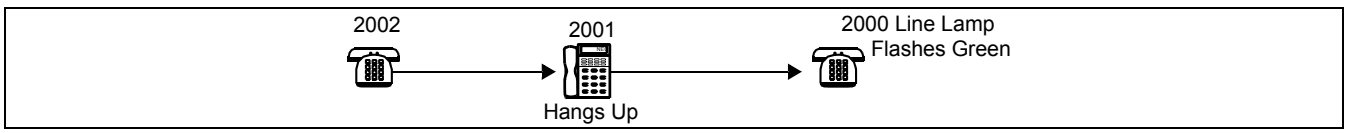
2. User at 2001 presses transfer and dials Station 2000.
3. Station 2000 begins to ring.

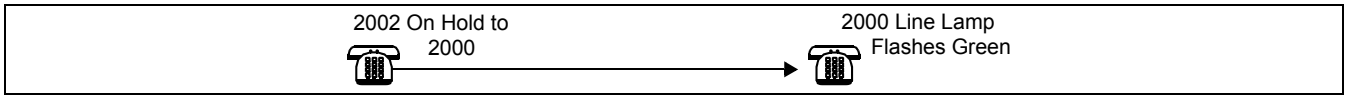


4. User at 2001 presses the REMOTE HOLD Key.

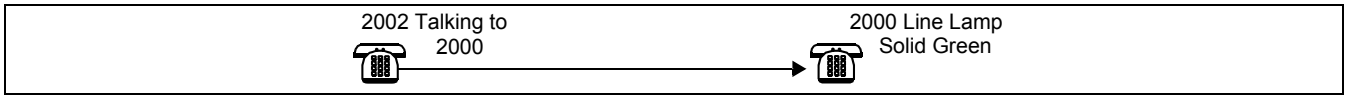


5. Station 2001 receives Service Set Tone and display shows HOLD 2000.
6. Station 2002 is placed in held state to 2000, a chime is sent to 2000 over the speaker, and the line lamp flashes green.
7. Station 2001 releases.





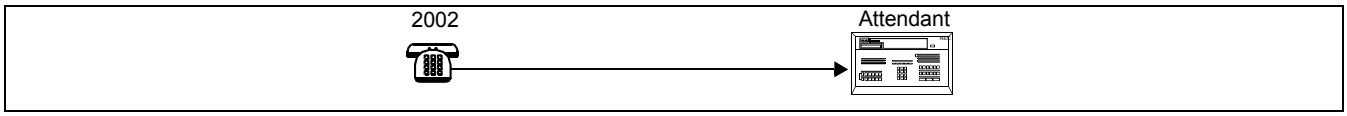
8. User can answer the call on 2000 by pressing the Line Key and going off-hook.



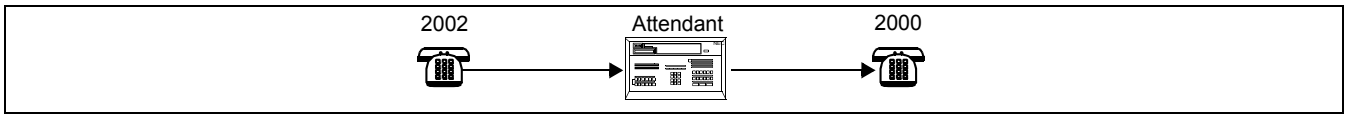
9. Normal line lamp functions now.

Attendant Console Procedure

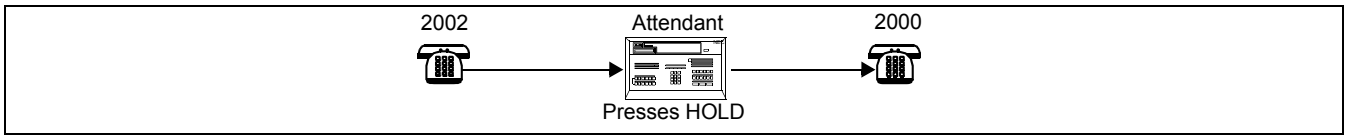
1. Attendant Console 0 receives a call from Station 2002.

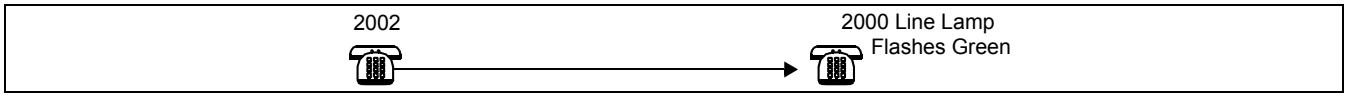


2. Attendant extends the call to 2000 by dialing 2000.

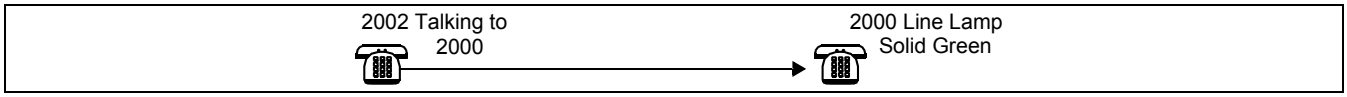


3. Station 2000 begins to ring.
4. Attendant presses the HOLD Key.





5. Station 2002 is placed in held state to Station 2000 and a chime is sent to 2000 over the speaker.
6. Attendant returns to is released and returns to idle state.
7. User can answer the call on 2000 by pressing the Line Key and going off-hook.



Trunk Call Procedure

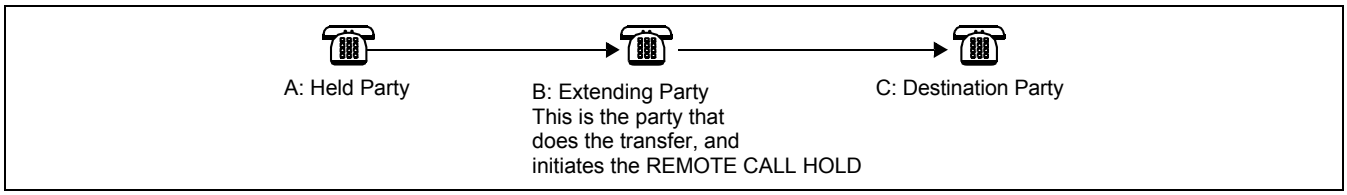
The Trunk Call procedure is the same as the procedure for the Attendant Console, but the Trunk Party will have no indication of being on hold, other than music on hold, if provided.

Service Conditions

1. The Destination (Target) of this service must be a Desktop terminal (TEC 12) or Virtual (TEC 18) Station. The destination cannot be a Trunk, Wireless Terminal, Attendant Console, Analog Station or any other type of Terminal/Connection.
2. Activation of this feature can only be from a Desktop terminal or the Attendant Console.
3. The Originating Party of the call to the party that will set REMOTE CALL HOLD may be a Station or External Trunk.
4. This feature will be set to the Ringing Party (not the Dialed Party). If Call Forwarding or Station Hunting occur, the call will be placed on hold on the Station that it is currently ringing on, not the originally dialed Station.
5. Remote Call Hold does not function over CCIS.
6. When this service is set by a Desktop terminal (after ASYDL System Data 1, Index 666, Bits 0-6 is set), the REMOTE CALL HOLD's timer will attempt to call the Station that set the feature if the timer expires.
   a.) If that Station is busy or in a state that will not allow the call to return the call, it will continue to hold at the Destination Station. The timer will be reset so the call can attempt another recall after the timer times out again. This will continue until the Calling Party disconnects.
   b.) If the Station that sets this feature is in DND when the Recall Timer expires, the call will override DND and chime the Recalled Party. This is referred to as DND Override on Recall Service.
7. When this service is set by the Attendant Console (after ASYDL System Data 1, Index 665, Bits 0-3 is set), (TC * 8 seconds) the REMOTE CALL HOLD's timer will attempt to call the Attendant Console if the timer expires.
   a.) This timer may be set to a maximum of 120 seconds (two minutes).
   b.) If ASYD System Data 1, Index 177, Bit 7 (Att Multiple Recall Service) = 1 (in service), Trunk Calls that are placed on remote hold from the Console will recall to all Consoles on the NANS Key. When this recall takes place, the call is released from the destination of the REMOTE CALL HOLD. Station Calls will recall to a Loop Key of the Attendant that placed the call on REMOTE CALL HOLD. Attendant Multiple Recall Service is not valid when a Station Call is remote held (these calls will function the same as if ASYD System Data 1, Index 177, Bit 7 = 0).
8. If ASYD System Data 1, Index 177, Bit 7 (Att Multiple Recall Service) is = 0 (out of service), all calls that are placed on REMOTE CALL HOLD will recall on a Loop Key of the Attendant that placed the call on remote hold.
   a.) If the ATT is in Night mode or position busy, the recall will override and ring on a Loop Key.
   b.) If the Destination Station answers the call before the Attendant, the Attendant will be released and the destination is connected to the Held Party.
   c.) This recall can be answered by pressing the corresponding Loop Key.
   d.) The Loop Key on the DESKCON will flash green until the Destination Party answers. When the destination answers, the Loop Key will change to solid green.
   e.) This call will remain on hold to the destination of the Remote Hold, but the line lamp will change from flashing green to flashing red.
   f.) If the Attendant presses the DEST Key, the Calling Party is placed in a Consultation Hold. If the Hold Key is pressed, the call will be placed back on REMOTE CALL HOLD. If the Cancel Key is pressed, the Destination Station becomes idle. If the Release Key is pressed, the call is transferred to the Destination Station with no change in the Destination Stations indication.
   g.) If the SRC Key is pressed, the ATT will be connected back to the Calling Party, with no change in the Destination Station.
9. When this service is activated, an audible chime indication is given to the My Line of the Remote Party, if the Remote Party is talking on a Sub Line, the chime is still sent.

a.) The My Line Lamp of the Held Party flashes green at 240 pulses per minute (ppm).
b.) All other Line Appearances of the LENS associated with the My Line of the Held Party (and on other Desktop terminals), will flash red at 60 ppm. These Line Appearances do not receive the audible chime.
c.) After the call is answered, normal line lamp control will be in place.

10. This feature cannot be used directly with Call Hold Conference.
    a.) The REMOTE CALL HOLD must be answered before it can be used with CALL HOLD CONFERENCE.
11. This feature cannot be used from the Console if Split Hold is in service (ASYD System Data 1, Index 161, Bit 3 = 1). These features are mutually exclusive.
12. This feature is not available from a Night Attendant Console.
13. Usage matrix for REMOTE CALL HOLD:



	A: HELD PARTY	B: EXTENDING PARTY	C: DESTINATION PARTY
ALLOWED	Analog station, Desktop terminal, External trunk (COT, ISDN, TIE Line, CCIS), PHS	Desktop terminal (with RHold key), Attendant	Desktop terminal (TEC 12), Virtual (TEC 18)
NOT ALLOWED	Attendant, ISDN Terminal	Analog station, PHS, ISDN Terminal	Analog station, PHS, ISDN Terminal, Attendant

Programming

Remote Hold Recall Timer for Station
STEP 1: **ASYDL** - System Data 1, Index 666, Bits 0-3 MTC/Bits 4-6 TC/MTC * TC = Timer Data

Remote Hold Recall Timer for Attendant
STEP 1: **ASYDL** - System Data 1, Index 665, Bits 0-3 MTC/eight seconds * MTC = Timer Data
STEP 2: **AKYD** - Assign Feature Key RHOLD [133] to a Function Key.
STEP 3: **ADSL** - Assign Feature Key RHOLD [133] to Soft Key Pattern for State 15.

R-57D Ringing Line Pickup -D

General Description

This feature allows Desktop terminal users to answer Incoming Calls to a Sub Line by going Off-Hook or pressing Answer Key. Line Preference data is to be set through PCPro (AKYD).

Operating Procedure

To set Answer pre-Setting Mode

1. Press FEATURE Key.
2. Press ANSWER Key.
3. ANSWER Key lights red.

To cancel Answer pre-Setting Mode

1. Press FEATURE Key.
2. Press ANSWER Key.
3. ANSWER Key is extinguished.

Service Conditions

1. This feature is available for Desktop terminals.
2. This feature can be activated only when ASYD, System Data 1, Index 79, Bit 0 = 1. Stations must be placed in Answer pre-Setting Mode in advance.
3. When Answer pre-Setting Mode is set, ANSWER Key lights red.
4. When an Incoming Call terminates to Station accommodated in the following circuit cards, ANSWER Key is extinguished (Default) or flashes red according to the System Data setting (ASYD, System Data 1 Index 385, Bit 3).

    - SP-3656 16 ELCJB PROG-E Issue 1A or later **Note 1**
    - SP-3752 16 ELCJB PROG-F Issue 1A or later

    Note 1: (Only for SP-3656 16 ELCJB PROG-E Issue 1A) When Incoming Calls terminate to Station accommodated in this circuit card, ANSWER Key always flashes red.

5. When answering Incoming Calls, ANSWER Key rights green (Default) or red according to the System Data setting (ASYD, System Data 1 Index 385, Bit 0).
6. Line Preference data for each Station is to be set by AKYD.

Programming

STEP 1: **ASYD** - System Data 1, Index 79, Bit 0. Assign as data "1" (Line Preference Service via AKYD 0/1 = Out/In Service).
System Data 1, Index 385, Bit 0. Answer Key Lamp Indication for answering incoming call. 0/1 =Lights Green/Red
System Data 1, Index 385, Bit 3. Answer Key Lamp Indication for call termination. 0/1 = Extinguished/Flash Red

Note: Initialize the ELC circuit card (by flipping its MB key OFF → ON → OFF) for the change to take effect.

STEP 2: **AKYD** - Assign Ringing Line Preference data according to the following parameters.
TN: Tenant Number
STN: Station Number
LN PRE: Line Preference Assign "1" (Line Preference data is to be set according to ANS and ORG.
ANS: Line Preference for Answer Key.
This data is only available in case of LN PRE=1.
0=Incoming Line Preference (in the order of Key arrangement)
1=Incoming Line Preference (in the order of Key arrangement)
2=Ringing Line Preference

R-58 RTP Information Output

This feature consists of the following functions:

- Call/Fault Log Collection

  (a) Call Log Collection

  - A history (includes call time, opponent, IP address, MAC address, and voice quality) per call is collected and output.
  - If a user claims against voice quality, information related to the claims can be extracted from call log to confirm and investigate its cause.

    <For Call Log Filter feature> **Note 1**

    Filtering the call logs on the Telephony Server side, the logs are more specified and help in investigating the cause for claims or fault. The following filters are available:

    - Station Filter Setting
      Output the filtered call logs by using the specified settings such as station number and its type.
    - RTP Warning Tone **Note 2**
      Output the filtered call logs that generated RTP Warning Tone.
    - Received RTP Packet Loss **Note 2**
      Output the filtered call logs of RTP Packet Loss with a count that is greater than or equal to the threshold value.

    Note 1: This feature is available since FP95-114 V4.

    Note 2: These settings are not affected by Station Filter Setting.

  (b) Fault Log Collection

  - When an event such as autonomous reset occurs on terminals, the system outputs reset factor, its time. If it caused due to packet fault, sending/receiving times, and retransmitted times, which will be collected for cause investigation.
  - If a user claims against voice quality, information related to the claims can be extracted from call log to confirm and investigate its cause.

- RTP Warning Tone

  Playback and Cancel of RTP packet communication error tone (hereinafter also called as “RTP warning tone”).

  (a) RTP Warning Tone Playback

  When the RTP packet loss reaching the predetermined threshold and the terminal cannot play back a voice due to delay of the network or packet loss during the conversation, RTP Warning Tone is generated to indicate poor voice quality.

(b) RTP Warning Tone Cancel

RTP Warning Tone can be canceled by assigning the system data. No warning tone is heard in case of RTP packet communication error when it is canceled.

- DTMF Control

Playback and Cancel of DTMF tone sending to the destination party.

- This function can choose whether to send or cancel the DTMF tone sent to the destination terminal during the conversation. More comfortable call is available by canceling the DTMF tone to be heard when the dial pad of opposite party is pressed. DTMF signal tone can be canceled on a system basis.

Call/Fault Log Collection

General Description

This function outputs log information such as the causes of poor voice quality and abnormal status of IP devices. By this feature network administrator can determine the exact cause for the problem. Call/Fault Log Collection function can be output by using SNMP Manager and PCPro.

Operating Procedure

To operate the Call/Fault Log output

1. When SNMP Manager is used
   See SNMP [S-148] in Data Programming Manual - Business.
2. When PCPro is used
   See EVLOG command described in Operations and Maintenance Manual.

To display the Call/Fault Log as system message on PCPro

1. To display the log as system message on PCPro, set ASYDL, SYS1, Index 996, Bit 2=1. For more detailed information about the system messages, refer to system messages described in Operations and Maintenance Manual.

For Terminal Claim Notification:

1. See TERMINAL CLAIM HANDLING [T-65] in Data Programming Manual - Business.

Service Conditions

1. Call Log Collection
   a.) Station Number, IP Address, MAC Address and Location ID are read out from the database of the system. And call time (start and end time), communication opponent, number of received/sent RTP, and number of received RTP NG are read out from the stations.
   b.) To report all of the information collected, connect PCPro to read out data from memory in advance or set destination point for the SNMP Manager in advance.
   c.) When a Station originates a trunk call and abandons the call while it is ringing, the call log information is output. However, if the abandoned call occurs while a Station calls another Station without using a trunk, the call log information is not output.
   d.) Note the following for Call Log Filter:
      (1) This feature is available only when the system data is available (ASYDL, SYS1, Index 999, Bit 3=1 (**Note 3**)) and one or more filtering is set by the ACLFL command.

      Note 3: After enabling this system data, the following are available by using the ACL-FL command:
      - Start/stop the Call Log Collection
      - Enable/disable the SNMP output
      - Configuration of filter settings

      (2) This feature does not work for Fault Log.

(3) Supported IP devices for the each filtering (Station Filter Setting/RTP Warning Tone/ Received RTP Packet Loss) is as follows:

X: Available for the Filter N: Not available for the Filter

IP Device	Station Filter Setting	RTP Warning Tone	Received RTP Packet Loss
IP station (TEC=12)	X	X **Note 4**	X **Note 4**
IP station (except TEC=12)	N	X **Note 4**	X **Note 4**
MG	N	X **Note 4**	X **Note 4**
Other than the above (such as VS-32)	N	X **Note 4**	X **Note 4**

Note 4: For the supported version, see the table of IP Device Call Information Output Feature For Telephony Server.

(4) Station Filter Setting works only for the following IP devices that are set as an IP station (TEC=12) by AISTL command.

Note: SIP Handler controlled SIP terminal is not supported.

(5) The conditions of the filtered log output are as follows:

- Station Filter Setting works for the call log that is sent from an available station to the Telephony Server after the call.
- Both settings of Station Filter Setting and Fault Kind are not affected by each other. When both of them are set by the ACLFL command, each filtered result is output.
- When Call Log Collect is set to START/STOP by the ACLFL command, the call log is not output if the setting is done to a call in progress.
  When Call Log Collect is set to START before the call and filtering data is set on the call by the ACLFL command, the log is output if the filtering condition is suitable.
- When the connection between Telephony Server and IP terminals/devices is disconnected, the log is not output.

(6) When the IP station data for the IP terminal that specified by Station Filter Setting is deleted by the AISTL command, the setting of Station Filter Setting is not available.

(7) As the ACLFL command data synchronizes with some system data in the ASYDL command, this feature does not work if the synchronized system data is changed accidentally by the ASYDL command.

Note: For the synchronized system data, see the ACLFL command in Command Manual.

(8) When making a call with a terminal accommodating Multiple Lines, My Line is used for Station Filter Setting.

(9) LOCATION DIVERSITY [L-55] is not supported with this feature.

(10) When using Geographic Redundancy, note the following conditions for the log output location:
- In spite of Operating mode or Standby mode, the log is output on the node that controls the station.
- When the connection between Telephony Server and IP terminals/equipment is disconnected during the Transition to Operating, the log is not output.

(11) When using UMGi (**Note 5**), note the following conditions for the log output location
- The log that is generated by CCN or RN in normal mode is output on the CCN side.
- The log that is generated by RN in survival mode is not output.
- When the connection between CCN/RN and IP terminals/equipment is disconnected during the normal/survival mode switchover, the log is not output.

Note 5: When using Geographic Redundancy with UMGi, note the following conditions for the log output location:
- The log that is generated by CCN (Operating/Standby mode) is output on the node that controls the station.
- The log that is generated by RN (Normal/Operating mode) is output on the CCN (Operating mode) side.
- The log that is generated by RN (Survival/Operating mode) or RN (Survival/Standby mode) is not output.
- When the connection between CCN/RN and IP terminals/equipment is disconnected during the Transition to Operating, the log is not output.

2. Fault Log Information Collection
   a.) The times of read-out number can be determined such as once a day or certain period of a day.
   b.) The read-out process per terminal is completed within 15 seconds as a default. When several read-out processes are required to proceed in a large amount of information, the maximum of consecutive times can be determined in the system.
   c.) Other than periodical read-out process, a log will be read out when each terminal has registered.
   d.) To report all of the information collected, connect PCPro to read out data from memory in advance or set destination point for the SNMP Manager in advance.
   e.) If a trouble (such as WDT alarm and power failure) that cannot make any log occurs, its fault log cannot be stored as well.

3. Devices supporting Call log and Fault log information collection function are as follows:

Call Log and Fault Log Information Collection

Name of the IP Equipment		Supported firmware version
DT830/DT830G		Ver. 3.0.0.0 or later
DT820		Available since the first version
DT700 Series		Available since the first version
DtermIP	Proprietary Protocol (Including 16LD)	Ver. 02.80 or later
	SIP	Ver. 3.0.0.0 or later
IP Enabled Dterm		Ver. 02.80 or later
DtermIP INASET (240G/320GC)		Ver. 02.80 or later

Name of the IP Equipment		Supported firmware version
Soft Client SP350		Available since the first version
DtermSP30	Proprietary Protocol	Ver. 6.0.0.0 or later
	SIP	Ver. 9.0.0.11 or later

4. When the IP devices send the call log to the Telephony Server, the supported firmware version and each feature activity of IP devices as follows:

IP Device Call Information Output Feature For Telephony Server

X: Applicable N: Not Applicable

IP Device			Supported Firmware Version	Call Log Output	RTP Warning Tone Notification	Received RTP Packet Loss Count Notification
DT830/DT830G			Ver. 3.0.0.0 or later	X	X	X
DT820			Available since the first version	X	X	X
DT700 Series			Available since the first version	X	X	X
IP Enabled Dterm			Ver. 2.82 or later	X	X	X
DtermIP			Ver. 2.81 or later	X	X	X
DtermIP (SIP)			Ver. 3.0.7.0 or later	X	N Note 9	X
Soft Client SP350			Available since the first version	X	X	X
DtermSP30	Proprietary Protocol		Ver. 6.3.0.4 or later	X	X	X
	SIP		Ver. 11.0.0.0 or later	X	N Note 9	X
Analog MC			SP-3753 2AMC PROG-B Issue 12 or later	X	X	X
8LC Card	SCA-8LCA		SP-3893 8MC PROG-A Issue 2 or later	X	X	X
	SCA-8LCC		SP-3893 8MC PROG-B Issue 1 or later	X	X	X
MG (PRI) Card/Box [SCA-24PRIA/MG-24PRIA]		Proprietary Protocol	SP-3884 MG PRI PROG-A Issue 1 or later	X	X	X
		SIP	SP-3884 MG PRI PROG-A Issue 5 or later	X	X	X
MG (PRI) Card/Box [SCA-30PRIA/MG-30PRIA]		Proprietary Protocol	SP-3884 MG PRI PROG-A Issue 1 or later	X	X	X
		SIP	SP-3884 MG PRI PROG-A Issue 5 or later	X	X	X

IP Device			Supported Firmware Version	Call Log Output	RTP Warning Tone Notification	Received RTP Packet Loss Count Notification
MG (BRI) Card/Box [SCA-2BRIA/SCA-2BRIA-B/ MG-2BRIA]		SIP	SP-3826 BRI MG SIP BSC PROG-A Issue 1 or later	X	X	X
MCMG Card [SCA-4LC2COTA]			SP-3778 MCMG PROG-A Issue 8 or later	X	X	X
MCMG Box [MG-4LC2COTA]			SP-3927 MCMG PROG-B Issue 1 or later	X	X	X
MG-COT [SCA-6COTA]			SP-3875 6MGCOT PROG-A Issue 2 or later	X	X	X
MG-COT [SCA-6COTC]			SP-3948 6MGCOT PROG-A Issue 1 or later	X	X	X
MG (SIP)	Software-based MG-SIP **Note 6**		Available since the first version	X	N **Note 9**	X
	MG-SIP128 [MG-128SIPMGL-A] **Note 7**		Available since the first version	X	N **Note 9**	X
	MG-SIP128 [MG-128SIPMGG-B/ MG-128SIPMGG]		Available since the first version	X	N **Note 9**	X
	MG-SIP128 [MG-128SIPMGJ-B/ MG-128SIPMGJ]		Available since the first version	X	N **Note 9**	X
	MG-SIP96		SP-3987 MGSIP PROG-B Issue 3 or later	X	N **Note 9**	X
	MG-SIP16 Card/Box [SCA-16SIPMGA/MG-16SIP-MGA]		Available since the first version	X	N **Note 9**	X
VS32	Software-based VS32	Proprietary Protocol	Available since the first version	X	N **Note 9**	X
	VS-32 (Dual) Card [SCA-VS32VA/MG-VS32VA] VS-32 (Dual) Box [MG-VS32VA]	Proprietary Protocol	SP-3891 CA-VS32 PROG-A Issue 1 or later	X	N **Note 9**	X
		SIP	SP-3891 CA-VS32 PROG-A Issue 3 or later	X	N **Note 9**	X
IPPAD PKG	PA-32IPDA		SP-3835 IPPAD2 PROGRAM-A Issue 3 or later	X	N **Note 9**	X
	PA-32IPDB		SP-3835 IPPAD2 PROGRAM-A Issue 1 or later	X	N **Note 9**	X
	CH-IPDA		Available since the first version	X	N **Note 9**	X
UG50 **Note 8**			Available since the first version	X	X	X
Standard SIP terminal			It depends on the terminal.			

Note 6: For SP-4085, collected data is only voice RTP payload. Video is not supported.

Note 7: Collected data is only voice RTP payload. Video is not supported.

Note 8: Available only for IPG mode in DLC function. Refer to the Peripheral Equipment Description (UG50) for detailed conditions for UG50.

Note 9: Not applicable because the terminal outputs only “0.”

Programming

STEP 1: **ASYDL** - SYS1, Index 996
Bit 0: 0/1=None/Call Log Collection is enabled. **Note 10**
Bit 1: 0/1=None/Call Log SNMP Output is enabled. **Note 10**, **Note 11**, **Note 12**
Bit 2: 0/1=None/Fault Log Output is enabled. **Note 13**
Bit 3: 0/1=Fault Log read-out process immediately after registry is enabled./None
Bit 4: 0/1=Fault Log is read out once a day/periodically
Bit 5: 0/1=None/Fault log is read out periodically.

Note 10: As the filtering is configured by using the ACLFL command if Call Log Filter is enabled, this setting is not required.

Note 11: This is available when SYS1, Index 996, Bit 0=1.

Note 12: When call log is output via SNMP, refer to SNMP [S-148] in Data Programming Manual - Business.

Note 13: A system message containing fault log occurred immediately after registry is output regardless of this bit. To stop the system message output, set SYS1, Index 996, Bit 3=1.

- SYS1, Index 997: Cycle of fault log readout
00H: 30 seconds
01H-FFH: 1-255 (× 15 seconds)

- SYS1, Index 998: A maximum of consecutive times to read out fault log
00H: Eight times
01H-FFH: 1-255 (times)

- SYS1, Index 999: Call Log Filter function **Note 14**
Bit 3: 0/1=Disable/Enable

Note 14: Available since FP95-114 V4. When using Call Log Filter, assign “1” (Enable).

STEP 2: **ACLFL** - Assign Call Log Filter control data according to the following parameters.

Note: Available since FP95-114 V4.

FPC: Input the Fusion Point Code of the target node. **Note 15**

Note 15: This parameter is displayed only when logging in to NCN.

Setting Kind: Select Common Setting or Station Filter Setting.

<Common Setting>
Call Log Collect: Select Stop or Start for Call Log collection.
Output Call Log to SNMP: Call Log is output to SNMP when placing a check mark on this check box to ON. **Note 16**

Note 16: When this parameter is enabled, SNMP configuration is required. For details, refer to SNMP [S-148] in Data Programming Manual - Business.

Fault Kind

RTP Warning Tone: When choosing this check box, Call Log that generated an RTP warning tone is output in spite of Station Filter Setting.

Received RTP Packet Loss: When choosing this check box, Call Log that generated Received RTP Packet loss times over than the specified threshold is output in spite of Station Filter Setting.

Threshold: Specify the times for over the threshold value of Received RTP Packet Loss.

<Station Filtering Setting>

Station Number Type: Select TELN(NDM), TELN(LDM) or STN.

UGN (START/END): Specify the range of user group number. **Note 17**

TELN (START/END): Specify the range of telephone number. **Note 17**

TN (START/END): Specify the range of tenant. **Note 18**

STN (START/END): Specify the range of station number. **Note 18**

Note 17: These parameters are displayed when “TELN(NDM)” or “TELN(LDM)” is selected for Station Number Type.

Note 18: These parameters are displayed when “STN” is selected for Station Number Type.

<Station Data List>

Setting: Select the check box to ON for filtering station.

Note: When filtering for all of getting stations, place a check mark on “All Filter Setting” check box to ON.

RTP Warning Tone

General Description

This function generates an audible alarm when the packet loss rate reaches a threshold value of 20% with a duration of at least three seconds. The alarm, an intermittent dual tone consisting of 770 Hz and 1663 Hz sinusoids (DTMF tone for key B), makes five short beeps (five beeps per second, each with 120 ms duration and a 100 ms pause in between).

Packet loss is checked every one second. If RTP packet loss continues, the Warning Tone will be generated in 10 seconds after the first one.

RTP Packet Loss Alarm

RTP Packet Loss Check Period		1 second
RTP Alarm Conditions	Received RTP Packet Loss Threshold	20% of expected packets
	Reported Packet Loss Interval	3 seconds or more
	Time Between Warning Tone Beeps	10 seconds
Alarm Tone	Alarm Sound Pattern	5 short beeps per second
	Tone Duration	120 ms
	Tone Pause	100 ms

Operating Procedure

No operation is required.

Service Conditions

<Important Notice>

1. RTP warning tone can be played back or canceled by the setting of system data (ASYDL SYS1, Index 831, Bit 5).

   Note: For IP devices such as MG and MC, the setting of system data (ASYDL, SYS1, Index 831, Bit 5) will not be reflected. Therefore, the setting of MG/MC is required. For details, refer to the command section (SET WARNING_TONE) of each MG/MC in Peripheral Equipment Description - IP Devices.

   Note: The setting will be valid for the next call after it is set.

2. To play back or cancel the RTP warning tone can be chosen to follow the system data setting or to follow the setting of the each terminal individually.
   - Automatic: Follow the setting of system data setting (ASYDL, SYS1, Index 831, Bit 5).
   - ON: Play back the RTP warning tone. (Does not follow the setting of the system data)
   - OFF: Does not play back the RTP warning tone (Does not follow the setting of the system data).

   Note: For the setting for each terminal, see the following:
   - DT830 Series
     "RTP Alarm" for Configuration Setup of DT830 Series in Chapter 3 of Peripheral Equipment Description - Multi-Line Telephone
   - DT820 Series
     "RTP Alarm" for Configuration Setup of DT820 Series in Chapter 3 of Peripheral Equipment Description - Multi-Line Telephone
   - DT700 Series
     "RTP Alarm" for Configuration Setup of DT700 Series in Chapter 3 of Peripheral Equipment Description - Multi-Line Telephone
   - DtermIP/IP Enabled Dterm
     "How to Switch DTMF/RTP Alarm ON and OFF" for Configuration Setup of DtermIP/IP Enabled Dterm in Chapter 3 of Peripheral Equipment Description - Multi-Line Telephone
   - DtermIP (SIP)
     "How to Switch DTMF/RTP Alarm ON and OFF" for Configuration Setup of DtermIP (SIP) in Chapter 3 of Peripheral Equipment Description - Multi-Line Telephone
   - Softphone
     The operation manual attached to the Softphone

3. RTP warning tone is generated by DTMF as dial information. When a user is engaged in a call and an opposite party presses any key during the conversation, another tone will be generated by DTMF as dial information. Then, the user might be confused between the RTP warning tone and another DTMF tone. To avoid this, disable the DTMF tone used for RTP warning tone so that the user can distinguish it from the DTMF tones used for other purposes such as dial information. For more information, see " DTMF Control".

<Hardware/Software>

1. The firmware version of terminal needs to support this function to perform it. The playback/cancel of the RTP warning tone depends on terminal when it does not support this function.
2. Terminals/equipment supporting RTP warning tone playback function are as follows:

RTP Warning Tone Playback

Name of the IP Equipment		Supported firmware version
DT830/DT830G		Ver. 3.0.0.0 or later
DT820		Available since the first version
DT700 Series		Available since the first version
DtermIP	Proprietary Protocol	Ver. 2.08 or later
	SIP	Ver. 3.0.0.0 or later
IP Enabled Dterm		Ver. 2.04 or later
Soft Client SP350		Available since the first version
DtermSP30	Proprietary Protocol	Ver. 6.3.0.4 or later
	SIP	Ver. 9.0.0.11 or later
8LC Card [SCA-8LCA] 8LC Card [SCA-8LCA-EMEA] **Note 19**		SP-3893 8MC PROG-A Issue 2
MCMG Card [SCA-4LC2COTA] MCMG Box [MG-4LC2COTA] **Note 19**		SP-3778 MCMG PROG-A Issue 5.0 or later SP-3927 MCMG PROG-B Issue 1.0 or later
MG-T1(SIP) [SCA-24DTIA] **Note 19**		Available since the first version
MG(PRI) Card [SCA-24PRIA] MG(PRI) Box [MG-24PRIA]	Proprietary Protocol	SP-3884 MG PRI PROG-A Issue 1.0 or later
	SIP	SP-3884 MG PRI PROG-A Issue 5.0 or later
MG(PRI) Card [SCA-30PRIA] MG(PRI) Box [MG-30PRIA]	Proprietary Protocol	SP-3884 MG PRI PROG-A Issue 1.0 or later
	SIP	SP-3884 MG PRI PROG-A Issue 5.0 or later
MG(BRI) Card [SCA-2BRIA] MG(BRI) Box [MG-2BRIA] **Note 19**	SIP	SP-3826 BRI MG SIP BSC PROG-A Issue 1.0 or later
UG50 **Note 20**		Any version

Note: DtermIP INASET (240G/320GC), VS32, IPPAD and MG(SIP) do not support “RTP Warning Tone Playback.”

Note: Analog MC is not supported. However, UG50-2MC and UG50-8LC support “RTP Warning Tone Playback.”

Note 19: This function is set “enabled” as default.

Note 20: Refer to the Peripheral Equipment Description (UG50) for detailed conditions for UG50.

3. Terminals/equipment supporting cancel of RTP warning tone function are as follows:

RTP Warning Tone Cancel

Name of the IP Equipment		Supported firmware version Note 21
DT830/DT830G		Ver. 3.0.0.0 or later
DT820		Available since the first version
DT700 Series		Available since the first version
DtermIP	Proprietary Protocol	Ver. 3.00 or later
	SIP	Ver. 3.0.0.0 or later
IP Enabled Dterm		Ver. 3.06 or later
Soft Client SP350		Available since the first version
DtermSP30	Proprietary Protocol	Ver. 9.0.0.11 or later
	SIP	Ver. 9.0.0.11 or later
8LC Card [SCA-8LCA] Note 22		SP-3893 8MC PROG-A Issue 2
MCMG Card [SCA-4LC2COTA] MCMG Box [MG-4LC2COTA] Note 22		SP-3778 MCMG PROG-A Issue 5.0 or later SP-3927 MCMG PROG-B Issue 1.0 or later
MG-T1(SIP) [SCA-24DTIA]		Available since the first version
MG(PRI) Card [SCA-24PRIA] MG(PRI) Box [MG-24PRIA] Note 24	Proprietary Protocol	SP-3884 MG PRI PROG-A Issue 1.0 or later
	SIP	SP-3884 MG PRI PROG-A Issue 5.0 or later
MG(PRI) Card [SCA-30PRIA] MG(PRI) Box [MG-30PRIA] Note 24	Proprietary Protocol	SP-3884 MG PRI PROG-A Issue 1.0 or later
	SIP	SP-3884 MG PRI PROG-A Issue 5.0 or later
MG(BRI) Card [SCA-2BRIA] MG(BRI) Box [MG-2BRIA] Note 23	SIP	SP-3826 BRI MG SIP BSC PROG-A Issue 1.0 or later
UG50 Note 25		Not supported

Note 21: The setting of RTP warning tone is different for each terminal/device.

Note 22: This function is set “enable” as default. Only this function can be set “disable”.

Note 23: This function is set “enable” as default. When this function is disabled, the sound of other functions that give a beep though handset are disabled as well.

Note 24: This function is set “disable” as default. When this function is disabled, the sound of other functions that give a beep though handset are disabled as well.

Note 25: Refer to the Peripheral Equipment Description (UG50) for detailed conditions for UG50.

Programming

ASYDL - SYS1, Index 831 (RTP warning tone), Bit 5: 0/1=Play back the warning tone/Cancel the warning tone.

Note: For the terminals except IP terminal, Softphone and SIP Multiple Line terminal, the warning tone cannot be canceled if this system data is assigned as “1”.

DTMF Control

General Description

This function enables/disables DTMF tone to the opposite party. When RTP warning tone is enabled, the opposite party might be confused with RTP warning tone when hearing it. To avoid this, disable DTMF tone used for RTP warning tone so that the users can distinguish it form DTMF tones used for other purposes such as dial information. It also provides the user with more comfortable conversation.

Operating Procedure

No operation is required.

Service Conditions

1. Terminal firmware version needs to support the canceling function to disable the DTMF tone. Otherwise, it will be unconditionally played back. Terminals supporting cancel of DTMF sending function are as follows:

DTMF Sending Cancel

Name of the IP Equipment		Supported firmware version
DT830/DT830G		Ver. 3.0.0.0 or later
DT820		Available since the first version
DT700 Series		Available since the first version
DtermIP	Proprietary Protocol	Ver. 3.00 or later
	SIP	Ver. 3.0.0.0 or later
IP Enabled Dterm		Ver. 3.06 or later
Soft Client SP350		Available since the first version
DtermSP30	Proprietary Protocol	Ver. 9.0.0.11 or later
	SIP	Ver. 9.0.0.11 or later

Note: MG(PRI/BRI/COT), 8LC Card, VS32 (SIP mode) and IPPAD do not support this function. Refer to the Peripheral Equipment Description (UG50) for detailed conditions for UG50.

2. Playback/Cancel of DTMF tone to the destination party can be assigned with the SFI:225 of ASFC command.
3. The setting will be valid for the next communication after it is set.
4. To play back or cancel the DTMF tone can be chosen to follow the ASFC command data setting or to follow the setting of the each terminal individually.
   - Automatic: Follow the setting of ASFC command data setting (SFI: 225).
   - ON: Play back the DTMF tone. (Does not follow the setting of the ASFC command)
   - OFF: Does not play back the DTMF tone (Does not follow the setting of the ASFC command).

Note: For the setting for each terminal, see the following:
- DT830 Series
  "RTP Alarm" for Configuration Setup of DT830 Series in Chapter 3 of Peripheral Equipment Description - Multi-Line Telephone

- DT820 Series
  “RTP Alarm” for Configuration Setup of DT820 Series in Chapter 3 of Peripheral Equipment Description - Multi-Line Telephone
- DT700 Series
  “RTP Alarm” for Configuration Setup of DT700 Series in Chapter 3 of Peripheral Equipment Description - Multi-Line Telephone
- DtermIP/IP Enabled Dterm
  “How to Switch DTMF/RTP Alarm ON and OFF” for Configuration Setup of DtermIP/IP Enabled Dterm in Chapter 3 of Peripheral Equipment Description - Multi-Line Telephone
- DtermIP (SIP)
  “How to Switch DTMF/RTP Alarm ON and OFF” for Configuration Setup of DtermIP (SIP) in Chapter 3 of Peripheral Equipment Description - Multi-Line Telephone
- Softphone
  The operation manual attached to the Softphone

5. DTMF tone cannot be canceled for IPPAD, MG (PRI/BRI/COT), 8LC Card, and VS32 (SIP mode). DTMF tone is unconditionally played back for these equipments.

Programming

STEP 1: **ASYD**

- SYS1, Index 75, Bit 1 (Service Feature Restriction Class Data [ASFC/ASFCN command])
 0/1=Common or Separate Day/Night Data Tables

- SYS1, Index 93, Bit 0 (Service Feature Restriction Class Data [ASFC/ASFCN command])
 0/1=Separate or Common Tenant Data table development

STEP 2: **ASFC/ASFCN**

Assign data below to the station class that specifies DTMF signal tone playback.
SFC: Service Feature Restriction Class (1-15)
SFI = 225 (DTMF tone playback)
RES= 0 (DTMF tone playback is canceled)/1 (DTMF tone playback)

R-60 Redial

General Description

This feature allows the user of each DT800 Series/DT700 Series/DT400 Series/DT300 Series to redial the last 20 dialed numbers stored in the memory.

Operating Procedure

How to redial the dialed numbers stored in the memory

1. Press the Last Number Called key or the Redial key.
2. Press the cursor key (Up/Down) or the Soft key (Up/Down) for displaying the dialed number for redialing by changing the display on an LCD as follows.

   No.1 → No.2 → No.3 →...No.18 → No.19 → No.20 → No.1 → No.2...

No.1	The last dialed number
No.2	The number dialed 2-times ago
No.3	The number dialed 3-times ago
.	.
.	.
.	.
No.18	The number dialed 18-times ago
No.19	The number dialed 19-times ago
No.20	The number dialed 20-times ago

3. Go off-hook, or press the Speaker key/the OK key in the center of the cursor key.
4. The designated destination number is dialed automatically. Then, the calling station hears the Ringback Tone (RBT).

Note: Note for the following conditions when redialing.
- After pressing the Redial key, the dialing keys are disabled.
- When no dialed number is stored, “No list” is displayed for 10 seconds.

How to delete one dialed number stored in the memory

1. Press the Last Number Called key or the Redial key.
2. Press the cursor key (Up/Down) or the Soft key (Up/Down) for displaying the dialed number for deleting by changing display on an LCD as follows.

   No.1 → No.2 → No.3 →...No.18 → No.19 → No.20 → No.1 → No.2...

No.1	The last dialed number
No.2	The number dialed 2-times ago
No.3	The number dialed 3-times ago
.	.
.	.
.	.
No.18	The number dialed 18-times ago
No.19	The number dialed 19-times ago
No.20	The number dialed 20-times ago

3. Press the Delete key.
4. Press the One key for deleting the displayed number. After the number is deleted, the next dialed number is displayed.

How to delete all dialed numbers stored in the memory

1. Press the Last Number Called key or the Redial key.
2. Press the Delete key.
3. Press the All key for deleting the stored numbers. After the dialed number history is deleted, “No list” is displayed for 10 seconds.

Service Conditions

<Important Notices>

1. This feature is available only for DT800/DT700/DT400/DT300 Series assigned the System Data (ASYDL/ASYDN SYS1 Index 1257 Bit 2=1). For DT800/DT700/DT400/DT300 Series assigned the System Data (ASYDL/ASYDN SYS1 Index 1257 Bit 2=0), and Desktop terminals other than DT800/DT700/DT400/DT300 Series, Last Number Called is enabled. For details of Last Number Called, refer to Last Number Called -D [L-6D] in this manual.

   Note: When the System Data (ASYDL/ASYDN SYS1 Index 1257 Bit 2) is changed, each terminal needs to be restarted.

2. When the System Data (ASYDL SYS1 Index 673 Bit 1=1, Bit 0=0) is assigned, and the display location of User Information is set to the position where Soft key function is displayed, the Redial key becomes disabled.

<Interactions with Other Features>

1. When a Feature Key is assigned as the Last Number Called key, the Feature key also works as the Redial key.
2. Call origination by using Speed Calling feature with the Last Number Called key or the Redial key is unavailable. To use Speed Calling feature, be sure to disable this feature.

3. The dialed numbers cannot be registered for System Phone Book and Terminal Phone Book by using this feature

<Hardware/Software>

This feature is only available for DT800/DT700/DT400/DT300 Series. For terminals other than DT800/DT700/DT400/DT300 Series, refer to Last Number Called -D [L-6D] in this manual.

<Interactions with Networks>

None

<Conditions Specific to This Feature>

1. A maximum of 20 destinations can be stored.

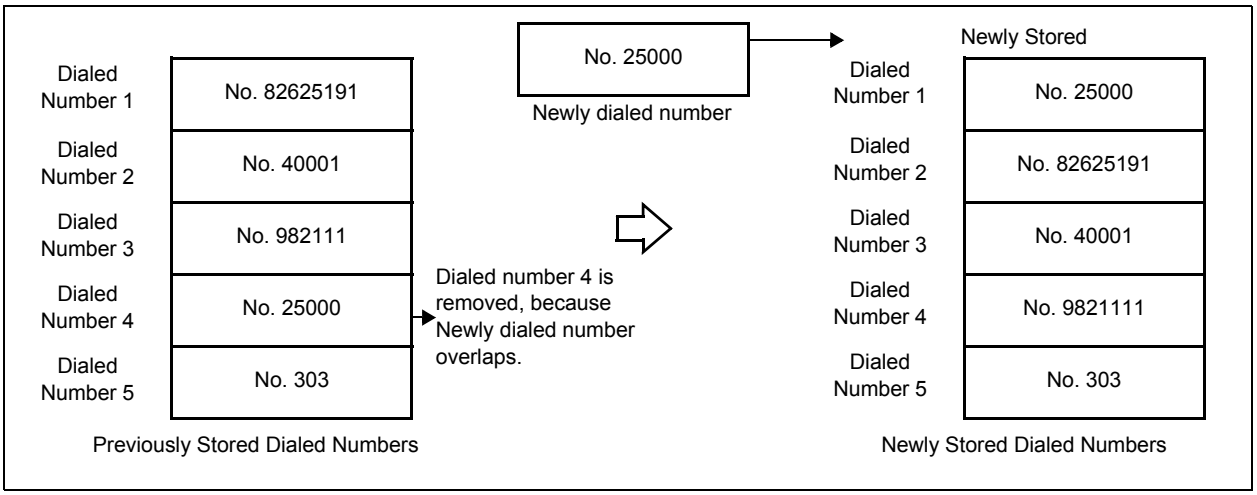
   Note: A maximum of 5 destinations can be stored by using Last Number Called -D [L-6D].

2. A maximum of 32 digits (the destination number + the number dialed during the call) can be stored (including “pause”).
3. The number dialed during the call can be stored with a maximum of 32 digits (the destination number + the number dialed during the call).
4. When the Last Number Called key or the Redial key is pressed in an idle state, DT800/DT700/DT400/DT300 Series displays the stored dialed numbers.
5. After pressing the Last Number Called key or the Redial key, DT800/DT700/DT400/DT300 Series displays the first 24 digits of a stored dialed number of 32 digits.
6. To finish displaying the stored dialed numbers, press the on of the following keys.

   - The Last Number Called key or the Redial key **Note 1**
   - The Exit key
   - The End key on the Soft key

   Note 1: The left button of the cursor key is disabled.

7. When a DT800/DT700/DT400/DT300 Series user dials the same destination that has already been stored in the system, the newly dialed number is stored in the following manner:



8. While DT800/DT700/DT400/DT300 Series displays the dialed numbers, LCD contrast cannot be operated by the cursor keys (Up/Down).
9. When receiving the following calls while displaying the dialed numbers, DT800/DT700/DT400/DT300 Series finishes displaying the dialed numbers, and displays a call indication.
   - A call for Prime Line
   - A call for the line during line-pre-selection
10. When making a call from DT800/DT700/DT400/DT300 Series during displaying the dialed numbers, even if you may hear Dial Tone, there is no problem with the operation of DT800/DT700/DT400/DT300 Series.

<Other Conditions>

None

Programming

STEP 1: **ASYDL/ASYDN** - Assignment of System Data (LDM/NDM)
SYS1 Index 1257 Bit 2 = 1 (Type of outgoing call history used by DT800/DT700/DT400/DT300 Series: Redial [R-60])

Note: When this System Data is changed, each terminal needs to be restarted.

Note: For details of Last Number Called when this System Data is assigned to “0” (Last Number Called -D [L-6D]), refer to Last Number Called -D [L-6D] in this manual.

R-61 Retention of Customized Terminal Settings

General Description

This feature allows Desktop terminals to start up with a preassigned terminal setting pattern, such as Receiver Volume, Ringer Tone Volume, LCD Contrast and Handset/Headset Switching, after system initialization or terminal re-registration. A maximum of eight options (seven Customized Terminal Setting Patterns and one Default Terminal Setting Pattern) can be preassigned through the PCPro.

Operating Procedure

None.

Service Conditions

<Important Notices>

1. When the system started up while this feature is disabled (ASYDL/ASYDN SYS1 Index 1248 Bit 3=0): If the setting of the system data is changed (ASYDL/ASYDN SYS1 Index 1248 Bit 3=1) during operation, system initialization or Internal PHE module initialization with the ADPM command is required for the change to take effect.

   Note: When Internal PHE initialization is executed, all other devices controlled by Internal PHE are also initialized.

2. When the system started up while this feature is enabled (ASYDL/ASYDN SYS1 Index 1248 Bit 3=1): If the setting of the system data is changed during the operation, the new setting is applied to each terminal which performs a registration after the data change. The setting on the each terminal is determined according to the state of the data setting at the time of terminal registration.

   Note: The setting of the terminal which performs a registration before the system data change is not reflected.

3. Be sure to assign (enable) the system data below. If disabled, receiver volume for Desktop terminal returns to the default after going on-hook/off-hook or switching headset/handset mode.

   ASYD, System Data1, Index 384, Bit 5=1: Retains Handset/Headset receiver volume changed during a call (after the call is released).

<Interactions with Other Features>

None

<Hardware/Software>

1. This feature is available for DtermIP (Dterm Series i) and DT800/DT700 Series.

   Note: LCD contrast of terminal with color LCD such as DT750 and DT730G (12/24CG) cannot be set by this feature.

2. When using a DT700 Series terminal (except DT730G) with a firmware version earlier than V5.0.3.0, do not start the terminal with Headset enabled for a Customized Terminal Setting Pattern. If you do so, no operation will be possible on the terminal.

Note: Manual Handset/Headset switching on the currently operating terminal does not affect the functionality.

<Interactions with Networks>

None

<Conditions Specific to This Feature>

1. When this feature is enabled, the terminals to which no Customized Terminal Setting Pattern is assigned start up with Default Terminal Setting Pattern. If Default Terminal Setting Pattern data is also not assigned, the terminals start up with the factory default settings.
2. A maximum of seven Customized Terminal Setting Patterns (eight options including a Default Terminal Setting Pattern) can be preassigned through the PCPro.
3. When terminal settings are changed by key operations on the terminal while logged on to the system, the settings return to Customized Terminal Setting Pattern assigned through the PCPro/QuickPro after system initialization or terminal re-registration.
4. When Customized Terminal Setting Pattern assigned to a terminal or setting values of the Customized Terminal Setting Pattern are changed through the PCPro/QuickPro while the terminal is logging on to the system, the changes take effect after the terminal re-registration.
5. When a terminal is in the following status, the terminal starts up with the setting that Handset is used even if Headset is chosen in Customized Terminal Setting Pattern.
   - Headset key is not registered on the terminal.
   - "S=0" is assigned by AKYD command.
6. If the status of a DtermIP (Dterm Series i) meets all the following conditions, the DtermIP user may momentarily hear DT from the headset on re-registration.
   - "S=0" is assigned by the AKYD command.
   - Headset is not chosen in Customized Terminal Setting Pattern.
   - Headset is connected to the terminal.
   - Headset key is not registered on the terminal.

<Other Conditions>

When this feature is disabled (ASYDL/ASYDN SYS1 Index 1248 Bit 3=0), customized settings of a Desktop terminal are not retained after the terminal is turned off and comes to a complete stop or when a long time has elapsed since its last logout. The Desktop terminal operates with default settings on the next start-up or login.

Programming

STEP 1: **ASYD** - System Data 1, Index 384, Bit 5. Set to 1 to retain Handset/Headset receiver volume for Desktop terminal changed during a call (after the call is released).

STEP 2: **ASYDL** - System Data 1, Index 678, Bit 0. Set to 0 to retain the setting of handset receiver volume changed during a call.

STEP 3: **ASYDL/ASYDN** - System Data 1, Index 1248, Bit 3. Set to 1 to enable RETENTION OF CUSTOMIZED TERMINAL SETTINGS [R-61].

STEP 4: Assign detailed information on Customized Terminal Setting for each pattern.

Note: When the following data settings are changed, terminal re-registration is required to enable the new settings.

ASYDL/ASYDN

System Data 1, Index 1264, Bits 0-3: Receiver Volume Setting for **Customized Terminal Setting Pattern 1**.

Bits				Volume Level (dB)	Scale on LCD	Remarks
b3	b2	b1	b0			
0	0	0	0	0	6	**Note 1**
0	0	0	1	-10	1	Minimum (Bias value ±0dB)
0	0	1	0	-8	2	Minimum (Bias value ±2dB)
0	0	1	1	-6	3	Minimum (Bias value ±4dB)
0	1	0	0	-4	4	Minimum (Bias value ±6dB)
0	1	0	1	-2	5	
0	1	1	0	0	6	Default for terminal
0	1	1	1	2	7	
1	0	0	0	4	8	
1	0	0	1	6	9	
1	0	1	0	8	10	
1	0	1	1	10	11	
1	1	0	0	12	12	
1	1	0	1	14	13	Maximum
1	1	1	0	0	6	**Note 1**
1	1	1	1	0	6	**Note 1**

Note 1: This value is the same as default setting of the terminal side.

System Data 1, Index 1264, Bits 4-7: Ringer Tone Volume Setting for **Customized Terminal Setting Pattern 1**.

Bits				Volume Level (dB)	Scale on LCD	Remarks
b7	b6	b5	b4			
0	0	0	0	−20	8	**Note 2**
0	0	0	1	−48	1	Minimum
0	0	1	0	−44	2	
0	0	1	1	−40	3	
0	1	0	0	−36	4	
0	1	0	1	−32	5	
0	1	1	0	−28	6	
0	1	1	1	−24	7	
1	0	0	0	−20	8	Default for terminal
1	0	0	1	−16	9	
1	0	1	0	−12	10	
1	0	1	1	−8	11	
1	1	0	0	−4	12	
1	1	0	1	0	13	Maximum
1	1	1	0	−20	8	**Note 2**
1	1	1	1	−20	8	**Note 2**

Note 2: This value is the same as default setting of the terminal side.

System Data 1, Index 1265,
Bits 0-3: LCD Contrast Setting for **Customized Terminal Setting Pattern 1**.

Bits				Contrast	Scale on LCD	Remarks
b3	b2	b1	b0			
0	0	0	0		5	**Note 3**
0	0	0	1	Low	1	Minimum
0	0	1	0	.	2	
0	0	1	1	.	3	
0	1	0	0	.	4	
0	1	0	1	.	5	Default for terminal
0	1	1	0	.	6	
0	1	1	1	.	7	
1	0	0	0	High	8	Maximum
1	0	0	1		5	**Note 3**
1	0	1	0		5	**Note 3**
1	0	1	1		5	**Note 3**
1	1	0	0		5	**Note 3**
1	1	0	1		5	**Note 3**
1	1	1	0		5	**Note 3**
1	1	1	1		5	**Note 3**

Note 3: This value is the same as default setting of the terminal side.

Bit 4: Handset/Headset Switching for **Customized Terminal Setting Pattern 1**.
0/1=Handset/Headset

System Data 1, Index 1266
Bits 0-3: Receiver Volume Setting for **Customized Terminal Setting Pattern 2**.
Bits 4-7: Ringer Tone Volume Setting for **Customized Terminal Setting Pattern 2**.

Note: For details on setting of each bit, see “Index 1264.”

System Data 1, Index 1267
Bits 0-3: LCD Contrast Setting for **Customized Terminal Setting Pattern 2**.

Note: For details on setting of each bit, see “Index 1265.”

Bit 4: Handset/Headset Switching for **Customized Terminal Setting Pattern 2**.
0/1=Handset/Headset

System Data 1, Index 1268

Bits 0-3: Receiver Volume Setting for **Customized Terminal Setting Pattern 3**.
Bits 4-7: Ringer Tone Volume Setting for **Customized Terminal Setting Pattern 3**.

Note: For details on setting of each bit, see “Index 1264.”

System Data 1, Index 1269
Bits 0-3: LCD Contrast Setting for **Customized Terminal Setting Pattern 3**.

Note: For details on setting of each bit, see “Index 1265.”

Bit 4: Handset/Headset Switching for **Customized Terminal Setting Pattern 3**.
0/1=Handset/Headset

System Data 1, Index 1270
Bits 0-3: Receiver Volume Setting for **Customized Terminal Setting Pattern 4**.
Bits 4-7: Ringer Tone Volume Setting for **Customized Terminal Setting Pattern 4**.

Note: For details on setting of each bit, see “Index 1264.”

System Data 1, Index 1271
Bits 0-3: LCD Contrast Setting for **Customized Terminal Setting Pattern 4**.

Note: For details on setting of each bit, see “Index 1265.”

Bit 4: Handset/Headset Switching for **Customized Terminal Setting Pattern 4**.
0/1=Handset/Headset

System Data 1, Index 1272
Bits 0-3: Receiver Volume Setting for **Customized Terminal Setting Pattern 5**.
Bits 4-7: Ringer Tone Volume Setting for **Customized Terminal Setting Pattern 5**.

Note: For details on setting of each bit, see “Index 1264.”

System Data 1, Index 1273
Bits 0-3: LCD Contrast Setting for **Customized Terminal Setting Pattern 5**.

Note: For details on setting of each bit, see “Index 1265.”

Bit 4: Handset/Headset Switching for **Customized Terminal Setting Pattern 5**.
0/1=Handset/Headset

System Data 1, Index 1274
Bits 0-3: Receiver Volume Setting for **Customized Terminal Setting Pattern 6**.
Bits 4-7: Ringer Tone Volume Setting for **Customized Terminal Setting Pattern 6**.

Note: For details on setting of each bit, see “Index 1264.”

System Data 1, Index 1275
Bits 0-3: LCD Contrast Setting for **Customized Terminal Setting Pattern 6**.

Note: For details on setting of each bit, see “Index 1265.”

Bit 4: Handset/Headset Switching for **Customized Terminal Setting Pattern 6**.
0/1=Handset/Headset

System Data 1, Index 1276
Bits 0-3: Receiver Volume Setting for **Customized Terminal Setting Pattern 7**.
Bits 4-7: Ringer Tone Volume Setting for **Customized Terminal Setting Pattern 7**.

Note: For details on setting of each bit, see “Index 1264.”

System Data 1, Index 1277
Bits 0-3: LCD Contrast Setting for **Customized Terminal Setting Pattern 7**.

Note: For details on setting of each bit, see “Index 1265.”

Bit 4: Handset/Headset Switching for **Customized Terminal Setting Pattern 7**.
0/1=Handset/Headset

System Data 1, Index 1278
Bits 0-3: Receiver Volume Setting for **Default Terminal Setting Pattern.**
Bits 4-7: Ringer Tone Volume Setting for **Default Terminal Setting Pattern**.

Note: For details on setting of each bit, see “Index 1264.”

System Data 1, Index 1279
Bits 0-3: LCD Contrast Setting for **Default Terminal Setting Pattern.**

Note: For details on setting of each bit, see “Index 1265.”

Bit 4: Handset/Headset Switching for **Default Terminal Setting Pattern**.
0/1=Handset/Headset

STEP 5: **ASGD** - Assign Customized Terminal Pattern Number to each Station.

TN: Tenant Number

STN: Station Number

F: Function Number =7 (Retention of Customized Terminal Settings)

GN: Customized Terminal Pattern Number (1-7)

GN=1: Applies the settings of ASYDL/ASYDN, System Data 1, Indexes 1264 and 1265.

GN=2: Applies the settings of ASYDL/ASYDN, System Data 1, Indexes 1266 and 1267

GN=3: Applies the settings of ASYDL/ASYDN, System Data 1, Indexes 1268 and 1269

GN=4: Applies the settings of ASYDL/ASYDN, System Data 1, Indexes 1270 and 1271

GN=5: Applies the settings of ASYDL/ASYDN, System Data 1, Indexes 1272 and 1273

GN=6: Applies the settings of ASYDL/ASYDN, System Data 1, Indexes 1274 and 1275

GN=7: Applies the settings of ASYDL/ASYDN, System Data 1, Indexes 1276 and 1277

Note: For setting of the GN parameter, consider the following.

- Terminal re-registration is required for this data change to take effect.
- When no data is set for the GN parameter, the setting s of ASYDL/ASYDN, System Data 1, Indexes 1278 and 1279 is applied.

R-62 Ring Volume in Headset Mode

General Description

When a Desktop terminal in the Headset Mode receives an incoming call, Ringer Tone Volume from a speaker of the Desktop terminal is adjusted to 20 dB lower than the actual volume setting. However, this feature enables to apply the actual volume setting to a Desktop terminal in the Headset Mode **Note 1**.

Note 1: The term "Headset Mode" refers to the condition that a Headset is activated by pressing the Handset/Headset key, and a Desktop terminal user hears tone such as Dial Tone (DT), Ring-back Tone (RBT) or Reorder Tone (ROT), or line selection is still not performed (AKYD, S = 1: (Off-hook suppression is restricted)).

Operating Procedure

1. Press the cursor key (Up/Down) and adjust Ringer Tone Volume when a Desktop terminal that uses the Handset (the lamp of the Handset/Headset key is off) and is in idle status receives a call (Dial Tone is heard). **Note 2**
2. Press the Handset/Headset key and switch to the Headset (the Handset/Headset key lamp turns on).
3. When the Desktop terminal is in the Headset Mode, Ringer Tone Volume adjusted in the Operating Procedure 1 is used for the incoming call to the lines assigned to the Desktop terminal.

Note 2: This Ringer Tone Volume setting is for the normal use of the Desktop terminal. By this feature, this setting becomes available also for the Desktop terminal in the Headset Mode.

Service Conditions

<Important Notices>

1. When a change is made to the data setting of ASFC/ASFCN SFI263 (Ringer Tone Volume of Desktop Terminal in Headset Mode), re-registration of the Desktop terminal using this feature is required to reflect the new setting.

<Interactions with Other Features>

1. When using this feature, Ringer Tone Volume for the Headset Mode is subject to Ringer Tone Volume setting of the Desktop terminal in idle status. If Ringer Tone Volume setting of the Desktop terminal in idle status is changed by other features (for example, assigning ASYDL, SYS1, Index 678, bit 0 = 0 (retaining the setting of handset receiver volume changed during a call)), this feature is subject to the newly changed setting.
2. When using Location Diversity [L-55] with this feature, the Desktop terminal is subject to the data setting of the node to which the Desktop terminal is registered.

<Hardware/Software>

1. This feature is released for DT800/DT700 Series. Use the terminals with the following firmware versions.

Terminals	Firmware Version	Remarks
DT750	V5.0.3.0 or later	-
DT730		
DT730G (24CG/24DG)	V1.0.1.0 or later	
DT710	not supported	This feature is not supported by DT710/DT770G because they cannot be connected to a Headset.
DT770G		
DT830	V3.0.0.0 or later	-
DT830G		

<Interactions with Networks>

None

<Conditions Specific to This Feature>

1. When a Desktop terminal user is on the 1st call using a Handset and the 2nd call arrives at another line, this feature does not work during the following situations:

   - a Desktop terminal is receiving a call
   - a Desktop terminal user is hearing tones

   In these situations, Ringer Tone Volume becomes 20 dB lower than the actual volume setting.
2. Ringer Tone Volume cannot be adjusted while talking on a Headset or hearing tones with a Headset. In this situation, only the Receiver Volume can be adjusted. However, Ringer Tone can be muted in this situation by pressing Off Hook Ringing key (assigned by AKYD FKY = 123) or Ringer Mute key (assigned by AKYD FKY = 117).

<Other Conditions>

None

Programming

STEP 1: **ASFC/ASFCN**- Assign a Service Feature Restriction Class

SFC: 0 - 15 (Service Feature Restriction Class)

D/N: D/N (Day/Night Mode)

TN: Tenant Number

SFI: 263 = 1 (Enable Ring Volume in Headset Mode feature; Ringer Tone Volume is not adjusted to 20 dB lower than the actual Ringer Tone Volume.) **Note 3**

Note 3: To reflect the new setting, re-registration of the Desktop terminal using this feature is required.

R-63 Real-Time Call Information on Softphone

General Description

This feature enables the telephony server to send information about the opposite party and call records to softphones in real time by using an interface similar to that used for SIP Multiple Line terminals.
To implement this feature, you must configure the system settings to allow the softphones to receive the following information:

- Opposite party information
  Allow opposite party information to be passed to softphones.
- Call record information
  Decide whether calls picked up by non-ring station are classified as an answered call or a missed call.

Operating Procedure

No manual operation is required.

Service Conditions

<Common Conditions>

1. When the system is configured to enable integration between a softphone and a PS/WLAN Handset/wired phone (Soft Phone - PS [S-158]/Soft Phone WLAN Handset [S-159]/Soft Phone Wired-Phone Handset [S-147]), if a call comes in to a PS/WLAN Handset/wired phone during the temporary release of the softphone integration and is abandoned before being answered, the call is not recorded in the call log of the softphone.

   Note: In the above case, if the call is answered on the softphone, the call record is stored in the call log of the softphone.

2. When you retrieve a parked call, the call is recorded in the incoming call log (not in the outgoing call log) of your softphone.
3. When a call ringing terminal A is picked up by terminal B, the call is recorded as an answered call on terminal B. It depends on the system data (ASYDL/ASYDN SYS1 Index 681 Bit 2) whether the call is recorded as a missed call or an answered call on terminal A. This preference setting also applies to the terminal rung with Internal Zone Paging [I-21] or DUAL STATION CALL [D-156].
4. The Real-Time Call Information on Softphone feature is available for Soft Client SP350 R6.0.0.0 or later.

<Conditions for Opposite Party Information>

1. Opposite party information is sent to your softphone when:

   - An incoming call terminates to My Line of your station if the same number as My Line is assigned to its Prime Line.
   - You answer an incoming call or a held call by pressing a Line key on the station with Terminal Phonebook Display on Line Pre-selection enabled (ASYDL SYS1 Index 679 Bit 0 = 1).
   - You start a telephone conversation.

- You place a call.

2. A maximum of 24 digits can be sent for each caller ID. If the length of a caller ID exceeds the limit, the telephony server truncates it based on the system data setting (ASYD SYS1 Index 241 Bit 6), and sends it to the softphone without adding a truncation symbol (asterisk).

<Conditions for Call Record Information>

1. Call record information is sent to your softphone when:

   - You answer an incoming call.
   - An incoming call is abandoned before you answer it.
   - An incoming call is picked up by a non-ring station.
   - You retrieve a held or parked call.
   - You pick up a call ringing on another station.

2. The following two types of call records are sent to softphones.

   - Answered calls
   - Missed calls

3. A maximum of 24 digits can be sent for each caller ID. If the length of a caller ID exceeds the limit, the telephony server truncates it based on the system data setting (ASYD SYS1 Index 241 Bit 6), and sends it to the softphone without adding a truncation symbol (asterisk).
4. When My Line of Station A is accommodated on a Sub Line of Station B, if Station B answers a call placed to Station A by pressing the corresponding Sub Line key, the call record is sent to both Station A and Station B.
5. When a call is answered through Call Pickup, the call record is sent to both stations which the call is originally placed to and which the call is answered on.
6. When a call below is answered, the call record is not sent to the answering station:

   - Calls terminating directly at a particular station instead of a line (for example, DID calls)

7. When a transferred call terminates, calling number sent as call record information is as follows:

   - Call Transfer - All Calls [C-11]:

     The number of Intermediate station (the first terminal to receive the call)

   - Blind Transfer To Attendant [B-17]/Blind Transfer To Station [B-18]:

     The number of originally calling station

<Other Conditions>

When a softphone and a DT800/DT700 Series terminal are functioning as an integrated unit, opposite party information and call record information are sent to both phones.

Programming

STEP 1: **ASYD**- System Data 1, Index 241, Bit 6. (Truncation of Caller ID exceeding the display on the LCD)
0/1: Only the former/latter part of data appears on the LCD

STEP 2: **ASYDL/ASYDN** - System Data 1, Index 678, Bit 1. Set to 1 to enable notification of opposite party information to softphones.

Note: To apply the changes, initialize the system.

STEP 3: **ASYDL/ASYDN** - System Data 1, Index 681, Bit 2. (Preference for classifying calls picked up by non-ring station as an answered call or a missed call)
0/1: Answered call/Missed call.

S-1 Single-Digit Station Calling

General Description

This feature allows the assignment of single digits for Station Numbers.

Operating Procedure

To operate

1. Lift the handset. Receive Dial Tone.
2. Dial the Single-digit (preprogrammed) Telephone Number.
3. The desired service station will be rung.

Service Conditions

1. The single-digit number must be different from the first digit of:
   a.) Trunk Access Codes
   b.) Service Feature Access Codes
   c.) Other Telephone Numbers
2. Timing start is not available with this feature.

Interactions

For each station with single-digit access, an entire Number Level is eliminated. Do not plan to have more than one or two such phones in a system.

Programming

STEP 1: **ASYD** - System Data 1, Index 16, Bit 0. Assign this Bit as data "1" enabling single-digit Station Numbers.

STEP 2: **ANPD** - Reserve a Number Level for station access. Assign an NND = 1, and BLF is ON.

STEP 3: **ASPA** - For the desired Station Number assign SRV = "STN: Station Number". NND is not required. Assign Connection Indexes (CI) of "N: Normal" and "H: Hooking".

STEP 4: **ASDT** - Assign the Station Number to a specific station within the system.

FCCS Programming

This data setting describes only the additional or replacement commands used to activate FCCS Services. When used as an FCCS feature, these commands replace the non-network commands (for example, AAED is replaced by AAEDN). Note that the data setting for FCCS Service varies depending on the Node.

Note: When using ASYDL or ASYDN to program an FCCS feature, the basic programming of ASYD is required in addition to ASYDL or ASYDN.

STEP 1: **ANPDN** - Reserve a Number Level for station access. Assign an NND = 1, and BLF is ON.

STEP 2: **ASPAN** - For the desired Station Number assign SRV = "TELN: Telephone Number". NND is not required. Assign Connection Indexes (CI) of "N: Normal" and "H: Hooking".

STEP 3: **ALGSN** - Assign Type = 2 / UGN1
TELN = Station Number/Telephone Number
FPC = "x"
Tenant and Station assigned by ASDT.

S-2 Special Dial Tone

General Description

This feature provides a distinctive Dial Tone to a Station user after the switch hook has been pressed, enabling the user to activate a specific feature.

Operating Procedure

To operate

1. While connected with the first party, Station user presses the switch hook. Receives SPECIAL DIAL TONE.
2. User may activate a feature, such as:

CALL HOLD	[C-6]
CALL TRANSFER-ALL CALLS	[C-11]

Service Conditions

1. SPECIAL DIAL TONE rating is 350/440 Hz at 240 IPM.
2. If dialing is not initiated within 12 seconds after receiving SPECIAL DIAL TONE, Reorder Tone will be sent to the Station. If the Station has not gone back on hook or initiated dialing after 30 seconds of Reorder Tone, the Station will be placed in the LINE LOCKOUT [L-3] mode.
3. If a Station user attempts CALL FORWARDING - ALL CALLS [C-5], -BUSY LINE [C-2] - DON'T ANSWER [C-3], etc., SPECIAL DIAL TONE will be sent to that Station after Access Code dialing.

Interactions

1. No other programming is required. This tone is generated for all momentary pressing of the Hooking feature applications.
2. LINE LOCKOUT [L-3]: SPECIAL DIAL TONE is sent for 12 seconds, Reorder Tone for 30 seconds, followed by Line Lockout.

Programming

ASYD - System Data 1, Index 42, Bit 7. Is SPECIAL DIAL TONE continuous or a burst? 0/1: Continuous/Burst. Leave as Data 0. System Data 2, Index 1, Bit 0 and 1. Assign as data "1" for both Bits to provide CONSULTATION HOLD capability on a per Tenant basis.

System Data 3, Index 2. Assign data 91H for Normal (N) and Hooking (H) values.

S-3 Speed Calling - System

General Description

This feature allows a Desktop terminal user or ATTENDANT CONSOLE [A-3] to call frequently Dialed Numbers using fewer digits (Abbreviated Call Codes) than would normally be required.

Operating Procedure

To originate a call using Access Code

1. Lift handset. Receive Dial Tone,
2. Dial SPEED CALLING - SYSTEM Access Code (maximum three digits).
3. Dial Abbreviated Call Code (maximum of four digits).
4. Call is completed.

<The following procedure is applied for Desktop terminals only.>

To originate a call using Line/Feature Key or Directory Key

1. Lift the handset or press the SPEAKER key. Receive Dial Tone.
2. Press the SPEED CALLING-SYSTEM (SPEED-SYS) Key or Directory Key. **Note 1 Note 2**
3. Dial the Abbreviated Call Code (maximum of four digits). The LCD displays:



Abbreviated Call Code

Note 1: Directory Key is available for Dterm Series i. For detailed information on the operation, refer to the related User's Guide.

Note 2: If the Desktop terminal does not have the SPEED-SYS Key, dial the SPEED CALLING-SYSTEM Access Code, then the Abbreviated Call Code.

Service Conditions

1. SPEED CALLING - SYSTEM codes programmed into System Data are accessible by all Stations with a CLASS OF SERVICE - INDIVIDUAL [C-15] permitting SPEED CALLING - SYSTEM.
2. Each Abbreviated Call Code can store a maximum of 24 digits, including Access Codes and pauses.
3. SPEED CALLING - SYSTEM numbers are programmed via System Data entry from the PCPro.
4. Stations may be denied access to SPEED CALLING - SYSTEM via CLASS OF SERVICE [C-15] assignment in System Data. SPEED CALLING may also be allowed or denied on either a Station or Tenant-wide basis.
5. TOLL DENIAL/TOLL DIVERSION [T-6] and TOLL RESTRICTION-3/6-DIGIT [T-7] may or may not be applied to SPEED CALLING - SYSTEM, as desired.
6. LEAST-COST ROUTING-3/6-DIGIT [L-5] may be provided via SPEED CALLING - SYSTEM access.
7. If the Station is restricted from dialing Outside via programming, the Station can still access SPEED CALLING - SYSTEM.
8. A maximum of 10,000 codes can be assigned per System.

9. The Outgoing Call using a Trunk in the Remote Node is applied to this feature.
10. When this feature is activated by dialing the Speed Calling - System Feature Access Code + the Abbreviated Call Code, only the Speed Calling-System Feature Access Code and the Abbreviated Call Code are displayed on the LCD of the Desktop terminal.

<The following conditions are applied for Desktop terminals only.>

1. When assigning the Access Code to Line/Feature Key, Abbreviated Call Code must be set at two to four digits.
2. When outgoing dialing with this feature, Speed Calling Number can be displayed on the LCD instead of Speed Calling-System Feature Access Code and the Abbreviated Call Code by ASYDL command (ASYDL SYS 1 Index 673 bit 6=1). The following conditions are about this feature.

   Note: This feature is available only with outgoing call (LCR/LCRS/OGC/OGCA).

   a.) When you register the Abbreviated Call Codes as a Forwarding Destination with Speed Calling Number Display on Desktop terminal service, only SPEED CALLING - SYSTEM Access Code and Abbreviated Code are displayed on LCD. Speed Calling Number is not displayed.
   b.) Although Speed Calling Number Display on Desktop terminal service is available, SPEED CALLING-SYSTEM Access Code and Abbreviated Code are displayed on LCD as a dialed numbers memory, not Speed Calling Number.
   c.) When Speed Calling Number Display on Desktop terminal service is available, the number is displayed within 16 digits. More than 16 digits number is displayed as only the first 16 digits.
   d.) This feature is available only while calling the party. If the LCD is re-displayed with other feature such as Name Display-OAI-CCIS or the called party is being busy, this feature is not available.
   e.) When you use this feature, register the Speed Calling Number correctly in ASPD command. If the last digit part of number is not registered, Speed Calling Number is not displayed.
   (When calling “8-26-70000” with the use of this service, you may register “826” as a Speed Calling Number. In this case, press the Access Code and Abbreviated Code, and dial “70000”.
   At the time you call “7”, Speed Calling Number display becomes invalid.)
   f.) While calling with this feature, if you press the Abbreviated Call Codes, Speed Calling Number is not displayed.

Interactions

1. FORCED ACCOUNT CODES [F-7] and AUTHORIZATION CODES [A-20] cannot be programmed for use via the SPEED CALLING-SYSTEM.
2. Pauses are programmable from the PCPro using the “P” Key of the Keyboard. Pauses consume one digit of the available 24 digits. A pause lasts for about 3 seconds. A pause cannot be added directly after an LCR Access Code. A pause cannot be added to ISDN, or CCIS Calls.

   Note: A pause to MG-COT Calls can be added. In this case, a pause lasts about three seconds.

   Note: Registered pauses will not be activated when originating a call to ISDN Lines or to Analog Lines via MG-COT.

3. TOLL DIVERSION/TOLL DENIAL [T-6] and TOLL RESTRICTION 3/6 DIGIT [T-7] will interact with this feature if SPEED CALLING-SYSTEM is toll restricted.

4. Special Dial Tone may be returned after dialing SPEED CALLING-SYSTEM if AUTHORIZATION CODES [A-20] are operational in the system and Toll Restriction applies to SPEED CALLING-SYSTEM.
5. See SPEED CALLING OVERRIDE-SYSTEM [S-61].

Programming

STEP 1: **ASYD** - System Data 2, Index 1, Bit 7. Does TOLL RESTRICTION [T-7] apply to SPEED CALLING-SYSTEM? 0/1: No/Yes.

STEP 2: **<For Desktop terminal only>**
ASYDL - System Data 1, Index 673, bit 6. Speed Calling Number Display Service on Desktop terminal (Outgoing Call only) 0/1=Out/In Service

STEP 3: **ANPD** - Reserve a Number Level for Feature Access. Select Normal Service (N), Hooking Service (H) or Busy Service (B) from the combo box for Connection Indexes. Assign NND in accordance with a predetermined Numbering Plan.

STEP 4: **ASPA** - Assigning an Access Code to SPEED CALLING-SYSTEM, assign SRV = "SSC: Service Code", SID = 15. Program for "N: Normal" and "H: Hooking". NND = Access Code plus Abbreviated Digit Code (ADC). NND1 = Abbreviated Digit Code (ADC). See ASPD.

Example: 6 + 010 1 digit + 3 digits = NND = 4.

STEP 5: **ASFC** - Assign a Service Feature Restriction Class that allows SFI 12 to Stations that will have SPEED CALLING-SYSTEM.

STEP 6: **ASPD** - Enter Tenant Number (TN), Abbreviated Digit Code (ADC), and the Called Number (CD) to be sent. Include the Access Code of the route in the CD.

STEP 7: **AKYD** - Assign SPEED CALLING to Desktop terminal key. Assign FKY15 to the Desktop terminal key.

STEP 8: **ADSL** - Assign S-SPD Key; Assign SN = 1 (Originating; DT Connection) and FKY = 15 (Speed Calling - System).

This command is used for assigning a function to each Soft Key.

FCCS Programming

This data setting describes only the additional or replacement commands used to activate FCCS Services. When used as an FCCS feature, these commands replace the non-network commands (for example, AAED is replaced by AAEDN). Note that the data setting for FCCS Service varies depending on the Node.

Note: When using ASYDL or ASYDN to program an FCCS feature, the basic programming of ASYD is required in addition to ASYDL or ASYDN.

STEP 1: **ANPDL** - Reserve a Number Level for Feature Access. Select Normal Service (N), Hooking Service (H), or Busy Service (B) from the combo box for Connection Indexes. Assign NND in accordance with a predetermined Numbering Plan.

STEP 2: **ASPAL** - Assigning an Access Code to SPEED CALLING-SYSTEM, assign SRV = "SSC: Service Code", SID = 15. Program for "N: Normal". NND = Access Code plus Abbreviated Digit Code (ADC). See ASPD.

Example: 6 + 010 1 digit + 3 digits = NND = 4.

S-4 Splitting

General Description

This feature allows the ATTENDANT CONSOLE [A-3] to speak privately with one party on an Attendant-handled Connection without the other party overhearing.

Operating Procedure

To speak with the Called Party only

1. Dial the desired Telephone Number.
2. Station class/Telephone Number is displayed.
3. Wait for the party to answer.

To speak with the Calling Party only

1. Press the SRC Key.
2. Trunk Group/Number is displayed.
3. Proceed with the conversation.

To return to the Called Party

1. Press the DEST Key.
2. Station class/number displayed.

To speak with both parties

1. Press the TALK Key.

To release from the Attendant Console

1. Press the RELEASE Key.

To disconnect all parties involved in a THREE-WAY CONFERENCE [T-2]

1. Press the DEST Key.
2. Press the CANCEL Key twice.

Service Conditions

1. The ATTENDANT CONSOLE [A-3] may alternate between the Called and Calling Parties, and THREE-WAY CONFERENCE [T-2], as desired.
2. If SERIAL CALL [S-15] is provided, the call is automatically split when the ATTENDANT presses the Loop Key to answer a recall initiated by a Station user.
3. The SPLITTING feature is a standard Attendant feature.
4. SPLITTING is automatic when the Attendant begins call completion or answers a recall.
5. SPLITTING is manual when the SRC, DEST, or TALK Key is pressed on the ATTENDANT CONSOLE [A-3].
6. If a Station is engaged in a SERIAL CALL [S-15] with an Outside Party and LOCKOUT [A-5] is not provided (optional), upon pressing the associated Loop Key, will bridge on the existing Connection, and SPLITTING is not provided.

Interactions

1. The ATTENDANT CONSOLE [A-3] must first be provided.
2. If Conference Calling is required, THREE-WAY CALLING [T-2] must first be provided.

Programming

No programming is required.

S-4D Splitting -D

General Description

This feature allows a Dterm user to alternately converse between two separate parties while one party is connected and one party remains on hold. The Station user uses the TRANSFER button to alternate conversations between the two parties.

Operating Procedure

While a conversation is in progress

1. Press the TRANSFER Key. Hear Special Dial Tone.
2. Dial the new number. The first party is held, the Second Party answers.
3. Press the TRANSFER Key. Return to the first party; the Second Party is held.
4. Repeat as often as needed.

Service Conditions

THREE-WAY CALLING [T-2] may be established at any time by pressing the CONF Key.

Interactions

1. SPLITTING is inherent to the system and is operated by using the TRANSFER Key.
2. The Conference Trunk is only required when a Three-way Connection is desired.

Programming

STEP 1: **ASYD** - System Data-2, Index 1, Bits 0 and 1. Consultation Hold allowed or denied. 00: Denied; 01: Originating and Terminating Calls allowed, Tandem Call Denied; 11: All calls allowed. Assign data "11".

System Data 2, Index 6, Bit 2. Leave as data "0" to enable the CALL TRANSFER-ALL CALLS [C-11] feature.

STEP 2: **ATRK** - Assign these LENS as Route 909, Digital Conference Trunk. LENS consists of the following:

XX	X	XX	X
Module Group	Unit	Group	Level

Group and Level numbers for the Digital Conference Trunk are 210, 213, 216, 221, 224, 227, 232, and 235.

STEP 3: **MBTK** - Make idle the Digital Conference Trunk.

S-6 Station-Controlled Conference

General Description

This feature allows a Station user to establish a Conference. The Conference may be any combination of Stations and/or Trunks (inside and outside parties). The number of conferee per conference group varies depending on the Conference Trunk (CFT) as shown below:

CFT	NUMBER OF CONFEREES (including conference leader)
PA-CFTW	8-party × 2 groups
PA-CFTV	8-party × 4 groups
	16-party × 2 groups
	32-party × 1 group
CH-CFTA	8-party × 4 groups
	16-party × 2 groups
	32-party × 1 group

Operating Procedure

Procedure 1 (MEET-ME CONFERENCE)

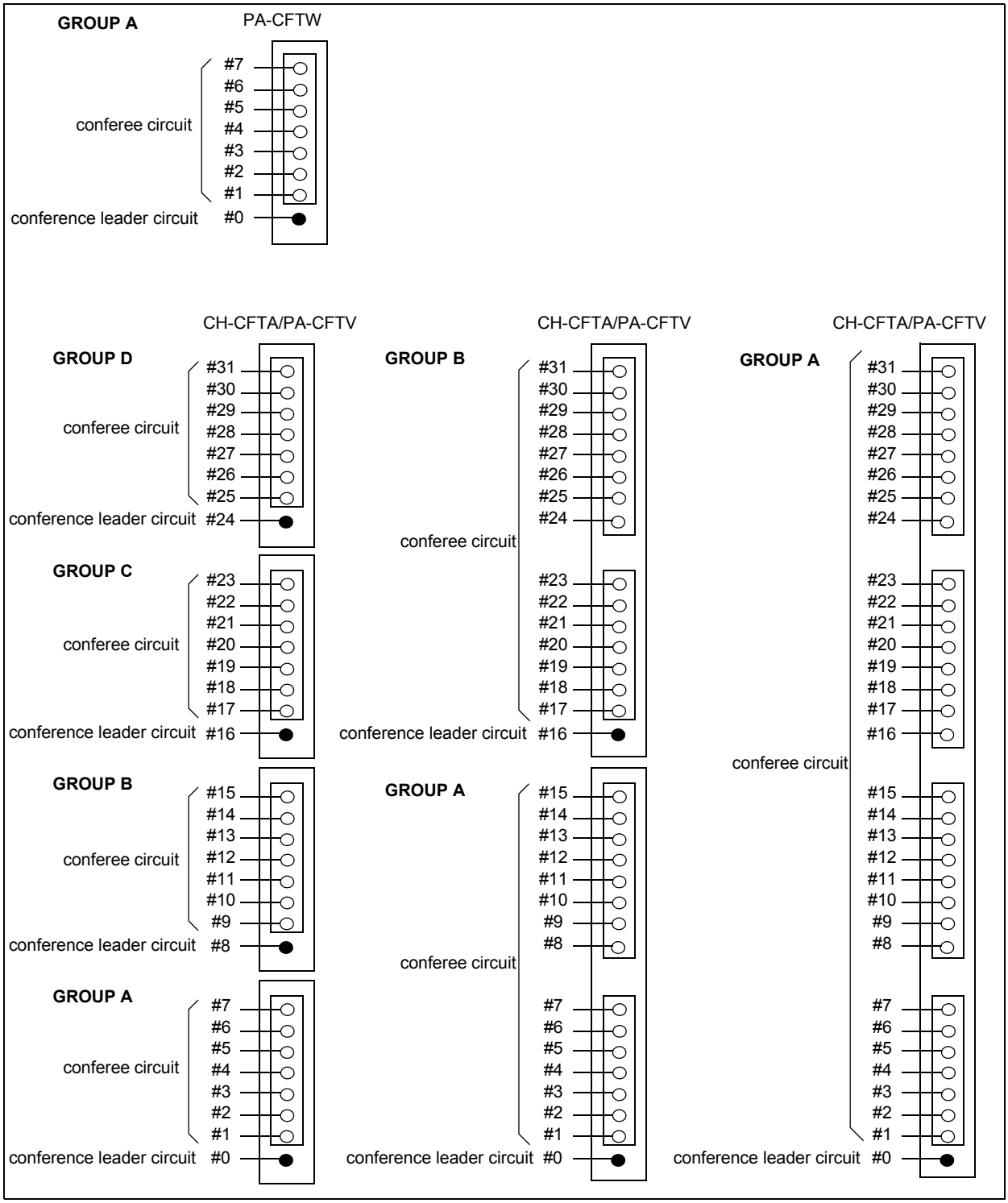
1. Station user wishing to establish the Conference (Conference Leader) calls each party desired and advises them of the Conference.
2. All Stations then call the Conference Telephone Number, and the Conference Leader Calls the special Conference number for the leader.
3. Conference Connection is automatic.
4. The Conference Leader may dial "0" to lock out additional parties.
5. If the Conference Leader connected to the Conference Circuit dials digit "#", all Conferees will be disconnected. Otherwise, the Conference is terminated only after all parties have hung up.
6. To release from a Conference Circuit individually, a Station user hangs up. The remaining Conferees may continue without interruption.

Procedure 2

1. Conference Leader dials the Conference Telephone Number on the Conference; Conference Circuit automatically answers.
2. Conference Leader makes a Switch Hook Flash (SHF).
3. Conference Leader dials the Telephone Number of conferee; receive Ringback Tone.
4. The conferee answers.
5. Conference Leader hangs up; the conferee is connected to the Conference Circuit (port 1).
6. Conference Leader sequentially calls all desired parties (up to 7, 15, or 31 parties) and connects them to the other Conferee Circuits (port 2 to 7/15/31).
7. Conference Leader dials the Conference Telephone Number on the Conference again.
8. Conference Leader may dial "0" to lock out additional parties.
9. If Conference Leader connected to the Conference Circuit dials digit "#", all Conferees will be disconnected. Otherwise, the Conference is terminated only after all parties have hung up.
10. To release from a Conference Circuit individually, the Station user hangs up. The remaining Conferees may continue without interruption.

Service Conditions

1. The Station connected to port #0 of PA-CFTW, and port #0/#8/#16/#24 of CH-CFTA/PA-CFTV becomes the Conference Leader.



2. While the Conference Unit provides for amplification, the condition of Central Office Connections will affect Conference transmission.

3. Each Conference Unit provides up to an/four 8-Party Conference, two 16-Party Conference, or 32-Party Conference.
4. Conference Unit extension numbers should be placed in a STATION HUNTING [S-7, 8, 9] Group with only the pilot numbers published.
5. Override services, such as, BUSY VERIFICATION [B-3], ATTENDANT OVERRIDE [A-7], or EXECUTIVE RIGHT-OF-WAY [E-1] are denied on a Conference Call.
6. You can disconnect the conference call by going on-hook.
7. When a Conferee exits the conference call, the other members may hear Reorder Tone (ROT) for a short time.
8. While a Desktop terminal is connected to one of conference ports in a conference trunk, if a DSS key where a station number for another conference port is assigned is pressed by mistake, the call is transferred to the conference port whose station number is assigned to the DSS Key. Even after the Desktop terminal is released, the connection between these conference ports is retained, and the conference trunk becomes unavailable.
   To prevent this, do not assign station numbers for conference ports on any DSS keys.

   Note: When assigning station numbers for conference ports on DSS keys is required for any operational reasons, set different tenant numbers to a Desktop terminal and a conference trunk respectively, and restrict TRI=0 (station within OGTN calls to a station within TMTN) in the ATNR command for tenants where the conference ports belong.

Programming

STEP 1: **ASYD** - Assign System Data, SYS1.
Index 20, Bit 6=0 (Line Circuit Reverse Relay Control when Reorder Tone is connected to a station/trunk.)
Index 64, Bit 0: Assign the compression method (0/1 = μ-law/A-law)

STEP 2: **ANPD** - A Number Level must be reserved for Station Numbering Plan. In an existing system, a Station Level will already be established. If a vacant Level does exist, it may be used to create two or three digit Station Numbering. For example:
TN: 1 DC: 3 CI: Select Normal Service (N), Hooking Service (H) or Busy Service (B) from the combo box for Connection Indexes.NND: 1BLF: 1-On

STEP 3: **ASPA** - For the desired Station Numbering assign SRV = “STN: Station Number”. For example:
ACC: 3
CI: “N: Normal”, “H: Hooking”, “B: Busy”
SRV = “STN: Station Number”

STEP 4: **ASDT** - Assign all ports of the CH-CFTA/PA-CFTW/PA-CFTV as DP/PB Stations; TEC = 3. Based on the example data assignments in ANPD, the Station Numbers will reside in Level 3.

Example data assignments. (CH-CFTA located in PIR 0, Unit=1, Group = 00):

CH-CFTA Port	LENS	Station Number	TEC	SFC	RSC	Remarks
0	001000	3200	3	1	1	Conference Leader
1	001001	3201	3	1	1	Conference Participant Ports
2	001002	3202	3	1	1	
3	001003	3203	3	1	1	
↓	↓	↓	↓	↓	↓	
28	001034	3228	3	1	1	
29	001035	3229	3	1	1	
30	001036	3230	3	1	1	
31	001037	3231	3	1	1	

STEP 5: **ASHP** - Set the type of Station Hunting to “Pilot” for stations connected to conferee ports on the CH-CFTA/PA-CFTW/PA-CFTV card.

Example (CH-CFTA): Levels 1-7 (all the conferees ports)

TN: 1
STN: 3201
CNT: 7
STN: 3202, 3203, 3204, 3205, 3206, 3207... 3221

In this example, Ext. 3201 is the Pilot number of the Conference Bridge. Do not assign the Conference Leader to this group. Before or after all Conference participants are connected to the CFT, the Conference Leader directly calls the Conference Leader Port; for this example Ext. 3200.

FCCS Programming

This data setting describes only the additional or replacement commands used to activate FCCS Services. When used as an FCCS feature, these commands replace the non-network commands (for example, the AAED command is replaced by the AAEDN command). Note that the data setting for FCCS Service varies depending on the node.

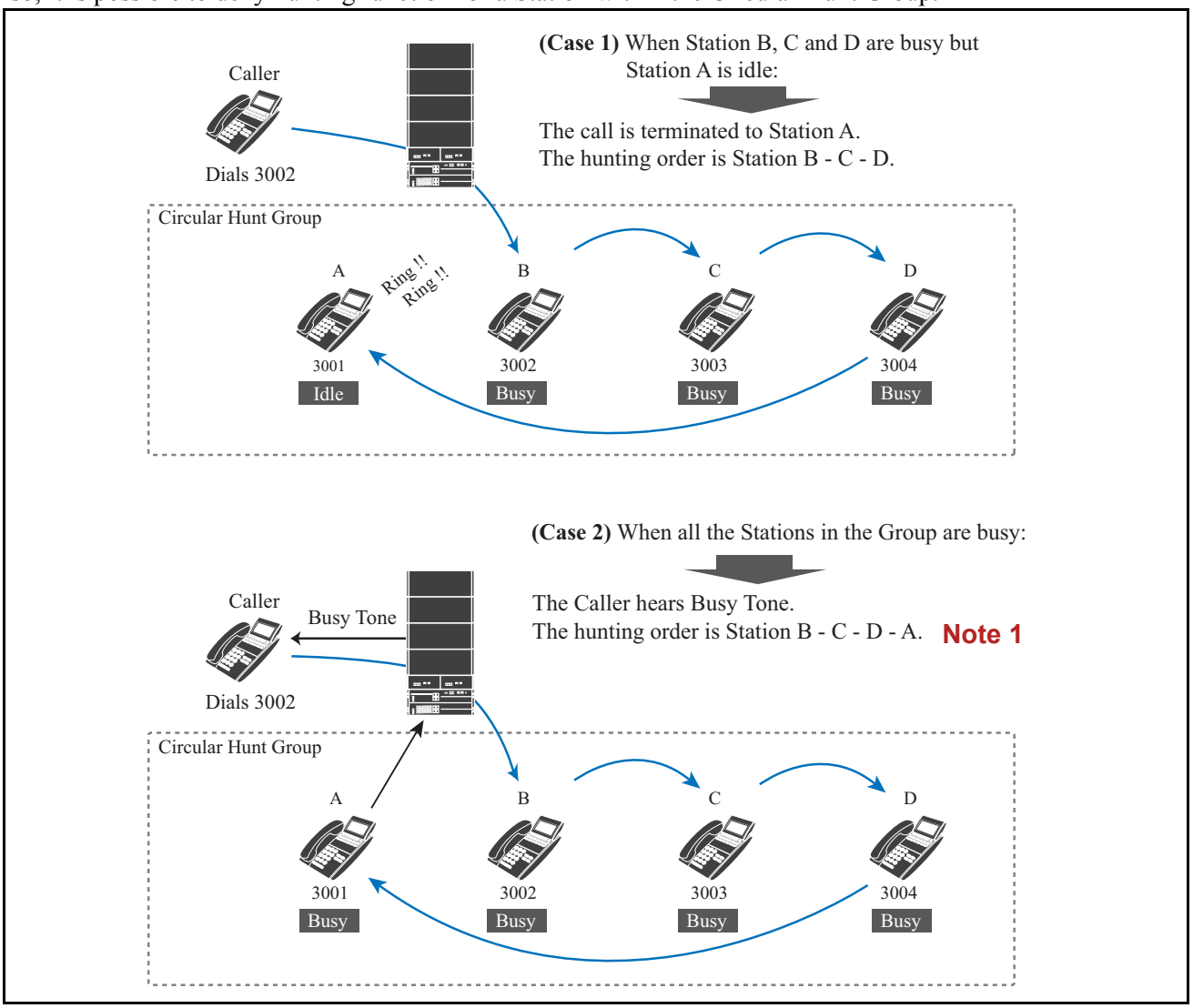
STEP 1: **ALGSN** - Assign a Telephone Number to the entry of the physical station data created with the ASDT command.

STEP 2: **ASHPN** - Set the type of Station Hunting to “Pilot for NDM” for stations connected to conferee ports on the CH-CFTA/PA-CFTW/PA-CFTV card.

S-7 Station Hunting - Circular

General Description

This feature permits a call directed to a Busy Station of a Circular Hunt Group to be automatically forwarded to other Stations within the Hunt Group, in a programmed order. The call arrives to one of the Stations of the group and, if it is in a busy state, the call is transferred to the next Station in the programmed order until it finds an idle Station that rings. For example, as illustrated in the following figure, if a call is placed to Station B and Station B is busy, the call is forwarded to Station C and so on. If all Stations are busy, the caller hears Busy Tone.
Also, it is possible to deny hunting function for a Station within the Circular Hunt Group.



Note 1: If a secretarial Station is assigned for the Circular Hunt Group and all the Stations in the Circular Hunt Group are busy, the call is forwarded to the secretarial Station. For the secretarial Station, see STATION HUNTING - SECRETARIAL [S-8].

Note: In this feature, Busy means the Station cannot receive a call (because of reasons such as talking, receiving other call, do not disturb being set).

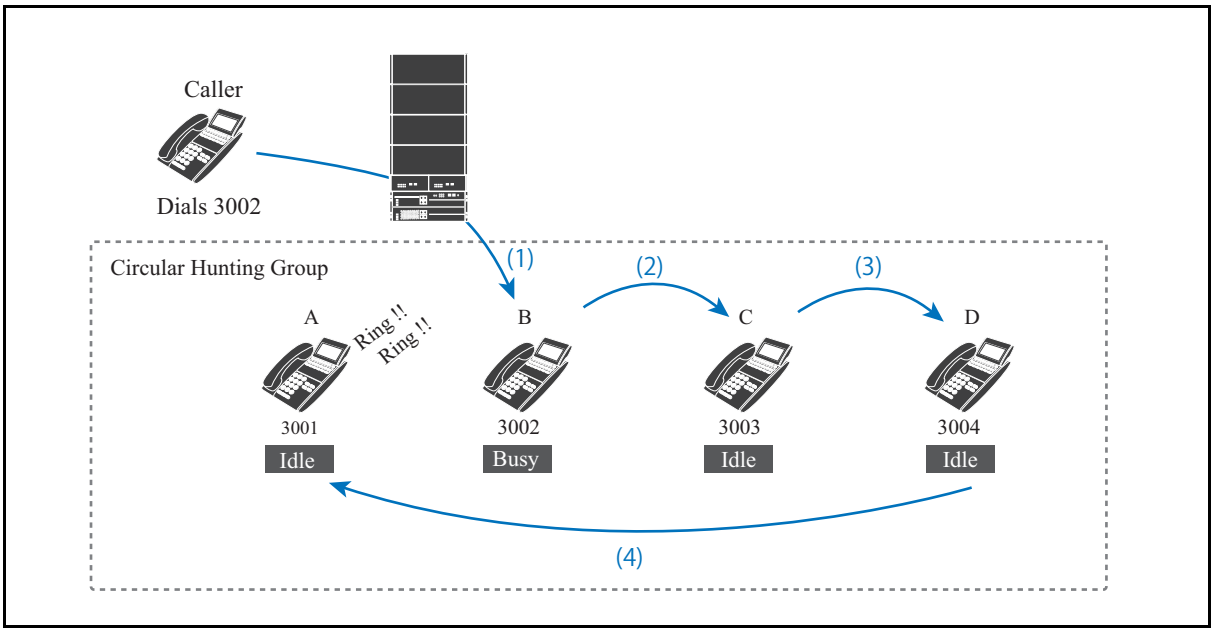
- TRANSFER OF UNANSWERED CALLS IN A HUNT GROUP

When the caller dials a Station in the Circular Hunt Group, the call arrives to a Station within the Circular Hunt Group, and the Station does not reply within a programmed period of time, the call can be transferred to another idle Station.

Note: Transfer of unanswered calls is available from FP95-114 V4 Issue 4.0.

As for the transfer of an unanswered call in a Circular Hunt Group, the figure below gives an example for the transfer pattern with the following steps.

1. A call is made to Station B.
2. Station B is busy, the call is transferred to the first idle station in the hunting order (Station C).
3. The call remains unanswered within a programmed period of time and is transferred to the next idle Station in the hunting order (Station D).
4. The call remains unanswered within a programmed period of time and is transferred to the next idle Station in the hunting order (Station A).



Note: In the example above, if all the Stations enter in a busy state, the call is transferred to a Secretarial Station if it has been set. For the Secretarial Station, see STATION HUNTING - SECRETARIAL [S-8].

Operating Procedure

No manual operation is required.

Service Conditions

<Important Notices>

1. Station Number can belong to one Hunt Group only, it cannot belong to another Hunt Group including that for STATION HUNTING - CIRCULAR [S-7], STATION HUNTING - TERMINAL [S-9], STATION HUNTING - PRIORITY [S-138], STATION HUNTING - PRIORITY WITH SWITCHBACK [S-139] or UNIFORM CALL DISTRIBUTION (UCD) [U-1].
2. A station can be assigned to a different Hunt Group by the Physical Station Number (DM) and the Telephone Number (LDM or NDM). For that reason, a Station already assigned to a Hunt Group that uses Physical Station Number (DM) can be also assigned to other Hunt Group that uses the Telephone Number (LDM or NDM). Also, when a Physical Station Number (DM) and Telephone Number (LDM or NDM) are identical, priority is attached to the Telephone Number (LDM or NDM). However, it is not possible to assign the same Physical Station Number or the same Telephone Number to more than one Hunt Group.

<Interactions with Other Features>

1. If the Station of a Hunt Group is in the following states or the following features are set in the Station of a Hunt Group, the following occurs.

   - **A**: The feature is activated and the call is forwarded.
   - **S**: The Station is skipped.

Feature/State	Direct Call to the Station	Call Transfered to the Station in the Hunting
CALL FORWARDING - ALL CALLS [C-5]	**A** Note 2	**S**
DO NOT DISTURB [D-11]	**S**	**S**
BUSY STATION	**S**	**S**
BUSY STATION + CALL FORWARDING - BUSY LINE [C-2]	**A** Note 2, Note 3	**S**
BUSY STATION + CALL FORWARDING - BUSY LINE [C-2] set with the ACFO/ACFOL/ACFON command (on a tenant/route basis)	**S**	**S**
LOGGED OUT STATION	**S**	**S**
LOGGED OUT STATION + LOGGED OUT IP STATION - CALL DESTINATION [L-53]	**A**	**S**
LOGGED OUT STATION + LOGGED OUT IP STATION - CALL DESTINATION [L-53] set with the ACFO/ACFOL/ACFON command (on a tenant/route basis)	**S**	**S**

Note 2: If CALL FORWARDING - OVERRIDE [C-26] is set to the forwarding destination, the call forwarding overriding from the forwarding destination to the Station of the Hunt Group will hunt if the Station of the Hunt Group is busy.

Note 3: When a call to a station in a Hunt Group is forwarded by CALL FORWARDING - BUSY LINE [C-2] and the forwarded party (also a member of another Hunt Group) is also busy, the resultant hunt target can be designated from the forwarding side Hunt Group or the forwarded side Hunt Group. This can be specified with ASYD/ASYDN SYS1 Index 69 Bit 5.

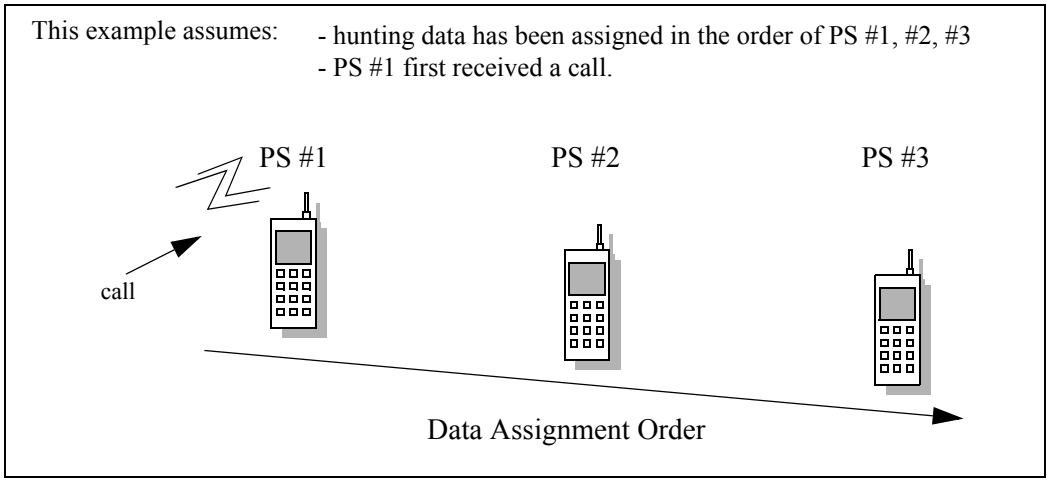
2. A Phantom Number can be assigned to the Station of a Hunt Group. However, when a caller dials the Phantom Number, hunting does not occur for the Station that receives the call.
3. Recall to a Station in the Hunt Group from the following features will not hunt:
   - CALL BACK [C-1]
   - CALL HOLD [C-6]
   - OUTGOING TRUNK QUEUING [O-2]

- PAGING TRANSFER [P-10]

4. For STATION MESSAGE DETAIL RECORDING [S-10] the number recorded is the Station Number where the call actually terminates.
5. HOTLINE [H-1] cannot be assigned to the Station of a Hunt Group.

<Hardware/Software>

1. A call from ATTENDANT CONSOLE [A-3] will not hunt.
2. The conditions when using PS are as follows:
   a.) If a call terminates to a PS in the Hunt Group while the PS is Out of Zone or its power is OFF, whether the call is processed as PS Out of Zone or hunting is executed is determined by System Data (ASYDL, SYS 1, Index 688, b0). When using this feature in an FCCS environment, assign the same data value to the system data in each node.
   b.) When hunting is selected in the system data, the caller hears no tone until the hunted PS answers. Total length of no tone is obtained by the following.
      - Time for Detecting Out of Zone (Maximum eight seconds) × Number of Hunting
   c.) When a Desk Console operator dials the number of PS that belongs a Hunt Group, the hunting is not available if the PS is Out of Zone.
   d.) When receiving an analog PSTN incoming call while all PSs in the Hunt Group are Out of Zone, the call processing will be stopped after detecting Out of Zone of the last PS in the Hunt Group. At this time, the calling party will receive RBT when the call is a Direct In Termination call. If Automated Attendant is used, the calling party will receive ROT after No Tone. Note that the call cannot be terminated to the destination even if the PS returns to the zone during RBT.
   e.) When receiving an ISDN incoming call while all PSs in the Hunt Group are Out of Zone, hunting starts after calling up the destination. However, timeout will occur at Central Office side, resulting in ROT connection (hunting is not available).
   f.) If all Stations in the Hunt Group are busy or PS Out of Zone, the call can be forwarded a Station in the group. The station is called secretarial Station. The caller hears Busy Tone if the secretarial Station is also busy.
   g.) A PS can be assigned as a secretarial Station.
   h.) If a PS assigned as a secretarial Station is placed in Out of Zone, the caller hears Busy Tone.
   i.) A PS in the Hunt Group can set PS Out of Zone - Call Forwarding. When the PS has been set as able of hunting, the incoming call will be forwarded only if the call cannot terminate to any stations in the group. When the PS has not been set as able of hunting the call can be forwarded only if the PS is directly called.
   j.) When a PS in the Hunt Group performs roaming, the PS is skipped. However, the PS is a station that receives the call directly, it will be roaming call termination.
   k.) If a PS receiving a call turns to Out of Zone, this may cause a network failure. In this case, the caller hears Busy Tone.
   l.) The following shows call processing when Out of Zone or busy is involved for this feature.



When Hunting is selected (ASYDL, SYS1, INDEX 688, Bit 0 = 0)

PS #1	PS #2	PS #3	Processing
Out of Zone (Forwarding has been set)	Out of Zone	Out of Zone	After calling up PS #1, PS #2, PS #3, the forwarding destination is called.
Out of Zone (Forwarding has not been set)			After calling up PS #1, PS #2, PS #3, the caller receives ROT or Automatic Announcement Called PS Out of Zone.
Busy	Out of Zone	Out of Zone	After calling up PS #2 and PS #3, the caller receives Busy Tone.
Busy	Busy	Out of Zone	After calling up PS #3, the caller receives Busy Tone.

When Out of Zone Processing is selected (ASYDL, SYS 1, INDEX 688, Bit 0 = 1)

PS #1	PS #2	PS #3	Processing
Out of Zone (Forwarding has been set)	Out of Zone	Out of Zone	After calling up PS #1, the forwarding destination is called.
Out of Zone (Forwarding has not been set)			After calling up PS #1, the caller receives ROT or Automatic Announcement Called PS Out of Zone.
Busy	Out of Zone	Out of Zone	After calling up PS #2, the caller receives Busy Tone.
Busy	Busy	Out of Zone	After calling up PS #3, the caller receives Busy Tone.

<Conditions Specific to This Feature>

1. The maximum number of Stations per group is 100. The feature is usable with all Stations and there is no limit on the allowable number of Hunt Groups within a system.

2. This feature enables a Station within Hunt Group to deny hunting function for an Incoming Call originated from Station (including PS) (ASFC/ASFCN SFI 181 =1). The service conditions are as follows:
   a.) When CALL FORWARDING-DON'T ANSWER [C-5] is set, a call to the Station is forwarded according to their service condition.
   b.) When CALL FORWARDING-BUSY LINE [C-2] is set, a call to the Station is forwarded according to their service condition.
   c.) A call to the Phantom Number assigned to the Station will not hunt.
   d.) External Trunk (such as DDD (TCL = 1)/FX (TCL = 2)/WATS (TCL = 3), etc) is not available.

3. The conditions for the transfer of unanswered calls in a Hunt Group feature are as follows: **Note 4**

Note 4: Available from FP95-114 V4 Issue 4.0.

   a.) The feature is only possible for Hunt Groups with data assigned via Telephone Numbers (LDM or NDM).
   b.) Be aware that if there is a value set in Timer Value of No Answer Timer set in the ADAI command for a Station in the group it will have priority over the value set with ASYDL/ASYDN, SYS1 Index 1396. If no value is set for ADAI command or ASYDL/ASYDN, SYS1 Index 1396, other Timer Value of No Answer Timer (ASYD SYS1, Indexes 139, 140, 141, 247) will be used.
   c.) The display of a Station receiving an incoming call transfer based on this feature will be the same as in the case of CALL FORWARDING - DON'T ANSWER [C-3]. Also, in the case of other external devices, it will be treated in the same way.

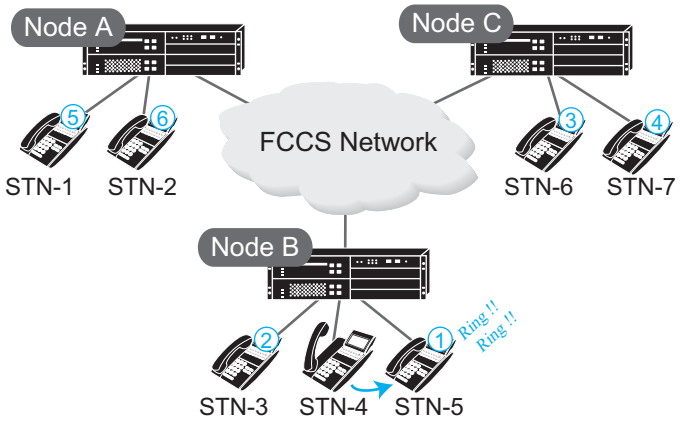
d.) If other unanswered calls transfer features are set in a Station of this feature group, those other unanswered calls transfer features have priority over the transfer of unanswered calls in a Hunt Group feature.

Priority Order	Transfer of Unanswered Calls in a Hunt Group Feature	
	Out of Service	In Service
1	CALL FORWARDING - DON'T ANSWER [C-3]	CALL FORWARDING - DON'T ANSWER [C-3]
2	-	CALL FORWARDING - BUSY LINE [C-2] **Note 5**
3	-	TRANSFER OF UNANSWERED CALLS IN A HUNT GROUP
4	CALL FORWARD BY SERVICE FEATURE CLASS [C-172]	CALL FORWARD BY SERVICE FEATURE CLASS [C-172]
5	CALL FORWARDING - DON'T ANSWER [C-3] set with the ACFO/ACFOL/ACFON command (on a tenant/route basis)	CALL FORWARDING - DON'T ANSWER [C-3] set with the ACFO/ACFOL/ACFON command (on a tenant/route basis)

Note 5: The Station with the incoming call is treated as Busy Station and the call is transfered.

FCCS Conditions

1. When activating this feature between nodes via FCCS, use only Hunt Groups assigned with Telephone Number (NDM).
2. Stations in up to four different nodes can be assigned to a Hunt Group.
   When the Hunt Group includes more than one node, the [Group by FPC] button of ASHCN must be used to set the order of the nodes in the Hunt Group. Without this node order grouping assignment, data cannot be assigned.
3. When Stations belonging to different nodes are assigned to a Hunt Group, the assigned order of the Stations and the actual order of the hunting may be different. The below explains the hunting order following an FCCS operations example.
   a.) Hunting takes place inside the node of the Station that first received the call (Node B).
   b.) In the case that not all the Stations inside the node (Node B) of the Station that first received the call are busy, hunting takes place inside the node (Node B) of the Station that first received the call.
   c.) In the case that all the Stations inside the node (Node B) of the Station that first received the call are busy, hunting takes place in the next node (Node C) of the office data order.



	Node A		Node B			Node C	
	STN-1	STN-2	STN-3	STN-4 Note 6	STN-5	STN-6	STN-7
Assigned Hunting order by the office data	1	2	3	4	5	6	7
Actual Hunting order	5	6	2	(Busy)	1 (Ringing)	3	4

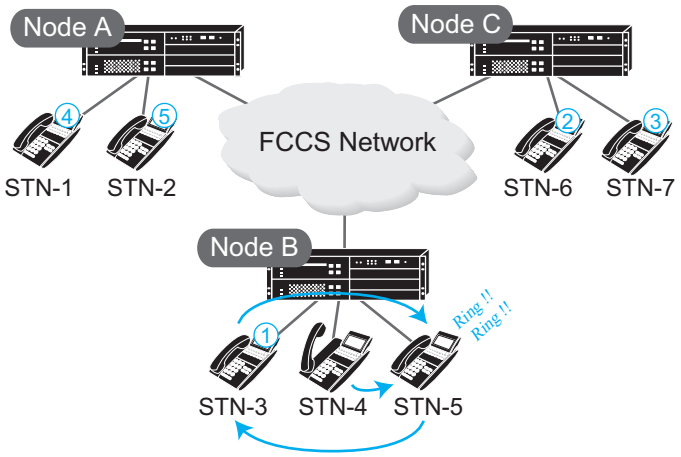
Note 6: This is the Station that first receives the call.

Note: The below explains the hunting order following an FCCS operations example when the transfer of unanswered calls in a Hunt Group is used.

a.) Hunting and transfer of unanswered calls takes place inside the node of the Station that first received the call (Node B).

b.) In the case that not all the Stations inside the node (Node B) of the Station that first received the call are busy, hunting and transfer of unanswered calls takes place inside the node (Node B) of the Station that first received the call.

c.) In the case that all the Stations inside the node (Node B) of the Station that first received the call are busy, hunting and transfer of unanswered calls takes place in the next node (Node C) of the office data order.



	Node A		Node B			Node C	
	STN-1	STN-2	STN-3	STN-4 Note 7	STN-5	STN-6	STN-7
Assigned Hunting order by the office data	1	2	3	4	5	6	7
Actual Hunting order (incoming call)	4	5	1	(Busy)	(Ringing)	2	3
Actual Hunting order (transfer 1)	4	5	(Ringing)	(Busy)	1	2	3
Actual Hunting order (transfer 2)	4	5	1	(Busy)	(Ringing)	2	3

Note 7: This is the Station that first receives the call.

4. The Incoming Call terminated between nodes via FCCS to a Station in the Hunt Group programmed by the Physical Station Number, is not hunted.

Programming

<When using Physical Station Numbers for creating the Hunt Group>

STEP 1: **ASYD** - Assign the following system data.
System Data 1, Index 69, Bit 5: Hunt Group when transferred party is busy (Station Hunting after Call Forwarding.-Busy Line):
0 = Hunt in Transferring Party's Hunt Group
1 = Hunt in Transferred Party's Hunt Group

STEP 2: **ASHC** - Assign the required data for the Hunt Group.
TN: Tenant Number
STN: Physical Station Number **Note 9**
SECRETARY STN:Physical Station Number of the Secretarial Station **Note 8**
[Station List in Group]
STN: Physical Station Numbers of the Stations included in the Hunt Group. **Note 9**

Note: Assignment of Numbers to a Circular Hunt Group may be in any order.

Note 8: If secretarial Station is used, assign the Physical Station Number of the Station used as such secretarial Station.

Note 9: Do not enter Phantom Number.

STEP 3: **ASFC** - (Optional) Assign the following Service Feature Restriction when hunting must be denied.
SFI 181 =1 (No Hunting to next station in group)

<When using Telephone Numbers for creating the Hunt Group>

Note: The following programming supposes that Telephone Numbers have been already assigned and does not include such data settings.

STEP 1: **ASYD/ASYDN** - Assign the following system data.
System Data 1, Index 69, Bit 5: Hunt Group when transferred party is busy (Station Hunting after C.F.-Busy Line):
0 = Hunt in Transferring Party's Hunt Group
1 = Hunt in Transferred Party's Hunt Group

STEP 2: **ASYDL** - (Optional) Assign the following System Data.
System Data 1, Index 688, Bit 0. Specify whether hunting is activated or not if the called PS is Out of Zone.
0 = Hunting
1 = PS Out of Zone processing

Note: Assign this data if PS is used in the Hunt Group.

STEP 3: **ASHCL/ASHCN** - Assign the required data for the Hunt Group.
UGN: User Group Number
TELN: Telephone Number in the Hunt Group **Note 11**
SECRETARY TELN:Secretary Telephone Number **Note 10**
[Station List in Group]
TELN: Telephone Numbers of the Stations included in the group. **Note 11**

Note: Assignment of Numbers to a Circular Hunt Group may be in any order.

Note 10: If secretarial Station is used, assign the Telephone Number of the Station used as such secretarial Station.

Note 11: Do not enter Phantom Number.

STEP 4: **ASFCN** - (Optional) Assign the following Service Feature Restriction when hunting must be denied.
SFI 181 =1 (No Hunting to next station in group)

<When activating the transfer of unanswered calls in a Circular Hunt Group> Note 12

STEP 5: **ASYD** - Assign the following System Data.
System Data 1, Index 69, Bit 0= 1 (No Recall, or execution of Call Forwarding –Don't Answer (when CF – Don't Answer is set at the transfer destination station)).
System Data 1, Index 69, Bit 1 (Send short tone when a call forwarded via CF – All Calls/Logout service is answered.)
0 = Not Required
1 = Required
System Data 2, Index 6, Bit 7 (CF-Don't Answer service when Attendant Console transfers call before

called station answers)
0 = Out of service (Recall to Attendant Console)
1 = In service

STEP 6: **ASYDL/ASYDN** - Assign the following System Data.
System Data 1, Index 1395, Bit 1=1 (transfer of unanswered calls in a Circular Hunt Group in service). **Note 12**
System Data 1, Index 1396= set the no answer timer for the transfer of unanswered calls in a Hunt Group. **Note 12**
Bit 0 - 3: 1 - F= MTC
Bit 4 - 6: 0 - 7= TC
TC: Timer Class, MTC: Miscellaneous Timer Counter
TC = 0: -
1: -
2: -
3: 2 seconds
4: 30 seconds
5: -
6: -
7: -

Note: Default value for this data is HEX 00 (data is not set). If data is not set, other Timer Value of No Answer Timer (ASYD SYS1, Indexes 139, 140, 141, 247) will be used.

Note: Do not assign "0" for MTC value when TC = 3 or 4.

Note: Assigned time will be used for all the Hunt Groups to which is available (Circular, Terminal, Priority), it cannot be set separately according to the type of Hunt Group.

Note: The time will be used for the transfer of unanswered calls regardless of whether transfer of unanswered calls in a Circular Hunt Group is in service or not.

Note 12: Available from FP95-114 V4 Issue 4.0.

STEP 7: **ASFC/ASFCN** - Assign the following Service Feature Restrictions.
SFI 8=1 (Call Forwarding-Don't Answer allowed).
SFI 103 (Assignment of No Answer Timer for Blind Transfer to Station/Blind Transfer to Attendant) **Note 13**
0 = Restricted (settings in ASYD, System Data 1, Index 140)
1 = Allowed (settings in ASYD, System Data 1, Index 247)
SFI 104 (Processing when the transferred station does not answer for a predetermined period of time in Blind Transfer) **Note 14**
0 = Transferred station continuously rings
1 = The transferred call is forwarded when Call Forwarding-Don't Answer is assigned at the transferred station and no-answer timer expires

Note 13: The time set for a Station with ASYDL/ASYDN System Data 1, Index 1396 (no answer timer for the transfer of unanswered calls in a Hunt Group) will be given priority.

Note 14: Set to 1 if the terminal in the Hunt Group is the destination of BLIND TRANSFER TO STATION [B-18].

STEP 8: **ATNR** - Assign the following Tenant Restriction Class.
TRI 0=1 (Station within OGTN calls to a station within TMTN allowed).
TRI 1=1 (Station within OGTN sets Call Forwarding-All Calls / Busy Line / Don't Answer/Logout to a station within TMTN allowed).

STEP 9: **ACFR/ACFRN** - Assign the following Call Forwarding Restriction:
CCI 8=1 (Call Forwarding-Don't Answer allowed).

S-8 Station Hunting - Secretarial

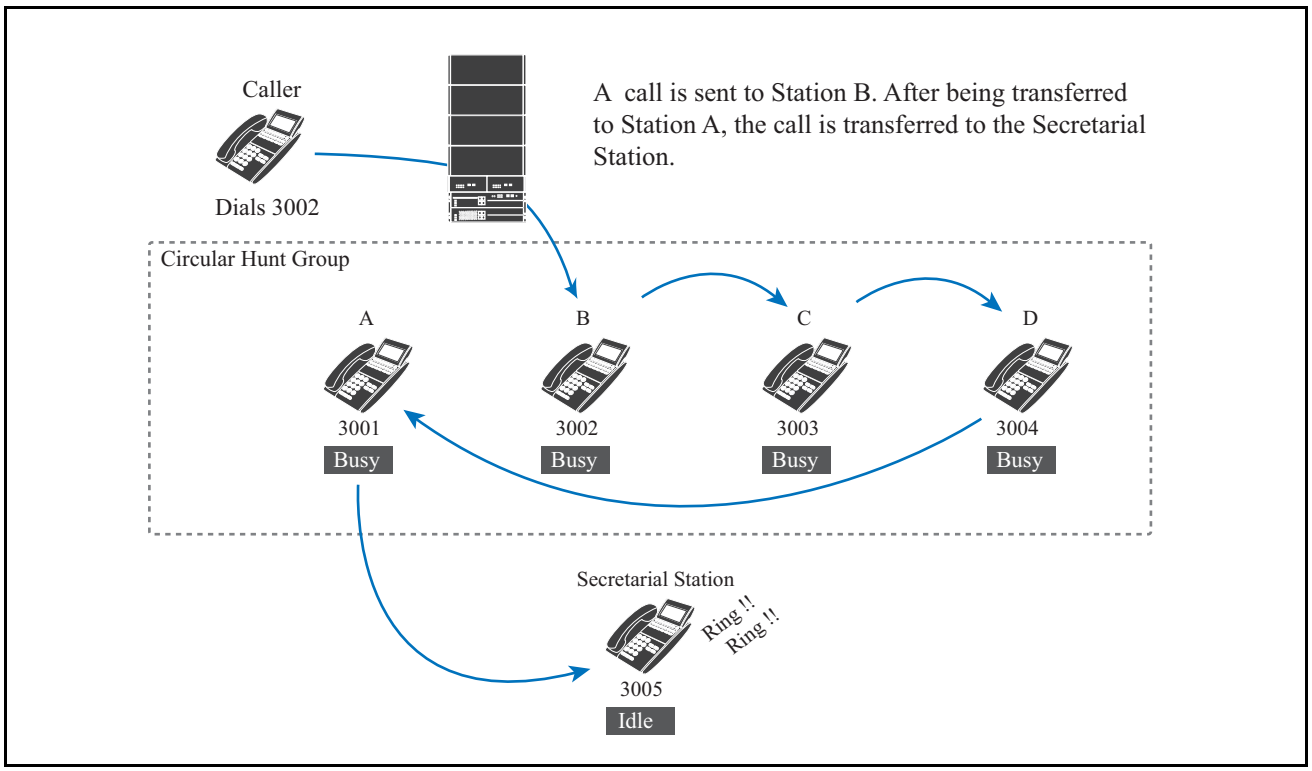
General Description

This feature, when all the Stations in the Hunt Group are busy, allows to the call of a Hunt Group to be hunted by one Station outside the Hunt Group. This Station is called the secretarial Station. If all the Stations in the Hunt Group and the secretarial Station are busy, the caller hears Busy Tone. The types of Hunt Groups that can use a secretarial Station are as follows:

- STATION HUNTING - CIRCULAR [S-7]
- STATION HUNTING - TERMINAL [S-9]
- STATION HUNTING - PRIORITY [S-138]
- STATION HUNTING - PRIORITY WITH SWITCHBACK [S-139]

Note: This feature is not a Boss - Secretary feature.

The following is an example of a secretarial Station used in a Circular Hunt Group.



Operating Procedure

No manual operation is required.

Service Conditions

1. A Hunt Group (Terminal, Circular, Priority, Switchback) may have one secretarial Station. A Station already assigned to a Hunt Group cannot be assigned as the secretarial Station outside the Hunt Group.
2. The secretarial Station can be shared with multiple Hunt Groups.

3. The maximum number of secretarial Stations that can be assigned per system differs depending of using the Physical Station Number (DM) or the Telephone Number (LDM or NDM).
    - Physical Station Number (DM): 256 secretarial Stations
    - Telephone Number (LDM or NDM): 4,096 secretarial Stations
4. The secretarial Station is not counted as one of the 100 allowed Stations per Hunt Group.
5. As the Hunt Groups can be created with the Physical Station Number (DM) or the Telephone Number (LDM or NDM), the secretarial Station must be respectively assigned with the Physical Station Number (DM) or the Telephone Number (LDM or NDM).

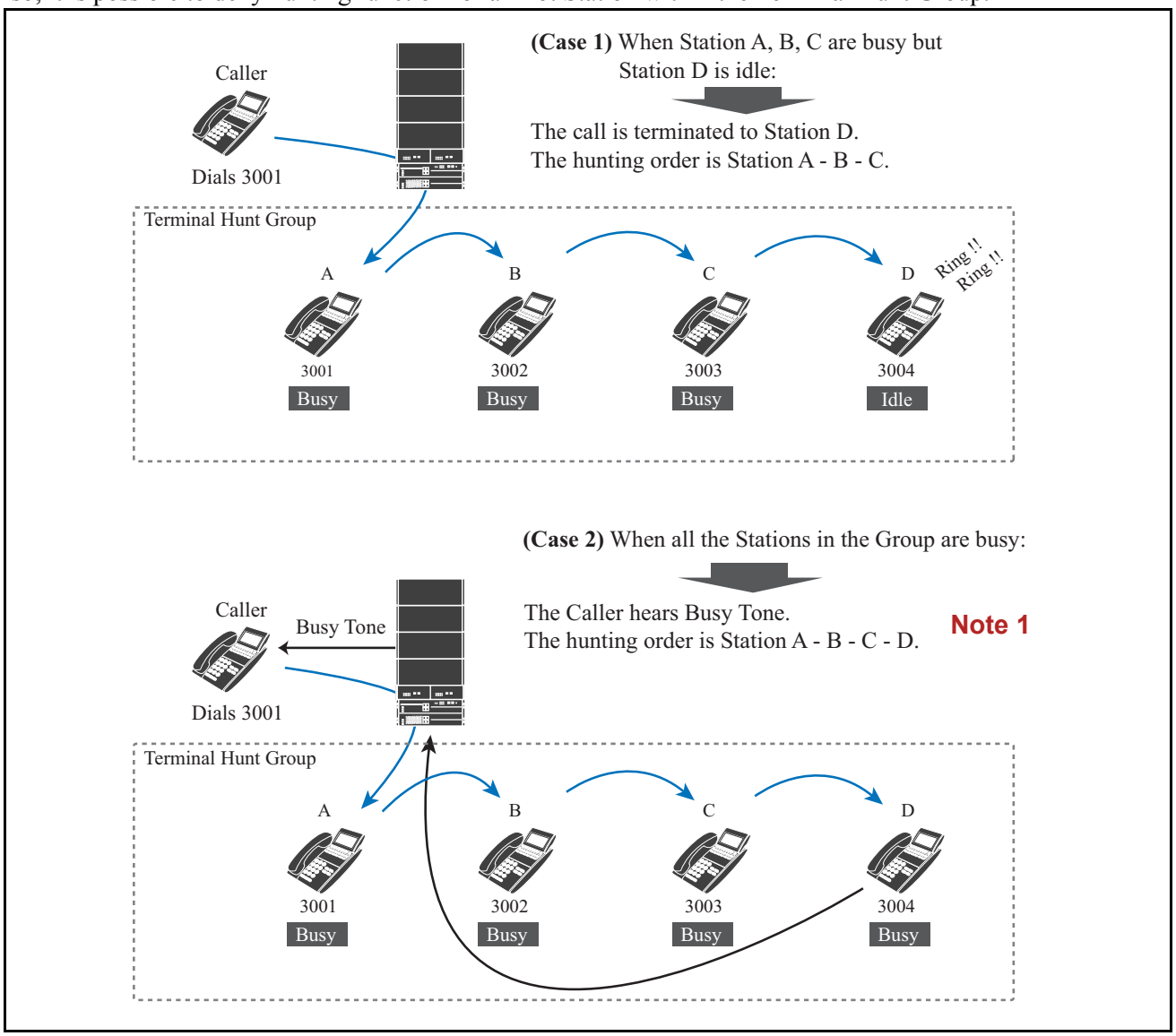
Programming

See STATION HUNTING-CIRCULAR [S-7], STATION HUNTING-TERMINAL [S-9], STATION HUNTING - PRIORITY [S-138], STATION HUNTING - PRIORITY WITH SWITCHBACK [S-139]. These types of Station hunting use secretarial Stations.

S-9 Station Hunting - Terminal

General Description

This feature permits a call to be processed automatically through a Terminal Hunt Group of busy Stations, in a programmed order. The call arrives to a specific Station of the group (the Pilot Station) and, if it is in a busy state, the call is transferred to the next Station in the programmed order until it finds an idle Station that rings. For example, as illustrated in the following figure, if a call is placed to Pilot Station A and Station A is busy, the call is forwarded to Station B and so on. If all Stations are busy, the caller hears Busy Tone.
Also, it is possible to deny hunting function for a Pilot Station within the Terminal Hunt Group.



Note 1: If a secretarial Station is assigned for the Hunt Group, the call is forwarded to the secretarial Station. For the secretarial Station, see STATION HUNTING - SECRETARIAL [S-8].

Note: In this feature, Busy means the Station cannot receive a call (because of reasons such as talking, receiving other call, do not disturb being set).

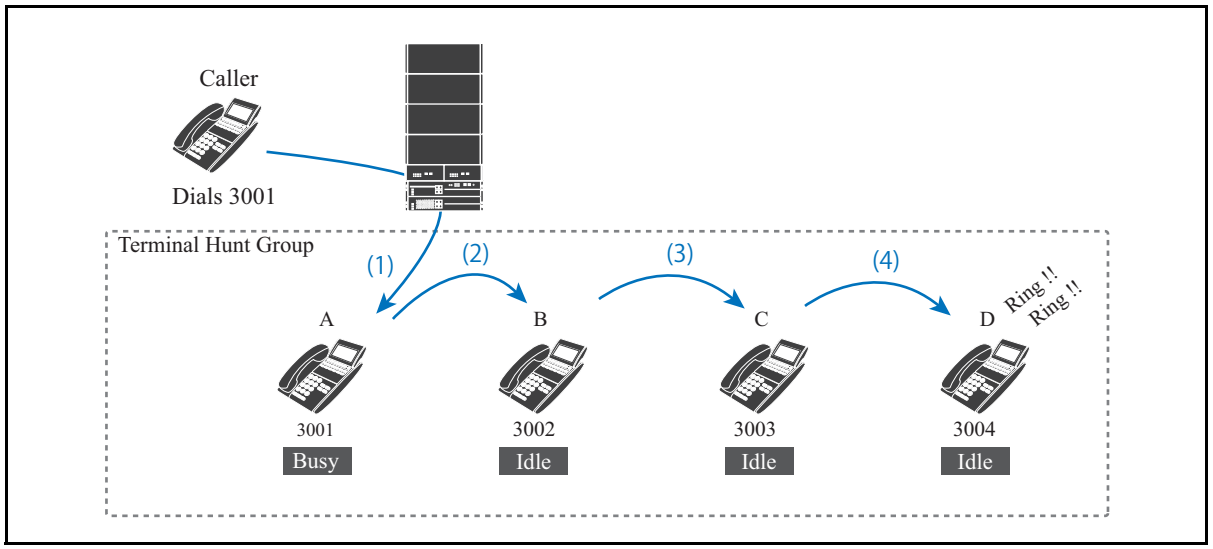
- TRANSFER OF UNANSWERED CALLS IN A HUNT GROUP

When the caller dials the Pilot Station in the Terminal Hunt Group, the call arrives to a Station within the Terminal Hunt Group, and the Station does not reply within a programmed period of time, the call can be transferred again to another idle Station.

Note: Transfer of unanswered calls is available from FP95-114 V4 Issue 4.0.

As for the transfer of an unanswered call in a Terminal Hunt Group, the figure below gives an example for the transfer pattern with the following steps.

1. A call is made to the Pilot Station (Station A).
2. Station A is busy, the call is transferred to the first idle station in the hunting order (Station B).
3. The call remains unanswered within a programmed period of time and is transferred to the next idle Station in the hunting order (Station C).
4. The call remains unanswered within a programmed period of time and is transferred to the next idle Station in the hunting order (Station D).



Note: In the example above, the call may be transferred to a secretarial Station if it has been set. For the secretarial Station, see STATION HUNTING - SECRETARIAL [S-8].

Operating Procedure

No manual operation is required.

Service Conditions

<Important Notices>

1. Station Number can belong to one Hunt Group only, it cannot belong to another Hunt Group including that for STATION HUNTING - CIRCULAR [S-7], STATION HUNTING - TERMINAL [S-9], STATION HUNTING - PRIORITY [S-138], STATION HUNTING - PRIORITY WITH SWITCHBACK [S-139] or UNIFORM CALL DISTRIBUTION (UCD) [U-1].
2. A station can be assigned to a different Hunt Group by the Physical Station Number (DM) and the Telephone Number (LDM or NDM). For that reason, a Station already assigned to a Hunt Group that uses Physical Station Number (DM) can be also assigned to other Hunt Group that uses the Telephone Number (LDM or NDM). Also, when a Physical Station Number (DM) and Telephone Number (LDM or NDM) are identical, priority is attached to the Telephone Number (LDM or NDM). However, it is not possible to assign the same Physical Station Number or the same Telephone Number to more than one Hunt Group.

<Interactions with Other Services>

1. If the Station of a Hunt Group is in the following states or the following features are set in the Station of a Hunt Group, the following occurs.

   - **A**: The feature is activated and the call is forwarded.
   - **S**: The Station is skipped.

Feature/State	Direct Call to the Pilot Station	Call Transfered to the Station in the Hunting
CALL FORWARDING - ALL CALLS [C-5]	**A** Note 2	**S**
DO NOT DISTURB [D-11]	**S**	**S**
BUSY STATION	**S**	**S**
BUSY STATION + CALL FORWARDING - BUSY LINE [C-2]	**A** Note 2, Note 3	**S**
BUSY STATION + CALL FORWARDING - BUSY LINE [C-2] set with the ACFO/ACFOL/ACFON command (on a tenant/route basis)	**S**	**S**
LOGGED OUT STATION	**S**	**S**
LOGGED OUT STATION + LOGGED OUT IP STATION - CALL DESTINATION [L-53]	**A**	**S**
LOGGED OUT STATION + LOGGED OUT IP STATION - CALL DESTINATION [L-53] set with the ACFO/ACFOL/ACFON command (on a tenant/route basis)	**S**	**S**

Note 2: If CALL FORWARDING- OVERRIDE [C-26] is set in the transfer destination of Pilot Station, the call forwarding overriding from the forwarding destination to the Station of the Hunt Group will hunt if the Station of the Hunt Group is busy.

Note 3: When a call to a station in a Hunt Group is forwarded by CALL FORWARDING - BUSY LINE [C-2] and the forwarded party (also a member of another Hunt Group) is also busy, the resultant hunt target can be designated from the forwarding side Hunt Group or the forwarded side Hunt Group. This can be specified with ASYD/ASYDN SYS1 Index 69 Bit 5.

2. A Phantom Number can be assigned to the Pilot Station of a Hunt Group. However, when a caller dials the Phantom Number, hunting does not occur for the Station that receives the call.
3. Recall to a Station in the Hunt Group from the following features will not hunt:
   - CALL BACK [C-1]
   - CALL HOLD [C-6]
   - OUTGOING TRUNK QUEUING [O-2]

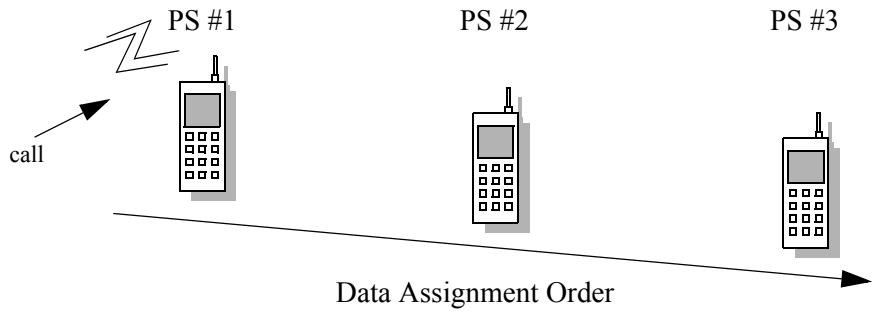
- PAGING TRANSFER [P-10]

4. For STATION MESSAGE DETAIL RECORDING [S-10] the number recorded is the Station Number where the call actually terminates.
5. HOTLINE [H-1] cannot be assigned to the Station of a Hunt Group.

<Hardware/Software>

1. A call from ATTENDANT CONSOLE [A-3] will not hunt.
2. The conditions when using PS are as follows:
   a.) If a call terminates to a PS in the Hunt Group while the PS is Out of Zone or its power is OFF, whether the call is processed as PS Out of Zone or hunting is executed is determined by System Data (ASYDL, SYS 1, Index 688, b0). When using this feature in an FCCS environment, assign the same data value to the system data in each node.
   b.) When hunting is selected in the system data, the caller hears no tone until the hunted PS answers. Total length of no tone is obtained by the following.
      - Time for Detecting Out of Zone (Maximum eight seconds) × Number of Hunting
   c.) When a Desk Console operator dials the number of PS that belongs a Hunt Group, the hunting is not available if the PS is Out of Zone.
   d.) When receiving an analog PSTN incoming call while all PSs in the Hunt Group are Out of Zone, the call processing will be stopped after detecting Out of Zone of the last PS in the Hunt Group. At this time, the calling party will receive RBT when the call is a Direct In Termination call. If Automated Attendant is used, the calling party will receive ROT after No Tone. Note that the call cannot be terminated to the destination even if the PS returns to the zone during RBT.
   e.) When receiving an ISDN incoming call while all PSs in the Hunt Group are Out of Zone, hunting starts after calling up the destination. However, timeout will occur at Central Office side, resulting in ROT connection (hunting is not available).
   f.) If all Stations in the Hunt Group are busy or PS Out of Zone, the call can be forwarded a Station in the group. The station is called secretarial Station. The caller hears Busy Tone if the secretarial Station is also busy.
   g.) A PS can be assigned as a secretarial Station.
   h.) If a PS assigned as a secretarial Station is placed in Out of Zone, the caller hears Busy Tone.
   i.) A PS in the Hunt Group can set PS Out of Zone - Call Forwarding. When the PS has been set as Pilot Station, the incoming call will be forwarded only if the call cannot terminate to any stations in the group. When the PS has not been set as Pilot Station can be forwarded only if the PS is directly called.
   j.) When a PS in the Hunt Group performs roaming, the PS is skipped. However, the PS is a station that receives the call directly, it will be roaming call termination.
   k.) If a PS receiving a call turns to Out of Zone, this may cause a network failure. In this case, the caller hears Busy Tone.
   l.) The following shows call processing when Out of Zone or busy is involved for this feature.

This example assumes: - hunting data has been assigned in the order of PS #1, #2, #3
- PS #1 first received a call.



When Hunting is selected (ASYDL, SYS1, INDEX 688, Bit 0 = 0)

PS #1	PS #2	PS #3	Processing
Out of Zone (Forwarding has been set)	Out of Zone	Out of Zone	After calling up PS #1, PS #2, PS #3, the forwarding destination is called.
Out of Zone (Forwarding has not been set)			After calling up PS #1, PS #2, PS #3, the caller receives ROT or Automatic Announcement Called PS Out of Zone.
Busy	Out of Zone	Out of Zone	After calling up PS #2 and PS #3, the caller receives Busy Tone.
Busy	Busy	Out of Zone	After calling up PS #3, the caller receives Busy Tone.

When Out of Zone Processing is selected (ASYDL, SYS 1, INDEX 688, Bit 0 = 1)

PS #1	PS #2	PS #3	Processing
Out of Zone (Forwarding has been set)	Out of Zone	Out of Zone	After calling up PS #1, the forwarding destination is called.
Out of Zone (Forwarding has not been set)			After calling up PS #1, the caller receives ROT or Automatic Announcement Called PS Out of Zone.
Busy	Out of Zone	Out of Zone	After calling up PS #2, the caller receives Busy Tone.
Busy	Busy	Out of Zone	After calling up PS #3, the caller receives Busy Tone.

<Conditions Specific to This Feature>

1. The Pilot Station is the first Station programmed in the ASHP/ASHPN/ASPHN command.
2. Only calls to the Pilot Station will cause the hunting sequence to operate. Calls to busy Stations, other than the Pilot Station, will receive a Busy Tone.

3. The maximum number of Stations per group is 100. The feature is usable with all Stations and there is no limit on the allowable number of Hunt Groups within a system.
4. This feature enables a Pilot Station within Hunt Group to deny hunting function for an Incoming Call originated from Station (including PS) (ASFC/ASFCN SFI 181 =1). The service conditions are as follows:
   a.) When CALL FORWARDING-DON'T ANSWER [C-5] is set, a call to the Pilot Station is forwarded according to their service condition.
   b.) When CALL FORWARDING-BUSY LINE [C-2] is set, a call to the Pilot Station is forwarded according to their service condition.
   c.) A call to the Phantom Number assigned to the Station will not hunt.
   d.) External Trunk (such as DDD (TCL = 1)/FX (TCL = 2)/WATS (TCL = 3), etc) is not available.

5. The conditions for the transfer of unanswered calls in a Hunt Group feature are as follows: **Note 4**

Note 4: Available from FP95-114 V4 Issue 4.0.

   a.) The feature is only possible for Hunt Groups with data assigned via Telephone Numbers (NDM or NDM).
   b.) The feature will be activated only for calls to the Pilot Station. Calls to other Stations of the Hunt Group will not be transferred.
   c.) Be aware that if there is a value set in Timer Value of No Answer Timer set in the ADAI command for a Station in the group it will have priority over the value set with ASYDL/ASYDN, SYS1 Index 1396. If no value is set for ADAI command or ASYDL/ASYDN, SYS1 Index 1396, other Timer Value of No Answer Timer (ASYD SYS1, Indexes 139, 140, 141, 247) will be used.
   d.) The display of a Station receiving an incoming call transfer based on this feature will be the same as in the case of CALL FORWARDING - DON'T ANSWER [C-3]. Also, in the case of other external devices, it will be treated in the same way.

e.) If other unanswered calls transfer features are set in a Station of this feature group, those other unanswered calls transfer features have priority over the transfer of unanswered calls in a Hunt Group feature.

Priority Order	Transfer of Unanswered Calls in a Hunt Group Feature	
	Out of Service	In Service
1	CALL FORWARDING - DON'T ANSWER [C-3]	CALL FORWARDING - DON'T ANSWER [C-3]
2	-	CALL FORWARDING - BUSY LINE [C-2] **Note 5**
3	-	TRANSFER OF UNANSWERED CALLS IN A HUNT GROUP
4	CALL FORWARD BY SERVICE FEATURE CLASS [C-172]	CALL FORWARD BY SERVICE FEATURE CLASS [C-172]
5	CALL FORWARDING - DON'T ANSWER [C-3] set with the ACFO/ACFOL/ACFON command (on a tenant/route basis)	CALL FORWARDING - DON'T ANSWER [C-3] set with the ACFO/ACFOL/ACFON command (on a tenant/route basis)

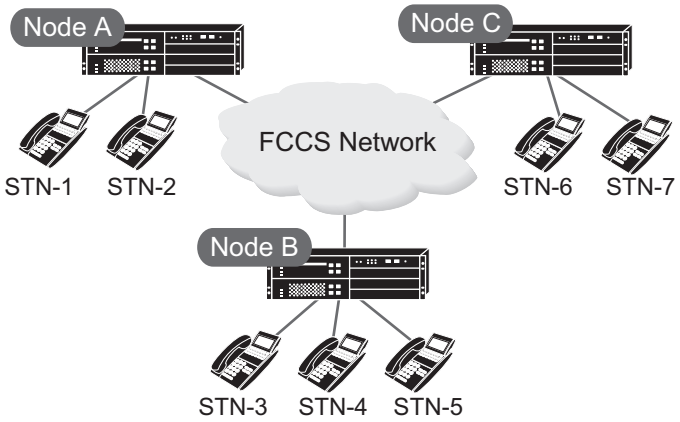
Note 5: The Station with the incoming call is treated as Busy Station and the call is transfered.

f.) When using this feature, do not include in the Hunt Group the following type of terminals except as the Pilot Station:

- PS
- SP Controlled SIP terminal

FCCS Conditions

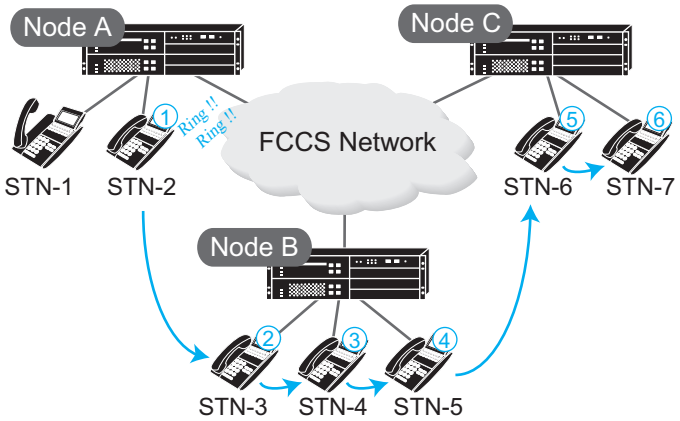
1. When activating this feature between nodes via FCCS, use only Hunt Groups assigned with Telephone Number (NDM).
2. Stations in up to four different nodes can be assigned to a Hunt Group.
   When the Hunt Group includes more than one node, the [Group by FPC] button of ASHPN must be used to set the order of the nodes in the Hunt Group. Without this node order grouping assignment, data cannot be assigned.
3. The Hunting order when the Hunt Group is created across more than one Node is as follows.



	Node A		Node B			Node C	
	STN-1 Note 6	STN-2	STN-3	STN-4	STN-5	STN-6	STN-7
Assigned Hunting order by the office data	1	2	3	4	5	6	7
Actual Hunting order	(Busy)	1	2	3	4	5	6

Note 6: This is the Pilot Station.

Note: The following is the hunting order in the case of the transfer of unanswered calls in a Hunt Group.



	Node A		Node B			Node C	
	STN-1 Note 7	STN-2	STN-3	STN-4	STN-5	STN-6	STN-7
Assigned Hunting order by the office data	1	2	3	4	5	6	7
Actual Hunting order	(Busy)	1 (Ringing)	2	3	4	5	6

Note 7: This is the Pilot Station.

4. The Incoming Call terminated between nodes via FCCS to a Station in the Hunt Group programmed by the Physical Station Number, is not hunted.

Programming

<When using Physical Station Numbers for creating the Hunt Group>

STEP 1: **ASYD** - Assign the following system data.
System Data 1, Index69, Bit 5: Hunt Group when transferred party is busy (Station Hunting after C.F.-Busy Line):
0 = Hunt in Transferring Party's Hunt Group
1 = Hunt in Transferred Party's Hunt Group

STEP 2: **ASHP** - Assign the required data for the Group.
TN: Tenant Number
STN: Physical Station Number **Note 9**
SECRETARY STN:Physical Station Number of the Secretarial Station **Note 8**
[Station List in Group]
STN: Physical Station Numbers of the Stations included in the Hunt Group. **Note 9**

Note: Assignment of Numbers to a Hunt Group may be in any order.

Note 8: If secretarial Station is used, assign the Physical Station Number of the Station used as such secretarial Station.

Note 9: Do not enter Phantom Number.

STEP 3: **ASFC** - (Optional) Assign the following Service Feature Restriction when hunting must be denied.
SFI 181 =1 (No Hunting to next station in group)

<When using Telephone Numbers for creating the Hunt Group>

Note: The following programming supposes that Telephone Numbers have been already assigned and does not include such data settings.

STEP 1: **ASYD/ASYDN** - Assign the following system data.
System Data 1, Index69, Bit 5: Hunt Group when transferred party is busy (Station Hunting after Call Forward - Busy Line):
0 = Hunt in Transferring Party's Group
1 = Hunt in Transferred Party's Group

STEP 2: **ASYDL** - (Optional) Assign the following System Data.
System Data 1, Index 688, Bit 0. Specify whether hunting is activated or not if the called PS is Out of Zone.
0 = Hunting
1 = PS Out of Zone processing

Note: Assign this data if PS is used in the Hunt Group.

STEP 3: **ASHPL/ASHPN** - Assign the required data for the Hunt Group.
Assign the Hunt Group using Telephone Number.
UGN: User Group Number
TELN: Telephone Number in the Hunt Group **Note 10**
SECRETARY TELN:Secretary Telephone Number **Note 11**
[Station List in Group]
TELN: Telephone Numbers of the Stations included in the group. **Note 10**

Note: Assignment of Numbers to a Hunt Group may be in any order.

Note 10: Do not enter Phantom Number.

Note 11: If secretarial Station is used, assign the Telephone Number of the Station used as such secretarial Station.

STEP 4: **ASFCN** - (Optional) Assign the following Service Feature Restriction when hunting must be denied.
SFI 181 =1 (No Hunting to next station in group)

<When activating the transfer of unanswered calls in a Terminal Hunt Group> Note 12

STEP 5: **ASYD** - Assign the following System Data.
System Data 1, Index 69, Bit 0= 1 (No Recall, or execution of Call Forwarding –Don't Answer (when CF – Don't Answer is set at the transfer destination station)).
System Data 1, Index 69, Bit 1 (Send short tone when a call forwarded via CF – All Calls/Logout service is answered.)
0 = Not Required
1 = Required
System Data 2, Index 6, Bit 7 (CF-Don't Answer service when Attendant Console transfers call before

called station answers)
0 = Out of service (Recall to Attendant Console)
1 = In service

STEP 6: **ASYDL/ASYDN** - Assign System Data 1, Index 1395, Bit 0=1 (transfer of unanswered calls in a Terminal Hunt Group in service). **Note 12**
System Data 1, Index 1396= set the no answer timer for the transfer of unanswered calls in a Hunt Group. **Note 12**
Bit 0 - 3: 1 - F= MTC
Bit 4 - 6: 0 - 7= TC
TC: Timer Class, MTC: Miscellaneous Timer Counter
TC = 0: -
1: -
2: -
3: 2 seconds
4: 30 seconds
5: -
6: -
7: -

Note: Default value for this data is HEX 00 (data is not set). If data is not set, other Timer Value of No Answer Timer (ASYD SYS1, Indexes 139, 140, 141, 247) will be used.

Note: Do not assign "0" for MTC value when TC = 3 or 4.

Note: Assigned time will be used for all the Hunt Groups to which is available (Circular, Terminal, Priority), it cannot be set separately according to the type of Hunt Group.

Note: The time will be used for the transfer of unanswered calls regardless of whether transfer of unanswered calls in a Terminal Hunt Group is in service or not.

Note 12: Available from FP95-114 V4 Issue 4.0.

STEP 7: **ASFC/ASFCN** - Assign the following Service Feature Restrictions:
SFI 8=1 (Call Forwarding-Don't Answer allowed).
SFI 103 (Assignment of No Answer Timer for Blind Transfer to Station/Blind Transfer to Attendant) **Note 13**
0 = Restricted (settings in ASYD, System Data 1, Index 140)
1 = Allowed (settings in ASYD, System Data 1, Index 247)
SFI 104 (Processing when the transferred station does not answer for a predetermined period of time in Blind Transfer) **Note 14**
0 = Transferred station continuously rings
1 = The transferred call is forwarded when Call Forwarding-Don't Answer is assigned at the transferred station and no-answer timer expires

Note 13: The time set for a Station with ASYDL/ASYDN System Data 1, Index 1396 (no answer timer for the transfer of unanswered calls in a Hunt Group) will be given priority.

Note 14: Set to 1 if the terminal in the Hunt Group is the destination of BLIND TRANSFER TO STATION [B-18].

STEP 8: **ATNR** - Assign the following Tenant Restriction Class.
TRI 0=1 (Station within OGTN calls to a station within TMTN allowed).
TRI 1=1 (Station within OGTN sets Call Forwarding-All Calls / Busy Line / Don't Answer/Logout to a station within TMTN allowed).

STEP 9: **ACFR/ACFRN** - Assign the following Call Forwarding Restriction.
CCI 8=1 (Call Forwarding-Don't Answer allowed).

S-10 Station Message Detail Recording (SMDR)

General Description

This feature provides a call record for all outgoing Station to Trunk Calls and Incoming Trunk to Station Calls. When the system is equipped with this feature, an RS232C output port or a SOCKET interface (LAN) is provided, permitting interface with a customer-owned computer system. All output is in the ASCII format, and includes the following:

- Calling Telephone Number
- Dialed Number (maximum of 24 digits)
- Route Number
- Start of call time
- Disconnect time
- Year, month and date
- Condition (Attendant handled, transfer, etc.)
- ACCOUNT CODE [A-18] (10 digits maximum)
- AUTHORIZATION CODE [A-28] (10 digits maximum)
- FORCED ACCOUNT CODE [F-7]

[Additional FCCS Format]

- Called Party Type (Attendant Console/Station)
- Call Start/Call end time (millisecond)
- Call metering
- FPC/User Group/Telephone Number (Calling Party)
- FPC/User Group/Telephone Number (Called Party)
- FPC of the node providing the route for the call
- Logical Route Number
- FPC of the node providing the first-choice route
- First-choice Logical Route Number
- Trunk arrival/sending time and Abandoned Call Information (for Incoming /Outgoing Call)

[Station-to-Station Basis SMDR]

SMDR can be output for Station-to-Station Calls by assigning data.

[SMDR by MA-ID via IP Network]

Multi Area ID (MA-ID) can be assigned on a network basis for identifying the location. Then, SMDR can be output for calls over IP network when MA-ID of both parties are not identical. For more information, see SMDR MULTI AREA ID VIA IP [S-144].

[SMDR - Multiple Time Zone]

SMDR output that takes into account time difference when a call is made over different time zones is available. For more information, see [M-114] MULTIPLE TIME ZONE.

Note: SMDR (Station Message Detail Record) collectively means billing information and service. When connecting to SMDR system, simultaneous use of LAN and RS-232C connection is not available in a single node.

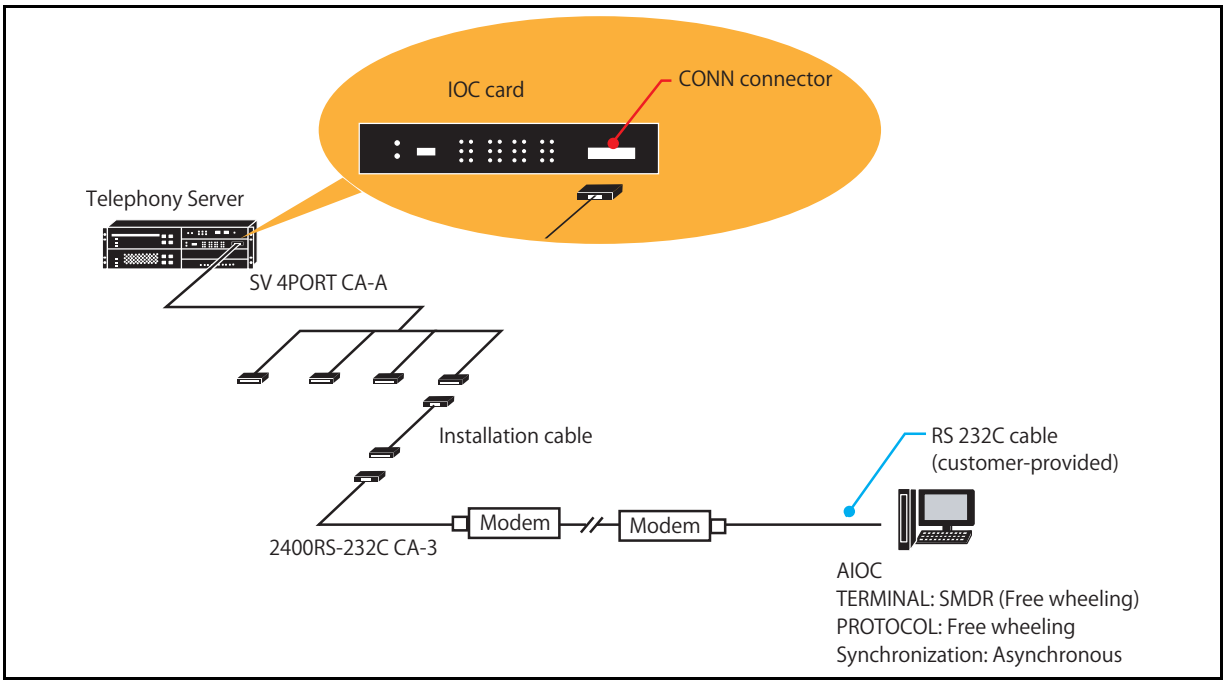
Operating Procedure

No manual operation is required. Call record outputs are produced automatically.

Service Conditions

<RS232C Interface Conditions>

1. When you use the SMDR system with a RS-232C interface, be aware of the following:
   - When connecting to the SMDR system, simultaneous use of LAN and RS-232C connection is not available in a single node.
   - When you change the interface for the SMDR system (from RS-232C to LAN and vice versa), all the data collected in the Telephony Server is erased. Therefore, before performing such operation, output all the collected data to the SMDR system.
2. SMDR information facilitates analysis of Trunk traffic and can be readily applied for cost allocation or billing purposes.
3. SMDR may be programmed to record either all Outgoing Calls or only Toll Calls, depending upon user requirements.
4. Both Incomplete and Abandoned Calls will be discarded by the SMDR.
5. When ROUTE ADVANCE [R-6] or LEAST COST ROUTING (LCR) - 3/6-DIGIT [L-5] is provided with the system, SMDR will record the Route Number selected.
6. The maximum number of digits dialed cannot exceed 24.
7. When customer-owned computer equipment is connected via RS232C interface, SMDR information will be transmitted directly to the computer as each call record is completed.
8. If the Outgoing Call is directed to a Trunk that cannot give Answer Supervision from the Central Office, the start of call time is 18 seconds after the last digit is dialed. This timer value is set on the System Data.
9. If the distance between the system and the processing computer exceeds 50 feet (15 meters), an asynchronous modem should be used.

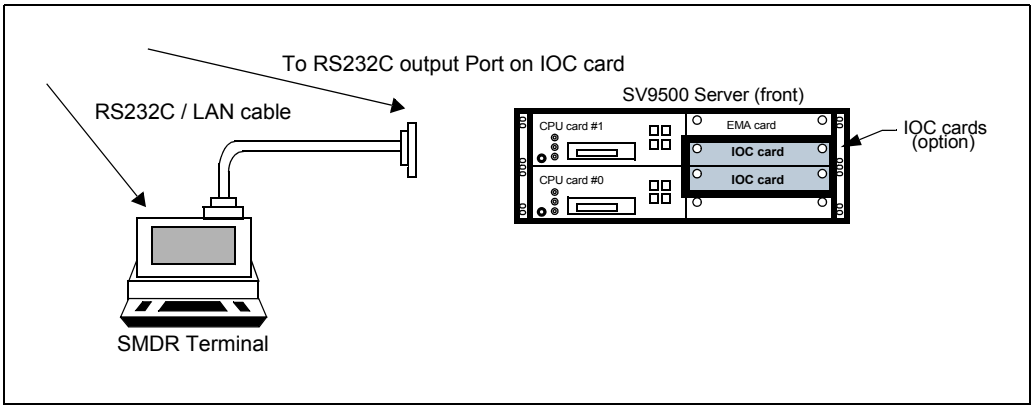


10. RS232C Interface specifications:

Synchronization	Asynchronous
Data Speed	300, 600, 1200, 2400, 4800, 9600bps
Code	ASCII 7-Bit + parity Bit
Maximum Distance	50 ft. (15m) without modem

11. Supervision on the status of the external RS232C terminal is not performed.
12. FORCED ACCOUNT CODE [F-7] can be recorded on SMDR, for INWATS and Central Office Incoming Calls, when used in conjunction with the REMOTE ACCESS TO SYSTEM [R-2] feature.
13. FORCED ACCOUNT CODE [F-7] cannot be registered on SMDR if the Incoming REMOTE ACCESS TO SYSTEM [R-2] Calls are terminated to Stations via night service or DID [D-8].
14. For REMOTE ACCESS TO SYSTEM [R-2] Tandem Connections, the FORCED ACCOUNT CODE [F-7] is registered for Outgoing Trunks only.
15. The PBX can accommodate up to 7 active ports or 4 redundant ports. Additional hardware is required.
16. AUTHORIZATION CODE [A-20] can be recorded on SMDR.
17. The maximum number of simultaneous calls can be stored in the system memory (active calls) is 47,000. When a call is completed, the record is sent to the output device and is removed from memory.
18. If an overflow occurs, calls are allowed to complete and are not recorded.
19. ATTENDANT-CONTROLLED CONFERENCE [A-2]: If the Attendant adds a Trunk to the Conference, the Attendant Conference line terminal is recorded.
20. LEAST COST ROUTING - 3/6-DIGIT [L-5]: The Calling Party can dial an ACCOUNT CODE [A-18] before the LCR Access Code. The Called Number is recorded (as modified for sending).
21. If an ACCOUNT CODE [A-18] is dialed, the maximum digits of the Called Number is 22.
22. CENTRALIZED ATTENDANT SERVICE (CAS) [C-20]: The SMDR record is provided for calls extended outward by a CAS Attendant.
23. DIRECT INWARD DIALING (DID) Trunk Calls [D-8]: If the call is to the Attendant, the Attendant can input an ACCOUNT CODE [A-18] before extending the call.
24. INDIVIDUAL Trunk Access [I-4]: The Attendant cannot dial in an SMDR account Code before using direct Trunk selection.
25. Incoming TIE Trunk Calls: If an Incoming Call to the Attendant is extended, the Attendant can input an ACCOUNT CODE [A-18].
26. SERIAL CALL [S-15]: The duration of the total call is recorded on the last Station Call.
27. STATION HUNTING [S-7, 8, 9]: The hunted-to Station is recorded in the Called Number field.
28. TANDEM TIE TRUNK [T-1] Calls: The Calling Number is the Incoming Trunk Route plus Trunk Number; the Called Number is the number dialed.
29. UNIFORM CALL DISTRIBUTION (UCD) [U-1]: The Calling Number is the Incoming Trunk Route plus Trunk Number. If the call is to a Station, the Station answering the number is recorded as the Dialed Number.
30. On an Incoming Trunk Call to the Attendant, the Attendant Call record will start upon answering and complete when the Attendant releases. A Station Call record will start when the Attendant releases and complete when the Station goes on-hook. If required, a Station record will not be generated if the Station does not answer.
31. A system option is available to allow or restrict call completion in the case where a failure occurs in transmitting SMDR data from the processor to the I/O port.

32. Additional hardware is required.



33. When either of the 2 parties goes On-Hook (or presses Transfer Key, Hold Key, Answer Key, Over Key) during Station-to-Station Calling [S-11], output of the billing information will be completed.
34. In a CCIS environment if a node has a PHC and IPTRK coexisting that node cannot be the centralize point for call accounting.
35. SMDR data can be output via LAN cable.
36. A maximum of four SMDR apparatuses per node are available.
37. It is normal operation that ACC and ANI data is sent to SMDR if ACNP is assigned.
38. The LAN and RS232C interfaces cannot be used simultaneously for connecting SMDR equipment in the same node.

<FCCS Conditions>

1. Call Time specification:
    - The base for Call Start Time and Call End Time is the time in the Calling Party-side node.
    - In case there is time difference between each node, related time difference data, based on the UCT (Universal Coordinated Time) standard, should be written in Network Data Memory (NDM) by using the ATDF command.
    - Because billing information gathering is performed in the Called Party-side node, the Call Start Time in the originating node is determined in a way the deduction of time between Calling and Called Party nodes is added or subtracted to/from the Called Party-side time. (If there is no time difference data in any [both] of the nodes, the Call Start Time is specified automatically by that in the Called Party-side node.)
    - A counter, not the clock providing the current time, is used for Call End Time calculation. The calculation method is as follows:

    a.) Call Start Time (by using the clock) and the current counter value are registered to the Call Base Table, soon after a Line Connection is established.
    b.) Upon the call completion, the deduction between the counter value shown in a.) and that renewed during the call exchange is calculated.
    c.) The deduction is transformed to a time (hour, minute, second, millisecond) format.
    d.) The transformed time is added to the Call Start Time registered in a.)-End.
        - The counter-based time is renewed in every 16.384 milliseconds.

2. The maximum number of calls available for concurrent storage in the system memory (active calls) is:

System Type	Maximum Buffer Capacity for Storing SMDR
IMG/MMG system	47,000 calls / System
UMGi/RNIP system	47,000 calls / 1 Node
UMG system	47,000 calls / 1 LMG

Note: The same buffer memory is used for storing Centralized Billing data and Billing of calls that are made in the center node.

Note: The upper limit of polling buffer rate can be specified by ASYDL, SYS 1, Index 586. When the value has not been set, default 00 (Hex) = 50% will be applied.

3. When the center node stores the billing information of calls that are made in itself, the center node checks the number of completed calls in all nodes on a CCIS/FCCS network including itself. The system message "6-O (SMDR Output Buffer Overflow Alarm message)" and "6-P (SMDR Output Buffer Overflow Release)" are issued if SMDR output buffer in the center node is in the following conditions:

   "6-O" is issued when:

   - SMDR output buffer is not overflowing, and
   - SMDR output buffer usage rate exceeds the value assigned to ASYD, SYS1, Index 249.

   "6-P" is issued when:

   - SMDR output buffer is overflowing, and
   - SMDR output buffer usage rate is less than the value assigned to ASYD, SYS1, Index 250.

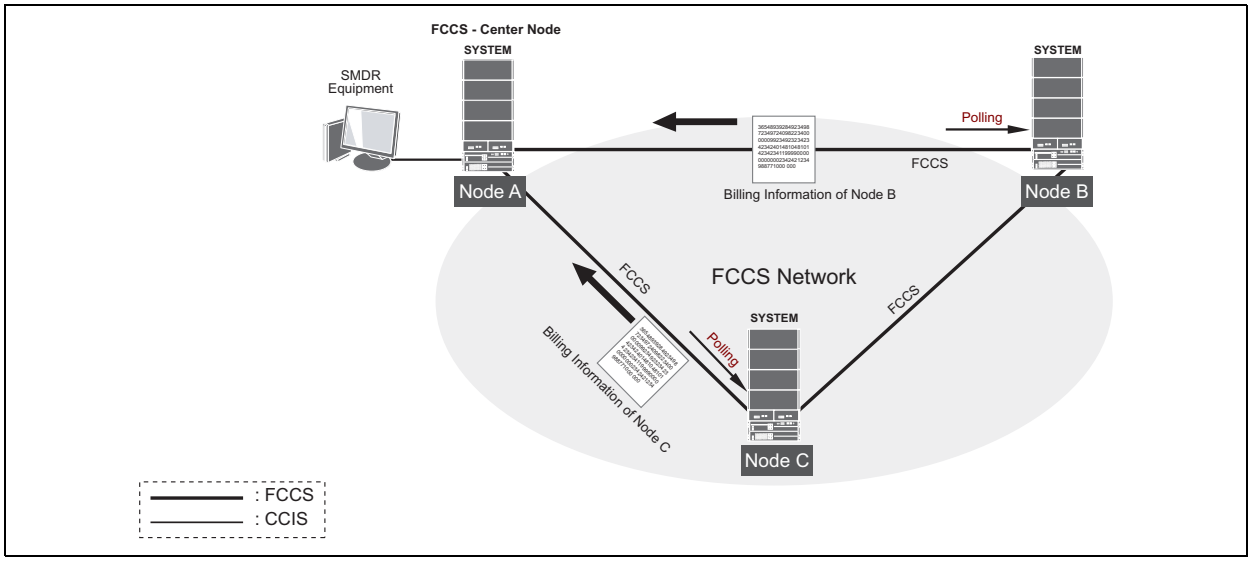
   Even if the SMDR output buffer usage rate in the center node exceeds the value assigned to ASYD, SYS1, Index 249 due to polling Local Nodes, the system message "6-O" is not issued if there are no billing information of calls made in the center node afterwards.

4. If an overflow occurs, calls are allowed to complete but are not recorded.
5. To output the FCCS-related billing data, the external SMDR terminal should be compatible also with the IMX text format patterns.
6. When a Station originates an outgoing Trunk Call using a Trunk in the Remote Node (the Station and the Trunk belong to an FCCS Network), the billing information for the Station to Trunk Connection is not recorded.
7. Because the FCCS Service achieves a completely fused link between multiple distant nodes, Intra-Station Connections and Node-to-Node Connections are theoretically considered identical.
8. When a Line Connection is established on the FCCS, the billing data concerned is collected fully in the Called Station-side node, not in the Calling Station side. The collected data is then transmitted to the SMDR terminal as the call finishes.
9. Multiple Center Nodes can be assigned on a FCCS Network for centralized billing purposes.
10. Transmission of call base table with 264 bytes.
11. When the user performs system changeover (using MBR Key, CMOD command or etc.), system massages (26-V, 26-W) are displayed. However, there are no defective conditions.

12. The figures on the following pages show the Service Conditions of Polling Method and the sample data assignment, focusing on when CCIS links are involved.
13. SMDR data assignment is required for all nodes in the FCCS network when using Centralized Billing - FCCS feature. (See Example 9 for the sample data assignment.)
14. SMDR data will be stored in a node whose incoming trunks received the data.
15. When SMDR equipment is not connected to the Center Node, polling is failed from a Center Node to a Local Node. If the Center Node is a Local Node used by Centralized Billing - CCIS [C-55], polling is available.
16. After the Center Node receives billing information from a Local Node, it returns ACK to the Local Node. If the Local Node fails to receive ACK due to a network failure, the Local Node sends billing information again. Therefore the billing information is output redundantly.

Example 1

- Node A (FCCS - Center Node) collects the billing information of Nodes B and C via FCCS using the polling method.



[Common Data]

ASYD	System Data 1	Indexes 32, 34	
	System Data 2	Index 3	
ASYDL	System Data 1	Index 576 b0 = 1	

[Node A]

ASYD	System Data 1	Index 288	Assign when RS-232C output is used.
ASYDL	System Data 1	Index 576	Assign Bit 2 = 1 (when using LAN interface for SMDR)
		Index 577	Assign 00H.
		Index 578	Assign when LAN output is used.(Bit 4 **Note 1** of each Index)
		Index 608	Assign Node B and Node C.

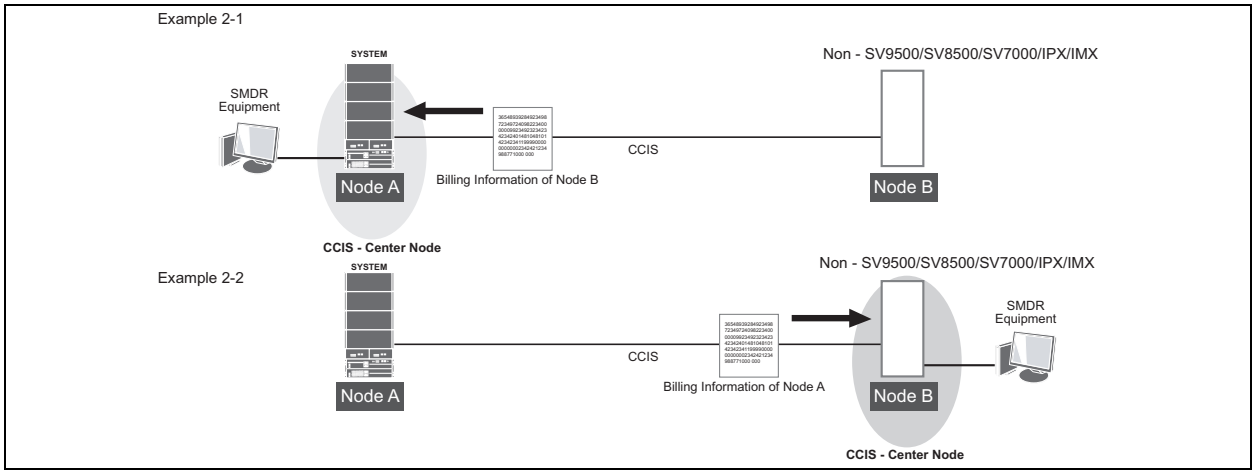
Note 1: When using LAN Interface for SMDR, a TCP/IP LAN Address, Subnet Mask Address, and Gateway Address must be assigned to the Active CPU. Use either ASYDL, Sys-1, Indexes 515 through 526 to assign these addresses or the ADTM command.

[Node B, Node C]

ASYD	System Data 1	Indexes 288 - 295	Assign all 00H.
ASYDL	System Data 1	Index 577	Assign the FPC of Node A.
		Index 578 - 581	Assign all 00H.

Example 2

- The following shows an example of Centralized Billing - CCIS when non-SV9500/SV8500/SV7000/IPX/IMX is used.



Example 2-1

[Common Data]

ASYD	System Data 1	Indexes 32, 34	
	System Data 2	Index 3	

[Node A]

ASYD	System Data 1	Index 180	Assign the PC of Node A.
		Index 182	Assign 00H.
		Index 288	Assign when RS-232C output is used.
ASYDL	System Data 1	Index 576	Assign Bit 2 = 1 (when using LAN interface for SMDR)
		Index 578	Assign when LAN output is used. (Bit 4 **Note 2** of each Index)

Note 2: When using LAN Interface for SMDR, a TCP/IP LAN Address, Subnet Mask Address, and Gateway Address must be assigned to the Active CPU. Use either ASYDL, Sys-1, Indexes 515 through 526 to assign these addresses or the ADTM command.

[Node B]

ASYD	System Data 1	Index 180	Assign the PC of Node B.
		Index 182	Assign the PC of Node A.
		Indexes 288 - 295	Assign all 00H.

Example 2-2

[Common Data]

ASYD	System Data 1	Indexes 32, 34	
	System Data 2	Index 3	

[Node A]

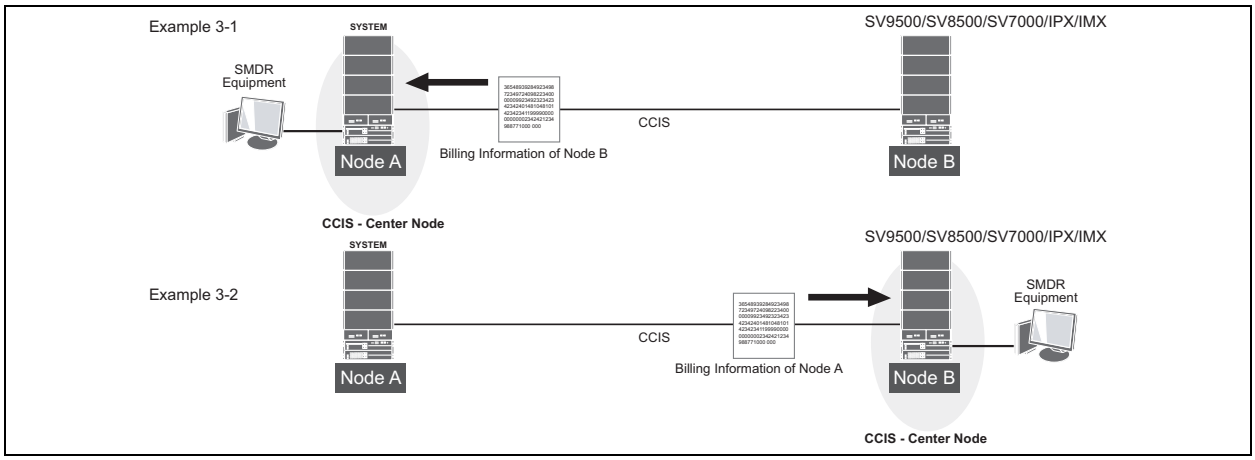
ASYD	System Data 1	Index 180	Assign the PC of Node A.
		Index 182	Assign the PC of Node B.
		Indexes 288 - 295	Assign all 00H.
ASYDL	System Data 1	Index 578	Assign all 00H.

[Node B]

ASYD	System Data 1	Index 180	Assign the PC of Node B.
		Indexes 288	Assign when RS-232C output is used.

Example 3

- The following shows an example of Centralized Billing - CCIS when SV9500/SV8500/SV7000/IPX/IMX is not used.



Example 3-1

[Common Data]

ASYD	System Data 1	Indexes 32, 34	
	System Data 2	Index 3	
ASYDL	System Data 1	Index 576	Assign Bit 2 = 1 (when using LAN interface for SMDR)

[Node A]

ASYD	System Data 1	Index 180	Assign the PC of Node A.
		Index 182	Assign 00H.
		Index 288	Assign when RS-232C output is used.
ASYDL	System Data 1	Index 578	Assign when LAN output is used. (Bit 4 **Note 3** of each Index)

Note 3: When using LAN Interface for SMDR, a TCP/IP LAN Address, Subnet Mask Address, and Gateway Address must be assigned to the Active CPU. Use either ASYDL, SYS1, Indexes 515 through 526 to assign these addresses or the ADTM command.

[Node B]

ASYD	System Data 1	Index 180	Assign the PC of Node B.
		Index 182	Assign the PC of Node A.
		Indexes 288 - 295	Assign all 00H.
ASYDL	System Data 1	Index 578	Assign all 00H.

Example 3-2

[Common Data]

ASYD	System Data 1	Indexes 32, 34	
	System Data 2	Index 3	
ASYDL	System Data 1	Index 576	Assign Bit 2 = 1 (when using LAN interface for SMDR)

[Node A]

ASYD	System Data 1	Index 180	Assign the PC of Node A.
		Index 182	Assign the PC of Node B.
		Indexes 288 - 295	Assign all 00H.
ASYDL	System Data 1	Index 578	Assign all 00H.

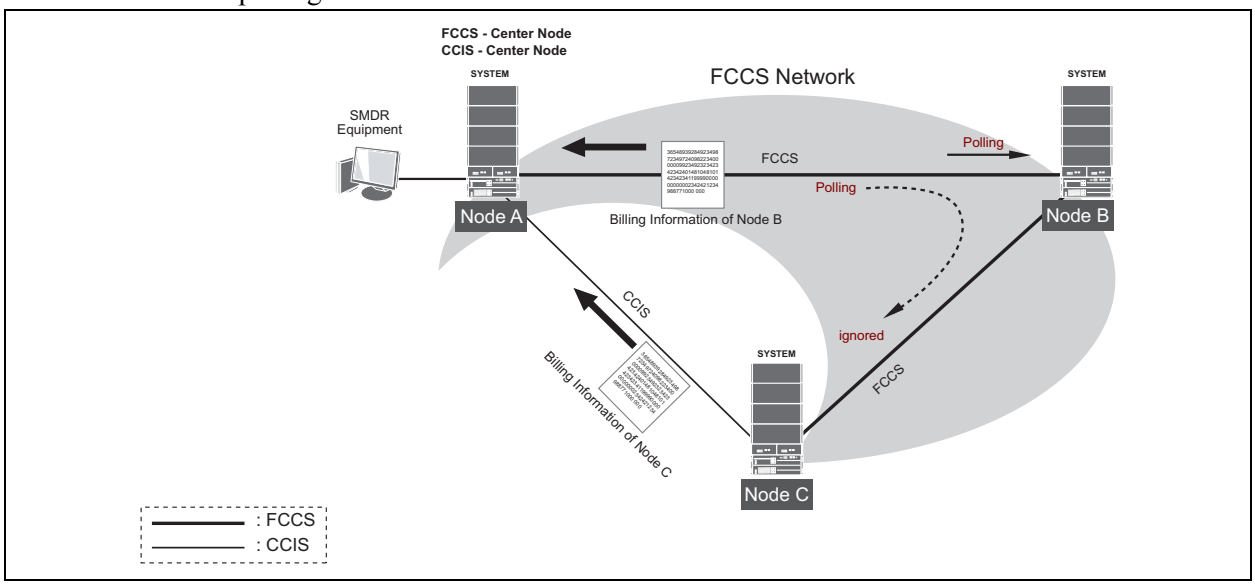
[Node B]

ASYD	System Data 1	Index 180	Assign the PC of Node B.
		Index 182	Assign 00H.
		Index 288	Assign when RS-232C output is used.
ASYDL	System Data 1	Index 578	Assign when LAN output is used. (Bit 4 **Note 4** of each Index)

Note 4: When using LAN Interface for SMDR, a TCP/IP LAN Address, Subnet Mask Address, and Gateway Address must be assigned to the Active CPU. Use either ASYDL, SYS1, Indexes 515 through 526 to assign these addresses or the ADTM command.

Example 4

- Node A tries to collect the billing information of Node B and Node C via FCCS, using the polling method (Node A cannot collect the billing information of Node C via FCCS).
- Node C sends the billing information via CCIS, ignoring polling from Node A.
- Node C deletes the Point Code (CCIS) of Node A for centralized billing data when required to send billing information for polling from Node A.



[Common Data]

ASYD	System Data 1	Indexes 32, 34
	System Data 2	Index 3

[Node A]

ASYD	System Data 1	Index 180	Assign the PC of Node A.
		Index 288	Assign when RS-232C output is used.
ASYDL	System Data 1	Index 576	Assign Bit 0 for Centralized FCCS SMDR In Service.

Index 576	Assign Bit 2 = 1 (when using LAN interface for SMDR)
Index 577	Assign 00H.
Index 578	Assign when LAN output is used. (Bit 4 **Note 5** of each Index)
Index 608	Assign Node B.

Note 5: When using LAN Interface for SMDR, a TCP/IP LAN Address, Subnet Mask Address, and Gateway Address must be assigned to the Active CPU. Use either ASYDL, SYS1, Indexes 515 through 526 to assign these addresses or the ADTM command.

[Node B]

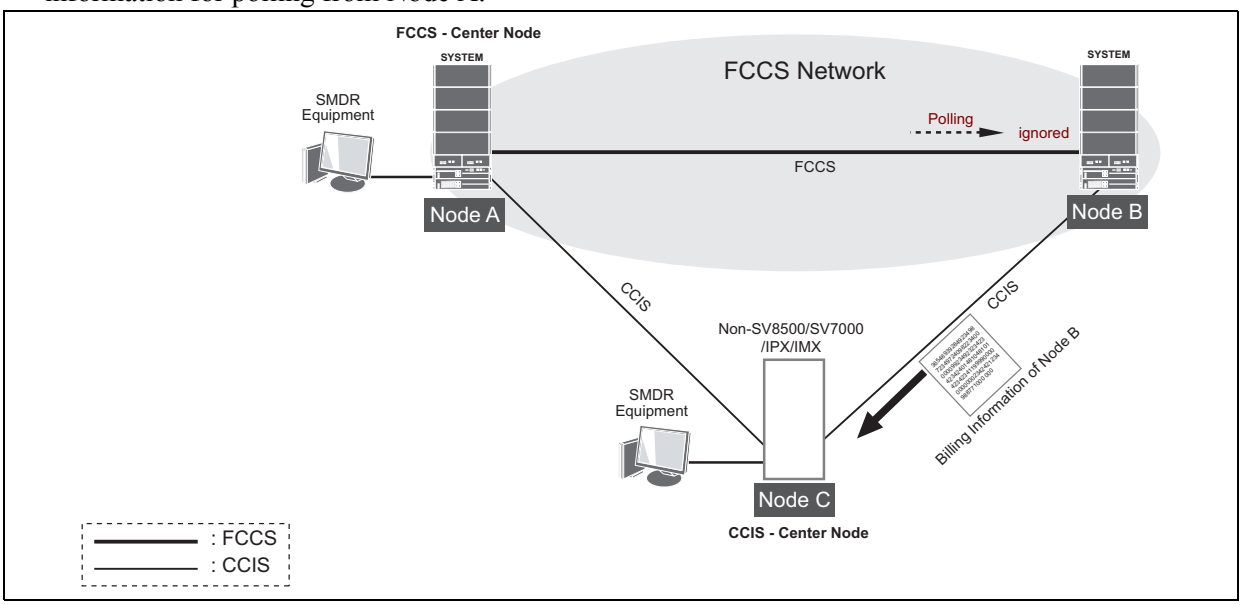
ASYD	System Data 1	Indexes 288 - 295	Assign all 00H.
ASYDL	System Data 1	Index 576	Assign 01H.
		Index 577	Assign the FPC of Node A.
		Indexes 578 - 581	Assign all 00H.

[Node C]

ASYD	System Data 1	Index 180	Assign the PC of Node C.
		Index 182	Assign the PC of Node A.
		Indexes 288 - 295	Assign all 00H.
ASYDL	System Data 1	Index 576	Assign Bit 2 = 1 (when using LAN interface for SMDR)
		Index 577	Assign 00H.
		Indexes 578 - 581	Assign all 00H.

Example 5

- Node A tries to collect the billing information of Node B via FCCS, using the polling method (Node A cannot collect the billing information of Node B via FCCS).
- Node B sends the billing information via CCIS, ignoring polling from Node A.
- Node B deletes the Point Code (CCIS) of Node C for centralized billing data when required to send billing information for polling from Node A.



[Common Data]

ASYD	System Data 1	Indexes 32, 34
	System Data 2	Index 3

ASYDL	System Data 1	Index 576	Assign Bit 2 = 1, when using LAN interface for SMDR.

[Node A]

ASYD	System Data 1	Index 180	Assign the PC of Node A.
		Index 288	Assign when RS-232C output is used.
ASYDL	System Data 1	Index 577	Assign 00H.
		Index 578	Assign when LAN output is used. (Bit 4 **Note 6** of each Index)

Note 6: When using LAN Interface for SMDR, a TCP/IP LAN Address, Subnet Mask Address, and Gateway Address must be assigned to the Active CPU. Use either ASYDL, SYS1, Indexes 515 through 526 to assign these addresses or the ADTM command.

[Node B]

ASYD	System Data 1	Index 180	Assign the PC of Node B.
		Index 182	Assign the PC of Node C.
		Indexes 288 - 295	Assign all 00H.
ASYDL	System Data 1	Index 577	Assign 00H.
		Indexes 578 - 581	Assign all 00H.

[Node C]

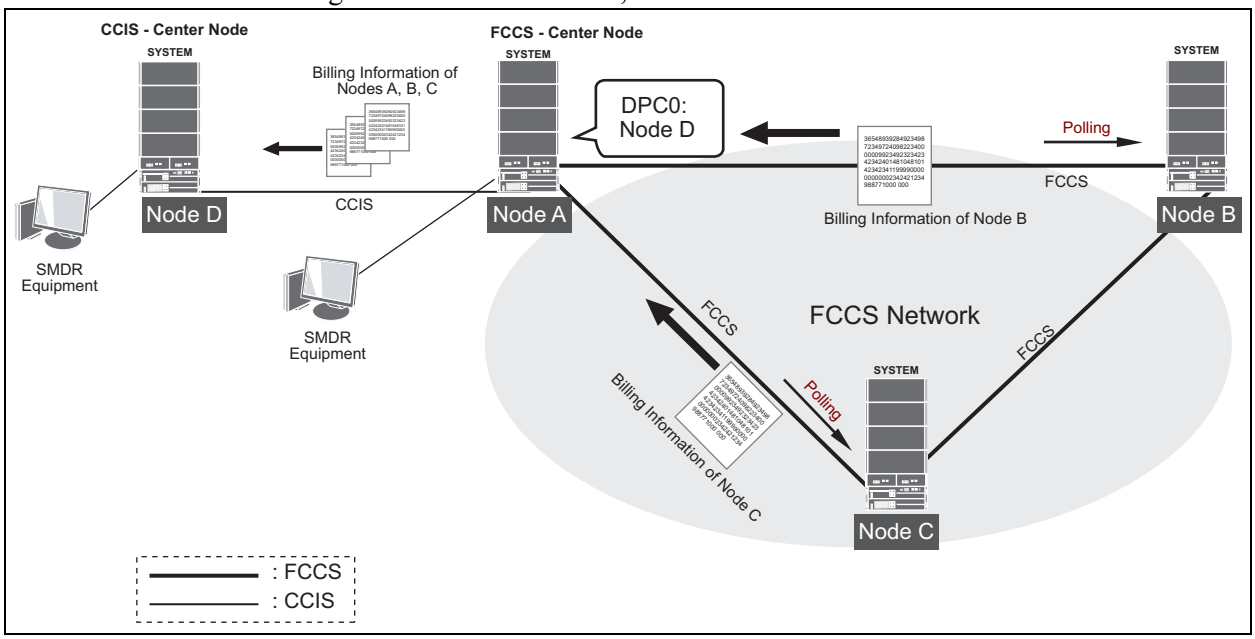
ASYD	System Data 1	Index 180	Assign the PC of Node C.
		Indexes 288 - 295	Assign when RS-232C output is used.
ASYDL	System Data 1	Index 578	Assign when LAN output is used. (Bit 4 **Note 7** of each Index)

Note 7: When using LAN Interface for SMDR, a TCP/IP LAN Address, Subnet Mask Address, and Gateway Address must be assigned to the Active CPU. Use either ASYDL, SYS1, Indexes 515 through 526 to assign these addresses or the ADTM command.

Example 6

- Node A collects the billing information of Node B and Node C via FCCS, using the polling method.

- Node A sends the billing information of Node A, Node B and Node C to Node D.



[Common Data]

ASYD	System Data 1	Indexes 32, 34	
	System Data 2	Index 3	

[Node A]

ASYD	System Data 1	Index 180	Assign the PC of Node A.
		Index 182	Assign the PC of Node D.
		Index 288	Assign when RS-232C output is used.
ASYDL	System Data 1	Index 576	Assign Bit 0 = 1 for Centralized FCCS SMDR In Service.
		Index 576	Assign Bit 2 = 1 (When using LAN interface for SMDR)
		Index 577	Assign 00H.
		Index 578	Assign when LAN output is used. (Bit 4 **Note 8** of each Index)
		Index 608	Assign Node B and Node C.

Note 8: When using LAN Interface for SMDR, a TCP/IP LAN Address, Subnet Mask Address, and Gateway Address must be assigned to the Active CPU. Use either ASYDL, SYS1, Indexes 515 through 526 to assign these addresses or the ADTM command.

[Node B and Node C]

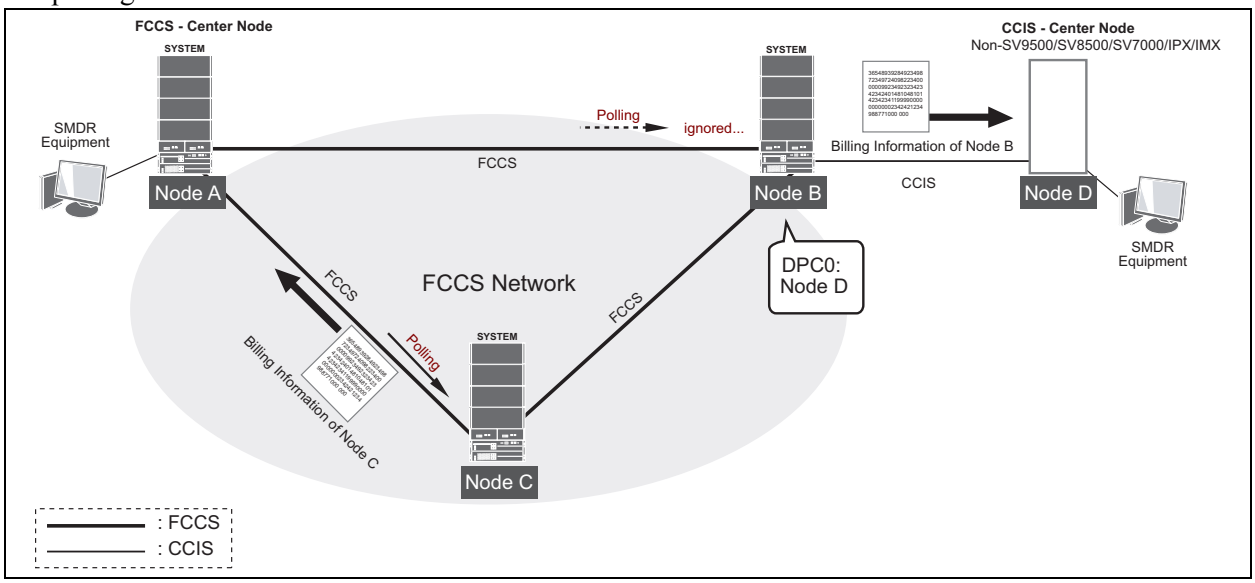
ASYD	System Data 1	Indexes 288 - 295	Assign all 00H.
ASYDL	System Data 1	Index 576	Assign 01H.
		Index 577	Assign the FPC of Node A.
		Indexes 578 - 581	Assign all 00H.

[Node D]

ASYD	System Data 1	Index 180	Assign the PC of Node D.
		Index 288	Assign when RS-232C output is used.

Example 7

- Node A collects the billing information of Node B and Node C via FCCS, using the polling method (Billing information of Node B cannot be collected).
- Node B sends the billing information to Node D via CCIS, ignoring polling from Node A.
- Node B deletes the Point Code (CCIS) for Node D when Node B wants to send billing information for polling from Node A.



[Common Data]

ASYD	System Data 1	Indexes 32, 34	
	System Data 2	Index 3	

[Node A]

ASYD	System Data 1	Index 288	Assign when RS-232C output is used
ASYDL	System Data 1	Index 576	Assign Bit 0 = 1 for Centralized FCCS SMDR In Service.
		Index 576	Assign Bit 2 = 1 (when using LAN interface for SMDR)
		Index 577	Assign 00H.
		Index 578	Assign when LAN output is used. (Bit 4 **Note 9** of each Index)
		Index 608	Assign Node C.

Note 9: When using LAN Interface for SMDR, a TCP/IP LAN Address, Subnet Mask Address, and Gateway Address must be assigned to the Active CPU. Use either ASYDL, SYS1, Indexes 515 through 526 to assign these addresses or the ADTM command.

[Node B]

ASYD	System Data 1	Index 180	Assign the PC of Node B
		Index 182	Assign the PC of Node D
		Indexes 288 - 295	Assign all 00H.
ASYDL	System Data 1	Index 576	Assign 00H.
		Index 577	Assign 00H.
		Indexes 578 - 581	Assign all 00H.

[Node C]

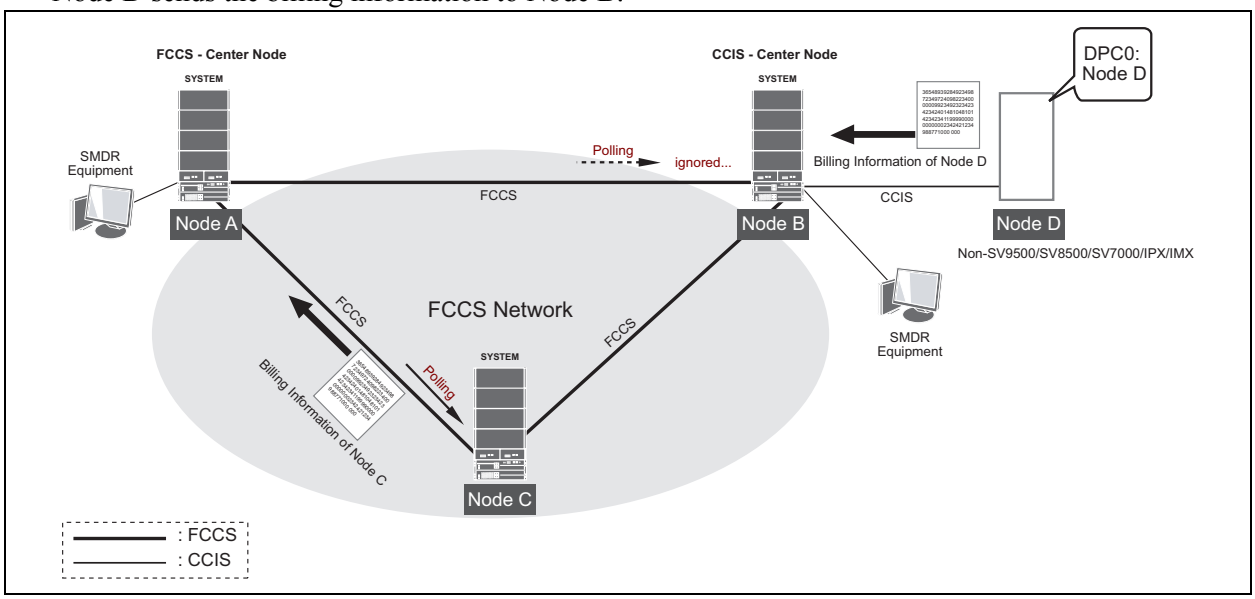
ASYD	System Data 1	Indexes 288 - 295	Assign all 00H.
ASYDL	System Data 1	Index 576	Assign 01H
		Index 577	Assign the FPC of Node A.
		Indexes 578 - 581	Assign all 00H.

[Node D]

ASYD	System Data 1	Index 180	Assign the PC of Node D.
		Index 288	Assign when RS-232C output is used.

Example 8

- Node A collects the billing information of Node B, Node C and Node D, which are stored in Node B by polling method.
- Node D sends the billing information to Node B.



[Common Data]

ASYD	System Data 1	Indexes 32, 34
	System Data 2	Index 3

[Node A]

ASYD	System Data 1	Index 288	Assign when RS-232C output is used.
ASYDL	System Data 1	Index 576	Assign Bit 0 = 1 for Centralized FCCS SMDR In Service.
		Index 576	Assign Bit 2 = 1 (when using LAN interface for SMDR)
		Index 577	Assign 00H.
		Index 578	Assign when LAN output is used. (Bit 4 **Note 10** of each Index)
		Index 608	Assign Node B and Node C.

Note 10: When using LAN Interface for SMDR, a TCP/IP LAN Address, Subnet Mask Address, and Gateway Address must be assigned to the Active CPU. Use either ASYDL, SYS1, Indexes 515 through 526 to assign these addresses or the ADTM command.

[Node B]

ASYD	System Data 1	Index 180	Assign the PC of Node B
		Index 288	Assign when RS-232C output is used.
ASYDL	System Data 1	Index 576	Assign Bit 0 = 1 for Centralized FCCS SMDR In Service.
		Index 576	Assign Bit 2 = 1, when using LAN interface for SMDR.
		Index 577	Assign the FPC of Node A.
		Indexes 578 - 581	Assign when LAN output is used. (Bit 4 **Note 11** of each Index).

Note 11: When using LAN Interface for SMDR, a TCP/IP LAN Address, Subnet Mask Address, and Gateway Address must be assigned to the Active CPU. Use either ASYDL, SYS1, Indexes 515 through 526 to assign these addresses or the ADTM command.

[Node C]

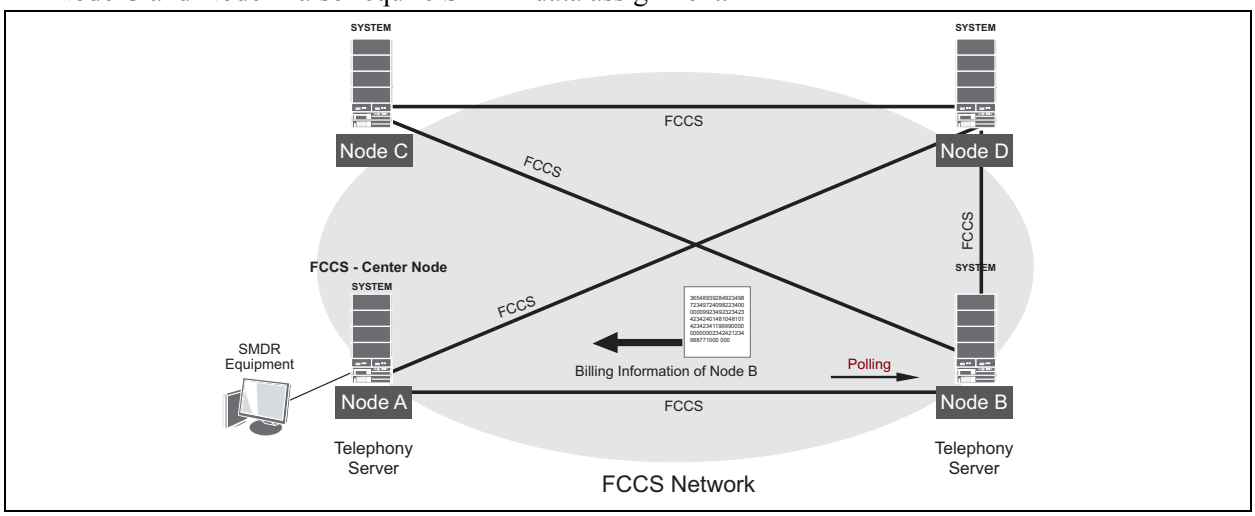
ASYD	System Data 1	Indexes 288 - 295	Assign all 00H.
ASYDL	System Data 1	Index 576	Assign 01H
		Index 577	Assign the FPC of Node A.
		Indexes 578 - 581	Assign all 00H.

[Node D]

ASYD	System Data 1	Index 180	Assign the PC of Node D.
		Index 182	Assign the PC of Node B.
		Indexes 288 - 295	Assign all 00H.

Example 9

- Node A collects the billing information of Node B and the Self Node.
- Node C and Node D also require SMDR data assignment.



[Common Data]

ASYD	System Data 1	Indexes 32, 34	
	System Data 2	Index 3	
ASYDN		Index 583	Assign 00H.

[Node A]

ASYD	System Data 1	Index 288	Assign when RS-232C output is used.
ASYDL	System Data 1	Index 576	Assign Bit 0 = 1 for Centralized FCCS SMDR In Service.
		Index 576	Assign Bit 2 = 1 (when using LAN interface for SMDR)

		Index 577	Assign 00H.
		Index 578	Assign when LAN output is used. (Bit 4 **Note 12** of each Index)
		Index 608	Assign Node B.

Note 12: When using LAN Interface for SMDR, a TCP/IP LAN Address, Subnet Mask Address, and Gateway Address must be assigned to the Active CPU. Use either ASYDL, SYS1, Indexes 515 through 526 to assign these addresses or the ADTM command.

[Node B]

ASYD	System Data 1	Indexes 288 - 295	Assign all 00H.
ASYDL	System Data 1	Index 576	Assign 01H
		Index 577	Assign the FPC of Node A.
		Indexes 578 - 581	Assign all 00H.

[Node C and Node D]

ASYD	System Data 1	Indexes 288 - 295	Assign all 00H.
ASYDL	System Data 1	Index 576	Assign 00H
		Index 577	Assign 00H.
		Indexes 578 - 581	Assign all 00H.

<SOCKET Interface (LAN Interface) Conditions>

1. When you use the SMDR system with a LAN interface, be aware of the following:
   - When connecting to the SMDR system, simultaneous use of LAN and RS-232C connection is not available in a single node.
   - When you change the interface for the SMDR system (from LAN to RS-232C and vice versa), all the data collected in the Telephony Server is erased. Therefore, before performing such operation, output all the collected data to the SMDR system.
2. SMDR data can be output via LAN cable.
3. A maximum of four SMDR apparatuses per node are available.
4. When providing this feature with LAN interface, IMX text format must be designated by the ASYDL command (System Data 1, Index 578-581, Bit 0).
5. The LAN and RS232C interfaces cannot be used simultaneously for connecting SMDR equipment in the same node.

<Relocation Condition for SMDR>

When relocating SMDR equipment to another node, reassign the billing information-related data. The relocation conditions are shown below.

1. Make sure there is no billing information in each node from the Center Node before changing the SMDR location. (make sure billing information is output to the SMDR by sending test signal.)
2. Exact billing information may not be output to SMDR while changing the SMDR location.
3. When removing the SMDR equipment from the Center Node, first delete all of the billing information-related data, and also when connecting the SMDR to new-Center Node, the user must, first of all, assign appropriate billing information-related data.

4. Assign the Fusion Point Code (FPC) of the new-Center Node to all nodes including the previous Center Node by ASYDL (System Data 1 Index 577).

<Trunk Arrival/Sending Time and Abandoned Call Information (for Incoming/Outgoing Trunk Call)>

The following information on Incoming/Outgoing Trunk Call can be output to SMDR:

- Trunk Sending Time (when an outgoing trunk call is sent from the trunk) **Note 13**
- Abandoned Call as billing information for outgoing trunk call **Note 13**
- Trunk Arrival Time (when an incoming trunk call is terminated to the terminal) **Note 14**
- Abandoned Call as billing information for incoming trunk call **Note 14**.

Note: Answered call and abandoned call are defined for this feature as follows:

- Answered Incoming Trunk Call:
  Calling party releases the call after sending answer signal to the incoming trunk.
  Call transferred to Announcement trunk by [C-25] Call Forwarding - Intercept/Announcement/[C-69] Call Forwarding - All Calls - Announcement/[O-21] Outgoing Trunk Busy Announcements/[S-106] Slumber Time - Do Not Disturb (routed to Announcement Trunk) is counted as answered call.
  Call disconnected by [A-28] Automatic Announcement - Disconnected PS Out of Zone is counted as answered call (until the PS is in out of zone state).
- Answered Outgoing Trunk Call:
  Calling party seizes a trunk and releases the call after receiving answer signal to the outgoing trunk.
- Abandoned Incoming Trunk Call:
  Calling party releases the call before sending answer signal to the incoming trunk.
- Abandoned Outgoing Trunk Call:
  Calling party seizes a trunk and releases the call before receiving answer signal to the outgoing trunk.

Note 13: Available from FP95-113 V3.

Note 14: Available in EMEA from FP95-111 V1 and available in other countries from FP95-113 V3.

1. By using Trunk Arrival/Sending Time information, SMDR can calculate the ringing time (the time elapsed between Trunk Arrival/Sending and Call Start). The calculation method is as follows:

   - In the case of Answered Incoming Trunk Call:

     Ringing Time = Call Start Time - Trunk Arrival Time

   - In the case of Abandoned Incoming Trunk Call:

     Ringing Time = Call End Time (Ringing stop time) - Trunk Arrival Time

- In the case of Answered Outgoing Trunk Call:

  Ringing Time = Call Start Time - Trunk Sending Time

- In the case of Abandoned Outgoing Trunk Call:

  Ringing Time = Call End Time (Ringing stop time) - Trunk Sending Time

Examples:

- In the case of Answered Incoming Trunk Call:

Trunk Arrival Time	2015/12/01	15:30:02:000
Call Start Time	2015/12/01	15:30:24:500
Call End Time	2015/12/01	15:35:15:500

- In the case of Abandoned Incoming Trunk Call:

Trunk Arrival Time	2015/12/01	15:30:02:200
Call Start Time	Blank (**Note 15**)	
Call End Time	2015/12/01	15:30:57:500

- In the case of Answered Outgoing Trunk Call:

Trunk Sending Time	2015/12/01	15:30:02:000
Call Start Time	2015/12/01	15:30:24:500
Call End Time	2015/12/01	15:35:15:500

- In the case of Abandoned Outgoing Trunk Call:

Trunk Sending Time	2015/12/01	15:30:02:200
Call Start Time	Blank (**Note 15**)	
Call End Time	2015/12/01	15:30:57:500

Note 15: A call is considered abandoned when the Call Start Time is blank and there is an entry for Call End Time.

2. Incoming and outgoing billing information messages must be activated to respectively output Trunk Arrival Time and Abandoned Incoming Call information and Trunk Sending Time and Abandoned Outgoing Call information. Data cannot be output with station to station billing information messages.
3. Trunk Arrival/Sending Time and Abandoned Call information can be provided when a call from or to C.O. line is terminated to or sent from Station, Attendant Console or trunk.
4. Trunk Arrival/Sending Time and Abandoned Call information can be provided on a route basis.
5. Called Party Number information on Incoming Abandoned Call is as follows:
   a.) When called Station or Attendant Console can be identified, its number is output.
   b.) When called Station cannot be identified ([T-23D] Trunk Line Appearance -D, etc.), the following information is output:

   ORIG: 0 (Station)
   Tenant: Tenant Information of Incoming Trunk
   Called Party Number: Blank (Called Party cannot be identified)

c.) When called Attendant Console cannot be identified ([D-114] Delay Announcement - Attendant, [A-55] Automatic Change of Night Service, Ring-down termination or [S-106] Slumber Time - Do Not Disturb (routed to Attendant Console), etc.), the following information is output:

ORIG: 1 (Attendant Console)
Tenant: Tenant Information of Incoming Trunk
Called Party Number: Blank (Called Party cannot be identified)

d.) When [A-82] Automated Attendant/[R-2] Remote Access to System call is abandoned while the Calling Party Number is dialed, the following information is output:

ORIG: 0 (Station)
Tenant: Tenant Information of Incoming Trunk
Called Party Number: Blank (Called Party cannot be identified)

e.) When an incoming call is forwarded by the following features, the forwarding destination number is output as the Called Party Number:

[C-5] Call Forwarding - All Calls
[C-2] Call Forwarding - Busy Line
[L-53] Logged Out IP Station - Call Destination
[C-103] Calling Party Recognition Service (Call Forwarding-All Calls/Busy Line)

f.) When an incoming call is forwarded by the following features, the number of the party that set the forwarding destination is output as Called Party Number. Even when the call is forwarded to second hop forwarding destination, the first forwarding source number is output.

[C-3] Call Forwarding - Don't Answer
[C-103] Calling Party Recognition Service (Don't Answer)
[C-74] Call Forwarding - Don't Answer - Outside
[C-47] Call Forwarding - Don't Answer - CCIS
[C-155] Call Forwarding PS Incoming Call Incomplete

g.) When an incoming call is forwarded by the following features, the trunk number that is used for the forwarding feature is output as Called Party Number.

[C-28] Call Forwarding - All Calls - Outside
[C-45] Call Forwarding - All Calls - CCIS
[C-60] Call Forwarding - Busy Line - Outside
[C-46] Call Forwarding - Busy Line - CCIS
[T-1] Tandem Switching of TIE Trunks - 2/4-Wire

h.) When an incoming call is forwarded by the following features, Blank is output as Called Party Number.

Unused Number/Dead Level Number-Announcement

i.) For features activated while in busy state such Transfer-All Calls and Blind Transfer, billing information on abandoned call is not provided.

j.) When a multiline terminal receives a call to a subline/virtual line, the subline/virtual line number is used for called party information.

k.) When a PS/SIP line is set to a Line/Feature key of the Desktop terminal as a subline by PS Line

Appearance on Dterm [P-71] and the Desktop terminal receives a call to the PS/SIP line, the PS/SIP line number is used for called party information.

6. Calling Party Number information on Outgoing Abandoned Call is as follows:
   a.) The number of the station or Attendant Console that seized the trunk is displayed. For stations, the information is displayed as follows:

   ORIG: 0 (Station)
   Tenant: Tenant Information of Outgoing Trunk
   Calling Party Number: The station that makes the call

   For attendant consoles, the information is displayed as follows:

   ORIG: 1 (Attendant Console)
   Tenant: Tenant Information of Outgoing Trunk
   Calling Party Number: The Attendant Console that makes the call

   b.) In the following features, when ASYD, SYS1, Index 20, Bit 7=0, the billing party is the originator of the transferred call. When Bit 7=1, the billing party is the station that has set the feature.

   [C-28] Call Forwarding - All Calls - Outside
   [[C-60] Call Forwarding - Busy Line - Outside
   [C-74] Call Forwarding - Don't Answer - Outside
   [L-53] Logged Out IP Station - Call Destination

   c.) When a multiline terminal seizes a trunk using a subline/virtual line, the my line number is used for calling party information.
   d.) When a PS/SIP line is set to a Line/Feature key of a Desktop terminal as a subline by PS Line Appearance on Dterm [P-71] and the Desktop terminal seizes a trunk with the PS/SIP line, the PS/SIP line number is used for called party information.
7. In the following cases, billing information of the terminal receiving the call is not output to SMDR (Forwarding feature is not set): **Note 16**

   - Target terminal is busy
   - Target terminal is unassigned terminal **Note 17**
   - Connection between target terminal and incoming trunk is restricted **Note 17**
   - Target terminal is in Do Not Disturb mode
   - Target terminal is logged out
   - Register is busy **Note 18**

Note 16: If a forwarding feature such as Call Forwarding - All Calls/Busy/Don't Answer is set for the terminal receiving the call, the specific billing information output conditions of the feature are applied.

Note 17: When ASYD, SYS1, Index 7, Bit 5 = 1:
If unassigned number or restricted number is dialed, the following information is output:
- ORIG: 0 (Station)
- Tenant: Tenant Information of Incoming Trunk
- Called Party Number: Blank (Called Party cannot be identified)

Note 18: Only for features that use the register for incoming calls.

8. In the following cases, billing information of the terminal sending the call is not output to SMDR (Forwarding feature is not set): **Note 19**
   - The seized trunk is busy
   - Connection between the terminal and incoming trunk is restricted
   - Target terminal is busy
   - Target terminal is logged out
   - Target terminal is in Do Not Disturb mode
   - Target terminal is unassigned terminal

   Note 19: If a forwarding feature such as Call Forwarding - All Calls/Busy/Don't Answer is set for the target terminal, the specific billing information output conditions of the feature are applied.

9. In the case of a tandem connection, both Incoming and Outgoing Abandoned Calls information is output.
10. When total billing is used, if a call is abandoned after being transferred via blind transfer, the following call information is provided:
    a.) In an outgoing/incoming trunk call of terminal B, terminal A transfers the call to terminal C (connected with external line), but terminal B abandons the call before terminal C answers the call.
    - Call information between terminal B and the trunk is output as an answered call
    - Call information between terminal C and the trunk is output as an abandoned call

    b.) In an outgoing/incoming trunk call of terminal B, terminal A transfers the call to terminal C (connected with extension line), but terminal B abandons the call before terminal C answers the call.
    - Call information between terminal A and the trunk is output as an answered call

11. When split billing is used, if a call is abandoned after being transferred via blind transfer, the following call information is provided:
    a.) In an outgoing/incoming trunk call of terminal B, terminal A transfers the call to terminal C (connected with external line), but terminal B abandons the call before terminal C answers the call.
    - Call information between terminal A and the trunk is output as an answered call
    - Call information between terminal B and the trunk is output as an answered call
    - Call information between terminal C and the trunk is output as an abandoned call

    b.) In an outgoing/incoming trunk call of terminal B, terminal A transfers the call to terminal C

(connected with extension line), but terminal B abandons the call before terminal C answers the call.

- Call information between terminal A and the trunk is output as an answered call

12. In an Incoming Abandoned Call, neither the MA-ID of both the sender and the receiver is output.
13. The relationship between the time until occurrence of billing buffer overflow and the number of incoming/outgoing calls (Answered Call and Abandoned Call) per one hour is as follows:

5,000 calls/hour → 9.40 hours
10,000 calls/hour → 4.70 hours
15,000 calls/hour → 3.13 hours
20,000 calls/hour → 2.35 hours
20,000 calls/hour → 1.88 hours
30,000 calls/hour → 1.56 hours
*billing buffer: 47, 000 calls

14. As in the following, if Split Billing is applied, the same value is output as Trunk Arrival Time for the conversations between "Trunk C and Terminal A" and "Trunk C and Terminal B".
    1.) An Incoming Trunk C Call is terminated to terminal A.
    2.) Trunk C and terminal A are engaged in a two-way conversation.
    3.) Trunk C is placed on hold, and the call is transferred to terminal B from terminal A.
    4.) Terminal A goes on-hook. trunk and terminal B are engaged in a two-way conversation. (Billing Information for the conversation between trunk C and terminal A is output.)
    5.) Conversation between trunk C and terminal B ends. (Billing Information for the conversation between trunk C and terminal B is output.)
15. When using Centralized Billing - CCIS feature, Trunk Arrival/Sending Time and Abandoned Call Information of Local Node is not sent to Center node.
16. Incoming Abandoned Call information is not output for the following features:
    [S-15] Serial Call
    [S-52] Serial Call - CCIS
17. When a call is established or released while in the following states, Trunk Arrival Time is not output:

- UCD Queuing
- Delay Announcement - UCD
- Overflow - UCD
- Call Waiting

18. The following shows the output availability of billing information on Incoming/Outgoing Abandoned Call:

Feature Name	Abandoned Call			
	Outgoing Call		Incoming Call	
	Output of Billing Information	Calling Party Number	Output of Billing Information	Called Party Number
ISDN Features				
[D-123] Direct Inward Dialing (DID) Addressing	-	-	×	Called Party
[C-104] Calling Party Recognition Service (Direct-In Termination [DIT])	-	-	×	Called Party
[C-103] Calling Party Recognition Service (CF-All Calls)	-	-	×	Forwarding Destination Terminal
[C-103] Calling Party Recognition Service (CF-Busy Line)	-	-	×	Forwarding Destination Terminal
[C-103] Calling Party Recognition Service (CF-Don't Answer)	-	-	×	Terminal that has been set as forwarding destination
PCS/PHS Features				
[P-71] PS Line Appearance on Dterm	× Note 20	The My Line of the Desktop terminal that accommodates the PS line	×	PS
[A-129] Automatic Announcement - Called PS Out of Zone	-	-	×	PS
[C-155] Call Forwarding PS Incoming Call Incomplete	-	-	×	PS
[A-128] Automatic Announcement - Disconnected PS Out of Zone	-	-	Answered Call	PS
Incoming Call Features				
[D-8] Direct Inward Dialing (DID)	-	-	×	Called Party
[R-2/A-82] Remote Access to System/Automated Attendant	-	-	×	Called Party
[D-7] Direct-In Termination (DIT)	-	-	×	Called Party
[T-23D] Trunk Line Appearance-D	-	-	×	Blank

Feature Name	Abandoned Call			
	Outgoing Call		Incoming Call	
	Output of Billing Information	Calling Party Number	Output of Billing Information	Called Party Number
[S-106] Slumber Time - Do Not Disturb (routed to Attendant Console)	-	-	×	Blank
[S-106] Slumber Time - Do Not Disturb (routed to Announcement Trunk)	-	-	Answered Call	Announce-ment Trunk
Outgoing Call Features				
[O-2] Outgoing Trunk Queuing **Note 21**	×	Calling Party	-	-
[O-7] Off-Hook Queuing **Note 21**	×	Calling Party	-	-
[R-6] Route Advance	×	Calling Party	-	-
[S-3] Speed Calling - System [S-21] Speed Calling - Station [S-23] Speed Calling - Group	×	Calling Party	-	-
Other Features				
[U-1] Uniform Call Distribution (UCD)	-	-	×	Called Party
[D-31] Delay Announcement - UCD	-	-	N	-
[O-13] Overflow - UCD **Note 22**	-	-	N	-
[C-25] Call Forwarding - Intercept / Announce-ment	-	-	Answered Call	Announce-ment Trunk
[A-20] Authorization Code	×	Calling Party	-	-
CCIS Features				
[C-45] Call Forwarding - All Calls - CCIS	-	-	×	Trunk
[C-46] Call Forwarding - Busy Line - CCIS	-	-	×	Trunk
[C-47] Call Forwarding - Don't Answer - CCIS	-	-	×	Terminal that has been set as forwarding destination
[S-52] Serial Call - CCIS	-	-	N	-
Attendant Console Features				
[S-15] Serial Call	-	-	N	-
[D-114] Delay Announcement - Attendant	-	-	×	Blank
[A-55] Automatic Change of Night Service	-	-	×	Blank

Feature Name	Abandoned Call			
	Outgoing Call		Incoming Call	
	Output of Billing Information	Calling Party Number	Output of Billing Information	Called Party Number
Station Features				
[C-12] Call Waiting - Terminating	-	-	N	-
Unused Number/Dead Level Number-Announcement	-	-	×	Blank
[C-5] Call Forwarding - All Calls	-	-	×	Forwarding Destination Terminal
[C-2] Call Forwarding - Busy Line	-	-	×	Forwarding Destination Terminal
[C-3] Call Forwarding - Don't Answer	-	-	×	Terminal that has been set as forwarding destination
[C-28] Call Forwarding - All Calls - Outside	-	-	×	Trunk
[C-60] Call Forwarding - Busy Line - Outside	-	-	×	Trunk
[C-74] Call Forwarding - Don't Answer - Outside	-	-	×	Terminal that has been set as forwarding destination
[S-7] Station Hunting - Circular [S-8] Station Hunting - Secretarial [S-9] Station Hunting - Terminal [S-138] Station Hunting - Priority [S-139] Station Hunting - Priority with Switchback	-	-	×	Called Terminal
[C-69] Call Forwarding - All Calls - Announcement	-	-	Answered Call	Announcement Trunk
[C-28] Call Forwarding - All Calls - Outside **Note 23**	×	Calling party/ Service setting party	-	-
[C-60] Call Forwarding - Busy Line - Outside **Note 23**	×	Calling party/ Service setting party	-	-
[C-74] Call Forwarding - Don't Answer - Outside **Note 23**	×	Calling party/ Service setting party	-	-
[C-11 Call Transfer - All calls] **Note 24**	×	Transfer party	-	-

Feature Name	Abandoned Call			
	Outgoing Call		Incoming Call	
	Output of Billing Information	Calling Party Number	Output of Billing Information	Called Party Number
[B-18 Blind Transfer to Station] **Note 24**	×	Transfer party	-	-
Desktop Terminal Features				
Multiline **Note 25**	×	The My Line of the Desktop terminal that accommodates the Virtual Line/Sub Line line	-	-
Network Features				
[T-1] Tandem Switching of TIE Trunks - 2/4-Wire	×	Calling party	×	Trunk
[O-21] Outgoing Trunk Busy Announcements	-	-	Answered Call	Announce-ment Trunk
[F-41] Flexible Network Re-routing **Note 26**	×	Calling party	-	-
IP Features				
[L-53]Logged Out IP Station - Call Destination	× **Note 27**	Calling party/ Service set-ting party	×	Destination Terminal

Note 20: In case the call is abandoned after the PS line accommodated in the Desktop terminal is seized and the call is sent from the trunk.

Note 21: When a call is abandoned after a trunk becomes idle and is used to send the call.

Note 22: When a call is abandoned while waiting in the first UCD queue:

Note 23: In case the call is abandoned after the transfer terminal seizes a trunk and the call is sent.
When ASYD, SYS1, Index 20, Bit 7=0, the billing party is the originator of the transferred call. When Bit 7=1, the billing party is the station that has set the feature.

Note 24: In case the call is abandoned after the transfer to an external line is performed.

Note 25: In case the call is abandoned after the Sub Line/Virtual Line accommodated in the Desktop terminal is seized and the call is sent from the trunk.

Note 26: In case the call is abandoned after a network failure is detected and an external call based on the automatic rerouting is performed.

Note 27: When the call is abandoned after the transfer terminal seizes a trunk and sends the call. When ASYD, SYS 1, Index 20, Bit 7= 0, the billing party is the originator of the transferred call. When Bit 7= 1, the billing party is the station that has set the feature.

<SMDR for Trunk Calls Made Using Access Code with Variation in Number of Digits>

The destination number of an outgoing call recorded in SMDR is either the actual number dialed by the user or the number sent by the system. This is basically determined by the System Data (ASYD, System Data 1, Index 32, Bit 6). In the case of an outgoing trunk call via an Access Code, when there is a variation in the number of digits of Access Codes, which of the two is shown on SMDR is determined by the factors listed below:

- System Data (ASYD, System Data 1, Index 32, Bit 6)
- The necessary number of digits for Access Codes (the value specified for the NND parameter of the ANPD command)
- The number of digits of Access Code (the number of digits of the value specified for the ACC parameter of the ASPA command)

The following shows how these factors affect the destination number information to be recorded in SMDR (where "N" represents the value specified for the NND parameter of the ANPD command, and "A" for the number of digits of the value specified for the ACC parameter of the ASPA command):

(a) When N is less than A

ASYD SYS 1, Index 32	ARTD/ARTDN CDN 121: CONV = 0 (Converted Number)	ARTD/ARTDN CDN 121: CONV = 1 (Dialed Number)
Bit 6 = 0 (Number sent by system)	Access Code **Note 28** + Number sent to the trunk **Note 29**	Access Code **Note 28** + Number sent to the trunk **Note 29**
Bit 6 = 1 (Number dialed by user)	Actual number dialed by user	Actual number dialed by user

Note 28: When an Access Code is set to be included in the destination number of an outgoing call recorded in SMDR(ASYD, System Data 1, Index 32, Bit 5 = 1), the system recognizes the first N digits of the dialed number as an Access Code (where N is the value specified by the NND parameter of the ANPD, ANPDL or ANPDN command).

Note 29: Additional Digit Information (DC), if any, followed by a string obtained by deleting the first x digits from the dialed number (where x is the value specified for the SKIP parameter of the AOPR, AOPRL or AOPRN command).

Example

Case Where User Dials “1#1 58059”

ANPD (1stDC = 1, NND = 2), ASPA (ACC = 1#1), AOPR (SKIP = 3, PNL ≠ 0), AADC (DC = 99)			
		ARTD/ARTDN CDN 121: CONV = 0 (Converted Number)	ARTD/ARTDN CDN 121: CONV = 1 (Dialed Number)
ASYD SYS 1, Index 32	Bit 6 = 0 (Number sent by system)	1# 99 58059	1# 99 58059
	Bit 6 = 1 (Number dialed by user)	1#1 58059	1#1 58059

(b) When N is greater than or equal to A

		ARTD/ARTDN CDN 121: CONV = 0 (Converted Number)	ARTD/ARTDN CDN 121: CONV = 1 (Dialed Number)
ASYD SYS 1, Index 32	Bit 6 = 0 (Number sent by system)	Access Code **Note 30** + Number sent to the trunk **Note 31**	Actual number dialed by user
	Bit 6 = 1 (Number dialed by user)	Actual number dialed by user	Actual number dialed by user

Note 30: When an Access Code is set to be included in the destination number of an outgoing call recorded in SMDR(ASYD, System Data 1, Index 32, Bit 5 = 1), the system recognizes the first N digits of the dialed number as an Access Code (where N is the value specified by the NND parameter of the ANPD, ANPDL or ANPDN command).

Note 31: Additional Digit Information (DC), if any, followed by a string obtained by deleting the first x digits from the dialed number (where x is the value specified for the SKIP parameter of the AOPR, AOPRL or AOPRN command).

Example 1

Case Where User Dials "1# 58059"

ANPD (1stDC = 1, NND = 2), ASPA (ACC = 1#), AOPR (SKIP = 2, PNL ≠ 0), AADC (DC = 99)			
		ARTD/ARTDN CDN 121: CONV = 0 (Converted Number)	ARTD/ARTDN CDN 121: CONV = 1 (Dialed Number)
ASYD SYS 1, Index 32	Bit 6 = 0 (Number sent by system)	1# 99 58059	1# 58059
	Bit 6 = 1 (Number dialed by user)	1# 58059	1# 58059

Example 2

Case Where User Dials "1#1 58059"

ANPD (1stDC = 1, NND = 3), ASPA (ACC = 1#1), AOPR (SKIP = 3, PNL ≠ 0), AADC (DC = 99)			
		ARTD/ARTDN CDN 121: CONV = 0 (Converted Number)	ARTD/ARTDN CDN 121: CONV = 1 (Dialed Number)
ASYD SYS 1, Index 32	Bit 6 = 0 (Number sent by system)	1#1 99 58059	1#1 58059
	Bit 6 = 1 (Number dialed by user)	1#1 58059	1#1 58059

Example 3

Case Where User Dials "1# 58059"

ANPD (1stDC = 1, NND =3), ASPA (ACC = 1#), AOPR (SKIP = 2, PNL ≠ 0), AADC (DC = 99)			
		ARTD/ARTDN CDN 121: CONV = 0 (Converted Number)	ARTD/ARTDN CDN 121: CONV = 1 (Dialed Number)
ASYD SYS 1, Index 32	Bit 6 = 0 (Number sent by system)	1#5 99 58059	1# 58059
	Bit 6 = 1 (Number dialed by user)	1# 58059	1# 58059

Interactions

1. If LCR(S) is provided, the LCR(S) Access Code is listed in ARNP. If an outgoing route is not used in LCR(S), then list the OGCA Access Code in ARNP.
2. To test SMDR output, it is recommended that a serial printer, breakout box, or a TCP/IP based tool be used to determine if the system is sending data.
3. CALL TRANSFER-ALL CALLS [C-11]: Either the first Station, last Station, or split billing is allowed with SMDR. Split billing provides a separate call record for both before and after a Call Transfer.
4. ATTENDANT CAMP-ON WITH TONE INDICATION [A-1]: The Called Station Number is recorded for Incoming Central Office Calls via the Attendant Console (RRI 2 of ARSC). To record the time a Destination Station is on a call, split billing or last Station billing is required. With split billing, the time the Attendant Console is on the call is separated from the time the Destination Station is on the call. For last Station billing, the Destination Station is charged with the entire time of Connection.
5. LEAST-COST ROUTING-3/6-DIGIT [L-5]: SMDR will record the digits sent rather than the digits dialed.
6. The Destination Station is recorded on SMDR for the following features:

CALL FORWARDING	[C-2, 3, 5]
CALL PICKUP	[C-7, 30]

   Note: CALL PICKUP - GROUP [C-7] and CALL PICKUP - DIRECT [C-30]: When a billing after transfer over different MA-IDs is in service (ASYDL, SYS 1, Index 589, Bit 3 = 1), the SMDR record is created.

7. CALL FORWARDING-INTERCEPT [C-25]: An Incoming Trunk Call responds to Call Forwarding-Intercept to the Attendant Console. SMDR records the call as if it is a direct call to the Attendant Console.
8. NIGHT CONNECTION-FIXED/FLEXIBLE [N-1, 2]: Calls terminated to these Stations are recorded on SMDR.
9. OUTGOING TRUNK QUEUING [O-2]: SMDR records the Outgoing Call when the system recalls the user and the digits are sent, not when the Trunk is queued.
10. REMOTE ACCESS TO SYSTEM [R-2]: The Incoming Route and Trunk Number is recorded as the Calling Party. If FORCED ACCOUNT CODE [F-7] is used, it will be recorded on SMDR. REMOTE ACCESS TO SYSTEM via a DID will not record FORCED ACCOUNT CODE on SMDR.
11. TRUNK-TO-TRUNK CONNECTION [T-10] and TIE LINE ACCESS [T-3]: SMDR records an Incoming TIE Line that uses TRUNK-TO-TRUNK CONNECTION to make an Outgoing Call. The TIE Line Route and Trunk Number is recorded as the Calling Party. The outgoing route is programmed for SMDR in ARTD, not the Incoming TIE Line Route.
12. CALL WAITING-TERMINATING [C-12]: SMDR will begin recording the call when the call is answered, not when a Call Waiting Tone is heard.

Programming

STEP 1: **ASYD** - System Data 1, Index 20, Bit 7. Charging to a call transferred by CALL FORWARDING-OUTSIDE Service 0/1: Charging is made to the caller of the Transferred Call / charging is made to the Station that has set CALL FORWARDING-OUTSIDE service.

System Data 1, Index 32
Bit 1 = 1 (records Tenant information)
Bit 2 = 1 (records route information)
Bit 5 = 1 (records Route Access Code)
Bit 6: 0/1 (records Number Sent/Number Dialed as SMDR Called Party Number Registration Type)
Note 32
Bit 7 = 1 (enables SMDR service)

Note 32: Please consider the following:

- In the case of an outgoing call made using a Speed Calling feature or a Last Number Called feature, the number sent by the Telephony Server is recorded as the SMDR destination number even if Bit 6 is set to 1.
- For the SMDR destination number in the case where there is a variation in the number of digits of Access Codes, see "<SMDR for Trunk Calls Made Using Access Code with Variation in Number of Digits>."

System Data 1, Index 33. Assign this Index for Split Billing. When Split Billing is used, assign 00H. When Total Billing is used, Bits 3 and 4 represent the following:

ASYD System Data 1, Index 33		
Bit 4	**Bit 3**	**Telephone Subject to Total Billing**
0	0	Telephone called first (regardless of calling party kind)
0	1	Telephone called last (regardless of calling party kind)
1 (**Note 33**)	0	First telephone after the call has been handled by ATT
1 (**Note 33**)	1	Telephone called last (regardless of calling party kind)

Note 33: When Bit 4 is set, the intermediate ATT information will be output to SMDR.

Bit 5 must be assigned as data "1" to provide total billing for Incoming Calls.
Bit 6 must be assigned as data "1" to provide total billing for Outgoing Calls.

Note: When using Split Billing, if an outgoing call is made via ISDN, billing information will not be displayed.

Note: When using Split Billing, the following condition must be considered.
When a Station transfers the trunk call to Attendant Console via BLIND TRANSFER [B-18], Total Billing for Incoming Call to the answered Station is output to SMDR.

System Data 1, Index 34. Assign data 01H or 21H. Bit 0 enables RS-232C transmission.
Bit 0: SMDR RS-232 C Output. 0/1 = Out of Service/In Service

Bit 1 and 2: Parity check of RS-232C Output. Bit 1=0, Bit 2=0 means no parity (parity bit fixed to "0").
Bit 5: Specify whether or not to send a converted number to the SMDR. 0/1 = Original Number is sent. / Converted Number (the first 6 digits) is sent.

Note: If you choose to send a converted number, the maximum length of the dial code output in the outgoing call record will be limited to 18 digits. If the number of digits dialed exceeds 18, the first 18 digits will be displayed as the dial number, and the remaining digits will be displayed as the first 6 digits of a converted number. If you do not want the converted number to be added to the dial code, set ASYDL System Data 1, Index 1245, Bit 6=1.

System Data 1, Index 58, Bit 7. Is the Specific Attendant Number, assigned by ASAT/ASATN command output to SMDR terminal? 0/1: No/Yes.

Note: When the Originating Point Code (OPC) is assigned by System Data 1 Index 180 and 181 and Attendant Console originates the call, the Specific Attendant Number is output even if System Data 1 Index 58 Bit 7 = 0.

System Data 1, Index 95. Maximum Buffer Capacity for Storing SMDR (RS-232C)
Bit 0-3: 0-9 (10-minute basis)
Bit 4-7: 0-9 (100-minute basis)

Note: Used for SMDR by RS-232C only. When 00H is assigned, 180 minutes (= 3hours) will be applied. 10-990 minutes (=16.5hours) can be assigned for billing by 10-minutes basis.

System Data 1, Index 157. Outgoing Trunk, Soft Hold Timer. The point in time when the system will recognize an Outgoing Call as answered. Default is 18 seconds. Because of interaction with momentary pressing of the hook-switch, this data may be reassigned as 31H for two seconds. Take the SMDR aspect into consideration when reassigning this data.

System Data 1, Index 174, Bit 5. SMDR output method in CCIS/ISDN tandem node. Specify the information n type of calling party in the outgoing call via CCIS (Calling Party Number in ISDN). (0/1 = Station Number/Route Number and Trunk Number of Incoming Trunk)

Note: In CCIS Network, the number is limited to six digits including the node number of RT=0 in the ARNP command.

Note: Set Bit 5 when using IS-11572/Dp Channel so that billing may be processed by using Route Number of Incoming Trunk. If 0 is assigned in this instance, billing will not applied correctly on the ground that the Calling Party Number cannot be referred to.

Note: If 0 is assigned for this data, when [T-1] Tandem Switching of TIE Trunks - 2/4-Wire is used in CCIS network, the billing information output at tandem node will be changed depending on the System Data - ASYD, Sys 1, Index 241, Bit 6. (Process to be executed when the length of Caller ID exceeds the limits)

[Example]

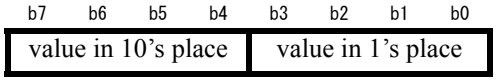
When Calling Party Number is 123456789 and limited to 8 digits:

In the case of Index 241, bit 6 =0, the calling party number will be set as 1234567*. (8 digits will be displayed including * that indicates that there is an information not being displayed. The six digits from the top digit will be output as the billing information. (123456))

In the case of Index 241, bit 6=1, the calling party number will be set as *3456789. (8 digits will be displayed including * that indicates that there is an information not being displayed. The six digits from the top digit will be output as the billing information. (*34567))

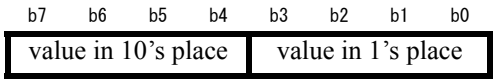
System Data 1, Index 249. SMDR output buffer usage rate to *output* Alarm Message (default = 80%)
Bit 0-3: value in the 1's place of buffer usage rate (0 - 9)
Bit 4-7: value in the 10's place of buffer usage rate (0 - 9)

Usage Rate to output Alarm Message:



System Data 1, Index 250. SMDR output buffer usage rate to *clear* Alarm Message (default = 50%)
Bit 0-3: value in the 1's place of buffer usage rate (0 - 9)
Bit 4-7: value in the 10's place of buffer usage rate (0 - 9)

Usage Rate to clear Alarm Message:



System Data 1, Indexes 288-295 (Required for SMDR by RS-232C only)
Designation of the attributes of the SMDR.
SMDR output port will be designated. For this designation, the attributes of the terminal that outputs the detail call information should have been designated by Indexes 117-123. (Assigned by the AIOC command.)
Also for detecting a fault of the SMDR, set the SMDR fault detect timer.
Index 288 = Charging information port of SMDR A.
Bit 4: IOC Equipment Number (0/1 = No.0/No.1)
Bit 5: Expanded SMDR Output (0/1 = Out/In Service)
Index 289 = SMDR fault detect timer of SMDR A. **Note 34**, **Note 35**
Bit 0-3: Fault detection timer of SMDR A Value for the first digit (minutes)
Bit 4-7: Fault detection timer of SMDR A Value for the second digit (minutes)
Index 290 = Charging information port of SMDR B.

Bit 4: IOC Equipment Number (0/1 = No.0/No.1)
Bit 5: Expanded SMDR Output (0/1 = Out/In Service)
Index 291 = SMDR fault detect timer of SMDR B. **Note 34**, **Note 35**
Bit 0-3: Fault detection timer of SMDR B Value for the first digit (minutes)
Bit 4-7: Fault detection timer of SMDR B Value for the second digit (minutes)
Index 292 = Charging information port of SMDR C.
Bit 4: IOC Equipment Number (0/1 = No.0/No.1)
Bit 5: Expanded SMDR Output (0/1 = Out/In Service)
Index 293 = SMDR fault detect timer of SMDR C. **Note 34**, **Note 35**
Bit 0-3: Fault detection timer of SMDR C Value for the first digit (minutes)
Bit 4-7: Fault detection timer of SMDR C Value for the second digit (minutes)
Index 294 = Charging information port of SMDR D.
Bit 4: IOC Equipment Number (0/1 = No.0/No.1)
Bit 5: Expanded SMDR Output (0/1 = Out/In Service)
Index 295 = SMDR fault detect timer of SMDR D. **Note 34**, **Note 35**
Bit 0-3: Fault detection timer of SMDR D Value for the first digit (minutes)
Bit 4-7: Fault detection timer of SMDR D Value for the second digit (minutes)

Note 34: (0-9) HEX is used only. (A-F) HEX is not used. If (A-F) HEX is assigned for bit 0-3 and bit 4-7, SMDR fault cannot be detected.

Note 35: In the case of 00 HEX (default), system message will not be output for SMDR fault.

System Data 2, Index 3. Bit 0 enables SMDR registration per Tenant.
System Data 2, Index 3. Optional assignments for Station-to-Station Connection.
Bit 5: Code Type (0/1 = Authorization/Forced Account) assign as "1" for Authorization Code to be present in SMDR.
Bit 7: Station-to-Station SMDR in service (0/1 = No/Yes)
System Data 1, Index 18, Bit 4. To enable the output of Trunk Arrival Time and Abandoned Call information, Assign as "1".

STEP 2: **ASYDL -** System Data 1, Index 576

Bit 0: Centralized Billing-FCCS (Polling Method): 0/1 = Not/To be provided. Required both for Center Node and each Local Node.

Bit 1: SMDR for Station to Station Connections in a Self Node 0/1 = output/not output **Note 36**

Bit 2: SMDR Interface Type 0/1 = RS-232C interface is used/LAN interface is used.

System Data 1, Index 577.

Fusion Point Code (FPC) of Center Node.

Provided that "1" is flagging in Bit 0 of Index 576 (System Data 1), let every Local Node recognize its belonging Center Node for centralized billing-FCCS. Assign data on Center Node FPC at each Local Node. Given this data, the Local Nodes can acknowledge the SMDR polling only from the Center Node specified here (the polling from other parties will be ignored).

At the Center Node, assign 00H.

System Data 1, Index 578-581, Bit 0. Assign correct data (0/1) for regulation of output text format pattern for centralized billing-FCCS. Index 578 corresponds to SMDR A, 579 to SMDR B, 580 to SMDR C and 581 to SMDR D. **Note 37** The pattern to be adopted will be:

Bit 0 = 0: ICS text format.

Bit 0 = 1: IMX format.

System Data 1, Index 582, Bit 0-3. Assign the number of nodes to be polled at once.

0000 = 1 0001 = 1 0010 = 2 0011 = 3 0100 = 4 0101 = 5
0110 = 6 0111 = 7 1000 = 8 1111 = 8 **Note 38**

Note 36: This data is valid when the data is assigned at System Data 1, Index 512.

Note 37: The same billing information can be output to a maximum of four pieces of SMDR equipment.

Note 38: For the numbers more than 8, assign the data value as 8.

System Data 1, Index 583, Bit 0-2. The timing for FCCS billing data transmission from the node that accommodates the Outgoing Trunk or the node that accommodates the Called Station to the node that accommodates the Calling Party (Valid when System Data 1, Index583, Bit 7 = 0 in ASYDN).

Bit 2 1 0
0 0 0 = eight seconds
0 0 1 = two seconds
0 1 0 = four seconds
0 1 1 = six seconds
Bit 2 1 0
1 0 0 = eight seconds
1 0 1 = 10 seconds
1 1 0 = 12 seconds
1 1 1 = 14 seconds

System Data 1, Index 608-639.

Fusion Point Code (FPC) of SMDR polling destination.

After the Center Node (multiple CN: possible on a network) is assigned for centralized billing-FCCS, let the Center Node recognize its polling destination (specified by the FPC), referring to the table below. In the table, each Bit represents Fusion Point Code ranging from 1 to 253. When "1" is set, the FPC is specified as the polling destination.

Index	b7	b6	b5	b4	b3	b2	b1	b0
608	7	6	5	4	3	2	1	-
609	15	14	13	2	11	10	9	8
610	23	22	21	20	19	18	17	16
611	31	30	29	28	27	26	25	24
612	39	38	37	36	35	34	33	32
613	47	46	45	44	43	42	41	40
614	55	54	53	52	51	50	49	48
615	63	62	61	60	59	58	57	56
616	71	70	69	68	67	66	65	64
617	79	78	77	76	75	74	73	72
618	87	86	85	84	83	82	81	80
619	95	94	93	92	91	90	89	88
620	103	102	101	100	99	98	97	96
621	111	110	109	108	107	106	105	104
622	119	118	117	116	115	114	113	112
623	127	126	125	124	123	122	121	120
624	135	134	133	132	131	130	129	128

Index	b7	b6	b5	b4	b3	b2	b1	b0
625	143	142	141	140	139	138	137	136
626	151	150	149	148	147	146	145	144
627	159	158	157	156	155	154	153	152
628	167	166	165	164	163	162	161	160
629	175	174	173	172	171	170	169	168
630	183	182	181	180	179	178	177	176
631	191	190	189	188	187	186	185	184
632	199	198	197	196	195	194	193	192
633	207	206	205	204	203	202	201	200
634	215	214	213	212	211	210	209	208
635	223	222	221	220	219	218	217	216
636	231	230	229	228	227	226	225	224
637	239	238	237	236	235	234	233	232
638	247	246	245	244	243	242	241	240
639	-	-	253	252	251	250	249	248

System Data 1, Index 641,
Bit 0: Type of the Station Number for the ICS text format.
0/1 = Physical Station Number/Telephone Number.

Note: If the Telephone Number is within 6 digits, the Telephone Number will be output.

Bit 3: Kind of the Route Number for the ICS text format.
0/1 = Physical Route Number/Logical Route Number.
System Data 1, Index 529, Bit 0-1: Assign the methods of parity check

00 = Odd Parity (Default)
01 = Odd Parity
10 = Even Parity
11 = No Parity check

System Data 1, Index 576, Bit 2: SMDR Interface Type 0/1 = RS-232C interface is used/LAN interface is used.
System Data 1, Index 578-581, Bit 4:Assign the usage of the SMDR.
Index 578, Bit 4: 0/1 = SMDR A is not used/used
Index 579, Bit 4: 0/1 = SMDR B is not used/used
Index 580, Bit 4: 0/1 = SMDR C is not used/used
Index 581, Bit 4: 0/1 = SMDR D is not used/used
Index 588, Bit 0: Output station number of transferring party to SMDR as originating party in Blind Transfer 0/1 = Out of service/In service

Note: For setting billing information for Outgoing Trunk Sending Time and Abandoned Call for Outgoing Trunk Call set Bit 0=1 for indexes 578, 579, 580 and 581 (available from FP95-113 V3).

Note: This data is valid when ASYD SYS1, Index 20, Bit 7=1 and Index 33=60 or 68 HEX. is assigned.

STEP 3: **ASYDN** - This data should be set "0" for ASYDL, System Data 1, Index 583, Bit 0-2. Index 583, Bit 7.
The node for saving FCCS billing data:
0 = The Node accommodating the Calling Party
1 = The Node accommodating the Outgoing Trunk or the Called Station

STEP 4: **AIOC** - Assign the function and attribute data of the IOC ports.
This data is required when using RS-232C interface, not necessary when using LAN interface.

STEP 5: **ADTM** - Specify the Port Number used for SMDR within the range of 1024 to 65534. When "0" is set, the default value (60010) is automatically used.

STEP 6: **ARTD** - CDN 10: SMDR. Enable outgoing routes to be included for SMDR registration.
CDN 16: SMDR2. Assign Detail Billing Information data. For details, see Command Manual.

Note: To enable Billing for Incoming Call, assign 1 to CDN16 (SMDR2).

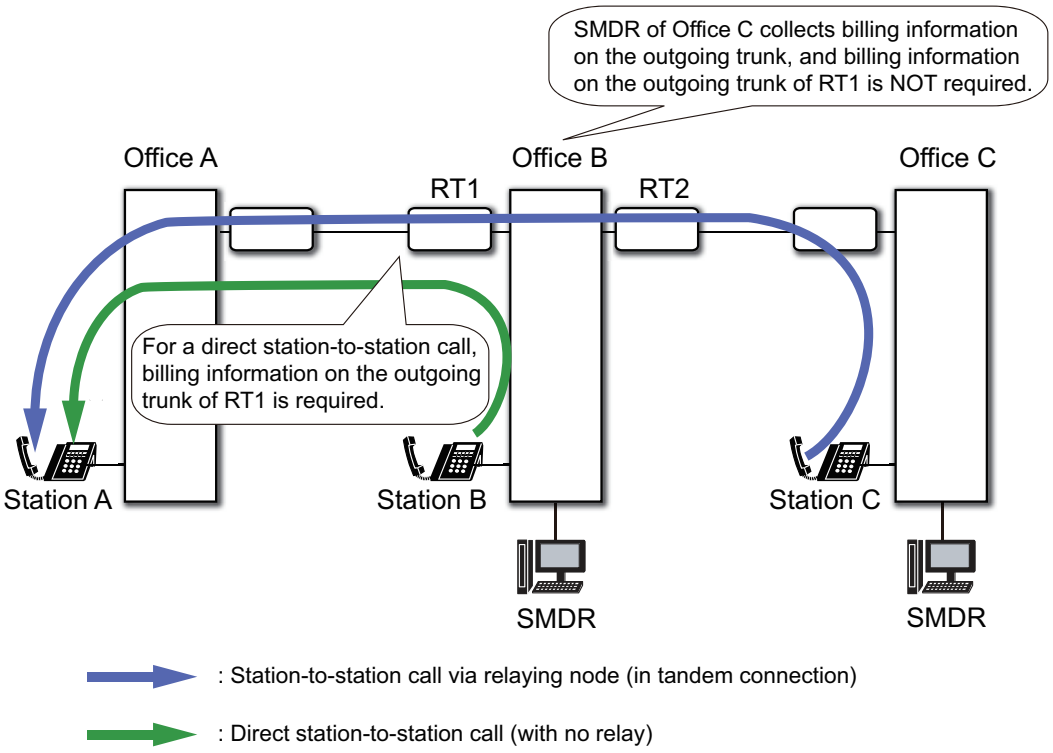
Note: When assigning Logical Route, use ARTDN command.

Note: To change the data for existing route (RT), initialize or reboot the trunk device such as circuit card or MG.

CDN 56: SMDR3. Detailed billing for outgoing in tandem connection.
0/1 = Required/Not required.

Note: Make this setting for the incoming route (ICRT) at the relaying node. For details, see <Case Where Detailed Billing for Outgoing in Tandem Connection is Not Required (CDN 56 = 1)>.

<Case Where Detailed Billing for Outgoing in Tandem Connection is Not Required (CDN 56 = 1)>



In the network shown above, Office B functions as a tandem switch. Office A has no SMDR.

When Station C places a call to Station A, the SMDR of Office C collects billing information on the outgoing trunk. Therefore, billing information on the outgoing trunk of RT1 is not required.

When Station B places a call to Station A, the billing information on the outgoing trunk of RT1 is required since Office A has no SMDR.

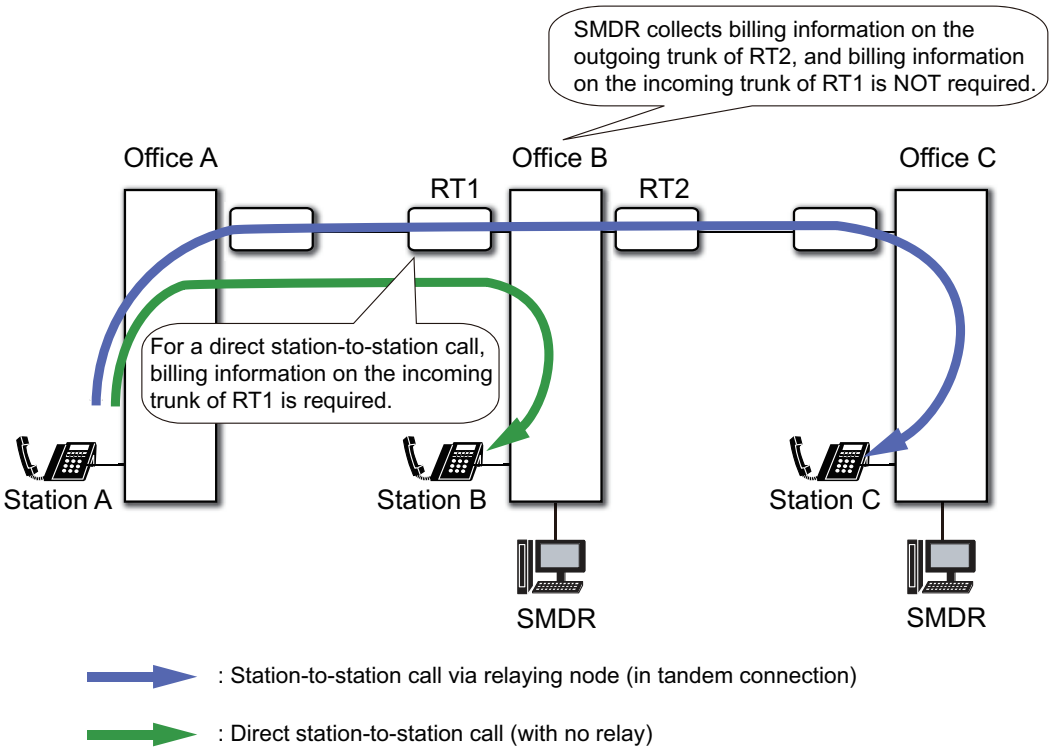
By assigning the Route Class data (ARTD, CDN56: SMDR3 = 1) to the incoming trunk of RT2, you can program the system so that the SMDR of Office B gathers billing information on RT1 only when the outgoing call is originated from a station in Office B.

CDN 69: SMDR4. Detailed billing for incoming in tandem connection.

0/1 = Required/Not required.

Note: Make this setting for the outgoing route (OGRT) at the relaying node. For details, see <Case Where Detailed Billing for Incoming in Tandem Connection is Not Required (CDN 69 = 1)>.

<Case Where Detailed Billing for Incoming in Tandem Connection is Not Required (CDN 69 = 1)>



In the network shown above, Office B functions as a tandem switch. Office A has no SMDR.

When Station A places a call to Station C, the SMDR of Office B collects billing information on the outgoing trunk of RT2. Therefore, billing information on the incoming trunk of RT1 is not required.

When Station A places a call to Station B, the billing information on the incoming trunk of RT1 is required since Office A has no SMDR.

By assigning the Route Class data (ARTD, CDN69: SMDR4 = 1) to the outgoing trunk of RT2, you can program the system so that the SMDR of Office B gathers billing information on RT1 only when the incoming call terminates to a station in Office B.

STEP 7: **ARTI** - To enable the output of Trunk Arrival/Sending Time and Abandoned Call information (for Incoming/Outgoing Trunk Call), assign CDN 90 (SMDS5) = 1.

Note: When assigning Logical Route, use ARTIN command.

Note: To change the data for existing route (RT), initialize or reboot the trunk device such as circuit card or MG.

Note: Also, to enable this feature, it is necessary to enable SMDR billing for incoming/outgoing call by the data setting. The following tables and figure show the office data setting pattern for Incoming/Outgoing Call Billing:

Data Setting of Billing for Incoming Call, Abandoned Incoming Call and Trunk Arrival Time

ARTD/ARTDN CDN 16 (SMDR2) Bit 0	ARTI/ARTIN CDN 90 (SMDS5)	ASFC SFI 14 (System Message Detail Recording (SMDR) - Trunk Basis)	
		1 (Allowed)	0 (Restricted)
1	1	Billing for Incoming Call Abandoned Incoming Call Trunk Arrival Time	Not Output
1	0	Billing for Incoming Call	Not Output
0	1	Not Output	Not Output
0	0	Not Output	Not Output

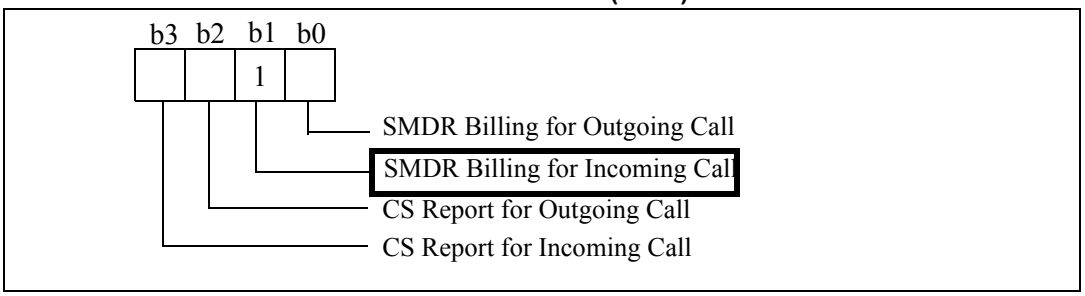
Note: For countries other than EMEA:

- This feature is available since FP95-113 V3.
- This feature cannot be used in conjunction with CS Report.

Note: For EMEA market:

- This feature is available since FP95-111 V1.
- This feature can be used in conjunction with CS Report until FP95-112 V2. (From FP95-113 V3 they cannot be used at the same time.)
  Until FP95-112 V2, when CS Report is in service, the setting value of CSMDS (CDN70) is enabled. In that case, note the following:
  - *When no ARTI/ARTIN data is set or CSMDS (CDN70) bits 0-3= 0 or CSMDS (CDN70) bit 1= 1, Billing for Incoming Call is enabled.*
  - *To disable Billing for Incoming Call, set CSMDS (CDN70) bit 1 to "0" and bit 0 or bits 2-3 to "1".*

CDN: 70 CSMDS (ARTI)



Data Setting of Billing for Outgoing Call, Abandoned Outgoing Call and Trunk Sending Time

ARTD/ARTDN CDN 10 (SMDR1)	ARTI/ARTIN CDN 90 (SMDS5)	ASFC SFI 14 (System Message Detail Recording (SMDR) - Trunk Basis)	
		1 (Allowed)	0 (Restricted)
1	1	Billing for Outgoing Call Abandoned Outgoing Call Trunk Sending Time	Not Output
1	0	Billing for Outgoing Call	Not Output
0	1	Not Output	Not Output
0	0	Not Output	Not Output

Note: This feature is available since FP95-113 V3.

Note: This feature cannot be used in conjunction with CS Report.

STEP 8: **ASFC** - Assign data "1" to SFI 14 of a Service Feature Restriction Class to all traffics (Trunk Basis). Assign data "1" to SFI 58 of a Service Feature Restriction Class to all traffics of which is to be registered on SMDR (STN to STN Basis).

Note: The setting enables all traffics to be billing targets. If an unnecessary SMDR data is output, please discard it at the SMDR terminal.

STEP 9: **ARNP** - For the routes flagged for SMDR in ARTD, list the Route Number and the Outgoing Trunk Access Code or Outgoing Trunk Advance Access Code here. If LCR(S) is used, list the route numbers and the LCR(S) Access Code.

STEP 10: **AMND/AMNDL** - Assign the number of digits the Register is to receive for every dialing Code. List the Toll Number ID Data Code for the system to distinguish the Toll Dialing patterns for SMDR registration. The dialing Code listings are identical for LCR(S). Refer to the AMND descriptions for LCR-3/6-DIGIT [L-5] and TOLL RESTRICTION-3/6-DIGIT [T-7].

Note: Toll/Local data is used when ARTD CDN16 (SMDR2) Bit 1 = 1, where SMDR is output for an Outgoing Toll Calls only.

FCCS Programming

This data setting describes only the additional or replacement commands used to activate FCCS Services. When used as an FCCS feature, these commands replace the non-network commands (for example, AAED is replaced by AAEDN). Note that the data setting for FCCS Service varies depending on the node.

Note: When using ASYDL or ASYDN to program an FCCS feature, the basic programming of ASYD is required in addition to ASYDL or ASYDN.

STEP 1: **ALRTN** - Assign this command to connect the Logical Route to the Physical Route.

STEP 2: **ATDF** - Assign time difference data if the time between the nodes and the UCT (Universal Coordinated Time) may not be identical on a FCCS Network. The assigned data (time difference margin [Plus/Minus, Hour, Minute] between each FPC and the UCT) is to be written in Network Data Memory (NDM) of the Network Control Node (NCN).

Programming for Centralized Billing - FCCS (Polling Method)

STEP 1: **ASYDL** - System Data 1, Index 529, Bits 0 and 1. Set a parity check type (Odd/Even/Non)

Bit 1	Bit 0	Parity Check Type
0	0	Odd Parity (Default)
0	1	Odd Parity
1	0	Even Parity
1	1	Non Parity

System Data 1, Index 576,
Bit 0 = 1 (FCCS Centralized Billing Service (Polling Method) is in service.
Bit 2 = 0/1 (SMDR Interface Type: RS232C/LAN). **Note 39**

Note 39: Set an appropriate interface type (RS232C/LAN) for SMDR equipment that is installed at Center Node only. This bit is not required for Local Node when no SMDR equipment is installed at the Local Node.

System Data 1, Index 577, Bits 0 - 7. Set FPC as follows.

Node	Data for Index 577
Center Node	00 [Hex]
Local Node	Assign FPC of Center Node in hexadecimal (01-FD [Hex]).

System Data 1, Index 578, Bit 0 and Bit 4. Assign data for SMDR A
Bit 0 = 0/1 (ICS format/IMX format)
Bit 4 = 0/1 (SMDR A with LAN interface is not used/used)

System Data 1, Index 579, Bit 0 and Bit 4. Assign data for SMDR B
Bit 0 = 0/1 (ICS format/IMX format)
Bit 4 = 0/1 (SMDR B with LAN interface is not used/used)

System Data 1, Index 580, Bit 0 and Bit 4. Assign data for SMDR C
Bit 0 = 0/1 (ICS format/IMX format)
Bit 4 = 0/1 (SMDR C with LAN interface is not used/used)

System Data 1, Index 581, Bit 0 and Bit 4. Assign data for SMDR D
Bit 0 = 0/1 (ICS format/IMX format)
Bit 4 = 0/1 (SMDR D with LAN interface is not used/used)

System Data 1, Index 582, Bits 0-3. Assign the number of nodes to be polled in a polling. When default value is used, polling is performed to one node at each polling.

Bit 3	Bit 2	Bit 1	Bit 0	Number of Nodes to be polled
0	0	0	0	1 node
0	0	0	1	1 node
0	0	1	0	2 nodes
0	0	1	1	3 nodes
0	1	0	0	4 nodes
0	1	0	1	5 nodes
0	1	1	0	6 nodes
0	1	1	1	7 nodes
1	0	0	0	8 nodes
|	|	|	|	
1	1	1	1	

System Data 1, Index 583, Bits 0-2. Assign the cycle to send billing data to the node which accommodates the calling station. The billing data is sent from node which accommodates the originating trunk (Outgoing Call SMDR) or the called station (Station-to-Station SMDR).

Bit 2	Bit 1	Bit 0	Cycle to Send
0	0	0	8 seconds
0	0	1	2 seconds
0	1	0	4 seconds
0	1	1	6 seconds
1	0	0	8 seconds
1	0	1	10 seconds
1	1	0	12 seconds
1	1	1	14 seconds

Note: By default, the data is transmitted every 8 seconds.

System Data 1, Index 585, Bits 0-7. Modify the polling cycle between nodes at the Center Node if required. The default cycle (2 seconds) can be modified in one of the following cases to avoid buffer overflow at Local Node:

- FCCS network consists of many Local Nodes
- Local Node is specified as a Main Node of Centralized Billing - CCIS

Bit 2	Bit 1	Bit 0	Cycle to Send
0	0	0	8 seconds
0	0	1	2 seconds
0	1	0	4 seconds
0	1	1	6 seconds
1	0	0	8 seconds
1	0	1	10 seconds
1	1	0	12 seconds
1	1	1	14 seconds

System Data 1, Index 586, Bits 0-7. Modify the upper limit of polling buffer usage rate (01% - 99%) if required. The default value is 50%. If the usage rate exceeds this value, polling will not be performed.

b7	b6	b5	b4	b3	b2	b1	b0
Tens				Unit			

Note: When the Centralized Billing Office cannot receive any billing information for 20 seconds, billing information created at local offices is not to be sent to the Centralized Billing Office. In this case, when there is no other destination for a local office to output the billing information than the Centralized Billing Office-CCIS, the information will be deleted. As long as this incapability of sending information persists, the transmission status is to be checked in order from the oldest.

Note: This setting applies to “No.7 Centralized Billing -CCIS”, “Centralized Billing -FCCS”, and “CS Report [for UMGi/RNIP system] [C-149]”.

System Data 1, Indexes 608-639. Assign FPCs of nodes that are polled by Center Node.

System Data 1, Index 641 Output Number and Route Number designation that is applied for SMDR - ICS format only.
Bit 0 = 0/1 (Station Number/Telephone Number)
Bit 4 = 0/1 (Physical Route Number/Logical Route Number)

Note: If the number of digits of Telephone Number is within 6 digits, the Telephone Number will be output when Bit 0 = 1.

STEP 2: **ASYDN** - System Data 1, Index 583 Bit 7. Specify node to which SMDR data should be stored.
0/1 = (Node accommodating the calling party/Node accommodating the outgoing trunk or the called party)

Note: Be sure to assign Bit 7= 1 when the SMDR terminal is not connected to the Local Node in order to collect the billing information quickly.

S-11 Station-to-Station Calling

General Description

This feature permits any Station user to directly dial another Station within the system without operator assistance.

Operating Procedure

To operate

1. Lift the handset. Receive Dial Tone.
2. Dial the desired Telephone Number.

Service Conditions

1. If the Called Station is off-hook, the originating Station receives a Busy Tone. Otherwise, the Called Station will ring and the caller will hear a Ringback Tone.
2. If the Called Station remains unanswered for longer than a preset interval, the call will not be transferred to the ATTENDANT CONSOLE [A-3], unless CALL FORWARDING-DON'T ANSWER [C-3] is provided.
3. If the Called Station is busy, the call will not be transferred to the Attendant Console, unless CALL FORWARDING-BUSY LINE [C-2] is provided.
4. Station-to-Station Calling between TENANTs [T-12], will not be restricted, unless programming for such restriction is entered in System Data.
5. The maximum number of digits per Telephone Number is 16.
6. The same Telephone Numbers cannot be duplicated in different TENANT SERVICE [T-12] Groups.
7. Calling Party Release/Called Party Release/Both Party Release/First Party Release can be selected on a system basis.
8. The following features, if activated, may result in the Calling Station user to be connected to a Station other than the dialed Station:

CALL FORWARDING - ALL CALLS/BUSY/DON'T ANSWER	[C-5, 2, 3]
CALL PICKUP	[C-7, 30]
STATION HUNTING	[S-7, S-8, S-9]
UCD	[U-1]

9. When HOTLINE [H-1] service is provided, the Calling Station user automatically connects to a preset number by going off-hook. The preset number may be another Station, an Attendant, or an Outside Party.
10. When the Calling Station user dials an unassigned Telephone Number, the call will be routed to Reorder Tone or Announcement (option).
11. An option per Station Line is available to eliminate the STATION 5dB PAD [S-62].

Interactions

1. The ATNR command is required for all Station-to-Station Calling.
2. The normal 5 dB PAD loss inserted for Station-to-Station Calling may be eliminated on a per-Station basis. For these Stations, assign a Service Feature Restriction Class that allows SFI 50, Removal of 5dB pad for Off-Premises Extensions (OPX).

Programming

STEP 1: **ASYD** - System Data 1, Index 4, Bits 2 and 3. Assign data "10" for First-Party Release.

STEP 2: **ATNR** - Assign for Tenant Restriction Index (TRI) 0. This Index allows for Station-to-Station Calling between the same Tenant and between different Tenants.

DAY/NIGHT: Day/Night Mode
- D: Day Mode
- N: Night Mode

OGTN: Originating Tenant

TMTN: Terminating Tenant

TRI: Tenant Restriction Index (0-15)

RES: Restriction Data
- 0: Restricted
- 1: Allowed

TRI: Tenant Restriction Index
- 0: Allows Intra and Inter-Tenant Station Calls. Allow call within Tenant.
- 1: Allows CALL FORWARDING-ALL CALLS, -DON'T ANSWER, and -BUSY LINE on an Intra and Inter-Tenant basis. Allow this feature within Self-Tenant.
- 2: Allows or restricts Incoming Calls via the Night Attendant Console from Tenant to Tenant. This allows the Tenant, as assigned to a Trunk, to be terminated to a Tenant, as assigned to a Station.
- 3: Allows the CALL FORWARDING-ALL CALLS, DON'T ANSWER, and BUSY LINE on an Intra and Inter-Tenant basis, as when set by the Attendant Console.
- 4: Allows or restricts Incoming Calls to the Attendant Console on a per Tenant basis. Thus, an Attendant may handle calls only for a specific Tenant.
- 5: Day/Night Mode Change Restriction.
- 6: Not used
- 7: Not used

S-12 Station-to-Station Calling - Operator Assistance

General Description

This feature allows a Station user to call another Station within the System, with the assistance of the ATTENDANT CONSOLE [A-3] operator.

Operating Procedure

To operate

1. Lift the handset. Receive Dial Tone.
2. Dial the operator Access Code (normally “0”).
3. Relay the desired Telephone Number to the Attendant.
4. The Attendant dials the Telephone Number. The Called Party Station rings and the Attendant presses the RELEASE Key.
5. Calling and Called Parties are connected when the Called Station answers the call.

Service Conditions

1. The Attendant may drop out of the Connection before or after the Called Party has answered.
2. If the Called Station does not answer within 30 seconds, AUTOMATIC RECALL [A-8] will be activated. The Recall Timer length is set in System Data.
3. Using this feature, the Attendant may connect the Calling Party to a fully-restricted Station.
4. STATION-TO-STATION CALLING-OPERATOR ASSISTANCE between TENANTs [T-2] will not be restricted, unless such a restriction is programmed in System Data.
5. The Fusion Point Code of the Node which accommodates the Called Attendant Console must be assigned after System Data assignment.

Interactions

1. The ATNR command is required for all Station-to-Station Calling.
2. The normal 5 dB PAD loss inserted for Station-to-Station Calling may be eliminated on a per-Station basis. For these Stations, assign a Service Feature Restriction Class that allows SFI 50 [Off-Premise Extension (No Howler/No Pad Control)], Removal of 5dB pad for Off-Premises Extensions (OPX).
3. The Attendant Console must first be provided.
4. ATTENDANT CAMP-ON WITH TONE INDICATION [A-1]: The Attendant cannot Camp-On a Station-to-Station Call.

Programming

STEP 1: **ASYD** - System Data 1, Index 4, Bits 2 and 3. Assign data "10" for First-Party Release.

STEP 2: **ATNR** - Assign for Tenant Restriction Index (TRI) 0. This Index allows for STATION-TO-STATION CALLING between the same Tenant and between different Tenants. The Tenant, in which the Attendant resides, must also be allowed STATION-TO-STATION CALLING to assist in placing the call.

D/N: Day/Night Mode
- Day Mode
- Night Mode

OGTN: Originating Tenant

TMTN: Terminating Tenant

TRI: Tenant Restriction Index (0-15)

RES: Restriction Data

0: Restricted

1: Allowed

S-13 Step Call

General Description

This feature allows the Attendant or Station user, upon calling a busy Station, to call an Idle Station by dialing an additional digit. This feature will operate only if the number of the Idle Station is identical to that of the busy Station in all respects, except the last digit.

Operating Procedure

To operate

1. The dialed Station (220) is busy.
2. Dial "5" (or any digit between 0 and 9).
3. If Station 225 is idle, the call will be connected there.

Service Conditions

1. This feature can be provided for all Stations and the Attendant Console(s).
2. STEP CALLING can be in any order, 0-9 or 9-0.
3. If the second selected Station is also busy, STEP CALLING can continue until a Station rings.
4. If a Station is set for any type of CALL FORWARDING (ALL CALLS, BUSY LINE NO ANSWER [C-2, 3, 5]), the Call Forwarding will take effect.
5. CALL FORWARDING - ALL CALLS/BUSY LINE [C-2, 5]: The activation of a Call Forwarding feature by the Called Station can cause a STEP CALL to be completed to a Station other than the dialed Station.
6. CONSULTATION HOLD - ALL CALLS [C-17]: STEP CALL can be activated when a Busy Tone is returned on a Consultation Hold attempt.
7. The STEP CALL feature is not operational for Trunk Incoming Calls to the SV9500 Stations.

   Note: STM:2 and STS:6 should be assigned as 1 by ASTD command, to restrict Step Call on DID.

8. A STEP CALL to a Station in LINE LOCKOUT [L-3] or maintenance busy state will result in a Reorder Tone.
9. This feature can be activated between an IP terminal and a SIP terminal (a SIP Multiple Line terminal, a Soft Phone (SIP), a WLAN Handset (MH Series) and a SIP Handler Controlled SIP terminal), under the following conditions:
   - From IP terminal to SIP terminal: Available
   - From SIP terminal to IP terminal: Not available
   - From SIP terminal to SIP terminal: Not available

Interactions

1. A STEP CALL can search through an entire group to find an Idle Station.
2. CALL FORWARDING [C-2, 3, 5]: STEP CALL will respond according to the numbering scheme of the dialed Station. Station A (7526) Call Forwards his phone to Station B (7128). Station B is busy. The Calling Station dials 7526 and receives Busy Tone. The Calling Station then dials 2. The call terminates at 7522.

3. CONSULTATION HOLD-ALL CALLS [C-17]: STEP CALL can be used when encountering a busy Station via CONSULTATION HOLD.
4. LINE LOCKOUT [L-3]: A STEP CALL to a Station in LINE LOCKOUT [L-3] results in Reorder Tone. Another digit can be dialed to reach an Idle Station.
5. When STEP CALL [S-13] is used after LAST NUMBER CALL [L-6, 6D] service with the Access Code, the destination of STEP CALL is changed depending on the Access Code of the LAST NUMBER CALLED [L-6, 6D] service because the System recognizes the Access Code of LAST NUMBER CALLED as the Telephone Number. In other words, the target number for STEP CALL is not the Telephone Number dialed before LAST NUMBER CALLED, but the number dialed as the LAST NUMBER CALLED Access Code. For example:
   a.) Station A places a call to Station B (4000) by dialing "300", which is the Access Code of LAST NUMBER CALLED.
   b.) Station A hears Busy Tone since Station B is busy.
   c.) Station A dials "1" using STEP CALL.
   d.) The System recognizes that "301" is dialed.
6. System data that enables to receive digits when the destination is in busy (ASYDL, SYS1, Index1382 Bit 0 = 0) is required to use this feature through SIP Handler Controlled SIP terminals.
7. A Step Call through a TIE Line is available, except when a TIE Line call to a PS/SP Controlled SIP terminal is forwarded to a busy station via CALL FORWARDING PS INCOMING CALL INCOMPLETE [C-155].

Programming

STEP 1: **ASYD** - System Data 1, Index 59, Bit 2. DID Busy Condition 0/1: No Tone/Busy Tone (Assign 1)

STEP 2: **ASTD** - STM 2, STS 6, ST 1 to restrict DID step call.

S-15 Serial Call

General Description

This feature allows the Attendant to arrange for a recall from a Station before releasing a Central Office Call to that Station. When the Station subsequently disconnects from the call, the Central Office party automatically rings back to the Attendant.

Operating Procedure

1. Answer the Incoming Central Office Call.
2. Extend the call to the desired Station.
3. Press the SERIAL CALL (SC) Key. The HOLD Lamp corresponding to the Loop will light.
4. The Called Station and Incoming Station are connected.
5. The Called Station hangs up. The HOLD Lamp (and the ATND Lamp) on the Attendant Console flash at 60 IPM. If the Attendant position is available, an audible indication will be provided.
6. Press the associated Loop Key to return to the Incoming Calling Party.

Service Conditions

1. This feature is effective only for Incoming Calls from external Trunks.
2. If the Attendant enters a Loop that is already set up for SERIAL CALL, the service is automatically canceled.
3. SERIAL CALL (SC) is not provided for Station-to-Station Calling.
4. SERIAL CALL can be provided on a per Tenant basis.
5. SERIAL CALLING is allowed when a Station is involved in an ATTENDANT CONTROLLED CONFERENCE [A-2].
6. STATION MESSAGE DETAIL RECORDING (SMDR) [S-10]: The duration of the total call is recorded on the last Station Call.
7. When the ATTENDANT LOCKOUT [A-5] feature is provided, the Attendant cannot reenter the Connection without being recalled.
8. This feature is not applicable for Tandem Connections.
9. In the Loop Release system, once this service is set, the Attendant position enters the idle state, and the established call is released from the Loop, enabling all hooking services at the extension side. ASYD, System Data 1, Index 186, Bit 4 = 1.
10. The Attendant sets this service by pressing the SC Key when the extension user answers.
11. It is not possible to release this service once it is set.
12. If the operator presses the SC Key before the called extension answers (Non-Delay mode service setting), the external Calling Party receives a Ringback Tone, and the Attendant position enters the idle state.
13. In the Non-Delay mode service, setting the extension Loop release does not occur until the Called Extension answers, even though the Loop Release system is specified by the System Data setting (System Data 2 Index 2 Bit 4 = 1). The operator is allowed to reenter the Connection before the extension answers, though the SERIAL CALL service remains set.
14. CALL PICKUP GROUP [C-7] feature can be used for answering a call handled by the operator in the Non-Delay service setting mode.
15. CALL FORWARDING-DON'T ANSWER [C-3] feature cannot be used if SERIAL CALL is set.

16. SERIAL CALL termination is indicated to the Attendant by means of the HOLD or SC/RC lamp and buzzer indication. Ringback Tone is sent to the external Calling Party up to the time the Attendant answers the indicated Serial Call.
17. SC termination indication goes off if the external Calling Party is released before the Attendant answers the indicated Serial Call (in case of TIE Line Connection).
18. Either the SC/RC Key or the ANSWER Key is used to answer a Serial Call if ASYD System Data 1, Index 186, Bit 4 is "1".
19. SERIAL CALL service is automatically released when the operator answers the indicated Serial Call.
20. The Day-To-Night mode change does not interrupt the indication of Serial Call Termination.
21. Attendant supervision of a SERIAL CALL on the ATTENDANT CONSOLE [A-3] is provided via the Loop status lamps. After an Incoming Trunk Call is extended to a Station and the Called Station goes on-hook, the Attendant is alerted by the flashing HOLD indicator lamp.
22. The following features are denied toward a line or Trunk involved in a SERIAL CALL if ASYD, System Data 1, Index 186, Bit 4 = 0:

ATTENDANT CAMP-ON WITH TONE INDICATION	[A-1]
ATTENDANT OVERRIDE	[A-7]
BUSY VERIFICATION	[B-3]
BOSS - SECRETARY OVERRIDE -D	[B-6D]
CALL HOLD	[C-6, 6D]
CALL WAITING - ORIGINATING	[C-31]
CALL WAITING - TERMINATING	[C-12]
EXECUTIVE RIGHT-OF-WAY	[E-1, 1D]

23. When CALL WAITING - TERMINATING [C-12], CALL WAITING - ORIGINATING [C-31], Attendant Camp-On, or Attendant Camp-On - CCIS is set to the Station already engaged in a SERIAL CALL, the following conditions occur:
    a.) If the Station answers the Overriding Call, the party formerly engaged in the SERIAL CALL is placed on hold. When the Station releases and does not return to the Held Call, the Attendant is recalled.
    b.) If the Station does not answer the Overriding Call, Call Waiting or Attendant Camp-On is activated.
24. When a call is directed to Attendant Console from CCIS Trunk:
    If the Incoming Trunk is released due to the reason such as call transfer to the Calling Party, etc., this feature is canceled.

Interactions

1. Features requiring the switch hook to be flashed are available. These include CALL TRANSFER-ALL CALLS [C-11], CALL PICKUP [C-7, 30], CONSULTATION HOLD [C-17], and CALL HOLD [C-6] if ASYD System Data 1, Index 186, Bit 4 is "1".
2. EXECUTIVE RIGHT-OF-WAY [E-1], BUSY VERIFICATION [B-3], and ATTENDANT OVERRIDE [A-7] are available towards a Station/Trunk involved in a Serial Call, if ASYD System Data 1, Index 186, Bit 4 is "1".
3. If a Serial Call is transferred to a second Station, when the second Station hangs up, the Serial Call will return to the Attendant Console if ASYD, System Data 1, Index 186, Bit 4 is "1".

Note: The SC (Serial Call) Key is the Key right of the DEST Key. The SC/RC (Serial Call Recall) Key replaces the CCSA Key if ASYD System Data 1, Index 186, Bit 4 is "1".

Programming

ASYD - System Data 1, Index 186, Bit 4. Serial Call Loop Release 0/1: OUT/In Service When data "1" is assigned, CCSA Calls will not ring the ATT. In this case, change ARTD CDN 6:TCL from CCSA to DDD Line, TIE Line, etc.

System Data 2, Index 7, Bit 2. Is SERIAL CALL to be enabled on a per Tenant basis? 0/1: No/Yes.

System Data 1, Index 43, Bit 4. Is Attendant able to reenter Loop of Serial Call? 0/1: No/Yes.

S-19 Single-Digit Feature Code

General Description

This feature allows the system to be programmed so that features used most frequently may be accessed by dialing a single digit.

Operating Procedure

To access a desired feature, refer to the Operating Procedure section of that particular Feature Description.

Service Conditions

1. When accessing SINGLE-DIGIT FEATURE CODEs, consideration must be given to the overall Numbering Plan of the particular system, unless timing start has been provided.
2. Timing start is not available with this feature.

Interactions

1. For each feature with one-digit access, an entire Number Level is eliminated. Do not plan to have more than one or two such features in a system.
2. The corresponding feature Cancel Code does not have to be a single digit.
3. TRUNK ANSWER ANY STATION [T-8], SPEED CALLING-SYSTEM [S-3], and DIAL ACCESS TO THE ATTENDANT [D-2] are three applications for SINGLE-DIGIT FEATURE CODE.
4. The digit "*" or "#" can be used as the first digit of a Trunk Access Code and can be used as a Feature Access Code.

Programming

STEP 1: **ANPD** - Reserve a Number Level for Feature Access. Select Normal Service (N), Hooking Service (H) or Busy Service (B) from the combo box for Connection Indexes. (applicable to any feature assigned to this Level). Assign NND in accordance with a predetermined Numbering Plan. BLF is not in service.

STEP 2: **ASPA** - Assign the Tenant and Access Code for the desired feature. Assign Connection Indexes, CI = "N: Normal", "H: Hooking" and/or "B: Busy". Assign the SRV = "SSC: Service Code" or "SSCA: Service Code Appendix", and Service Feature Index SID or SIDA for the required feature.

STEP 3: **ASFC** - Assign to the Stations to receive this feature an SFC that allows the corresponding SFI.

S-21 Speed Calling - Station

General Description

This feature allows a Station user to dial certain frequently-called exchange network numbers using fewer digits (Abbreviated Call Codes) than normally required. Using this feature, a Station user can establish his own abbreviated codes.

Operating Procedure

To operate

1. Lift handset or press SPEAKER Key. Receive Dial Tone.
2. Dial SPEED CALLING. Enter Access Code (one to three digits). **Note 1**
3. Dial the Abbreviated Call Code (one to three digits). Call is completed.

To set, using the Primary Station only

1. Lift handset. Receive Dial Tone.
2. Dial SPEED CALLING. Enter Feature Access Code (one or two digits). **Note 1**
3. Dial the Abbreviated Call Code (one to three digits). At this point no tone is returned.
4. Dial the destination telephone number. A brief pause will occur before receiving Service Set Tone.

Note 1: If one digit Code is used, the format must be 00 to 99.

To operate from a Desktop terminal

1. Press the REDIAL Key. Receive Dial Tone. The LCD displays:



2. Press the desired Call Code. The call is completed.

Service Conditions

1. SPEED CALLING-STATION codes are programmed into RAM memory per Local Processor.
2. Each Abbreviated Call Code can store a maximum of 24 digits, including Access Codes and pauses.
3. A maximum of 100 Stations per Module Group can be equipped with SPEED CALLING-STATION.
4. **<Speed Calling - Station>**
   100 Memory Blocks per Module Group are assigned for Speed Calling. A maximum of 10 numbers can be stored in each Memory Block. This allows for a total of 1,000 numbers per Module Group.
   <Expanded Speed Calling - Station>
   A maximum of 1400 Memory Blocks can be assigned per LP. A maximum of 10 numbers can be stored in each Memory Block. This allows for a total of 14,000 numbers per LP.
5. Each Station can be assigned 10 memory blocks for Speed Calling; therefore, it can be programmed for 10 ~ 100 numbers per MG. In Expanded Speed Calling, each station can be programmed for 140 ~1400 numbers per LP.
6. TOLL DENIAL/TOLL DIVERSION [T-6] and TOLL RESTRICTION-3/6-DIGIT [T-7] may or may not be applied to SPEED CALLING, as desired.
7. LEAST-COST ROUTING-3/6-DIGIT [L-5] may be provided via SPEED CALLING-STATION access.

8. Each user of an individual Station equipped with this feature can program a personal Speed Calling Numbers, via the Single Line Telephone.
9. If SPEED CALLING-STATION is assigned to a Local Processor, SPEED CALLING-GROUP [S-23] shares the same memory tables. Therefore, careful consideration should be given when programming.
10. When outgoing dialing with this feature, Speed Calling Number can be displayed on the LCD instead of Speed Calling-System Feature Access Code and the Abbreviated Call Code by ASYDL command (ASYDL SYS 1 Index 673 bit 6=1). The following conditions are about this feature.

    Note: This feature is available only with outgoing call (LCR/LCRS/OGC/OGCA).

    a.) When you register the Abbreviated Call Codes as a Forwarding Destination with Speed Calling Number Display on Dterm service, only SPEED CALLING - SYSTEM Access Code and Abbreviated Code are displayed on LCD. Speed Calling Number is not displayed.
    b.) Although Speed Calling Number Display on Dterm service is available, SPEED CALLING-SYSTEM Access Code and Abbreviated Code are displayed on LCD as a dialed numbers memory, not Speed Calling Number.
    c.) When Speed Calling Number Display on Dterm service is available, the number is displayed within 16 digits. More than 16 digits number is displayed as only the first 16 digits.
    d.) This feature is available only while calling the party. If the LCD is re-displayed with other service such as Name Display-OAI-CCIS or the called party is being busy, this feature is not available.
    e.) When you use this feature, register the Speed Calling Number correctly in ASPD command. If the last digit part of number is not registered, Speed Calling Number is not displayed.
    (When calling "8-26-70000" with the use of this service, you may register "826" as a Speed Calling Number. In this case, press the Access Code and Abbreviated Code, and dial "70000". At the time you call "7", Speed Calling Number display becomes invalid.)
    f.) While calling with this feature, if you press the abbreviated call codes, Speed Calling Number is not displayed.
11. SPEED CALLING-STATION data will be retained in memory, even if the system is reinitialized or in the event of a power failure, by means of a hard disk for a system.
12. A pause may be programmed, using the "*****" Key of a PB pad. This will take one of the 24 digits available. Rotary type phones cannot store a pause. A pause lasts about 3 seconds. A pause cannot be added to ISDN, or CCIS Calls.

    Note: A pause to MG-COT Calls can be added. In this case, a pause lasts about three seconds.

    Note: Registered pauses will not be activated when originating a call to ISDN Lines or to Analog Lines via MG-COT.

13. It is not necessary to erase a stored number in order to store another number. The second number replaces the first.
14. When a caller places a call via Outgoing Call Alternative (OGCA), and when the first outgoing route and the Station setting SPEED CALLING - STATION are restricted, this service cannot be set. In this case, the service can be set via the following procedure:
    a.) assign a dummy route.
    b.) clear the restriction between the dummy route and all the Stations.
    c.) assign the dummy route as the first route of OGCA.

15. When this feature is activated by pressing the REDIAL Key, only the Abbreviated Call Code is displayed on the LCD of the Dterm. Expanded Speed Calling - Station cannot be activated by the REDIAL Key.
16. When this feature is activated by dialing the Speed Calling Feature Access Code + the Abbreviated Call Code, only the Speed Calling Feature Access Code and the Abbreviated Call Code are displayed on the LCD of the Dterm.
17. After switch hook flash, station user cannot originate a call by Abbreviated Call Code. However, this is available from attendant console.
18. Speed Calling - Station can be activated from ISDN terminals. However, Expanded Speed Calling - Station cannot be activated from ISDN terminals.

Interactions

1. Blocks are not assigned sequentially (1-100), but are assigned as a quantity of blocks in the AISA command. These quantities will never exceed the value of 10.
2. If one block is assigned, the Abbreviated Call Code will be digits 0-9. If two blocks are assigned, the Abbreviated Call Code will be digits 00 - 19. If three blocks are assigned, the Abbreviated Call Code will be digits 00 - 29, and so on, to a maximum of 10 blocks with Abbreviated Call Codes 00 - 99.
3. Only the Primary Station will be allowed to use the SPEED CALLING-ASSIGN Access Code. When assigning the abbreviated SPEED CALLING Access Code and number, a delay of six to eight seconds will occur until Service Set Tone is returned.
4. Only those Stations assigned as Secondary Stations are considered to be members of the SPEED CALLING-GROUP.
5. Stations of a Speed Calling - Group (Individual) must be in the same Module Group (MG). Error 0238, “MG number of MY Line/Primary Station mismatch” is displayed.
   Stations of an Expanded Speed Calling - Group (Individual) must be in the same LP.
6. SPEED CALLING-INDIVIDUAL requires the assignment of senders, route 905. See UNIVERSAL SENDER [U-3].
7. OUTGOING TRUNK QUEUING [O-2]. This feature may be used with SPEED CALLING-INDIVIDUAL (GROUP) if all Trunks are busy.
8. ROUTE ADVANCE [R-6] and LEAST-COST ROUTING-3/6 DIGIT [L-5] may be used with this feature. The ASPA Access Code to either ROUTE ADVANCE [R-6] or LEAST-COST ROUTING-3/6 DIGIT [L-5] will be stored in the memory of the SPEED CALLING-GROUP number.
9. To set/modify this feature setting from a SIP Handler Controlled SIP terminal, dial all required digits: SPEED CALLING Setting Access Code, Abbreviated Call Code, and Call Origination Access Code with the destination station number.
   To place a call from the SIP Handler Controlled SIP terminal, dial the entire Access Code and Abbreviated Call Code.

Programming

STEP 1: **ASYD** - System Data 2, Index 1, Bit 3. Does TOLL RESTRICTION apply to SPEED CALLING-STATION? 0/1: Yes/No. (If yes, refer to TOLL RESTRICTION-3/6 DIGIT [T-10]).

STEP 2: **ASYDL** - System Data 1, Index 673, bit 6. Speed Calling Number Display Service on Dterm (Outgoing Call only) 0/1=Out/In Service

STEP 3: **ANPD** - Reserve a Number Level for Feature Access. Select Normal Service (N), Hooking Service (H) or Busy Service (B) from the combo box for Connection Indexes. (applicable to any feature assigned to this Level). Assign NND in accordance with a predetermined Numbering Plan.

STEP 4: **ASPA** - Two Access Codes must be assigned for this feature.

For the individual Station Speed Calling Number assignment, assign the Tenant and Access Code (ACC). Assign for a Connection Index of "N: Normal". For SRV = "SSC: Service Code", assign SID 14.

For individual Station Speed Calling Access, assign the Tenant and Access Code (ACC). Assign for a Connection Index of "N: Normal". For SRV = "SSC: Service Code", assign SID 21.

For the Expanded Speed Calling Number assignment, assign the Tenant and Access Code (ACC). Assign for a Connection Index of "N: Normal". For SRV = "SSCA: Service Code Appendix", assign SIDA 134.

For the Expanded Speed Calling Access, assign the Tenant and Access Code (ACC). Assign for a Connection Index of "N: Normal". For SRV = "SSCA: Service Code Appendix", assign SIDA 135.

STEP 5: **ASFC** - For Stations allowed this feature, assign a Service Feature Restriction Class that allows SFI 19 (Speed Calling - Station/Group). If necessary, assign a Service Feature Restriction Class that allows SFI 197 (Expanded Speed Calling - Station/Group).

STEP 6: **AISA** - Program the Tenant, Station, type of Station, and the number of SPEED CALLING blocks to be assigned to the Primary Station.

TN: Tenant Number
STN: Station Number
MST/SLV: Primary/Secondary
1: MST (Master Station)
2: SLV (Slave Station)
BLOCK: Number of Blocks
MST STN: When assigning SLV (Secondary) Station, assign the MST (Primary) Station Number.

STEP 7: **AIEA** - Program the Tenant, Station, Type of Station, and the number of Expanded Speed Calling blocks to be assigned to the MST Station.

STN: Station Number
STATION TYPE = 1: MST (Master Station)
BLOCK: Number of Blocks (1-10)
When assigning secondary station;
STN: Station Number
STATION TYPE = 2: SLV (Slave Station)
MST STN: MST (Primary) Station Number

STEP 8: **AKYD** - Assign Feature Key Speed Calling - Station/Group [26]. If necessary, assign Feature Key Expanded Speed Calling - Station/Group [146].

STEP 9: **ADSL** - Assign G_SPD Key; Assign SN = 1 (Originating; DT Connection) and FKY = 26 (Speed Calling Station/Group)/FKY = 146 (Expanded Speed Calling Station/Group).

This command is used for assigning a function to each Soft Key.

STEP 10: **AISD** - This command is used to assign, delete, and display the destination number of Speed Calling.

STEP 11: **AIED** - This command is used to assign, delete, and display the destination number of Expanded Speed Calling.

FCCS Programming

This data setting describes only the additional or replacement commands used to activate FCCS Services. When used as an FCCS feature, these commands replace the non-network commands (for example, AAED is replaced by AAEDN). Note that the data setting for FCCS Service varies depending on the Node.

Note: When using ASYDL or ASYDN to program an FCCS feature, the basic programming of ASYD is required in addition to ASYDL or ASYDN.

STEP 1: **ANPDL** - Reserve a Number Level for Feature Access. Select Normal Service (N), Hooking Service (H), or Busy Service (B) from the combo box for Connection Indexes. (applicable to any feature assigned to this Level). Assign NND in accordance with a predetermined Numbering Plan.

STEP 2: **ASPAL** - Two Access Codes must be assigned for this feature.

For the individual Station Speed Calling Number assignment, assign the Tenant and Access Code (ACC). Assign for a Connection Index of "N: Normal". For SRV = "SSC: Service Code", assign SID 14.

For individual Station Speed Calling Access, assign the Tenant and Access Code (ACC). Assign for a Connection Index of "N: Normal". For SRV = "SSC: Service Code", assign SID 21.

For the Expanded Speed Calling Number assignment, assign the Tenant and Access Code (ACC). Assign for a Connection Index of "N: Normal". For SRV = "SSCA: Service Code Appendix", assign SIDA 134.

For the Expanded Speed Calling Access, assign the Tenant and Access Code (ACC). Assign for a Connection Index of "N: Normal". For SRV = "SSCA: Service Code Appendix", assign SIDA 135.

S-23 Speed Calling - Group

General Description

This feature allows a Station user to share a set of common Speed Calling Numbers with other Station users in the group.

Operating Procedure

To record

1. Lift handset or press SPEAKER Key. Receive Dial Tone.
2. Dial the SPEED CALLING Access Code (one to three digits).
3. Dial the Abbreviated Call Code (one to three digits) and the called exchange network numbers. Receive Service Set Tone.

Note 1: If one digit code is used, the format must be 00 to 99.

To operate

1. Lift handset. Receive Dial Tone.
2. Dial the SPEED CALLING Feature Access Code.
3. Dial the Abbreviated Call Code (one or two digits).
4. Call is completed.

To operate from a Dterm

1. Press the REDIAL Key. Receive Dial Tone. The LCD displays:



2. Press the desired Call Code. The call is completed.

Service Conditions

1. **<Speed Calling - Group>**
   A maximum of 100 Memory Blocks per Module Group can be equipped for SPEED CALLING-GROUP. A maximum of ten Speed Calling Numbers per Memory Block can be assigned. This allows for a total of 1,000 numbers per Module Group.
   <Expanded Speed Calling - Group>
   A maximum of 1400 Memory Blocks can be assigned per LP. A maximum of 10 numbers can be stored in each Memory Block. This allows for a total of 14,000 numbers per LP.
2. A Primary Station will be assigned memory blocks, and associate Stations in that group are assigned as Secondary Stations. The Primary Station will have the capability to program 10 memory blocks commonly used as Speed Calling Numbers for that group; therefore, a group can consist of 100 Speed Calling Numbers maximum.
3. Each Abbreviated Call Code can store a maximum of 24 digits, including Access Codes and pauses.
4. A Station can be assigned as a Primary Station for only one group.
5. There is no limit to the number of Secondary Stations that may be assigned to one group.
6. Secondary Stations and their associated Primary Station must belong to the same Module Group.
7. A Secondary Station can only be assigned to one Primary Group.

8. TOLL DENIAL/TOLL DIVERSION [T-6] and TOLL RESTRICTION-3/6-DIGIT [T-7] may or may not be applied to Speed Calling, as desired.
9. LEAST-COST ROUTING-3/6-DIGIT [L-5] may be provided via Speed Calling access.
10. SPEED CALLING-GROUP and SPEED CALLING-STATION [S-21] use the same block of memory tables.
11. SPEED CALLING-GROUP data remains, even if the system is reinitialized or there is a power failure.
12. A pause may be programmed using the "*" button of a PB pad. This will take one of the 24 digits available. Rotary type phones cannot store a pause. A pause lasts about 3 seconds. A pause cannot be added to an ISDN or CCIS Call.

    Note: A pause to MG-COT Calls can be added. In this case, a pause lasts about three seconds.

    Note: Registered pauses will not be activated when originating a call to ISDN Lines or to Analog Lines via MG-COT.

13. Stations can be added or removed from a group only via PCPro.
14. When a caller places a call via Outgoing Call Alternative (OGCA), and when the first outgoing route and the Station setting SPEED CALLING - GROUP are restricted, this service cannot be set. In this case, the service can be set via the following procedure:
    a.) Assign a dummy route.
    b.) Clear the restriction between the dummy route and all the Stations.
    c.) Assign the dummy route as the first route of OGCA.
15. When this feature is activated by pressing the REDIAL Key, only the Abbreviated Call Code is displayed on the LCD of the Dterm. Expanded Speed Calling - Group cannot be activated by the REDIAL Key.
16. When this feature is activated by dialing the Speed Calling Feature Access Code + the Abbreviated Call Code, only the Speed Calling Feature Access Code and the Abbreviated Call Code are displayed on the LCD of the Dterm.
17. After switch hook flash, station user cannot originate a call by Abbreviated Call Code. However, this is available from attendant console.
18. Speed Calling - Group can be activated from ISDN terminals. However, Expanded Speed Calling - Group cannot be activated from ISDN terminals.

Interactions

1. Blocks are not assigned sequentially (1-100), but are assigned as a quantity of blocks in the AISA command. These quantities will never exceed the value of 10.
2. If one block is assigned, the Abbreviated Call Code will be digits 0-9. If two blocks are assigned, the Abbreviated Call Code will be digits 00-19. If three blocks are assigned, the Abbreviated Call Code will be digits 00-29, and so on, to a maximum of 10 blocks with Abbreviated Call Codes 00-99.
3. Only the Primary Station will be allowed access to the SPEED CALLING-ASSIGN Access Code.
4. Only those Stations assigned as Secondary Stations are considered to be members of the SPEED CALLING-GROUP.
5. Stations of a Speed Calling - Group must be in the same Module Group (MG).
   Stations of an Expanded Speed Calling - Group must be in the same LP.
6. SPEED CALLING-GROUP requires the assignment of senders, route 905. See UNIVERSAL SENDER [U-3].

7. OUTGOING TRUNK QUEUING [O-2]: This feature may be used with SPEED CALLING GROUP if all Trunks are busy.
8. ROUTE ADVANCE [R-6] and LEAST-COST ROUTING-3/6 DIGIT [L-5] may be used with this feature. The ASPA Access Code to either ROUTE ADVANCE [R-6] or LEAST-COST ROUTING-3/6 DIGIT [L-5] will be stored in the memory of the SPEED CALLING-GROUP number.

Programming

STEP 1: **ASYD** - System Data 2, Index 1, Bit 3. Does TOLL RESTRICTION apply to SPEED CALLING-STATION? 0/1: Yes/No. (If yes, refer to TOLL RESTRICTION-3/6 DIGIT [T-10]).

STEP 2: **ANPD** - Reserve a Number Level for Feature Access. Select Normal Service (N), Hooking Service (H) or Busy Service (B) from the combo box for Connection Indexes. (applicable to any feature assigned to this Level). Assign NND as "2". This should be in accordance with a predetermined Numbering Plan.

STEP 3: **ASPA** - Two Access Codes must be assigned for this feature. For the Speed Calling Number assignment, assign the Tenant and Access Code (ACC). Assign a Connection Index of "N: Normal". For SRV = "SSC: Service Code", assign SID 14. (Available only to the Primary Station.)

For the Speed Calling Access, assign the Tenant and Access Code (ACC). Assign a Connection Index of "N: Normal". For SRV = "SSC: Service Code", assign SID21.

For the Expanded Speed Calling Number assignment, assign the Tenant and Access Code (ACC). Assign for a Connection Index of "N: Normal". For SRV = "SSCA: Service Code Appendix", assign SIDA 134. (Available only to the Primary Station.)

For the Expanded Speed Calling Access, assign the Tenant and Access Code (ACC). Assign for a Connection Index of "N: Normal". For SRV = "SSCA: Service Code Appendix", assign SIDA 135.

STEP 4: **ASFC** - For Stations allowed this feature, assign a Service Feature Restriction Class that allows SFI 19 (Speed Calling - Station/Group). If necessary, assign a Service Feature Restriction Class that allows SFI 197 (Expanded Speed Calling - Station/Group).

STEP 5: **AISA** - Program the Tenant, Station, type of Station, and the number of Speed Calling blocks to be assigned to the Primary Station.

TN: Tenant Number
STN: Station Number
MST/SLV: Primary/Secondary
1: MST (Master Station)
2: SLV (Slave Station)
BLOCK: Number of Blocks
MST STN: When assigning SLV (Secondary) Station, assign the MST (Primary) Station Number.

STEP 6: **AIEA** - Program the Tenant, Station, Type of Station, and the number of Expanded Speed Calling blocks to be assigned to the MST Station.

STN: Station Number
STATION TYPE = 1: MST (Master Station)
BLOCK: Number of Blocks (1-10)
When assigning secondary station;
STN: Station Number
STATION TYPE = 2: SLV (Slave Station)
MST STN: MST (Primary) Station Number

STEP 7: **AKYD** - Assign Feature Key Speed Calling - Station/Group [26]. If necessary, assign Feature Key

Expanded Speed Calling - Station/Group [146].

STEP 8: **ADSL** - Assign G_SPD Key; Assign SN = 1 (Originating; DT Connection) and FKY = 26 (Speed Calling Station/Group)/FKY = 146 (Expanded Speed Calling Station/Group).

This command is used for assigning a function to each Soft Key.

STEP 9: **AISD** - This command is used to assign, delete, and display the destination number of Speed Calling.

STEP 10: **AIED** - This command is used to assign, delete, and display the destination number of Expanded Speed Calling.

FCCS Programming

This data setting describes only the additional or replacement commands used to activate FCCS Services. When used as an FCCS feature, these commands replace the non-network commands (for example, AAED is replaced by AAEDN). Note that the data setting for FCCS Service varies depending on the Node.

Note: When using ASYDL or ASYDN to program an FCCS feature, the basic programming of ASYD is required in addition to ASYDL or ASYDN.

STEP 1: **ANPDL** - Reserve a Number Level for Feature Access. Select Normal Service (N), Hooking Service (H), or Busy Service (B) from the combo box for Connection Indexes. (applicable to any feature assigned to this Level). Assign NND as "2". This should be in accordance with a predetermined Numbering Plan.

STEP 2: **ASPAL** - Two Access Codes must be assigned for this feature. For the Speed Calling Number assignment, assign the Tenant and Access Code (ACC). Assign a Connection Index of "N: Normal". For SRV = "SSC: Service Code", assign SID 14. (Available only to the Primary Station.)

For the Speed Calling Access, assign the Tenant and Access Code (ACC). Assign a Connection Index of Normal (N). For SRV = "SSC: Service Code", assign SID21.

For the Expanded Speed Calling Number assignment, assign the Tenant and Access Code (ACC). Assign for a Connection Index of "N: Normal". For SRV = "SSCA: Service Code Appendix", assign SIDA 134.

For the Expanded Speed Calling Access, assign the Tenant and Access Code (ACC). Assign for a Connection Index of "N: Normal". For SRV = "SSCA: Service Code Appendix", assign SIDA 135.

S-24D Save And Repeat -D

General Description

This feature allows you to save the phone number you just dialed in your Desktop terminal. This feature also allows you to retain the calling phone number from a station or an analog PSTN, ISDN, or CCIS trunk.

Operating Procedure

1. Go off-hook, seize any idle line, and dial a number.
2. When the number has been dialed, press the SAVE AND REPEAT (S&R) Feature Key. The Dialed Number is stored for future use. The associated LED lights. The LCD displays:

Example: Saved Local Number

S&R	**9NX-XXXX**
(Time Display)	

Note: Dialed number displays up to 16 digits when monitoring. However, 32 digits can actually be stored in memory.

3. To access this number later, go off-hook, receive Dial Tone, and press the S&R Key; the saved number is dialed.

Service Conditions

1. One number per Station can be saved.
2. The save function may be set at any time after the number has been dialed and before going on-hook.
3. It is not necessary to erase a saved number in order to save another. The second number replaces the first.
4. A Desktop terminal user may save the number of an Incoming Call, if the call originates at a Station in the system.
5. If necessary, dialing can be added after a repeat operation.
6. To monitor the saved digits, press the S&R Key while idle. The saved digits will be displayed on the LCD.
7. If the LED associated with S&R goes out, the memory is erased. The S&R Key must be pressed again, using the above procedure, to retain the original memory.
8. Up to 32 digits can be saved.

Note: When dialed digits exceed 33 digits, only the first 32 digits of all dialed digits are saved.

9. When digits are dialed during the line is engaged in a call, those digits, Calling Number + dialed digits during the line is engaged in a call, are saved up to 32 digits.
10. When originating a call out of the office by pressing the call origination button (such as function keys):
    - A maximum of 16 digits can be stored in memory.
    - If the called party does not answer it, this feature cannot be registered correctly.
11. To memorize the Telephone Number (TELN) of the Calling Station (versus the Physical Number), assign ASYDL, System Data 1, Index 641, Bit 0 as data “1”.
    Caution: *This bit is also used for SMDR, use care when assigning the data.*

12. You can save the caller ID information if available from an analog PSTN, ISDN, or CCIS trunk. For further information on how to save the caller ID, refer to the description regarding the manual saving of calling number in CALL RETURN [C-151].
13. The SAVE AND REPEAT - D feature is not available for ACD calls.

Interactions

AKYD is assigned after ASDT is assigned. Assign a LENS as TEC 12.

Programming

STEP 1: **AKYD** - Assign the Desktop terminal according to the following parameters:

TN: 1
STN: Station Number
TP: Type of Desktop Terminal (0-3)
KYN: Key Number (1-40)
KYI: Service Index
0: Key Not Used
1: Feature Key
2: Multi Line Key
KD: Not assigned for Feature Key
FKY: Feature Key Number
FKY: = 7 (Save and Repeat)

STEP 2: **ADSL** - Assign SN = 0, 1, 2, 3, 4, 5, 6 and FKY = 7 (Save and Repeat).
This command is used for assigning a function to each Soft Key.

S-25D Service Display -D

General Description

This feature generates LCD displays corresponding to the various features as they are initiated.

Operating Procedure

Each display is automatically generated when a feature is initiated.

Service Conditions

1. Each display is controlled by System Data and is produced accordingly. The service display may be allowed or denied in System Data on a per-system basis.
2. The contrast on the LCD Display Panel is controlled by two buttons on the top front of the Dterm. Press the right (UP) button for a darker image or the left (DOWN) button for a lighter image.
3. The LCD will display one of the features being used.

TRANSFER - CALL TRANSFER	[C-11]
PICKUP - CALL PICKUP	[C-7]
CF ALL - CALL FORWARDING - ALL CALLS	[C-5]
CF BUSY - CALL FORWARDING - BUSY LINE	[C-2]
CF NANS - CALL FORWARDING- DON'T ANSWER	[C-3]
HUNTING - STATION HUNTING	[S-7,8,9]
UCD UNIFORM CALL DISTRIBUTION	[U-1]

4. For a Transferred Call, the LCD displays:

	Transferring Station	Transferred Station
TRANSFER	**XXXX**	**XXXX**
	(Time Display)	

Note: When the Telephone Number of the Transferring/Transferred Station exceeds 8 digits, the last 8 digits are displayed. However, a maximum of a 16-digit transferred Telephone Number can be displayed on the middle line of the LCD, depending on System Data.

Programming

ASYD - System Data 1, Index 78, Bit 1. Is SERVICE DISPLAY in service? 0/1: No/Yes.

S-26D Speed Calling - One Touch -D

General Description

This feature allows Desktop terminal users to dial Telephone Numbers by pressing a Single Key. It is also available to transfer calls to the specified destination by pressing a single Key. Up to 12/24 numbers may be programmed by the Desktop terminal station user.

Operating Procedure

To program (when using SPD setting key)

Note: This procedure is not available for Softphone.

1. Press the SPD setting key
2. Press the SPEED CALLING-ONE TOUCH button when the operation guide appears on the LCD. If the phone number is registered in the terminal, press “Edit” button to change the phone number.
3. Press the desired phone number to dial.
4. Press “SET” key.

To program (when using Feature Key)

1. Press the Feature Key.
2. Press the SPEED CALLING-ONE TOUCH button.
3. Dial the desired number.
4. Press the Feature Key again.
5. Display will show SPEED SET.

Note: The page-change function works as follows: If the Page button is not lit, Page 0 Speed Dial memory is being programmed. (This is the same memory that would be used if the Page feature were not being used.) To program Page 1, press the FEATURE, the SPEED CALLING-ONE TOUCH button, the desired number, and the Feature Key again. Use of the Page feature allows twice the number of equipped speed buttons to be used.

Note: On a Desktop terminal where outgoing call from the Prime Line is restricted with key data (AKYD, S = 1), the following steps are also available:
(1) Press the SPEED CALLING-ONE TOUCH button.
(2) Dial the desired number.
(3) Press the SPEED CALLING-ONE TOUCH button again to complete the setting.

To verify

- Press the Feature Button and the desired SPEED CALLING - ONE TOUCH button: The first 16 characters is displayed.

To use

- **To originate a call**

1. Press the SPEED CALLING-ONE TOUCH button. The call will complete.

- **To transfer a call using the following:**
  - SPEED CALLING-ONE TOUCH button
  - Line/Feature Key programmed as SPEED CALLING-ONE TOUCH Key

1. Press Transfer button during speech. LCD displays “TRANSFER.”
2. Press the following:
   - SPEED CALLING-ONE TOUCH button
   - Line/Feature Key programmed as SPEED CALLING-ONE TOUCH Key
   The button/key above must be programmed with the desired station number. **Note 1**
3. Hang up the call. The call will be transferred.

Note 1: Step 1 may be omitted if “Transfer” button, pause and the desired station number are programmed in the one-touch button.

Service Conditions

1. A maximum of 32 (2 × 16) SPEED CALLING - ONE TOUCH Keys (FKY = 49) can be programmed to the programmable Line/Feature Key of a Desktop terminal with a PAGE Key (FKY = 55). The setting of key pattern can be assigned by ADKS command. For details, see ADKS command in Command Manual.
2. A programmable Line/Feature Key is used for the SPEED CALLING - ONE TOUCH Keys.

   Note: For using the programmable Line/Feature Key, assign the SPEED CALLING - ONE TOUCH Keys by using AKYD command. A maximum of 16 SPEED CALLING - ONE TOUCH Keys (FKY = 49) can be programmed to any Key number (KYN) in consecutive order. if assigning 1 in LKP parameter of ADKS command (16 Line/Feature buttons + DSS keys) or ADKS command is not assigned, My Line number is required to be assigned on a Line/Feature Key. Therefore, SEED CALLING ONE TOUCH Keys can be programmed up to 30 (2 × 15).

3. A stored number may have up to 32 digits.
4. Numbers from 0 to 9, #, *, pauses and hooking can be registered in a dial number.
5. Only the first 16 digits are displayed on the LCD.
6. SPEED CALLING-ONE TOUCH numbers may only be stored while the Desktop terminal is idle.
7. Storing a SPEED CALLING-ONE TOUCH number erases the previously stored number.
8. After dialing the new number, the user must press the SPEED CALLING-ONE TOUCH Key before the LCD displays the time. If the LCD displays the time before the user presses the Key, the number has not been stored. The LCD displays SPEED SET when the number has been stored.
9. Pressing the RECALL Key while programming a SPEED CALLING-ONE TOUCH number inserts a pause. A dash (“-”) is displayed on the LCD.
10. Pressing the RECALL Key as the first digit of a SPEED CALLING-ONE TOUCH numbers inserts a flash entry. A “!” is displayed on the LCD.

11. If a VOICE CALL [V-2] feature is programmed, VOICE CALL initiation can be programmed onto a SPEED CALLING-ONE TOUCH Key. The user dials the number and then presses the TRANSFER Key when programming SPEED CALLING. The LCD will display "V" after the number. Pressing the SPEED CALLING-ONE TOUCH Key initiates an automatic Voice Call to the programmed Station.
12. FLASH ENTRY and VOICE CALL features may be combined on a SPEED CALLING-ONE TOUCH Key to activate an automatic voice-announce transfer. To program, press the SPEED CALLING-ONE TOUCH Key, press the RECALL Key, dial the number, press the TRANSFER Key, then press the SPEED CALLING-ONE TOUCH Key.
13. A PAGE CHANGEOVER Key must be assigned to a programmable Line/Feature Key to activate the second page of SPEED CALLING-ONE TOUCH numbers.
14. VOICE CALL-CCIS [V-7] cannot be activated via this feature.
15. When you program the SPEED CALLING-ONE TOUCH Key using SPD setting key, the following condition must be considered.
    a.) Press any key of the Exit, SPD setting, or Cancel keys to quit the programming operation of the SPEED CALLING-ONE TOUCH Key. If you press a key which is not related to the programming operation, the function assigned to the key is executed.
    b.) When you have an incoming call on the accommodated line while programming the SPEED CALLING-ONE TOUCH Key, you can continue with the programming if you do not answer the call. However, if you answer the call, the conversation will be started and the programming operation will be canceled.
    c.) If you press another SPEED CALLING-ONE TOUCH Key while programming a key for this feature, the number already assigned to the SPEED CALLING-ONE TOUCH Key will be dialed.
    d.) You cannot edit the SPEED CALLING-ONE TOUCH numbers from MA4000 while programming the SPEED CALLING-ONE TOUCH Key.
    e.) When collaboration with Softphone is in operation, if the target station is programming the SPEED CALLING-ONE TOUCH Key, the key operation from the Softphone is not available. **Note 2**

Note 2: If SPEED CALLING-ONE TOUCH Key is programmed on the Softphone, the key name will not be displayed.

16. The same SPEED CALLING - ONE TOUCH Key (FKY = 49) is programmed to the programmable Line/ Feature Key for every 16 keys. Therefore, when the same SPEED CALLING - ONE TOUCH Key is programmed to another key number by AKYD command, the One-Touch Speed Calling data will be inherited to the another key number. The allocation of SPEED CALLING - ONE TOUCH Key is as follows.

Example: KYN 1 and KYN 17 are the same SPEED CALLING - ONE TOUCH Key.
Therefore, when the SPEED CALLING - ONE TOUCH Key assigned on KYN 1 shifts the location to KYN 17, the One-Touch Speed Calling number of KYN 1 will be inherited to KYN 17.

SPEED CALLING - ONE TOUCH Key	KEY NUMBER	
No.1	KYN 1	KYN 17
No.2	KYN 2	KYN 18
No.3	KYN 3	KYN 19
No.4	KYN 4	KYN 20
No.5	KYN 5	KYN 21
No.6	KYN 6	KYN 22
No.7	KYN 7	KYN 23
No.8	KYN 8	KYN 24
No.9	KYN 9	KYN 25
No.10	KYN 10	KYN 26
No.11	KYN 11	KYN 27
No.12	KYN 12	KYN 28
No.13	KYN 13	KYN 29
No.14	KYN 14	KYN 30
No.15	KYN 15	KYN 31
No.16	KYN 16	KYN 32

17. To program the SPEED CALLING - ONE TOUCH Key (FKY = 49) to another key number or delete the SPEED CALLING - ONE TOUCH Key by AKYD command, back up the One-Touch Speed Calling data by BOSD command, and load the backup data after the terminal reregistration.
18. When using MA4000, assign the system data (ASYD, SYS1, Index 386, Bit 1 = 1).

Interactions

1. The PAGE CHANGEOVER Key is useful for changing the page of Feature Keys programmed as SPEED CALLING - ONE TOUCH (AKYD, FKY = 49) or the Feature Key programmed as Call Forwarding/Split Call Forwarding.
2. The Page Changeover function can be activated from any condition (Idle, Busy, Conference, etc.).

3. When assigning SPEED CALLING-ONE TOUCH Numbers to Desktop terminal, if a previous number was not assigned to a particular SPEED CALLING-ONE TOUCH button, the display will not change when that particular button is pressed for programming (it will appear that nothing has happened; the middle portion of the LCD display will continue to display the date and time). Once a number is entered, it will be displayed on the upper portion of the LCD display.
4. The same SPEED CALLING-ONE TOUCH buttons are used for Page 0 and Page 1 numbers. The PAGE CHANGE-OVER Key determines which page is selected.
5. There is no deletion of SPEED CALLING-ONE TOUCH numbers. Previously assigned numbers can only be overwritten.
6. SPEED CALLING-ONE TOUCH may be used to store FORCED ACCOUNT CODES [F-7] and AUTHORIZATION CODES [A-20]. Other forms of Speed Calling may not be used for FORCED ACCOUNT CODES [F-7] and AUTHORIZATION CODES [A-20].
7. When SPEED CALLING-ONE TOUCH is used on a programmable Line/Feature Key, only eight digits may be stored.
8. When using a Desktop terminal, SPEED CALLING - ONE TOUCH can be programmed to any Keys in consecutive order. A maximum of 16 one touch Keys per LENS are allowable.
9. No more than three pauses can be entered in a row.
10. If Abbreviated Call Codes using Speed Calling - Station, Group, or System features are registered in the SPEED CALLING-ONE TOUCH Key, the registered pauses will not be activated when originating a call to ISDN Lines or to Analog Lines via MG-COT.

Programming

No programming is required for the 12 existing SPEED CALLING-ONE TOUCH Keys for the Desktop terminal.

For the detail of backup for One-Touch Speed calling data, see “Backup for One-Touch Speed Calling Data” in Chapter 2 of Operations and Maintenance Manual.

In order to activate the second page of SPEED CALLING-ONE TOUCH numbers, the following must be programmed:

AKYD - Assign a PAGE CHANGEOVER Key to a programmable Line/Feature Key on the Desktop terminal. Assign Type:Function:PAGE [55].

If a SPEED CALLING-ONE TOUCH button is to be used on a programmable Line/Feature Key, complete the following:

Assign Type:Function:SPEED CALL-ONE TOUCH [49].

S-27 Service Feature Peg Count

General Description

This feature provides a statistical usage report on various features, generated at the PCPro. This report will provide the Communications Manager with detailed users' habits and planning data for the communications system.

Operating Procedure

Refer to TRAFFIC MANAGEMENT of CHAPTER 2 in the Operation and Maintenance Manual.

Service Conditions

The number of attempts to invoke the following features will be reported through the PCPro:

1. CALL HOLD [C-6] count: Number of times CALL HOLD Code is dialed by Stations for which the feature is allowed.
2. CALL BACK [C-1] count: Number of times CALL BACK set Code is dialed by Stations for which the feature is allowed.
3. CALL BACK [C-1] cancel count: Number of times CALL BACK is canceled by Stations that have set the feature.
4. EXECUTIVE RIGHT-OF-WAY [E-1] count: Number of times the EXECUTIVE RIGHT-OF-WAY Code is dialed by Stations for which the feature is allowed.
5. CALL WAITING-ORIGINATING [C-31] count: Number of times the Access Code for CALL WAITING-ORIGINATING is dialed by Stations for which the service is allowed.
6. CALL WAITING-TERMINATING [C-12] count: Number of times CALL WAITING-TERMINATING is set to Stations for which the feature is allowed.
7. CALL PICKUP-GROUP [C-7] count: Number of times the CALL PICKUP-GROUP Code is dialed by Stations for which the feature is allowed.
8. CALL PICKUP-DIRECT [C-30] count: Number of times the CALL PICKUP-DIRECT Code is dialed by Stations for which the feature is allowed.
9. CALL FORWARDING-ALL CALLS [C-5] entry count: Number of times a destination number is entered for CALL FORWARDING-ALL CALLS by Stations for which the feature is allowed.
10. CALL FORWARDING-ALL CALLS [C-5] cancel count: Number of times the CALL FORWARDING-ALL CALLS Cancel Code is dialed by Stations for which the feature is allowed.
11. CALL FORWARDING-BUSY LINE [C-2] entry count: Number of times a destination number is entered for CALL FORWARDING-BUSY LINE by Stations for which the feature is allowed.
12. CALL FORWARDING-BUSY LINE [C-2] cancel count: Number of times the CALL FORWARDING-BUSY LINE Cancel Code is dialed by Stations for which the feature is allowed.
13. CALL FORWARDING-DON'T ANSWER [C-3] entry count: Number of times a destination number is entered for CALL FORWARDING-DON'T ANSWER by Stations for which the feature is allowed.
14. CALL FORWARDING-DON'T ANSWER [C-3] cancel count: Number of times the CALL FORWARDING-DON'T ANSWER Cancel Code is dialed by Stations for which the feature is allowed.
15. SPEED CALLING ACCESS-SYSTEM [S-3] count: Number of times a fixed abbreviated number is dialed by Stations for which the feature is allowed.
16. SPEED CALLING ACCESS-STATION [S-21] count: Number of times a variable abbreviated number is dialed by the Stations for which the feature is allowed.

17. SPEED CALLING ACCESS-GROUP [S-23] count: Number of times a variable abbreviated number is registered by Stations for which the feature is allowed.
18. TRUNK ANSWER FROM ANY STATION (TAS) [T-8] count: Number of times the TAS Answer Code is dialed by Stations for which the feature is allowed.
19. OFF-HOOK QUEUING [O-7] count: Number of times an extension accesses an Outgoing Trunk via OFF-HOOK QUEUING after initially encountering a condition where all Outgoing Trunks are busy. The extension must actually access a Trunk in order for the count to be incremented.
20. ON-HOOK OUTGOING QUEUING-ENTRY [O-2] count: Number of times an extension accesses the ON-HOOK OUTGOING QUEUING feature by dialing access or going on-hook (if automatic queuing is provided) after initially encountering a condition where all Outgoing Trunks are busy. The count will be incremented each time the feature is accessed, even if the queuing request is canceled before a Trunk is accessed.
21. ON-HOOK OUTGOING QUEUING [O-2] cancel count: Number of times ON-HOOK QUEUING is canceled by dialing the Cancel Code.
22. CONSULTATION HOLD-ALL CALLS [C-17] count: Number of times CONSULTATION HOLD is executed.
23. CALL TRANSFER-ALL CALLS [C-11] count: Number of times a call is transferred to another extension before the Called Party answers, without Attendant involvement. A call originated internally or from the Outside will be registered in the count.
24. THREE-WAY CALLING [T-2] count: Number of times THREE-WAY CALLING occurs without Attendant assistance.
25. The maximum Tenant Number Counting SERVICE FEATURE PEG COUNT is 63 Tenants.

Note: Additional hardware is required.

Programming

STEP 1: **ASYD**

System Data 1, Index 47, Bit 0. Unit of Traffic Measurement: 0/1 = CCS/Erlang
System Data 1, Index 47, Bit 7. Are traffic measurements required? 0/1: No/Yes

STEP 2: **ATRF/ATRFN** - Assignment of Traffic Measurement Order (CM03)

TYPE: 6: Service Peg Count

PORT: Port number where Traffic Data is to be output.

INTERVAL(min):

Output Interval must be assigned in units of 10 minutes (range of between 30-120 minutes), or data "0" must be assigned, which is the instruction for assigning output time.

Start TIME, End TIME:

Assign the traffic measurement time period between Start TIME and End TIME longer than an hour. Also, if the measurement should be executed throughout a day, assign the same data in both Start TIME and End TIME parameters.

Output TIME:This data entry is valid only when INTERVAL(min)=0.

STEP 3: **DTF201/DTF201N** - Display of Traffic Data 2

For more information, refer to TRAFFIC MANAGEMENT of CHAPTER 2 in the Operation and Maintenance Manual.

S-31 Supervisory Call

General Description

An attendant console can monitor the state of a call after a central office line or TIE line is connected to a station. The console operator can enter into the monitored call if necessary.

Operating Procedure

1. Receive an incoming call from a trunk.
2. Dial the forwarding destination station number.
3. After the station answers the call, press the Talk (three-way conference) key to start a three-way call with the central office line/TIE line and the station.
4. Press SVC (Supervisory Call Set Key) to release the three-way call and enter into a Supervisory Call state.
5. In the case of a call back due to the hooking from the monitored station:
   Line lamp for Supervisory Call starts flashing, and Tone Ringer rings. Answer by pressing the Loop (line) key (Supervisory Call is released).
6. In the case of entering into the monitored call:
   Press the appropriate Loop (line) key to return to a three-way call with Supervisory Call being released.

Service Conditions

<Important Notice>

1. If a station user performs a hooking operation while Supervisory Call is on, it recalls. Therefore, all hooking services except for recalling such as CALL TRANSFER-ALL CALLS [C-11] will be invalid.

<Interactions with Other Features>

1. This service is also used for NON-DELAY OPERATION [N-3].
2. With NON-DELAY OPERATION [N-3], an operator cannot set Supervisory Call before the called party on the external line answers. (In this case, by pressing the Release key the operator can leave and the connection between the station and the called party remains.)

<Hardware/Software>

1. If the called station is PS, Supervisory Call cannot be set before PS answers.
2. When connecting to an ISDN terminal, Supervisory Call cannot be set before the called ISDN terminal answers. (Blind Transfer is not also allowed)

<Interactions with Networks>

1. This service is also valid when using an ISDN Trunk/ISDN terminal as a central office line/station.

<Conditions Specific to This Feature>

None.

<Others>

None.

Programming

STEP 1: **ASYD**

System Data 2

Index 7, Bit=3

Assign as data “1” to enable this service (Supervisory Call).

S-33 Software Line Appearance

General Description

This feature permits ports that do not physically exist to be used as Sub Lines on Dterm Stations. Ports normally reserved on a Dterm for data Stations only, can be assigned as a SOFTWARE LINE APPEARANCE. Additionally, 64 software ports have been added to a Port Interface Module. The 64 software-assigned ports can be assigned as Virtual Line Appearances on Dterm Stations. The use of the data ports and the 64 software-assigned ports conserve valuable hardware for Stations and Trunks.

Operating Procedure

Call Origination via a SOFTWARE LINE APPEARANCE on a Dterm

1. Press the Software Line Appearance Key, then either press the SPEAKER Key or lift the handset.
2. Upon hearing Dial Tone, dial the desired number.

Calling a SOFTWARE LINE APPEARANCE from a Station/Trunk/Attendant Console

1. The Calling Party dials the Telephone Number of the SOFTWARE LINE APPEARANCE.
2. Any Dterm accommodating the SOFTWARE LINE APPEARANCE will receive ringing and have an Incoming Lamp Display. The Dterm Station user presses the Line Key and goes off-hook to answer the Incoming Call.

Service Conditions

1. When using a SOFTWARE LINE APPEARANCE assigned to a data port (TEC:12), Call Origination from that Line Appearance can be made if the card is mounted in that slot.
2. A mix of data ports and Sub Lines can be assigned on a Dterm package.
3. All Sub Lines of Dterms must be located in the same Local Processor as the MY Line of that Dterm.
4. A SOFTWARE LINE APPEARANCE cannot be assigned as a Hotline Station, but can be assigned as a BROKERAGE - HOTLINE [B-12] Station.
5. A SOFTWARE LINE APPEARANCE cannot activate:

CALL BACK	[C-1]
OUTGOING TRUNK QUEUING	[O-2]
NAILED-DOWN CONNECTION	[N-9]

6. When activating CALL PICKUP-GROUP [C-7], the Sub Line will pick up calls assigned to the CALL PICKUP-GROUP of the MY Line of the Dterm on which the Sub Line appears.
7. When accessing or entering SPEED CALLING [S-2] Group/Station data, a SOFTWARE LINE APPEARANCE will enter or access the SPEED CALLING [S-2] Group/Station data of the MY Line of the Dterm on which the Sub Line appears.
8. The STATION MESSAGE DETAIL RECORDING (SMDR) [S-10] of the SOFTWARE LINE APPEARANCE Call activity will be recorded as the MY Line Call activity.
9. The SOFTWARE LINE APPEARANCE cannot be the Called Station of the following features:

VOICE CALL	[V-2]
MESSAGE REMINDER	[M-22D]
BOSS - SECRETARY OVERRIDE	[B-5]

10. Station PEG COUNT [P-7] data will be recorded for SOFTWARE LINE APPEARANCEs as a MY Line of the Dterm.
11. If the DO NOT DISTURB D [D-11D] Key is activated by the SOFTWARE LINE APPEARANCE, ringing will not be sent to the port of the MY Line of the Dterm.

Programming

STEP 1: **ASDT** - Assign the Station as a TEC = 18 for a Virtual Circuit. The Line Equipment Numbers (LENS) of a SOFTWARE LINE APPEARANCE will exist in Groups 24-31.

STEP 2: **AKYD** - This command assigns for Dterms the SOFTWARE LINE APPEARANCE to a programmable Line/Feature Key.

TN: 1

STN: Station Number

TP: Type of Dterm (0-3)

PL TN STN: Prime Line Tenant and Station Number

KYN: Key Number (1-40)

KYI: Service Index

- 0: Key Not Used
- 1: Feature Key
- 2: Multi Line Key

KD: Not assigned for Feature Key

TN STN: Tenant Number and Station Number of SOFTWARE LINE APPEARANCE.

RG: Ringing Assignment per Multi Line Appearance

- 0: No Ringing.
- 1: Ringing in Day Mode Only.
- 2: Ringing in Night Mode Only.
- 3: Not Used
- 4: Ringing in Day and Night Mode - 30 second delay.
- 5: Ringing in Day and Night Mode - 20 second delay.
- 6: Ringing in Day and Night Mode - 10 second delay.
- 7: Ringing in Day and Night Mode.

S-46 Sender/Mark - Out Signal

General Description

This feature is an additional function for use when the Telephony Server station originates a call to a sender line by the register-sender method. This service allows the station user to originate the dial number for the above type of origination when dialing service code etc. after receiving Dial Tone or information.

Note: This feature is not available in North America and Australia.

Operating Procedure

Sender/Mark-out Signal origination:

1. Dial the access code for the relevant line.
2. Dial the desired party's number.
3. Dial the service code after receiving Dial Tone or information from the receiving end.
4. Receive Ringback Tone from the receiving end.

Service Conditions

1. Origination by this feature is possible from a DTMF telephone, DP telephone of Dterm.
2. A maximum of 10 service code digits may be originated by this feature.
3. When the signaling system is DTMF, a 4-wire trunk is required for interface.
4. Service code is included in the Speed Calling code when Speed Calling is set. In this case, manual dialing will not be performed.
5. When the following service are in progress, consecutive dialing cannot be performed:
   a.) After origination by Speed Calling [S-3, S-21, S-23]
   b.) Outgoing/Trunk Queuing [O-2]
   c.) Individual Trunk Access [I-4]
   d.) CCITT No. 7 CCIS
   e.) MFC signaling

Programming

STEP 1: **ASYD** - System Data 1. Index 128. Assign the timer value for the period from receiving connection acknowledgement signal from the distant office till dialing of the service code.

STEP 2: **ANPD** - Assigns minimum necessary number of digits for the first number of the LCR/LCRS access code.

STEP 3: **ASPA** - Assign the LCR/LCRS access code by providing a dummy route.

STEP 4: **AMND** - Assign the maximum number of necessary digits to the Dial (Area/Office) Code.

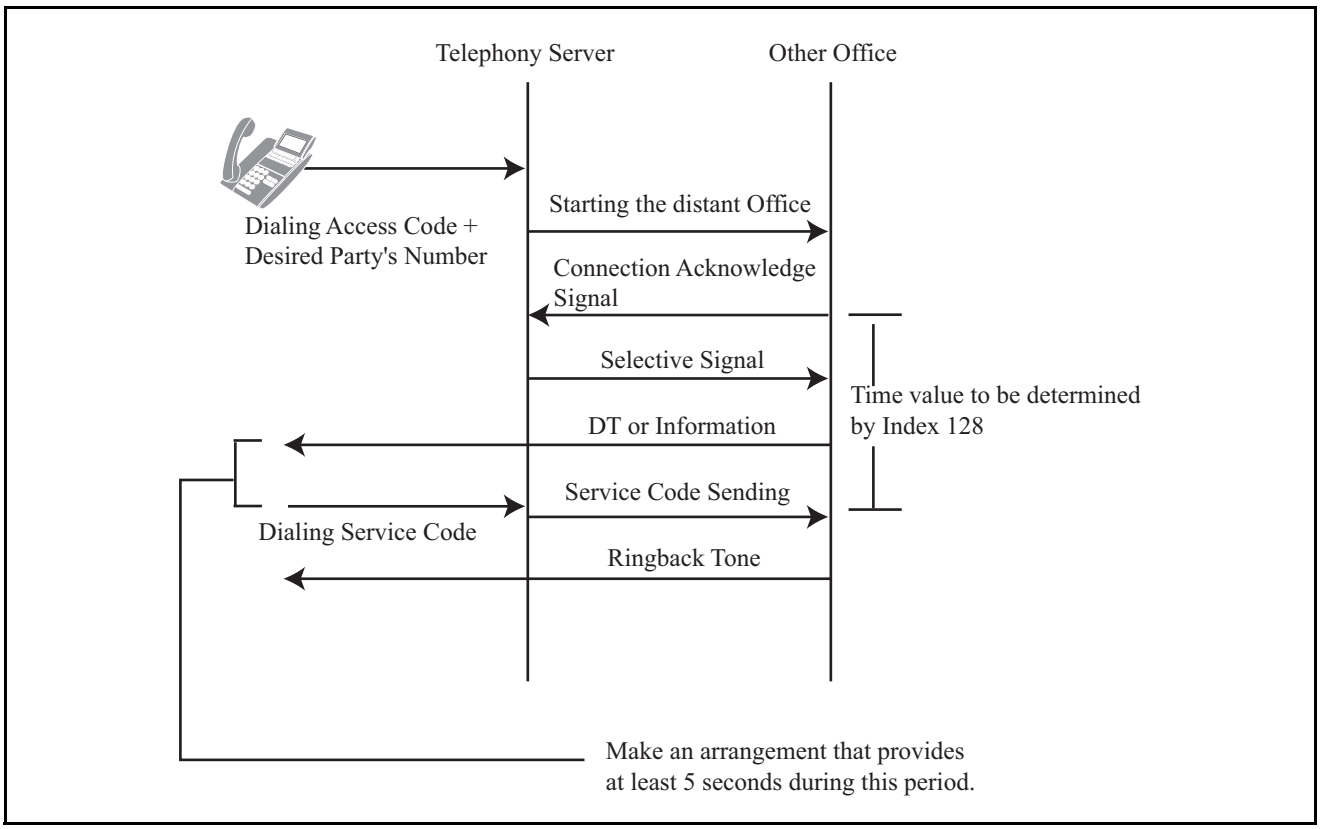
Note: Assign the total number of digits of Destination Code including Access Code and those of Service Code to parameter MND.

STEP 5: **ACDN** - Assign the number of digits of Service Code.
TN: Tenant Number
DC: Destination Code (Assign the same value as that of DC for AMND)
CDN: Number of digits of Service Code.

STEP 6: **ARTD** - Assign various items of route data to the route number.
Assign the dummy route. Assign 1 at all times for the AC parameter of CDN 13.
The route data of the dummy route is TCL = 4, AC =1, and L/T = 1. For the others, assign 0 in all cases.
For the route to which service code can be sent out, assign CD = 1.

Note: To change the data for existing route (RT), initialize or reboot the trunk device such as a circuit card or MG.

STEP 7: **ATRK** - Assign the LENS data of each trunk.
STEP 8: **MBTK** - Release these trunks make busy.
STEP 9: **AFRS** - Assign the Number Pattern Code (NPC) and Outgoing Route Selection Pattern (OPR) to the dummy route.
STEP 10: **AOPR** - Assign the route number including the alternative route to the Outgoing Route Selection Pattern (OPR). Assign the number of skipped digits and Pattern Number Location (PNL) to each route.
STEP 11: **ASDC** - Perform validity/invalidity and near distance toll judge for the Office Code using FX, WATS and the other patterns, and select a suitable pattern.
STEP 12: **ARSC** - Release the dummy route and station connection restriction.



S-48 Station Message Detail Recording - Printout

General Description

The SMDR printout feature is one of the SMDR facilities of the System.
This service provides automatic printer output of the following SMDR information on all Central Office originations, both DOD calls from stations and attendant - handled Central Office origination calls. This feature is available for a printer that is connected via an RS-232C interface.

- Calling telephone number (a maximum of six digits)
- Dialed number (a maximum of 11 digits)
- Start time of call (month, date, hour, minute, second)
- Call duration (a maximum of six digits representing seconds)

Note: This feature is not available in North America and Australia.

Operating Procedure

No manual operation is required.

Service Conditions

1. Two types of call charge start timing can be selected by the system data programming:
   a.) The point when polarity reversal of the Central Office exchange is made, when the called party answers.
   b.) A predetermined time after sending the called party number, when polarity reversal cannot be made from the Central Office exchange.
2. Contents of the printout are as follows:

- In case of call origination

1) CALLING
   STN XXXXX: Origination from station No. XXXXX.
   ATT XX: Origination from attendant console No. XX.
   RT XX: Origination from trunk (origination from route No. XX).
2) ATT*: "*" is printed when the call is handled by attendant.
3) ORIGINATING TIME
   XX,XX XX:XX,XX: Call start time-month, date hour:minute, second are printed.
4) DURATION XXXXXX: Time from call start to end is shown in seconds. (a maximum of six digits; if the time exceeds 1,000,000 seconds, "999,999" is printed.)
5) CALLED NUMBER
   XXXXXXXXXXX: Dialed number is printed (a maximum of 12 digits).
6) C.O. COUNT XXXXXX: Metering count is printed when metering pulse is sent from Central Office "0" is printed if there is no metering pulse (a maximum of four digits, if the value exceeds 10,000, "9,999" is printed).
7) C.O. CHARGE
   XXXXXXX: Product of count and charging unit (set by system data) will be printed when metering pulse is sent from Central Office "0" will be displayed when there is no metering pulse.

- In case of call termination

  1) CALLING
     STN XXXXX: Shows the terminating station.
  5) CALLED NUMBER
     XXXXXXXXXXX: "INCOMING" is printed.

  As for 2), 3), 4), 6), 7), see the "In case of call origination" above.

3. The interface between the Telephony Server and printer is as follows:
   a.) Electrical specification: RS232C (Asynchronous)
   b.) Protocol: Freewheeling
4. SMDR traffic that can be handled by this service is a maximum of 5200 calls per hour (9600 bps).
5. Commercial AC power is used for printer's power source.
6. The distance between printer and the Telephony Server is 15m maximum.
7. Additional hardware is required for this service.
8. The SMDR information that can be stored in case of the printer failure (power failure, paper out), the line failure is as follows:
   1 IMG: 28,000 calls
   2-4 IMG: 47,000 calls
   UMGi: 47,000 calls per CCN/RN
9. The SMDR information is stored for the period of time determined in the system data. After the determined time, the data is erased.
10. The data on the Account Code, Authorization Code, and the Forced Account Code is not printed.
11. The page number is printed from 1 to 255. After page 255, page 1 comes again.
12. This service is not available when AP is accommodated.
13. In case of CCIS centralized Billing System service, the SMDR information of the remote office can be printed out at the central office, however, the calling telephone number and the billing number is not printed out.
14. The ISDN billing information is not printed out.

Programming

STEP 1: **ASYD** - Assign appropriate data in the following indexes.

System Data 1
<Index 32>
b1 (Tenant Data for SMDR): 0/1 = Not Recorded/Recorded
b2 (Selected Route Information): 0/1 = Access Code/Selected Route
b7 (SMDR): 0/1 Out/In Service
<Index 33>

b4	b3	Telephone subject to Total Billing
0	0	Telephone which is called first.
0	1	Telephone which is called last.
1	0	First Telephone after the call has been handled by ATT.
1	1	Last Telephone after the call has been handled by ATT.

b5 (Billing for transferred Incoming Call) 0/1 = Split/Total Billing
b6 (Billing for transferred Outgoing Call) 0/1 = Split/Total Billing

<Index 34>
b0 (SMDR RS-232C Output) 0/1 = Out/In Service
<Index 36>
b0-b7 (Call Record Information Output Port Designation for SMDR-Printout)
Each bit corresponds to the port number of IOC Card. (For example, bit 3 → port 3)

Note: Even if multiple ports are designated as output ports, the SMDR information appears from the port which has the smallest number.

<Index 38>
b0-b7 (Printer Accommodation Designation)
Each bit corresponds to the port number of IOC Card. (For example, bit 3 → port 3)
<Index 44>
b1-b7 (Call Charge Rate per count of Metering Pulse)
The Call Charge Rate ranges from 1 to 127. When no data is assigned here, the default data (=10) is used as Call Charge Rate. The Call Charge is calculated as follows.
C.O. Charge = C.O. Count × Call Charge Rate (default data = 10)
<Index 95>
b0-b7 (SMDR RS-232C Variable Timer)
The System stores SMDR information, when the SMDR information cannot be output for some reasons. (For example, a failure occurred at the terminal device.)

System Data 2
<Index 3>
b0 (SMDR service on a Tenant basis) 0/1 = Out/In Service
b7 (SMDR <STN-STN> service on a Tenant basis) 0/1 = Out/In Service

STEP 2: **AIOC** - Assign appropriate data in the following parameters.
PORT No.: 0-7 (Enter the same port No. which has been assigned in Index 36.)
TERMINAL: PRINTER
SPEED: Assign appropriate data which conforms to protocol of the printer.
PARITY BIT: Assign appropriate data which conforms to protocol of the printer.
CHARACTER BIT: Assign appropriate data which conforms to protocol of the printer.

STEP 3: **ASFC** - Remove restriction of the desired features.
SFI 14 (Station Message Detail Recording System <SMDR> - Trunk Basis) 0/1 = Out/In Service
SFI 58 (Station Message Detail Recording System <SMDR> - for station-to-station call) 0/1 = Out/In Service

Note: The setting enables all traffics to be billing targets. If an unnecessary SMDR data is output, please discard it at the SMDR terminal.

STEP 4: **ARTD** - Assign route data related to SMDR.
CDN 10 (SMDR) 0/1 = Out/In Service for this route
CDN 16 (SMDR2) Assign detailed billing data referring to the table below.

Note: To change the data for existing route (RT), initialize or reboot the trunk device such as a circuit card or MG.

INPUT DATA / CONTENTS		0	1	2	3	4	5	6	7	8	9	10	11	12	13	14	15
b0	0-Detailed billing is not required for incoming calls.	×		×		×		×									
	1-Detailed billing is required for incoming calls.		×		×		×		×								
b1	0-Detailed billing is required for toll and local outgoing calls.	×	×			×	×										
	1-Detailed billing is required only for toll local outgoing calls.			×	×			×	×								
b2,3	0-																
	1-Metering pulses are used.					×	×	×	×								
	2-																
	3-																

S-61 Speed Calling Override - System

General Description

This feature allows numbers programmed for SPEED CALLING to be available to Stations on a system-wide basis, determined by Service Feature Restriction Class (SFC). For example, SPEED CALLING-SYSTEM [S-3] numbers, available to Stations that are not restricted, can be made available to Stations that are restricted.

Operating Procedure

To operate

1. Lift the handset (includes SPEAKER Key operation on Dterm). Listen for Dial Tone.
2. Dial the SPEED CALLING-SYSTEM [S-3] Access Code (maximum three digits).
3. Dial the Abbreviated Code (maximum four digits).
4. If the Station is allowed this call, the call is completed. If not, the Station will hear a Reorder Tone.

To operate from a Dterm

1. Press the REDIAL Key. Receive Dial Tone. The LCD displays:

LNR [*] / SPD [_ _]
(Time Display)

2. Press the desired Call Code.
3. If the Station is allowed this call, the call is completed. If not, the Station will hear a Reorder Tone.

Service Conditions

1. This feature cannot be used on an Outgoing Call which uses OUTGOING TRUNK QUEUING [O-2].
2. This service can be used in combination with CONSECUTIVE SPEED CALLING - SYSTEM [C-62].
3. This feature cannot be used in MFC signaling.
4. This feature cannot be used if the Calling Station is holding another line.
5. This Attendant Console cannot use this feature.
6. SPEED CALLING-SYSTEM [S-3] must first be provided.
7. When this feature is activated by pressing the REDIAL Key, only the Abbreviated Call Code is displayed on the LCD of the Dterm.
   When this feature is activated by dialing the Speed Calling-System Feature Access Code + the Abbreviated Call Code, only the Speed Calling-System Feature Access Code and the Abbreviated Call Code are displayed on the LCD of the Dterm.
8. Restriction is done via the ARSC/ARSCN command.

Interactions

1. SPEED CALLING-SYSTEM [S-3] must first be provided.
2. FORCED ACCOUNT CODES [F-7] and AUTHORIZATION CODES [A-20] cannot be programmed for use via SPEED CALLING-SYSTEM [S-3].
3. Pauses are programmable from the PCPro, using the “P” Key. Pauses consume one digit of the 24 digits available. A pause lasts for about 3 seconds. A pause cannot be added to ISDN, or CCIS Calls.

Note: A pause to MG-COT Calls can be added. In this case, a pause lasts about three seconds.

Note: Registered pauses will not be activated when originating a call to ISDN Lines or to Analog Lines via MG-COT.

4. TOLL DIVERSION/TOLL DENIAL [T-6] and TOLL RESTRICTION 3/6 DIGIT [T-7] will not interact with this feature if a Station is allowed access to an ADC via Service Feature Restriction Class.
5. AUTHORIZATION CODES [A-20] are never required to complete a call.

Programming

STEP 1: **ASYD** - System Data 2, Index 1, Bit 6. Is SPEED CALLING OVERRIDE-SYSTEM to be enabled? 0/1: No/Yes. System Data 2, Index 1, Bit 7. Toll Restriction for SPEED CALLING-SYSTEM [S-3]. Assign data "0".

STEP 2: **ANPD** - Reserve a Number Level for Feature Access. Select Normal Service (N), Hooking Service (H) or Busy Service (B) from the combo box for Connection Indexes. Assign NND in accordance with a predetermined Numbering Plan.

STEP 3: **ASPA** - Assigning an Access Code to SPEED CALLING-SYSTEM [S-3], assign SRV = "SSC: Service Code", SID = 15. Program for "N: Normal". NND = Access Code plus Abbreviated Digit Code (ADC). NND1 = Abbreviated Digit Code (ADC). See ASPD.

Example: 6 + 010

1 digit + 3digits = NND = 4

STEP 4: **ASFC** - Assign a Service Feature Restriction Class that allows SFI 12 to Stations that will have SPEED CALLING-SYSTEM [S-3].

STEP 5: **ASPD** - Enter Tenant Number (TN), Abbreviated Digit Code (ADC), and Called Number (CD). Include the Access Code of the route in the CD.

STEP 6: **AABD** - Assign the Tenant (TN); and the Abbreviated Speed Calling Code, ADC; as assigned in the ASPD command. Assign the Service Feature Restriction Class (SFC) and Restriction Data (RES).

RES 0: Connection is restricted.

RES 1: Connection is allowed.

STEP 7: **AKYD** - Assign SPEED CALLING-SYSTEM [S-3] to a programmable Line/Feature Key. Assign Type:Function:SPEED CALLING-SYSTEM [15].

STEP 8: **ADSL** - Assign S_SPD Key; Assign SN = 1 (Originating; DT Connection) and FKY = 15 (Speed Calling - System). This command is used for assigning a function to each Soft Key.

S-62 Station 5db PAD

General Description

This feature allows a Single Line Station user to be affected by the 5db loss through the system on Station-to-Station Calls.

Operating Procedure

No manual operation is required for this software control feature.

Service Conditions

1. STATION 5db PAD [S-62] is inserted only for intra-office, two-way, Station-to-Station Connections.
2. STATION 5db PAD is inserted into the Called Party of the Station.
3. STATION 5db PAD (both Stations) is not inserted by the Class of Service Feature Restriction Class and/or telephone class when a Station is talking to an Off-Premise Station (OPX).

Programming

ASYD - System Data 1, Index 4, Bit 5, 5db PAD on Single Line Stations? 0/1: Yes/No

S-64 Serial Call - Loop Release

General Description

This feature allows an ATTENDANT CONSOLE [A-3] Loop Key to become available after setting SERIAL CALL [S-15].

Operating Procedure

To operate

1. The Attendant answers an Incoming Central Office Call.
2. The Attendant extends the call to the desired Station, and the desired Station answers.
3. The Attendant presses the SC Key. The Loop is now available for another call.
4. The Called Station and the Incoming Call are connected.
5. The Called Station hangs up. The CCSA lamp on the ATTENDANT CONSOLE flashes, indicating Incoming Call IDENTIFICATION. If the Attendant position is available, an audible indication will be provided.
6. The Attendant presses the CCSA Key or ANSWER Key to return to the Incoming Calling Party.

Service Conditions

1. This feature is available only for Incoming Central Office Trunk Calls.
2. The Attendant can set SC/RC by pressing the SC Key after answering the Station.
3. The Station can switch hook flash after setting SERIAL CALL-Loop RELEASE.
4. It is not permitted to cancel SERIAL CALL-Loop RELEASE before recall.
5. The Attendant Loop will be maintained until the Station is answered, even if the SC Key is pressed. In this case, the following features are not available:

CALL PICK-UP	[C-7, 30]
CALL FORWARDING-DON'T ANSWER	[C-3]

6. SERIAL CALL [S-15] recalling indication continues, even if the system status is changed from Day Mode to Night Mode.
7. When this service is available, the ATTENDANT CONSOLE [A-3] cannot have a CCSA Key.
8. When CALL WAITING-TERMINATING, CALL WAITING-ORIGINATING, ATTENDANT CAMP-ON WITH TONE INDICATION or ATTENDANT CAMP-ON - CCIS is set to the Station already engaged in SERIAL CALL, the following conditions occur:
   a.) If the Station answers the Overriding Call, the previous party engaged in SC is placed on hold. If the Station releases and does not return to the Held Call, the Attendant is recalled.
   b.) If the Station does not answer the Overriding Call, Call Waiting or Attendant Camp-On is activated.
9. When using the CCSA Key on the DESK CONSOLE, an additional module is required.

Programming

ASYD - System Data 1, Index 186, Bit 4 SERIAL CALL-Loop RELEASE 0/1: Out (CCSA Key) /In Service (No CCSA Key)

System Data 2, Index 2, Bit 5. Serial Call service for ATT: 0/1: Out/In service.

System Data 2, Index 7, Bit 2. SERIAL CALL-Loop RELEASE. 0/1: Out/In service.

See SERIAL CALL S-15.

S-65 Short Break Hooking

General Description

This feature enables a Station user using a PB telephone with hooking button, which sends out break pulses to perform all the available hooking services by using this button.

Note: This feature is available in Australia only.

Operating Procedure

The Station user using a PB telephone with the hooking button simply presses the hooking button to perform an available hooking service.

Service Conditions

1. This feature is available in Australia only.
2. The hooking button facility is enabled when the telephone class is assigned as “PB” (TEC = 2).
3. All DP signals are invalid during Call Origination from a PB Telephone (TEC = 2).
4. If Ground Button for Switch Hook Flash is specified as valid in the System Data (System Data 1, Index 68, Bit 6 = 1), the hooking button becomes invalid.

Programming

ASDT - Assign the Station data.
TEC (Telephone Equipment Class) = 2 (PB)

S-99 Split Call Forwarding

General Description

This feature allows a Station to set two different target Stations for CALL FORWARDING-ALL CALLS (FORWARD)/BUSY LINE (FORWARD-BY)/DON'T ANSWER (FORWARD-DA), depending on whether the Incoming Call is from an internal Station or an Outside Party (Central Office TIE, DID, etc.).

Operating Procedure

To operate

1. Station A sets or is set for CALL FORWARDING-ALL CALLS (**Note 1**)/BUSY LINE/DON'T ANSWER. Station B is set as the transfer destination for an Incoming Call from a Station. Station C is set as the transfer destination for an Incoming Call from an Outside Party.
2. An Incoming Call from a Station terminates to Station A. The call is transferred to Station B. **Note 2**
3. An Incoming Call from an Outside Party terminates to Station A. **Note 3**
4. The call is transferred to Station C. **Note 2**

Note 1: From FP95-114 V4 Issue 2.0, CALL FORWARDING - ALL CALLS can be canceled with the entry Access Code (ASYDL/N SYS1, Index 686, Bit 0= 1).

Note 2: For the transfer procedure, refer to the Operation Procedure of CALL FORWARDING-ALL CALLS [C-5], CALL FORWARDING-BUSY LINE [C-2], or CALL FORWARDING-DON'T ANSWER [C-3].

Note 3: The call may be transferred to Station B (transfer destination for an Incoming Call from a Station), depending on the Route Class Data [ARTD, CDN 68:SCF] of the Trunk involved.

To set SPLIT CALL FORWARDING using an Access Code

To set Call Forwarding from an individual Station (Single Line Telephone)

1. Setting Call Forwarding for an Incoming Call from a Station:
   a.) Lift the handset. Receive Dial Tone.
   b.) Dial the Access Code for FORWARD, FORWARD-BY, or FORWARD-DA for an Incoming Call from a Station. Receive Special Dial Tone.
   c.) Dial the number of the desired target Station; receive Service Set Tone.
   d.) Replace the handset.
2. Setting Call Forwarding for an Incoming Call from an Outside Party:
   a.) Lift the handset. Receive Dial Tone.
   b.) Dial the Access Code for FORWARD, FORWARD-BY, or FORWARD-DA for an Incoming Call from an Outside Party. Receive Special Dial Tone.
   c.) Dial the number of the desired target Station. Receive Service Set Tone.
   d.) Replace the handset.

To cancel Call Forwarding from an individual Station (Single Line Telephone)

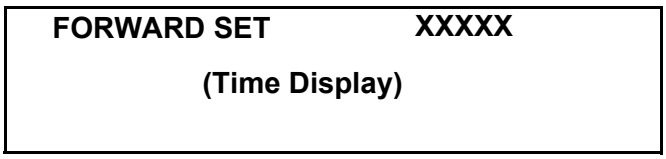
1. Canceling Call Forwarding for an Incoming Call from a Station:
   a.) Lift the handset. Receive Dial Tone.
   b.) Dial the Cancel Code for FORWARD, FORWARD-BY, or FORWARD-DA for an Incoming Call from a Station. Receive Special Set Dial Tone.

c.) Replace the handset.

2. Canceling Call Forwarding for an Incoming Call from an Outside Party:
   a.) Lift the handset. Receive Dial Tone.
   b.) Dial the Cancel Code for FORWARD, FORWARD-BY, or FORWARD-DA for an Incoming Call from an Outside Party. Receive Special Set Tone.
   c.) Replace the handset.

To set SPLIT CALL FORWARDING using Dterm Feature Key
To set Call Forwarding from an individual Station (Dterm)

1. Setting Call Forwarding for an Incoming Call from a Station:
   a.) Confirm that the PAGE* (One-touch changeover) lamp is ON. If the lamp is OFF, turn the lamp on by pressing the PAGE Key.
   b.) Lift the handset or press the SPEAKER Key. Receive Dial Tone.
   c.) Press the FORWARD, FORWARD-BY, or FORWARD-DA Key. Receive Special Dial Tone.
   d.) Dial the number of the desired target Station. Receive Service Set Tone. The following is displayed on the LCD, when the destination is a Station:



When the destination is an ATTENDANT CONSOLE, the Dterm displays:



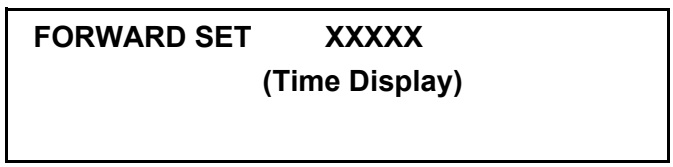
When the destination is an Outside Trunk, The Dterm displays:



The FORWARD, FORWARD-BY, or FORWARD-DA lamp turns on.

   e.) Replace the handset or press the SPEAKER Key. *PAGE Feature Key is a paging button for SPEED DIALING - One-Touch -D [S-26D], (FKY = 55).

2. To set Call Forwarding for an Incoming Call from an Outside Party:
   a.) Confirm that the PAGE (One-Touch Changeover) lamp is OFF. If the lamp is ON, turn the lamp off by pressing the PAGE Key.
   b.) Lift the handset or press the SPEAKER Key. Receive Dial Tone.
   c.) Press the FORWARD, FORWARD-BY, or FORWARD-DA Key. Receive Special Dial Tone.
   d.) Dial the number of the desired target Station. Receive Service Set Tone. The following is displayed on the LCD when the destination is a Station:



When the destination is an Attendant Console, the LCD displays:

```
FWD SET OPR
          (Time Display)
```

When the destination is an Outside Trunk, the Dterm displays:

```
FORWARD SET      XXXXX
          (Time Display)
```

To cancel Call Forwarding from an individual Station (Dterm)

1. Cancelling Call Forwarding for an Incoming Call from a Station:
   a.) Confirm that the PAGE (One-Touch Changeover) lamp is ON. If the lamp is OFF, turn the lamp on by pressing the PAGE Key.
   b.) Lift the handset or press the SPEAKER Key. Receive Service Set Tone.
   c.) Press the FORWARD, FORWARD-BY, or FORWARD-DA Key. Receive Service Set Tone. The LCD displays:

```
FORWARD CANCEL
          (Time Display)
```

   The FORWARD, FORWARD-BY, or FORWARD-DA lamp turns off.
   d.) Replace the handset or press the SPEAKER Key.
2. Cancelling Call forwarding for an Incoming Call from an Outside Party:
   a.) Confirm that the PAGE lamp is OFF. If the lamp is ON, turn the lamp off by pressing the PAGE Key.
   b.) Lift the handset or press the SPEAKER Key. Receive Dial Tone.
   c.) Press the FORWARD, FORWARD-BY, or FORWARD-DA Key. Receive Service Set Tone. The LCD displays:

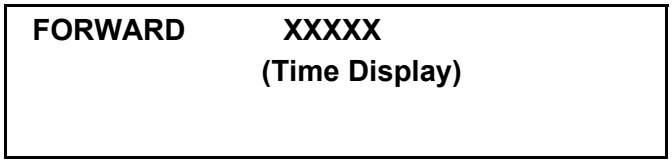
```
FORWARD CANCEL
          (Time Display)
```

   Also, the FORWARD, FORWARD-BY, or FORWARD-DA lamp turns off.
   d.) Replace the handset or press the SPEAKER Key.

To monitor the assigned data using the Dterm Feature Key

1. To monitor the assigned data for calls from Incoming Trunks (Outside):
   a.) Confirm that the LED of the PAGE Feature Key on the Dterm is not illuminated. (If it is illuminated, press PAGE Key to turn it off.)
   b.) While the Dterm is in the idle state (on-hook), press the CALL FORWARDING - ALL CALLS/ BUSY LINE or DON'T ANSWER Feature Key.

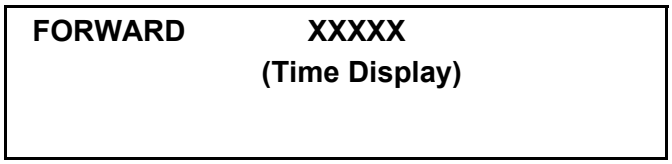
When the destination is a Station, the Dterm displays:



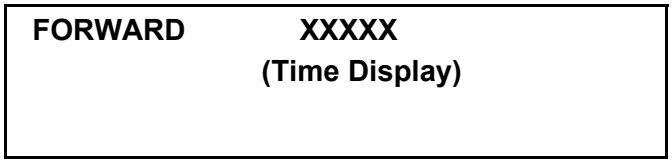
When the destination is an ATTENDANT CONSOLE, the Dterm displays:



When the destination is an Outside Trunk, the Dterm displays:



2. To monitor the assigned data for Station-to-Station Calls:
   a.) Confirm that the PAGE Key LED on the Dterm is illuminated. (If it is not illuminated, press the PAGE Key to turn it on.)
   b.) While the Dterm is in the idle state (on-hook), press the CALL FORWARDING - ALL CALLS/ BUSY LINE or DON'T ANSWER Feature Key.
   When the destination is a Station, the Dterm displays:



When the destination is an ATTENDANT CONSOLE, the Dterm displays:



To set Call Forwarding from the Attendant Console

1. Setting Call Forwarding for an Incoming Call from a Station:
   a.) Press an Idle Loop Key and dial the FORWARD, FORWARD-BY, or FORWARD-DA Access Code for an Incoming Call from a Station. Receive Special Dial Tone.
   b.) Dial the Tenant Number (two digits) of the originating Station, the originating Telephone Number, and the desired target Telephone Number. Receive Service Set Tone.
   c.) Press the RELEASE Key.
2. Setting Call Forwarding for an Incoming Call from an Outside Party:
   a.) Press an Idle Loop Key and dial the FORWARD, FORWARD-BY, or FORWARD-DA Access Code for an Incoming Call from an Outside Party. Receive Special Dial Tone.
   b.) Dial the Tenant Number (two digits) of the originating Station, the originating Telephone Number, and the desired target Telephone Number. Receive Service Set Tone.

c.) Press the RELEASE Key.

To cancel Call Forwarding from the Attendant Console

1. Cancelling Call Forwarding for an Incoming Call from a Station:
   a.) Press an Idle Loop Key and dial the Cancel Code for FORWARD, FORWARD-BY, or FORWARD-DA for an Incoming Call from a Station. Receive Special Dial Tone.
   b.) Dial the Tenant Number (two digits) of the originating Station and the originating Telephone Number.
   c.) Press the RELEASE Key; receive Service Set Tone.
2. Cancelling Call Forwarding for an Incoming Call from an Outside Party:
   a.) Press an Idle Loop Key and dial the FORWARD, FORWARD-BY, or FORWARD-DA Access Code for an Incoming Call from an Outside Party. Receive Special Dial Tone.
   b.) Dial the Tenant Number (two digits) of the originating Station and the originating Telephone Number. Receive Service Set Tone.
   c.) Press the RELEASE Key.

Service Conditions

1. The same Station can be set as the transfer target Station for an Incoming Call from an Outside Party and/or from a Station.
2. For an Incoming Call from a Trunk, whether Call Forwarding destination for an Incoming Call from a Station or for an Incoming Call from an Outside Party is to be used can be designated on each Incoming Route basis. (According to ARTD command, CDN 68:SCF, data.)
3. When an Incoming Call from a Trunk is handled via the Attendant/Station, the destination of Call Forwarding is selected according to the Route Data of the Incoming Trunk (as assigned by ARTD command, CDN 68:SCF).
4. Enter Route Class Data by assigning "1" to CDN 68 (SCF) of the ARTD command, for all the routes that are used to forward incoming trunk calls with CALL FORWARDING - ALL CALLS [C-5], CALL FORWARDING-BUSY LINE [C-2], or CALL FORWARDING-DON'T ANSWER [C-3].
5. Call Forwarding table to be referred to for incoming call is determined by System Data, ARTD (CDN68) and Incoming Call Type as shown below.

   Table 1: forwarding destination assigned in SID=8, 10, 12 of ASPA command
   Table 2: forwarding destination assigned in SIDA=86, 87, 88 of ASPA command

SYSTEM DATA	INCOMING CALL TYPE		
	FROM STATION/ATT	FROM C.O. / TIE LINE	
		ARTD SCF = 0	ARTD SCF = 1
SYS1, Index 79, Bit 2 = 0 Split Call Forwarding is Out of Service	Table 1	Table 1	Table 1
SYS1, Index 79, Bit 2 = 1 Split Call Forwarding is In Service	Table 2	Table 2	Table 1

6. When enabling Split Call Forwarding for the first time from the disabled state, consider the following:

   - To forward trunk calls (C.O./Tie Line) by using Split Call Forwarding

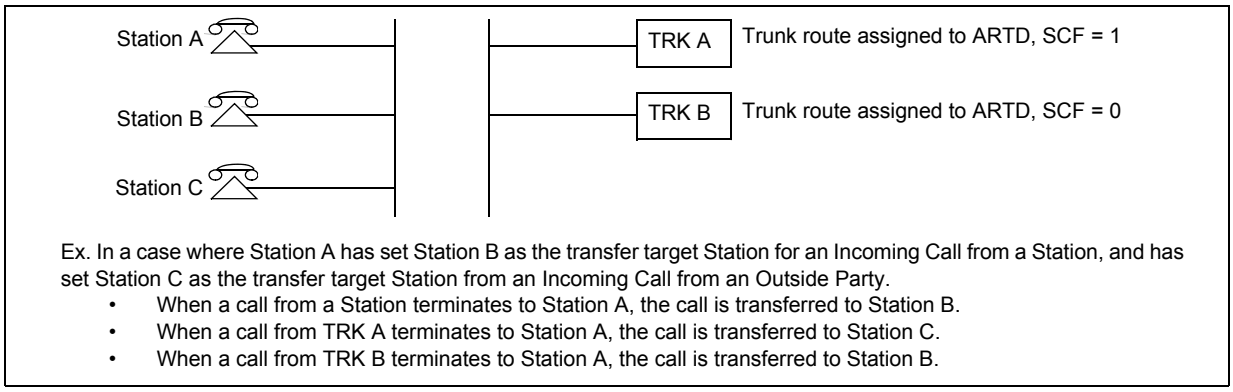
You can use the existing call forwarding data (for CALL FORWARDING-ALL CALLS [C-5], CALL FORWARDING-BUSY LINE [C-2], and CALL FORWARDING-DON'T ANSWER [C-3]).

- To forward station calls by using Split Call Forwarding

  Create data for forwarding station calls by setting SIDA = 86, 87, or 88 of the ASPA command.

Note: As shown in item 5. above, when CDN68 (SCF) of the ARTD command is set to "1," the data in Call Forwarding Table 1 is always used to forward incoming trunk calls whether Split Call Forwarding is enabled or not. Therefore, the existing call forwarding data for incoming trunk calls is still available after the status of this feature is changed (enabled or disabled).
However, in the case of station calls, different Call Forwarding Tables are used, depending on the status of this feature. Thus, if you enable this feature for the first time from the disabled state, you need to create data for forwarding station calls.

7. This service is invalid on Call Forwarding by the system. (Call Forwarding by the ACFO command.)
8. Multiple Call Forwarding is available for those Stations where SPLIT CALL FORWARDING has been set.
9. This service can be set from the Attendant Console.



Ex. In a case where Station A has set Station B as the transfer target Station for an Incoming Call from a Station, and has set Station C as the transfer target Station from an Incoming Call from an Outside Party.

- When a call from a Station terminates to Station A, the call is transferred to Station B.
- When a call from TRK A terminates to Station A, the call is transferred to Station C.
- When a call from TRK B terminates to Station A, the call is transferred to Station B.

10. Indication of SPLIT CALL FORWARDING (ALL CALLS/BUSY LINE/DON'T ANSWER) feature can be accomplished only if the Service Feature Restriction Class and Access Codes are assigned correctly.
11. This feature is available for Stations where CALL FORWARDING services are available by CLASS OF SERVICE.
12. Multiple Call Forwarding is available for those Stations where SPLIT CALL FORWARDING has been set.
13. In the UCD procedure, if SPLIT CALL FORWARDING - ALL CALLS (either for Station-to-Station or Incoming Trunk) is set on a Station in the group, that Station is skipped.
14. If SPLIT CALL FORWARDING is set on a Station in the Hunting Group (STATION HUNTING - CIRCULAR [C-7], - TERMINAL [S-9], that Station is skipped in hunting. However, if the Station has set SPLIT CALL FORWARDING for Station-to-Station Calls only, the Station is hunted for Incoming Calls from Outside. If the Station has set SPLIT CALL FORWARDING for Outside Calls only, the Station is hunted for Station Calls.

15. In this feature, consultation calls (associated with CALL TRANSFER - ALL CALLS [C-11] or BLIND TRANSFER TO STATION [B-18]) are forwarded to the appropriate destination according to whether the original call placed on hold is from a trunk or a station.
16. This service cannot be activated concurrently with CALL FORWARDING - I'M HERE [C-81].
17. A user (via a Station or an Attendant Console) can assign a maximum of 24-digit forwarding Station Number including the Trunk Access Code.
18. A Dterm can display only the first 8 digits of forwarding number on the LCD, even though 24-digit number is assigned.
19. This feature is not available for ISDN terminals.
20. By registering the Call Forwarding - All Calls Access Code + the desired Announcement Trunk Access Code to the CF Setting button of a Line/Feature key on a Desktop terminal, it is possible to enable and disable the settings just by pressing the CF Setting button. For a detailed explanation, see CALL FORWARDING (CF) BUTTON - ONE TOUCH [C-208].

Note: CALL FORWARDING (CF) BUTTON - ONE TOUCH [C-208] is available from FP95-114 V4 Issue 2.0.

Programming

STEP 1: **ASYD** - System Data 1, Index 4, Bit 6. 0/1 = When a call terminates to a terminal that Call Forwarding-All Calls is set, one burst ringing is not sent/sent.

System Data 1, Index 5, Bit 0. 0/1 = Access Code for Call Forwarding-Busy Line and Don't Answer is same/separate.

System Data 1, Index 5, Bit 1-3. Maximum number of Multi-hops for Call Forwarding-All Calls and Busy Line. Note that this setting is valid when "System Data 1, Index 69, Bit 7 = 1" is assigned.

Bit 3	Bit 2	Bit 1	Maximum Number of Multi-hops
0	0	0	Once
0	0	1	Once
0	1	0	Twice
0	1	1	Three times
1	0	0	Four times
1	0	1	Five times
1	1	0	Five times
1	1	1	Five times

System Data 1, Index 69, Bit 7. Multiple Call Forwarding-Busy Line/All Calls: 0/1 = Out/In service.

System Data 1, Index 79, Bit 2 = 1. Split Call Forwarding Service: 0/1 = Out/In service.

System Data 1, Index 248, Bit 7. 0/1 = When going off-hook at Station that Call Forwarding-All Calls is set, Special Dial Tone (SPDT) is not sent/sent.

System Data 2, Index 6, Bit 4. Call Origination Restriction for Station that Call Forwarding-All Calls is set: 0/1 = Restricted/Not restricted. Note that this data is not effective when Split Call Forwarding is set.

STEP 2: **ASYDL/N** - SYS1, Index 686, Bit 0=1.
Disabling Call Forwarding - All Calls with entry Access Code is in service.

Note: Available from FP95-114 V4 Issue 2.0.

STEP 3: **ANPD** - Assign the minimum necessary number of digits for the first number of the Access Code for Split Call Forwarding.

STEP 4: **ASPA** - Assign the Access Code for Split Call Forwarding-All Calls.

CI = "N: Normal", SRV = "SSC: Service Code", SID = 8: Split Call Forwarding-All Calls (for Central Office/TIE): Entry.

CI = "N: Normal", SRV = "SSC: Service Code", SID = 9: Split Call Forwarding-All Calls (for Central Office/TIE): Cancel. **Note 4**

CI = "N: Normal", SRV = "SSC: Service Code", SID = 10: Split Call Forwarding-Busy Line (for Central Office/TIE): Entry.

CI = "N: Normal", SRV = "SSC: Service Code", SID = 11: Split Call Forwarding-Busy Line (for Central Office/TIE): Cancel.

CI = "N: Normal", SRV = "SSC: Service Code", SID = 12: Split Call Forwarding-Don't Answer (for Central Office/TIE): Entry.

CI = "N: Normal", SRV = "SSC: Service Code", SID = 13: Split Call Forwarding-Don't Answer (for Central Office/TIE): Cancel.

CI = "N: Normal", SRV = "SSCA: Service Code Appendix", SIDA = 86: Split Call Forwarding-All Calls (for Station): Entry.

CI = "N: Normal", SRV = "SSCA: Service Code Appendix", SIDA = 87: Split Call Forwarding-Busy Line (for Station): Entry.

CI = "N: Normal", SRV = "SSCA: Service Code Appendix", SIDA = 88: Split Call Forwarding-Don't Answer (for Station): Entry.

CI = "N: Normal", SRV = "SSCA: Service Code Appendix", SIDA = 89: Split Call Forwarding-All Calls (for Station): Cancel. **Note 4**

CI = "N: Normal", SRV = "SSCA: Service Code Appendix", SIDA = 90: Split Call Forwarding-Busy Line (for Station): Cancel.

CI = "N: Normal", SRV = "SSCA: Service Code Appendix", SIDA = 91: Split Call Forwarding-Don't Answer (for Station): Cancel.

Note 4: When SYS1, Index 686, Bit 0=1 (Disabling Call Forwarding - All Calls with entry Access Code is in service), Call Forwarding - All Calls cancel is not required because the cancellation can be performed with the entry Access Code.

STEP 5: **AKYD** - Assign the following data to the Line/Feature Keys of the Dterm when this feature is provided from a Dterm:

Type: Function:CALL FORWARDING-BUSY LINE [1]

Type: Function:CALL FORWARDING-ALL CALLS [2]

Type: Function:CALL FORWARD-DON'T ANSWER [22]

Type: Function:PAGE [55]

STEP 6: **ADSL** - Assign the following soft Keys:

SN = 1, FKY = 1: Call Forwarding - Busy Line (FDB)

SN = 1, FKY = 2: Call Forwarding - All Calls (FDA)

SN = 1, FKY = 22: Call Forwarding - Don't Answer (FDN)

SN = 1, FKY = 55: Call Forwarding - One - Touch Changeover (PAGE)

This command is used for assigning a function to each Soft Key.

STEP 7: **ASFC** - Allows Call Forwarding-All Calls (SFI = 7), Call Forwarding-Busy Line (SFI = 9), or Call Forwarding-Don't Answer (SFI = 8) to the SFC of the Station.

STEP 8: **ARTD** - Assign Route Data to the route of the Trunks which validates C.F.-All Calls, C.F.-Busy Line or C.F.-Don't Answer for an Incoming Call from an Outside Party. CDN 68:SCF = 1)

Note: To change the data for existing route (RT), initialize or reboot the trunk device such as circuit card or MG.

STEP 9: **ACFS** - This feature can be set by this command.

FCCS Programming

This data setting describes only the additional or replacement commands used to activate FCCS Services. When used as an FCCS feature, these commands replace the non-network commands (for example, AAED is replaced by AAEDN). Note that the data setting for FCCS Service varies depending on the Node.

Note: When using ASYDL or ASYDN to program an FCCS feature, the basic programming of ASYD is required in addition to ASYDL or ASYDN.

STEP 1: **ALRTN** - Assign this command to connect the Logical Route to the Physical Route.

STEP 2: **ARTDN** - Assign Route Data to the route of the Trunks which validates Call Forwarding - All Calls, Call Forwarding - Busy Line or Call Forwarding - Don't Answer for an Incoming Call from an Outside Party. CDN 68:SCF = 1)

Note: To change the data for existing route (RT), initialize or reboot the trunk device such as circuit card or MG.

S-106 Slumber Time - Do Not Disturb

General Description

This service allows a slumber time for up to four times per day on a Station Group basis. During the slumber time, all Stations in the group concerned are placed into "Do Not Disturb" mode; and Incoming Calls (Station/DID/DIT) are routed to the Attendant or an Announcement Machine.

Operating Procedure

No manual operation is required.

Service Conditions

1. The Stations for which this feature is set are determined by the Telephone Number of the group preassigned by the MAINTENANCE ADMINISTRATION TERMINAL (PCPro) and Slumber Time data, not by the Service Feature Restriction Class.
2. This feature cannot be set for Data Terminals.
3. This feature cannot be set for Night Stations. (NIGHT CONNECTION [N-1] [N-2] take precedence over SLUMBER TIME - DO NOT DISTURB.)
4. There can be up to seven Slumber Time Groups per system. There is no limitation to the number of Stations that can be assigned to one group.
5. A maximum of four Slumber Time assignments can be made for one Slumber Time Group. When setting more than one assignment, assure that the Times do not overlap each other. Also, a transfer destination can be assigned on each Slumber Time Data basis.
6. A Station cannot be assigned to more than one Slumber Time Group.
7. A Night Transfer Station cannot be assigned to a Slumber Time Group.
8. During Slumber Time, Incoming Calls (Station/DID/DIT) are routed to an Attendant Console or Announcement Machine. When an Attendant Console is designated as the transfer destination, the transfer call is terminated to the ICPT Key on the Attendant Console. When an Announcement Machine is designated as the transfer destination but is not assigned in data by AAED command, the caller receives a Busy Tone (BT).
9. An Announcement Machine with one channel can be connected to each Station Group. When connecting several Station Groups to an Announcement Machine, all groups must be assigned to the same single route and Trunk.
10. Call transfer to a Station in Slumber time mode after holding the call by CONSULTATION HOLD [C-17] is possible only when an Attendant Console is designated as the transfer destination. (The call is transferred to the Attendant Console.) If the Announcement Machine is designated as the transfer destination, the Transfer Call is routed to Reorder Tone (ROT) Connection.
11. The caller is not always connected to the announcement from its beginning.
12. Call origination is available during Slumber Time.
13. Station Group or time is changed, assigned, or canceled by PCPro command.
14. There is no external indication showing Slumber Time set.
15. Any of the following services take precedence over the Slumber Time service:

CALL FORWARDING-ALL CALLS	[C-5]
CALL FORWARDING-BUSY LINE	[C-2]
STATION HUNTING	[S-7, 9]
UNIFORM CALL DISTRIBUTION	[U-1]
DO NOT DISTURB -D	[D-11D]

16. CALL BACK [C-1] and CALL WAITING-ORIGINATING [C-31] can be set from a Slumber Time Station.
17. Each of the following services is restricted to a Slumber Time Station:

CALL WAITING-ORIGINATING	[C-31]
CALL BACK	[C-1]
EXECUTIVE RIGHT-OF-WAY	[E-1]

18. Each of the following terminals can access to a Slumber Time Station during a Slumber Time. (Slumber Time Override Service):
    - Attendant Console
    - Station (when SFI 107 is set to 1 in the ASFC command)
    - Trunk (when CDN109 (SLOV) is set to 1 in the ARTD command)
19. The Announcement Machine uses DAT card and VS32.

Note: In case of using VS32, apart from this feature service conditions, also service conditions of VS32 apply. For detailed explanations, see 4. Service Conditions for Announcement of the Service Conditions section in VS32 CONFERENCE SERVER [V-26].

20. To provide this service in the FCCS Network, all the Nodes in the network must be assigned the same Group Number.
21. Errors of time data are about + 1 minute.
22. This service is not available for Incoming Calls from Q-sig/IS-11572 line/MG-SIP.

Programming

Note: In case of using VS32, see 3. Data Programming for Announcement of the Programming section in VS32 CONFERENCE SERVER [V-26].

STEP 1: **ASLU1** - Assign Slumber Time Group.
TN = Tenant Number
STN = Station Number
G = 1-7: Slumber Time Group Number

STEP 2: **ASLU2** - Assign Slumber Time Data
G = 1-7: Slumber Time Group Number
CNT = Count of Assigned Transfer Destination/Time Data
TFR = 1/2: Transfer to Attendant/Announcement
FROM = Slumber Time Start Time
TO = Slumber Time End Time **Note 1**

] Assign as many data as assigned by CNT

Note 1: If the slumber service is required 24 hours a day, enter the same time data in to both the FROM HOUR/FROM MINUTE parameters and TO HOUR/TO MINUTE parameters.

STEP 3: **ASFC** - Assign the service feature restriction class.
SFI 107 = 0 (Slumber Time is effective)
However, in the case you want to enable incoming calls even during Slumber Time, set the following service feature restriction class of calling terminals.
SFI 107 = 1 (Slumber Time Override is effective)

STEP 4: **ARTD** - Assign the Route Class Data Route Number of the Announcement Trunk and the following Route Class Data.
CDN 109 (SLOV) = 0 (Slumber Time Override Out of Service)
However, in the case you want to enable incoming calls even during Slumber Time, set the following Route Class Data of incoming trunk.
CDN 109 (SLOV) = 1 (Slumber Time Override in Service)

Note: To change the data for existing route (RT), initialize or reboot the trunk device such as circuit card or MG.

STEP 5: **ATRK** - Assign the Trunk data of the Announcement Machine. **Note 2**

STEP 6: **MBTK** - Cancel the make-busy of the Announcement Trunk assigned by ATRK. **Note 2**

STEP 7: **AAED** - Assign the data of the Announcement Machine as shown below: **Note 2**

EQP = 36: Announcement Machine for Slumber Time Group 1
37: Announcement Machine for Slumber Time Group 2
38: Announcement Machine for Slumber Time Group 3
39: Announcement Machine for Slumber Time Group 4
40: Announcement Machine for Slumber Time Group 5
41: Announcement Machine for Slumber Time Group 6
42: Announcement Machine for Slumber Time Group 7

C = Connection time: Disconnected in 30 seconds/connected until caller releases the Station (= 0/1)

R = RBT TRANSMISSION: RBT transmission/No RBT transmission (= 0/1)

A = Call is answered directly/call is extended to an answering device.

M = 1: Multiple Connection

RT = When using a One-channel Announcement Machine by more than one Slumber

TK = Time Group, assign the same data.

Note 2: The data is to be assigned only when Announcement Machine is designated as the transfer destination.

FCCS Programming

This data setting describes only the additional or replacement commands used to activate FCCS Services. When used as an FCCS feature, these commands replace the non-network commands (for example, AAED is replaced by AAEDN). Note that the data setting for FCCS Service varies depending on the Node.

STEP 1: **ALRTN** - Assign this command to connect the Logical Route to the Physical Route.

STEP 2: The Station to use the service is registered by Physical Station Number in ASLU1 command.

STEP 3: The identical ASLU2 data should be programmed at all FCCS Nodes.
By assigning the data, this service is available for the Incoming Calls to be dialed Telephone Number.

S-127D Serial Call -D

General Description

This feature allows the Dterm user to set SERIAL CALL to the central office line/TIE Line Call and extend the call to the desired Station/Trunk. When the communication with the desired party ends, the announcement instructing the caller to dial the next desired party number is sent out. The caller, by following the instructions, can be connected to a number of destinations without hanging up.

Note: This feature is not available in Australia.

Operating Procedure

To operate

1. The Dterm user answers the Incoming Central Office Line/TIE Line Call.
2. The caller requests SERIAL CALL, and the Dterm user presses the SERIAL CALL Key.
3. The caller tells the desired Station/Trunk Number to the Dterm user. The Dterm user extends the call to the desired Station/Trunk using Consultation Hold.
4. When the conversation ends, the caller receives the announcement requesting the caller to dial the next Telephone/Trunk Number.
5. The caller dials the desired Station/Trunk Number. When the Called Party answers, communication can begin.
6. When the conversation ends, the caller receives the announcement requesting the caller to dial the next Telephone/Trunk Number. The caller may successively dial the desired Telephone/Trunk Number.

Note: For the Dterm equipped with a 24-digit LCD, the 16 digits on the left side of the LCD is used for the service set/cancel indication. The 8-digit space on the right side is blank.

Service Conditions

1. Whether to activate this service can be determined on a system basis, but not on a Tenant or Incoming Route basis. Therefore, it is not available to activate this service for each Incoming Route.
2. SERIAL CALL can be set only for the Incoming Central Office Line, TIE Line, and ISDN Line. It cannot be set for the Station or the Outgoing Trunk.
3. The Station that can set/cancel SERIAL CALL is the Dterm with the SERIAL CALL Key. In case of the Incoming Call from a Ringdown Trunk, the call has to be transferred from the Attendant to the Dterm, or the Dterm should be previously designated as the Direct-In Termination Station.
4. The Dterm may set/cancel SERIAL CALL by pressing the SERIAL CALL Key any time while engaged in a call with the Incoming Trunk Party. (The LCD indicates set (SERIAL SET)/cancel (SERIAL CANCEL) for five seconds for every attempt, and the SERIAL CALL Key lights red while SERIAL CALL is set.) However, the Dterm user must not have any other calls when using this service (i.e. this service cannot be used together with CALL TRANSFER-ALL CALLS [C-11], CONSULTATION HOLD-ALL CALLS [C-17], CALL WAITING-TERMINATING [C-12], etc.).
5. The caller can call the Station, central office line, TIE Line, or the Attendant successively by activating SERIAL CALL.
6. When the call terminates on the Central Office Line/TIE Line in the Multi Line and the Called Station answers, the SERIAL CALL Key lights red. However, in case the Dterm user answers by CALL PICKUP - GROUP [C-7]/CALL PICKUP - DIRECT [C-30], it does not light.

7. If SERIAL CALL is not set to the Incoming Trunk, the caller is not connected to the announcement when the conversation with the first party ends (after being placed on Consultation Hold), but receives Reorder Tone.
8. This service is available to the Incoming Route where AUTOMATED ATTENDANT [A-82] or REMOTE ACCESS TO SYSTEM [R-2] are set.
9. When the Registers are all busy when Serial Call is attempted, the caller receives Reorder Tone.
10. Whether to send the announcement can be determined on a system basis. If the announcement is not sent, the caller receives Dial Tone. Both the first and the second announcement is connected in Single Connection, and the caller receives no tone until the announcement is sent.
11. When Serial Call is attempted, and the caller encounters all Announcement Trunks busy or when there is no Announcement Trunk, the caller receives Dial Tone.
12. The caller can dial the desired Telephone/Trunk Number during the announcement.
13. This service is provided only when the caller uses the DTMF telephone.
14. The caller is connected to the announcement when the caller is engaged in a Two-party Connection with the Station, Central Office Line, TIE Line, or the Attendant and then the Station, Central Office Line, TIE Line or the Attendant releases.
15. If the caller does not dial any number after the announcement, the caller receives no tone until the Register prepause timer expires.
16. In case the Register prepause timer expires after the announcement, whether to recall to the operator Dterm user or to release the call automatically can be selected on a system basis.
17. When recalling to the Dterm, whether to end or continue this setting can be selected by System Data. However, in case the Central Office Trunk works with Loop Start system (Route Data LSG = 0), release cannot be detected; therefore, when recall to the Dterm, this setting is ended. If this setting is ended, the Dterm user can press the SERIAL CALL Key again and reset.

    * Recall to the Dterm is activated with the following steps and ringing is according to the ringer pattern 5 ASYD System Data 3 Index 7.

    a.) In case the Dterm setting SERIAL CALL is free, the destination is the set line and only the Dterm is called. If the set line is accommodated in the other Dterm Multi Line, the other Dterm becomes busy status (lights in red) and CALL FORWARDING – ALL CALLS [C-5] and FORWARDING - DON'T ANSWER [C-3] is not available. When the set line is the My Line of the Dterm, the LCD displays RECALL. (Ordinary transaction for recall to the Dterm user)

    b.) In case the Dterm setting SERIAL CALL is busy, another free line among the Dterms is selected by the same transaction as ordinary Serial Calling. Therefore, CALL FORWARDING - ALL CALLS [C-5], CALL FORWARDING - BUSY LINE [C-2] and STATION HUNTING - CIRCULAR [S-7] are available, and the LCD on the Dterm which is selected as a Prime Line indicates RECALL.
    CALL FORWARDING - DON'T ANSWER [C-3] is not available. In case to end the setting is selected at item 18, the Central Office Line/TIE Line is released. (Reorder Tone is sent out); in case to continue is selected, dialing of the next number is allowed. (Dial tone is sent out.)
18. Conditions for restriction are as follows:
    a.) When the dead number or the Dead Level is dialed (even if the dead number announcement data is assigned, the call is not connected to this announcement, but it is recognized as a Redial Call. If the second announcement data is assigned [AAED EQP = 35], this announcement has priority).
    b.) In case of inter digit timer time-out while the Telephone Number is dialed.
    c.) When the Called Station does not answer for 30 seconds (if CALL FORWARDING - DON'T ANSWER [C-3] is set, the call is forwarded.)

d.) When the Called Station is busy (if CALL FORWARDING - BUSY LINE [C-2] is set, the call is forwarded.)

e.) When the Connection is restricted
- when the Telephone Number where the Connection with the Central Office Line/TIE Line is restricted is dialed.
- when the Access Code dialing for Trunk Call Origination is restricted.
- when the Access Code is dialed for Trunk Call Origination, and inter digit timer expires before the Call Originating Route is determined, resulting in restriction.
- when the Outgoing Trunks are all busy (if Tandem Trunk Queuing service is set, the service is activated.)

19. After the second announcement is sent out, whether to allow dialing of the next number or to release the call automatically can be selected on a system basis.
20. The announcement can be repeated and have different durations depending on the Key setting of the Announcement Trunk (CH-DATA or PA-4DATB-A). The Announcement Trunk should always be connected in Single Connection. For a detailed explanation see CHAPTER 3 CARDS IN 7U-PIR and CHAPTER 4 CARDS IN 8U-PIR in the Circuit Card Description manual.

	SETTING	STANDARD SETTING
CH-DATA	SW82-1 to SW82-2 (64-second mode, 128-second mode, 256-second mode, 512-second mode)	SW26-1 to SW26-8 (ON/OFF= played once/repeated continuously)
PA-4DATB-A	SW5-1 to SW5-2 (When SP-702: 16-second mode, 32-second mode, 60-second mode) (When SP-882: 64-second mode, 128-second mode, 240-second mode)	SW6-1 to SW6-4 (ON/OFF= played once/repeated continuously)

21. When the caller attempts Trunk Call Origination by this service and encounters restriction, the class change service via Authorization Code dialing is not available.
22. The following services are denied toward a line or Trunk involved in Serial Call:

CALL FORWARDING - ALL CALLS	[C-5]
CALL FORWARDING - BUSY LINE	[C-2]
CALL FORWARDING - DON'T ANSWER	[C-3]
CALL PICKUP - DIRECT	[C-30]
CALL PICKUP - GROUP	[C-7]
CALL WAITING - ORIGINATING	[C-31]
CALL WAITING - TERMINATING	[C-12]
STATION HUNTING - CIRCULAR	[S-7]
STATION HUNTING - SECRETARIAL	[S-8]
STATION HUNTING - TERMINAL	[S-9]

23. When the Called Station via Serial Call is busy, STEP CALL [S-13] is not available.
24. When the call is extended to a Trunk, the call is charged to the Incoming Trunk (the caller).
25. This feature is not available when receiving calls from MG (Media Gateway) trunk.

Programming

STEP 1: **ASYD** - System data 1, Index 007, Bit 5 = 0/1: when dialing a vacant Level, Unused Number, Timeout, Recall, or Restricted Connection, the caller will be (0) able to redial / (1) disconnected.
System data 1, Index 435, Bit 0 = 0/1: Serial Call in service No/Yes.
System data 1, Index 435, Bit 1 = 0/1: after pre-pause timer, recall/disconnect.
System data 1, Index 435, Bit 2 = 0/1: when recalling, the service will end/continue.

STEP 2: **AKYD** - FKY 84 = Serial Call.

STEP 3: **ADSL** - Assign SCALL Key; Assign SN = 4 (Communication in Progress) and FKY = 84 (Serial Call).
This command is used for assigning a function to each Soft Key.

STEP 4: **AAED** - EQP 34 (Auto Attendant first announcement)

Duration of Connection:	Disconnect in 30 seconds
Do not Send RBT:	Set to ON
Send Answer Signal:	Set to ON
Multiple Connection:	Set to OFF

EQP 35 (Auto Attendant second announcement)

Duration of Connection:	Disconnect in 30 seconds
Do not Send RBT:	Set to ON
Send Answer Signal:	Set to ON
Multiple Connection:	Set to OFF

FCCS Programming

This data setting describes only the additional or replacement commands used to activate FCCS Services. When used as an FCCS feature, these commands replace the non-network commands (for example, AAED is replaced by AAEDN). Note that the data setting for FCCS Service varies depending on the Node.

Note: When using ASYDL or ASYDN to program an FCCS feature, the basic programming of ASYD is required in addition to ASYDL or ASYDN.

STEP 1: **AAEDN**

- EQP 34 (Auto Attendant first announcement)
  Equipment Kind = DAT / Announcement Machine.
  Duration of Connection = Disconnect in 30 seconds.
  Check the Do Not Send RBT box.
  Check the Send Answer Signal box.
  Multiple Connections = Multiple.

- EQP 35 (Auto Attendant second announcement)
  Duration of Connection = Disconnect in 30 seconds.
  Check the Do Not Send RBT box.
  Check the Send Answer Signal box.
  Multiple Connections = Multiple.

STEP 2: **AAEDN** - EQP 34 (Auto Attendant first announcement) C = 0, R = 1, A = 1, M = 0.
EQP 35 (Auto Attendant second announcement) C = 0, R = 1, A = 1, M = 0.

S-129 SMDR Output Expansion - ANI/CPN

General Description

This feature expands SMDR information output, by which Calling Numbers from Central Office may be output to the expanded area.

Note: This feature is available in North America only.

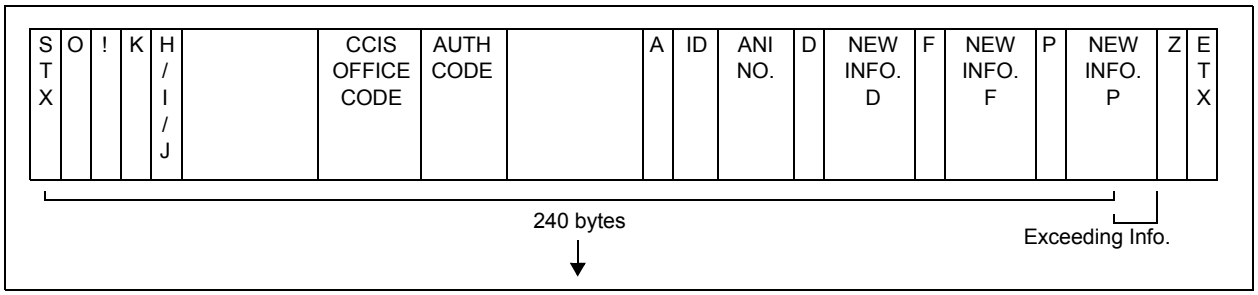
Operating Procedure

No manual operation is required.

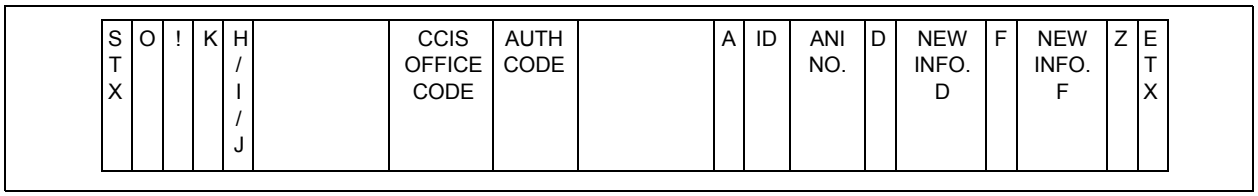
Service Conditions

1. When the expanded area information is used, SMDR equipment must corresponds to the expanded output format.
2. Users may assign 24 kinds of output information (maximum 240 bytes) combined as they like. Information exceeding 240 bytes is not output from the beginning.

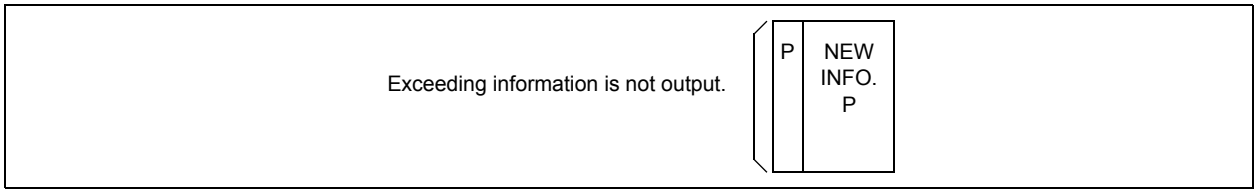
When current information + ANI information (A) + new information D + F + P:



Output information:



No output information:



3. Multi output and dual output are available. (If dual output is assigned, it is possible that the formats are different at each port depending on System Data [e.g. expanded output is available/not available]).

4. Regarding the ANI information identifier, the numbers 0, 2, 3, 4 and 5 are output only when the information is sent from Central Office.

   Note: Meanings of ANI information identifier numbers:
   0 = Not output
   1 = Display
   2 = Not display
   3 = Not provide service
   4 = Outgoing Central Office Call
   5 = Service Condition

5. When an office receiving a Calling Number from Central Office extends the call to another office through CCIS and charging is performed at the extended office, the maximum Calling Number is 16 digits. However, only when ANI information identifier sent from Central Office is 1 (Display), the Calling Number is output.
6. This service is available when using CENTRALIZED BILLING – CCIS [C-55]. (A main office receives charging information from its remote office as it received. However, if the main office makes this service invalid, the information in the expanded area is not output but transferred by using current identifiers (outgoing: KA, incoming: KE, both Station: KB).

   The following is the table showing relationship between a main office and a remote office when remote charging is established.

Transfer Format From A Remote Office / Output Format From A Main Office	Current Transfer Format	Expanded Transfer Format By Activating This Service
This service is activated.	Expanded area information ID is output only. Information is output in space.	Information is output in the expanded format according to System Data of the main office. In case information required to be output at a main office is not in the transfer format from a remote office, information is output same as the current transfer format.
This service is not activated.	Information is output in the current format.	Information in the expanded area is not output but transferred in the current format.

7. This service is not supported in offices performing remote charging.

Programming

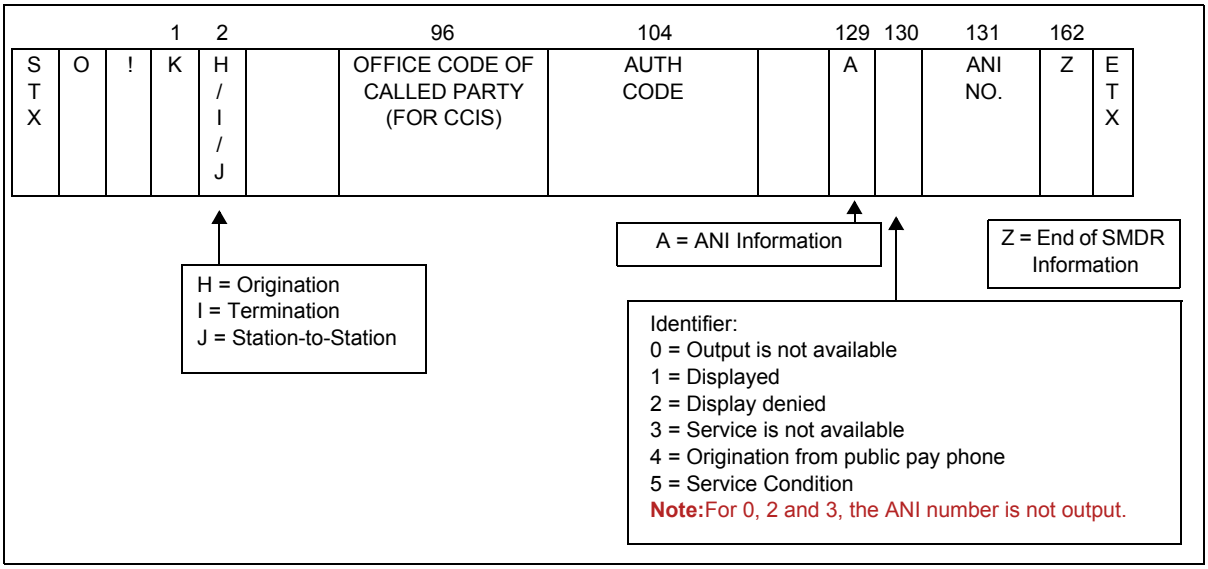
Also see TRUNK LINE APPEARANCE [T-23] for information on how to accommodate Central Office line on a Dterm Line/Feature Key.

STEP 1: **ASYD** - System Data 1, Index 241, Bit 6. 0/1 = When the length of Caller ID exceeds the limit, only the former/latter part of data appears on the LCD.

System Data 1, Index 390, Bit 0. Caller ID Display Pattern on the LCD. 0/1 = Calling Number/Calling Name has priority.

System Data 1, Index 390, Bit 2. when using Dterm Two-line LCD, the lower LCD can be used for displaying Caller ID. 0/1 = Out/In service.

When Calling Number is output to SMDR, the following expanded format is used:



System Data 1, Index 288, Bit 5. Expanded SMDR Output for SMDR A Apparatus. 0/1 = Out/In Service.

System Data 1, Index 290, Bit 5. Expanded SMDR Output for SMDR B Apparatus. 0/1 = Out/In Service.

System Data 1, Index 292, Bit 5. Expanded SMDR Output for SMDR C Apparatus. 0/1 = Out/In Service.

System Data 1, Index 294, Bit 5. Expanded SMDR Output for SMDR D Apparatus. 0/1 = Out/In Service.

System Data 1, Index 296, Bit 0. Output of Calling Number (ANI Information). 0/1 = Out/In Service. Assign data "1".

System Data 1, Index 300, Bit 0. 0/1 = Billing information is sent to the Main Node in Expanded SMDR format if CENTRALIZED BILLING - CCIS [C-55].

STEP 2: **ASFC** - Assign a Service Feature Restriction Class that allows SFI 14 to Stations allowed STATION MESSAGE DETAIL RECORDING [S-10]. SFI 58: SMDR for Station to Station

Note: The setting enables all traffics to be billing targets. If an unnecessary SMDR data is output, please discard it at the SMDR terminal.

STEP 3: **ARTD** - Assign Route Class Data for the Central Office Trunk.

15-LSG :3 50-DPLY :1

When Calling Number is output to SMDR:

10-SMDR :1

Note: To change the data for existing route (RT), initialize or reboot the trunk device such as circuit card, MG, or VS32.

STEP 4: **ATRK** - Assign Trunk data for the Central Office Trunk.

STEP 5: **MBTK** - Cancel the make busy of the Central Office Trunk.

STEP 6: **AKYD** - Assign Call Id Change Display Key. Assign Type:Function:Call ID Change Display [113]. When Call Id Change Display Key is pressed, Display Pattern on the LCD assigned by ASYD command changes according to the conditions below:

When My Line is Prime Line, Call Id Change Display Key is effective while terminating or in conversation.

When My Line is not Prime Line, Call Id Change Display Key is effective while in conversation.

However, Display Pattern returns to the original setting when answering, holding, or transferring the call.

FCCS Programming

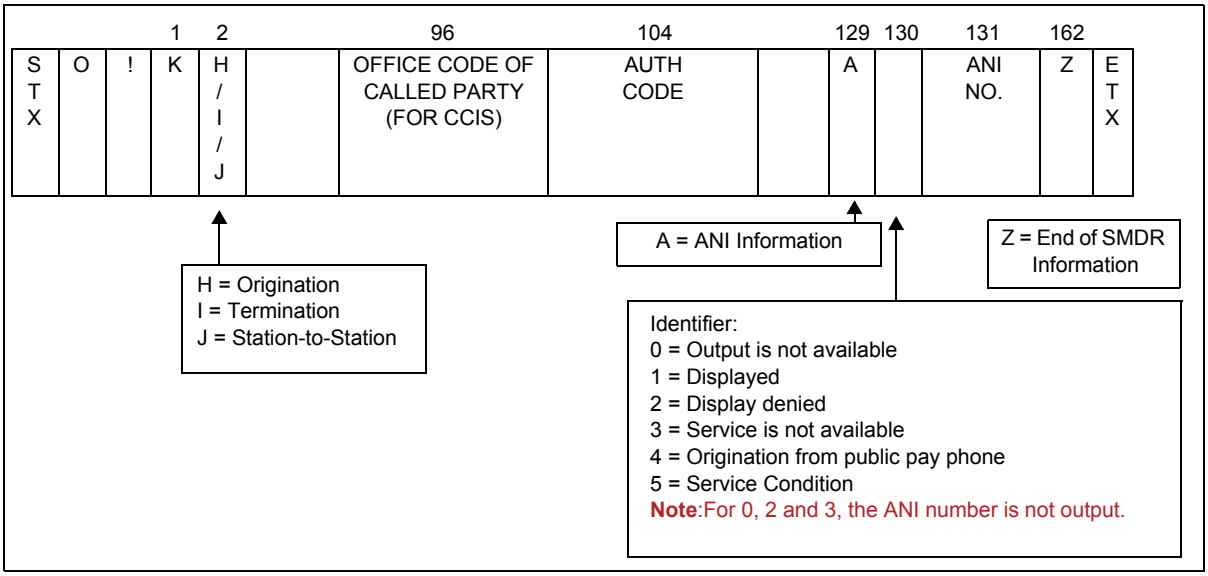
This data setting describes only the additional or replacement commands used to activate FCCS Services. When used as an FCCS feature, these commands replace the non-network commands (for example, AAED is replaced by AAEDN). Note that the data setting for FCCS Service varies depending on the Node.

STEP 1: **ASYD** - System Data 1, Index 241, Bit 6. 0/1 = When the length of Caller ID exceeds the limit, only the former/latter part of data appears on the LCD.

System Data 1, Index 390, Bit 0. Caller ID Display Pattern on the LCD. 0/1 = Calling Number/Calling Name has priority.

System Data 1, Index 390, Bit 2. when using Dterm Two-line LCD, the lower LCD can be used for displaying Caller ID. 0/1 = Out/In service.

When Calling Number is output to SMDR, the following expanded format is used:



System Data 1, Index 288, Bit 5. Expanded SMDR Output for SMDR A Apparatus. 0/1 = Out/In Service.

System Data 1, Index 290, Bit 5. Expanded SMDR Output for SMDR B Apparatus. 0/1 = Out/In Service.

System Data 1, Index 292, Bit 5. Expanded SMDR Output for SMDR C Apparatus. 0/1 = Out/In Service.

System Data 1, Index 294, Bit 5. Expanded SMDR Output for SMDR D Apparatus. 0/1 = Out/In Service.

System Data 1, Index 296, Bit 0. Output of Calling Number (ANI Information). 0/1 = Out/In Service. Assign data "1".

System Data 1, Index 300, Bit 0. 0/1 = -/Billing information is sent to the Main Node in Expanded SMDR format if CENTRALIZED BILLING - CCIS [C-55].

STEP 2: **ASYDL** - System Data 1, Index 578-581, Bit 0.

STEP 3: **ARTDN** - Assign Route Class Data for the Central Office Trunk.

15-LSG :3 50-DPLY :1

When Calling Number is output to SMDR:

10-SMDR :1

Note: To change the data for existing route (RT), initialize or reboot the trunk device such as circuit card, MG, or VS32.

S-137 Selectable Station Call Pickup

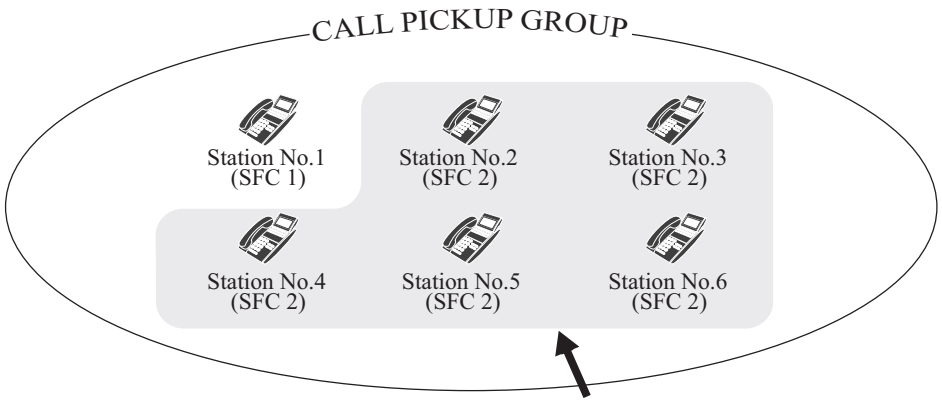
General Description

This feature enables the Station/PS user to pick up the Incoming Call to Station, which are not specified by Service Feature Restriction Class (SFC) data, within Call Pickup Group/Call Pickup Expand Group by dialing Selectable Station Call Pickup/Selectable Station Call Pickup Group Access Code. Incoming Call to pre-specified Stations/PSs cannot be picked up. The figure below shows an example of the feature.

Example: Call Pickup Group

- An incoming call to Station No.1 can be picked up by Station No.2-No.6 using Selectable Station Call Pickup Access Code.
- Incoming calls to Station No.2-No.6 cannot be picked up by all stations using Selectable Station Call Pickup Access Code.

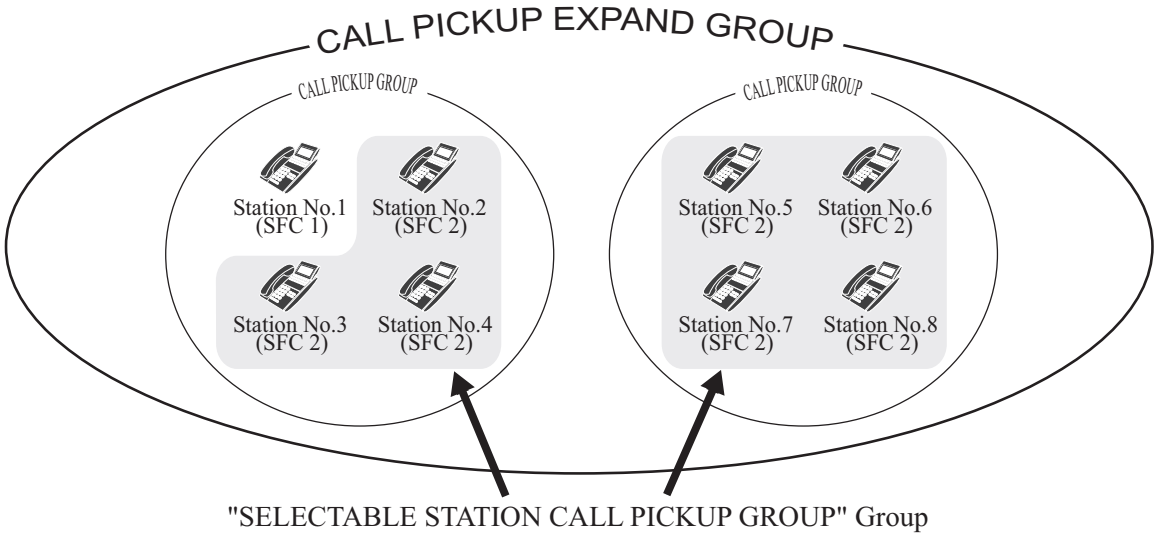
SFC
SFC 1: SFI 183 = 0
SFC 2: SFI 183 = 1



Example: Call Pickup Expand Group

- An incoming call to Station No.1 can be picked up by Station No.2-No.8 using Selectable Station Call Pickup Group Access Code.
- Incoming calls to Station No.2-No.6 cannot be picked up by all stations using Selectable Station Call Pickup Group Access Code.

SFC
SFC 1: SFI 183 = 0
SFC 2: SFI 183 = 1



Operating Procedure

To pick up an Incoming Call by Dialing Access Code

1. A Station within the Call Pickup Group/Call Pickup Expand Group rings.
2. Lift the handset (receive Dial Tone).
3. Dial the Access Code for Selectable Station Call Pickup/Selectable Station Call Pickup Group.
4. The call is connected to the Station.

To pick up an Incoming Call by Pressing Function Key

1. A Station within the Call Pickup Group/Call Pickup Expand Group rings.
2. Press speaker/monitor button (receive Dial Tone).
3. Press Function Key that Access Code for Selectable Station Call Pickup/Selectable Station Call Pickup Group is programmed.
4. The call is connected to the Station.

To pickup an Incoming Call by preset dialing (Only PS)

1. A Station within the Call Pickup Group rings.
2. Dial Access Code for Selectable Station Call Pickup/Selectable Station Call Pickup Group.
3. The call is connected to the Station.

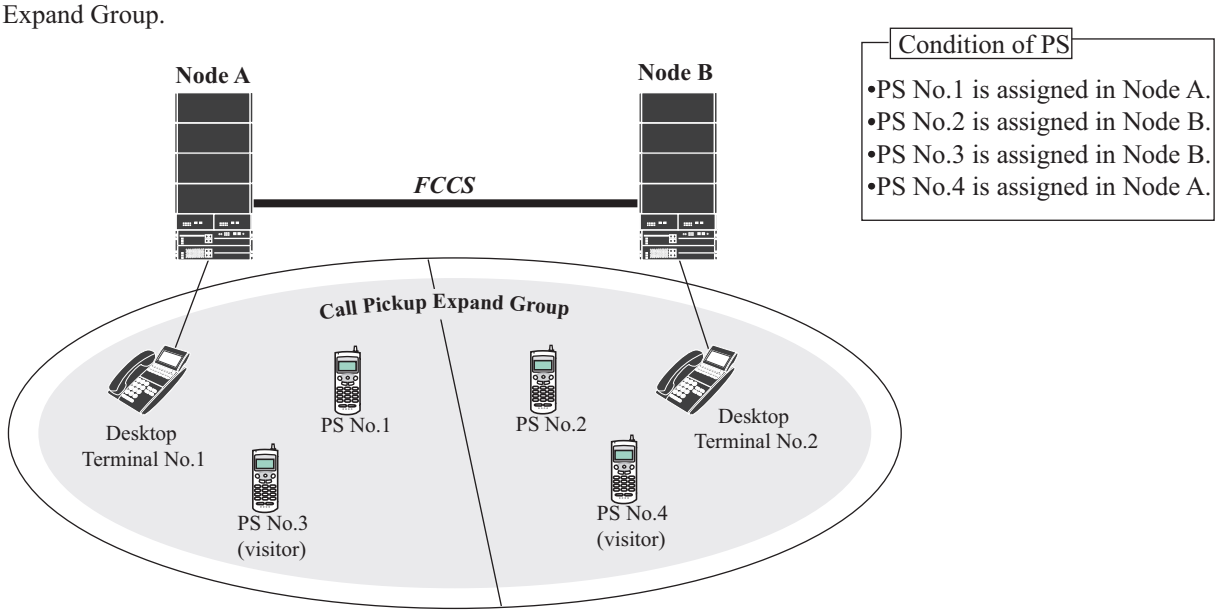
Service Conditions

1. For Service Conditions of Call Pickup Group and Call Pickup Expand Group, refer to Call Pickup Group [C-7].
2. Access Code for Selectable Call Pickup [S-137] and Access Code for Call Pickup Group [C-7] are not identical.
3. When dialing the Access Code for the feature while an Incoming Call is terminating to a Station that "SFI183 = 1" is assigned, Reorder Tone is send. If Access Code for Call Pickup Group [C-7] is dialed, the Incoming Call can be picked up.
4. Call Pickup Group [C-7] and Selectable Station Call Pickup [S-137] can be set to a single Call Pickup Group assigned by ACPG/ACPGL/ACPGN command.

FCCS Conditions

In FCCS Network, the Station/PS cannot pick up a call to PS in visitor Node by this feature. For more details, refer to figure below.

Example:
When Desktop Terminal No.1, Desktop Terminal No.2, and PS No.1 through No.4 are registered in Call Pickup Expand Group.



A Call to PS No.3 and PS No.4 are not picked up by terminal in Call Pickup Expand Group using Selectable Station Call Pickup Group Access Code. Details about other terminals are shown in table below:

Terminating Station	Operating Station					
	Desktop Terminal No.1	Desktop Terminal No.2	PS No.1	PS No.2	PS No.3 (visitor)	PS No.4 (visitor)
Desktop Terminal No.1		×	×	×	×	×
Desktop Terminal No.2	×		×	×	×	×
PS No.1	×	×		×	×	×
PS No.2	×	×	×		×	×
PS No.3 (visitor)	–	–	–	–		–
PS No.4 (visitor)	–	–	–	–	–	

× : Picked up – : Not Picked up

Interactions

1. When this feature is used for Call Pickup Expand Group, dial Access Code for Selectable Station Call Pickup Group.
2. Service Feature Index (SFI) is shared between Call Pickup Group and Call Pickup Expanded Group.
3. Peg count for the feature is counted as Call Pickup Group [C-7].

Programming

- Assignment for Selectable Station Call Pickup

STEP 1: **ANPD/ANPDL** - Reserve a Number Level for Feature Access and cancel.

TN: Tenant Number

1stDC: The first digit of service Access Code. (0-9, *, #)

CI: Select Normal Service (N), Hooking Service (H) or Busy Service (B) from the combo box for Connection Indexes.

NND: Number of Necessary Digits for service Access Code (max. 6-digit)

BLF: Busy Lamp Field Display Activated on ATT (0/1 = Out of Service/In Service)

STEP 2: **ASPA/ASPAL** - Assign an Access Code for this feature.

TN: Tenant Number

ACC: Access Code for this feature (max. 6-digit)

CI: Connection status Index =”N: Normal”

SRV: Kind of Service = “SSCA: Service Code Appendix”

SIDA: Service Index A = 120 (Selectable Station Call Pickup)

STEP 3: **ASFC** - Assign the service feature restriction class of this feature. Stations, assigned the data, cannot be picked up by Access Code for this feature.

D/N: Day/Night Mode

TN: Tenant Number

SFI: Service Feature Index = 183 (Selectable Station Call Pickup)

Res: Restriction Data = 1 (Allowed)

STEP 4: **ACPG/ACPGL** - Register Station Number to make a Call Pickup Group.

TN/UGN: Tenant Number/User Group Number (1-255)

STN/TELN: Station Number/Telephone Number

CNT: Number of Stations to be registered to the Call Pickup Group. (max. 100 Stations)

STEP 5: **AKYD** - Optional. Assign a programmable Line/Feature Key as a Selectable Station Call Pickup Group.

STN: Station Number

KYN: Key Number

KYI: Service Condition = 1 (Function Key)

FKY: Feature Key Code = 136 (Selectable Station Call Pickup)

STEP 6: **ADSL** -Optional (only Desktop terminal). Assign a programmable Soft Key as a Selectable Station Call Pickup.

SN: Status Number = 1 (Originating (DT Connection))

FKY: Function Key Data = 136 (Selectable Station Call Pickup)

- Assignment for Selectable Station Call Pickup Group

STEP 1: **ANPD/ANPDL** - Reserve a Number Level for Feature Access and cancel.

TN: Tenant Number

1stDC: The first digit of service Access Code. (0-9, *, #)

CI: Select Normal Service (N), Hooking Service (H) or Busy Service (B) from the combo box for Connection Indexes.

NND: Number of Necessary Digits for service Access Code (max. 6-digit)

BLF: Busy Lamp Field Display Activated on ATT (0/1 = Out of Service/In Service)

STEP 2: **ASPA/ASPAL** - Assign an Access Code for this feature.
TN: Tenant Number
ACC: Access Code for this feature (max. 6-digit)
CI: Connection status Index = "N: Normal"
SRV: Kind of Service = "SSCA: Service Code Appendix"
SIDA: Service Index A = 121 (Selectable Station Call Pickup Group)

STEP 3: **ASFC** -Assign the service feature restriction class of this feature. Stations, assigned the data, cannot be picked up by Access Code for this feature.
D/N: Day/Night Mode
TN: Tenant Number
SFI: Service Feature Index = 183 (Selectable Station Call Pickup Group)
Res: Restriction Data = 1 (Allowed)

STEP 4: **ACPE/ACPEL** - Register Station/Telephone Number to make a Call Pickup Expand Group.
TN/UGN: Tenant Number/User Group Number (1-255)
STN/TELN: Station Number/Telephone Number
CNT: Number of Stations to be registered to the Call Pickup Expand Group. (max. 100 Stations)

STEP 5: **AKYD** -Optional. Assign a programmable Line/Feature Key as a Selectable Station Call Pickup Group.
STN: Station Number
KYN: Key Number
KYI: Service Condition = 1 (Function Key)
FKY: Feature Key Code = 137 (Selectable Station Call Pickup Group)

STEP 6: **ADSL** -Optional (only Desktop terminal). Assign a programmable Soft Key as a Selectable Station Call Pickup Group.
SN: Status Number = 1 (Originating (DT Connection))
FKY: Function Key Data = 137 (Selectable Station Call Pickup Group)

FCCS Programming

- Assignment for Selectable Station Call Pickup

STEP 1: **ANPDN** - Reserve a Number Level for Feature Access and cancel.
TN: Tenant Number
1stDC: The first digit of service Access Code. (0-9, *, #)
CI: Select Normal Service (N), Hooking Service (H), or Busy Service (B) from the combo box for Connection Indexes.
NND: Number of Necessary Digits for service Access Code (max. 6-digit)
BLF: Busy Lamp Field Display Activated on ATT (0/1 = Out of Service/In Service)

STEP 2: **ASPAN** - Assign an Access Code for this feature.
TN: Tenant Number
ACC: Access Code for this feature (max. 6-digit)
CI: Connection status Index = "N: Normal"
SRV: Kind of Service = "SSCA: Service Code Appendix"
SIDA: Service Index A = 120 (Selectable Station Call Pickup)

STEP 3: **ASFC** - Assign the service feature restriction class of this feature. Stations, assigned the data, cannot be picked up by Access Code for this feature.
D/N: Day/Night Mode
TN: Tenant Number
SFI: Service Feature Index = 183 (Selectable Station Call Pickup)
Res: Restriction Data = 1 (Allowed)

STEP 4: **ACPGN** - Register Station Number to make a Call Pickup Group.
UGN: User Group Number (1-255)
TELN: Telephone Number
CNT: Number of Stations to be registered to the Call Pickup Group. (max. 100 Stations)

STEP 5: **AKYD** - Optional. Assign a programmable Line/Feature Key as a Selectable Station Call Pickup.
STN: Station Number
KYN: Key Number
KYI: Service Condition = 1 (Function Key)
FKY: Feature Key Code = 136 (Selectable Station Call Pickup)

STEP 6: **ADSL** - Optional (only Desktop terminal). Assign a programmable Soft Key as a Selectable Station Call Pickup.
SN: Status Number = 1 (Originating (DT Connection))
FKY: Function Key Data = 136 (Selectable Station Call Pickup)

- Assignment for Selectable Station Call Pickup Group

STEP 1: **ANPDN** - Reserve a Number Level for Feature Access and cancel.
TN: Tenant Number
1stDC: The first digit of service Access Code. (0-9, *, #)
CI: Select Normal Service (N), Hooking Service (H), or Busy Service (B) from the combo box for Connection Indexes.
NND: Number of Necessary Digits for service Access Code (max. 6-digit)
BLF: Busy Lamp Field Display Activated on ATT (0/1 = Out of Service/In Service)

STEP 2: **ASPAN** - Assign an Access Code for this feature.
TN: Tenant Number
ACC: Access Code for this feature (max. 6-digit)
CI: Connection status Index = "N: Normal"
SRV: Kind of Service = "SSCA: Service Code Appendix"
SIDA: Service Index A = 121 (Selectable Station Call Pickup Group)

STEP 3: **ASFC** - Assign the service feature restriction class of this feature. Stations, assigned the data, cannot be picked up by Access Code for this feature.
D/N: Day/Night Mode
TN: Tenant Number
SFI: Service Feature Index = 183 (Selectable Station Call Pickup)
Res: Restriction Data = 1 (Allowed)

STEP 4: **ACPEN** - Register Telephone Number to make a Call Pickup Expand Group.
UGN: User Group Number (1-255)
TELN: Telephone Number
CNT: Number of Stations to be registered to the Call Pickup Expand Group. (max. 100 Stations)

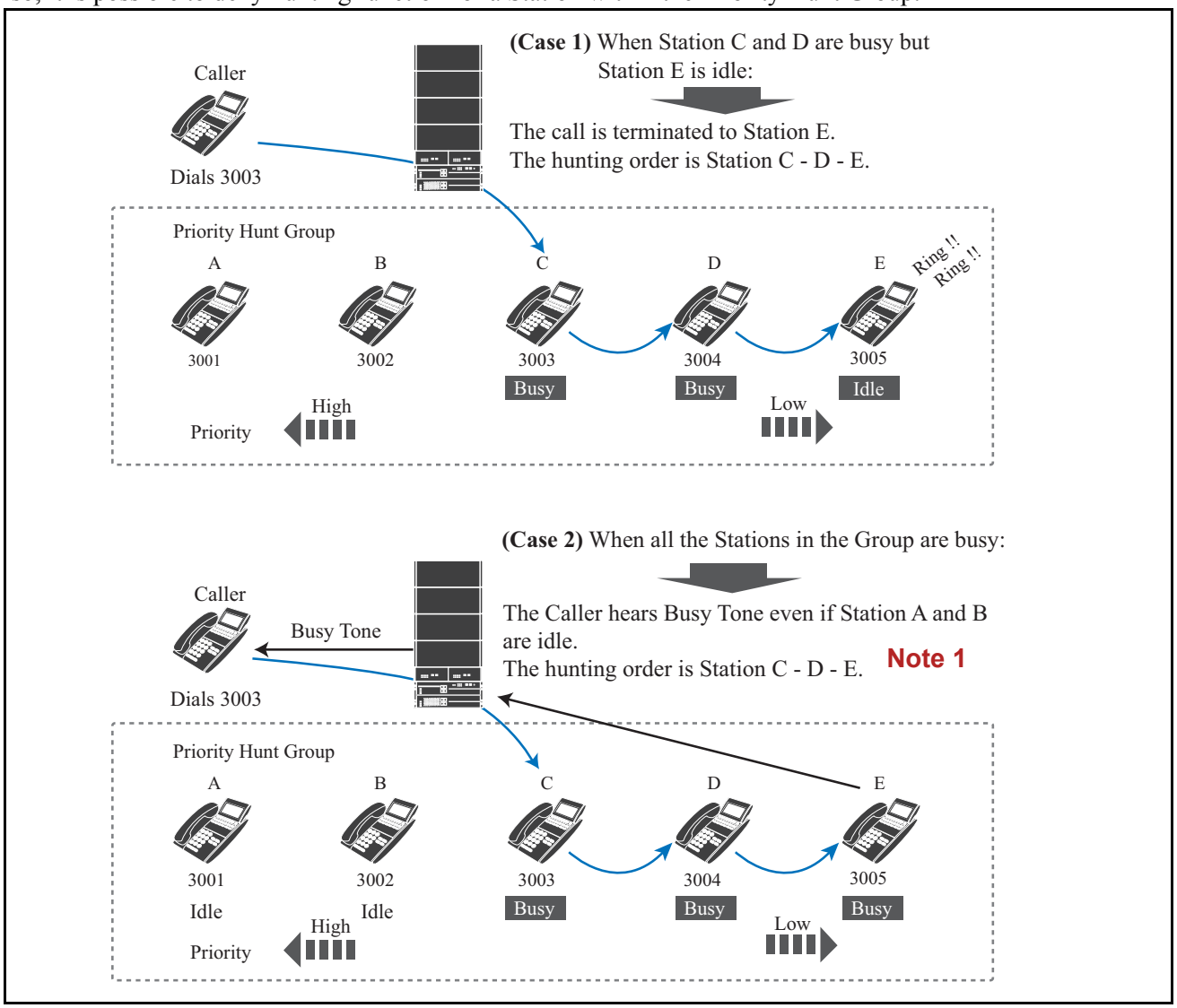
STEP 5: **AKYD** - Optional. Assign a programmable Line/Feature Key as a Selectable Station Call Pickup Group.
STN: Station Number
KYN: Key Number
KYI: Service Condition = 1 (Function Key)
FKY: Feature Key Code = 137 (Selectable Station Call Pickup Group)

STEP 6: **ADSL** - Optional (only Desktop terminal). Assign a programmable Soft Key as a Selectable Station Call Pickup Group.
SN: Status Number = 1 (Originating (DT Connection))
FKY: Function Key Data = 137 (Selectable Station Call Pickup Group)

S-138 Station Hunting - Priority

General Description

This feature permits a call directed to a Busy Station of a Priority Hunt Group to be automatically forwarded to other Stations within the Hunt Group, in a programmed order. The call arrives to the busy Station of the group and is forwarded from Higher Priority Stations to Lower Priority Stations until it finds an idle Station that rings. When the Lowest Priority station is also busy, the caller receives Busy Tone. For example, as illustrated in the following figure, if a call is placed to Station C and Station C is busy, the call is forwarded to Station D and so on, in sequence. If Station E (Lowest Priority Station) is busy, the call receives Busy Tone.
Also, it is possible to deny hunting function for a Station within the Priority Hunt Group.



Note 1: If a secretarial Station is assigned for the Hunt Group, the call is forwarded to the secretarial Station. For the secretarial Station, see STATION HUNTING - SECRETARIAL [S-8].

Note: In this feature, Busy means the Station cannot receive a call (because of reasons such as talking, receiving other call, do not disturb being set).

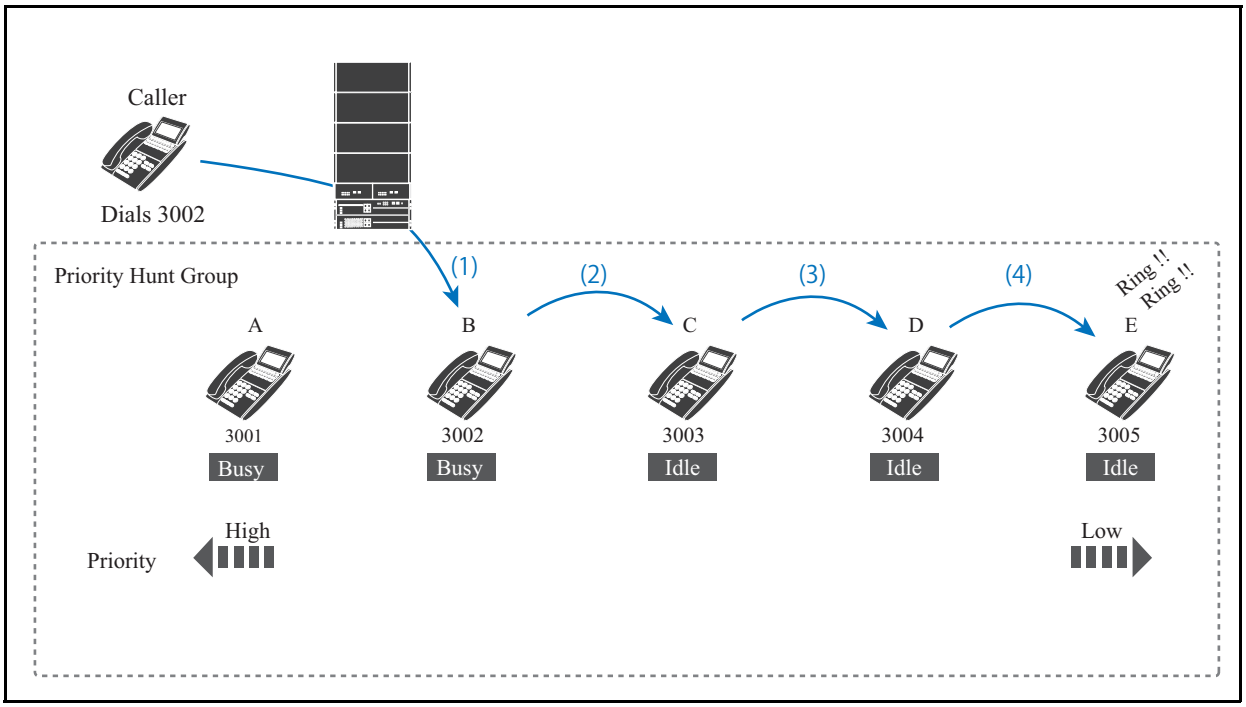
- TRANSFER OF UNANSWERED CALLS IN A HUNT GROUP

When the caller dials a Station in the Priority Hunt Group, the call arrives to a Station within the Priority Hunt Group, and the Station does not reply within a programmed period of time, the call can be transferred again to another idle Station.

Note: Transfer of unanswered calls is available from FP95-114 V4 Issue 4.0.

As for the transfer of an unanswered call in a Priority Hunt Group, the figure below gives an example for the transfer pattern with the following steps.

1. A call is made to Station B.
2. Station B is busy, the call is transferred to the first idle station in the hunting order (Station C).
3. The call remains unanswered within a programmed period of time and is transferred to the next idle Station in the hunting order (Station D).
4. The call remains unanswered within a programmed period of time and is transferred to the next idle Station in the hunting order (Station E).



Note: In the example above, the call may be transferred to a secretarial Station if it has been set. For the secretarial Station, see STATION HUNTING - SECRETARIAL [S-8].

Operating Procedure

No manual operation is required.

Service Conditions

<Important Notices>

1. Station Number can belong to one Hunt Group only, it cannot belong to another Hunt Group including that for STATION HUNTING - CIRCULAR [S-7], STATION HUNTING - TERMINAL [S-9], STATION HUNTING - PRIORITY [S-138], STATION HUNTING - PRIORITY WITH SWITCHBACK [S-139] or UNIFORM CALL DISTRIBUTION (UCD) [U-1].
2. A station can be assigned by the Telephone Number (LDM or NDM). However, it is not possible to assign the same Telephone Number to more than one Hunt Group.

<Interactions with Other Services>

1. If the Station of a Hunt Group is in the following states or the following features are set in the Station of a Hunt Group, the following occurs.

   - **A**: The feature is activated and the call is forwarded.
   - **S**: The Station is skipped.

Feature/State	Direct Call to the Station	Call Transfered to the Station in the Hunting
CALL FORWARDING - ALL CALLS [C-5]	**A** Note 2	**S**
DO NOT DISTURB [D-11]	**S**	**S**
BUSY STATION	**S**	**S**
BUSY STATION + CALL FORWARDING - BUSY LINE [C-2]	**A** Note 2, Note 3	**S**
BUSY STATION + CALL FORWARDING - BUSY LINE [C-2] set with the ACFO/ACFOL/ACFON command (on a tenant/route basis)	**S**	**S**
LOGGED OUT STATION	**S**	**S**
LOGGED OUT STATION + LOGGED OUT IP STATION - CALL DESTINATION [L-53]	**A**	**S**
LOGGED OUT STATION + LOGGED OUT IP STATION - CALL DESTINATION [L-53] set with the ACFO/ACFOL/ACFON command (on a tenant/route basis)	**S**	**S**

Note 2: If CALL FORWARDING - OVERRIDE [C-26] is set to the forwarding destination, the call forwarding overriding from the forwarding destination to the Station of the Hunt Group will hunt if the Station of the Hunt Group is busy.

Note 3: When a call to a station in a Hunt Group is forwarded by CALL FORWARDING - BUSY LINE [C-2] and the forwarded party (also a member of another Hunt Group) is also busy, the resultant hunt target can be designated from the forwarding side Hunt Group or the forwarded side Hunt Group. This can be specified with ASYD/ASYDN SYS1 Index 69 Bit 5.

2. A Phantom Number can be assigned to the Station of a Hunt Group. However, when a caller dials the Phantom Number, hunting does not occur for the Station that receives the call.
3. Recall to a Station in the Hunt Group from the following features will not hunt:
   - CALL BACK [C-1]
   - CALL HOLD [C-6]

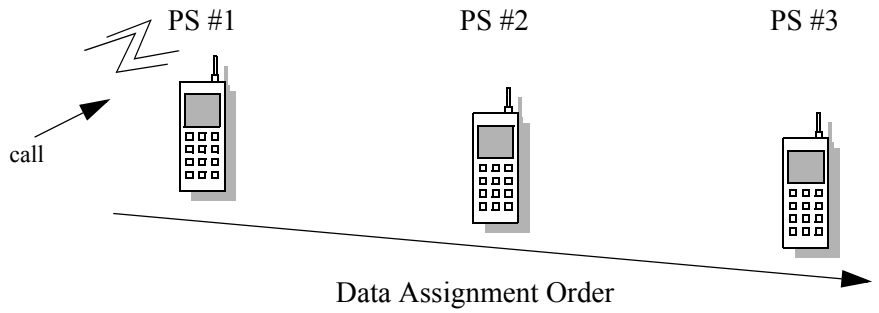
- OUTGOING TRUNK QUEUING [O-2]
- PAGING TRANSFER [P-10]

4. For STATION MESSAGE DETAIL RECORDING [S-10] the number recorded is the Station Number where the call actually terminates.
5. HOTLINE [H-1] cannot be assigned to the Station of a Hunt Group.

<Hardware/Software>

1. A call from ATTENDANT CONSOLE [A-3] will not hunt.
2. The conditions when using PS are as follows:
   a.) If a call terminates to a PS in the Hunt Group while the PS is Out of Zone or its power is OFF, whether the call is processed as PS Out of Zone or hunting is executed is determined by System Data (ASYDL, SYS 1, Index 688, b0). When using this feature in an FCCS environment, assign the same data value to the system data in each node.
   b.) When hunting is selected in the system data, the caller hears no tone until the hunted PS answers. Total length of no tone is obtained by the following.
      - Time for Detecting Out of Zone (Maximum eight seconds) × Number of Hunting
   c.) When a Desk Console operator dials the number of PS that belongs a Hunt Group, the hunting is not available if the PS is Out of Zone.
   d.) When receiving an analog PSTN incoming call while all PSs in the Hunt Group are Out of Zone, the call processing will be stopped after detecting Out of Zone of the last PS in the Hunt Group. At this time, the calling party will receive RBT when the call is a Direct In Termination call. If Automated Attendant is used, the calling party will receive ROT after No Tone. Note that the call cannot be terminated to the destination even if the PS returns to the zone during RBT.
   e.) When receiving an ISDN incoming call while all PSs in the Hunt Group are Out of Zone, hunting starts after calling up the destination. However, timeout will occur at Central Office side, resulting in ROT connection (hunting is not available).
   f.) If all Stations in the Hunt Group are busy or PS Out of Zone, the call can be forwarded a Station in the group. The station is called secretarial Station. The caller hears Busy Tone if the secretarial Station is also busy.
   g.) A PS can be assigned as a secretarial Station.
   h.) If a PS assigned as a secretarial Station is placed in Out of Zone, the caller hears Busy Tone.
   i.) A PS in the Hunt Group can set PS Out of Zone - Call Forwarding. When the PS has been set as able of hunting, the incoming call will be forwarded only if the call cannot terminate to any stations in the group. When the PS has not been set as able of hunting the call can be forwarded only if the PS is directly called.
   j.) When a PS in the Hunt Group performs roaming, the PS is skipped. However, the PS is a station that receives the call directly, it will be roaming call termination.
   k.) If a PS receiving a call turns to Out of Zone, this may cause a network failure. In this case, the caller hears Busy Tone.
   l.) The following shows call processing when Out of Zone or busy is involved for this feature.

This example assumes: - hunting data has been assigned in the order of PS #1, #2, #3
- PS #1 first received a call.



When Hunting is selected (ASYDL, SYS1, INDEX 688, Bit 0 = 0)

PS #1	PS #2	PS #3	Processing
Out of Zone (Forwarding has been set)	Out of Zone	Out of Zone	After calling up PS #1, PS #2, PS #3, the forwarding destination is called.
Out of Zone (Forwarding has not been set)			After calling up PS #1, PS #2, PS #3, the caller receives ROT or Automatic Announcement Called PS Out of Zone.
Busy	Out of Zone	Out of Zone	After calling up PS #2 and PS #3, the caller receives Busy Tone.
Busy	Busy	Out of Zone	After calling up PS #3, the caller receives Busy Tone.

When Out of Zone Processing is selected (ASYDL, SYS 1, INDEX 688, Bit 0 = 1)

PS #1	PS #2	PS #3	Processing
Out of Zone (Forwarding has been set)	Out of Zone	Out of Zone	After calling up PS #1, the forwarding destination is called.
Out of Zone (Forwarding has not been set)			After calling up PS #1, the caller receives ROT or Automatic Announcement Called PS Out of Zone.
Busy	Out of Zone	Out of Zone	After calling up PS #2, the caller receives Busy Tone.
Busy	Busy	Out of Zone	After calling up PS #3, the caller receives Busy Tone.

<Conditions Specific to This Feature>

1. The maximum number of Stations per group is 100. The feature is usable with all Stations and there is no limit on the allowable number of Hunt Groups within a system.

2. This feature enables a Station within Hunt Group to deny hunting function for an Incoming Call originated from Station (including PS) (ASFC/ASFCN SFI 181 =1). The service conditions are as follows:
   a.) When CALL FORWARDING-DON'T ANSWER [C-5] is set, a call to the Station is forwarded according to their service condition.
   b.) When CALL FORWARDING-BUSY LINE [C-2] is set, a call to the Station is forwarded according to their service condition.
   c.) A call to the Phantom Number assigned to the Station will not hunt.
   d.) External Trunk (such as DDD (TCL = 1)/FX (TCL = 2)/WATS (TCL = 3), etc) is not available.

3. The conditions for the transfer of unanswered calls in a Hunt Group feature are as follows: **Note 4**

Note 4: Available from FP95-114 V4 Issue 4.0.

   a.) The feature is only possible for Hunt Groups with data assigned via Telephone Numbers (LDM or NDM).
   b.) Be aware that if there is a value set in Timer Value of No Answer Timer set in the ADAI command for a Station in the group it will have priority over the value set with ASYDL/ASYDN, SYS1 Index 1396. If no value is set for ADAI command or ASYDL/ASYDN, SYS1 Index 1396, other Timer Value of No Answer Timer (ASYD SYS1, Indexes 139, 140, 141, 247) will be used.
   c.) The display of a Station receiving an incoming call transfer based on this feature will be the same as in the case of CALL FORWARDING - DON'T ANSWER [C-3]. Also, in the case of other external devices, it will be treated in the same way.

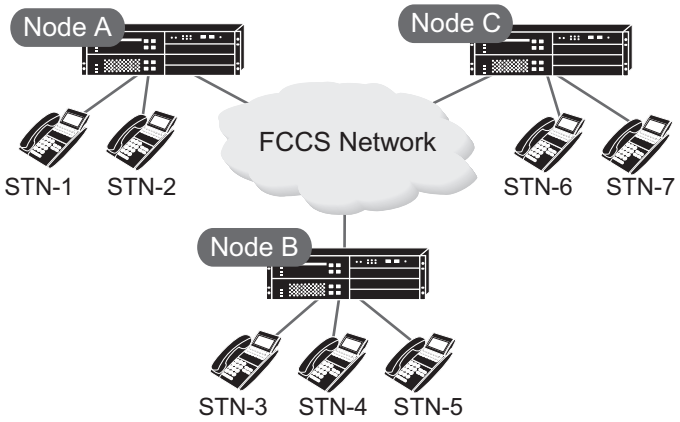
d.) If other unanswered calls transfer features are set in a Station of this feature group, those other unanswered calls transfer features have priority over the transfer of unanswered calls in a Hunt Group feature.

Priority Order	Transfer of Unanswered Calls in a Hunt Group Feature	
	Out of Service	In Service
1	CALL FORWARDING - DON'T ANSWER [C-3]	CALL FORWARDING - DON'T ANSWER [C-3]
2	-	CALL FORWARDING - BUSY LINE [C-2] **Note 5**
3	-	TRANSFER OF UNANSWERED CALLS IN A HUNT GROUP
4	CALL FORWARD BY SERVICE FEATURE CLASS [C-172]	CALL FORWARD BY SERVICE FEATURE CLASS [C-172]
5	CALL FORWARDING - DON'T ANSWER [C-3] set with the ACFO/ACFOL/ACFON command (on a tenant/route basis)	CALL FORWARDING - DON'T ANSWER [C-3] set with the ACFO/ACFOL/ACFON command (on a tenant/route basis)

Note 5: The Station with the incoming call is treated as Busy Station and the call is transfered.

FCCS Conditions

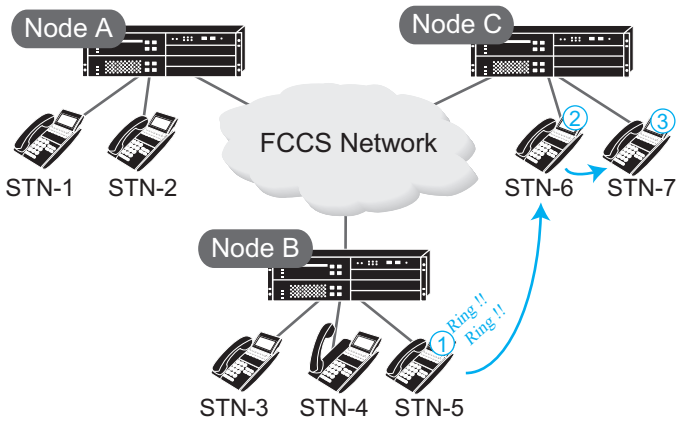
1. When activating this feature between nodes via FCCS, use only Hunt Groups assigned with Telephone Number (NDM).
2. Stations in up to four different nodes can be assigned to a Hunt Group.
   When the Hunt Group includes more than one node, the [Group by FPC] button of ASHTN must be used to set the order of the nodes in the Hunt Group. Without this node order grouping assignment, data cannot be assigned.
3. The Hunting order when the Hunt Group is created across more than one Node is as follows.



	Node A		Node B			Node C	
	STN-1	STN-2	STN-3	STN-4 Note 6	STN-5	STN-6	STN-7
Assigned Hunting order by the office data	1	2	3	4	5	6	7
Actual Hunting order	-	-	-	(Busy)	1	2	3

Note 6: This is the Station that first receives the call.

Note: The following is the hunting order in the case of the transfer of unanswered calls in a Hunt Group.



	Node A		Node B			Node C	
	STN-1	STN-2	STN-3	STN-4 Note 7	STN-5	STN-6	STN-7
Assigned Hunting order by the office data	1	2	3	4	5	6	7
Actual Hunting order	-	-	-	(Busy)	1 (Ringing)	2	3

Note 7: This is the Station that first receives the call.

4. The Incoming Call terminated between nodes via FCCS to a Station in the Hunt Group programmed by the Physical Station Number, is not hunted.

Programming

Note: The following programming supposes that Telephone Numbers have been already assigned and does not include such data settings.

STEP 1: **ASYD/ASYDN** - Assign the following System Data.
System Data 1, Index 69, Bit 5. Resultant Hunt when the Target Party is also busy after CALL FORWARDING - Busy Line [C-2].
0 = Hunt in Called Party's Hunt Group
1 = Hunt in Target Party's Hunt Group

STEP 2: **ASYDL** - (Optional) Assign the following System Data.
System Data 1, Index 688, Bit 0. Specify whether hunting is activated or not if the called PS is Out of Zone.
0 = Hunting
1 = PS Out of Zone processing

Note: Assign this data if PS is used in the Hunt Group.

STEP 3: **ASHTL/ASHTN** - Assign Hunt Group data.
UGN: User Group Number
TELN: Telephone Number in the Hunt Group. **Note 9**
SECRETARY TELN: Telephone Number of a secretarial Station **Note 8**
[Station List in Group]
TELN: Enter the Telephone Number of Stations in the Hunt Group. **Note 9**

Note: Assignment of Numbers to a Priority Hunt Group may be in any order.

Note 8: If secretarial Station is used, assign the Telephone Number of the Station used as such secretarial Station.

Note 9: Do not enter Phantom Number.

STEP 4: **ASFCN** - (Optional) Assign the following Service Feature Restriction when hunting must be denied.
SFI 181 =1 (No Hunting to next station in group)

<When activating the transfer of unanswered calls in a Priority Hunt Group> Note 10

STEP 5: **ASYD** - Assign the following System Data.
System Data 1, Index 69, Bit 0= 1 (No Recall, or execution of Call Forwarding –Don't Answer (when CF – Don't Answer is set at the transfer destination station)).
System Data 1, Index 69, Bit 1 (Send short tone when a call forwarded via CF – All Calls/Logout service is answered.)
0 = Not Required
1 = Required
System Data 2, Index 6, Bit 7 (CF-Don't Answer service when Attendant Console transfers call before called station answers)

0 = Out of service (Recall to Attendant Console)
1 = In service

STEP 6: **ASYDL/ASYDN** - Assign the following System Data.
System Data 1, Index 1395, Bit 2=1 (transfer of unanswered calls in a Priority Hunt Group in service). **Note 10**
System Data 1, Index 1396= set the no answer timer for the transfer of unanswered calls in a Hunt Group. **Note 10**
Bit 0 - 3: 1 - F= MTC
Bit 4 - 6: 0 - 7= TC
TC: Timer Class, MTC: Miscellaneous Timer Counter
TC = 0: -
1: -
2: -
3: 2 seconds
4: 30 seconds
5: -
6: -
7: -

Note: Default value for this data is HEX 00 (data is not set). If data is not set, other Timer Value of No Answer Timer (ASYD SYS1, Indexes 139, 140, 141, 247) will be used.

Note: Do not assign "0" for MTC value when TC = 3 or 4.

Note: Assigned time will be used for all the Hunt Groups to which is available (Circular, Terminal, Priority), it cannot be set separately according to the type of Hunt Group.

Note: The time will be used for the transfer of unanswered calls regardless of whether transfer of unanswered calls in a Priority Hunt Group is in service or not.

Note 10: Available from FP95-114 V4 Issue 4.0.

STEP 7: **ASFC/ASFCN** - Assign the following Service Feature Restrictions.
SFI 8=1 (Call Forwarding-Don't Answer allowed).
SFI 103 (Assignment of No Answer Timer for Blind Transfer to Station/Blind Transfer to Attendant) **Note 11**
0 = Restricted (settings in ASYD, System Data 1, Index 140)
1 = Allowed (settings in ASYD, System Data 1, Index 247)
SFI 104 (Processing when the transferred station does not answer for a predetermined period of time in Blind Transfer) **Note 12**
0 = Transferred station continuously rings
1 = The transferred call is forwarded when Call Forwarding-Don't Answer is assigned at the transferred station and no-answer timer expires

Note 11: The time set for a Station with ASYDL/ASYDN System Data 1, Index 1396 (no answer timer for the transfer of unanswered calls in a Hunt Group) will be given priority.

Note 12: Set to 1 if the terminal in the Hunt Group is the destination of BLIND TRANSFER TO STATION [B-18].

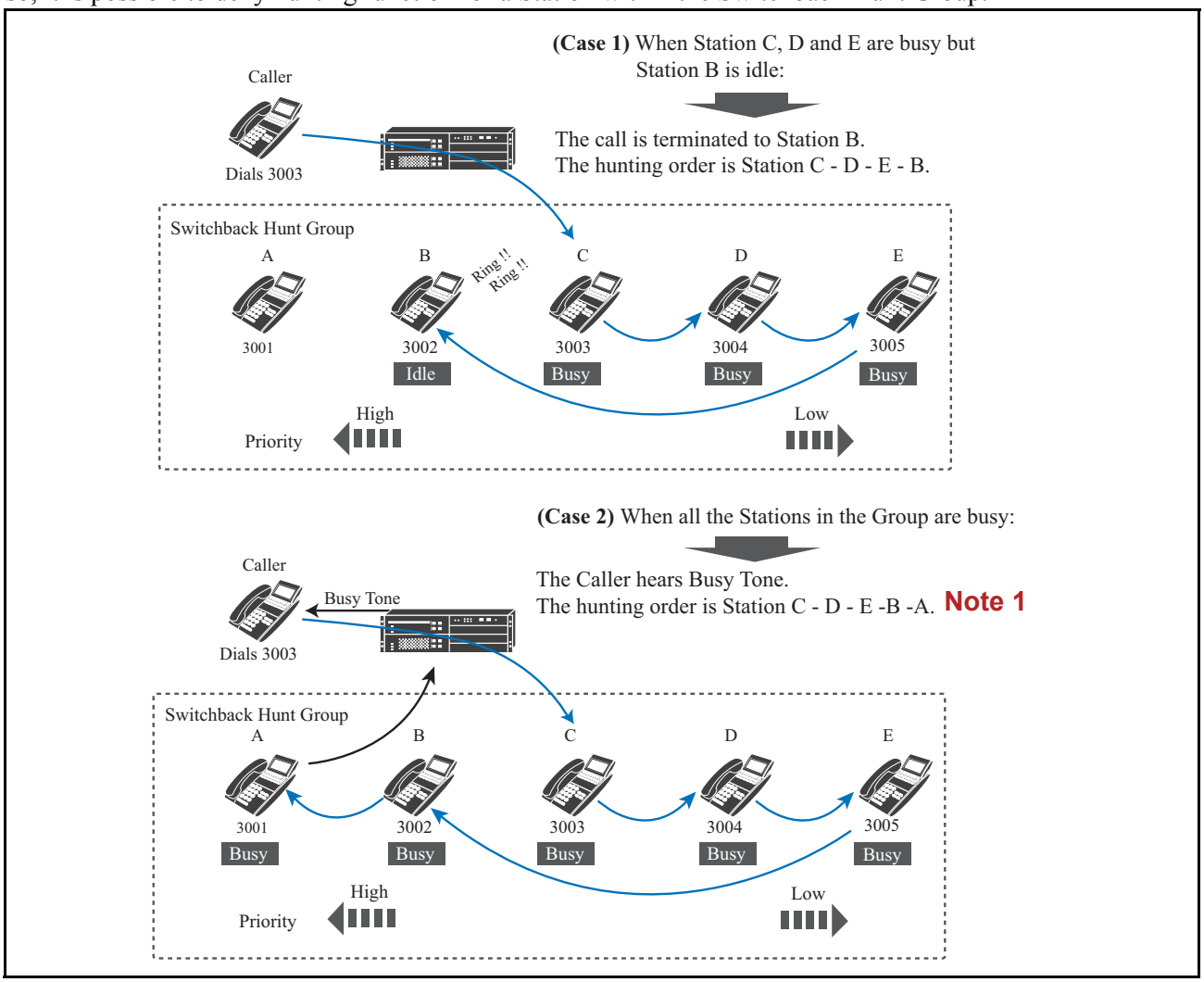
STEP 8: **ATNR** - Assign the following Tenant Restriction Class.
TRI 0=1 (Station within OGTN calls to a station within TMTN allowed).
TRI 1=1 (Station within OGTN sets Call Forwarding-All Calls / Busy Line / Don't Answer/Logout to a station within TMTN allowed).

STEP 9: **ACFR/ACFRN** - Assign the following Call Forwarding Restriction:
CCI 8=1 (Call Forwarding-Don't Answer allowed).

S-139 Station Hunting - Priority with Switchback

General Description

This feature permits a call directed to a Busy Station of a Switchback Hunt Group to be automatically forwarded to other Stations within the Hunt Group, in a programmed order. The call arrives to the busy Station of the group and is forwarded from Higher Priority Stations to Lower Priority Stations until it finds an idle Station that rings. For example, as illustrated in the following figure, if a call is placed to Station C and Station C is busy, the call is forwarded to Station D and so on, in sequence. If Station E (the Lowest Priority Station) is busy, the call is forwarded to Station B and then the call is processed from Lower Priority Station to High Priority Station. If Station A (the Highest Priority Station) is busy, the caller receives Busy Tone. However, if a secretarial Station is assigned for the group, the call is forwarded to the secretarial Station.
Also, it is possible to deny hunting function for a Station within the Switchback Hunt Group.



Note 1: If a secretarial Station is assigned for the Hunt Group, the call is forwarded to the secretarial Station. For the Secretarial Station, see STATION HUNTING - SECRETARIAL [S-8].

Note: In this feature, Busy means the Station cannot receive a call (because of reasons such as talking, receiving other call, do not disturb being set).

Operating Procedure

No manual operation is required.

Service Conditions

<Important Notices>

1. Station Number can belong to one Hunt Group only, it cannot belong to another Hunt Group including that for STATION HUNTING - CIRCULAR [S-7], STATION HUNTING - TERMINAL [S-9], STATION HUNTING - PRIORITY [S-138], STATION HUNTING - PRIORITY WITH SWITCHBACK [S-139] or UNIFORM CALL DISTRIBUTION (UCD) [U-1].
2. A station can be assigned by the Telephone Number (LDM or NDM). However, it is not possible to assign the same Telephone Number to more than one Hunt Group.

<Interactions with Other Services>

1. If the Station of a Hunt Group is in the following states or the following features are set in the Station of a Hunt Group, the following occurs.

   - **A**: The feature is activated and the call is forwarded.
   - **S**: The Station is skipped.

Feature/State	Direct Call to the Station	Call Transfered to the Station in the Hunting
CALL FORWARDING - ALL CALLS [C-5]	**A** Note 2	**S**
DO NOT DISTURB [D-11]	**S**	**S**
BUSY STATION	**S**	**S**
BUSY STATION + CALL FORWARDING - BUSY LINE [C-2]	**A** Note 2, Note 3	**S**
BUSY STATION + CALL FORWARDING - BUSY LINE [C-2] set with the ACFO/ACFOL/ACFON command (on a tenant/route basis)	**S**	**S**
LOGGED OUT STATION	**S**	**S**
LOGGED OUT STATION + LOGGED OUT IP STATION - CALL DESTINATION [L-53]	**A**	**S**
LOGGED OUT STATION + LOGGED OUT IP STATION - CALL DESTINATION [L-53] set with the ACFO/ACFOL/ACFON command (on a tenant/route basis)	**S**	**S**

Note 2: If CALL FORWARDING - OVERRIDE [C-26] is set to the forwarding destination, the call forwarding overriding from the forwarding destination to the Station of the Hunt Group will hunt if the Station of the Hunt Group is busy.

Note 3: When a call to a station in a Hunt Group is forwarded by CALL FORWARDING - BUSY LINE [C-2] and the forwarded party (also a member of another Hunt Group) is also busy, the resultant hunt target can be designated from the forwarding side Hunt Group or the forwarded side Hunt Group. This can be specified with ASYD/ASYDN SYS1 Index 69 Bit 5.

2. A Phantom Number can be assigned to the Station of a Hunt Group. However, when a caller dials the Phantom Number, hunting does not occur for the Station that receives the call.
3. Recall to a Station in the Hunt Group from the following features will not hunt:
    - CALL BACK [C-1]
    - CALL HOLD [C-6]
    - OUTGOING TRUNK QUEUING [O-2]
    - PAGING TRANSFER [P-10]
4. For STATION MESSAGE DETAIL RECORDING [S-10] the number recorded is the Station Number where the call actually terminates.
5. HOTLINE [H-1] cannot be assigned to the Station of a Hunt Group.

<Hardware/Software>

1. A call from ATTENDANT CONSOLE [A-3] will not hunt.
2. The conditions when using PS are as follows:
    a.) If a call terminates to a PS in the Hunt Group while the PS is Out of Zone or its power is OFF, whether the call is processed as PS Out of Zone or hunting is executed is determined by System Data (ASYDL, SYS 1, Index 688, b0). When using this feature in an FCCS environment, assign the same data value to the system data in each node.
    b.) When hunting is selected in the system data, the caller hears no tone until the hunted PS answers. Total length of no tone is obtained by the following.
        - Time for Detecting Out of Zone (Maximum eight seconds) × Number of Hunting
    c.) When a Desk Console operator dials the number of PS that belongs a Hunt Group, the hunting is not available if the PS is Out of Zone.
    d.) When receiving an analog PSTN incoming call while all PSs in the Hunt Group are Out of Zone, the call processing will be stopped after detecting Out of Zone of the last PS in the Hunt Group. At this time, the calling party will receive RBT when the call is a Direct In Termination call. If Automated Attendant is used, the calling party will receive ROT after No Tone. Note that the call cannot be terminated to the destination even if the PS returns to the zone during RBT.
    e.) When receiving an ISDN incoming call while all PSs in the Hunt Group are Out of Zone, hunting starts after calling up the destination. However, timeout will occur at Central Office side, resulting in ROT connection (hunting is not available).
    f.) If all Stations in the Hunt Group are busy or PS Out of Zone, the call can be forwarded a Station in the group. The station is called secretarial Station. The caller hears Busy Tone if the secretarial Station is also busy.
    g.) A PS can be assigned as a secretarial Station.
    h.) If a PS assigned as a secretarial Station is placed in Out of Zone, the caller hears Busy Tone.

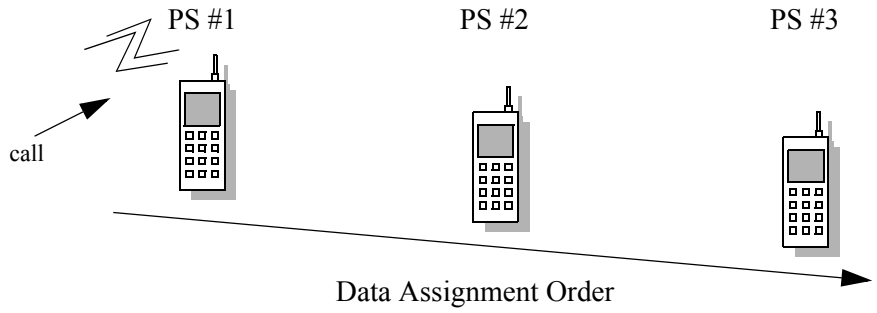
i.) A PS in the Hunt Group can set PS Out of Zone - Call Forwarding. When the PS has been set as able of hunting, the incoming call will be forwarded only if the call cannot terminate to any stations in the group. When the PS has not been set as able of hunting the call can be forwarded only if the PS is directly called.

j.) When a PS in the Hunt Group performs roaming, the PS is skipped. However, the PS is a station that receives the call directly, it will be roaming call termination.

k.) If a PS receiving a call turns to Out of Zone, this may cause a network failure. In this case, the caller hears Busy Tone.

l.) The following shows call processing when Out of Zone or busy is involved for this feature.

This example assumes: - hunting data has been assigned in the order of PS #1, #2, #3
- PS #1 first received a call.



When Hunting is selected (ASYDL, SYS1, INDEX 688, Bit 0 = 0)

PS #1	PS #2	PS #3	Processing
Out of Zone (Forwarding has been set)	Out of Zone	Out of Zone	After calling up PS #1, PS #2, PS #3, the forwarding destination is called.
Out of Zone (Forwarding has not been set)			After calling up PS #1, PS #2, PS #3, the caller receives ROT or Automatic Announcement Called PS Out of Zone.
Busy	Out of Zone	Out of Zone	After calling up PS #2 and PS #3, the caller receives Busy Tone.
Busy	Busy	Out of Zone	After calling up PS #3, the caller receives Busy Tone.

When Out of Zone Processing is selected (ASYDL, SYS 1, INDEX 688, Bit 0 = 1)

PS #1	PS #2	PS #3	Processing
Out of Zone (Forwarding has been set)	Out of Zone	Out of Zone	After calling up PS #1, the forwarding destination is called.
Out of Zone (Forwarding has not been set)			After calling up PS #1, the caller receives ROT or Automatic Announcement Called PS Out of Zone.
Busy	Out of Zone	Out of Zone	After calling up PS #2, the caller receives Busy Tone.

PS #1	PS #2	PS #3	Processing
Busy	Busy	Out of Zone	After calling up PS #3, the caller receives Busy Tone.

<Conditions Specific to This Feature>

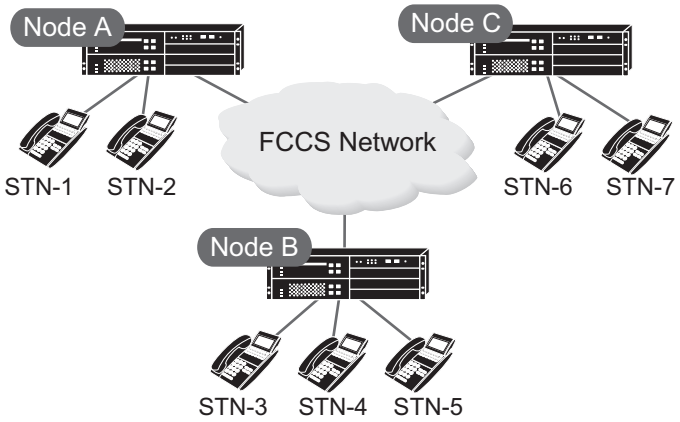
1. Transfer of unanswered calls in a Hunt Group is not available for this feature.

Note: Transfer of unanswered calls in Hunt Group is available for the following features:
- STATION HUNTING - CIRCULAR [S-7]
- STATION HUNTING - TERMINAL [S-9]
- STATION HUNTING - PRIORITY [S-138]

2. The maximum number of Stations per group is 100. The feature is usable with all Stations and there is no limit on the allowable number of Hunt Groups within a system.
3. This feature enables a Station within Hunt Group to deny hunting function for an Incoming Call originated from Station (including PS) (ASFC/ASFCN SFI 181 =1). The service conditions are as follows:
   a.) When CALL FORWARDING-DON'T ANSWER [C-5] is set, a call to the Station is forwarded according to their service condition.
   b.) When CALL FORWARDING-BUSY LINE [C-2] is set, a call to the Station is forwarded according to their service condition.
   c.) A call to the Phantom Number assigned to the Station will not hunt.
   d.) External Trunk (such as DDD (TCL = 1)/FX (TCL = 2)/WATS (TCL = 3), etc) is not available.

FCCS Conditions

1. When activating this feature between nodes via FCCS, use only Hunt Groups assigned with Telephone Number (NDM).
2. Stations in up to four different nodes can be assigned to a Hunt Group.
   When the Hunt Group includes more than one node, the [Group by FPC] button of ASHSN must be used to set the order of the nodes in the Hunt Group. Without this node order grouping assignment, data cannot be assigned.
3. The Hunting order when the Hunt Group is created across more than one Node is as follows.



	Node A		Node B			Node C	
	STN-1	STN-2	STN-3	STN-4 Note 4	STN-5	STN-6	STN-7
Assigned Hunting order by the office data	1	2	3	4	5	6	7
Actual Hunting order	6	5	4	(Busy)	1	2	3

Note 4: This is the Station that first receives the call.

4. The Incoming Call terminated between nodes via FCCS to a Station in the Hunt Group programmed by the Physical Station Number, is not hunted.

Programming

Note: The following programming supposes that Telephone Numbers have been already assigned and does not include such data settings.

STEP 1: **ASYD/ASYDN** - Assign the following System Data.
System Data 1, Index 69, Bit 5. Resultant Hunt when the Target Party is also busy after CALL FORWARDING - Busy Line [C-2].
0 = Hunt in Called Party's Hunt Group
1 = Hunt in Target Party's Hunt Group

STEP 2: **ASYDL** - (Optional) Assign the following System Data.
System Data 1, Index 688, Bit 0. Specify whether hunting is activated or not if the called PS is Out of Zone.
0 = Hunting
1 = PS Out of Zone processing

Note: Assign this data if PS is used in the Hunt Group.

STEP 3: **ASHSL/ASHSN** - Assign Hunt Group data.
UGN: User Group Number
TELN: Telephone Number in the Hunt Group.**Note 6**
SECRETARY TELN: Telephone Number of a secretarial Station **Note 5**
[Station List in Group]
TELN: Enter the Telephone Number of Stations in the Hunt Group **Note 6**

Note: Assignment of Numbers to a Switchback Hunt Group may be in any order.

Note 5: If secretarial Station is used, assign the Telephone Number of the Station used as such secretarial Station.

Note 6: Do not enter Phantom Number.

STEP 4: **ASFCN** - (Optional) Assign the following Service Feature Restriction when hunting must be denied.
SFI 181 =1 (No Hunting to next station in group)

S-140D Side Tone On/Off -D

General Description

This feature allows a Desktop terminal to switch Side Tone ON and OFF (**Note 1**). In addition, Side Tone volume value can be controlled by the system data settings with PCPro.

Note 1: Using PREFERENCE SETTING - MULTI-LINE TELEPHONE WITH PCPRO [P-90] allows PC-Pro to change the Side Tone volume value of each terminal (Available since FP95-114 V4 Issue 2.0).

Operating Procedure

1. Press the Feature key + digit key 0. (Side Tone ON to OFF)
2. Press the Feature key + digit key 0. (Side Tone OFF to ON)

Note: The state of Side Tone (ON/OFF) is displayed on the upper line of the Desktop terminal's LCD by pressing Feature key + digit key 0.

Service Conditions

1. The following table shows the desktop terminals' support status for Side Tone On/Off and volume value change:

X: Available N: Not available C: Conditionally Available

	Side Tone On/Off	Side Tone Volume Value	
		Change with ASYDL Note 2	Change with Terminal Configuration
DT800/DT700	X	X	N Note 3
DT400/DT300	X	X	N
DtermIP (Proprietary Protocol)	C Note 6	N	N
DtermIP (SIP)	N	N	X Note 4
IP terminal and Digital terminal other than above	C Note 7	N Note 5	N Note 5

Note 2: Assignment of Side Tone volume value is possible by using ASYDL, SYS1, Index 1256, Bits 4, 5 and 6 in PCPro.

Note 3: The Side Tone volume value assigned in terminal configuration is not reflected.

Note 4: For the terminal configuration, refer to Peripheral Equipment Description (Multi-Line Telephone) - Chapter 3 IP Station - Configuration Setup - DtermIP (SIP) - Audio & Visual Settings - Audio Settings - Sidetone Volume.

Note 5: When Side Tone is ON, the Side Tone volume value is fixed at -30 dB.

Note 6: Supported only for IP Enabled Dterm and DtermIP INASET.

Note 7: It depends on the terminals.

2. Once the Side Tone setting (Side Tone ON/OFF) is set at each terminal by pressing Feature key + digit key 0, the setting is maintained even if the insertion and extraction of Modular Cable, ELC circuit card initialization or system initialization is occurred.

   Note: The Side Tone setting (Side Tone ON/OFF) at each terminal is cleared as the following timing:

   Digital terminal : Change the location of the ELC circuit card.

   IP terminal : Program version of Telephony Server is updated. However, if the system data backup is performed before the update, the data is maintained.

3. When changing Side Tone volume value during system operation, perform the following to enable the new data:
   - DT800/DT700 Series: Initialize the system.
   - DT400/DT300 Series: Initialize the ELC circuit card

4. (For Digital terminal) To activate this feature, the following firmware is required to be equipped on the ELC card.

Circuit Card	Firmware
PA-16ELCJ-B	SP-3655 16ELCJB PROG-D or later
PA-16ELCN	
PA-16ELCN-A	
CH-16ELCA	

Programming

[For Side Tone volume value change]

ASYDL–SYS1, Index 1256, Bit 7: 1 = Side Tone volume is enabled.
Bits 6, 5 and 4: Assign applicable data.

System Data			Volume Value [dB]
Bit 6	Bit 5	Bit 4	
0	0	0	-54 (Minimum Value) **Note 8**
0	0	1	-48
0	1	0	-42
0	1	1	-36
1	0	0	-30
1	0	1	-24
1	1	0	-18
1	1	1	-12 (Maximum Value)

Note 8: This is default value.

S-141 Survivable Remote MGC

General Description

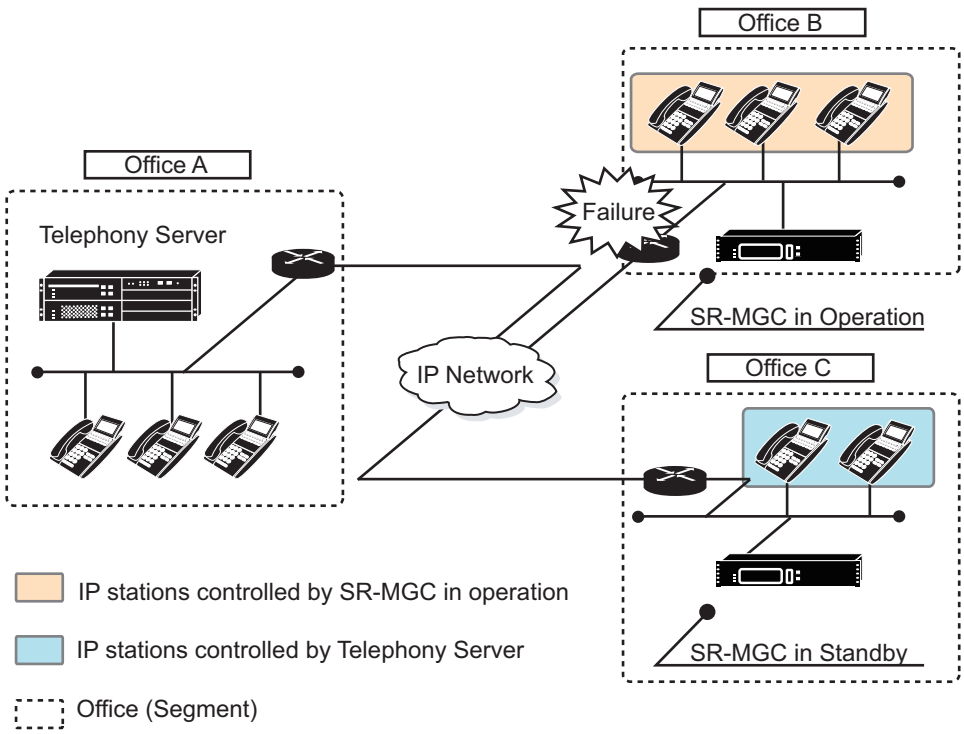
• Basic Functionality

Survivable Remote MGC (SR-MGC) is placed in a remote office to manage calls for SIP/IP terminals and trunk calls to/from PSTN via MG in the case of a network failure or the Telephony Server (hereinafter called "MGC") breakdown in the main office. In normal state, SR-MGC operates in Standby mode under monitoring of the main office MGC. Once the MGC cannot accept the registration of SIP/IP terminals in remote office, the operation is handed over to the SR-MGC. When the main office MGC recovers, the SR-MGC switches back the network equipment to the control of the MGC and it returns to Standby mode.

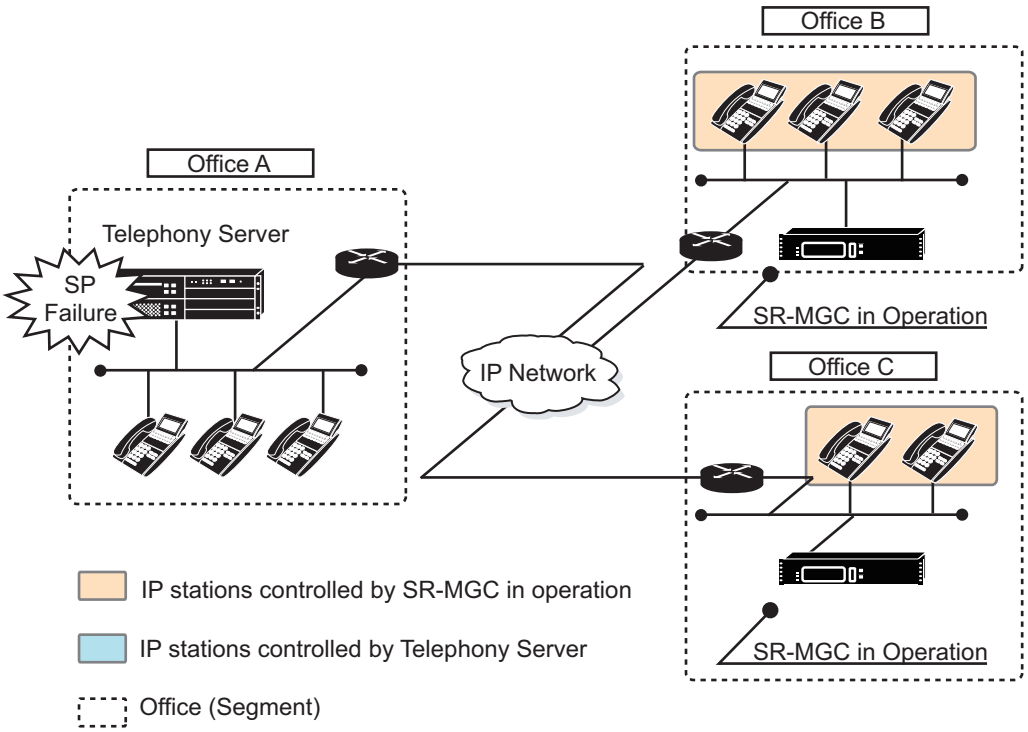
The hardware installation for the SR-MGC is described in the Peripheral Equipment Description (IP Devices). This describes Operation Procedure, Service Conditions, and Data Assignments for SR-MGC.

Note: Here we call the Telephony Server an MGC as mentioned, as well as all IP terminals and devices to be rescued by an SR-MGC just terminals for convenience.

In Case of Network Failure



In Case of SP Failure



When the terminals cannot be controlled by the Telephony Server of the main office because of a network failure or a failure in the SP part of the server, the SR-MGC in the remote office takes control of the assigned terminals. The SR-MGC periodically check the controlled terminals and, if it is possible to give back the control of the terminals to the main office Telephony Server, the SR-MGC indicates it to the terminals. The timing for giving back the control can be decided among several options depending on the data settings:

- When the communication with the main office Telephony Server is restored.
- When the terminals are reset with the visitor key or a maintenance console.
- When busy terminals exceed the period set for a forcible switch back.

In case of a failure in the SP part of the server, the automatic reboot because of a time out of health checks is performed only for the terminals controlled by the SP part of the server. Furthermore, during the SP failure, the registration of terminals in the server is restricted only for SP controlled terminals and such terminals are registered to the SR-MGC.

• Flexible Network Re-routing

This feature is useful when the Telephony Server or network failure occurs, and one of the nodes in a network is switched over to operation by SR-MGC, unable to connect to the opposite node via IP network. By assigning forwarding data to a trunk line in advance between the nodes, re-routing is automatically activated via a public network. Thus, this feature provides a connectivity equal to that could be attained by the Telephony Server via SR-MGC installed at each node.
For more detailed information about the feature, refer to "Flexible Network Re-routing [F-41]" in this manual.

• Visitor Key Procedures

When a terminal whose Visitor key has been assigned is registered in an SR-MGC as the backup MGC, the terminal is capable of switchback to primary MGC (set on the terminal side) by using the Visitor key as follows:

Note: If the system data (ASYDL, SYS1, Index 846, Bit 4 = 1) is assigned, the terminal registered in an SR-MGC cannot perform terminal reset by the Visitor key.

- When Visitor key is set to Function key

  [1] Make sure an appropriate Function key lights red and SR-MGC controls the terminal.

  [2] Push the Function key.

  [3] Make sure the Function key light is off and Primary MGC (set on the terminal side) controls the terminal.

- When Visitor key is set to the Soft key

  [1] Make sure VISIT display on an appropriate Soft key is flashing and SR-MGC controls the terminal.

  [2] Push the Soft key.

  [3] Make sure the Soft key light is on and Primary MGC (set on the terminal side) controls the terminal.

Service Conditions

Service Conditions are described in the following, consisting of several parts.

<Important Notices>

1. SR-MGC series is capable of rescuing the following equipment, depending on the system conditions.

   × : Available
   N : Not Available
   – : Not Applicable

Type of Equipment		Conditions				
IP Station/ Device	Protocol mode	Port Note 2	FCCS Cluster Note 3	No Encryption	Encryption	Detailed Information of Equipment
DT800 Series	SIP	Note 2	×	×	× Note 4	DT830/DT830G/DT820
DT700 Series	SIP		×	×	× Note 4	DT710/DT730/DT730G/DT750
DtermIP	Proprietary Protocol		×	×	× Note 4	DtermIP (Proprietary Protocol) IP Enabled Dterm DtermIP INASET
	SIP		×	×	× Note 4 Note 5 Note 6	DtermIP(SIP)
Soft Phone Note 1	Proprietary Protocol		×	×	–	DtermSP30 (Proprietary Protocol)
	SIP		×	×	SP350: × Note 4 DtermSP30: –	Soft Client SP350, DtermSP30 (SIP)
WLAN Handset (MH250)	SIP		N	×	–	
SP Controlled Standard SIP Note 7	SIP	–	N	N	–	SP Controlled Standard SIP Voice Terminal SP Controlled Standard SIP Video Terminal
MG(BRI)	SIP	4	×	×	× Note 4	MG(BRI) Card [SCA-2BRIA] MG(BRI) Box [MG-2BRIA]
MG(PRI)	Proprietary Protocol	T1: 24 E1: 30	×	×	–	MG(PRI) Card [SCA-24PRIA] MG(PRI) Card [SCA-30PRIA] MG(PRI) Box [MG-24PRIA] MG(PRI) Box [MG-30PRIA]
	SIP		×	×	× Note 4	MG(PRI) Card [SCA-24PRIA] MG(PRI) Card [SCA-30PRIA] MG(PRI) Box [MG-24PRIA] MG(PRI) Box [MG-30PRIA]
MG-T1(SIP)	SIP	24	×	×	× Note 4	MG-T1(SIP) Card [SCA-24DTIA]

Type of Equipment		Conditions				
IP Station/ Device	Protocol mode	Port Note 2	FCCS Cluster Note 3	No Encryption	Encryption	Detailed Information of Equipment
MG(SIP)	Proprietary Protocol	32	×	×	–	MG-SIP16 MG-SIP16 Card [SCA-16SIP-MGA] MG-SIP16 Box [MG-16SIPMGA] MG-SIP16 Card [SCA-16SIPMG(US)]
		96	×	×	–	MG-SIP96
		128	×	×	× Note 9	MG-SIP128 [MG-128SIPMGL-A] MG-SIP128 [MG-128SIPMGG-B] MG-SIP128 [MG-128SIPMGG] MG-SIP128 [MG-128SIPMGJ-B] MG-SIP128 [MG-128SIPMGJ]
		1–254	×	×	× Note 9	Software-Based MG-SIP
MC & MG -COT	Proprietary Protocol	8	×	×	–	MCMG Card [SCA-4LC2COTA] MCMG Box [MG-4LC2COTA]
MG -COT	Proprietary Protocol	8	×	×	–	MG-COT Card [SCA-6COTA] MG-COT Card [SCA-6COTB] MG-COT Card [SCA-6COTC]
	SIP	8	×	×	× Note 4	MG-COT Card [SCA-6COTA] MG-COT Card [SCA-6COTB] MG-COT Card [SCA-6COTC]
Analog MC	Proprietary Protocol	4	×	×	–	Analog 2MC
	SIP	8	×	×	× Note 4	8LC Card [SCA-8LCA] 8LC Card [SCA-8LCA-EMEA] 8LC Card [SCA-8LCC] 8LC Card [SCA-8LCC-EMEA]
VS32	Proprietary Protocol	32	×	×	–	VS-32 Card [SCA-VS32VA] VS-32 Box [MG-VS32VA] Software-based VS32
	SIP	32	×	×	× Note 4	VS-32 Card [SCA-VS32VA] VS-32 Box [MG-VS32VA]
UG50 (DLC)	Proprietary Protocol	8	× Note 8	×	–	GCD-8DLCA
		16				GCD-8DLCA+GPZ-8DLCB GCD-16DLCA
UG50 (LC)	Proprietary Protocol/SIP	4	× Note 8	×	–	CD-4LCA
		8				CD-4LCA+PZ-4LCA CD-8LCA
		12				CD-4LCA+PZ-8LCE CD-8LCA+PZ-4LCA
		16				CD-8LCA+PZ-8LCE
UG50 (PRT 1.5M)	Proprietary Protocol/SIP	32	× Note 8	×	–	GCD-PRTA

Type of Equipment		Conditions				
IP Station/ Device	Protocol mode	Port Note 2	FCCS Cluster Note 3	No Encryption	Encryption	Detailed Information of Equipment
UG50 (PRT 2M)	Proprietary Protocol/SIP	32	× Note 8	×	–	GCD-PRTA
UG50 (COT-Type1)	Proprietary Protocol	4	× Note 8	×	–	GCD-4COTA/B/C
		8				GCD-4COTA/B/C+GPZ-4COTE/F/G
UG50 (COT-Type2)	Proprietary Protocol/SIP	4	× Note 8	×	–	GCD-4COTA/B/C
		8				GCD-4COTA/B/C+GPZ-4COTE/F/G
UG50 (PGT-Type1)	Proprietary Protocol	4	–	×	–	GCD-PGTA
UG50 (PGT-Type2)	Proprietary Protocol/SIP	4	–	×	–	GCD-PGTA
UG50-IPG (Digital)	Proprietary Protocol	8	N	×	–	GCD-8DLCA
		16				GCD-8DLCA+GPZ-8DLCB GCD-16DLCA
UG50-IPG (Analog)	Proprietary Protocol	4	N	×	–	CD-4LCA
		8				CD-4LCA+PZ-4LCA CD-8LCA
		12				CD-4LCA+PZ-8LCE CD-8LCA+PZ-4LCA
		16				CD-8LCA+PZ-8LCE
UG50-2MC	Proprietary Protocol	4	×	×	–	CD-4LCA CD-8LCA
UG50-8LC	SIP	8	×	×	–	CD-8LCA
UG50-24PRIA	Proprietary Protocol/SIP	24	×	×	–	GCD-PRTA
UG50-30PRIA	Proprietary Protocol/SIP	30	×	×	–	GCD-PRTA
UG50-4LC2COTA (COT)	Proprietary Protocol	8	×	×	–	CD-4LCA CD-8LCA GCD-4COTA
UG50-4LC2COTA (PGT)	Proprietary Protocol	8	–	×	–	GCD-8LCA GCD-PGTA
UG50-6COT	Proprietary Protocol/SIP	8	×	×	–	GCD-4COTA/B/C GPZ-4COTE/F/G

Note 1: Soft Phone described here indicates one that is not being used with Softphone Wired Phone Handset [S-147].

Note 2: The maximum number of ports/stations that SR-MGC can rescue varies depending on the SR-MGC series.

Capacity of Each SR-MGC Port

SR-MGC Series	Number of Ports to be Rescued
SR-MGC50P PROG-Vxx	50
SR-MGC100P PROG-Vxx	100
SR-MGC200P PROG-Vxx	200
SR-MGC500P PROG-Vxx	500
SR-MGC1000P PROG-Vxx	1,000
SR-MGC2000P PROG-Vxx	2,000

** The port/station capacity of this software is fixed. To expand port/station capacity of SR-MGC, it is necessary to purchase new software (CF Card).*

** Use "Number of Ports to be Rescued" for Stations, MC, MG and VS32.*

Note 3: This item represents "SR-MGC with FCCS Cluster" feature. For the information about this feature, refer to "SR-MGC with FCCS Cluster" in this manual.

Note 4: SR-MGC supports VoIP Encryption [V-27] as in the case of MGC.

Note 5: SP350 can be rescued during failure when encryption is available.

Note 6: DtermSP30 is not applicable for this feature.

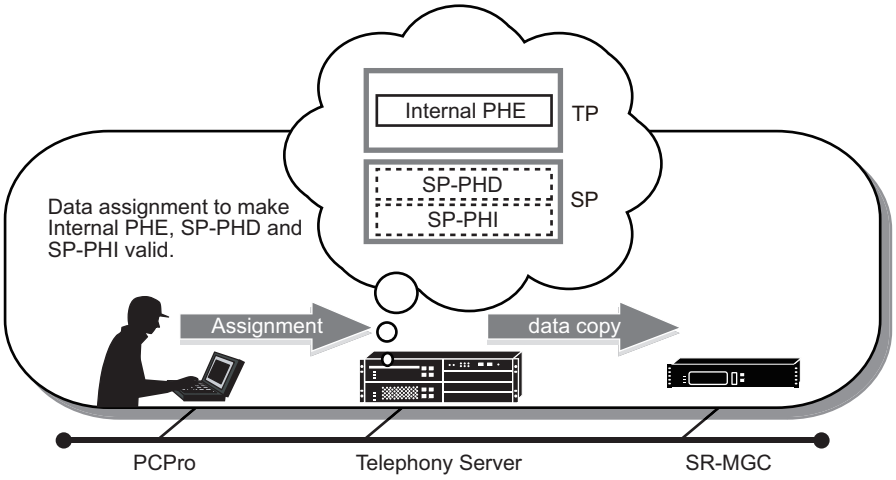
Note 7: Some types of SP Controlled SIP terminals cannot be retrieved. Contact your authorized NEC Sales Representative for more information about SP Controlled SIP terminals.

Note 8: Available from FP95-114 V4 Issue 2.0.

Note 9: Equipment available for RTP encryption are: MG-128SIPMGL-A, MG-128SIPMGJ-B/MG-128SIPMGJ, and Software-based MG (SIP). Available from FP95-114 V4 Issue 2.0.

2. The software issue of SR-MGC must be same or higher than that of the Telephony Server.
3. Minor Version Upgrade of SR-MGC can be executed remotely with the CMNT command. For details on Remote Version Upgrade, refer to "SR-MGC(E) [SN8178 MGCEV-B]/SR-MGC(E) [SN8178 MGCEV-A]" - "Remote Version Upgrade" in the Peripheral Equipment Description (IP Devices).
4. LAN1 port of the Telephony Server is used as the interface for the communication (Health Check) between SR-MGC and MGC, so communication failure/recovery is detected based on the LAN1 condition.

5. SR-MGC retrieves PH functions from MGC. Therefore, PHs used by SR-MGC need to be valid in MGC. As the figure below shows, data assignment on MGC is necessary to make Internal PHE, SP-PHD and SP-PHI valid.



Note: SR-MGC supports VoIP Encryption [V-27]. The devices/stations can continue to execute the encryption feature, even if they are rescued by SR-MGC.

6. Terminals' switchback (**Note 1**) from SR-MGC can be determined by using the ASYDL SYS1 Index 852 Bit 5.

Note 1: Switchback means to return the registration destination of the terminals to MGC from SR-MGC after communication between SR-MGC and MGC (LAN1 of the Telephony Server) is recovered.

A.) When **Automatic Switchback** (ASYDL SYS1 Index 852 Bit 5=1) is selected

<Factors of switchback>

(a) When "1" is selected in Changeover/Recover Mode in the CMNT command, SR-MGC makes the terminal register to MGC when the SR-MGC detects the recovery of the MGC.

(b) When "2" is selected in Changeover/Recover Mode in the CMNT command, SR-MGC makes the terminal register to MGC when the SR-MGC detects the normality of communication between SR-MGC and MGC, MGC and Pilot terminal. **Note 1**, **Note 2**, **Note 3**

Note 1: WLAN Handset (MH250) cannot be specified as a pilot terminal. If no terminal registers to SR-MGC except WLAN Handset (MH250), SR-MGC makes the terminal register to MGC when the SR-MGC detects the recovery of the MGC, as in the case of Mode 1 (MGC does not execute "ping" to check the connectivity).

Note 2: Pilot terminal is specified as follows:

LOC-ID Selection / Terminal Selection	ASYDL SYS1 Index 846 Bit 1=0	ASYDL SYS1 Index 846 Bit 1=1
ASYDL SYS1 Index 846 Bit 0=0	Search terminals registered in SR-MGC, and select a terminal accommodated in the smallest number of accommodated location.	Search terminals registered in SR-MGC by LOC-ID, and select a terminal accommodated in the smallest number of accommodated location.
ASYDL SYS1 Index 846 Bit 0=1	Search terminals connected to MGs registered in SR-MGC, and select a terminal accommodated in the smallest number of accommodated location.	Search terminals connected to MGs registered in SR-MGC by LOC-ID, and select a terminal accommodated in the smallest number of accommodated location.

Note 3: Put both of LAN1 and LAN2 IP addresses in the same segment in order to make this mode available.

<Terminal conditions when switchback is executed>

(a) SR-MGC checks the status of the terminals, if its status is released, the SR-MGC switches the terminal back to MGC. SR-MGC switches multiple-channel terminal back to MGC when all the channels are released.

(b) In multiple-channel terminals, SR-MGC switches the terminal back to MGC when all the channels in the terminal are released.

(c) If the terminals in busy exceed a certain period (determined by ASYDL SYS1 Index 847 **Note 4**, **Note 5**, **Note 7**), SR-MGC forcibly switches them back to MGC. In IP terminal except for Soft Phone and SIP Multiple Line terminal, from 3 minutes before forcible switchback, alarm tone that indicating terminal reset will be executed as soon as transmitted to the terminals per minute. **Note 8**

Note 4: When ASYDL SYS1 Index 847 Bits0-3 are set to "F", forcible switchback is not executed while the active calls remain.

Note 5: An SR-MGC may not execute forcible switchback to maintain terminal's states:

- An SR-MGC maintains following PHS's calls: an originating call (when a PHS user is hearing ringback tone); an ongoing call between PHSes under control of an IPBS.
- An SR-MGC maintains following WLAN Handset (MH Series)'s calls: an originating call (when a WLAN Handset user is hearing ringback tone); an ongoing call between WLAN Handsets.
- An SR-MGC maintains an ongoing call between a WLAN Handset and a PHS under control of an IPBS.
- An SR-MGC maintains the state of MC part on an MC&MG-COT if all of three conditions (device, firmware version, state) are true.

Device	Firmware Version	State
MG-4LC2COTA	Earlier than SP-3778 MCMG PROG-A Issue 13	Having an off-hook terminal. **Note 6**
	SP-3778 MCMG PROG-A Issue 13 or later	Having an ongoing call.
SCA-4LC2COTA	SP-3778 MCMG PROG-A Issue 1 or later	Having an ongoing call.
UG50 (COT-TYPE1)	Any version	Having an ongoing call.
UG50 (COT-TYPE2)	Any version	Having an ongoing call.
UG50-4LC2COTA (COT)	Any version	Having an ongoing call.
UG50-6COT	Any version	Having an ongoing call.
UG50 (PGT-TYPE1)	UG50 software version 2.00 or later	Having an ongoing call.
UG50 (PGT-TYPE2)	UG50 software version 2.00 or later	Having an ongoing call.
UG50-4LC2COTA (PGT)	UG50 software version 2.00 or later	Having an ongoing call.

Note 6: "Off-hook terminal" is a terminal which you are manipulating for a conversation including dialing, hearing ringback tone and conversation itself.

Note 7: When the switchback target product is Analog 2MC or VS32, and all channels are not vacant, the forcible switchback may not be executed immediately even though the set time of the forcible switchback passes. It requires one minute at the maximum from the set time of the forcible switch back to starting of its execution.

Note 8: Alarm before forcible switchback (ASYDL, SYS1, Index 847, Bit 4=1) supports IP terminal except for Soft Phone and SIP Multiple Line terminal only.

B.) When **Manual Switchback** (ASYDL SYS1 Index 852 Bit 5=0) is selected

<Factors of switchback>

(a) After the MGC is recovered, the terminal can be switched back by using the CMNT command (Condition in the Setting tab). Also, assigning Visitor key to terminals provides the terminal users with manual switchback.

<Terminal conditions when switchback is executed>

(b) Manual switchover by the CMNT command makes the terminals register to MGC regardless of its status (busy/unbusy).

7. After the office data assignment, you need to set an SR-MGC's IP address on each terminal. Set the SR-MGC's IP address as the DRS of the terminal.

- Set IP address of SR-MGC for the following:

  IP terminals including Soft Phone (Proprietary Protocol mode)
  Analog 2MC
  VS32 (Proprietary Protocol mode)
  MG(PRI) (Proprietary Protocol mode)
  MC&MG-COT
  MG(SIP)
  MG-COT (Proprietary Protocol mode)

- Set IP address (or URL) of SR-MGC for the following:

  SIP Multiple Line terminal
  Soft Phone (SIP mode)
  WLAN Handset (MH Series)
  8LC Card
  MG(BRI) Card/Box
  MG(PRI) (SIP mode)
  VS32(Dual) (SIP mode)
  MG-COT (SIP mode)
  MG-T1(SIP)

Note: Protocol mode (Proprietary Protocol mode/SIP mode) is added at the end of each IP device/station name whose required IP address varies depending on its protocol mode.

8. The Operating mode (**Note 9**) of SR-MGC will be greatly influenced by the communication status between MGC and SR-MGC and the settings of the CMNT command.

Note 9: SR-MGC has the following two modes:
Standby Mode: IP devices/stations cannot register to SR-MGC
Operating Mode: IP devices/stations can register to SR-MGC
If IP devices/stations register to SR-MGC in Standby Mode, each equipment operates differently.

The following explains the status change of SR-MGC by the CMNT command.

【Explanatory Note】

[ ]/( ) : Indicates Status of SR-MGC

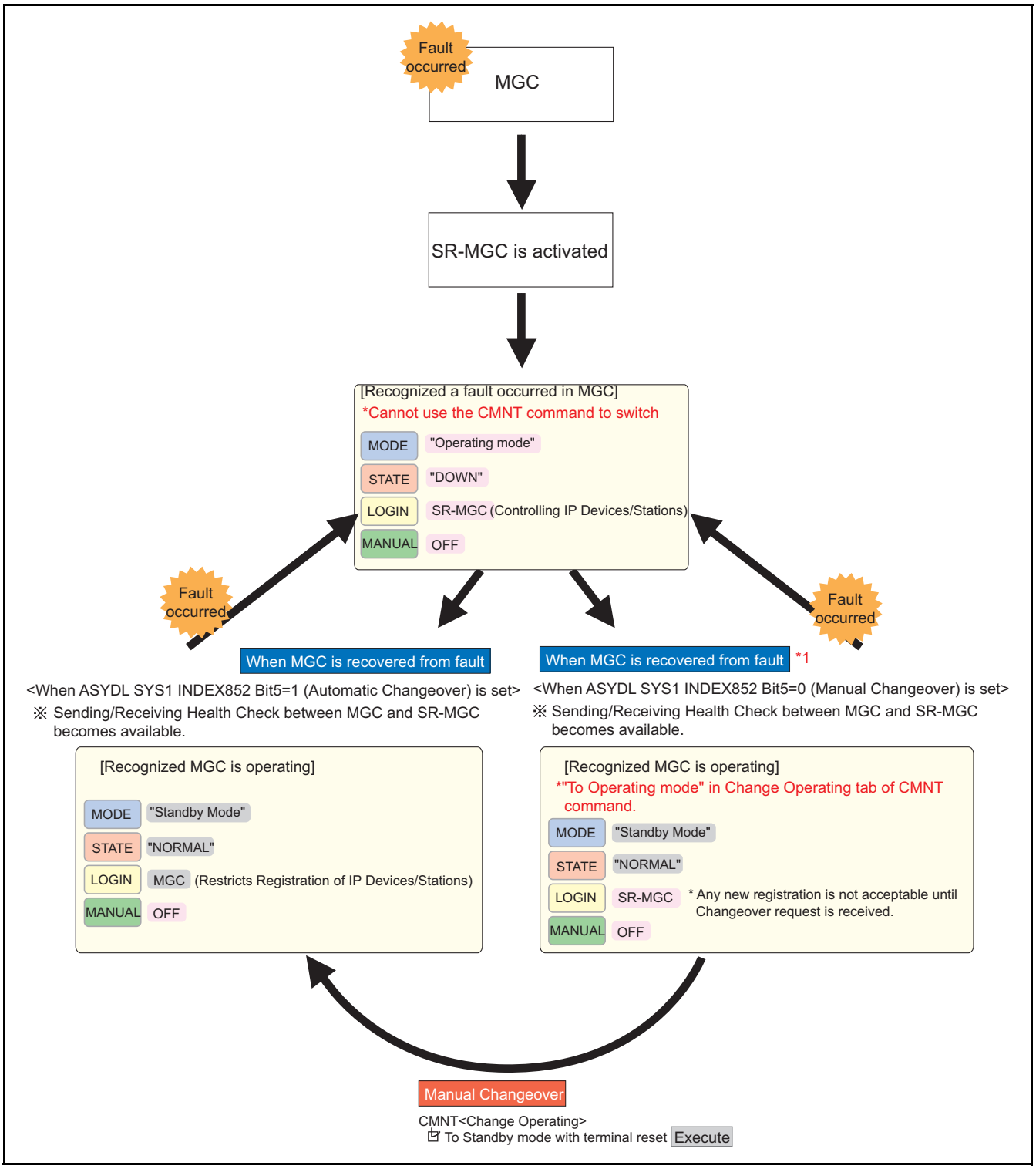
MODE
Indicates SR-MGC allows/restricts the registration of IP devices/stations.
"Standby Mode" : IP devices/stations cannot register to SR-MGC
"Operating Mode" : IP devices/stations can register to SR-MGC

STATE
Status of MGC
"NORMAL" : MGC is normally operating
"DOWN" : Fault occurred in MGC

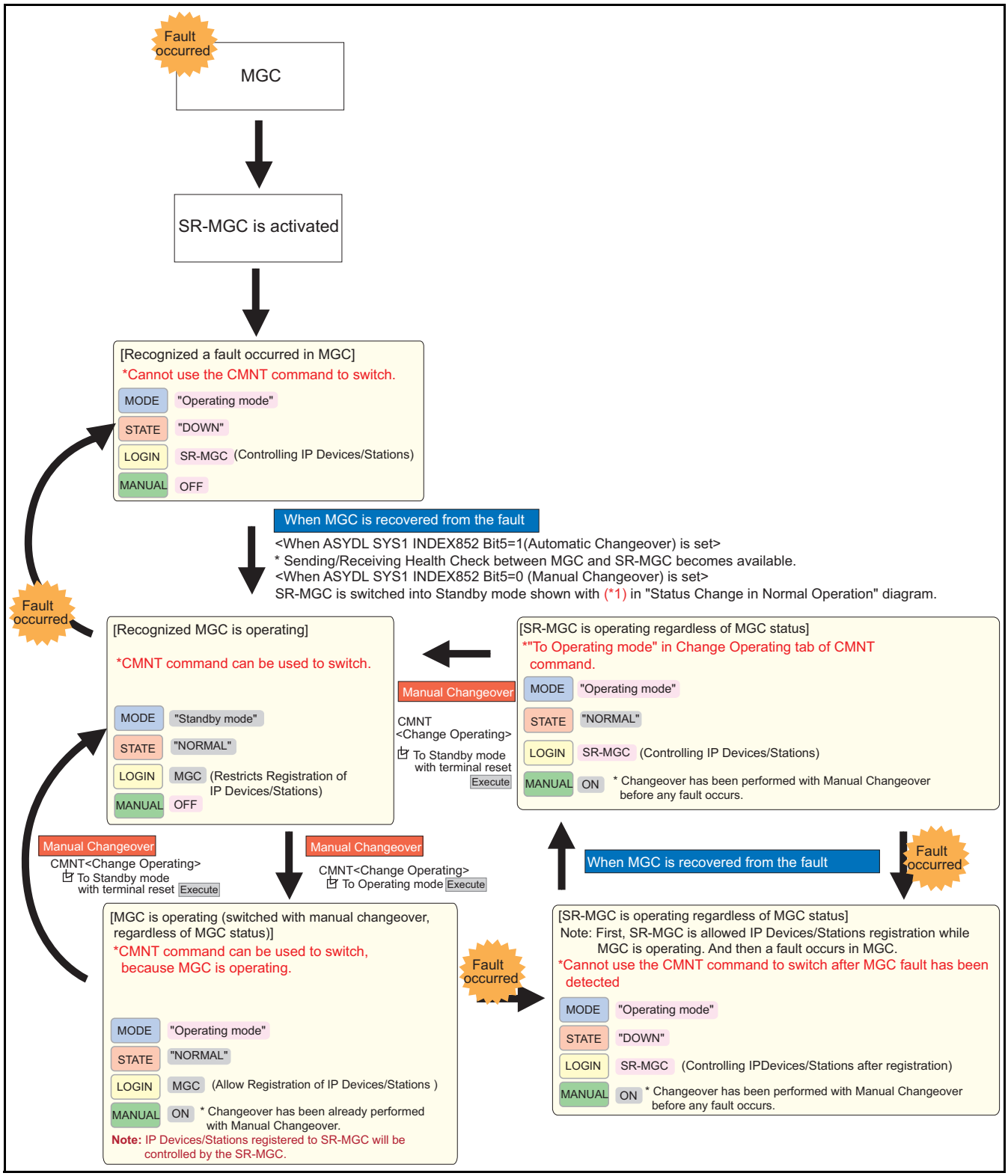
LOGIN
Destination where IP devices/stations register
MGC : Registered in MGC
SR-MGC : Registered in SR-MGC

MANUAL
Status of Operating Change by Manual Changeover (CMNT command)
ON : "To Operating mode" is performed
OFF : "To Operating mode" is not performed

- Status Change in Normal Operation



- Status Change in Test mode (Normally, SR-MGC's status is not changed in the following



9. The following table shows how IP equipment (a terminal or device) operates when the equipment switches its registration destination from an MGC to an SR-MGC or vice versa, which depends on the equipment kind and its state:

Note: The table applies when an opposite terminal is a station, and the station does not stop sending or receiving RTP packets nor abandon an originating or terminating call.

Note: If forcible switchback (ASYDL SYS1 Index 847 Bits 0-3) is enabled, an SR-MGC forcibly switches equipment back to the Telephony Server after the predetermined period of time expires regardless of the following. About operations of IP equipment regarding Call Retention, refer to "Operation of IP Equipment/Terminals When a Fault Occurs During a Conversation" in INTRODUCTION of Peripheral Equipment Description.

a.) If a communication failure occurs between IP equipment and an MGC while the MGC is operating (Switch from an MGC to an SR-MGC):

Equipment State / Equipment Kind	During Conversation	During Ringback Tone	During Ringing Tone
IP terminal Soft Phone (Proprietary Protocol)	The terminal maintains an ongoing call; after you disconnect the call by on-hook, the terminal resets itself to register to an SR-MGC.	After ringback tone lasts for two or three minutes, an IP terminal resets itself to register to an SR-MGC.	After ringing tone lasts for two or three minutes, an IP terminal resets itself to register to an SR-MGC.
SIP Multiple Line terminal		After ringback tone lasts for two to six minutes, a SIP Multiple Line terminal resets itself to register to an SR-MGC.	After ringing tone lasts for two to six minutes, a SIP terminal resets itself to register to an SR-MGC.
Soft Phone (SIP)		After ringback tone lasts for a while, a Soft Phone (SIP) discards the call to register to an SR-MGC.	
WLAN Handset (MH250)		After ringback tone lasts for a minute or two, a WLAN Handset resets itself to register to an SR-MGC.	After ringing tone lasts for five minutes, a WLAN Handset resets itself to register to an SR-MGC.

Equipment State / Equipment Kind	During Conversation	During Ringback Tone	During Ringing Tone
MC&MG-COT: **- SCA-4LC2COTA** **- MG-4LC2COTA**	The device maintains an ongoing call; after you disconnect the call by on-hook, the device resets itself to register to an SR-MGC.	After ringback tone lasts for two to four minutes, an MC&MG-COT resets itself to register to an SR-MGC.	After ringing tone lasts for two to four minutes, the device resets itself to register to an SR-MGC.
Analog 2MC	After a few minutes, an Analog 2MC resets itself to register to an SR-MGC.	After ringback tone lasts for two to four minutes, an Analog 2MC resets itself to register to an SR-MGC.	
8LC Card	An 8LC Card maintains an ongoing call; after you disconnect the call by on-hook, the 8LC Card resets itself to register to an SR-MGC.	After ringback tone lasts for a while, an 8LC Card discards the call and resets itself to register to an SR-MGC.	
IPG Digital Note 1	After a minute or two, the call is disconnected. An IPG resets itself to register to an SR-MGC in a few minutes.	After ringback tone lasts for about a few minutes, an IPG resets itself to register to an SR-MGC. (DT connection when off-hook)	After ringing tone lasts for two to three minutes, the device resets itself to register to an SR-MGC.
IPG Analog	After two or three minutes, the call is disconnected by an IPG reset. It will be registered to a SR-MGC by on-hook in a few minutes.	After ringback tone lasts for about a few minutes, an IPG resets itself. It will be registered to an SR-MGC by on-hook.	
UG50 (Multi-slot mode) • UG50 (DLC)	After a minute or two, the call is disconnected. the UG50 resets itself to register to an SR-MGC in a few minutes.	After ringback tone lasts for about a few minutes, the UG50 resets itself to register to an SR-MGC. (DT connection when off-hook)	
UG50 (Multi-slot mode) • UG50 (LC) [Proprietary] • UG50 (LC) [SIP]	The UG50 maintains an ongoing call; after you disconnect the call by on-hook, the UG50 resets itself to register to an SR-MGC.	After ringback tone lasts for a while, the UG50 discards the call and resets itself to register to an SR-MGC.	After ringing tone lasts for two to four minutes, the UG50 resets itself to register to an SR-MGC.
UG50 (IPG mode) • UG50-IPG (Digital)	After a minute or two, the call is disconnected. the UG50 resets itself to register to an SR-MGC in a few minutes.	After ringback tone lasts for about a few minutes, the UG50 resets itself to register to an SR-MGC. (DT connection when off-hook)	After ringing tone lasts for two to three minutes, the UG50 resets itself to register to an SR-MGC.

Equipment State / Equipment Kind	During Conversation	During Ringback Tone	During Ringing Tone
UG50 (IPG mode) • UG50-IPG (Analog)	The UG50 maintains an ongoing call; after you disconnect the call by on-hook, the UG50 resets itself to register to an SR-MGC.	After ringback tone lasts for a while, the UG50 discards the call and resets itself to register to an SR-MGC.	After ringing tone lasts for two to four minutes, the UG50 resets itself to register to an SR-MGC.
UG50 (Retrofit mode) • UG50-2MC • UG50-8LC • UG50-4LC2COTA (COT) • UG50-4LC2COTA (PGT)			

Note 1: IPG Analog resets itself to perform re-registration after two to three minutes, regardless of the set time before forcible switchback is executed (ASYDL SYS1 Index 847 Bit 0~3).

b.) If the system recovers from a communication failure between IP equipment and an MGC: (Switchback from SR-MGC to MGC) with the following system data set:

ASYDL SYS1 Index 852 Bit 5=1: Automatic Switchback

Equipment State / Equipment Kind	During Conversation	During Ringback Tone	During Ringing Tone
IP terminal **Soft Phone (Proprietary Protocol)**	The terminal maintains an ongoing call; after you disconnect the call, it takes a few minutes for the terminal to register to the Telephony Server.	**If you abandon the call before connected:** After ringback tone lasts for a few minutes, the terminal registers to the Telephony Server. **If the call connects to a peer:** The terminal maintains the ongoing call; after you disconnect the call, it takes a few minutes for the terminal to register to the Telephony Server.	**If you abandon the call without answering:** Ringing tone lasts; if the calling party disconnects the call by on-hook, it takes a few minutes for the terminal/device to register to the Telephony Server. **If you answer the call:** The terminal/device maintains the ongoing call; after you disconnect the call, it takes a few minutes for the terminal/device to register to the Telephony Server.
SIP Multiple Line terminal			
Soft Phone (SIP)			
WLAN Handset (MH250)			
MC&MG-COT: **- SCA-4LC2COTA** **- MG-4LC2COTA**	The device maintains an ongoing call; after you disconnect the call, the device registers to the Telephony Server.	Ringback tone lasts until you go on-hook; the device registers to the Telephony Server.	
Analog 2MC			
8LC Card			

ASYDL SYS1 Index 852 Bit 5=1: Automatic Switchback

Equipment State / Equipment Kind	During Conversation	During Ringback Tone	During Ringing Tone
IPG Digital	IPG maintains a call until the set forcible switchback time executes (5mins x (1~14)). IPG resets itself and will be registered to the Telephony Server by on-hook.	Ringback tone lasts until the set forcible switchback time executes (5mins x (1~14)). IPG resets itself to register to the Telephony Server. (DT connection when off-hook)	Ringing tone lasts until the set forcible switchback time executes (5mins x (1~14)). IPG resets itself to register to the Telephony Server.
IPG Analog Note 1	After two or three minutes, the call is disconnected by an IPG reset. It will be registered to the Telephony Server by on-hook in a few minutes.	After ringback tone lasts for about a few minutes, IPG resets itself. It will be registered to the Telephony Server by on-hook.	After ringing tone lasts for two to three minutes, the device resets itself to register to the Telephony Server.
UG50 (Multi-slot mode) • UG50 (DLC)	The UG50 maintains a call until the set forcible switchback time executes (5mins x (1~14)). The UG50 resets itself and will be registered to the Telephony Server by on-hook.	Ringback tone lasts until the set forcible switchback time executes (5mins x (1~14)). The UG50 resets itself to register to the Telephony Server. (DT connection when off-hook)	Ringing tone lasts until the set forcible switchback time executes (5mins x (1~14)). The UG50 resets itself to register to the Telephony Server.
UG50 (Multi-slot mode) • UG50 (LC) [Proprietary] • UG50 (LC) [SIP]	The device maintains an ongoing call; after you disconnect the call, the device registers to the Telephony Server.	Ringback tone lasts until you go on-hook; the device registers to the Telephony Server.	**If you abandon the call without answering:** Ringing tone lasts; if the calling party disconnects the call by on-hook, it takes a few minutes for the terminal/device to register to the Telephony Server. **If you answer the call:** The terminal/device maintains the ongoing call; after you disconnect the call, it takes a few minutes for the terminal/device to register to the Telephony Server.
UG50 (IPG mode) • UG50-IPG (Digital)	UG50 maintains a call until the set forcible switchback time executes (5mins x (1~14)). The UG50 resets itself and will be registered to the Telephony Server by on-hook.	Ringback tone lasts until the set forcible switchback time executes (5mins x (1~14)). The UG50 resets itself to register to the Telephony Server. (DT connection when off-hook)	Ringing tone lasts until the set forcible switchback time executes (5mins x (1~14)). The UG50 resets itself to register to the Telephony Server.

ASYDL SYS1 Index 852 Bit 5=1: Automatic Switchback

Equipment State / Equipment Kind	During Conversation	During Ringback Tone	During Ringing Tone
UG50 (IPG mode) • UG50-IPG (Analog)	The UG50 maintains an ongoing call; after you disconnect the call, the UG50 registers to the Telephony Server.	Ringback tone lasts until you go on-hook; the UG50 registers to the Telephony Server.	**If you abandon the call without answering:** Ringing tone lasts; if the calling party disconnects the call by on-hook, it takes a few minutes for the UG50 to register to the Telephony Server. **If you answer the call:** The UG50 maintains the ongoing call; after you disconnect the call, it takes a few minutes for the UG50 to register to the Telephony Server.
UG50 (Retrofit mode) • UG50-2MC • UG50-8LC • UG50-4LC2COTA (COT) • UG50-4LC2COTA (PGT)			

Note 1: IPG Analog resets itself to perform re-registration after two to three minutes, regardless of the time before forcible switchback is executed (ASYDL SYS1 Index 847 Bit 0~3).

ASYDL SYS1 INDEX 852 Bit 5= 0: Manual Switchback

Equipment State / Equipment Kind	During Conversation	During Ringback Tone	During Ringing Tone
IP terminal **Soft Phone (Proprietary Protocol)**	The CMNT command makes the terminal register to an MGC: CMNT command; Change Operating tab; “To Operating mode”		
SIP Multiple Line terminal			
Soft Phone (SIP)			
WLAN Handset (MH250)			

ASYDL SYS1 INDEX 852 Bit 5= 0: Manual Switchback

Equipment State / Equipment Kind	During Conversation	During Ringback Tone	During Ringing Tone
MC&MG-COT: **- SCA-4LC2COTA** **- MG-4LC2COTA**	The CMNT command makes the device register to an MGC after you disconnect an ongoing call: CMNT command; Change Operating tab; "To Operating mode"	The CMNT command makes an MC&MG-COT register to an MGC: CMNT command; Change Operating tab; "To Operating mode"	The CMNT command makes the device register to an MGC: CMNT command; Change Operating tab; "To Operating mode" to register to the Telephony Server.
Analog 2MC	The CMNT command makes an Analog 2MC register to an MGC: CMNT command; Change Operating tab; "To Operating mode"	The CMNT command makes an Analog 2MC register to an MGC: CMNT command; Change Operating tab; "To Operating mode"	
8LC Card	The CMNT command makes an 8LC Card register to an MGC after you disconnect an ongoing call: CMNT command; Change Operating tab; "To Operating mode"	The CMNT command makes an 8LC Card register to an MGC after you go on-hook: CMNT command; Change Operating tab; "To Operating mode"	
IPG Digital	The CMNT command makes IPG register to an MGC after you disconnect an ongoing call: CMNT command; Change Operating tab; "To Operating mode"	The CMNT command makes IPG register to an MGC after you go on-hook: CMNT command; Change Operating tab; "To Operating mode"	
IPG Analog	The CMNT command makes IPG register to an MGC after you disconnect an ongoing call: CMNT command; Change Operating tab; "To Operating mode"	The CMNT command makes IPG register to an MGC after you go on-hook: CMNT command; Change Operating tab; "To Operating mode"	
UG50 (Multi-slot mode) • UG50 (DLC)	The CMNT command makes the UG50 register to an MGC after you disconnect an ongoing call: CMNT command; Change Operating tab; "To Operating mode"	The CMNT command makes the UG50 register to an MGC after you go on-hook: CMNT command; Change Operating tab; "To Operating mode"	
UG50 (Multi-slot mode) • UG50 (LC) [Proprietary] • UG50 (LC) [SIP]	The CMNT command makes IPG register to an MGC after you disconnect an ongoing call: CMNT command; Change Operating tab; "To Operating mode"	The CMNT command makes IPG register to an MGC after you go on-hook: CMNT command; Change Operating tab; "To Operating mode"	

ASYDL SYS1 INDEX 852 Bit 5= 0: Manual Switchback

Equipment State / Equipment Kind	During Conversation	During Ringback Tone	During Ringing Tone
UG50 (IPG mode) • UG50-IPG (Digital)	The CMNT command makes IPG register to an MGC after you disconnect an ongoing call: CMNT command; Change Operating tab; "To Operating mode"	The CMNT command makes IPG register to an MGC after you go on-hook: CMNT command; Change Operating tab; "To Operating mode"	The CMNT command makes the device register to an MGC: CMNT command; Change Operating tab; "To Operating mode" to register to the Telephony Server.
UG50 (IPG mode) • UG50-IPG (Analog)	The CMNT command makes IPG register to an MGC after you disconnect an ongoing call: CMNT command; Change Operating tab; "To Operating mode"	The CMNT command makes IPG register to an MGC after you go on-hook: CMNT command; Change Operating tab; "To Operating mode"	
UG50 (Retrofit mode) • UG50-2MC • UG50-8LC • UG50-4LC2COTA (COT) • UG50-4LC2COTA (PGT)	The CMNT command makes IPG register to an MGC after you disconnect an ongoing call: CMNT command; Change Operating tab; "To Operating mode"	The CMNT command makes IPG register to an MGC after you go on-hook: CMNT command; Change Operating tab; "To Operating mode"	

c.) If the terminal operating under SR-MGC presses Visitor key

Terminal Reset by Visitor Key	Status of Terminal			
	Idle	During Conversation	During Ringback Tone	During Ringing Tone
ASYDL SYS 1 Index 846 Bit 4=0	Key operation is available	Key operation is Not available	Key operation is Not available	Key operation is Not available
ASYDL SYS 1 Index 846 Bit 4=1	Key operation is Not available	Key operation is Not available	Key operation is Not available	Key operation is Not available

d.) If switchback is performed during call termination to MC&MG-COT that is controlled by SR-MGC, the incoming call scenario will be as follows depending on Incoming Signal Interface of the channel used for the call.

MC&MG-COT: Incoming Signal Interface	Call Scenario
Ring Down	SR-MGC will not retain the incoming call, executing switchback. Note that if other channel (including an MC channel) is in use, the call will be retained.
Other than Ring Down	SR-MGC will retain the incoming call.

<Interactions with Other Features>

1. While SR-MGC is in operating mode, the billing information is stored in the remote office where the SR-MGC is located. When MGC is recovered, billing information accumulated until that time is transferred to the recovered MGC. The billing information can be stored up to 47,000 calls.

2. Features that need to group terminals, such as Multi-Line and Station Hunting, might not be available, depending on the system configuration.

3. SR-MGC does not support Hotel, CCIS over IP, FCCS, and Attendant Console features.

4. ACD and OAI services are available with SR-MGC.

5. To check the status of terminal registration on SR-MGC, use DISD command. Also, DISD command is used to send Soft Reset packets to the SR-MGC. For details on DISD command, refer to Operations and Maintenance Guide.

6. A program upload from the maintenance PC to the CF card in the SR-MGC is available via FTP. (Login account of SR-MGC must be registered in advance to use this function. See "Office Data Management via FTP" in Operations and Maintenance Guide for detailed procedure of program uploading.) Be sure to initialize the SR-MGC by using the CMNT command after updating the program.

7. If new SIP equipment performs Terminal Authentication using a one-time password, backup memory data by MEM_HDD/MEM_HDD_N command. After the backup of memory data, copy the office data to SR-MGC by using CMNT command or periodical diagnostic function. If the information on Terminal Authentication is not reflected on MGC, the SIP equipment cannot be rescued.

8. SIP Terminal Authentication [S-152] is supported in this feature. SR-MGC can rescue operating SIP terminals that have already completed SIP Terminal Authentication [S-152] with MGC (Telephony Server). To rescue SIP terminals with SR-MGC, save the Office Data including data related to SIP Terminal Authentication feature by MEM_HDD command. And then copy it into SR-MGC by CMNT command.

9. SR-MGC has supported VoIP Encryption [V-27].

10. When Softphone using Soft Phone Wired Phone Handset/Soft Phone Wireless Handset moves to SR-MGC, the following conditions must be considered. The conditions differ according to the software version of Softphone.

    Even if the Softphone moves to SR-MGC, the setting of "DRS Host Name or IP Address (Secondary/3rd/4th)" on the DtermSP30 (Proprietary Protocol) DRS Setting or "VoIP Host Name (2/3/4)" on the DtermSP30 (SIP) VoIP Setting (DtermSP30 Configuration Utility) is enabled. However, do not set the DRS Host Name of SR-MGC to "DRS Host Name or IP Address (Secondary/3rd/4th)" on the DtermSP30 (Proprietary Protocol) DRS Setting or "VoIP Host Name (2/3/4)" on the DtermSP30 (SIP) VoIP Setting because Soft Phone Wired Phone Handset/Soft Phone Wireless Handset might be activated incorrectly in SR-MGC. For the conditions of FCCS Cluster, see Location Diversity [L-55].

11. If Automatic Switchback is performed while a wired telephone which is being used as handset of Soft Phone by Soft Phone Wired-Phone Handset [S-147] is engaged in a call, the connection can be maintained and the Soft Phone can return to the main office MGC.

<Hardware/Software>

1. A maximum of 256 units of SR-MGC can be installed in a system.
2. External FTP server must be provided with 8GB or more free space for the hard disk.
3. The FTP Client feature of SR-MGC is operated in the Passive Mode. Use an external FTP server which supports the Passive Mode. Furthermore, set the Time to Stop the Program Copy late enough to complete the Program Copy. It takes about 2 hours in minimum to complete the Program Copy.

   Note: Passive Mode is an FTP connection mode which sends a connection request from the client to the server for data communication once again after the client is connected to the server.

4. Free space of 2 GB or more is required for the CF card of SR-MGC to execute a Program Copy. Otherwise, the Program Copy may fail.

   Note: Check the free space of a CF card in the System Status display area of Telephony Server Maintenance Menu.

5. When a communication error occurs:
   IP terminal becomes available to use in approximately 2 to 4 minutes.
   SIP Multiple Line terminal becomes available to use in approximately 2 to 7 minutes.

6. After an SR-MGC is powered on but has not been setup, or after its IP address is changed (**Note 2**) needs to communicate with the MGC for a Health Check.

   Note 2: Changing an SR-MGC IP address is only for the purpose of setting the SR-MGC back to default, so that it can be re-installed.

7. SR-MGC requires VS32 when terminals belonging to the SR-MGC use voice conference features such as Conference and Announcement.

8. This feature supports WLAN Handset (MH250) under the following conditions:
   a.) In a system with Location Diversity [L-55], SR-MGC cannot rescue WLAN Handsets (MH250).
   b.) Backup memory data by using MEM_HDD/MEM_HDD_N command after changing password of WLAN Handset (MH250). After the backup of memory data, copy the office data to SR-MGC by using CMNT command or periodical diagnostic function. If the password change is not reflected on MGC, WLAN Handset (MH250) cannot be rescued.

9. The following describe the setting of LAN interface for the applications (**Note 3**) communicating with SR-MGC from the Telephony Server side.

   To communicate with SR-MGC, the Telephony Server uses the specified LAN interface. By default, LAN1 interface is used for the applications communicating with SR-MGC (**Note 4**). This communication includes Health Check between the Telephony Server and SR-MGC. Therefore, a failure occurred on the LAN2 port cannot be notified to SR-MGC.

Note 3: The applications communicating with SR-MGC refer to all applications, i.e., Health Check, SMDR, Office Data Copy, DRS and Internal PHx.

Note 4: OS version upgrading function uses LAN1 for performing the program copy, regardless of the setting of LAN interface for the applications used by SR-MGC.

10. Only LAN1 interface can be used for communicating with the Telephony Server. LAN2 interface cannot be selected.
11. The operation of SR-MGC that performs Health Check between the Telephony Server and SR-MGC by using LAN1 of the Telephony Server varies depending on the status of the LAN interfaces.

    *By default, a rescue by SR-MGC is not performed for the failure occurred on LAN2 only. Also, the IP equipment that is generally operated by LAN2 is not rescued by SR-MGC when a failure occurred on LAN1 only.

Relationship between LANs Conditions and SR-MGC Operations

LAN Port		Rescue by SR-MGC	Remarks
LAN1	LAN2		
Normal	Normal	-	
Normal	Failure	NOT Available	
Failure	Normal	Available	IP equipment using LAN2 can be also rescued by the office data setting. Refer to the system data (Changeover to SR-MGC is performed on both LAN1 and LAN2 sides of Telephony Server: ASYDL, SYS1, Index 846, bit 2).
Failure	Failure	Available	

12. When the IP equipment (MG(PRI) [Proprietary Protocol]/ UG50 (PRT 1.5M) [Proprietary Protocol]/ UG50(PRT 2M) [Proprietary Protocol]/ UG50-24PRIA[Proprietary Protocol]/UG50-30PRIA[Proprietary Protocol]) to be rescued by SR-MGC uses LAN2 port, the IP equipment must be assigned the same Location ID (LOC-ID) as the SR-MGC. SR-MGC does not rescue IP equipment that is assigned different Location ID. However, by the office data setting, the IP equipment using LAN2 can be switched to the operation by SR-MGC. If the following conditions are met, the equipment using LAN2 also can be rescued by SR-MGC by assigning the system data (ASYDL, SYS1, Index846, bit 2 = 1):
    - LAN1/LAN2 of the Telephony Server is assigned to different segments (IP addresses)
    - LOC-IDs are assigned per SR-MGC

    Assign ASYDL, SYS1, Index846, bit 2 = 0 if the above conditions cannot be met.

    Note: If installing SR-MGC onto the system where LAN2 accommodates MG(PRI) in Proprietary Protocol mode and LAN1 accommodates other IP equipment, pay close attention to the office data assignment and the system environments/conditions. Due to an incorrect office data setting or operating status of SR-MGC, MG(PRI) in Proprietary Protocol mode may suddenly become unavailable.

13. When assigning "2" to Changeover/Recover Mode in CMNT command, it is required to put both of LAN1 and LAN2 IP addresses in the same segment.

Note: If installing SR-MGC on a location where both of LAN1 and LAN2 are used by IP devices/stations to be rescued, special attention must be paid to the system - data, circumstances, conditions. Some factors related to them may cause IP devices/stations using LAN2 (MG(PRI) in Proprietary Protocol mode) to stop operating suddenly.

<Interactions with Networks>

1. SR-MGC cannot be used as a node for CCIS over IP or FCCS network. Therefore, only terminals under the same SR-MGC (including connections via Media Gateways) can communicate with each other.

<Conditions Specific to This Feature>

1. Assign IP address of LAN1 (ACT) as Telephony Server's address on SR-MGC, which is set by AIPA command.

2. When initially starting up an SR-MGC, other than the following commands necessary for startup are restricted until an MGC finishes a first office data copy toward the SR-MGC: AIPA, ATIM, ASPC, MEM_HDD, SINZ.

3. SR-MGC automatically assigns an IP address as Internal PH to itself.

4. A call can be retained even if a fault occurs during the conversation. For the detail of call retention function that each terminals have individually, see "Operation of IP Equipment/Terminals when a Fault Occurs during a Conversation" in Chapter 3 IP Equipment of System Description.

5. Health check from MGC to SR-MGC is performed routinely by assigning the System Data (ASYDL, SYS1, Index 850, Index 851) in the MGC. If the SR-MGC does not receive Health Check packets for a certain period, the SR-MGC assumes the MGC might be down, and it changes to operating mode in order to rescue the MGC equipment.

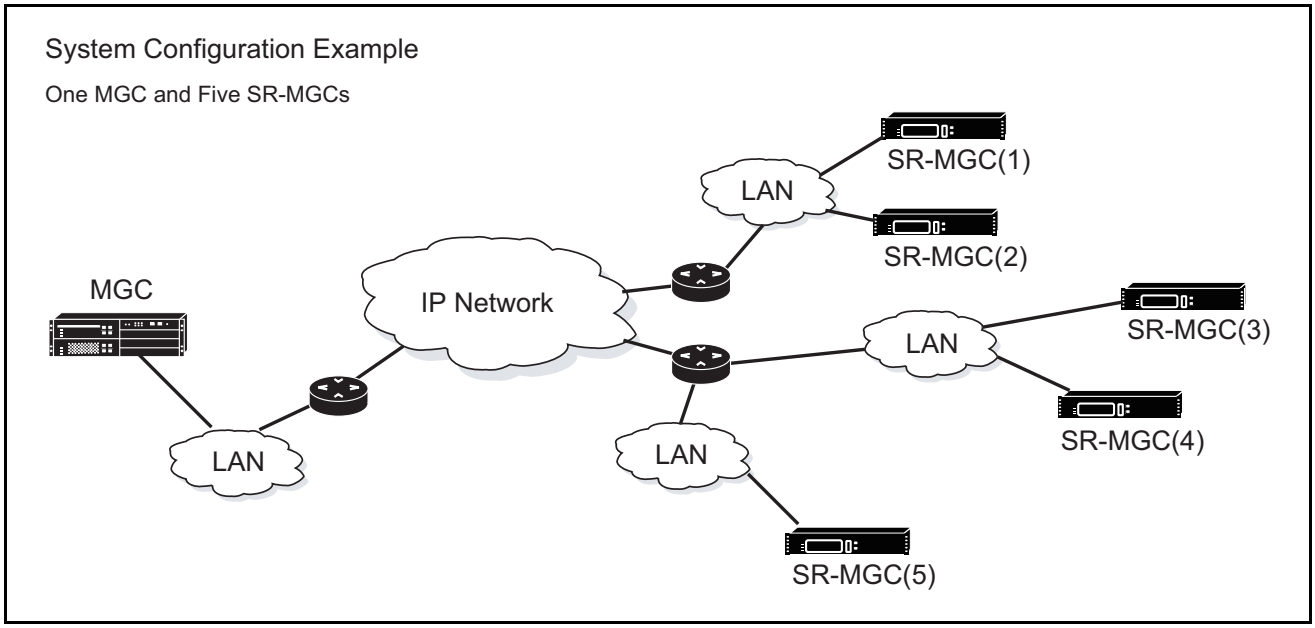
Note: Note the following conditions:
- It is recommended to assign at least 30 seconds (default) for Health Check cycle when 200 or more SR-MGC units are set for a MGC (Telephony Server).
- To change the health check data with the ASYDL, SYS1, Index 850, follow the procedure explained in Command Manual. For details, see the note on ASYDL, SYS1, Index 850 in Command Manual.

6. Office data of MGC and SR-MGC can be reflected by using the following: **Note 6**

(a) To reflect office data automatically

Office data is automatically transmitted to SR-MGC after all the items have been completed through periodical diagnosis.

- Example for Automatic Office Data Update of SR-MGCs after Periodical Diagnosis



The following table shows scheduling for automatic office data update of SR-MGCs in the above case by setting ASYDL, SYS1, Index 844.

ASYDL SYS1 INDEX 844	Day						Description
	1st	2nd	3rd	4th	5th	6th	
5	SR-MGC (1),(2),(3), (4),(5)	SR-MGC (1),(2),(3), (4),(5)	SR-MGC (1),(2),(3), (4),(5)	SR-MGC (1),(2),(3), (4),(5)	SR-MGC (1),(2),(3), (4),(5)	SR-MGC (1),(2),(3), (4),(5)	Every SR-MGC is updated every day. (All five SR-MGCs are updated every day.)
4	SR-MGC (1),(2),(3), (4)	SR-MGC (5)	SR-MGC (1),(2),(3), (4)	SR-MGC (5)	SR-MGC (1),(2),(3), (4)	SR-MGC (5)	Every SR-MGC is updated once two days. (Four SR-MGCs are updated on the first day, and the rest on the second day.)
3	SR-MGC (1),(2),(3)	SR-MGC (4),(5)	SR-MGC (1),(2),(3)	SR-MGC (4),(5)	SR-MGC (1),(2),(3)	SR-MGC (4),(5)	Every SR-MGC is updated once two days. (Three SR-MGCs are updated on the first day, and the rest on the second day.)
2	SR-MGC (1),(2)	SR-MGC (3),(4)	SR-MGC (5)	SR-MGC (1),(2)	SR-MGC (3),(4)	SR-MGC (5)	Every SR-MGC is updated once three days. (Two SR-MGCs are updated on the first day, another two on the second day, and the rest on the third day.)
1	SR-MGC (1)	SR-MGC (2)	SR-MGC (3)	SR-MGC (4)	SR-MGC (5)	SR-MGC (1)	Every SR-MGC is updated once five days. (Only one SR-MGC is updated each day.)
0	-	-	-	-	-	-	No SR-MGCs are updated.

Note: Time required for office data copying depends on office data volume, transmission line capacity and copy method. When setting Automatic Office Data Update of SR-MGCs, allow enough time for office data copying, based on the measured copy time taken for the setup.

(b) To reflect office data immediately

Using the CMNT command transmits the office data immediately.

Note: The following table shows whether or not office data is available to copy to the SR-MGC.

X: Available N: Not Available

Data Type	SR-MGC
Data Memory	X
Name Display	X
Wireless Call Forwarding	X
Call Forwarding	X
Speed Dialing	X
User Assign Key	X
Number Sharing	X
ACD Data Memory	X
Call Block	X
ICB (**Note 3**) and DR	X
Expanded Speed Calling (Not used)	(Not used)
Call Forwarding-Logout	X
One Touch Speed Calling Data for DtermIP	X
SPD Name Data	X
H.323 Terminal Call Transfer (Not used)	(Not used)
Message Area ID Data	X
SR-MGC Data **Note 4**	N
SIP Terminal Call Transfer **Note 5**	X
Mobile Service Data	X
Day/Night Data **Note 3**	N
Local Phone book Data	N
Soft Switch Data	N

Note 3: Available only for North America.

Note 4: This item is generated only on SR-MGC.

Note 5: SIP Terminal Call Transfer indicates the transfer information of SP Controlled SIP terminal.

Note: The following conditions must be considered when performing office data copy.

- Do not log in to SR-MGC by PCPro when performing office data copy.
- Do not perform any PCPro operations other than CMNT command to the Telephony Server during office data copy. The PCPro operation such as login/logoff can cause a copy failure.
- Functions other than the Refresh button and the File Control tab cannot be used during office data copy. Office data copy may not be executed concurrently as well (CMNT command can only be executed one at a time).
- When using the manual copy function of CMNT command, it must be completed before the office data copy of the periodic Health Check starts. The manual copy function is canceled when the periodic Health Check starts. Once the office data copy was canceled, it would not be performed on that day regardless of ASYDL, SYS1, Indexes 844 and 845 settings.
- * To work around the above problem, assign data FF HEX to both ASYD SYS1 Index 87 and Index 88 to stop the periodic Health Check program before starting the manual office data copy.
- During office data copy, another tab in the CMNT command is not available. Also, another office data copy process is not available: you cannot open more than one CMNT command window.
- In a dual configuration system, if system changeover is performed during office data copy, the copy fails.
- If you abort office data copy, data in the SR-MGC will not be updated: the existing data will be used.
- If a failure occurs over a network between an MGC and the SR-MGC, data in the SR-MGC will not be updated: the existing data will be used.
- Do not execute CMNT command with a PCPro-installed PC while executing office data copy with another one.

Note 6: The following must be considered.

- MGC office data transmission to SR-MGC must be performed at least once 29 days.
  If more than 29 days has elapsed since the last SR-MGC data update, a system message, 10-A: System Activation Report (SUP), is output. The request notice will continue to be sent every day until a new update is made, and vary depending on the time SR-MGC has left not updated, as shown below:
  For more than 29 days and less than 45 days → 10-A: System Activation Report (SUP)
  For more than 44 days and less than 60 days → 10-B: System Activation Report (MN)
  For 60 days or more → 10-C: System Activation Report (MJ)
  If you want to stop the message without copying office data to SR-MGCs, use the CMNT command and execute the initialization of the SR-MGC on the Initialize Tab.

7. SR-MGC starts its operation when a breakdown of MGC or a network failure occurs. **Note 7**

Note 7: LAN1 on the Telephony Server is used as the interface for Health Check between MGC and SR-MGC. It means that a failure only on LAN2 does not activate SR-MGC because Health Check is performed normally as long as LAN1 can communicate with SR-MGC.

8. Program Copy can be executed from the File Control tab. The conditions for Program Copy are as follows:

   (a) Do not perform any PCPro operations other than CMNT command (such as login/logoff of PCPro) to the Telephony Server or SR-MGC during office data copy.

   (b) When the Stop button is pressed during the manual program copy process, the process is canceled before it is completed, and the temporary file is discarded.

   (c) When the manual program copy process is still running at the time assigned with Appoint Stop Time, the process is canceled.

9. "Program update" function provides program upgrade within same version. The following are conditions for program update.

   Note: Execute the Program Copy/Program Switch from the File Control tab when updating the program.

   a.) All of the following requirements must be satisfied between MGC (copy source) and a SR-MGC (copy destination).

   - Using the same version of MAIN program.
   - Using the same version of BASE system.
   - Using the same issue number for the BASE system including digits after the decimal point.

   b.) During the program update process, MGC does not perform office data copy after periodic Health Check, which means no office data copy on the day.

   c.) During office data copy process after periodic Health Check, program update is not available.

   d.) For details on the program to be updated, see CMNT command in the Operations and Maintenance Manual.

<Conditions Specific to SR-MGC and FCCS>

The following explains SR-MGC conditions when FCCS network is involved.

SR-MGC Coverage Area (Example 1)

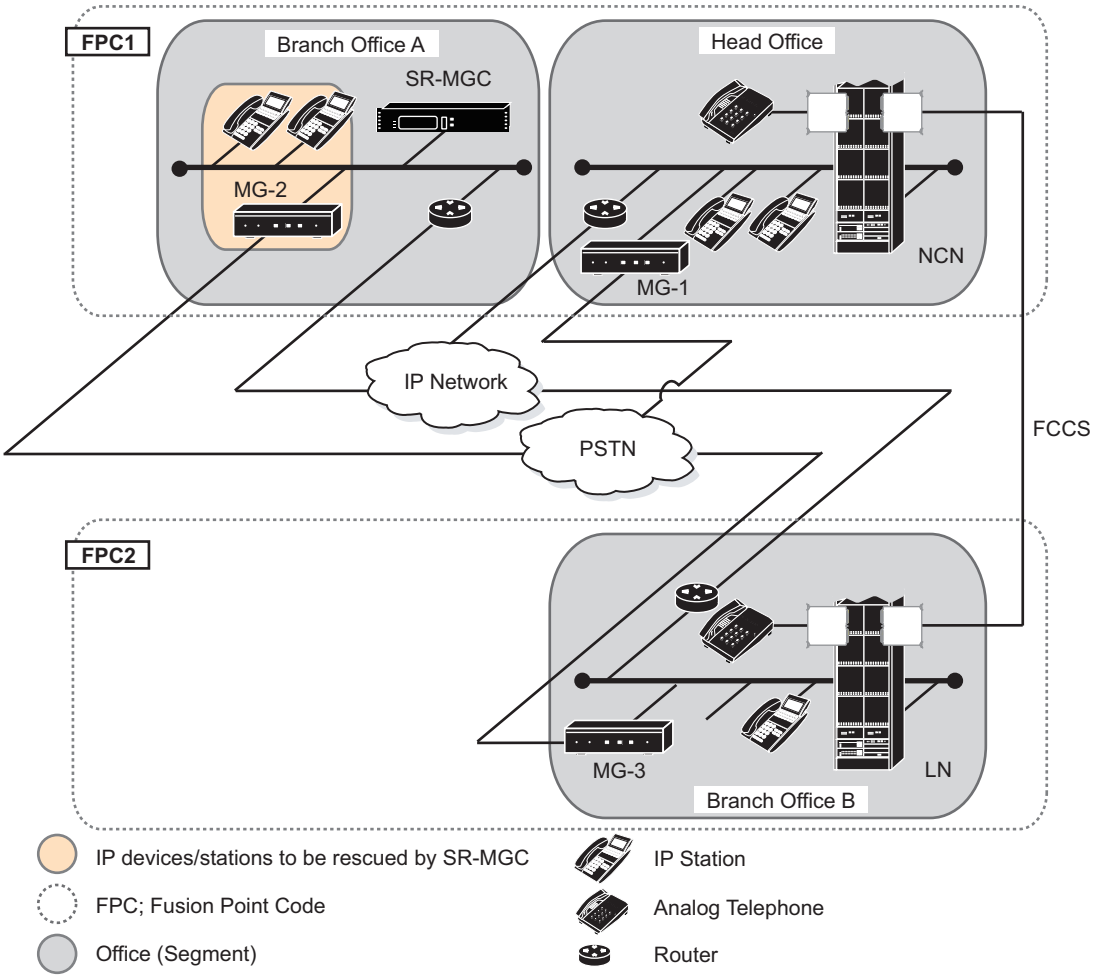
In the following configuration, the SR-MGC can back up IP terminals on LAN of Branch Office A only. Other IP terminals cannot be backed up by the SR-MGC.

System Configuration:

- Head Office and Branch Office B are connected via FCCS.
- SR-MGC that corresponds to NCN is installed in Branch Office A. **Note 8**
- In normal case, IP terminals in Branch Office A are controlled by NCN that is installed in Head Office.

Note 8: In an FCCS network, each node, which was previously called the Telephony Server, may be referred to as NCN or LN. NCN is an abbreviation of Network Control Node that is the main node in an FCCS network. LN is an abbreviation of Local Node which are nodes other than NCN in the FCCS network. For more information, refer to the Data Programming Manual - FCCS.

SR-MGC in FCCS Network - Example 1



IP Terminal Operationability in FCCS Nodes - Example 1

No.	Calling Party	Called Party	Calling Party Operationability	Called Party Operationability
1	Terminals in Head Office	Terminals in Office A	Receives RBT.	Cannot be called.
2	Terminals in Office A			Can be called.
3	Terminals in Office B			Cannot be called.
4	Terminals in Office A	IP Terminals in Head Office		
5		Non-IP Terminals in Head Office	Receives ROT with 'RST' indication on the LCD.	
6		IP Terminals in Office B	Receives ROT without indication on the LCD.	
7		Non-IP Terminals in Office B		

Note: When LOGGED OUT IP STATION - CALL DESTINATION has been set, the call is forwarded in the case of items 1, 3, 4.

Note: Items 6 and 7 are the same status such as an FCCS link-all-busy is encountered.

MG Operationability (When all MGs are operating by FCCS Nodes (NCN/LN)) - Example 1

No.	Calling Party	Available MGs for Making Outgoing Calls		
1	Terminals in Head Office	MG-1	MG-2	MG-3
2	Terminals in Office A	MG-1	MG-2	MG-3
3	Terminals in Office B	MG-1	MG-2	MG-3

Note: Shaded Media Gateways (MGs) are available.

MG Operationability (When MG-2 is operating by SR-MGC) - Example 1

No.	Calling Party	Available MGs for Making Outgoing Calls		
1	Terminals in Head Office	MG-1	MG-2 (Not Available)	MG-3
2	Terminals in Office A	MG-1 (Not Available)	MG-2	MG-3 (Not Available)
3	Terminals in Office B	MG-1	MG-2 (Not Available)	MG-3

Note: Media Gateways (MGs) are available. MGs cannot be used from the office.

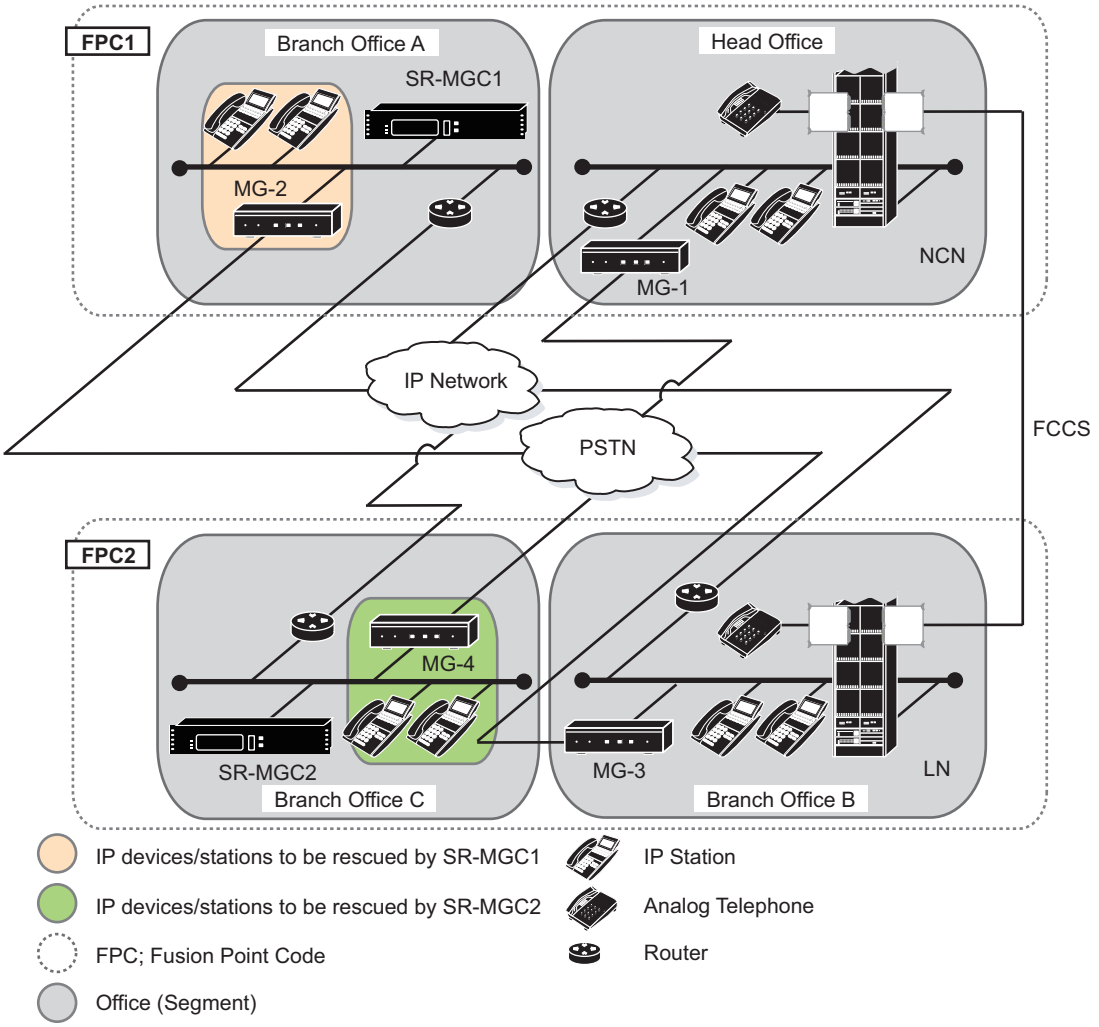
SR-MGC Coverage Area (Example 2)

In the following configuration, SR-MGC1 can back up IP terminals on LAN of Branch Office A and SR-MGC2 can back up IP terminals on LAN of Branch Office C only. Other IP terminals cannot be backed up by these SR-MGCs.

System Configuration

- Head Office and Branch Office B is connected via FCCS.
- SR-MGC1 that corresponds to NCN is installed in Branch Office A and SR-MGC2 that corresponds to LN is installed in Branch Office C.
- In normal status, IP terminals in Branch Office A are controlled by NCN and IP terminals in Branch Office C are controlled by LN.

SR-MGC in FCCS Network - Example 2



IP Terminal Operationability in FCCS Nodes - Example 2

No.	Calling Party	Called Party	Calling Party Operationability	Called Party Operationability
1	Terminals in Head Office (Terminals in Office B)	Terminals in Office A (Terminals in Office C)	Receives RBT.	Cannot be called.
2	Terminals in Office A (Terminals in Office C)			Can be called.
3	Terminals in Office B (Terminals in Head Office)			Cannot be called.
4	Terminals in Office A (Terminals in Office C)	Terminals in Head Office (Terminals in Office B) (IP Terminals)		
5	Terminals in Office A (Terminals in Office C)	Terminals in Head Office (Terminals in Office B) (Non-IP Terminals)	Receives ROT with 'RST' indication on the LCD.	
6		Terminals in Office B (Terminals in Head Office) (IP Terminals)	Receives ROT without indication on the LCD.	
7		Terminals in Office B (Terminals in Head Office) (Non-IP Terminals)		
8		Terminals in Office C (Terminals in Office A)		

Note: When LOGGED OUT IP STATION - CALL DESTINATION has been set, the call is forwarded in the case of 1, 3, and 4.

Note: Items 6, 7 and 8 are the same status such as an FCCS link-all-busy is encountered.

MG Operationability (When all MGs are operating by FCCS Nodes (NCN/LN)) - Example 2

No.	Calling Party	Available MGs for Making Outgoing Calls			
1	Terminals in Head Office	MG-1	MG-2	MG-3	MG-4
2	Terminals in Office A	MG-1	MG-2	MG-3	MG-4
3	Terminals in Office B	MG-1	MG-2	MG-3	MG-4
4	Terminals in Office C	MG-1	MG-2	MG-3	MG-4

Note: Shaded MGs are available.

MG Operationability (When MG-2 is operating by SR-MGC) - Example 2

No.	Calling Party	Available MGs for Making Outgoing Calls			
1	Terminals in Head Office	MG-1	MG-2 (Not Available)	MG-3	MG-4
2	Terminals in Office A	MG-1 (Not Available)	MG-2	MG-3 (Not Available)	MG-4 (Not Available)
3	Terminals in Office B	MG-1	MG-2 (Not Available)	MG-3	MG-4
4	Terminals in Office C	MG-1	MG-2 (Not Available)	MG-3	MG-4

Note: Shaded MGs are available. Unshaded MGs cannot be used from the office.

MG Operationability (When MG-4 is operating by SR-MGC) - Example 2

No.	Calling Party	Available MGs for Making Outgoing Calls			
1	Terminals in Head Office	MG-1	MG-2	MG-3	MG-4 (Not Available)
2	Terminals in Office A	MG-1	MG-2	MG-3	MG-4 (Not Available)
3	Terminals in Office B	MG-1	MG-2	MG-3	MG-4 (Not Available)
4	Terminals in Office C	MG-1 (Not Available)	MG-2 (Not Available)	MG-3 (Not Available)	MG-4

Note: Shaded MGs are available. Unshaded MGs cannot be used from the office.

MG Operationability (When MG-2 and MG-4 are operating by SR-MGCs) - Example 2

No.	Calling Party	Available MGs for Making Outgoing Calls			
1	Terminals in Head Office	MG-1	MG-2 (Not Available)	MG-3	MG-4 (Not Available)
2	Terminals in Office A	MG-1 (Not Available)	MG-2	MG-3 (Not Available)	MG-4 (Not Available)
3	Terminals in Office B	MG-1	MG-2 (Not Available)	MG-3	MG-4 (Not Available)
4	Terminals in Office C	MG-1 (Not Available)	MG-2 (Not Available)	MG-3 (Not Available)	MG-4

Note: Shaded MGs are available. Unshaded MGs cannot be used from the office.

SR-MGC Behavior in case of FCCS Node / Network Failure

In Case of NCN Failure

If a failure occurs in the NCN, it affects the system as follows:

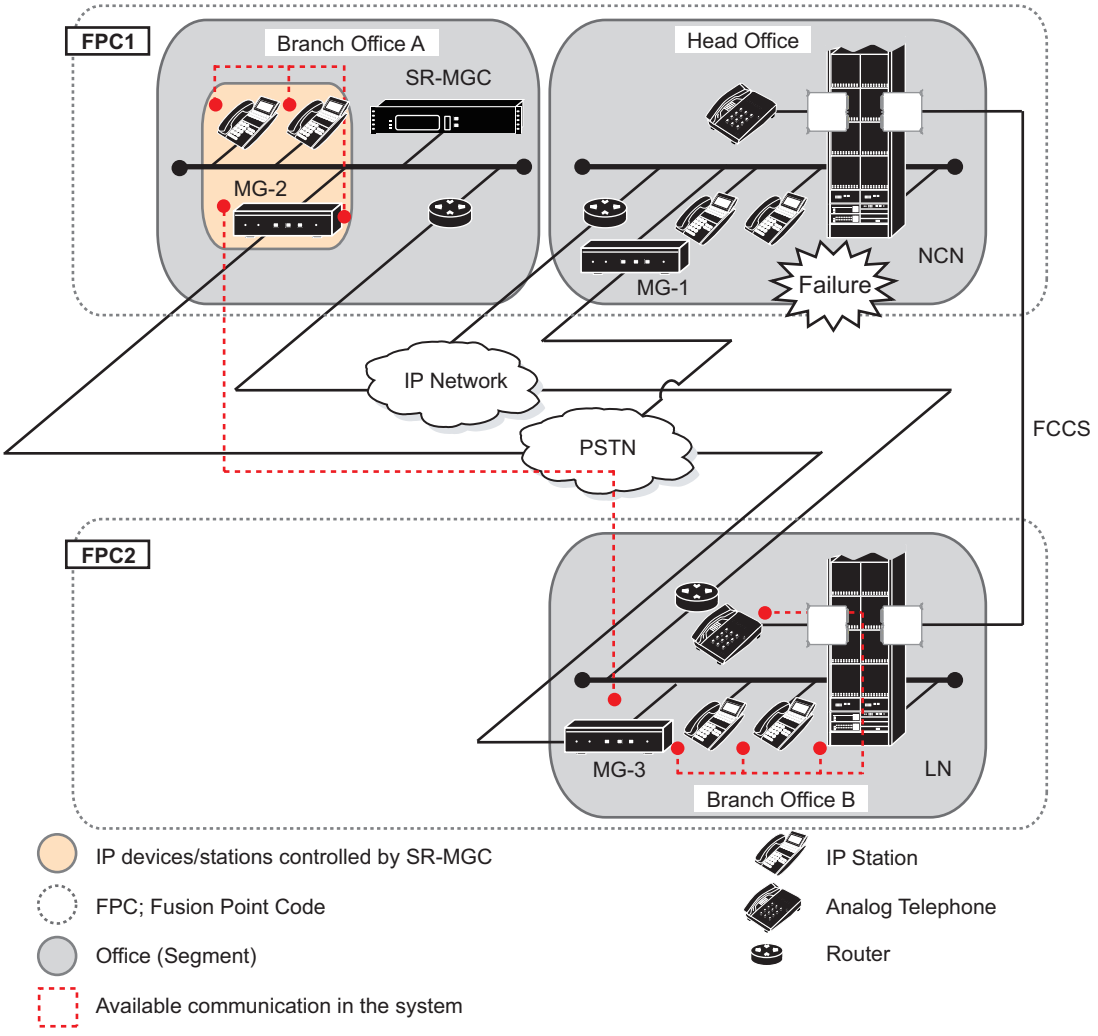
(1) FCCS calls cannot be made between IP devices/stations registered to SR-MGC and ones in FPC2. **Note 9**

(2) Calls can be made between IP devices/stations registered to SR-MGC.

(3) Calls can be made between IP devices/stations registered to SR-MGC and ones in FPC2 via MG.

Note 9: By using Automatic Re-routing when SR-MGC is in Operating Mode, calls can be automatically rerouted from SR-MGC to FPC2 via MGs.

SR-MGC Behavior in case of NCN Failure



In Case of Network Failure in Head Office

If a failure occurs on network (access to WAN) in Head Office, it affects the system as follows:

(1) FCCS calls cannot be made between IP devices/stations registered to SR-MGC and ones in FPC2. **Note 10**

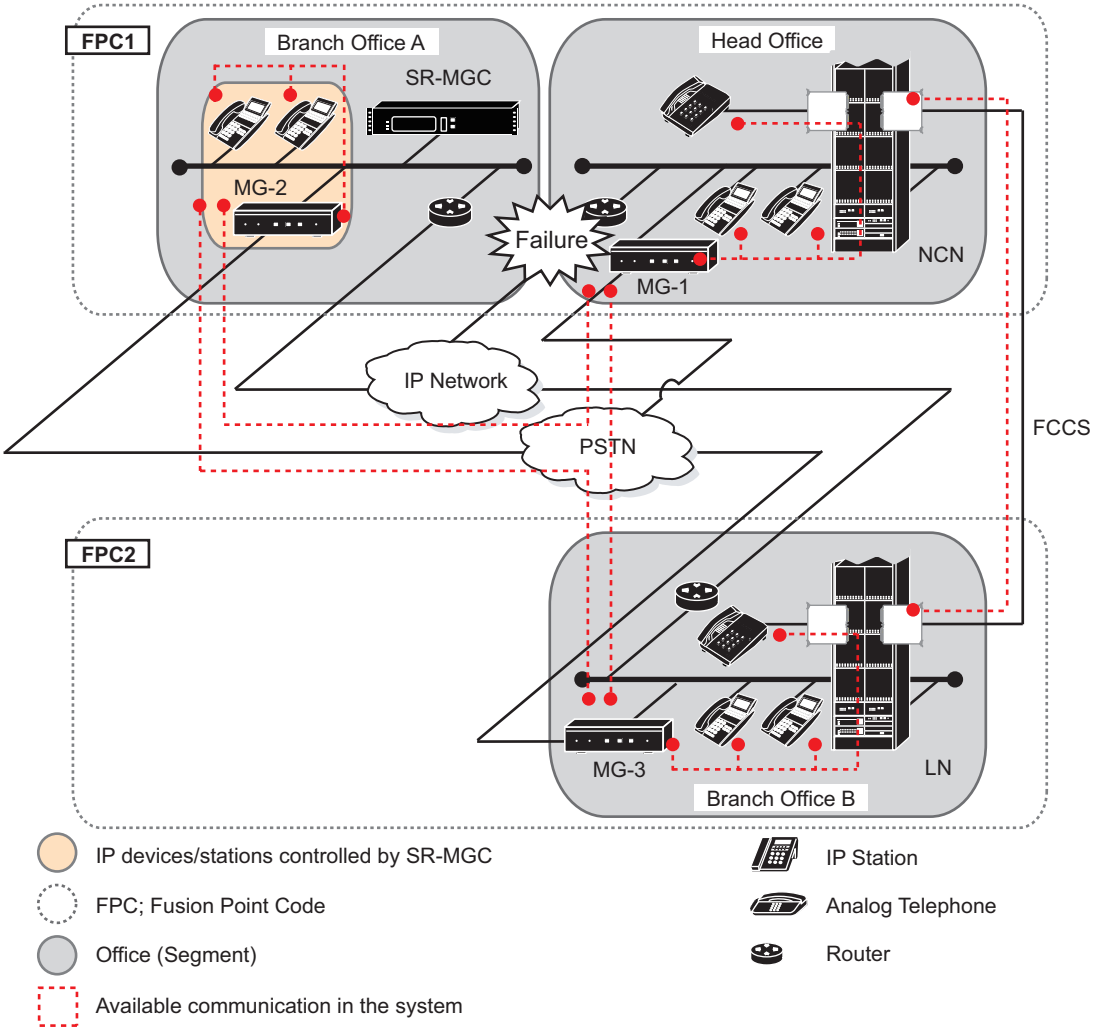
(2) Calls can be made between IP devices/stations registered to SR-MGC.

(3) FCCS calls can be made between NCN and LN.

(4) Calls can be made via MG.

Note 10: By using Automatic Re-routing when SR-MGC is in Operating Mode, calls can be automatically rerouted from SR-MGC to FPC2 via MGs.

SR-MGC Behavior in case of Network Failure in Head Office



In Case of Network Failure in both Nodes - NCN and LN

If a failure occurs on network (access to WAN) in both Head Office and Branch Office B, it affects the system as follows:

(1) FCCS calls cannot be made between IP devices/stations registered to SR-MGC and ones in FPC2. **Note 11**

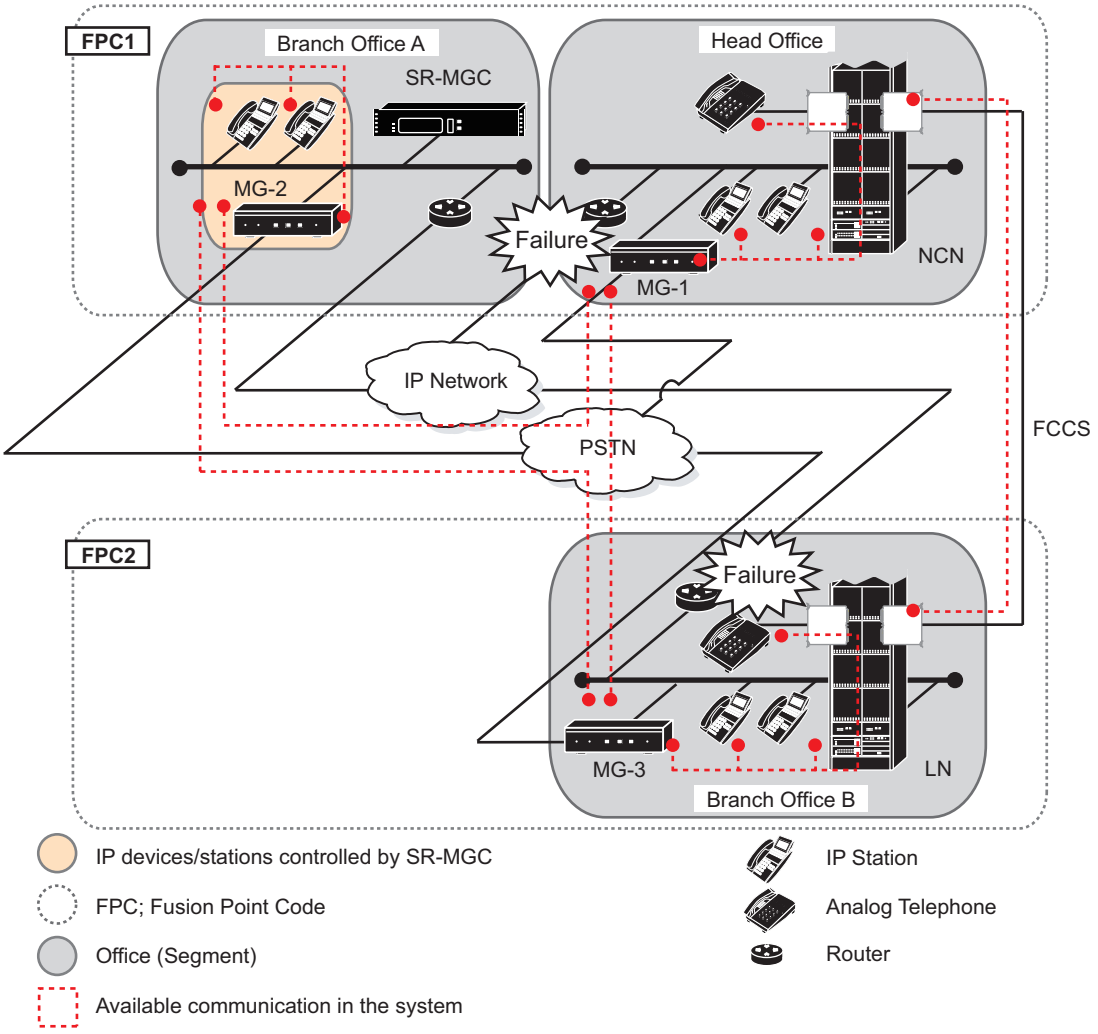
(2) Calls can be made between IP devices/stations registered to SR-MGC.

(3) FCCS calls can be made between NCN and LN.

(4) Calls can be made via MG.

Note 11: By using Automatic Re-routing when SR-MGC is in Operating Mode, calls can be automatically rerouted from SR-MGC to FPC2 via MGs.

SR-MGC Behavior in case of Network Failure in both Nodes



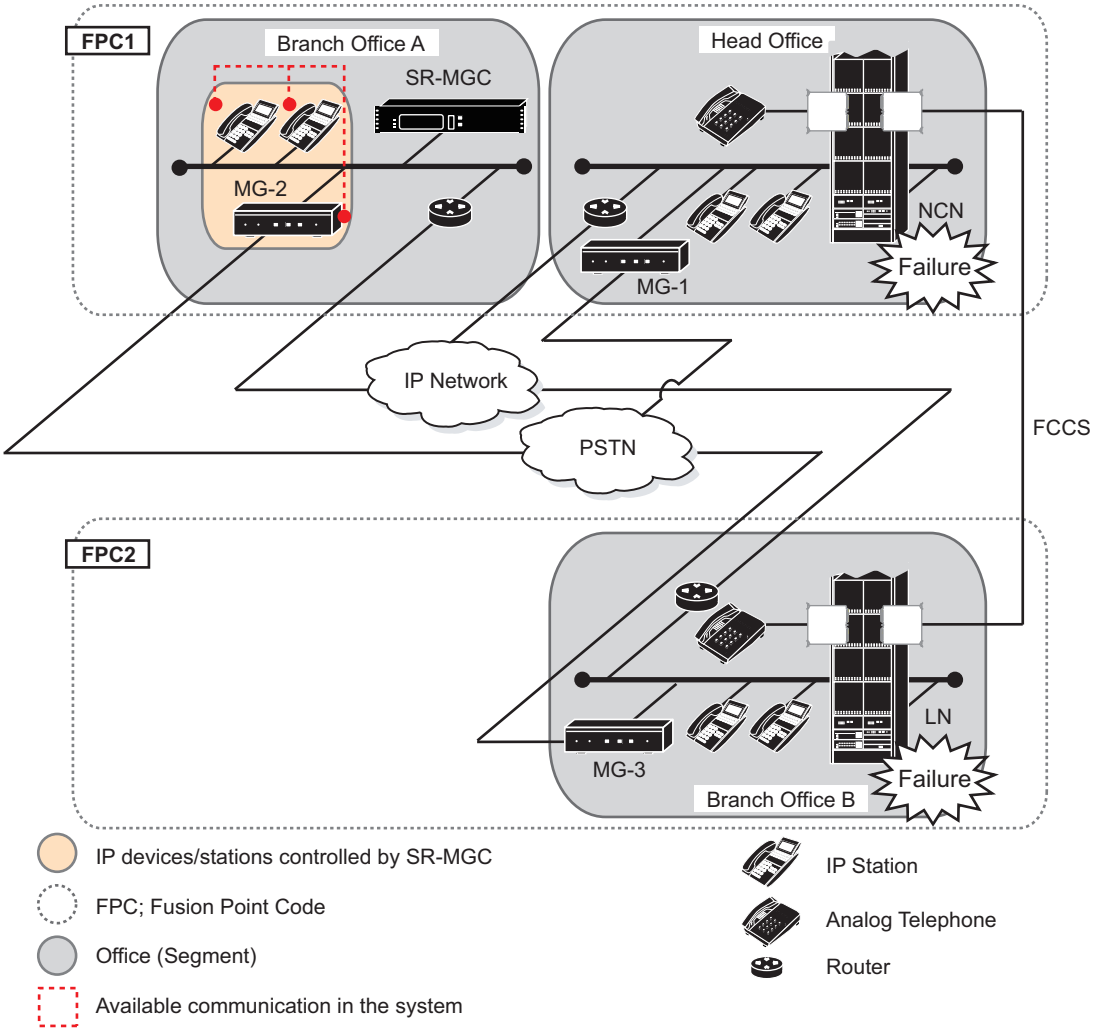
In Case of Node Failure in both FCCS Nodes - NCN and LN (Example 1)

If a failure occurs in both FCCS nodes - NCN and LN, it affects the system as follows:

(1) Calls can be made between IP devices/stations registered to SR-MGC.

(2) FCCS calls cannot be made between FPC1 and FPC2.

SR-MGC Behavior in case of Node Failure in both FCCS Nodes (Example 1)



In Case of Node Failure in both FCCS Nodes - NCN and LN (Example 2)

If a failure occurs in both FCCS nodes - NCN and LN, it affects the system as follows:

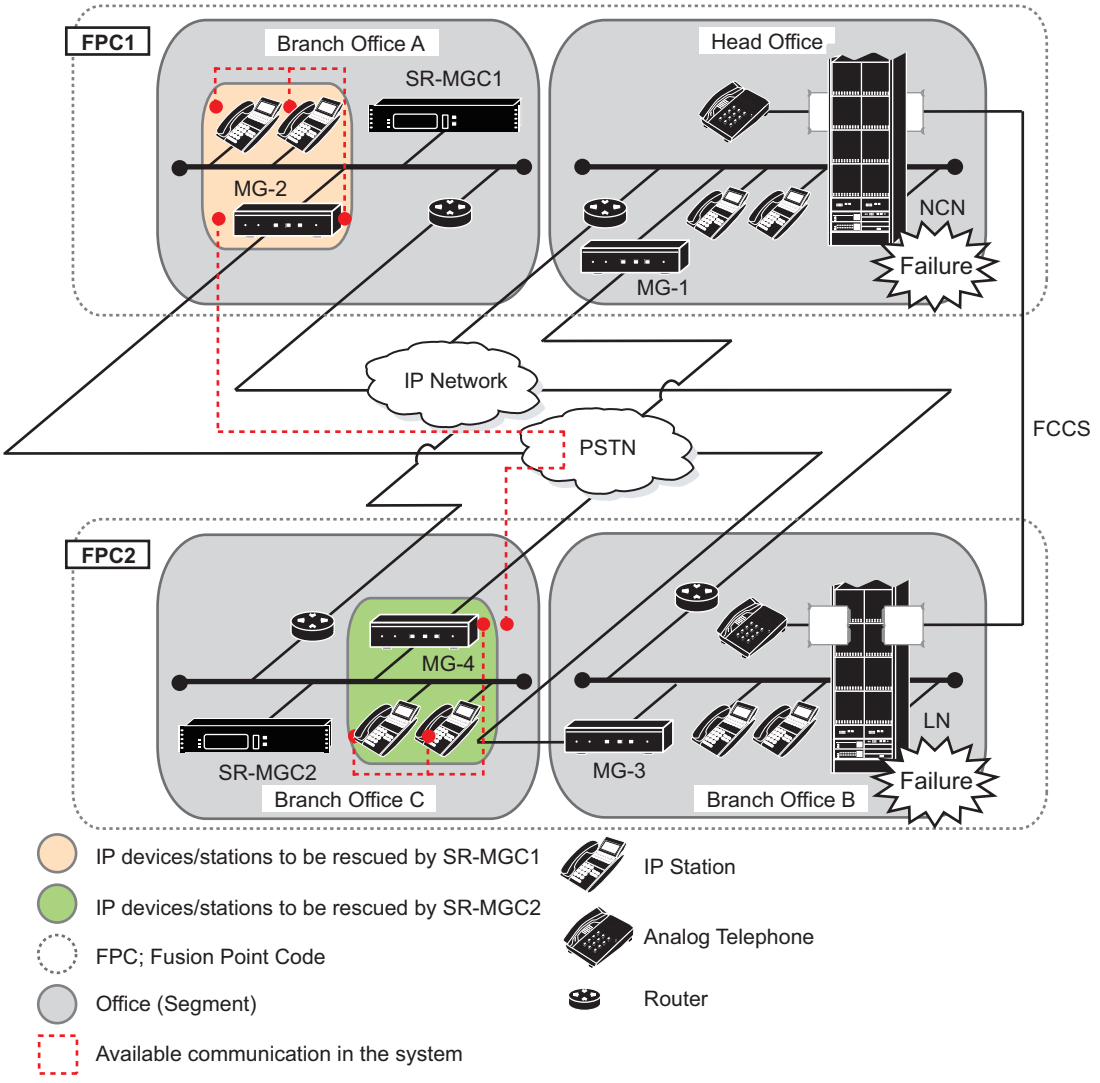
(1) Calls can be made between IP devices/stations registered to each SR-MGC.

(2) Calls can be made between IP devices/stations registered to SR-MGC1 and SR-MGC2 via MGs. **Note 12**

(3) FCCS calls cannot be made between FPC1 and FPC2.

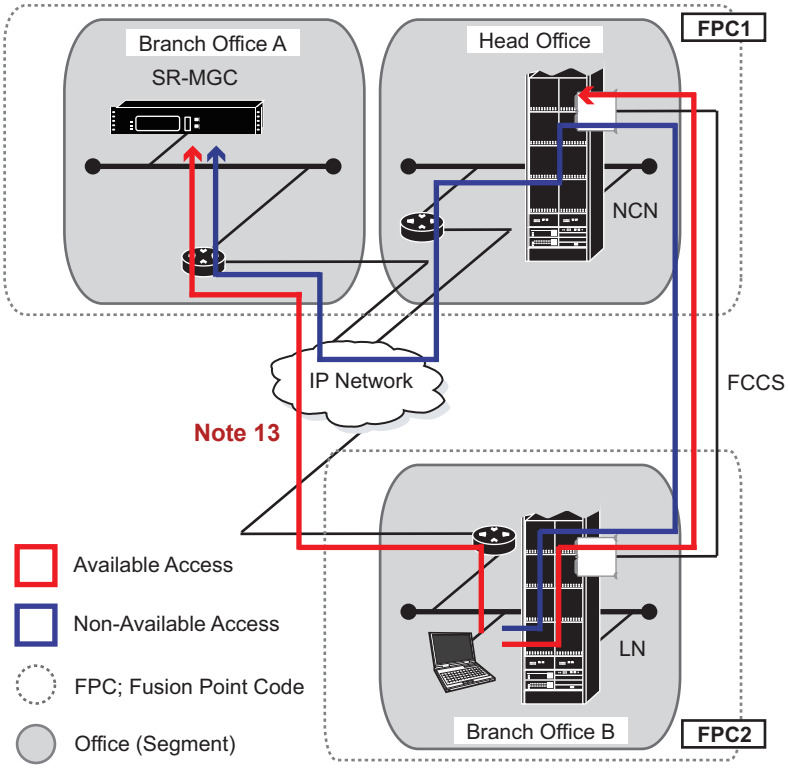
Note 12: By using Automatic Re-routing when SR-MGC is in Operating Mode, calls can be automatically rerouted from SR-MGC1 to SR-MGC2 via MGs.

SR-MGC Behavior in case of Node Failure in both FCCS Nodes (Example 2)



In order to do maintenance work, maintenance personnel are required to log in to the system. NCN can be accessed from Branch Office B via FCCS as shown below. However, login to the SR-MGC cannot be done in the same way. The personnel must directly log in to the SR-MGC by specifying the IP address of the SR-MGC from the PC.

Login to SR-MGC



Note 13: Maintenance personnel can gain access to the SR-MGC by specifying the IP address of the SR-MGC.

Broadcast Control

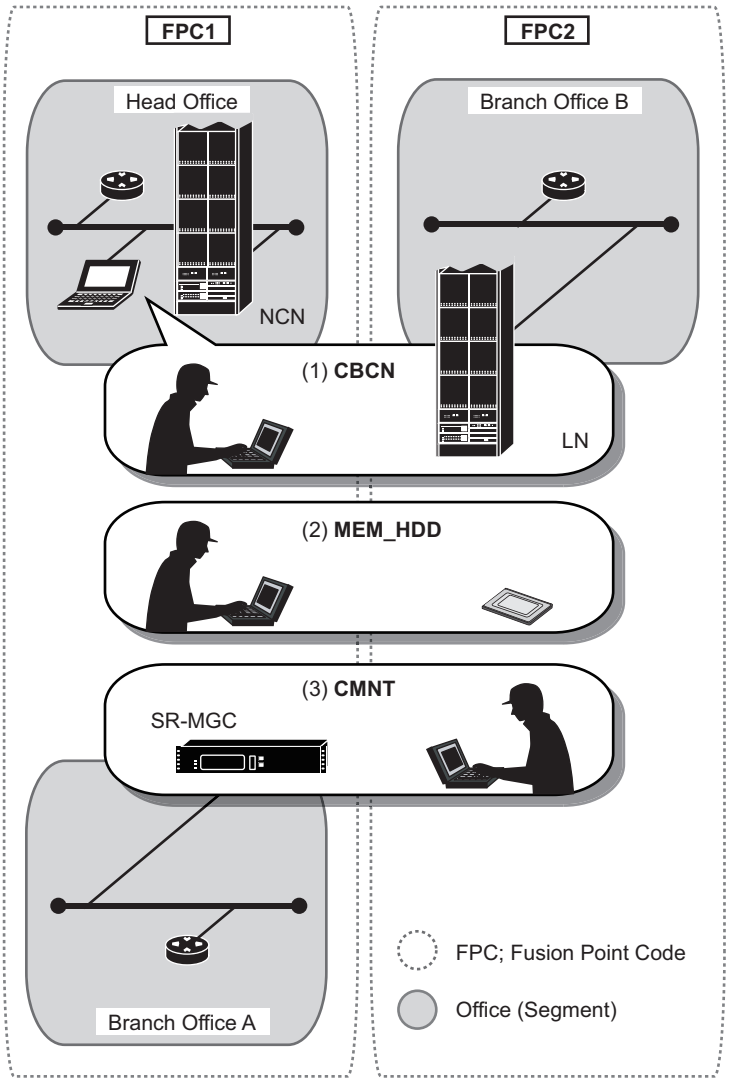
When performing the broadcast of NDM by using CBCN command between NCN and LN, the contents of the NDM are not reflected to SR-MGC. To reflect the data to the SR-MGC, perform the following steps:

STEP 1: Perform the broadcast of NDM by using CBCN command.

STEP 2: Back up the office data to the flash card using MEM_HDD command at NCN.

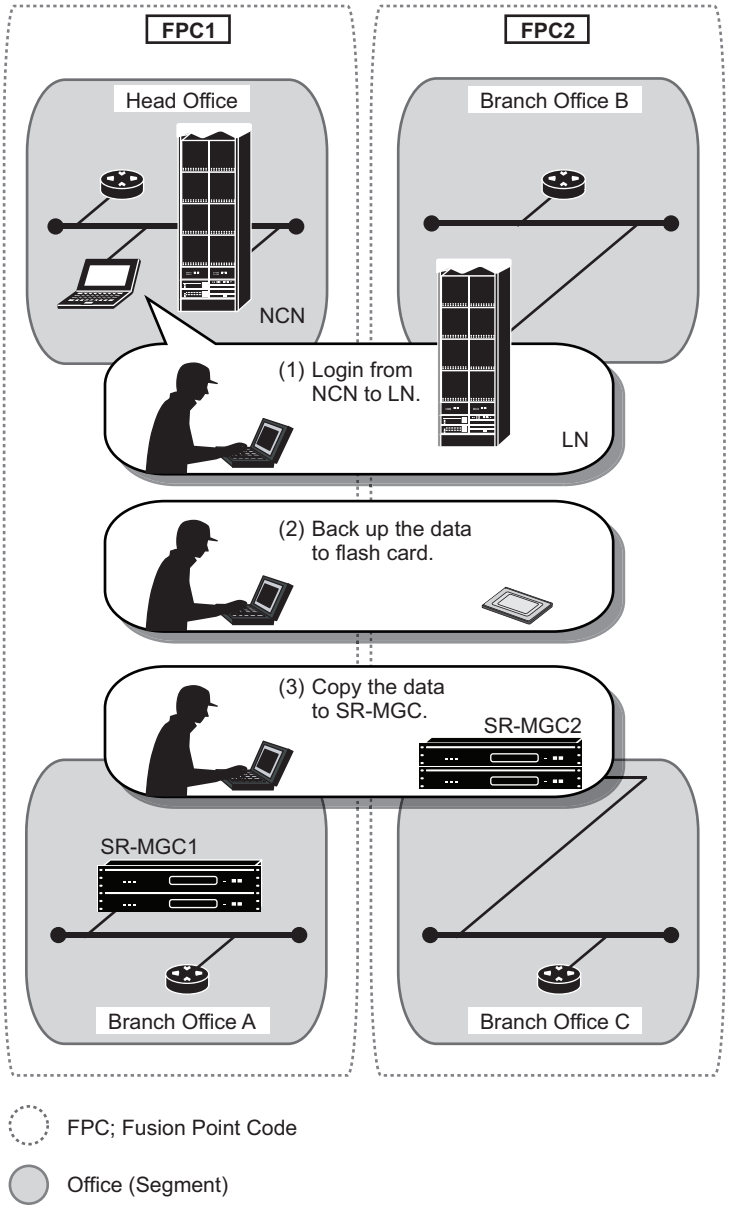
STEP 3: Perform "Office Data Copy" by using CMNT command at the NCN.

Broadcast Control (Example 1)



If login from NCN to LN is available, Office Data Copy from LN to SR-MGC2 installed in Branch Office B (FPC2) is available.

Broadcast Control (Example 2)



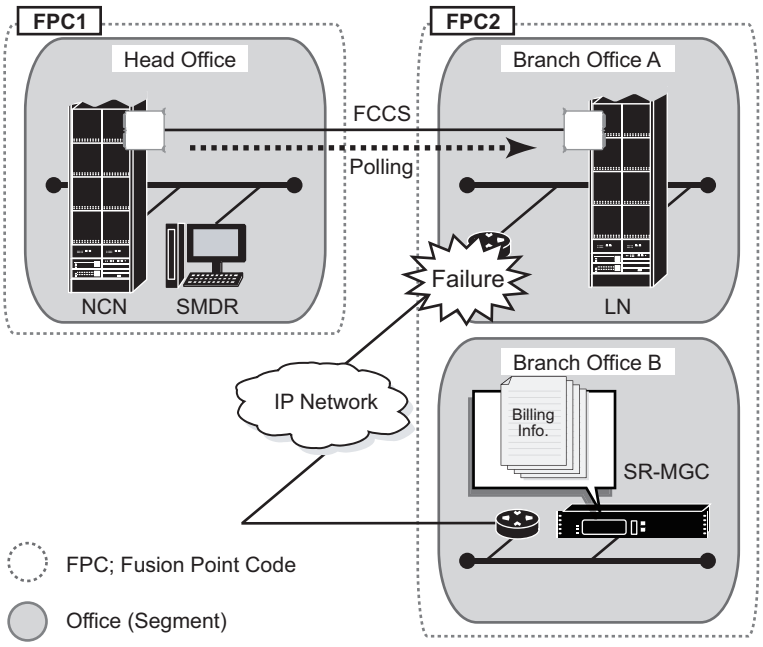
CENTRALIZED BILLING in a System with SR-MGC and FCCS

When LN to which SR-MGC is Registered is Failed

If SR-MGC starts operating, billing information is stored in the SR-MGC.

- A maximum of 47,000 calls can be recorded. If the number of records exceeds the maximum, the subsequent call records will be discarded.
- NCN keeps on polling LN regardless of the status of LN as conforming to the specifications of FCCS CENTRALIZED BILLING.
- NCN does not perform polling of SR-MGC for FCCS CENTRALIZED BILLING.

FCCS CENTRALIZED BILLING when LN is Failed

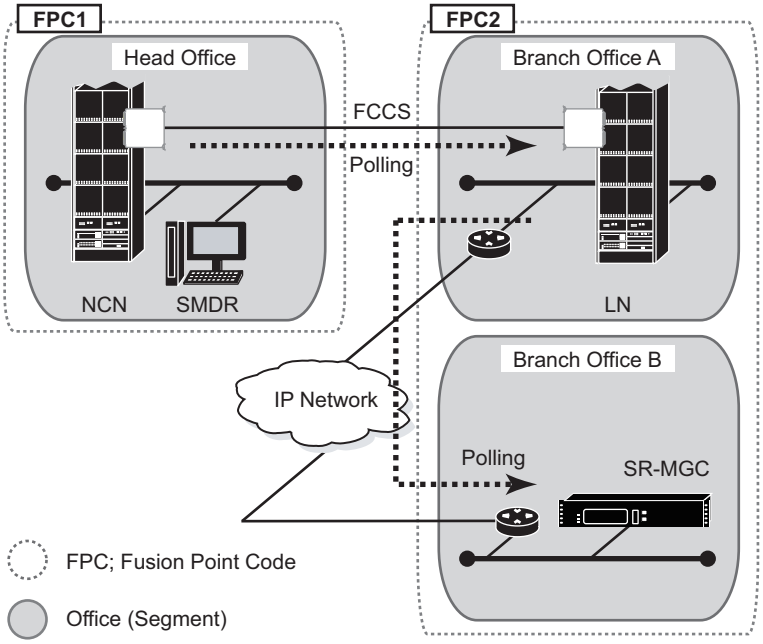


When LN is Recovered

When LN recovers, it starts polling of SR-MGC after receiving reply Health Check packets from the SR-MGC.

- A maximum of four units of information per polling can be exchanged between LN and SR-MGC.
- FCCS CENTRALIZED BILLING is executed by NCN even if the polling is performed between LN and SR-MGC.
- LN does not poll SR-MGC if the total amount of billing information exceeds the upper limit specified by ASYDL, SYS1, Index 586. When the value goes down below the limit, the polling will be resumed.

FCCS CENTRALIZED BILLING when LN is Recovered

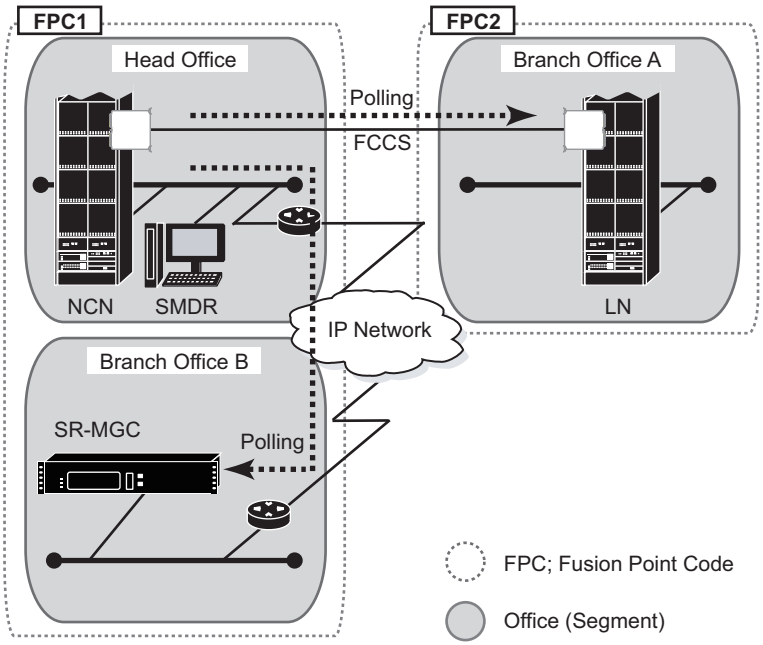


FCCS Centralized Billing when SR-MGC is installed in NCN

When SR-MGC is installed in NCN, it polls LN (polling for FCCS CENTRALIZED BILLING) and SR-MGC.

- NCN stops polling SR-MGC and LN (polling for FCCS Centralized Billing) if the total amount of billing information exceeds the upper limit specified by ASYDL, SYS1, Index 586. When the value goes down below the limit, the polling will be resumed.
- Priority is considered between the polling for FCCS CENTRALIZED BILLING and the polling for LN in the current specifications.

FCCS Centralized Billing when NCN is Failed

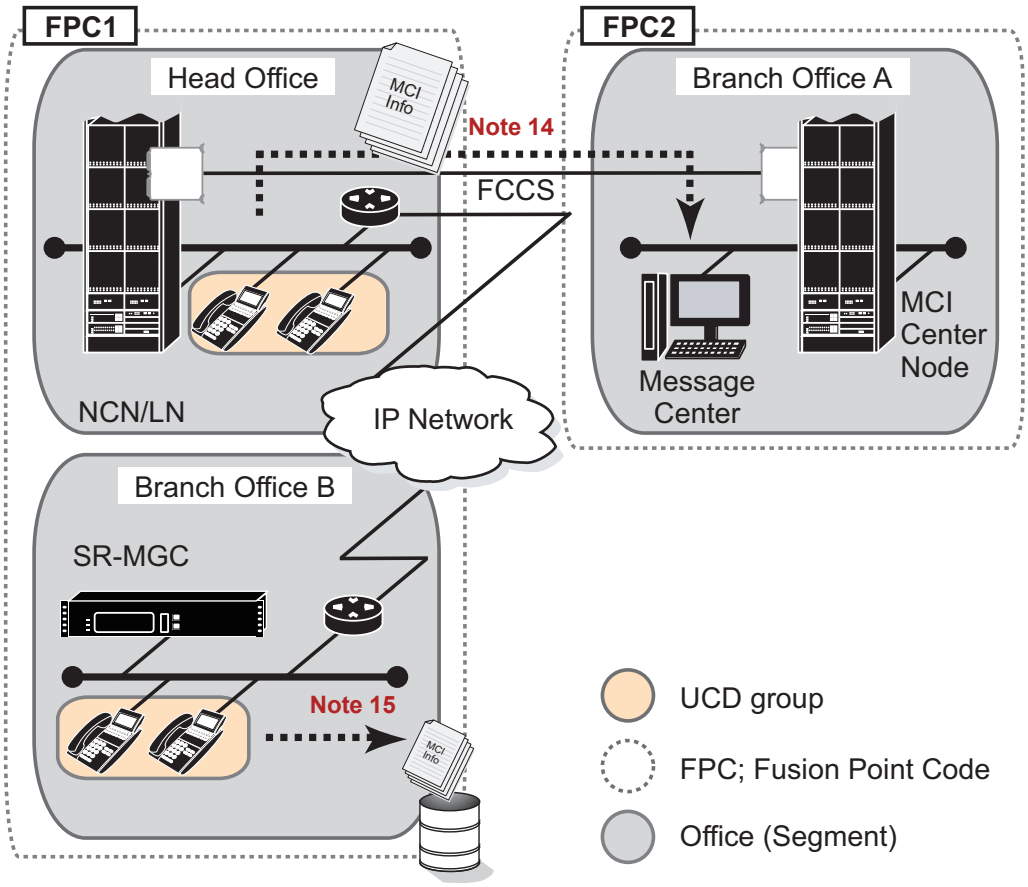


FCCS Message Center Interface (MCI) Features

FCCS MCI features are available when both FCCS nodes (Node 1 and Node 2) are in normal operation.

- MCI information created while the SR-MGC is in operation will be discarded.

FCCS MCI Features



Note: Either node in Head Office or Branch Office A can be NCN/LN.

Note 14: Head Office node information is sent to Branch Office A node that is specified as MCI Center Node.

Note 15: MCI information created while the SR-MGC is in operation will be discarded.

NCN and SR-MGC

Conditions for SR-MGC that backs up an NCN

The following is not available:

- SR-MGC cannot be used as an NCN even if the SR-MGC is placed into operation to back up the NCN.
- SR-MGC cannot change NDM-related office data.
- SR-MGC cannot perform Broadcast Control over LNs.

Others Conditions

- The basic operation of SR-MGC does not differ depending on the node (NCN or LN) to be backed up.
- The LM issue of SR-MGC is not necessary to be adjusted to an node even if the node is an FCCS node.

<Conditions on SR-MGC with FCCS CLUSTER>

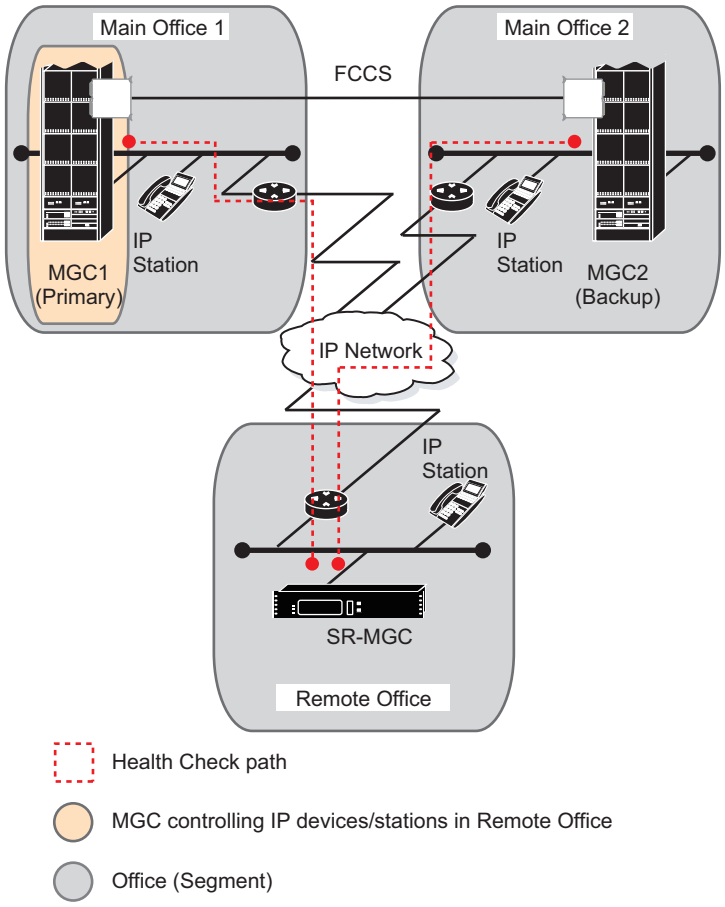
The following explains conditions on SR-MGC using the SR-MGC with FCCS Cluster. In this case, SR-MGC routinely checks the MGCs used in FCCS CLUSTER by Health Check. If a failure occurs in the network, SR-MGC starts to operate and keep the terminal's connections. The settings are as follows:

SETTINGS		
SR-MGC	Primary	MGC1
	Backup	MGC2
Terminals in Remote Office	Primary DRS	MGC1
	Secondary DRS	MGC2
	Tertiary DRS	SR-MGC

Note: "MGC" represents the Telephony Server here.

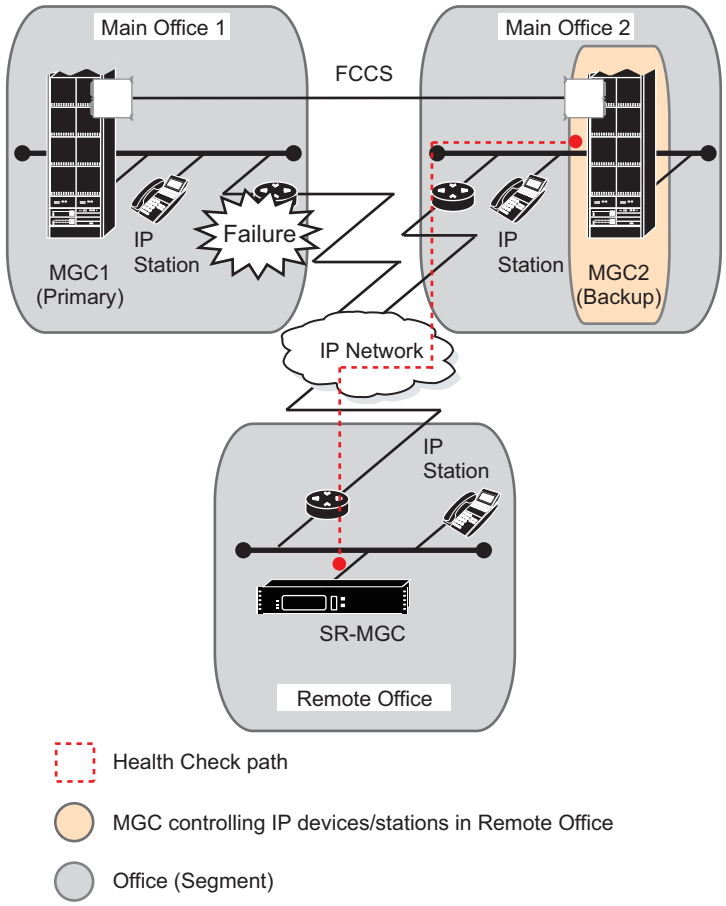
Normal Conditions

Note: Both MGC1 and MGC2 transmit Health Check request to SR-MGC in Remote Office. The SR-MGC sends back the Health Check responses to them.



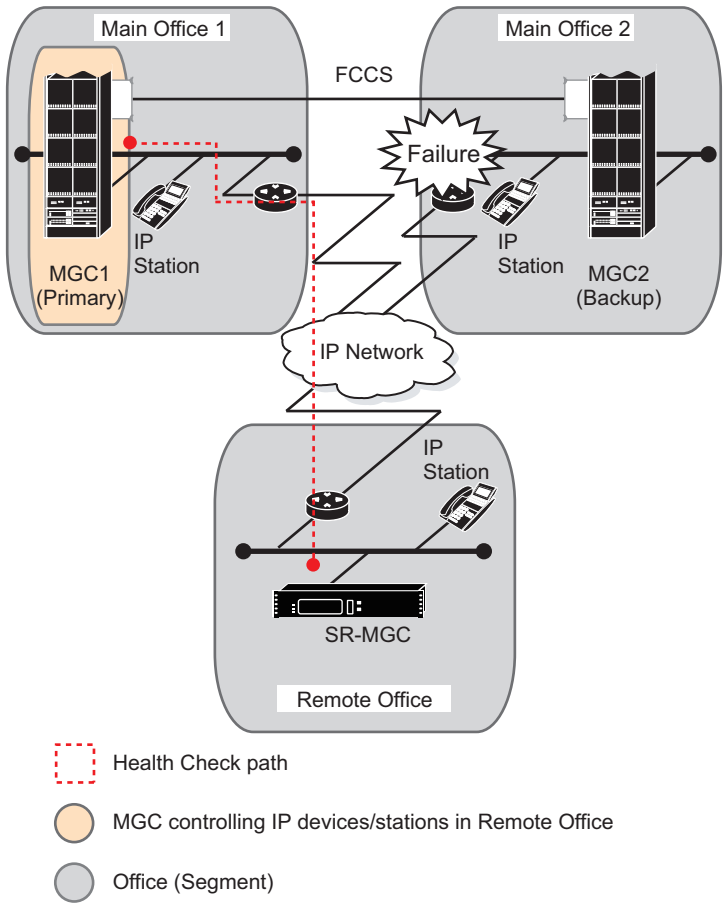
When network failure occurs in the Main Office 1

OFFICE NAME	EQUIPMENT	STATUS
Remote Office	SR-MGC	SR-MGC determines network failure might occur in Main Office 1 because of no Health Check request from MGC1. However, the request is transmitted from MGC2 that is the backup. Therefore, SR-MGC1 is not activated and it remains Standby mode.
	Terminals	The terminals register to MGC 2 by using FCCS Cluster feature.



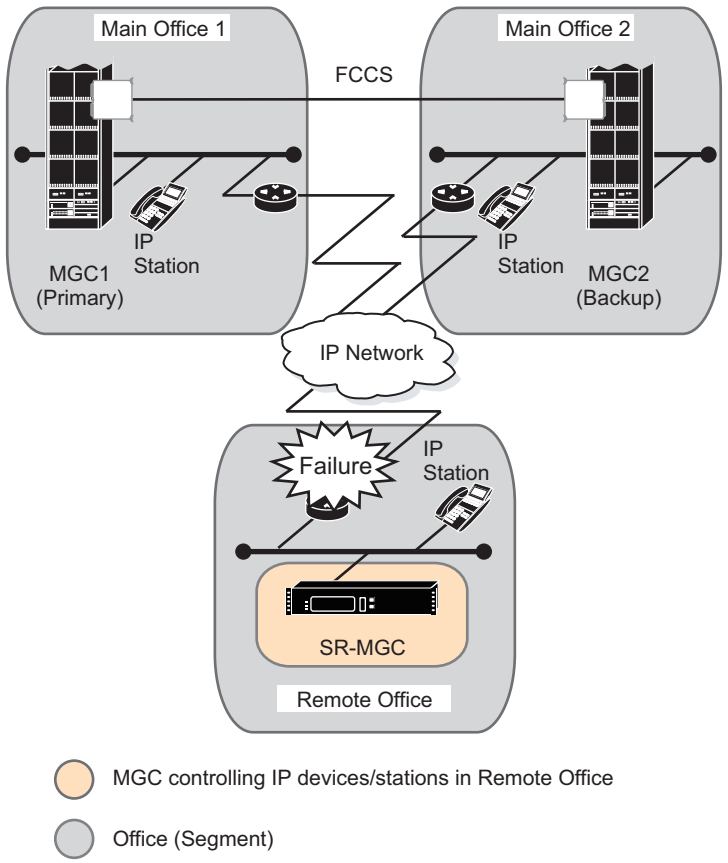
When network failure occurs in the Main Office 2

OFFICE NAME	EQUIPMENT	STATUS
Remote Office	SR-MGC	SR-MGC determines network failure might occur in Main Office 2 because of no Health Check request from MGC2. However, the request is transmitted from MGC1 that is the primary. Therefore, SR-MGC is not activated and it remains Standby mode.
	Terminals	The terminals are controlled by MGC 1 as usual.



When network failure occurs in the Remote Office 1

OFFICE NAME	EQUIPMENT	STATUS
Remote Office	SR-MGC	SR-MGC determines network failure might occur because of no Health Check request from MGC1 and MGC2. Therefore, SR-MGC changes its state to active mode to rescue the terminals in the Remote Office.
	Terminals	The terminals register to the SR-MGC.



<Conditions Specific to UMGi/RNIP System>

When UMGi/RNIP system is used as MGC, Remote Call Forwarding (RCF) Data cannot be used in SR-MGC; SR-MGC will be activated without receiving RCF data for PHS/PCS.

Register the MPH office data in the CCN or RN where are accommodated the terminals to be controlled by SR-MGC.

<Other Conditions>

None.

Programming

Perform the data settings on the Telephony Server (MGC) in main office. The Telephony Server (MGC) is required to recognize the SR-MGC in remote office by assigning the IP address, office name, and system data for Health Check. This explains each of those procedures as follows:

Data Assignment on the Telephony Server (MGC)

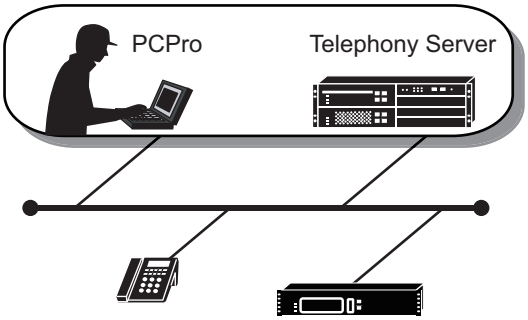
Data Assignment on SR-MGC

Data Assignment on Terminals

Data Assignment on the Telephony Server (MGC)

This item explains the procedures for setting up the Telephony Server (MGC). Log in to the Telephony Server (MGC) by PCPro and take the following procedures.

Note: We recommend to back up the data within this section to the CF card. Use the MEM_HDD command to back up the data. Configure the target data to be backed up according to your specification. Do not select “SR-MGC” in the Data Type Select.



[1] Use the **ASYDL** command

System Data 1

Index 843: Assign the settings for SP down detection mode.

bit 0-1: Detection mode for SP down

0/1=Detects when all SP-PHDs and SP-PHIs are down/Not detect when all SP-PHDs and SP-PHIs are down.

bit 5-7: The number of counts, which is recognized as SP down.

0: Two counts is applied
1-7: The number of counts

Indexes 844 and 845: Assign the number of SR-MGCs to which office data copy is executed per day.

Index 846: Determine the way of finding a pilot terminal (bits 0 and 1), range of LAN interface for changeover (bit 2) and enable /disable terminal reset function of the terminal registered in SR-MGC (bit 4).

bit 0: Specify a pilot terminal for connectivity check (PING), which can trigger switchback. **Note 1**

0/1=None/Search for Media Gateway

bit 1: Specify a pilot terminal for connectivity check (PING), which can trigger switchback. **Note 1**

0/1=None/Search for the terminals by LOC-ID

bit 2: Changeover to SR-MGC is performed on both LAN1 and LAN2 sides of the Telephony Server. **Note 2**

0=Only on LAN interface used by SR-MGC/NMS application port

1=On both LAN interfaces

bit 4: Enable/Disable Visitor key of terminals

0/1=None/Terminals registered in SR-MGC are able to perform terminal reset by using the Visitor key.

If the Telephony Server (MGC) is configured redundant, the decision as to which the Telephony Server (MGC) (MGC#0 or MGC#1) data will be copied depends on the following data assignment. However, Active-side data is always copied with the faster copy method regardless of this setting.

bit 7: Method of specifying the Telephony Server (MGC) when copying the office data.

0/1=MGC#0 **Note 3**/Active-side MGC **Note 4**

Note 1: This is available when Mode in the CMNT command is set to 2.

Note 2: Assign 1 for enabling the rescue by SR-MGC when a failure occurred only on LAN1 while MG(PRI) in Proprietary Protocol mode is operated on LAN2 port. The following conditions are required to enable this data setting:
- LAN1/LAN2 of the Telephony Server is assigned to different segments (IP addresses)
- LOC-IDs are assigned per SR-MGC

Assign 0 if the above conditions cannot be met. Otherwise, MG(PRI) in Proprietary Protocol mode may become unavailable due to the system conditions, such as during construction periods.

Note 3: When MGC#0 is down, MGC#1 is specified.

Note 4: Even if the data files in MGC#0 and MGC#1 are identical, the different time stamps may be written into the data files because Standby-side MGC data is backed up after the backup of the Active-side MGC data. Therefore, the office data copy after the system changeover is available for the data files with different time stamps even if the data files are not updated.

Index 847: Assign a time to execute forceful changeover against terminals.

bit 0-3: Timer Counter for forceful switchover

0: 70 minutes (default)
1-E: time assigned at intervals of 5 minutes
F: Forceful switchover is not executed while the active calls remain.

bit 4: Alarm is sent before forceful switchover is executed

0/1=Not alarmed/Alarmed per minute

bit 6: SR-MGC office data copy method

0/1 = Auto/Fixed to the existing method

Index 850: Assign SR-MGC Health Check cycle. **Note 5**, **Note 6**

00 HEX. = 30 seconds (default)
01 ~ 0F HEX. = 15 seconds (Not allowed to set)
10~ FE HEX. = 16 ~ 254 seconds

Note 5: It is recommended to set SR-MGC Health Check cycle for the default value (30 seconds) or more when more than 200 units of SR-MGC are assigned to the Telephony Server.

Note 6: To change the health check data with the ASYDL, SYS1, Index 850, follow the procedure explained in Command Manual. For details, see the note on ASYDL, SYS1, Index 850 in Command Manual.

Index 851: Assign Time-out to receive Health Check response. **Note 7**

00 HEX. = 5 seconds (default)
01 ~ 05 HEX. = 5 seconds
06 ~ FE HEX. = 6 ~ 254 seconds

Note 7: Be sure not to assign the Time-out timer exceeding the value of Health Check cycle assigned by ASYDL, SYS1, Index 850 (except when FF HEX. is assigned).

Index 852: Assign Error detection between the Telephony Server (MGC) and SR-MGC.

bits0-3: Determine the number of times of Health Check error to consider the communication failure between MGC and SR-MGC.

00 HEX. = Twice (default)
01 ~ 02 HEX. = Twice
03 ~ 0F HEX. = 3 ~ 15 times

Note: When changing the default value (twice), be sure to assign an identical value (the number of times of Health Check error) to all the nodes.

bit 4: System Message output for [10-F] Communication Failure and [10-G] Communication Recovery between MGC and SR-MGC.

0/1 = Out of service/In service

bit 5: IP terminal registration changeover from SR-MGC to the Telephony Server (MGC).

0/1= Manual/Automatic changeover

Indexes 853 and 854. Assign Port Number of SR-MGC Health Check (receiving side). **Note 8**, **Note 9**
Index 855. Assign CBT polling cycle from MGC to SR-MGC for SMDR Information management.

00 = 5 seconds (default)
01 ~ FF = 1 ~ 255 seconds

Indexes 856 and 857. Assign Port Number of SR-MGC Health Check (sending side). **Note 8**
Indexes 858 and 859. Assign Port Number of SMDR Information (receiving side). **Note 8**
Indexes 860 and 861.Assign Port Number of SMDR Information (sending side). **Note 8**
Index 862. Assign a cycle to check whether the SMDR information is stored on SR-MGC.

Note 8: Use **ADTM** command when changing the Default Port Number.

Note 9: The Port Number assigned by Indexes 853 and 854 must correspond to the system data in both the Telephony Server (MGC) and SR-MGC.

[2] Use the **ASYD** command

System Data 1, Indexes 87 and 88. Assign the time to perform the routine diagnosis.

[3] Use the **CMNT** command

Assign the SR-MGC data such as IP address, and office name.

<Setting Tab>

MGC No. [0-255]
Specify the SR-MGC number in the target remote office.
Office Name [Maximum 20 characters]
Enter the office name for corresponding with the MGC Number.
LP No. [0/4/8/12/16/20/24/28] **Note 10**
Select the corresponding Local Processor (LP) Number.

Note 10: This parameter is required when the remote office is configured by UMGi system.

IP Address
Enter the IP Address of SR-MGC.
Mode
This mode determines the timing when the changeover to SR-MGC is executed.

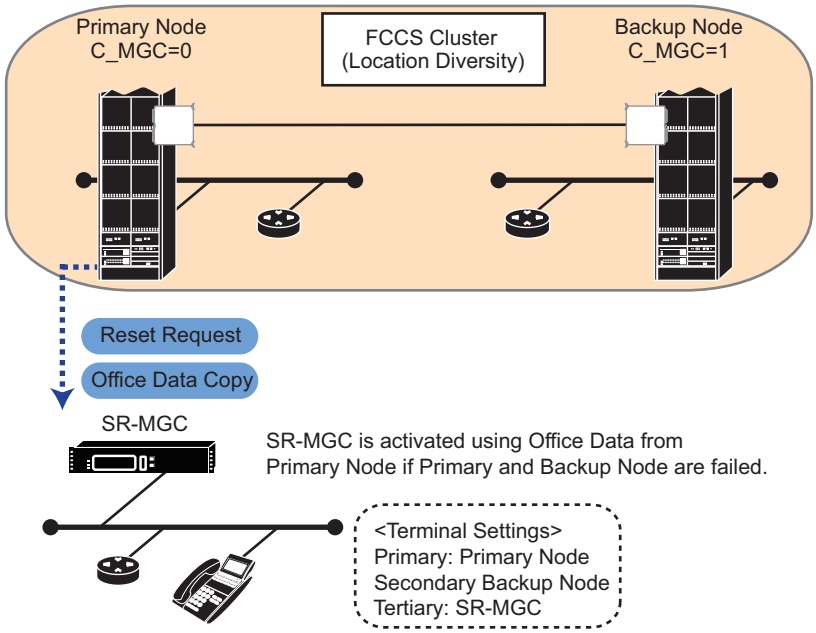
1= Automatically switched to SR-MGC operation when communication failure has been detected between the Telephony Server (MGC) and SR-MGC.
Automatically switched back to the Telephony Server (MGC) operation when communication recovery has been detected between the Telephony Server (MGC) and SR-MGC.
In FCCS Cluster network, if SR-MGC detects normality of communication between SR-MGC and Primary node or SR-MGC and Backup nodes, SR-MGC will switch the rescued terminals back to the recovered node.

2= Automatically switched to SR-MGC operation when communication failure has been detected between the Telephony Server (MGC) and SR-MGC.
Automatically switched back to the Telephony Server (MGC) operation when communication recovery between the Telephony Server (MGC) and SR-MGC, and normality of communication between the Telephony Server (MGC) and a selected terminal have been detected.
In FCCS Cluster network, if SR-MGC detects normality of communication between SR-MGC and Primary node, Primary node and a selected terminal, SR-MGC will switch the rescued terminals back to the Primary node.

C_MGC
Select an Operating Mode for SR-MGC. Set “1” in the Backup node side if SR-MGC is used with FCCS Cluster feature (SR-MGC with FCCS Cluster). **Note 11**
Note that this parameter is displayed only when LOCATION DIVERSITY feature is provided.
0= Normal Operation
Office data in the Telephony Server (MGC) can be copied to the SR-MGC.

1= This option must be selected in the Backup node when the SR-MGC is used with FCCS Cluster (SR-MGC with FCCS Cluster)
Office data in the Telephony Server (MGC) cannot be copied into SR-MGC.

Note 11: The following is an example using C_MGC when SR-MGC is used with FCCS Cluster feature.



<Initialize Tab>

Initialization of SR-MGC can be activated on this tab.
Click either of the radio buttons and press the Execute button to execute initialization on the target SR-MGC.

KIND: Specify the SR-MGC which will be activated for initialization.
Click either of the radio buttons and press the Execute button to execute initialization on the target SR-MGC.

Individual: Specify the MGC Number when this button is selected.
MGC No.: Enter a unique MGC Number. **Note 12**

ALL: Select this button to execute the initialization on all the SR-MGCs in the network.
MGC Kind Select: Select “SR-MGC”. **Note 13**

Note 12: Do not specify the operating SR-MGC.

Note 13: The SR-MGCs which are selected by “ALL” of “KIND” are the Standby mode SR-MGCs only.

<Change Operating Tab>

MGC Kind Select: Select “SR-MGC”.

FUNC: “Change Operating” is displayed in gray (fixed).

CONDITION: Conditions in SR-MGC can be selected. **Note 14**, **Note 15**, **Note 16**

To Operating mode: Switches SR-MGC to Operating mode
To Standby mode with terminal reset: Switches SR-MGC to Standby mode

KIND: Specify the SR-MGC, which will be activated if the Telephony Server (MGC) is failed.

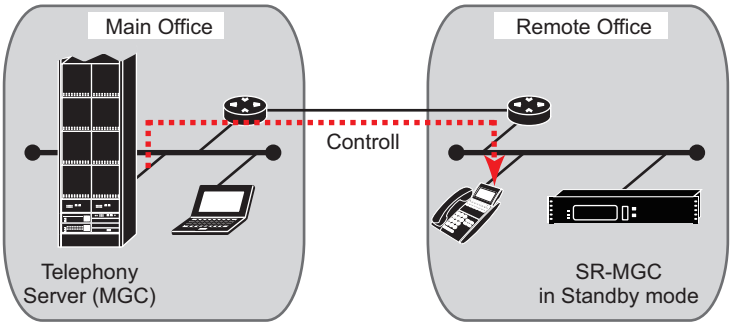
Individual: Specify the number of SR-MGC.
MGC No.: Enter a unique MGC Number.
ALL: Specify all the SR-MGCs.

Note 14: This parameter forcibly switches SR-MGC into Operating mode from Standby mode, which means SR-MGC allows terminals to register to SR-MGC. This parameter does not switch all the terminals registered in the Telephony Server (MGC) to SR-MGC.

Note 15: After the mode of SR-MGC has been switched by this parameter, the SR-MGC will be in Operating mode. To switch the SR-MGC into Standby mode, use “To Standby mode with terminal reset”. Note that the ASYDL settings (ASYDL SYS1 Index 852 Bit 5:IP terminal registration changeover) are not valid to this operation. When SR-MGC enters Standby mode, the Telephony Server (MGC) starts to operate.

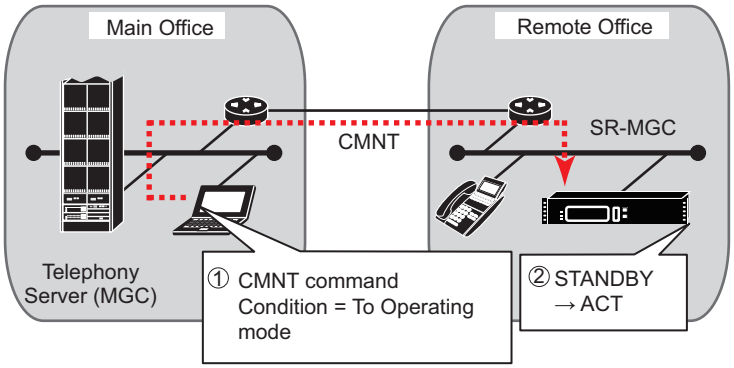
Note 16: The following are examples of using these parameters. If you use SR-MGC with FCCS Cluster (C_MGC = 1), see also the Service Conditions in SR-MGC with FCCS Cluster part of LOCATION DIVERSITY[L-55] in the Data Programming Manual - Business.

The Telephony Server (MGC) is located in Main Office, and SR-MGC is located in Remote Office. Terminals in Remote Office are controlled by the Telephony Server (MGC).



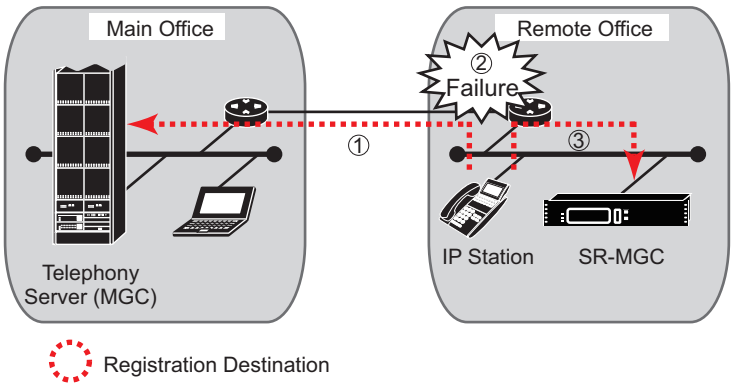
1. When “To Operating mode” is selected in CONDITION

SR-MGC is switched into Operating mode. All the terminals registered in the Telephony Server (MGC) are allowed to register to SR-MGC, regardless of network conditions. Note that it takes one cycle of Health Check time (30 seconds at default) to send a request.



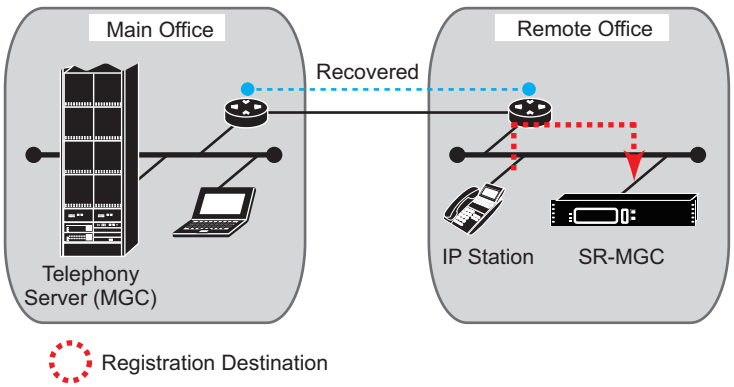
2. When fault occurs

When terminals located in Remote Office detect the network failure, Terminal Reset is performed and then the terminals re-register to SR-MGC.



3. When the Telephony Server (MGC) is recovered

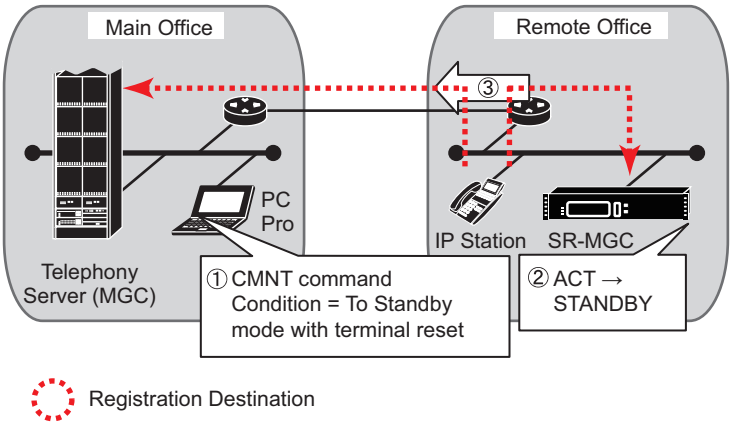
Even though the Telephony Server (MGC) is recovered from network failure, the setting (To Operating mode in CONDITION) keeps SR-MGC operating and controlling the terminals. (Terminal Reset will not be performed.)



Even when network fault and fault recovery often occur, the terminals are available using SR-MGC.

4. Back to normal operation

Select "To Standby mode with terminal reset" to switch SR-MGC into Standby mode, which sends Terminal Reset causing the terminals to re-register to the Telephony Server (MGC).



1. Use the **AKYD** command (Option)

   Assign the Visitor feature to the Line/Feature Key of the terminal, which makes it re-register to Primary Telephony Server (MGC).
   KYI=1 (Function Key)
   FKY:156 (Visitor)

2. Use the **ADSL** command (Option)

   Assign the Visitor feature to the Soft key of the terminal, which makes it re-register to Primary Telephony Server (MGC).
   SN=0 (Status Display)
   FKY=156 (Visitor)

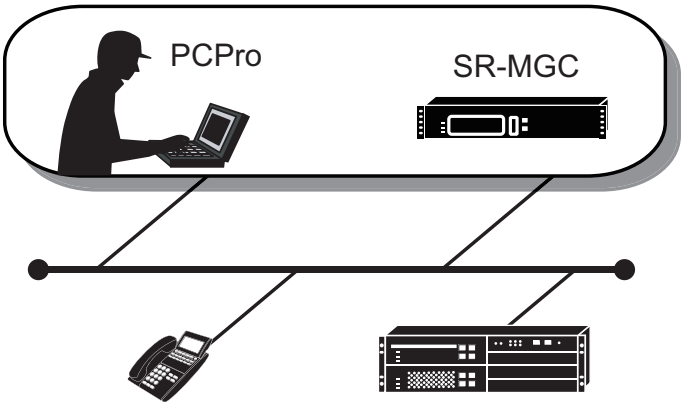
3. Use the **MEM_HDD** command

   Back up the office data into the Flash card in Telephony Server (MGC)

Note: Do not select SR-MGC in the Data Type Select.

Data Assignment on SR-MGC

This item explains the procedures for setting up SR-MGC. Log in to SR-MGC by PCPro.



[1] Use the **AIPA** command

Assign initial setting data including an IP address and transmitting speed on the SR-MGC.

<IP Address Setting Tab>

IP Address: Specify an SR-MGC's address.
Subnet Mask: Specify a subnet mask for the preceding address.
Default Gateway: Specify a router's address by which the SR-MGC's packets are routed.
MGC IP Address: Specify an MGC's (Telephony Server's) address to which the SR-MGC belongs. **Note 1**, **Note 2**

Note 1: Specify a LAN1 address (ACT) of the MGC.

Note 2: When UMGi/RNIP system is used, specify a LAN1 address (ACT) of the CCN.

<NIC Setting Tab>

Speed: Specify transmitting speed between the SR-MGC and a connected network device.
Duplex Mode: Specify a duplex mode between the SR-MGC and a connected network device. **Note 3**

Note 3: The setting is necessary only when specifying Fixed 10Mbps or Fixed 100Mbps in Speed.

[2] Use the **ATIM** command

Assign clock information on the SR-MGC.

<Date & Time Tab>

DATE: Specify a current date: Year/Month/Day.
DAY: Displays a day of the week depending on the preceding date.
TIME: Specify a current clock, 00:00 through 23:59.
Time Zone: Specify a time zone to which the SR-MGC belongs, GMT-13:00 through GMT+14:00.

<Daylight Savings Time Tab>

START/END
KIND: Select either of Date or Day.

[When selecting Date in KIND]

- Every Year: Checking it eliminates the following Year field.
- Month
- Day: Specify a day of the month.
- Year
- Hour
- Minute

[When selecting Day in KIND]

- Every Year: Checking it eliminates the following Year field.
- Month
- Day: Specify a day of the week.
- Year
- Hour
- Minute

[3] Use the **ASPC** command

Assign a SIP server ID on the SR-MGC.

Note: Perform the following only.

<SIP server ID>

System: Select System-0 (fixed).
Input Mode: Select Standard.
Function: Select SIP Server.
SIP Server ID: Specify a SIP server ID of the SR-MGC, 1 through 127. **Note 4**

Note 4: Enter the same value as a SIP server ID assigned on the MGC with the ASSDN/L command.

[4] Use the **MEM_HDD** command

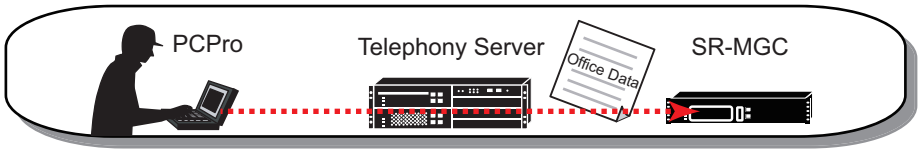
Back up the data set in the SR-MGC: select Memory to Hard Disk in Direction Select; check Data Memory and SR-MGC Data in Data Type Select.

Note: Perform Office Data Load & System Initialize with the SINZ command to enable data you assigned with the AIPA command.

[5] Copy Office Data by using the **CMNT** command

Log in to the Telephony Server (MGC) by PCPro and copy Office Data from the Telephony Server (MGC) to SR-MGC. Do not turn off the power of the SR-MGC during the Office Data copy. **Note 5**, **Note 6**

CMNT command provides SR-MGC with File Control tab. You can use this tab to copy a program in MGC to a SR-MGC after the program in MGC had been updated as is in the same version.
Execute CMNT command with PCPro; select "Program Copy" in PROCESS TYPE and a SR-MGC number in the grid; click the Execute button.



Note 5: When the Break button is pressed during office data copy, the operation is cut off after the currently copying file is completely copied. However, when DM/LDM/NDM copy is executed, the operation is cut off after all the DM/LDM/NDM files are copied.
File copy operation takes about two minutes per file. (The time varies depending on the connection status of the network.)

Note 6: If you power off the SR-MGC during the office data copy, check the data as follows:
STEP1: Use CMNT command to check the normality of communication between SR-MGC and the Telephony Server (MGC).
STEP2: If the status of the communication is normal, copy the Office Data into SR-MGC. If the communication is not established, set network settings of SR-MGC, and then copy the Office Data into SR-MGC.
When none of the above is performed at the situation, it is not guaranteed that SR-MGC operates normally.

<File Control Tab>

PROCESS TYPE: Processing type
Select the processing type to be executed.

- Office Data Copy & Load: Copies and loads the Office Data.
- Program Copy: Copies the program.
- Program Switch: Switches the program. **Note 7**

Note 7: Do not turn off SR-MGC during the program switch process. SR-MGC will not start up after it is turned off.

Appoint Stop Time **Note 8**
Check this box when the program copy needs to be stopped at the specified time.

- Time: Time to Stop the Program Copy [00:00-23:59]
  Specify the time to stop the program copy.

Restrict Concurrent Number **Note 8**
Check this box to restrict the number of SR-MGCs to allow concurrent copying.

- Concurrent Number: Parallel Copy Number [1-255]
  Specify the number of SR-MGCs to allow concurrent copying.

Note 8: This parameter is only displayed in the case of PROCESS TYPE = Program Copy is set.

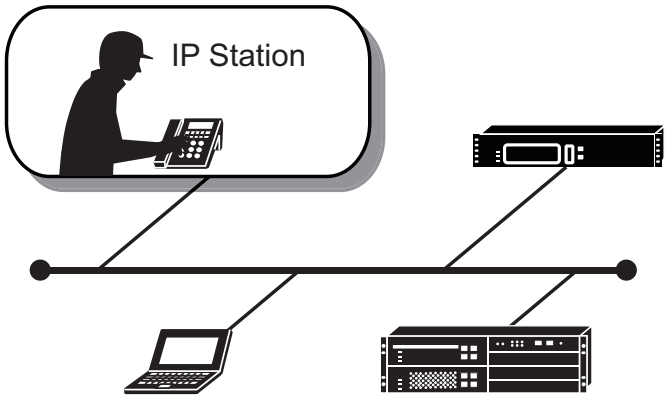
Select MGC No.
Select target SR-MGC to execute the process such as selected with the PROCESS TYPE parameter such as Office Data Copy & Load, Program Copy or Program Switch.

- Specifying SR-MGC:
  Specify the target SR-MGC number (0-255), the box of specified SR-MGC number is displayed in yellow. And then click the Execute button.
  To undo specifying SR-MGC, click the box again.
- All SR-MGC (en bloc specifying):
  Click the Select All button to execute office data or program copy to all the SR-MGCs in the network.

Note: Do not perform any PCPro operations other than CMNT command during office data copy. The PCPro operation such as login/logoff can cause a copy failure.

Data Assignment on Terminals

After SR-MGC and the Telephony Server (MGC) have been installed, the settings of each terminal are necessary to use the SR-MGC. The following must be assigned by each terminal.



- In the case of DtermIP(SIP)

[1] Select “2. SIP Settings” on the main menu.
[2] Select “2. Server Address & URI”.
[3] Select a server type and enter an IP Address (or URL) of SR-MGC.
[4] Select “OK”, and then select “Save” on the main menu.

- In the case of DtermIP(Proprietary Protocol)

[1] Select “3. BackUp-DRS Address” on the main menu.
[2] Select a server type and enter an IP Address of SR-MGC.
[3] Select “OK”, and then select “Save” on the main menu.

If Visitor key is set to your terminal, the terminal will be the following conditions when registered in SR-MGC.

<Line/Feature Key>
The key lights steadily

<Soft key>
The key flashes (red).

Under the above conditions, pushing the Visitor key will switch the terminal back to the Telephony Server (MGC).

S-143 Software Multiple My Line Appearance

General Description

This feature permits a Desktop terminal user to assign the same My Line numbers to the Sub Lines, which will be able to originate and terminate a call by the Sub Lines while another call is being originated or terminated by the My Line. A Sub Line assigned to the same My Line number is mentioned as the Same Number Sub Line afterwards.

Operating Procedure

To make a call using the Same Number Sub Line

1. When making a call, press a Sub Line key you have registered as the Same Number Sub Line. LCD on the called terminal displays the My Line number of the originating terminal.

To receive an incoming call on the Same Number Sub Line

1. When an incoming call arrives at a Sub Line you have registered as the Same Number Sub Line while your are talking, hold the in-progress call and press the Sub Line key.

Service Conditions

1. When this feature is activated with the PA-16ELCJ-B/PA-16ELCN/PA-16ELCN-A card, the following hardware setting is required.

   - SW10-1:
     ON: Previous LP-PM interface
     OFF: New LP-PM interface (When inter-frame multiple My line is used)

   - SW10-2= Fixed to OFF

2. When this feature is activated with the PA-16ELCJ-B/PA-16ELCN card, the following firmware is required.

   SP-3514 16ELCJB PROG-B or later (Firmware) / SP-3956 16ELCN PROG-A or later (Firmware)

3. The maximum number of Same Number Sub Line per Desktop terminal is four lines. When multiple Sub Lines have been registered on a terminal, the incoming call arrives at them in a sequential order.
4. The Same Number Sub Line can be registered to G24-31.
5. Uniform Call Distribution (UCD):
   a.) If the Desktop terminal belongs to NDM/LDM, UCD call distribution is available on My Line only.
   b.) If the Desktop terminal belongs to DM, UCD call distribution is available on My Line and all the Same Number Sub Lines. **Note 1**

   Note 1: If the Desktop terminal belongs to DM, the following must be considered. To work around the following, the Desktop terminal must belong to NDM/LDM.
   - A maximum of 100 Stations can be included in a Group.
   - Same Number Sub Line is counted as a Station.
   - If My Line and all the Same Number Sub Lines are busy, incoming calls are forwarded to other Station in the Group.

6. When the Desktop terminal (My Line) accommodating the Same Number Sub Line is included in a Station Hunting Group:
   a.) If the Desktop terminal belongs to NDM/LDM, Station Hunting - Terminal/Circular/Priority/Priority with Switchback is available on My Line only. **Note 2**

   Note 2: If the Desktop terminal (My Line) accommodating the Same Number Sub Line is assigned as a Pilot Station, incoming call is forwarded to the next Station in the Group when My Line and all the Same Number Sub Lines are busy.

   b.) If the Desktop terminal belongs to DM, Station Hunting features are available on My Line and all the Same Number Sub Lines. **Note 3**

   Note 3: If the Desktop terminal belongs to DM, the following must be considered.
   - A maximum of 100 Stations can be included in a Group.
   - Same Number Sub Line is counted as a Station.
   - For Station Hunting - Terminal/Circular:
     Incoming Calls are forwarded to the next Station when My Line and all the Same Number Sub Lines are busy.
     For Station Hunting - Priority/Priority with Switchback:
     Not available (Station Hunting - Priority/Priority with Switchback feature is originally activated via Telephone Number only)

7. When the service is provided to Desktop terminal in a group of Call Pickup

   When the Desktop terminal picks up a Station from the group
   a.) The call can be taken by using My Line and Sub Line.

   When the Desktop terminal is hunted by a Station from the group
   a.) When the Desktop terminal belongs to NDM/LDM, the call can be answered by using only My Line.
   b.) When the Desktop terminal belongs to DM, the call can be answered by using My Line and Sub Lines.
8. Camp-On from other Stations can be set to My Line, and the settings can be specified by the My Line. (Not Sub Line)
9. Name Display is supported on the Same Number Sub Line.
10. When a call with phantom number arrives at a Desktop terminal, if the My Line is being busy, the call will be transferred to the Same Number Sub Line.
11. When Call Forwarding has been set to the My Line, the Call Forwarding will be taken preferentially.

12. Transfer settings can be specified only to My Line. The actions are as follows:
    a.) Call Forwarding-All Calls: All the incoming call are transferred.
    b.) Call Forwarding-Busy Line: A call is transferred when all the lines are busy.
    c.) Call Forwarding-Don't Answer: All the incoming call are transferred.
13. 8/16-party conference with the My Line and the Same Number Sub Lines cannot be made.
14. Override
    a.) When another Station overrides a Station using this feature only the My Line can be taken by the call (The Same Number Sub Line cannot be taken.)
    b.) When a Station using this feature overrides another station, the My Line or the Same Number Sub Line either can be used to call it.
15. The Same Number Sub Line can be selected by Line Preference.
16. BLF lamp lights on when all of the My Line and the Same Number Sub Line are occupied.
17. In Hotel system, this feature is available for Administration Stations.
18. OAI/ACD System does not support this feature.
19. RSC and SFC of the Same Number Sub Lines are the same as those of My Line.
20. Same Number Sub Lines can be assigned to My Line only. It cannot be assigned to another Desktop terminal's Sub Line.
21. In Call Back service, only My Line can be registered to two-type extension that can specify the call back or can be specified as the target extension.
22. This feature is available via FCCS network.
23. The same number Sub Line can be assigned only to the Line/Feature Keys of the Desktop terminal. DSS Console is not available.

Interactions

1. Interactions are listed on the next page. These features are available in station-to-station speech.

SERVICE CONDITIONS WITH OTHER SERVICE

CATEGORY	SERVICE	MULTIPLE MY LINE APPEARANCE	
		MY LINE	SAME NUMBER SUB LINE
Station-to-Station Features	Off-Hook Alarm	X	X
	Add-On Conference - 8 Party	X	X
	Call Forwarding-All Calls-Outside	X	**Note 4**
	Call Forwarding-Don't Answer-Outside	X	**Note 4**
	Call Forwarding-Busy Line-Outside	X	**Note 4**
	Call Forwarding-All Calls	X	**Note 4**
	Call Forward All/Busy/No Answer Clear	X	-
	Blind Transfer	X	X
	Emergency Call	X	X
	Day/Night Class Of Service	X	X
	Call Pickup-Group	X	X
	Call Waiting	X	N
	Call Pickup	X	X
	Call Hold	X	X
	Three-Way Calling	X	X
	Exclusive Hold	X	X
	Timed Reminder	X	N
	Call Pickup-Direct	X	X
	Step Call	X	X
	Call Transfer-All Calls	X	X
	Call Pickup - One Touch Key	X	N
	Split Call Forwarding - PS	X	**Note 4**
	Hot Line	X	N
	Periodic Time Indication Tone	X	X
	Call Forwarding-I'm Here	X	-

SERVICE CONDITIONS WITH OTHER SERVICE

CATEGORY	SERVICE	MULTIPLE MY LINE APPEARANCE	
		MY LINE	SAME NUMBER SUB LINE
Station-to-Station Features	Class of Service-Individual	X	X
	Camp-on Busy	X	-
	Executive Right-of Way	X	**Note 5**
	Station Hunting	X	X
	Call Back	X	-
	Flexible Numbering of Station	X	N
	Dual Station Call	N	**Note 6**
	Station 5dB Pad	X	X
	Boss Secretary	X	N
	Call Forwarding-Don't Answer	X	**Note 4**
	Call Forwarding-All Calls-Announcement	X	**Note 4**
	Priority Call	X	X
	Line Lockout	X	X
	Automatic Recall	X	X
	Telephone Number on FCCS Network **Note 9**	X	-
	Call Forwarding-Busy Line	X	**Note 4**
	Call Forwarding - Busy Line Override	X	N
	Privacy-D	X	N
	Remote Call Forwarding Control	X	X
	Call Hold - Conference	X	**Note 7**
	Voice Call	X	N
	Exclusive Hold	X	N
	Call Forwarding Memory Clear	X	-
	Automatic Call Back Cancel	X	-
	Timing Start	X	-

SERVICE CONDITIONS WITH OTHER SERVICE

CATEGORY	SERVICE	MULTIPLE MY LINE APPEARANCE	
		MY LINE	SAME NUMBER SUB LINE
Trunk Outgoing Features	Alternate Routing	X	X
	Speed Calling-Station	X	X
	Direct Outward Dialing (DOD)	X	X
	Speed Calling-System	X	X
	Toll Denial/Toll Diversion	X	X
	Pushbutton-To-Rotary Conversion	X	X
	Toll Restriction -3/6-Digit	X	X
	Outgoing Trunk Queuing	X	N
	Off-Hook Queuing	X	N
	Miscellaneous Trunk Restriction	X	X
Trunk Incoming Features	External Music On Hold	X	X
	Direct-In Termination (DIT)	X	N
	Power Failure Transfer	-	-
	Slumber Time-Do Not Disturb	X	N
	Distinctive Ringing	X	X
	Automated Attendant	X	X
	Direct Inward Dialing (DID)	X	X
	Music-On-Hold	X	X
	Do Not Disturb-D	X	N **Note 8**

Note 4: Although the settings are unavailable, an incoming call is transferred to the destination set by My Line.
Note 5: Interrupting another extension is available but interrupted by another extension is not available.
Note 6: If both of the two extensions in My Line are in idle, they can be called at the same time.
Note 7: Not available by using the access code. Use a Line/Feature key (FKY=116).
Note 8: If My Line, which has set Do Not Disturb-D, is in busy status, incoming calls terminate to Same Number Subline.
Note 9: Do not assign the same Telephone Number used for Physical Station Number of My Line to other Stations.

Programming

STEP 1: **AKYD** - Assign the Station according to the following parameters:

TN: Tenant Number

STN: Station Number

PRI: Priority Of Answering Call [0-4]

PL TN: Prime Line Tenant

PL STN: Prime Line Station Number

S = 0

MWD: Message Waiting Data Display Selection

0: MW data display on the top line

1: MW data display on the bottom line

LN PRE:Line Preference For Speaker Button [0-3]

0: Prime Line pickup

1: Customize (Line Preference data are to be set for the other buttons)

SPK: Line Preference For Speaker Button [0-3]

0: Prime Line Preference

1: Idle Line Preference

2: Incoming Line Preference

3: Ringing Line Preference

ANS: Line Preference For Answer Button [0-2]

0: Prime Line Preference

1: Incoming Line Preference

2: Ringing Line Preference

ORG: Line Preference For Originating Button

0: Prime Line Preference

1: Idle Line Preference

TP: Type of Desktop Terminal

DATA	Dterm300 Series	Dterm Series *i*
0	2/6/12–button type	4/8/16–button type
1	24–button type with Add on Module	32-button type with Add on Module
2	2/6/12–button type + FLEX keys	4/8/16–button type + FLEX keys
3	24–button type with Add on Module + FLEX keys	32-button type with Add on Module + FLEX Keys

KYN: Key Number [1-40]

KYI: Service Index=2 (Multi Line Key)

KD: Line Kind

0: Line

1: Intercom
2: DSS Key
3: Not Used
4: Same Number Sub Line

Tenant Number of Multi-Line

When KYI = 2, the tenant assignment of the Multiple My Line Appearance must be the same tenant as the Prime line.

When KD = 4, the tenant same with the extension number is registered (Not Available)

MultipleLine EN:Accommodated Location of the Same Number Sub Line.

This parameters is valid when KD = 4.

STEP 2: **ASFC** - Assign the service feature restriction class of this feature.

D/N: Day/Night Mode

TN: Tenant Number [1-63]

SFI: Service Feature Index = 212 (Software Multiple My Line Appearance)

SFC: Service Feature Restriction Class [0-15]

RES: Restriction Data

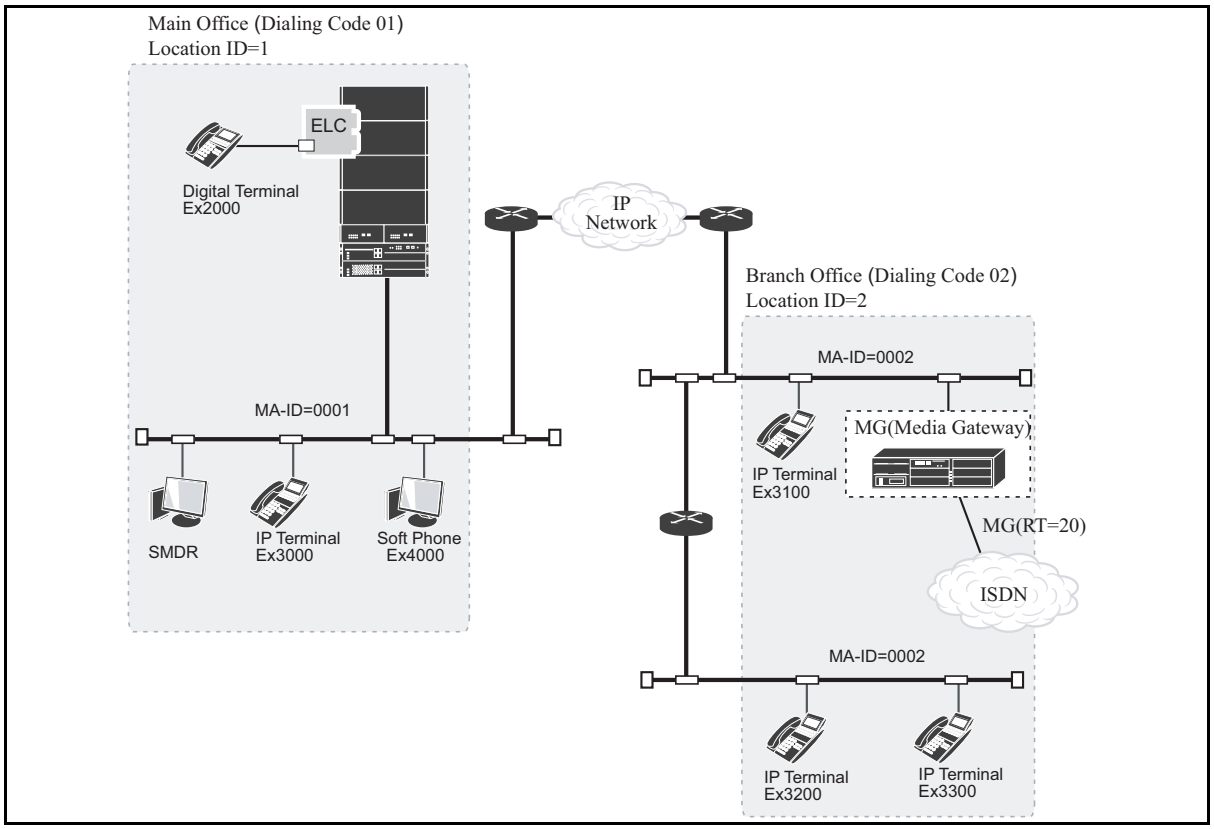
0: Restricted

1: Allowed

S-144 SMDR Multi Area ID via IP

General Description

This feature allows the system to record the billing information that terminal user makes a call via Media Gateway (MG). To perform this feature, each terminal must have an identification called MA-ID. This MA-ID is actually assigned to each location connected via IP network by using AMAA command. When the terminal in the network calls a party, it sends the location-assigned ID as the MA-ID so that the Station Detail Recording System (SMDR) can figure it out as the billing information. This feature is available when multiple Layer 3 networks are managed in the network. Note that MA-ID must be assigned to each location (called a Base) connecting IP network.



In IP network, assigning the MA-ID to each Base can create the billing information over the IP network. However, Station-to-Station SMDR is invalid, a speech between the same MA-ID terminals does not record any information.

- In the case that Extensions 2000 and 3000 call Extension 3100 having MA-ID=0002, the following are recorded after the call is terminated:

  (a) Extension 3000: MA-ID=0001 (assigned to a network that Extension 3000 belongs) and MA-ID=0002 (assigned to a network that Extension 3100 belongs) are recorded.

  (b) Extension 2000: MA-ID=0001 (assigned to a network that the Telephony Server belongs) and MA-ID=0002 (assigned to a network that Extension 3100 belongs) are recorded.

- In the case that Extension 3000 performs a trunk outgoing call (MG RT=20), MA-ID=0001 (assigned to a network that the Extension 3000 belongs) and MA-ID=0002 (assigned to a network that MG belongs) are recorded after the call is terminated.

Service Conditions

<Important Notices>

1. This feature requires the MA-ID assigned to each network segment.
2. All the network segment that contains the extensions and the trunks must be assigned MA-ID.
3. An unregistered network sends 0 as MA-ID to SMDR.
4. All network segments that have Stations and/or Trunks require to assign the MA-ID. If MA-ID is not assigned, MA-ID = 0 is recorded in SMDR when the terminal hangs up.
5. The same MA-ID can be assigned to multiple network segments.
6. Multiple MA-IDs cannot be assigned to one network segment.

<Interactions with Other Features>

1. A telephone using Sub Line sends its MA-ID.
2. When a call is made using a Sub Line or Virtual Line, the MA-ID of the My Line is passed.
3. When Centralized Billing-CCIS is used, MA-IDs are not recorded in SMDR.
4. In Hotel System, the billing information (including MA-ID) is not provided as [14-X] message to Property Management System (PMS).
5. MA-IDs are not included in a CS report.
6. The MA-ID is not added to a Status Monitor Facility Notification message (SMFN) of OAI.

<Hardware/Software>

1. This feature is valid in the SMDR of IMX format.
2. Non-IP terminal sends the network MA-ID that the Telephony Server connects.
   Note that network MA-ID is assigned to network via an external LAN by AMAA command.
3. PS sends the MA-ID of hunted CS.

<Interactions with Networks>

1. This feature is valid when multiple Layer 3 networks exists.
2. When the location of a terminal is changed (i.e from Head Office to Branch Office), the MA-ID of the current location is recorded in SMDR.
3. An FCCS network requires the MA-ID added to the SMDR text (ASYDL, SYS1, Index 589, B0=1).
4. A unified SFC table should be used in an FCCS Network.
5. When this feature is applied to an FCCS network, MA-ID must be assigned to all nodes and the network.

<Conditions Specific to This Feature>

1. An IP terminal MA-ID to be sent is an MA-ID assigned to a network where the IP terminal belongs.
2. This feature creates an SMDR record only for a Station-to-Station call between different offices (with different MA-IDs) via IP network.
3. By assigning system data (ASYDL, SYS 1, Index 589, Bit 3 = 1), SMDR output is provided when an ongoing call is transferred to another terminal that has a different MA-ID. Depending on terminal operations and the system data, billing is performed as follows.

OPERATION		When INDEX 589 Bit 3 = 0	When INDEX 589 Bit 3 = 1
Release while Talking		Billing ends.	• Billing ends.
Hold		Billing ends.	• Billing continues.
Transferring Operation (e.g. hooking)		Billing ends.	• Billing continues.
Blind Transfer	Intermediate Station Releases	No Billing	• Billing for the Original Call ends.
	Transfer Destination Answers	No Billing	• Billing for the previous party and the Transfer Destination starts.
	Intermediate Station Answers to Recall	No Billing	• Billing for the Original Call starts.
Call Transfer	Transfer Destination Answers	No Billing	• Billing for the Original Call continues. • Billing for the Intermediate Station and Transfer Destination starts.
	Intermediate Station Releases	No Billing	• Billing for the previous parties ends. • Billing for the Intermediate Station and Transfer Destination ends. • Billing for the previous party and the Transfer Destination starts.
Call Transfer is canceled.		No Billing	• Billing continues.

Note: Intermediate Station indicates the station that transferred the call. Original Call indicates the telephone call before the call transfer.

4. The following are conditions for billing after transferring a call over IP by assigning a system data (ASYDL, SYS1, Index 589, Bit 3 =1).

   - Split billing is applied for generating SMDR after transferring a call.
   - When a call is placed on hold by Call Park or by a Station having Multiple Line, SMDR continues for the first-answered station if the call is retrieved from other terminal.

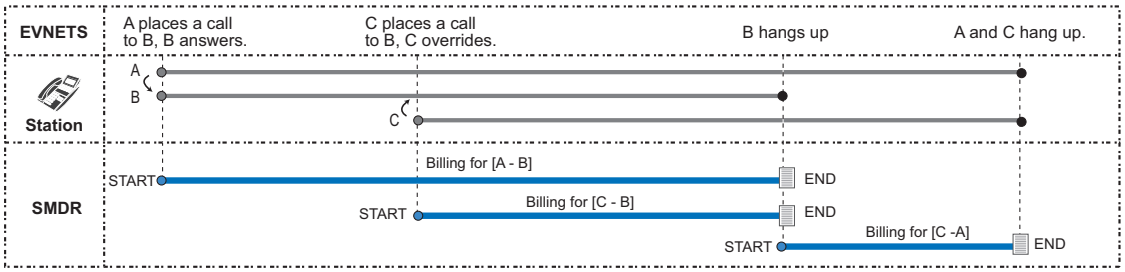
5. When a call from Station A is placed on hold by Station B and Call Park has been set, the SMDR is generated as follows when the held call is retrieved by Station C that belongs to the same Call Park Group.

OPERATION	SMDR
Station C retrieves the held call.	Billing output for: Stations A (Calling Party) and B (Called Party)

OPERATION	SMDR
Stations A and C talking, and one of the parties hangs up.	Billing output for: Stations A (Calling Party) and C (Called Party)

6. When a call is transferred by Blind Transfer, the billing starts after the intermediation station is released. If the Intermediate Station is recalled because Transfer Destination does not answer to the transferred call, the billing for the original call will start from when the intermediation station answers to the recall.
7. When a station is recalled by Call Hold, the billing starts when the station is recalled.
8. When an ongoing call changes to Three-party Conference, the previous billing continues for Two-way calling.
9. When a Three-way Calling is established by Executive Right-Of-Way, new billing starts for the overriding party and the overridden party. The billing for original Two-way call continues.
10. When a Tree-way Calling is established by Executive Right-Of-Way, and if the overridden party hangs up, the following two SMDR texts are output. (A) Billing for the overridden party and the original opposite party, (B) Billing for the overridden party and the overriding party. In addition, new billing starts for the original opposite party and the overriding party.

The following shows time chart for issuing SMDR information.



11. When eight-party conference is established, the billing is generated as follows.

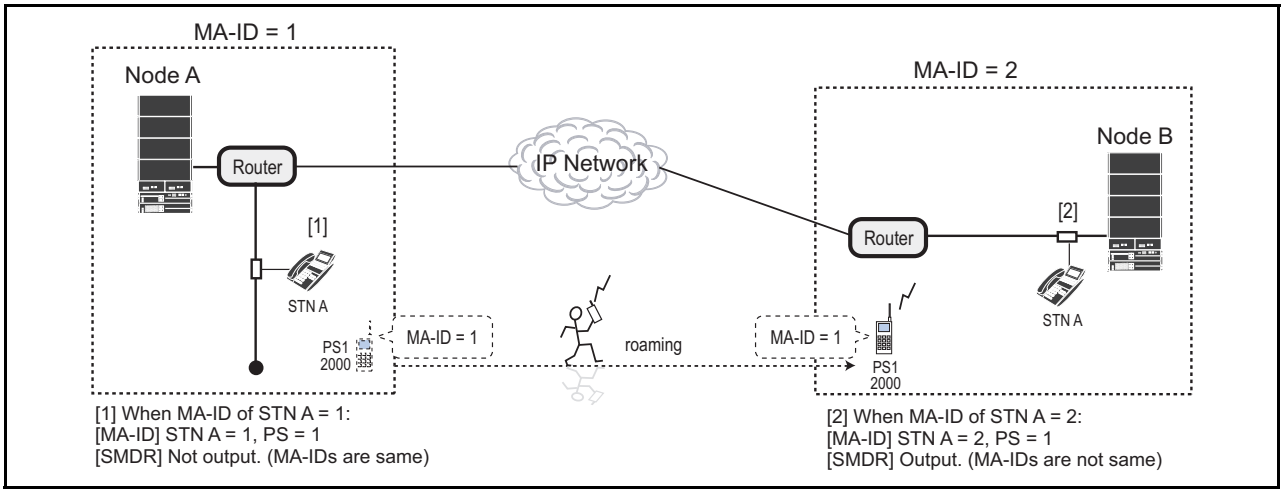
<Add-On>

OPERATION	Calling Party	Called Party
Conference Leader	Conference Leader's Number	8-party Conference Pilot Number
Conferees	Conference Leader's Number	Conferee's Number

<Meet-Me>

OPERATION	Calling Party	Called Party
Conferees	Conferee's Number	Conference Telephone Number

12. When a PS performs roaming while talking, the MA-ID used when the PS started talking is applied to the billing information.



13. In the offices not using Call Transfer Event Facility (ETF) that the OAI system uses, set ASYD, SYS1, Index 19, Bit 5=0. Unless, the SMDR information will not be output.
14. When using Station-to-Station SMDR, the billing from the PS in visitor node will be tallied by the account code (home node number and PS station number).

<Other Conditions>

None.

Programming

This section explains the data programming related to this feature. For data assignment of SMDR, refer to Station Message Detail Recording (SMDR) [S-10].

STEP 1: **ASYD** - Assign whether SMDR Station-to-Station connection data is output.
SYS 2, Index 3, Bit 7 = 0/1 (Not required/Required)

Note: To record the billing information in the Station-to-Station call via IP network, assign Bit 7 = 1 (SMDR Station-to-Station connection data is output.)

STEP 2: **ASYDL** - Assign the Extension according to the following parameters:
SYS 1, Index 578 - 581, Bit 4 = 1 (IMX format)
SYS 1, Index 589,
Bit 0 = 1 (MA-ID is added to SMDR text.)
Bit 3 = 0/1 (Billing after transfer over different MA-IDs is out of service/in service)
Index 801 Bit 1= 0/1 (Tenant/Share)

STEP 3: **AMAIL** - Assign the Area information.
MA-ID: Charged Area ID [1-9999 (0 is used to clear a MA-ID)]
AREA NAME: Charged Area Name (Maximum 16 digits)

STEP 4: **AMAA** - Assign the MA-ID information to Network ID.
KIND = Network
Type = Individual
IP ADDRESS: Enter the appropriate IP Address
MASK BIT: Select Mask Bit. [1-31]
Network Address: Assign the Network Address.
MA-ID: Assign Charged Area ID [1-9999 (0 is used to clear a MA-ID)]

STEP 5: **ASFC** - To record the billing information in the Station-to-Station call via IP network, assign the service feature restriction class of this feature.
SFC: Service Feature Restriction Class [0-15]
SFI: Service Feature Index 58 = 0 (Station-to-Station SMDR is restricted.)
SFI: Service Feature Index 219 = 1 (Station-to-Station SMDR via IP is allowed.)

FCCS Programming

This section explains the data programming related to this feature. For data assignment of SMDR, refer to Station Message Detail Recording (SMDR) [S-10].

STEP 1: **ASYD** - Assign whether output of SMDR Station-to-Station connection data is not required or required.
SYS 2, Index 3, Bit 7 = 0/1 (Not required/Required)

Note: To record the billing information in the Station-to-Station call via IP network, assign Bit 7 = 1 (SMDR Station-to-Station connection data is output.)

STEP 2: **ASYDL** - Assign the Extension according to the following parameters:
SYS 1, Index 578 - 581, Bit 4 = 1 (IMX format)
SYS 1, Index 589,
Bit 0 = 1 (MA-ID is added to SMDR text.)
Bit 1 = 1 (P2P FCCS SMDR is valid)
Bit 3 = 0/1 (Billing after transfer over different MA-IDs is out of service/in service)

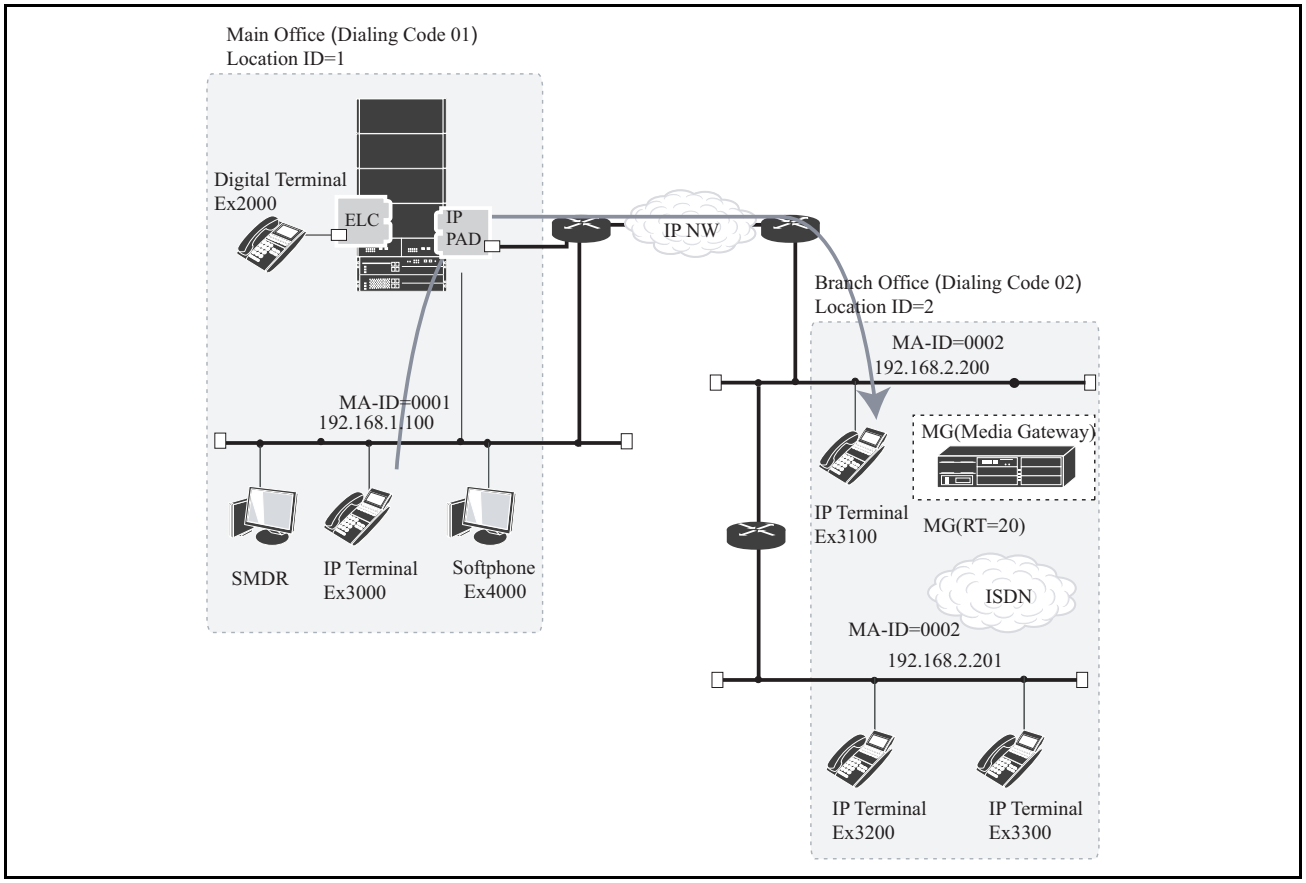
STEP 3: **AMAIN** - Assign the Area information.
MA-ID: Charged Area ID [1-9999 (0 is used to clear a MA-ID)]
AREA NAME: Charged Area Name (Maximum 16 digits)

STEP 4: **AMAA** - Assign the MA-ID information to Network ID.
KIND = Network
Type = Individual
IP ADDRESS: Enter the appropriate IP Address
MASK BIT: Select Mask Bit. [1-31]
Network Address: Assign the Network Address.
MA-ID: Assign Charged Area ID [1-9999 (0 is used to clear a MA-ID)]

STEP 5: **ASFCN** - To record the billing information in the Station-to-Station call via IP network, assign the service feature restriction class of this feature.
SFC: Service Feature Restriction Class [0-15]
SFI: Service Feature Index 58 = 0 (Station-to-Station SMDR is restricted.)
SFI: Service Feature Index 219 = 1 (Station-to-Station SMDR via IP is allowed.)

<AN EXAMPLE OF NETWORK SETTINGS>

The following shows an example in the case that Extension 3000 having MA-ID=0001 connects the Extension 3100 having MA-ID=0002.



STEP 1: Assign the following by using the ASYD command.
SYS 2, Index 3, Bit 7 = 1 (SMDR Station-to-Station connection data is output.)

STEP 2: Assign the following by using the ASYDL command.
SYS 1, Index 589
Bit 0 = 1 (MA-ID is added to SMDR text.)
Bit 3= 0/1 (Billing after transfer over different MA-IDs is out of service/in service)

STEP 3: Assign the following by using the AMAIL command.

- Main Office
  MA-ID = 0001
  AREA NAME = New York
- Branch Office
  MA-ID = 0002
  AREA NAME = Tokyo

STEP 4: Assign the following by using the AMAA command.

- Main Office

  KIND = Network
  Type = Individual
  IP ADDRESS = 10.168.1.100
  MASK BIT = 24 (255.255.255.0)
  Network Address = 10.168.1.0
  MA-ID = 0001

- Branch Office

  KIND = Network
  Type = Individual
  IP ADDRESS = 10.168.2.200
  MASK BIT = 24 (255.255.255.0)
  Network Address = 10.168.2.0
  MA-ID = 0002

STEP 5: Assign the following by using the ASFC command.
SFC: Service Feature Restriction Class [0-15]
SFI: Service Feature Index 58 = 0/1 (Station-to-Station SMDR Call is restricted/allowed.)
SFI: Service Feature Index 219 = 0/1 (Station-to-Station SMDR via IP is restricted/allowed.)

S-147 Soft Phone Wired-Phone Handset

General Description

This feature allows Soft Phone users to use a wired telephone (analog wired telephone/Desktop terminal) as handset of Soft Phone.

Operating Procedure

To originate calls

To originate a call from the wired telephone

1. Lift the handset of wired telephone.
2. Hear Dial Tone.
3. Dial the destination number.
4. Hear RBT through the handset of wired telephone.

To originate a call from Soft Phone after off-hooking the wired telephone

1. Lift the handset of wired telephone.
2. Hear Dial Tone.
3. Dial the destination number from the Soft Phone (The address book of the Soft Phone also can be used when dialing).
4. Hear RBT through the handset of wired telephone.

To originate a call from Soft Phone by dialing the destination number

- **When the Soft Phone is connecting with an analog wired telephone.**

1. Press Speaker button of the Soft Phone.
2. Dial destination number from the Soft Phone.
3. Hear RBT through the analog wired telephone handset.

- **When the Soft Phone is connecting with Desktop terminal**

1. Press Speaker button of the Soft Phone.
2. Hear DT from the speaker of Desktop terminal.
3. Dial destination number from the Soft Phone.
4. Hear RBT from the speaker of Desktop terminal.
5. Lift the Desktop terminal handset.
6. Hear RBT from the handset.

To originate a call from Soft Phone dialing with its address book

- **When the Soft Phone is connecting with analog phone.**

1. Click the destination number in the address book of the Soft Phone.
2. Lift the analog wired telephone handset.
3. Hear RBT through the handset.

- **When the Soft Phone is connecting with Desktop terminal**

1. Click the destination number in the address book of the Soft Phone.
2. Hear DT from the speaker of Desktop terminal.
3. Lift the Desktop terminal handset.
4. Hear RBT from the handset.

To receive a call

- **To receive calls with the wired telephone (Analog wired telephone/Desktop terminal)**

1. Receive a call with the wired telephone/Soft Phone.
2. Lift the handset of wired telephone.

Note: When the Soft Phone is connecting with analog wired telephones, pressing Speaker button is not effective.

Note: When the Soft Phone is connecting with Desktop terminal, intended party's voice can be heard through the speaker of Desktop terminal by pressing Speaker button of the Soft Phone.

- **To receive calls with other sub lines than lines of the wired telephone connecting with the Soft Phone.**

1. Receive a call with other lines than lines of the wired telephone connecting with the Soft Phone.
2. Press the line of the Soft Phone receiving the call.
3. Lift the handset of wired telephone.

Note: Above steps [(2) and (3)] can be done in reverse order.

Note: When the line receiving a call is pressed while the Soft Phone is connecting with analog wired telephone, the line of the analog wired telephone blinks. In this case, lift the handset in order to perform Call Pickup.

Note: When the Soft Phone is connecting with Desktop terminal, sub line can receive calls as usual.

To hold calls

Parking calls when Call Park data is already registered

- **Call Park with Hold button**

1. Press the Call Park button on the Desktop terminal /Soft Phone during the call (Intended party hears Hold Tone).
2. Call Park Group-D lump (Call Park Group-D Feature Key) on Line/Feature button of the Soft Phone blinks. Then, hear SST.
3. Hang up the handset of wired telephone.

- **Call Park using the Access Code**

1. Hook during the call (Intended party hears Hold Tone).

2. Hear SPDT.
3. Press Access Code of Call Park Group: Hold.
4. When Call Park Group-D lump on Line/Feature button of the Soft Phone blinks, hear SST (When "1" of Call Park Group-D is blinking, it means the Park Number is 1).
5. Hang up the handset of wired telephone.

Holding and answering calls when Call Park data is already registered

- **Holding and answering calls with Call Park key**

1. Lift the handset of wired telephone.
2. Press the blinking Call Park key (Call Park Group-D Feature Key).

Note: Above steps [(1) and (2)] can be done in reverse order.

- **Holding and answering calls with Access Code.**

1. Lift the handset of the wired telephone.
2. Press the Access Code of Call Park Group: Retrieval.
3. Hear SPDT.
4. Press the Park Number (When Call Park Group-D is blinking while holding a call, it means the Park Number is 1).
5. Start answering the parked call.Then, Call Park Group-D lump goes out.

- **Holding and answering calls through sublines as usual when Call Park data is not registered**

Note: This feature is available only when the Soft Phone is connecting with Desktop terminal. When it is connecting with analog wired telephone, this feature is not available.

- **Holding**

1. Press Hold button of the Soft Phone or Desktop terminal during conversation (Intended party hears Hold Tone).
2. Held Subline of Soft Phone or Desktop terminal blinks.
3. Hang up the handset of the wired telephone.

- **Answering and holding**

1. Lift the handset of the wired telephone.
2. Press the held subline of Soft Phone or Desktop terminal.

To use handsets

- **Using the handset of Soft Phone.**

1. Press Handset Switch key not lighting out on Soft Phone.
2. Handset Switch key starts lighting redly.

- **Using the wired telephone (analog wired telephone/Desktop terminal) as a handset of the Soft Phone**

1. Press Handset Switch key lighting out redly on Soft Phone.
2. Handset Switch key stops lighting.

Service Conditions

1. The terminal that can be used as a handset varies depending on the version of DtermSP30.

	DtermSP30 (Ver.2) (Proprietary Protocol)	DtermSP30 (Ver.4) (Proprietary Protocol)	DtermSP30 (Ver.7 or later) (Proprietary Protocol/SIP) and Soft Client SP350
Analog Telephone	N	N	X
DT400 Series/DT300 Series/Dterm/DtermIP/IP Enabled Dterm	N	N	X
DT800 Series/DT700 Series/DtermIP(SIP)	N	N	X
PHS	N	X	X
Wireless Handset (MH Series)	N	N	X

2. When the data for Soft Phone Wired-Phone Handset is registered, My Line to the Soft phone is restricted. Moreover, Prime Line of the Soft Phone must be set as the station number of the wired telephone connecting with.
3. When the data for Soft Phone Wired-Phone Handset is registered, the wired telephone functions as the handset of the Soft Phone every time the Soft Phone registers. However the Soft Phone or the wired telephone is being used (originating a call/a call is in progress), this feature between the wired telephone and the Soft Phone are canceled. After the call is released and the Handset Switch key is pushed, the wired telephone functions and Soft Phone are registered again.
4. Register Handset Switch key registered by AKYD command only to the Soft Phone. Note that the Handset Switch key is invalid when it is registered to Desktop terminal connecting with.
5. Type of wired telephones available to be connected with the Soft Phone are analog wired telephones (including terminals connected with Analog MC/MC&MG -COT) and Desktop terminal.
6. This feature requires to register Telephone Numbers (ALGSL/ALGSN).
7. When answering Sub Lines accommodated on the Multi Line of Soft Phone, it is required to set SFI 29 (Direct Call Pickup) to 1 using the ASFC/ASFCN command.
8. When an analog phone is connecting with the Soft Phone, calls cannot be originated or answered as far as the analog telephone handset is hanged up.
9. When an analog phone is connecting with the Soft Phone, the analog phone can receive calls if it is in on-hook state while the Soft Phone is calling or dialing.
10. Soft Phone and Desktop terminal to be connected together must belong to a same Sub Line. Moreover, do not accommodate My Line of the Soft Phone as other terminals' line.
11. When the connected wired telephone is in use, any soft phone dial key operation cannot send PB to intended party.
12. Once the data to connect the wired telephone and the Soft Phone is registered, Call Transfer data used for My line of the Soft Phone becomes invalid and the Call Transfer data registered to My line of the connected wired telephone becomes valid.

13. When ASYDL/ASYDN, System Data 1, Index 679, Bit 4 is assigned as "0" while CALL PARK GROUP -D [C-176D] feature is enabled, the wired telephone and the Soft Phone to be connected together and extensions assigned to the same group number must belong to the same MG.
14. When the Soft Phone is connecting with an analog wired telephone, Hold button of Desktop terminal becomes valid only if the data for Call Park using Hold button of Desktop terminal is registered.
15. When the Soft Phone is connecting with an analog wired telephone, Call Park Group-D with Hold button can be only handled with Soft Phone. Press the Call Park Hold Key blinking on Soft Phone and lift the handset of the analog telephone.
16. When calling by Soft Phone operation, it uses LCR/LCRS method.
17. When the handset of the connected wired telephone is off the hook, the Soft Phone cannot retrieve the call even if its speaker button is pressed.
18. When connecting IP terminals (terminals connected with DtermIP and Analog MC/MC&MG-COT) and Soft Phone together and use them, make sure that the IP terminals are logged in at first before using the Soft Phone. Moreover, in the case there is any IP terminal logged out, press Handset Switch key to cancel Soft Phone Wired-Phone Handset before using the Soft Phone, or log out the Soft Phone in order to use it.
19. When the IP terminal is used as the handset for this feature, the IP terminal is automatically specified as the handset whenever registration of IP terminal is performed. Even if the Handset Switch is applied, the IP terminal becomes the handset of Soft Phone when registration of IP terminal is performed.
20. Assign the same SFC and RSC to the Soft Phone and the wired telephone connecting with the Soft Phone.
21. This feature cannot be used for OAI/ACD system.
22. This feature cannot be used for Hotel system.

    Note: DT770G supports this feature for Hotel system.

23. When performing Speed Calling-System/Station, only wired telephone can be registered with AISA or AIEA (Soft Phone and wired telephone must be registered to the same Secondary Station).
24. When Soft Phone is using its handset and Soft Phone Wired-Phone Handset is not in use, the following features are not available from the Soft Phone:

    - Outgoing Trunk Queuing
    - Call Back
    - Camp-On Busy
    - Call Back Automatic Cancel
    - Call back-CCIS

25. When the Soft Phone is connecting with analog phones, the following features are not available from the Soft Phone:

    - Timed Reminder
    - Call Forwarding - All Calls - Announcement

26. In using Off-Hook Alarm, analog wired telephones cannot be assigned to specific lines of destination connected to.

27. In the case of answering Call Park and Call Park Group done by the Soft Phone operation with other terminals while Soft Phone Wired-Phone Handset is not in use, use Access Code and My line of the Soft Phone. Moreover, Line Number needs to be registered to My Line of Soft Phone in order to use Call Park.
28. UCD Call Waiting lamp does not light even when it is registered to the Soft Phone connected with.
29. If the intended party is using the Soft Phone connecting with analog wired telephone, the following features are not available:

    - Voice Call
    - Internal Zone Paging
    - Brokerage-Hotline

30. The following features are counted for wired telephone:

    - Service Feature Peg Count
    - Traffic Measurement
    - CS Report

31. Billing is charged for the wired telephone when using the following features:

    - Forced Account Code
    - Station Message Detail Recording (STN to STN)
    - Centralized Billing (NMS)
    - Station Message Detail Recording
    - Centered Billing-FCCS

32. Extensions having Number Display function cannot be accommodated to Multi Line. Therefore, they cannot be connected with Soft Phone.
33. MAC Authentication mode is not available for Soft Phone. However, when the terminal that is used for Soft Phone Telephone Handset is IP terminal or a station connected to Analog MC/MC&MG(COT), MAC Authentication mode is available.
34. Soft Phone is available to be used for VLAN with Windows or external devices.
35. In using UMG, both the Soft Phone and the wired telephone connecting together in Soft Phone Wired Phone Handset need to belong to a same LMG.
36. The following are conditions in terms of Calling Party Number-Name Assignment and Display, Redial, and Stack Number Called-D.

    - Numbers originated from Soft Phone are recorded in Call logs.
    - Numbers originated from wired telephones to stations are recorded in Call logs.
    - Numbers originated from a wired telephone to Trunks may not be recorded in Call logs.
    - Numbers originated from a wired telephone when Soft Phone Wired-Phone Handset is not in use are not recorded in Call logs. However, numbers originated from Soft Phone are recorded.

- Other kind of conditions, such as condition in receiving calls, are the same with conditions of Soft Phone not connected with wired telephones in Soft Phone Wired-Phone Handset.

37. In this feature, when assigning Call Pickup-Group and Expanded Call Pickup-Group data, you need to assign Station Number of Wired Phone only.
38. Refer to **Available Function Key List for Soft Phone Wired-Phone Handset** to know available Function Keys in using Soft Phone Wired-Phone Handset.
39. When Soft Phone Wireless Handset is used, Call Forwarding-All Calls is not canceled even if Call Forwarding-All Calls Automatic Canceled has been assigned by ASYD.
40. This feature is not available for a call via PHC with CCIS over IP.
41. In order to answer calls held by Call Park Group-D with analog telephones, the following needs to be registered.

    - CI=H, N (Retrieve only)
    - SRV=SSCA
    - SIDA=124 (Call Park Group -D: Hold)
    - SIDA=125 (Call Park Group -D: Retrieve)

<Conditions Specific to UMGi/RNIP System>

In using UMGi/RNIP system, both the Soft Phone and the wired telephone connecting together in Soft Phone Wired-Phone Handset need to belong to a same node.

Programming

The following programming steps are explained on the premise that the data for wired telephone and that of Soft Phone are already registered. (Refer to “Assignment of Basic Office Data” in Chapter 2 of “Data Programming Manual” to know how to configure office data for wired telephone.) Moreover, Extension Numbers assigned by ASDT and Soft Phone number assigned by AISTL need to be assigned as Telephone Numbers by ALGSL/ALGSN. (In using the Telephony Server, use ALGSN for this purpose.)
In order to perform Call Park Group-D, assign the office data for Call Park Group-D with Hold button (**Note 1**). For detail information, refer to Data Programming Manual when using the Telephony Server.

Note 1: The following explains required conditions for assigning the data for Call Park Group-D with Hold button.

- When assigning Group Number to each extension using ADGN, My line of the wired telephone and that of Soft Phone need to be registered to ADGN.
- When registering Call Park Group-D Feature Keys to Line/Feature button with AKYD, Call Park Group-D Keys need to be assigned to Soft Phone.
- In order to hold calls from Sub Line with Desktop terminal connecting with Soft Phone in Soft Phone Wired-Phone Handset, it is not necessary to configure the data for Call Park Group-D.

When using standalone configuration

STEP 1: **ASCL** (Station Class Data) - Assign the same Station Class to both Wired Phone and Soft Phone.

Note: When the Station Classes of the Soft Phone and the Wired Phone are different, some operations of this feature may not be performed from the Soft Phone.

Read Station Class Data of Wired Phone.

TN: Tenant Number
STN: Station Number of Wired Phone
TEC=12 (Dterm)
RSC: Route Restriction Class [0-15]
SFC: Service Feature Restriction Class [0-15]

For Soft Phone, assign the same Station Class as that of Wired Phone.

TN: Tenant Number
STN: Station Number of Soft Phone
TEC=12 (Dterm)
RSC: Route Restriction Class [0-15] (Same as RSC of Wired Phone)
SFC: Service Feature Restriction Class [0-15] (Same as SFC of the Wired Phone)

STEP 2: **AKYD** - Assign the PS/Wired-Station Number to the Prime Line of Soft Phone.

PL TN: Prime Line Tenant Number
PL STN: Prime Line Station Number
KYN: Key Number
KYI = 2 (Multiline)
Multi Line TN: Multi Line Tenant Number
Multi Line Station: Multi Line Station Number

Assign Handset Switch button to a function button of the Soft Phone when required.

KYN: Key Number
KYI = 1 (Function Key)
FKY = 157 (Handset Switch)

If Wired Phone is a Desktop terminal, assign Sub Lines accommodated on the Wired Phone to Soft Phone as well.

KYN: Key Number (1-40)
KYI=2 (Multiline)
Multi Line TN: Tenant Number of Multi Line accommodated on Wired Phone
Multi Line STN: Station Number of Multi Line accommodated on Wired Phone

STEP 3: **APJSL** - Assign the relationship between the My Line of Soft Phone and the PS.

UGN : PS/wired telephone User Group Number
TELN : PS/wired telephone Telephone Number
S_UGN : Soft Phone User Group Number
S_TELN : Soft Phone Telephone Number
SUBLF : Sub Line Flag (Whether to use Address Book via Soft Phone's Sub Line)
0/1 = Not used (default)/ used

When using FCCS

STEP 1: **ASCL/ASCL_T** (Station Class Data)

Assign the same Station Class to both Wired Phone and Soft Phone.

> **Note:** When the Station Classes of the Soft Phone and the Wired Phone are different, some operations of this feature may not be performed from Soft Phone.

Read Station Class Data of Wired Phone.

UGN: User Group Number
TELN: Telephone Number of Wired Phone
FPC: Fusion Point Code
TN: Tenant Number
STN: Station Number of Wired Phone
TEC=12 (Dterm)
RSC: Route Restriction Class [0-15]
SFC: Service Feature Restriction Class [0-15]

For Soft Phone, assign the same Station Class as that of Wired Phone.

UGN: User Group Number
TELN: Telephone Number of Soft Phone
TN: Tenant Number
STN: Station Number of Soft Phone
TEC=12 (Dterm)
RSC: Route Restriction Class [0-15] (Same as RSC of Wired Phone)
SFC: Service Feature Restriction Class [0-15] (Same as SFC of Wired Phone)

STEP 2: **AKYD** - Assign the Wired-Phone Station Number to the Prime Line of Soft Phone.

PL TN: Prime Line Tenant Number
PL STN: Prime Line Station Number
KYN: Key Number
KYI = 2 (Multiline)
Multi Line TN: Tenant Number of Multi Line accommodated on Wired Phone
Multi Line Station: Station Number of Multi Line accommodated on Wired Phone.

Assign Handset Switch button to a function button of the Soft Phone when required.

KYN: Key Number
KYI = 1 (Function Key)
FKY = 157 (Handset Switch)

If Wired Phone is a Desktop terminal, assign the Sub Lines accommodated on the Wired Phone to Soft Phone as well.

KYN: Key Number (1-40)
KYI=2 (MultiLine)
Multi Line TN: Tenant Number of Multi Line accommodated on Wired Phone
Multi Line STN: Station Number of Multi Line accommodated on Wired Phone

STEP 3: **APJSN** - Assign the relationship between the My Line of Soft Phone and the PS/Wired Phone.

UGN : PS/wired telephone User Group Number

TELN : PS/wired telephone Telephone Number
S_UGN : Soft Phone User Group Number
S_TELN : Soft Phone Telephone Number
SUBLF : Sub Line Flag (Whether to use Address Book via Soft Phone‘s Sub Line)
0/1 = Not used (default) /used

Data Assignments for Other Services

There are two types of services: services that need to be assigned for Wired Phone only, and services that need to be assigned for both Wired Phone and Soft Phone.

(a) Services that need to be assigned for Wired Phone only.

- Incoming Call services:
  A variety of Station Hunting services or UNIFORM CALL DISTRIBUTION (UCD) [U-1]
- Call Pickup services:
  CALL PICKUP-GROUP [C-7] and Expanded Call Pickup-Group
- Setting/Canceling of CF services:
  CALL FORWARDINFG-ALL CALLS [C-5], CALL FORWARDING -BUSY LINE [C-2], CALL FORWARDING-DON’T ANSWER [M-25] and LOGGED OUT IP STATION-CALL DESTINATION [L-53]

(b) Services that need to be assigned for both Wired Phone and Soft Phone. **Note 3**

- Multi Line Accommodation:
  For Soft Phone, assign the same Station Class as that of Wired Phone.
- CALL PARK GROUP-D [C-176D]:
  Wired Phone and Soft Phone must be assigned to the same Group of the same MG.
- LEAST COST ROUTING FOR MG-PRI [L-54] (MA-ID) (Assignment of AMAA is needed for Soft Phone.)
- Other services:
  Assign required data for Wired Phone and Soft Phone.

Note 3: When the service data is not assigned for Soft Phone, you may not operate the service from Soft Phone (even while Wired Phone is not used as the handset of Soft Phone).

Available Function Key List for Soft Phone Wired-Phone Handset

(X : **available,** N : **not available,** – : **not used)**

FKY	Service Feature	Analog Telephone	Desktop Terminal
1	Call Forwarding - Busy Line	X	X
2	Call Forwarding - All Calls	X	X
3	Executive Right of Way	X	X
4	Call Waiting	X	X
5	Call Back **Note 4**	X	X
6	Message Reminder	N	–
7	Save and Repeat	N	X
8	Last Number Call	X **Note 5**	X **Note 5**
9	Boss-Secretary - Message Waiting Lamp Control; Set	X	X
10	Call Hold	X	X
11	Boss-Secretary - Message Waiting Lamp Control; Cancel	X	X
12	Call Pickup - Group	X	X
13	Three-way Calling	X	X
14	Voice Call	X	X
15	Speed Calling - System	N	X
16	Function	N	X
17	CAS SHF	N	X
18	Call Transfer - All Calls	X	X
19	UCD Busy Out	X	X
20	Non-Exclusive Hold / Exclusive Hold (Hold) **Note 6**	X	X
21	Flash Button	X	X
22	Call Forwarding - Don’t Answer	X	X
23 - 24	-		
25	ISDN Malicious Call (Australia and U.A.E. only)	N	N
26	Speed Calling - Station	N	N
27	Originate (ORG)	N	X
28	Answer (ANS)	N	X
29	Data (DATA)	N	N

FKY	Service Feature	Analog Telephone	Desktop Terminal
30	Display / Auto (DSPY / AUTO)	N	N
31	Data Transfer (DAX)	–	–
32	-		
33	Message Waiting Lamp (MW)	N	N
34	OAI Key Code 1 (Break)	N	N
35	OAI Key Code 2 (Auto / Man)	N	N
36	OAI Key Code 3	–	–
37	OAI Key Code 4 (Work)	–	–
38	OAI Key Code 5 (Night)	–	–
39	OAI Key Code 6 (Trunk Trouble)	–	–
40	OAI Key Code 7 (Tally)	–	–
41	OAI Key Code 8 (Assistance Key)	–	–
42	OAI Key Code 9 (Emergency Key)	–	–
43	OAI Key Code 10 (Log ON / OFF)	–	–
44	OAI Key Code 11 (Answer)	N	N
45	OAI Key Code 12 (Monitor / Barge)	N	N
46	OAI Key Code 13	N	N
47	UCD Call Waiting Lamp / OAI Key Code 14 **Note 7**	N	N
48	Privacy Release	–	–
49	Speed Calling - One Touch	X	X
50	Do Not Disturb **Note 8**	X	X
51	Microphone	N	N
52	Speaker	X	X
53 - 54			
55	PAGE **Note 9**	X	X
56	-		
57	Recall	N	N
58	-		
59	Release Key	–	N
60	Manual Signaling Key	N	N

FKY	Service Feature	Analog Telephone	Desktop Terminal
61 - 82	-		
83	Retain Conference	N	N
84	Serial Call	N	X
85	Internal Zone Paging	X	X
86	-		
87	Wake Up Set	N	N
88	Wake Up Reset	N	N
89	Do Not Disturb Set	N	N
90	Do Not Disturb Reset	N	N
91	Room Cut Off Set	N	N
92	Room Cut Off Reset	N	N
93	Message Waiting Set	N	N
94	Message Waiting Reset	N	N
95	Check In	N	N
96	Check Out	N	N
97	Status	N	N
98	Audit	N	N
99	Enter	N	N
100	Clear Enter	N	N
101	END	N	N
102 - 112	-		
113	Call ID Change Display	N	N
114	-		
115	Call Pickup Expand	X	X
116	-		
117	Ringer Mute **Note 10**	–	N
118	Call Return	–	N
119	-		
120	Next Page of Software Keys	X	X
121	Previous Page of Software Keys	X	X

FKY	Service Feature	Analog Telephone	Desktop Terminal
122	Soft Key Assignment	N	N
123	Off Hook Ringing (OFR)	X	X
124	Mute (MUTE)	N	N
125	Headset / Handset	N	N
126	PS Key	–	–
127	Boss Move	–	–
128	Call Block	–	–
129	Call Trace	–	N
130	Calling Party CLIR / CLIP Control (Australia only)	–	–
131	Disconnect Key	–	–
132	Call Redirect (North America only)	–	–
133	Remote Call Hold (North America only)	–	–
134 - 135	-		
136	Selectable Station Call Pickup	N	N
137	Selectable Station Call Pickup Group	N	N
138 - 141	-		
142	LOGOUT (for IP Enabled Dterm)	N	N
143	Voice Coding Mode Change	–	–
144	Voice Coding Select	–	–
145 - 147	-		
148	Call Forwarding - Logout	Note 11	Note 11
149 - 150	-		
151	MA-ID Setting	N	N
152 - 156	-		
157	Change of Handset	X	X
158	Call Park Deluxe-Hold/Retrieve-1	N	X
159	Call Park Deluxe-Hold/Retrieve-2	N	X
160	Call Park Deluxe-Hold/Retrieve-3	N	X
161	Call Park Deluxe-Hold/Retrieve-4	N	X
162	Call Park Deluxe-Hold/Retrieve-5	N	X

FKY	Service Feature	Analog Telephone	Desktop Terminal
163	Privacy Key	X	X
164-177	-		
178	Call History	N	N
179	-		
180	Phone book	N	N
181-184	-		
185	Call Pickup Key Group-1	N	N
186	Call Pickup Key Group-2	N	N
187	Call Pickup Key Group-3	N	N
188	Call Pickup Key Group-4	N	N
189	Call Pickup Key Group-5	N	N
190 - 320	-		

Note 4: Soft Phone operation is not effective when Soft Phone Wired-Phone Handset is not in use.

Note 5: Originating calls by Redial button is unavailable when ASYD SYS1 Index 388 Bit 0=1.

Note 6: When Analog telephone is connecting with Soft Phone, this function only applies if Non-Exclusive Hold / Exclusive Hold (Hold) is registered to its connecting Soft Phone for the analog telephone to work.

Note 7: Keys of Desktop terminal connecting with Soft Phone are available to be used.

Note 8: When Soft phone is connecting with analog phone, calling party is always displayed as “Restricted”. Moreover, even when the analog phone is receiving calls and the FKY is pressed to enable Do Not Disturb at the same time, the analog phone keeps on ringing.

Note 9: If this function is applied while Soft Phone and its connecting wired telephone belong to different LPs, only lamp on the operated telephone indicates that the function is applied.

Note 10: Operated telephone‘s ring tone is disabled.

Note 11: This feature is only available when calls are made between the operated wired telephone and IP terminal or terminals connecting with Analog MC or MC&MG-COT.

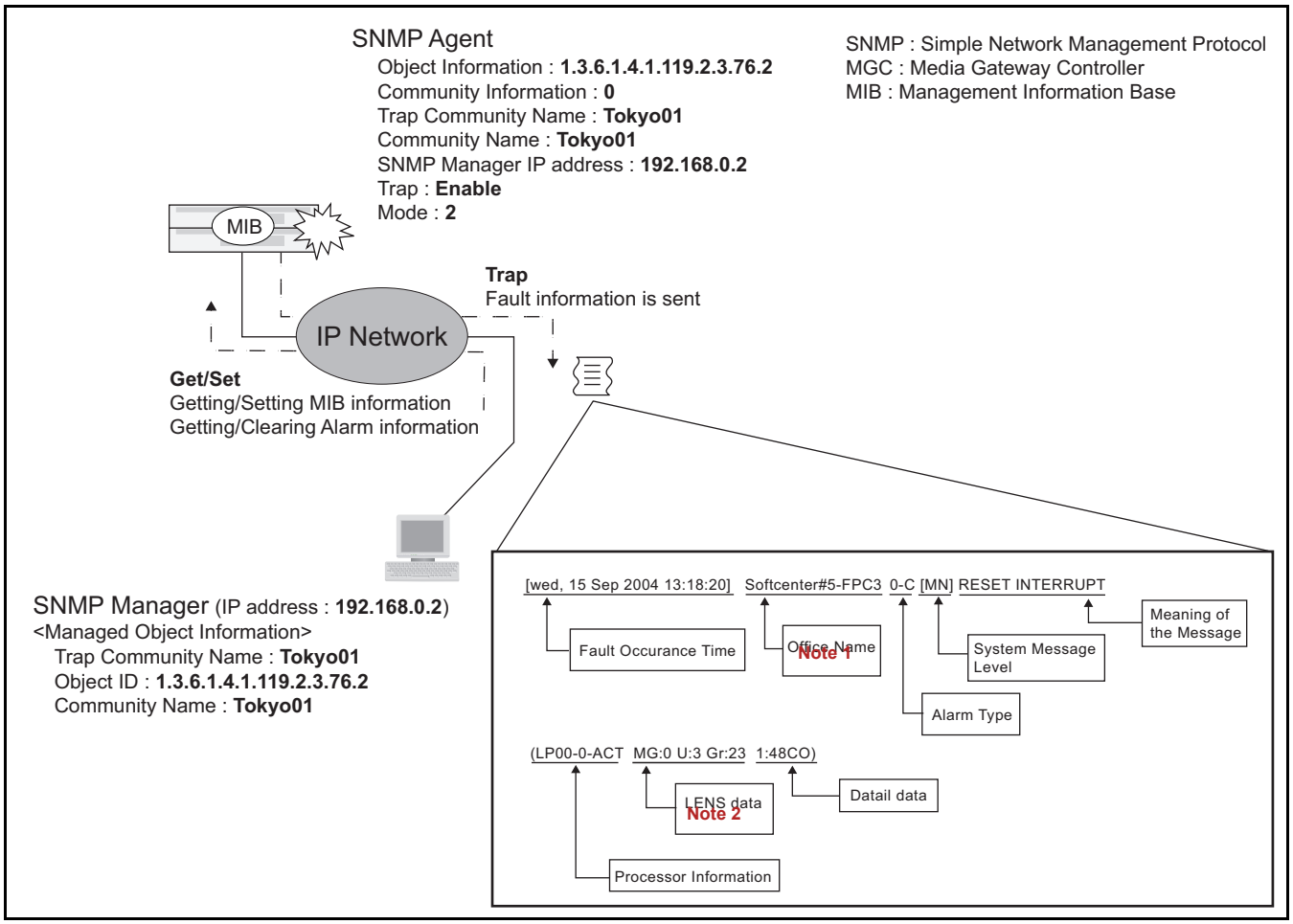
S-148 SNMP

General Description

SNMP, one of network management protocols defined by IETF, transfers and controls information on the TCP/IP basis between the SNMP manager and the SNMP agent. Generally, SNMP management software such as HP Open View and CiscoWorks needs to be installed in the SNMP manager, and operated to monitor the agents. On the other side, the SNMP agent collects network information and stores it as Management Information Base (MIB), which the SNMP manager can control. This system supports MIB-II based on RFC1158 and RFC1213. This setting provides the following functions:

- SNMP manager gets/sets MIB information of SNMP agent.
- SNMP manager monitors/clears the alarms for SNMP agent.
- SNMP agent sends trap (fault information) to SNMP manager.

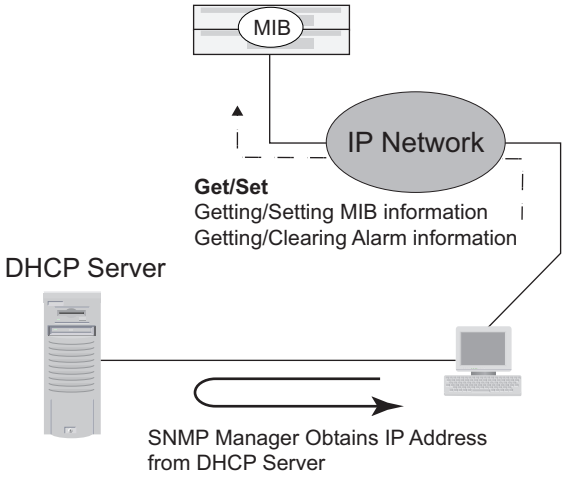
Other than standard MIB-II, the system supports NEC-specific MIB. The system configuration is shown below.



Note 1: Office Name is displayed only when ASYDL SYS1 Index 963 bit 5=1.

Note 2: LENS for MJ and MN Alarms are displayed when ASYDL SYS1 Index963 bit 4=1. For more information, refer to page 2286.

You can disable the checking of IP address of SNMP manager, which is necessary for accessing the SNMP information, by specifying "1.1.1.1" as an IP address of the SNMP manager when you cannot use a specific IP address of SNMP manager due to network conditions, such as when using DHCP.



Supports MIB information listed below. (TARP format is pursuant to SNMP v1.)

1. Function: GET

- Alarm Lamp Status
- MAJOR Lamp Status
- MINOR Lamp Status
- CPU Usage
- Basic System Setting
- Ready status Flag of Readout a Traffic
- Status of data setting for the Traffic Measurement
- Terminal Traffic
- Route Traffic
- Peg Count (Station)
- Peg Count (Att.)
- Peg Count (Route)
- Peg Count (Service)
- Peg Count (IP Service)
- Peg Count (Connection Route)
- Peg Count (UCD Route)
- Peg Count (UCD Group)
- Peg Count (UCD Station)
- Peg Count (Att. Answering)
- Peg Count (CS)
- Connection Route Traffic (Connection Route)
- IP Service Traffic (IP Service) **Note 1**
- Status of Circuit Card
- Status of Port
- Registration Information **Note 2**

Note 1: This item is available only with the IP Trunk circuit card (16 IPTC). It is not available with the Peer-to-Peer FCCS or Peer-to-Peer CCIS network.

Note 2: The following registration information can be obtained. The information is the same as the one which can be readout by DPHL command.

- LENS
- Maximum number of registration
- Number of registered terminal
- Number of terminal logged-in.
- Specified number of control stations
- MPH LENS
- MPH ID

2. **Function: GETNEXT**
    - CPU Usage
    - Ready status Flag of Readout a Traffic
    - Status of data setting for the Traffic Measurement
    - Terminal Traffic
    - Route Traffic
    - Peg Count (Station)
    - Peg Count (Att.)
    - Peg Count (Route)
    - Peg Count (Service)
    - Peg Count (IP Service)
    - Peg Count (Connection Route)
    - Peg Count (UCD Route)
    - Peg Count (UCD Group)
    - Peg Count (UCD Station)
    - Peg Count (Att. Answering)
    - Peg Count (CS)
    - Connection Route Peg Count
    - Connection Route Traffic (Connection Route)
    - IP Service Traffic (IP Service) **Note 1**
    - Status of Card
    - Status of Port
    - Registration Information
3. **Function: SET**
    - Clear of Alarm Lamp
    - Setting of Traffic Measurement
    - Clear the Ready Status flag of Readout a Traffic
    - Set/Release for Station Make Busy
    - Set/Release for Trunk Make Busy
4. **Function: TRAP**
    - Error Message Detail (Output all lamp information)
    - Traffic Measurement Ready Notification
    - RTP statistical information
    - Terminal Claim Handling
    - Error Message Detail (Output each lamp information individually)
        - *MAJOR Alarm*
        - *MINOR Alarm*
        - *SUPERVISORY Alarm*
        - *NO Alarm*

Note 1: This item is available only with the IP Trunk circuit card (16 IPTC). It is not available with the Peer-to-Peer FCCS or Peer-to-Peer CCIS network.

Service Conditions

1. When one of the following condition exists, the trap can be discarded due to the UDP that is a connection-less protocol.
   -When the SNMP manager is not activated
   -When the network is congested
   -When the line fault occurs between the Telephony Server and SNMP manager
2. When performing the traffic measurement by the SNMP manager, it is necessary to set the traffic measurement time interval from the SNMP manager. (The traffic measurement order cannot be invoked by using ATRF command when the traffic measurement is set from SNMP manager.)
3. When invoking the traffic measurement order, use the tool or command you used for setting. For example, use SNMP manager when you set the traffic measurement order from SNMP manager and ATRF command when you set from ATRF command.
4. When setting the traffic measurement time interval from the SNMP manager, it is available to set a shorter time interval than the time interval set via ATRF command.
5. To clear the traffic measurement from the SNMP manager, set 0 to Measurement Interval.
6. This service can output fault messages to multiple devices such as PCPro, E-mail and SNMP (Trap) simultaneously.
7. Message Filtering Feature is available. (**Note 2**) Message can be filtered and distributed with condition, such as Alarm higher then SUP to be reported to SNMP Manager. Alarm higher than MINOR to be reported to E-mail, etc. Information can be output depending on the alarm kind.

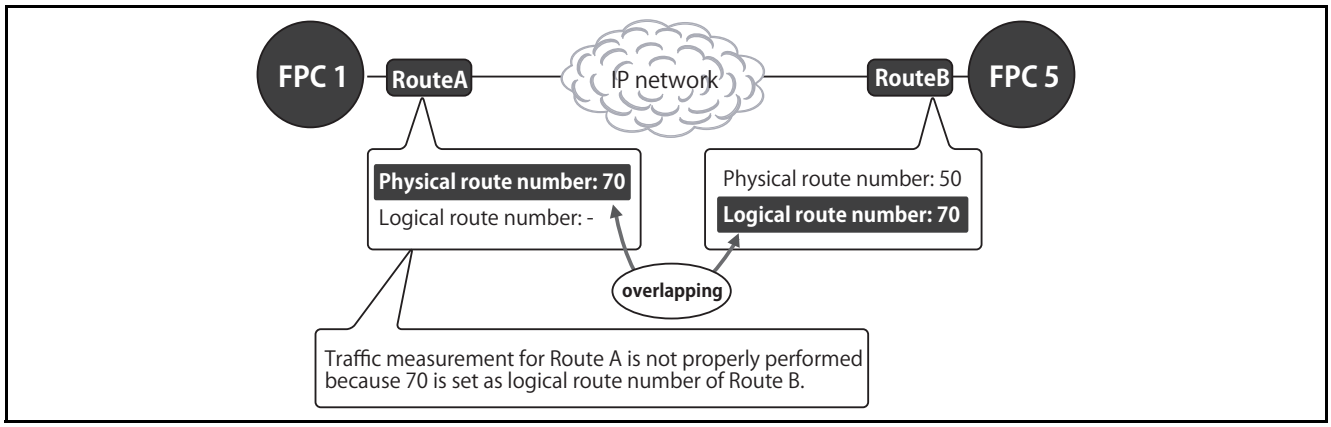
   Note 2: When message filtering has been set, the following restriction will be placed for 6-A (System Failure) message: 6-A turns on MN lamp of the system, however it will be output as a Non-Alarm.

8. System Message detail output of TRAP Notice can inform LENS as text data to the alarm grade MJ and MN. The table below lists the System Message to inform the LENS as text data.

SYSTEM MESSAGE	MJ	MN
1-S: Module Group Down Failure	√	
3-B: C-Level Infinite Loop (Permanent)		√
3-D: Lock-Up Failure (Permanent)		√
4-S: MUX Ready Failure		√
4-T: Both MUX Ready Failure		√
6-B: RGU & Howler Failure		√
15-B: VPS Failure (Permanent)		√
23-Y: MUX Clock Failure		√
23-Z: Both MUX Clock Failure		√
35-D: SP Failure		√

9. When you modified SNMP settings by ADTM command, TCP/IP Module Non Load initialization is required. When SNMP service is executed, ADTM command displays “SNMP initialization done” through Retrieve Information - Log Information Read.

10. Specify physical route numbers to observe the route traffic and route peg count by using SNMP manager. If the logical route number and physical route number of each route are the same, the traffic measurement is properly performed by SNMP manager. However, if the logical route number of another physical route is the same as the physical route number of the target route, the traffic measurement for the target route is not properly performed. For example, if the assumption is made that you perform the traffic measurement for Route A (in FPC 1) whose physical route number is 70. In this case, if FPC 5 has Route B whose logical route number is 70, you cannot perform the traffic measurement for the target route (Route A).



Programming

This section does not explain the procedures for installing the management software for SNMP managers. Refer to your SNMP product manual for the installation procedures.

<SNMP Settings>

The following enables the SNMP manager to manage the SNMP agent as shown in figure above.

STEP 1: **ASYD** - Assign the following data:

-SYS1 Index 86 Bit 4 (Storing System Messages associated with an SNMP Trap)

0: The System Messages associated with an SNMP Trap are stored until manually deleted via PCPro.

1: The System Messages associated with an SNMP Trap are not stored.

Note: Consider the following.

- When this bit is set to zero:
  Up to 512 System Messages are stored on the Telephony Server. When the limit is reached, subsequent System Messages are discarded and the corresponding SNMP trap will not be sent.
- When this bit is set to 1:
  Since no System Messages is stored on the Telephony Server, SNMP trap output will not fail because of the System Message overflow.

-SYS1 Index 86 Bit 7 = 1 (System Message detailed output is ON)

**Assign this bit to display the LENS as text data for System Message SNMP Trap Notice.*

STEP 2: **ASYDL** - Assign the following data:

- SYS1 Index 963 Bit 4 (LENS Display for System Message SNMP Trap Notice)
  - 0: Not Display
  - 1: Display
- SYS1 Index 963 Bit 5 (Office Name Display for System Message SNMP Trap Notice.) **Note 1**
  - 0: Not Display
  - 1: Display

Note 1: The following shows trap examples for SYS1 Index 963 Bit 5.

0: Not Display

Frame Length: 141 bytes
Community: public
OID: .1.3.6.1.4.1.119.2.3.76.2.2
Address: 172.16.2.2
sysUpTime: 0 days, 04:06:00
Generic: 6 - Enterprise Specific
Specific: 4
OID: .1.3.6.1.4.1.119.2.3.76.2.2.1.0
ASN1 Type: Octet String 0x04 (4)
Value: [Mon, 28 Feb 2005 13:52:19] 7-K PM MB KEY TURN ON (LP00-0-ACT 1:EB)

1: Display

Frame Length: 163 bytes
Community: public
OID: .1.3.6.1.4.1.119.2.3.76.2.2
Address: 172.16.2.2
sysUpTime: 0 days, 04:06:33
Generic: 6 - Enterprise Specific
Specific: 4
OID: .1.3.6.1.4.1.119.2.3.76.2.2.1.0
ASN1 Type: Octet String 0x04 (4)
Value: [Mon, 28 Feb 2005 13:52:53] ***NEC(A) HEAD OFF N301*** *7-K PM MB KEY TURN ON (LP00-0-ACT 1:EB)*

- SYS1 Index 963 Bit 6 (Specific Trap Type for System Message SNMP Trap Notice) **Note 2**

0: Specific Trap Type to be the Lamp Kind

Specific Trap Type	Lamp Kind
1	MJ Lamp
2	MN Lamp
3	SUP Lamp
4	No Lamp

1: Specific Trap Type to be the Fault Code kind

Specific Number XXYY (Decimal)
XX: Value between 00-37
(Indicates the Top two digits of Fault Code.)
YY: Value between 1-26
(Indicates the alphabet of Fault Code)

<Example1>: Reset Interrupt fault

Fault Code: = 0 - C
Specific Number = 3 (0 × 100 + 3)

<Example2>: System Initialization

Fault Code: = 7 - B
Specific Number = 702 (7 × 100 + 2)

Note 2: The following shows trap examples for SYS1 Index 963 Bit 6.

0: Specific Trap Type to be the Lamp Kind

Frame Length: 163 bytes
Community: public
OID: .1.3.6.1.4.1.119.2.3.76.2.2
Address: 172.16.2.2
sysUpTime: 0 days, 04:07:30
Generic: 6 - Enterprise Specific
Specific: 4
OID: .1.3.6.1.4.1.119.2.3.76.2.2.1.0
ASN1 Type: Octet String 0x04 (4)
Value: [Mon, 28 Feb 2005 13:53:50] NEC(A) HEAD OFF N301 7-K PM MB KEY TURN ON (LP00-0-ACT 1:EB)

1: Specific Trap Type to be the Fault Code Kind

Frame Length: 164 bytes
Community: public
OID: .1.3.6.1.4.1.119.2.3.76.2.2
Address: 172.16.2.2
sysUpTime: 0 days, 04:10:54
Generic: 6 - Enterprise Specific
Specific: 711
OID: .1.3.6.1.4.1.119.2.3.76.2.2.1.0
ASN1 Type: Octet String 0x04 (4)
Value: [Mon, 28 Feb 2005 13:57:20] NEC(A) HEAD OFF N301 7-K PM MB KEY TURN ON (LP00-0-ACT 1:EB)

<Specific Number Calculation>
Specific Number XXYY (Decimal)
Example for message: 7-K (Fault Code) PM MB KEY TURN ON (Fault Description)
XX: Value between 00 and 37 (Indicates the first two digits of Faule Code.)
The above message has value 7 in Specific Number.
YY: Value between 1 and 26 (Indicates the numeric position of Fault Code in alphabet)
The above message has letter K (11th place in alphabet) in Specific Number.
Therefore, Specific Number = 7 × 100 + 11 = 711

-SYS1 Index 963 Bit 7 (Output method selection for Object ID when sending System Message Trap)
Note 3

0: Fix to 1.3.6.1.4.119.2.3.76.2.2.1.0
1: Divide Alarm lamp into below 4 kinds.

MJ Lamp:	1. 3. 6. 1. 4. 119. 2. 3. 76. 2. 7. 1. 0
MN Lamp:	1. 3. 6. 1. 4. 119. 2. 3. 76. 2. 7. 2. 0
SUP Lamp:	1. 3. 6. 1. 4. 119. 2. 3. 76. 2. 7. 3. 0
No Lamp:	1. 3. 6. 1. 4. 119. 2. 3. 76. 2. 7. 4. 0

Note 3: The following shows trap examples for SYS1 Index 963 Bit 7.

0: Fix to 1.3.6.1.4.119.2.3.76.2.2.1.0

Frame Length: 170 bytes
Community: public
OID: .1.3.6.1.4.1.119.2.3.76.2.2
Address: 172.16.2.2
sysUpTime: 0 days, 05:03:57
Generic: 6 - Enterprise Specific
Specific: 419
OID: .1.3.6.1.4.1.119.2.3.76.2.2.1.0
ASN1 Type: Octet String 0x04 (4)
Value: [Mon, 28 Feb 2005 14:51:40] NEC(A) HEAD OFF N301 4-S [MN] MUX-PKG READY FAILURE (LP02-0-ACT 1:F306)

1: Specific Trap Type to be the Fault Code Kind

Frame Length: 164 bytes
Community: public
OID: .1.3.6.1.4.1.119.2.3.76.2.2
Address: 172.16.2.2
sysUpTime: 0 days, 04:10:54
Generic: 6 - Enterprise Specific
Specific: 711
OID: .1.3.6.1.4.1.119.2.3.76.2.7.2.0
ASN1 Type: Octet String 0x04 (4)
Value: [Mon, 28 Feb 2005 14:52:28] NEC(A) HEAD OFF N301 4-S [MN] MUX-PKG READY FAILURE (LP02-0-ACT 1:F306)

-SYS1 Index964 Bit 2 and Bit 3 (SNMP Output)

Bit 3	Bit 2	SNMP Trap Output
0	0	All messages will be output (Default)
0	1	Supervisory lamp or higher
1	0	Minor lamp or higher
1	1	Major lamp

STEP 3: **ADTM** - Select the Options tab for LAN2, then select Configure SNMP.

SNMP Activation: **Yes**
System Description: **SYS01**
Object ID: **1.3.6.1.4.1.119.2.3.76.2** fixed
Contact Address: Specify the contact name (maximum 64 characters).
System Name: Enter the System Name (maximum 64 characters).
Location: Specify the physical location where the system is configured (maximum 63 characters).
Trap: **Enable**
Community Name: **Tokyo01** **Note 4**
IP Address of SNMP Manager: **192.168.0.2** **Note 5**
Mode: **WRITING ALLOWED**
Trap Community Name: **Tokyo01** (Community transferred traps) **Note 4**
IP Address of SNMP Manager: **192.168.0.2** (IP address to be transferred traps)
Agent Address: Optional. Assign when the agent address of the trap packet needs to be different from own address.

Note 4: These parameters can be used only alphanumeric characters and underscore (“_”).

Note 5: You can disable the checking of IP address of SNMP manager, which is necessary for accessing the SNMP information, by specifying "1.1.1.1" as an IP address of the SNMP manager when you cannot use a specific IP address of SNMP manager due to network conditions, such as when using DHCP.

Note: After setting the above data, execute the restart of TCP/IP Module by TCP/IP Module Control window of ADTM command.

STEP 4: Register/Confirm the following data on the SNMP manager that manages the PBX (SNMP agent).

Note: The following information in SNMP manager is one of examples. Input necessary data according to middle ware installed in the SNMP manager.

Community name:**Tokyo01**

Registering the data shown above enables SNMP managers to manage SNMP agents.

<Details of ADTM Command>

The following table shows the field details of the ADTM command.

FIELD NAME		ATTRIBUTES	OPTIONS	DESCRIPTION
SNMP Activation	Activate SNMP	Required	Yes	SNMP service is activated
			No	SNMP service is not activated
System Group Information	System Description		-	Equipment information (maximum 128 characters)
	Object ID	Not Required	-	1.3.6.1.4.1.119.2.3.76.2 fixed
System, SNMP Group Information	Contact Address	Optional	-	Contact address (maximum 64 characters)
	System Name		-	Equipment name (maximum 64 characters)
	Location		-	Physical location for equipment (maximum 63 characters)
	Trap		Enable	Generate traps
			Disable	Not generate traps
Community Information **Note 1**	Community Name	Required	-	Community name (maximum 25 characters) **Note 3**
	IP Address of SNMP Manager		-	IP address for SNMP manager **Note 4**
	Mode		READ ONLY	Reading only by SNMP Manager
			WRITING ALLOWED	Reading and Writing by SNMP Manager
Trap Information **Note 2**	Trap Community Name	Optional	-	Community name sent traps (maximum 25 characters) **Note 3**
	IP Address of SNMP Manager		-	IP address to which sends traps
	Agent Address		-	

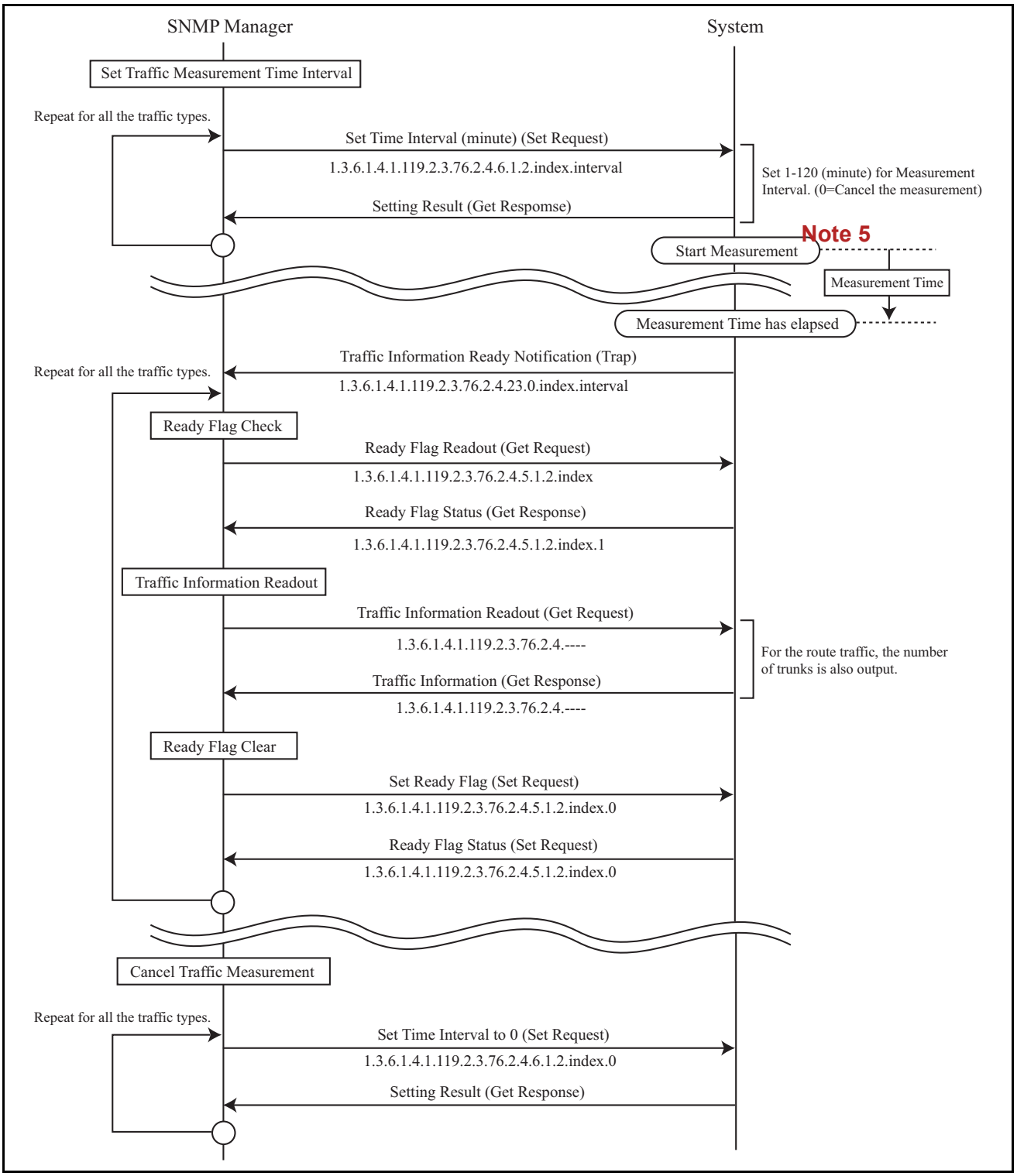
Note 1: Up to four SNMP managers can be created on the Community Information. Creating one SNMP manager requires three attributes (Community name, IP address, and Mode). When the SNMP manager is created, the activation is required by using TCP/IP module control.

Note 2: Up to four trap destinations can be created on the Trap Information. Creating one destination requires three attributes (Trap community name, IP address, and Agent address). If the Agent address is not determined, the main body's address will be used. When the trap destination is created, the activation is required by using TCP/IP module control.

Note 3: These parameters can use only alphanumeric characters and underscore ("_").

Note 4: You can disable the checking of IP address of SNMP manager, which is necessary for accessing the SNMP information, by specifying "1.1.1.1" as an IP address of the SNMP manager when you cannot use a specific IP address of SNMP manager due to network conditions, such as when using DHCP.

STEP 5: The following shows the Traffic Measurement procedure of SNMP:

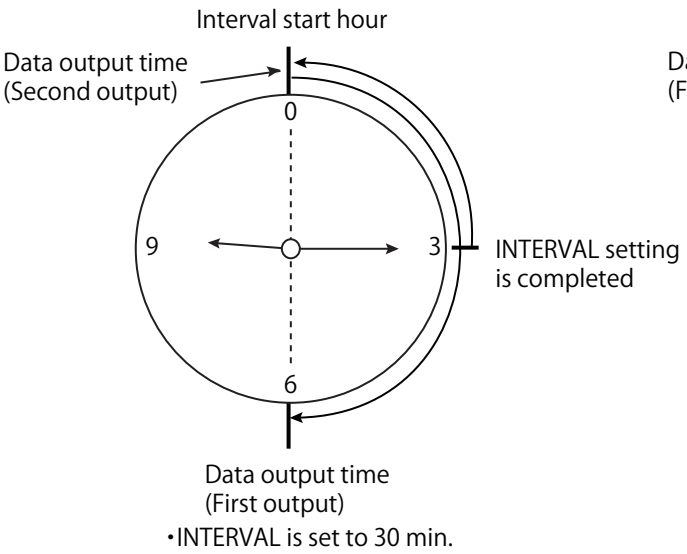


Note 5: Basically, the first output of data is provided when the time set for INTERVAL has elapsed since the Interval start hour, the time the INTERVAL setting is completed, which is represented with hours only.
For example, if you have completed setting INTERVAL to 30 minutes at 9:15, the Interval start hour is 9:00, and the data is generated at 9:30. (See Example 1.)

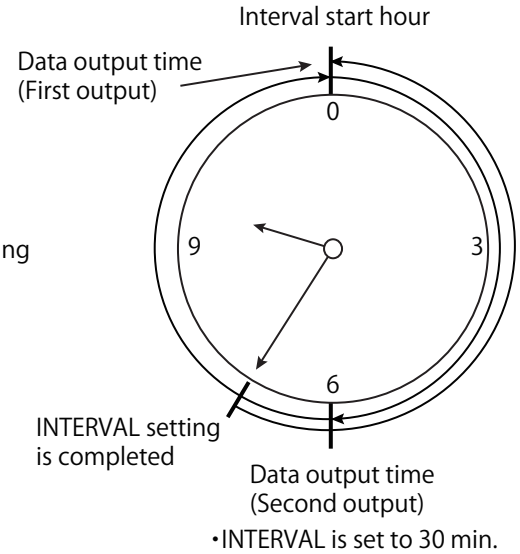
However, when INTERVAL setting is not completed before the first interval has elapsed, the output is generated when another interval has passed.

For example, if you have completed setting INTERVAL to 30 minutes at 9:35, the Interval start hour is 9:00, and the data is generated at 10:00. (See Example 2.)

Example 1



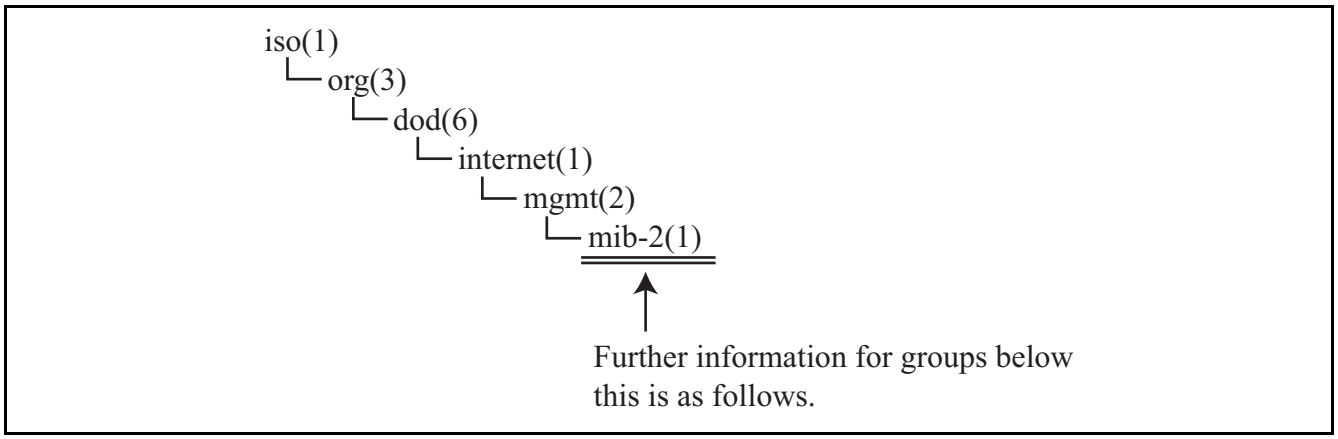
Example 2



<MIB Information>

MIB information can be loaded or cleared by using the SNMP manager (only with TP side). MIB information used in the Telephony Server is listed below.

Internet standard MIB (RFC compliance)



[Information for groups below mib-2]

[system(1) group]

sysDescr(1)
sysObjectID(2)
sysUpTime(3)
sysContact(4)
sysName(5)
sysLocation(6)
sysServices(7)

[interface(2) group]

ifNumber(1)		
ifTable(2)		
	ifEntry(1)	
		ifIndex(1)
		ifDescr(2)
		ifType(3)
		ifMtu(4)
		ifSpeed(5)
		ifPhysicalAddress(6)
		ifAdminStatus(7)
		ifOperStatus(8)
		ifLastChange(9)
		ifInOctets(10)
		ifInUcastPkts(11)
		ifInNUcastPkts(12)
		ifInDiscards(13)
		ifInErrors(14)
		ifInUnknownProtos(15)
		ifOutOctets(16)
		ifOutUcastPkts(17)
		ifOutNUcastPkts(18)
		ifOutDiscards(19)
		ifOutErrors(20)
		ifOutQLen(21)
		ifSpecific(22)

[ip(4) group]

ipForwarding(1)			
ipDefaultTTL(2)			
ipInReceives(3)			
ipInHdrErrors(4)			
ipInAddrErrors(5)			
ipForwDatagrams(6)			
ipInUnknownProtos(7)			
ipInDiscards(8)			
ipInDelivers(9)			
ipOutRequests(10)			
ipOutDiscards(11)			
ipOutNoRoutes(12)			
ipReasmTimeout(13)			
ipReasmReqds(14)			
ipReasmOKs(15)			
ipReasmFails(16)			
ipFragOKs(17)			
ipFragFails(18)			
ipFragCreates(19)			
ipAddrTable(20)			
	ipAddrEntry(1)		
		ipAdEntAddr(1)	
		ipAdEntIfIndex(2)	
		ipAdEntNetMask(3)	
		ipAdEntBcastAddr(4)	
		ipAdEntReasMaxSize(5)	
ipRoutingTable(21)			
ipNetToMediaTable(22)			
	ipNetToMediaEntry(1)		
		ipNetToMediaIfIndex(1)	
		ipNetToMediaPhysAd-dress(3)	
		ipNetToMediaNetAd-dress(4)	
		ipNetToMediaType(5)	
ipRoutingDiscards(23)			
ipForward(24)			
	ipForwardNumber(1)		
	ipForwardTable(2)		

		ipForwardEntry(1)	
			ipForwardDest(1)
			ipForwardMask(2)
			ipForwardPolicy(3)
			ipForwardNextHop(4)
			ipForwardIfIndex(5)
			ipForwardType(6)
			ipForwardProto(7)
			ipForwardAge(8)
			ipForwardInfo(9)
			ipForwardNextHopAS(10)
			ipForwardMetric1(11)
			ipForwardMetric2(12)
			ipForwardMetric3(13)
			ipForwardMetric4(14)
			ipForwardMetric5(15)

[icmp(5) group]

icmpInMsgs(1)
icmpInErrors(2)
icmpInDestUnreachs(3)
icmpInTimeExcds(4)
icmpInParmProbs(5)
icmpInSrcQuenchs(6)
icmpInRedirects(7)
icmpInEchos(8)
icmpInEchoReps(9)
icmpInTimeStamps(10)
icmpInTimestampReps(11)
icmpInAddrMasks(12)
icmpInAddrMaskReps(13)
icmpOutMsgs(14)
icmpOutErrors(15)
icmpOutDestUnreachs(16)
icmpOutTimeExcds(17)
icmpOutParmProbs(18)
icmpOutSrcQuenchs(19)
icmpOutRedirects(20)
icmpOutEchos(21)
icmpOutEchoReps(22)
icmpOutTimeStamps(23)
icmpOutTimestampReps(24)
icmpOutAddrMaks(25)
icmpOutAddrMaskReps(26)

[tcp(6) group]

tcpRtoAlgorithm(1)		
tcpRtoMin(2)		
tcpRtoMax(3)		
tcpMacConn(4)		
tcpActiveOpens(5)		
tcpPassiveOpens(6)		
tcpAttemptFails(7)		
tcpEstabResets(8)		
tcpCurrExtab(9)		
tcpInSegs(10)		
tcpOutSegs(11)		
tcpRetransSegs(12)		
tcpConnTable(13)		
	tcpConnEntry(1)	
		tcpConnState(1)
		tcpConnLocalAddress(2)
		tcpLocalPort(3)
		tcpConnRemAddress(4)
		tcpConnRemPort(5)
tcpInErrs(14)		
tcpOutRsts(15)		

[udp(7) group]

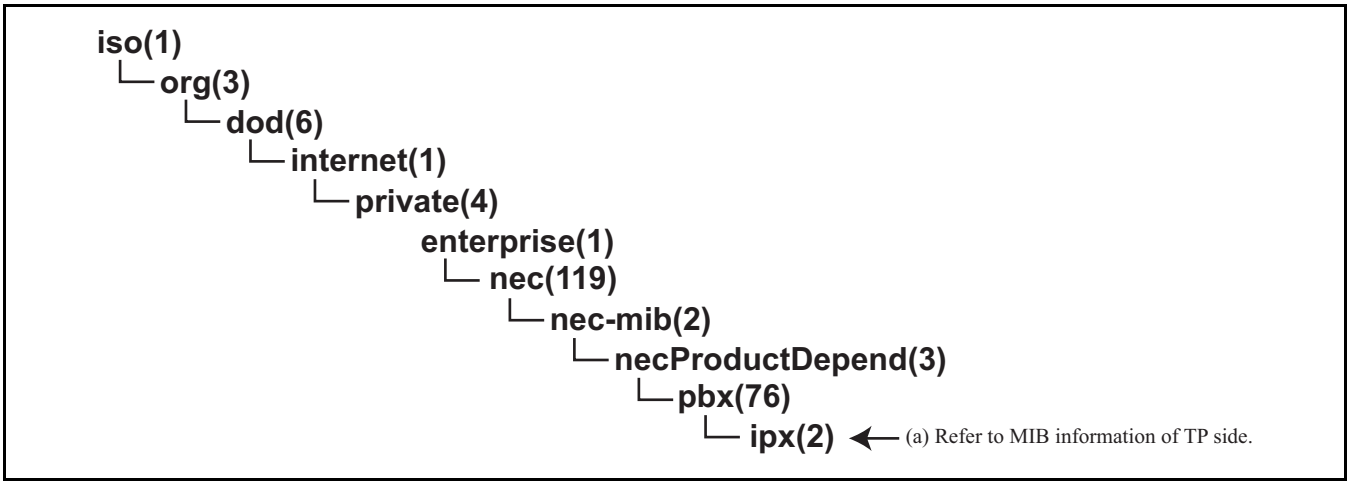
udpInDatagrams(1)		
udpNoPorts(2)		
udpInErrors(3)		
udpOutDatagrams(4)		
udpTable(5)		
	udpEntry(1)	
		udpLocalAddress(1)
		udpLocalPort(2)

[snmp(11) group]

snmpInPkts(1)
snmpOutPkts(2)
snmpInBadVersions(3)
snmpInBadCommunityNames(4)
snmpInBadCommunityUses(5)
snmpInASNParseErrs(6)
Is not used(7)
snmpTooBigs(8)
snmpInNoSuchNames(9)
snmpInBadValues(10)
snmpReadOnlys(11)
snmpInGenErrs(12)
snmpInTotalReqVars(13)
snmpInTotalSetVars(14)
snmpInGetRequests(15)
snmpInGetNexts(16)
snmpInSetRequests(17)
snmpInGetResponses(18)
snmpInTraps(19)
snmpOutTooBig(20)
snmpOutNoSuchNames(21)
snmpOutBadValues(22)
Is not used(23)
snmpOutGenErrs(24)
snmpOutGetRequests(25)
snmpOutGetNexts(26)
snmpOutSetRequests(27)
snmpOutGetResponses(28)
snmpOutTraps(29)
snmpEnableAuthenTraps(30)

NEC Private MIB

[Tree structure of MIB]



General Information of layer below pbx(76).ipx(2).

alarm(1): Used for Alarm
systemMessage(2): Failure message
cpu(3): CPU Group
traffic(4): Traffic Group
port(5): Port group
terminal(6): Terminal group
System Message(7): System message group
System license groups(11): Not used
Registration information group(12): Registration group

[ipx(2) group]

LAYER INFORMATION		MEANING
alarm(1)		Alarm Group
	ipxLampStatusClear(1)	Lamp status and lamp clear of the Telephony Server
	ipxMajorLampStatus(2)	Status of the Major alarm lamp
	ipxMinorLampStatus(3)	Status of the Minor alarm lamp
	ipxSupervisorLampStatus(4)	Status of the Supervisor alarm lamp
systemMessage(2)		Failure message
	ipxSystemMessage(0)	Failure Message Type
	ipxSystemMessageData(1)	Description of the failure message
cpu(3)		CPU Group
	cpuUsage(1)	CPU Usage
traffic(4)		Traffic Group
	moduleMax(1)	Number of Module Groups
	tenantMax(2)	Maximum Number of Tenants
	routeMax(3)	Maximum Number of Routes
	atnMax(4)	Maximum number of Attendant consoles
	readyFlag(5)	Ready Status Flag of Readout Traffic Data (GET/SET)
	obsOrder(6)	Traffic data Measurement Order Table (GET/SET)
	termTrafTable(7)	Readout Terminal Traffic Measurement
	rtTrafTable(8)	Readout Route Traffic Measurement
	staPegCntTable(9)	Readout Station Peg count
	attPegCntTNtable(10)	Readout Att. Peg count (Specify the Tenant No.)
	attPegCntATTTable(11)	Readout Att. Peg count (Specify the Att. No.)
	rtPegCntTable(12)	Readout Route Peg count
	svPegCntTable(13)	Readout Service Peg count

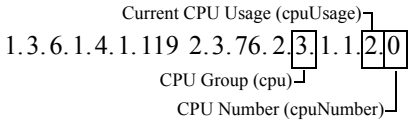
LAYER INFORMATION		MEANING
traffic(4)	ucdRTPegCntTable(14)	Readout the number of UCD Route Peg Counts
	ucdGrPegCntTable(15)	Readout the number of UCD Group Peg Counts
	ucdStaPegCntTable(16)	Readout the number of UCD Station Peg counts
	attAnsPegCntTable(17)	Readout the number of Peg counts per Att. Answering time
	csPegCntTable(18)	Readout the number of CS Peg Count
	crtPegCntTable(19)	Readout the number of Connection Route Peg Counts
	crtTrafTable(20)	Readout Connection Route Traffic
	ipSvPegCntTable(21)	Readout the Number of IP Service Peg Counts
	ipSvTrafTable(22)	Readout the IP Service Traffic
	obsReadyNotifyData(23)	Traffic Measurement Ready Notification
	obsReadyNotify(24)	Entity of Traffic Measurement Ready Notification
port(5)		Port Group
	pkgStsTable(1)	Readout the status of circuit card
	portStatsTable(2)	Readout of Port Status
	tkMBTable(3)	Readout/Setting of Information for Trunk Make Busy (GET/SET)
	staMBTable(4)	Readout/Setting of Information for Station Make Busy (GET/SET)
terminal(6)		Terminal Group
	terminalStatsData(1)	RTP statistics Information Notification
	terminalStats(2)	Entity of RTP statistics Information Notification
	claimNotifyData(3)	Terminal Claim Handling Notification
	claimNotify(4)	Entity of Terminal Claim Handling Notification

LAYER INFORMATION		MEANING
sysMessage(7)		System message group
	sysMessageMajor(1)	Trap of system message (major alarm)
	sysMessageMinor(2)	Trap of system message (minor alarm)
	sysMessageSupervisor(3)	Trap of system message (supervisory alarm)
	sysMessageNoAlarm(4)	Trap of system message (no alarm)
licenseStatus(11)		Not used
regInfo(12)		Registration information group
	regInfoTable(1)	Registration information table

Object ID for NEC Private MIB information

CONTENTS		OBJECT ID	GET	SET	TRAP
Alarm Group		1.3.6.1.4.1.119.2.3.76.2.1	-	-	-
	Lamp Status and Lamp Clear of Telephony Server	**Request** Lamp Status and Lamp Clear of Telephony Server (ipxLampStatusClear) 1.3.6.1.4.1.119 2.3.76.2.1.1.0 **Response** **<Input data for SET>** When you execute SET request, all lamps of the Telephony Server will be cleared. **<Output data for GET>** The lighting status of Major/Minor lamp is indicated as follows in integer. 0=Both Major lamp and Minor lamp are turned off 1=Major or Minor lamp is lighting	x	x	-
	Status of the Major Alarm Lamp	**Request** Status of the Major Alarm Lamp (ipxMajorLampStatus) 1.3.6.1.4.1. 119 2.3.76.2.1.2.0 **Response** The lighting status of Major lamp is indicated as follows in integer. 0=Major lamp is turned off 1=Major lamp is lighting	x	-	-
	Status of the Minor Alarm Lamp	**Request** Status of the Minor Alarm Lamp (ipxMinorLampStatus) 1.3.6.1.4.1. 119 2.3.76.2.1.3.0 **Response** The lighting status of Minor lamp is indicated as follows in integer. 0=Minor lamp is turned off 1=Minor lamp is lighting	x	-	-

CONTENTS			OBJECT ID	GET	SET	TRAP
Alarm Group	Status of the Supervisor Alarm Lamp		**Request** Status of the Supervisor Alarm Lamp (ipxSupervisorLampStatus) 1.3.6.1.4.1.119.2.3.76.2.1.4.0 **Response** The lighting status of Supervisor lamp is indicated as follows in integer. 0=Not available 1=Supervisor lamp is turned off 2=Supervisor lamp is lighting	x	-	-
Failure Message			1.3.6.1.4.1.119.2.3.76.2.2	-	-	-
	Failure Message Type		1.3.6.1.4.1.119.2.3.76.2.2.0	-	-	x
		Major Alarm	There is no specific operating instructions for this trap. However, when the failure message is generated, the object ID, “1.3.6.1.4.1.119.2.3.76.2.2.0.1”, is sent to SNMP Manager. Character strings of the failure notification are indicated (maximum 255 characters).	-	-	x
		Minor Alarm	There is no specific operating instructions for this trap. However, when the failure message is generated, the object ID, “1.3.6.1.4.1.119.2.3.76.2.2.0.2”, is sent to SNMP Manager. Character strings of the failure notification are indicated (maximum 255 characters).	-	-	x
		Supervisor Alarm	There is no specific operating instructions for this trap. However, when the failure message is generated, the object ID, “1.3.6.1.4.1.119.2.3.76.2.2.0.3”, is sent to SNMP Manager. Character strings of the failure notification are indicated (maximum 255 characters).	-	-	x
		No Alarm	There is no specific operating instructions for this trap. However, when the failure message is generated, the object ID, “1.3.6.1.4.1.119.2.3.76.2.2.0.4”, is sent to SNMP Manager. Character strings of the failure notification are indicated (maximum 255 characters).	-	-	x
	Description of the Failure Message		There is no specific operating instructions for this trap. However, when the failure message is generated, the object ID, “1.3.6.1.4.1.119.2.3.76.2.2.1.0”, is sent to SNMP Manager. Character strings of the failure notification are indicated (maximum 255 characters).	-	-	x

<table>
<tr><th colspan="3">CONTENTS</th><th>OBJECT ID</th><th>GET</th><th>SET</th><th>TRAP</th></tr>
<tr><td>CPU Group</td><td colspan="2"></td><td>1.3.6.1.4.1.119.2.3.76.2.3</td><td>-</td><td>-</td><td>-</td></tr>
<tr><td></td><td>CPU Usage</td><td></td><td>1.3.6.1.4.1.119.2.3.76.2.3.1.1</td><td>-</td><td>-</td><td>-</td></tr>
<tr><td></td><td></td><td>CPU Number</td><td>1.3.6.1.4.1.119.2.3.76.2.3.1.1.1.0</td><td>x</td><td>-</td><td>-</td></tr>
<tr><td></td><td></td><td>CPU Usage (Current value)</td><td>Request
 

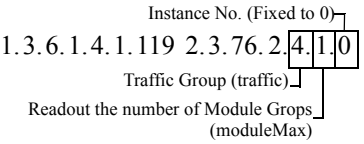
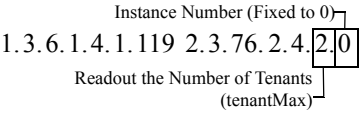
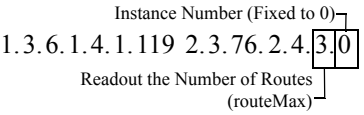
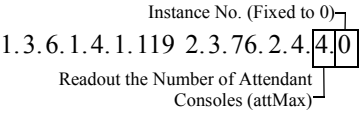
Response
Response data is an integer (0~99).
[For UMGi System]
When using Multiple Processor System, specifies the CPU Number according to the following table.
<table><tr><th>Node</th><th>IMG</th><th>Processor</th><th>CPU</th></tr><tr><td>CCN</td><td>IMG0</td><td>LP00</td><td>48</td></tr><tr><td></td><td>IMG1</td><td>-</td><td>52</td></tr><tr><td></td><td>IMG2</td><td>-</td><td>56</td></tr><tr><td></td><td>IMG3</td><td>-</td><td>60</td></tr><tr><td>RN</td><td>IMG4</td><td>LP08</td><td>64</td></tr><tr><td></td><td>IMG5</td><td>-</td><td>68</td></tr><tr><td></td><td>IMG6</td><td>LP12</td><td>72</td></tr><tr><td></td><td>IMG7</td><td>-</td><td>76</td></tr><tr><td></td><td>IMG8</td><td>LP16</td><td>80</td></tr><tr><td></td><td>IMG9</td><td>-</td><td>84</td></tr><tr><td></td><td>IMG10</td><td>LP20</td><td>88</td></tr><tr><td></td><td>IMG11</td><td>-</td><td>92</td></tr><tr><td></td><td>IMG12</td><td>LP24</td><td>96</td></tr><tr><td></td><td>IMG13</td><td>-</td><td>100</td></tr><tr><td></td><td>IMG14</td><td>LP28</td><td>104</td></tr><tr><td></td><td>IMG15</td><td>-</td><td>108</td></tr></table></td><td>x</td><td>-</td><td>-</td></tr>
</table>

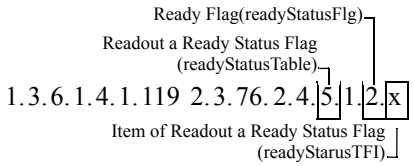
<table>
<tr><th colspan="3">CONTENTS</th><th>OBJECT ID</th><th>GET</th><th>SET</th><th>TRAP</th></tr>
<tr><td>CPU Group</td><td>CPU Usage</td><td>CPU Usage (10 minutes average per day)</td><td>Request
CPU Usage 1 day (cpuUsage1day)
1.3.6.1.4.1.119 2.3.76.2.3.1.1.3.0
CPU Number (cpuNumber)

Response
Response data is octet strings (288 octets).
Data of two octets times 144 is indicated by corresponding data indication from “00:00” to “23:50”.

1st Octet 2nd Octet
0:00 XX YY =XX.YY
0:10
0:20
|
23:30
23:40
23:50

The first two octets, xx, is the BCD indicating numbers within the range from 00 to 99. The latter octets, yy, is the BCD indicating numbers within the range from 0.00 to 0.99.</td><td>x</td><td>-</td><td>-</td></tr>
</table>

CONTENTS		OBJECT ID	GET	SET	TRAP
Traffic Group		1.3.6.1.4.1.119.2.3.76.2.4	-	-	-
	Number of Module Groups	**Request**  **Response** Response data is an integer (1~32) that is assigned by ASYD, SYS1, Index 0.	x	-	-
	Maximum Number of Tenants	**Request**  **Response** Response data is an integer (1~63) that is assigned by ASYD, SYS1, Index 8.	x	-	-
	Maximum Number of Routes	**Request**  **Response** Response data is an integer (1~800) that is assigned by ASYD, SYS1, Index 65 and Index 66.	x	-	-
	Maximum number of Att.	**Request**  **Response** Response data is an integer (1~60) that is assigned by ASYD, SYS1, Index 9.	x	-	-

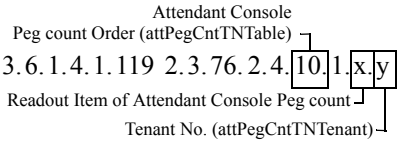
CONTENTS			OBJECT ID	GET	SET	TRAP
Traffic Group	Ready Status Flag of Readout Traffic Data (GET/SET)		1.3.6.1.4.1.119.2.3.76.2.4.5.1 **Request**  **<Input data for SET>** Ready Status Flag (Integer) 0=Not Ready **Response** **<Output data for GET>** Ready Status Flag (Integer) 0=Not Ready 1=Ready	-	-	-
		TFI No.	1.3.6.1.4.1.119.2.3.76.2.4.5.1.1.x ("GET" function is not available for this item)	x	-	-
		Traffic (Terminal)	1.3.6.1.4.1.119.2.3.76.2.4.5.1.2.1	x	x	-
		Traffic (Route)	1.3.6.1.4.1.119.2.3.76.2.4.5.1.2.2	x	x	-
		Peg Count (Station)	1.3.6.1.4.1.119.2.3.76.2.4.5.1.2.3	x	x	-
		Peg Count (Att.)	1.3.6.1.4.1.119.2.3.76.2.4.5.1.2.4	x	x	-
		Peg Count (Route)	1.3.6.1.4.1.119.2.3.76.2.4.5.1.2.5	x	x	-
		Peg Count (Service)	1.3.6.1.4.1.119.2.3.76.2.4.5.1.2.6	x	x	-
		Peg Count (UCD Route)	1.3.6.1.4.1.119.2.3.76.2.4.5.1.2.8	x	x	-
		Peg Count (UCD Group)	1.3.6.1.4.1.119.2.3.76.2.4.5.1.2.9	x	x	-
		Peg Count (UCD Station)	1.3.6.1.4.1.119.2.3.76.2.4.5.1.2.10	x	x	-

CONTENTS			OBJECT ID	GET	SET	TRAP
Traffic Group	Ready Status Flag of Readout Traffic Data (GET/SET)	Peg Count (Answering time of Att.)	1.3.6.1.4.1.119.2.3.76.2.4.5.1.2.15	x	x	-
		Peg Count (CS/ZT)	1.3.6.1.4.1.119.2.3.76.2.4.5.1.2.17	x	x	-
		Peg Count (Connection Route)	1.3.6.1.4.1.119.2.3.76.2.4.5.1.2.18	x	x	-
		Traffic (Connection Route)	1.3.6.1.4.1.119.2.3.76.2.4.5.1.2.19	x	x	-
		Peg Count (IP Service)	1.3.6.1.4.1.119.2.3.76.2.4.5.1.2.20	x	x	-
		Traffic (IP Service)	1.3.6.1.4.1.119.2.3.76.2.4.5.1.2.21	x	x	-
	Traffic data Measurement Order Table (GET/SET)		1.3.6.1.4.1.119.2.3.76.2.4.6.1 **Request** Traffic data mesurement interval reading (obsOrderTable) 1.3.6.1.4.1.119 2.3.76.2.4.6.1.2.x Mesurement interval readout (obsOrderInterval) Mesurement Item (obsOrderTFI) **<Input data for SET>** Set a Measurement Interval in integer (1~120 Minutes). To clear the traffic measurement, set 0 to Measurement Interval. **Response** **<Output data for GET>** Response data is an integer (1~120Min.). When 0 is set to Measurement Interval, the traffic measurement is not performed.	-	-	-
		TFI No.	1.3.6.1.4.1.119.2.3.76.2.4.6.1.1.x (“GET” function is not available for this item)	x	-	-
		Traffic (Terminal)	1.3.6.1.4.1.119.2.3.76.2.4.6.1.2.1	x	x	-
		Traffic (Route)	1.3.6.1.4.1.119.2.3.76.2.4.6.1.2.2	x	x	-
		Peg Count (Station)	1.3.6.1.4.1.119.2.3.76.2.4.6.1.2.3	x	x	-
		Peg Count (Att.)	1.3.6.1.4.1.119.2.3.76.2.4.6.1.2.4	x	x	-

CONTENTS			OBJECT ID	GET	SET	TRAP
Traffic Group	Traffic data Measurement Order Table (GET/SET)	Peg Count (Route)	1.3.6.1.4.1.119.2.3.76.2.4.6.1.2.5	x	x	-
		Peg Count (Service)	1.3.6.1.4.1.119.2.3.76.2.4.6.1.2.6	x	x	-
		Peg Count (UCD Route)	1.3.6.1.4.1.119.2.3.76.2.4.6.1.2.8	x	x	-
		Peg Count (UCD Group)	1.3.6.1.4.1.119.2.3.76.2.4.6.1.2.9	x	x	-
		Peg Count (UCD Station)	1.3.6.1.4.1.119.2.3.76.2.4.6.1.2.10	x	x	-
		PegCount (Answering time of Att.)	1.3.6.1.4.1.119.2.3.76.2.4.6.1.2.15	x	x	-
		Peg Count (CS/ZT)	1.3.6.1.4.1.119.2.3.76.2.4.6.1.2.17	x	x	-
		Peg Count (Connection Route)	1.3.6.1.4.1.119.2.3.76.2.4.6.1.2.18	x	x	-
		Traffic (Connection Route)	1.3.6.1.4.1.119.2.3.76.2.4.6.1.2.19	x	x	-
		Peg Count (IP Service)	1.3.6.1.4.1.119.2.3.76.2.4.6.1.2.20	x	x	-
		Traffic (IP Service)	1.3.6.1.4.1.119.2.3.76.2.4.6.1.2.21	x	x	-
	Readout Terminal Traffic Measurement		1.3.6.1.4.1.119.2.3.76.2.4.7.1	-	-	-
		Module No.	1.3.6.1.4.1.119 2.3.76.2.4.7.1.1.x.y.z Module No. (“GET” function is not available for this item)	x	-	-
		Unit No.	1.3.6.1.4.1.119 2.3.76.2.4.7.1.2.x.y.z Unit No. (“GET” function is not available for this item)	x	-	-

CONTENTS			OBJECT ID	GET	SET	TRAP
Traffic Group	Readout Terminal Traffic Measure-ment	Group No.	1.3.6.1.4.1.119 2.3.76.2.4.7.1.3.x.y.z Group No. (“GET” function is not available for this item)	x	-	-
		Terminal Traffic	**Request** Terminal Traffic Order (termTrafTable) 1.3.6.1.4.1.119 2.3.76.2.4.7.1.4.x.y.z Readout Terminal Trafic Mesurement (termTrafData) Module No. (termTrafModele) Unit No. (termTrafUnit) Group No. (termTrafGroup) **Response** Response data is coded as ASCII string. Terminal traffic data for each level are separated by commas. **<Example>** 0.123 , 0.456 , ~ 0.789 [Unit:Erlang] level 0 level 1 level 7 comma	x	-	-
	Readout Route Traffic Measurement		1.3.6.1.4.1.119.2.3.76.2.4.8.1	-	-	-
		Route No.	1.3.6.1.4.1.119.2.3.76.2.4.8.1.1.x (“GET” function is not available for this item)	x	-	-
		Number of Trunks	**Request** Route Traffic Order (rtTrafTable) 1.3.6.1.4.1.119 2.3.76.2.4.8.1.2.x Readout the Number of Measurement Routes, Trunks (rtTrafData) Mesurement of Rout No. (rtTrafRoute) **Response** Response data is an integer.	x	-	-
		Route Traffics	**Request** Route Traffic Measure Order (rtTrafTable) 1.3.6.1.4.1.119 2.3.76.2.4.8.1.3.x Readout the Number of Measure Routes, Traffics (rtTrafData) Mesurement Rout No. (rtTrafRoute) **Response** Response data is coded as ASCII string. Output data is total number of Route Traffics that are using at all of trunks. <Example> 012.345 [Unit:Erlang]	x	-	-

CONTENTS			OBJECT ID	GET	SET	TRAP
Traffic Group			1.3.6.1.4.1.119.2.3.76.2.4.9.1 **Request** Station Peg Count Order (staPegCntfTable) 1.3.6.1.4.1.119 2.3.76.2.4.9.1.x.y Readout Item of Station Peg count Tenant No. (staPegCntTenant) **Response** Response data is an integer.	-	-	-
	Readout Station Peg count		1.3.6.1.4.1.119.2.3.76.2.4.9.1.1.x (“GET” function is not available for this item)	x	-	-
		Tenant No.	1.3.6.1.4.1.119.2.3.76.2.4.9.1.2.x Name for the MIB entry: staPegCntOriginator	x	-	-
		Number of calls for originating	1.3.6.1.4.1.119.2.3.76.2.4.9.1.3.x Name for the MIB entry: staPegCntStaToStaOG	x	-	-
		Number of calls for Station-to-Station Outgoing	1.3.6.1.4.1.119.2.3.76.2.4.9.1.4.x Name for the MIB entry: staPegCntOutgoing	x	-	-
		Number of calls for Outgoing (C.O. Line or Tie Line)	1.3.6.1.4.1.119.2.3.76.2.4.9.1.5.x Name for the MIB entry: staPegCntOtherOG	x	-	-
		Others (relating to Outgoing	1.3.6.1.4.1.119.2.3.76.2.4.9.1.6.x Name for the MIB entry: staPegCntStaToStaIC	x	-	-
		Number of calls for Station-to-Station Incoming	1.3.6.1.4.1.119.2.3.76.2.4.9.1.7.x Name for the MIB entry: staPegCntIncoming	x	-	-
		Number of calls for Incoming (C.O. Line or Tie Line)	1.3.6.1.4.1.119.2.3.76.2.4.9.1.8.x Name for the MIB entry: staPegCntRinging	x	-	-
		Number of calls for Ringing				

CONTENTS			OBJECT ID	GET	SET	TRAP
Traffic Group	Readout Station Peg count	Number of lines for No Answer	1.3.6.1.4.1.119.2.3.76.2.4.9.1.9.x Name for the MIB entry: staPegCntNoAnswer30sec	x	-	-
		Number of Answers	1.3.6.1.4.1.119.2.3.76.2.4.9.1.10.x Name for the MIB entry: staPegCntAnswer	x	-	-
		Number of No Answers	1.3.6.1.4.1.119.2.3.76.2.4.9.1.11.x Name for the MIB entry: staPegCntNoAnswer	x	-	-
		Others (relating to Incoming	1.3.6.1.4.1.119.2.3.76.2.4.9.1.12.x Name for the MIB entry: staPegCntOtherIC	x	-	-
	Readout Att. Peg count (Specify the Tenant No.)		1.3.6.1.4.1.119.2.3.76.2.4.10.1 **Request**  **Response** Response data is an integer.	-	-	-
		Att. No.	1.3.6.1.4.1.119.2.3.76.2.4.10.1.1.x (“GET” function is not available for this item)	x	-	-
		C.O. Line	1.3.6.1.4.1.119.2.3.76.2.4.10.1.2.x Name for the MIB entry: attPegCntTNLdn	x	-	-
		Attendant/ Adminis-tration Station	1.3.6.1.4.1.119.2.3.76.2.4.10.1.3.x Name for the MIB entry: attPegCntTNAtndAdm	x	-	-
		Recall	1.3.6.1.4.1.119.2.3.76.2.4.10.1.4.x Name for the MIB entry: attPegCntTNRcl	x	-	-
		C.O. line 2	1.3.6.1.4.1.119.2.3.76.2.4.10.1.5.x Name for the MIB entry: attPegCntTNFx	x	-	-
		C.O. line 3	1.3.6.1.4.1.119.2.3.76.2.4.10.1.6.x Name for the MIB entry: attPegCntTNWats	x	-	-
		C.O. line 4	1.3.6.1.4.1.119.2.3.76.2.4.10.1.7.x Name for the MIB entry: attPegCntTNCcsa	x	-	-
		Tie Line	1.3.6.1.4.1.119.2.3.76.2.4.10.1.8.x Name for the MIB entry: attPegCntTNTie	x	-	-
		Busy	1.3.6.1.4.1.119.2.3.76.2.4.10.1.9.x Name for the MIB entry: attPegCntTNBusy	x	-	-
		No Answer	1.3.6.1.4.1.119.2.3.76.2.4.10.1.10.x Name for the MIB entry: attPegCntTNNans	x	-	-
		Restriction	1.3.6.1.4.1.119.2.3.76.2.4.10.1.11.x Name for the MIB entry: attPegCntTNIcpt	x	-	-

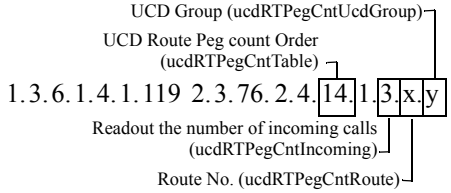
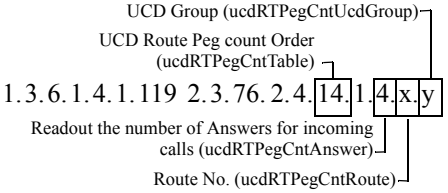
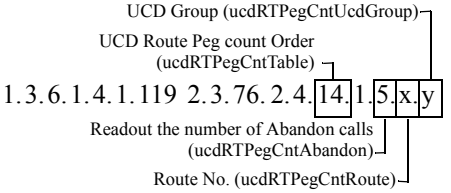
CONTENTS			OBJECT ID	GET	SET	TRAP
Traffic Group	Readout Att. Peg count (Specify the Tenant No.)	Special common battery/ House Phone Station	1.3.6.1.4.1.119.2.3.76.2.4.10.1.12.x Name for the MIB entry: attPegCntTNScbHp	x	-	-
		Inter Position Transfer	1.3.6.1.4.1.119.2.3.76.2.4.10.1.13.x Name for the MIB entry: attPegCntTNTf	x	-	-
		Do Not Disturb	1.3.6.1.4.1.119.2.3.76.2.4.10.1.14.x Name for the MIB entry: attPegCntTNDnd	x	-	-
		Guest Station	1.3.6.1.4.1.119.2.3.76.2.4.10.1.15.x Name for the MIB entry: attPegCntTNGst	x	-	-
		Howler Call	1.3.6.1.4.1.119.2.3.76.2.4.10.1.16.x Name for the MIB entry: attPegCntTNHwc	x	-	-
		Long Line Telephone	1.3.6.1.4.1.119.2.3.76.2.4.10.1.17.x Name for the MIB entry: attPegCntTNLt	x	-	-
		Total	1.3.6.1.4.1.119.2.3.76.2.4.10.1.18.x Name for the MIB entry: attPegCntTNTotal	x	-	-
	Readout Att. Peg count (Specify the Att. No.)		1.3.6.1.4.1.119.2.3.76.2.4.11.1 **Request** Attendant Console Peg count Order (attPegCntATTTable) 1.3.6.1.4.1.119 2.3.76.2.4.11.1.x.y Readout Item of Attendant Console Peg count Attendant Console No. (attPegCntATTnumber) **Response** Response data is an integer.	-	-	-
		Att. No.	1.3.6.1.4.1.119.2.3.76.2.4.11.1.1.x ("GET" function is not available for this item)	x	-	-
		C.O. Line	1.3.6.1.4.1.119.2.3.76.2.4.11.1.2.x Name for the MIB entry: attPegCntATTLdn	x	-	-
		Attendant/ Administration Station	1.3.6.1.4.1.119.2.3.76.2.4.11.1.3.x Name for the MIB entry: attPegCntATTAtndAdm	x	-	-
		Recall	1.3.6.1.4.1.119.2.3.76.2.4.11.1.4.x Name for the MIB entry: attPegCntATTRcl	x	-	-
		C.O. line 2	1.3.6.1.4.1.119.2.3.76.2.4.11.1.5.x Name for the MIB entry: attPegCntATTFx	x	-	-
		C.O. line 3	1.3.6.1.4.1.119.2.3.76.2.4.11.1.6.x Name for the MIB entry: attPegCntATTWats	x	-	-
		C.O. line 4	1.3.6.1.4.1.119.2.3.76.2.4.11.1.7.x Name for the MIB entry: attPegCntCcsa	x	-	-
		Tie Line	1.3.6.1.4.1.119.2.3.76.2.4.11.1.8.x Name for the MIB entry: attPegCntATTTie	x	-	-

CONTENTS			OBJECT ID	GET	SET	TRAP
Traffic Group	Readout Att. Peg count (Specify the Att. No.)	Busy	1.3.6.1.4.1.119.2.3.76.2.4.11.1.9.x Name for the MIB entry: attPegCntATTBusy	x	-	-
		No Answer	1.3.6.1.4.1.119.2.3.76.2.4.11.1.10.x Name for the MIB entry: attPegCntATTNans	x	-	-
		Restriction	1.3.6.1.4.1.119.2.3.76.2.4.11.1.11.x Name for the MIB entry: attPegCntATTIcpt	x	-	-
		Special common battery/ House Phone Station	1.3.6.1.4.1.119.2.3.76.2.4.11.1.12.x Name for the MIB entry: attPegCntATTScbHp	x	-	-
		Inter Position Transfer	1.3.6.1.4.1.119.2.3.76.2.4.11.1.13.x Name for the MIB entry: attPegCntATTTf	x	-	-
		Do Not Disturb	1.3.6.1.4.1.119.2.3.76.2.4.11.1.14.x Name for the MIB entry: attPegCntATTDnd	x	-	-
		Guest Station	1.3.6.1.4.1.119.2.3.76.2.4.11.1.15.x Name for the MIB entry: attPegCntATTGst	x	-	-
		Howler Call	1.3.6.1.4.1.119.2.3.76.2.4.11.1.16.x Name for the MIB entry: attPegCntATTHwc	x	-	-
		Long Line Telephone	1.3.6.1.4.1.119.2.3.76.2.4.11.1.17.x Name for the MIB entry: attPegCntATTLt	x	-	-
		Total	1.3.6.1.4.1.119.2.3.76.2.4.11.1.18.x Name for the MIB entry: attPegCntATTTotal	x	-	-
	Readout Route Peg count		1.3.6.1.4.1.119.2.3.76.2.4.12.1 **Request** Route No. (rtPegCntRoute) Route Peg Count Order (rtPegCntTable) 1.3.6.1.4.1.119 2.3.76. 2.4.12.1.x.y.z Readout Item of Route Peg count Trunk Type (PegCntTrunkType) 1=Internal Trunk / 2=External Trunk **Response** Response data is an integer.	-	-	-
		Trunk Type	1.3.6.1.4.1.119.2.3.76.2.4.12.1.1.x.y (“GET” function is not available for this item)	x	-	-
		Route No.	1.3.6.1.4.1.119.2.3.76.2.4.12.1.2.x.y (“GET” function is not available for this item)	x	-	-
		Number of attempts	1.3.6.1.4.1.119.2.3.76.2.4.12.1.3.x.y Name for the MIB entry: rtPegCntAttempts	x	-	-
		Number of Blocks	1.3.6.1.4.1.119.2.3.76.2.4.12.1.4.x.y Name for the MIB entry: rtPegCntBlock	x	-	-
		Number of establishments	1.3.6.1.4.1.119.2.3.76.2.4.12.1.5.x.y Name for the MIB entry: rtPegCntEstablished	x	-	-

CONTENTS			OBJECT ID	GET	SET	TRAP
Traffic Group	Readout Route Peg count	Number of Outgoing calls	1.3.6.1.4.1.119.2.3.76.2.4.12.1.6.x.y Name for the MIB entry: rtPegCntOutgoing	x	-	-
		Number of Incoming calls	1.3.6.1.4.1.119.2.3.76.2.4.12.1.7.x.y Name for the MIB entry: rtPegCntIncoming	x	-	-
		Number of access to ORT/IRT	1.3.6.1.4.1.119.2.3.76.2.4.12.1.8.x.y Name for the MIB entry: rtPegCntRegisterAccess	x	-	-
		Number of connec-tions to Att.	1.3.6.1.4.1.119.2.3.76.2.4.12.1.9.x.y Name for the MIB entry: rtPegCntAtt	x	-	-
		Number of connec-tions to station within the self office	1.3.6.1.4.1.119.2.3.76.2.4.12.1.10.x.y Name for the MIB entry: rtPegCntStation	x	-	-
		Number of Tandem Connec-tions	1.3.6.1.4.1.119.2.3.76.2.4.12.1.11.x.y Name for the MIB entry: rtPegCntTandem	x	-	-
		Others	1.3.6.1.4.1.119.2.3.76.2.4.12.1.12.x.y Name for the MIB entry: rtPegCntOther	x	-	-
		Virtual Tie Line (E-CCIS) a number of Outgoing setting	1.3.6.1.4.1.119.2.3.76.2.4.12.1.13.x.y Name for the MIB entry: rtPegCntEccisOut	x	-	-
		Virtual Tie Line (E-CCIS) a number of Incoming setting	1.3.6.1.4.1.119.2.3.76.2.4.12.1.14.x.y Name for the MIB entry: rtPegCntEccisIn	x	-	-

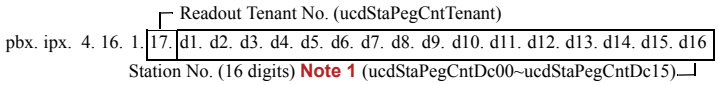
CONTENTS			OBJECT ID	GET	SET	TRAP
Traffic Group	Readout Service Peg count		1.3.6.1.4.1.119.2.3.76.2.4.13.1 **Request** Service Peg count Order(svPegCntTable) 1.3.6.1.4.1.119 2.3.76.2.4.13.1.x.y Readout Item of Service Peg count Tenant No. (svPegCntTenant) **Response** Response data is an integer.	-	-	-
		Tenant No.	1.3.6.1.4.1.119.2.3.76.2.4.13.1.1.x ("GET" function is not available for this item)	x	-	-
		Call Hold	1.3.6.1.4.1.119.2.3.76.2.4.13.1.2.x Name for the MIB entry: svPegCntCallHold	x	-	-
		Camp-On Set	1.3.6.1.4.1.119.2.3.76.2.4.13.1.3.x Name for the MIB entry: svPegCntCampOnSet	x	-	-
		Camp-On Release	1.3.6.1.4.1.119.2.3.76.2.4.13.1.4.x Name for the MIB entry: svPegCntCampOnRel	x	-	-
		Executive Right-of-Way	1.3.6.1.4.1.119.2.3.76.2.4.13.1.5.x Name for the MIB entry: svPegCntInterrupt	x	-	-
		Call Waiting - Originating	1.3.6.1.4.1.119.2.3.76.2.4.13.1.6.x Name for the MIB entry: svPegCntCallWaitOrg	x	-	-
		Call Waiting - Terminating	1.3.6.1.4.1.119.2.3.76.2.4.13.1.7.x Name for the MIB entry: svPegCntCallWaitTer	x	-	-
		Call Pickup	1.3.6.1.4.1.119.2.3.76.2.4.13.1.8.x Name for the MIB entry: svPegCntCallPickUp	x	-	-
		Call Pickup - Direct	1.3.6.1.4.1.119.2.3.76.2.4.13.1.9.x Name for the MIB entry: svPegCntCallPickUpSpec	x	-	-
		Call Forwarding - All Calls (Set)	1.3.6.1.4.1.119.2.3.76.2.4.13.1.10.x Name for the MIB entry: svPegCntCFAllSet	x	-	-
		Call Forwarding - All Calls (Release)	1.3.6.1.4.1.119.2.3.76.2.4.13.1.11.x Name for the MIB entry: scPegCntCFAllRel	x	-	-
		Call Forwarding - Busy Line (Set)	1.3.6.1.4.1.119.2.3.76.2.4.13.1.12.x Name for the MIB entry: svPegCntCFBusySet	x	-	-
		Call Forwarding - Busy Line (Release)	1.3.6.1.4.1.119.2.3.76.2.4.13.1.13.x Name for the MIB entry: svPegCntCFBusyRel	x	-	-

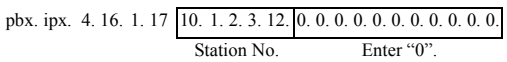
CONTENTS			OBJECT ID	GET	SET	TRAP
Traffic Group	Readout Service Peg count	Call Forwarding - Don't Answer (Set)	1.3.6.1.4.1.119.2.3.76.2.4.13.1.14.x Name for the MIB entry: svPegCntCFNoAnsSet	x	-	-
		Call Forwarding - Don't Answer (Release)	1.3.6.1.4.1.119.2.3.76.2.4.13.1.15.x Name for the MIB entry: svPegCntCFNoAnsRel	x	-	-
		Speed Calling - System	1.3.6.1.4.1.119.2.3.76.2.4.13.1.16.x Name for the MIB entry: svPegCntFixAbbreDial	x	-	-
		Speed Calling - Station	1.3.6.1.4.1.119.2.3.76.2.4.13.1.17.x Name for the MIB entry: svPegCntVariAbbreDial	x	-	-
		Speed Calling - Station (Set)	1.3.6.1.4.1.119.2.3.76.2.4.13.1.18.x Name for the MIB entry: svPegCntVariAbbreDialSet	x	-	-
		Distributed Answer	1.3.6.1.4.1.119.2.3.76.2.4.13.1.19.x Name for the MIB entry: svPegCntDistribAns	x	-	-
		Trunk Queuing	1.3.6.1.4.1.119.2.3.76.2.4.13.1.20.x Name for the MIB entry: svPegCntTrunkQue	x	-	-
		Outgoing Trunk Queuing (Set)	1.3.6.1.4.1.119.2.3.76.2.4.13.1.21.x Name for the MIB entry: svPegCntTrunkCamponSet	x	-	-
		Outgoing Trunk Queuing (Release)	1.3.6.1.4.1.119.2.3.76.2.4.13.1.22.x Name for the MIB entry: svPegCntTrunkCamponRel	x	-	-
		Call Transfer - All Calls	1.3.6.1.4.1.119.2.3.76.2.4.13.1.23.x Name for the MIB entry: svPegCntAutoTrans	x	-	-
		Blind Transfer	1.3.6.1.4.1.119.2.3.76.2.4.13.1.24.x Name for the MIB entry: svPegCntSimpleOpe	x	-	-
		Three-Way Calling	1.3.6.1.4.1.119.2.3.76.2.4.13.1.25.x Name for the MIB entry: svPegCntmeetings	x	-	-

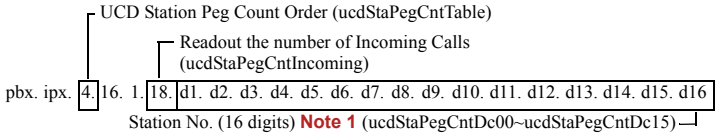
CONTENTS			OBJECT ID	GET	SET	TRAP
Traffic Group	Readout the number of UCD Route Peg Counts		1.3.6.1.4.1.119.2.3.76.2.4.14.1	-	-	-
		Route No.	1.3.6.1.4.1.119.2.3.76.2.4.14.1.1.x.y (“GET” function is not available for this item)	x	-	-
		UCD Group	1.3.6.1.4.1.119.2.3.76.2.4.14.1.2.x.y (“GET” function is not available for this item)	x	-	-
		Readout the number of Incoming calls	**Request**  **Response** Response data is an integer.	x	-	-
		Readout the number of Answers	**Request**  **Response** Response data is an integer.	x	-	-
		Readout the number of Abandon calls	**Request**  **Response** Response data is an integer.	x	-	-

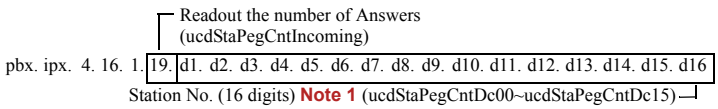
CONTENTS			OBJECT ID	GET	SET	TRAP
Traffic Group	Readout the number of UCD Group Peg Counts		1.3.6.1.4.1.119.2.3.76.2.4.15.1	-	-	-
		UCD Group	1.3.6.1.4.1.119.2.3.76.2.4.15.1.1.x ("GET" function is not available for this item)	x	-	-
		Readout the number of Incoming Calls	**Request** UCD Group Peg count Order (ucdGrPegCntTable) 1.3.6.1.4.1.119 2.3.76. 2.4.15.1.2.x Readout the number of Incoming calls (ucdGrPegCntIncoming) UCD Group (ucdGrPegCntUcdGroup) **Response** Response data is an integer.	x	-	-
		Read the number of Answers	**Request** UCD Group Peg count Order (ucdGrPegCntTable) 1.3.6.1.4.1.119 2.3.76. 2.4.15.1.3.x Readout the number of Answers (ucdRTPegCntIncoming) UCD Group (ucdGrPegCntUcdGroup) **Response** Response data is an integer.	x	-	-
		Readout the number of Abandon Calls	**Request** UCD Group Peg Count Order (ucdGrPegCntTable) 1.3.6.1.4.1.119 2.3.76. 2.4.15.1.4.x Readout the number of Abandon calls (ucdGrPegCntAbandon) UCD Group (ucdGrPegCntUcdGroup) **Response** Response data is an integer.	x	-	-
	Readout the number of UCD Station Peg counts		1.3.6.1.4.1.119.2.3.76.2.4.16.1 **Note:** "pbx.ipx" indicates "1.3.6.1.4.1.119.2.3.76.2" in this item.	-	-	-
		Station No. [1st Digit]	pbx.ipx.4.16.1.1.d1.d2.d3.d4.d5.d6.d7.d8.d9.d10.d11.d12.d13.d14.d14.d16 ("GET" function is not available for this item)	x	-	-
		Station No. [2nd Digit]	pbx.ipx.4.16.1.2.d1.d2.d3.d4.d5.d6.d7.d8.d9.d10.d11.d12.d13.d14.d15.d16 ("GET" function is not available for this item)	x	-	-
		Station No. [3rd Digit]	pbx.ipx.4.16.1.3.d1.d2.d3.d4.d5.d6.d7.d8.d9.d10.d11.d12.d13.d14.d15.d16 ("GET" function is not available for this item)	x	-	-
		Station No. [4th Digit]	pbx.ipx.4.16.1.4.d1.d2.d3.d4.d5.d6.d7.d8.d9.d10.d11.d12.d13.d14.d15.d16 ("GET" function is not available for this item)	x	-	-

CONTENTS			OBJECT ID	GET	SET	TRAP
Traffic Group	Readout the number of UCD Station Peg counts	Station No. [5th Digit]	pbx.ipx.4.16.1.5.d1.d2.d3.d4.d5.d6.d7.d8.d9.d10.d11.d12.d13.d14.d15.d16 ("GET" function is not available for this item)	x	-	-
		Station No. [6th Digit]	pbx.ipx.4.16.1.6.d1.d2.d3.d4.d5.d6.d7.d8.d9.d10.d11.d12.d13.d14.d15.d16 ("GET" function is not available for this item)	x	-	-
		Station No. [7th Digit]	pbx.ipx.4.16.1.7.d1.d2.d3.d4.d5.d6.d7.d8.d9.d10.d11.d12.d13.d14.d15.d16 ("GET" function is not available for this item)	x	-	-
		Station No. [8th Digit]	pbx.ipx.4.16.1.8.d1.d2.d3.d4.d5.d6.d7.d8.d9.d10.d11.d12.d13.d14.d15.d16 ("GET" function is not available for this item)	x	-	-
		Station No. [9th Digit]	pbx.ipx.4.16.1.9.d1.d2.d3.d4.d5.d6.d7.d8.d9.d10.d11.d12.d13.d14.d15.d16 ("GET" function is not available for this item)	x	-	-
		Station No. [10th Digit]	pbx.ipx.4.16.1.10.d1.d2.d3.d4.d5.d6.d7.d8.d9.d10.d11.d12.d13.d14.d15.d16 ("GET" function is not available for this item)	x	-	-
		Station No. [11th Digit]	pbx.ipx.4.16.1.11.d1.d2.d3.d4.d5.d6.d7.d8.d9.d10.d11.d12.d13.d14.d15.d16 ("GET" function is not available for this item)	x	-	-
		Station No. [12th Digit]	pbx.ipx.4.16.1.12.d1.d2.d3.d4.d5.d6.d7.d8.d9.d10.d11.d12.d13.d14.d15.d16 ("GET" function is not available for this item)	x	-	-
		Station No. [13th Digit]	pbx.ipx.4.16.1.13.d1.d2.d3.d4.d5.d6.d7.d8.d9.d10.d11.d12.d13.d14.d15.d16 ("GET" function is not available for this item)	x	-	-
		Station No. [14th Digit]	pbx.ipx.4.16.1.14.d1.d2.d3.d4.d5.d6.d7.d8.d9.d10.d11.d12.d13.d14.d15.d16 ("GET" function is not available for this item)	x	-	-
		Station No. [15th Digit]	pbx.ipx.4.16.1.15.d1.d2.d3.d4.d5.d6.d7.d8.d9.d10.d11.d12.d13.d14.d15.d16 ("GET" function is not available for this item)	x	-	-
		Station No. [16th Digit]	pbx.ipx.4.16.1.16.d1.d2.d3.d4.d5.d6.d7.d8.d9.d10.d11.d12.d13.d14.d15.d16 ("GET" function is not available for this item)	x	-	-

<table>
<tr><th colspan="3">CONTENTS</th><th>OBJECT ID</th><th>GET</th><th>SET</th><th>TRAP</th></tr>
<tr><td rowspan="3">Traffic Group</td><td rowspan="3">Readout the number of UCD Station Peg counts</td><td>Tenant</td><td>Request
 

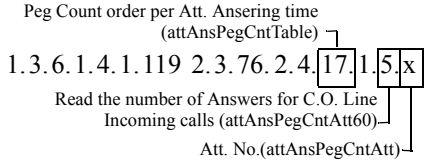
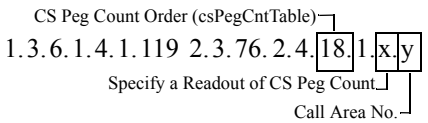
Note: “pbx.ipx” indicates “1.3.6.1.4.1.119.2.3.76.2”.
Note:1 Digits 0~12 can be entered to the Station No. as follows.
<table><tr><td>Station No.</td><td>no data</td><td>1</td><td>2</td><td>3</td><td>4</td><td>5</td><td>6</td><td>7</td><td>8</td><td>9</td><td>0</td><td>*</td><td>#</td></tr><tr><td>Input value</td><td>0</td><td>1</td><td>2</td><td>3</td><td>4</td><td>5</td><td>6</td><td>7</td><td>8</td><td>9</td><td>10</td><td>11</td><td>12</td></tr></table>Blank fields in Station number must be filled with “0”.
In the case of Station No.=”0123#” as shown in the example below:
 

Response
Response data is an integer (1~63).</td><td>x</td><td>-</td><td>-</td></tr>
<tr><td>Readout the number of Incoming calls</td><td>Request
 

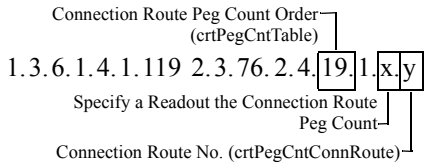
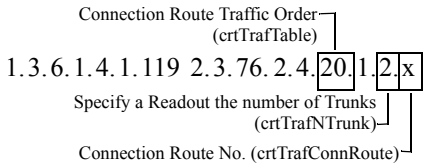
Note: “pbx.ipx” indicates “1.3.6.1.4.1.119.2.3.76.2”.
Response
Response data is an integer.</td><td>x</td><td>-</td><td>-</td></tr>
<tr><td>Readout the number of Answers</td><td>Request
 

Note: “pbx.ipx” indicates “1.3.6.1.4.1.119.2.3.76.2”.
Response
Response data is an integer.</td><td>x</td><td>-</td><td>-</td></tr>
</table>

CONTENTS			OBJECT ID	GET	SET	TRAP
Traffic Group	Readout the number of UCD Station Peg counts	Readout the number of Abandon calls	**Request** Specify the Readout the number of Abandon Calls (ucdStaPegCntIncoming) pbx. ipx. 4. 16. 1. 20. d1. d2. d3. d4. d5. d6. d7. d8. d9. d10. d11. d12. d13. d14. d15. d16 Station No. (16 digits) **Note 1** (ucdStaPegCntDc00~ucdStaPegCntDc15) **Note:** "pbx.ipx" indicates "1.3.6.1.4.1.119.2.3.76.2".	x	-	-
			Response Response data is an integer.			
	Readout the number of Peg counts per Att. Answering time		1.3.6.1.4.1.119.2.3.76.2.4.17.1	-	-	-
		Att. No.	1.3.6.1.4.1.119.2.3.76.2.4.17.1.1.x ("GET" function is not available for this item)	x	-	-
		Readout the number of Answers for C.O. Line Incoming calls (0~10 Sec.)	**Request** Peg Count order per Att. Ansering time (attAnsPegCntTable) 1.3.6.1.4.1.119 2.3.76.2.4.17.1.2.x Read the number of Answers for C.O. Line Incoming calls (0~10 Sec) (attAnsPegCntAtt0) Att. No.(attAnsPegCntAtt)	x	-	-
			Response Response data is an integer.			
		Readout the number of Answers for C.O. Line Incoming calls (10~30 Sec.)	**Request** Peg Count order per Att. Ansering time (attAnsPegCntTable) 1.3.6.1.4.1.119 2.3.76.2.4.17.1.3.x Readout the number of Answers for C.O. Line Incoming calls (10~30 Sec) (attAnsPegCntAtt10) Att. No.(attAnsPegCntAtt)	x	-	-
			Response Response data is an integer.			
		Readout the number of Answers for C.O. Line Incoming calls (30~60 Sec.)	**Request** Peg Count order per Att. Ansering time (attAnsPegCntTable) 1.3.6.1.4.1.119 2.3.76.2.4.17.1.4.x Readout the number of Answers for C.O. Line Incoming calls (attAnsPegCntAtt30) Att. No.(attAnsPegCntAtt)	x	-	-
			Response Response data is an integer.			

CONTENTS			OBJECT ID	GET	SET	TRAP
Traffic Group	Readout the number of Peg counts per Att. Answer-ing time	Readout the number of Answers for C.O. Line Incoming calls (more than 60 Sec.)	**Request**  **Response** Response data is an integer.	x	-	-
	Readout the number of CS Peg Count		1.3.6.1.4.1.119.2.3.76.2.4.18.1 **Request**  **Response** Response data is an integer.	-	-	-
		Call Area No.	1.3.6.1.4.1.119.2.3.76.2.4.18.1.1.x (“GET” function is not available for this item)	x	-	-
		Total num-ber of outgoing	1.3.6.1.4.1.119.2.3.76.2.4.18.1.2.x Name for the MIB entry: csPegCntOutgoing	x	-	-
		Number of Answers	1.3.6.1.4.1.119.2.3.76.2.4.18.1.3.x Name for the MIB entry: csPegCntAnswer	x	-	-
		Number of Ringing	1.3.6.1.4.1.119.2.3.76.2.4.18.1.4.x Name for the MIB entry: csPegCntRinging	x	-	-
		Number of PS Answers	1.3.6.1.4.1.119.2.3.76.2.4.18.1.5.x Name for the MIB entry: csPegCntPSAnswer	x	-	-
		Number of Line Key Answers	1.3.6.1.4.1.119.2.3.76.2.4.18.1.6.x Name for the MIB entry: csPegCntLineKeyAnswer	x	-	-
		Number of Paging Incoming	1.3.6.1.4.1.119.2.3.76.2.4.18.1.7.x Name for the MIB entry: csPegCntPaging	x	-	-
		Number of Paging Answers	1.3.6.1.4.1.119.2.3.76.2.4.18.1.8.x Name for the MIB entry: csPegCntPagingAnswer	x	-	-
		Number of No Answers	1.3.6.1.4.1.119.2.3.76.2.4.18.1.9.x Name for the MIB entry: csPegCntNoAnswer	x	-	-
		Number of Hand Over Attempts	1.3.6.1.4.1.119.2.3.76.2.4.18.1.10.x Name for the MIB entry: csPegCntHandOverAttempts	x	-	-
		Number of Hand Overs	1.3.6.1.4.1.119.2.3.76.2.4.18.1.11.x Name for the MIB entry: csPegCntHandOver	x	-	-

CONTENTS			OBJECT ID	GET	SET	TRAP
Traffic Group	Readout the number of CS Peg Counts	Total number of request a Position Assignment	1.3.6.1.4.1.119.2.3.76.2.4.18.1.12.x Name for the MIB entry: csPegCntRegistAttempts	x	-	-
		Number of Bch All Busy	1.3.6.1.4.1.119.2.3.76.2.4.18.1.13.x Name for the MIB entry: csPegCntBchAllBusy	x	-	-
		Number of Wrong Access	1.3.6.1.4.1.119.2.3.76.2.4.18.1.14.x Name for the MIB entry: csPegCntWrongAccess	x	-	-
		Number of Roaming to Home Office	1.3.6.1.4.1.119.2.3.76.2.4.18.1.15.x Name for the MIB entry: csPegCntHomeRoaming	x	-	-
		Number of Incoming calls to Home Office	1.3.6.1.4.1.119.2.3.76.2.4.18.1.16.x Name for the MIB entry: csPegCntHomeRoamingIC	x	-	-
		Roaming Count for Visitor Office	1.3.6.1.4.1.119.2.3.76.2.4.18.1.17.x Name for the MIB entry: csPegCntVisitRoaming	x	-	-
		Number of Wrong Access to Visitor Office	1.3.6.1.4.1.119.2.3.76.2.4.18.1.18.x Name for the MIB entry: csPegCntVisitWrongAccess	x	-	-
		Number of Tch Connection Failure	1.3.6.1.4.1.119.2.3.76.2.4.18.1.19.x Name for the MIB entry: csPegCntTchConnFail	x	-	-
		Number of Disconnection	1.3.6.1.4.1.119.2.3.76.2.4.18.1.20.x Name for the MIB entry: csPegCntDisconnection	x	-	-
		Number of Channel Change	1.3.6.1.4.1.119.2.3.76.2.4.18.1.21.x Name for the MIB entry: csPegCntChannelChange	x	-	-

CONTENTS			OBJECT ID	GET	SET	TRAP
Traffic Group	Readout the number of Connection Route Peg Counts		1.3.6.1.4.1.119.2.3.76.2.4.19.1 **Request**  **Response** Response data is an integer.	-	-	-
		Connection Route No.	1.3.6.1.4.1.119.2.3.76.2.4.19.1.1.x (“GET” function is not available for this item)	x	-	-
		Number of attempts	1.3.6.1.4.1.119.2.3.76.2.4.19.1.2.x Name for the MIB entry: crtPegCntAttempts	x	-	-
		Number of Blocks	1.3.6.1.4.1.119.2.3.76.2.4.19.1.3.x Name for the MIB entry: crtPegCntBlock	x	-	-
		Number of Outgoing calls	1.3.6.1.4.1.119.2.3.76.2.4.19.1.4.x Name for the MIB entry: crtPegCntOutgoing	x	-	-
		Number of Incoming calls	1.3.6.1.4.1.119.2.3.76.2.4.19.1.5.x Name for the MIB entry: crtPegCntIncoming	x	-	-
		Number of Connections	1.3.6.1.4.1.119.2.3.76.2.4.19.1.6.x Name for the MIB entry: crtPegCntIcConnection	x	-	-
		Number of Tandem Connections	1.3.6.1.4.1.119.2.3.76.2.4.19.1.7.x Name for the MIB entry: crtPegCntTandem	x	-	-
		Others	1.3.6.1.4.1.119.2.3.76.2.4.19.1.8.x Name for the MIB entry: crtPegCntOther	x	-	-
	Readout Connection Route Traffic		1.3.6.1.4.1.119.2.3.76.2.4.20.1	-	-	-
		Connection Route No.	1.3.6.1.4.1.119.2.3.76.2.4.20.1.1.x (“GET” function is not available for this item)	x	-	-
		Readout the number of Trunks	**Request**  **Response** Response data is an integer.	x	-	-

<table>
<tr><th colspan="3">CONTENTS</th><th>OBJECT ID</th><th>GET</th><th>SET</th><th>TRAP</th></tr>
<tr><td rowspan="9">Traffic Group</td><td rowspan="2">Readout Connec-tion Route Traffic</td><td>Readout the Traffic</td><td>Request
Connection Route Traffic Order (crtTrafTable)
1.3.6.1.4.1.119 2.3.76.2.4.[20].1.[3].[x]
Specify a Readout the Traffic (crtTrafData)
Connection Route No. (crtTrafConnRoute)
Response
Response data is coded as ASCII string.</td><td>x</td><td>-</td><td>-</td></tr>
<tr><td colspan="2">Readout the Number of IP Service Peg Counts</td><td>1.3.6.1.4.1.119.2.3.76.2.4.21.1
Request
IP Service Peg Count Order (crtPegCntTable)
1.3.6.1.4.1.119 2.3.76.2.4.[21].1.[x].[n0.n1.n2]
Specify a Readout the IP Service Peg Count
Node No. (3 digits) Note 2 (ipSvPegCntNode0~ipSvPegCnt-Node2)
Note:2 Digits 0~13 can be entered to the Node No. as follows.
<table><tr><td>Station No.</td><td>no data</td><td>1</td><td>2</td><td>3</td><td>4</td><td>5</td><td>6</td><td>7</td><td>8</td><td>9</td><td>0</td><td>*</td><td>#</td><td>P</td></tr><tr><td>Input value</td><td>0</td><td>1</td><td>2</td><td>3</td><td>4</td><td>5</td><td>6</td><td>7</td><td>8</td><td>9</td><td>10</td><td>11</td><td>12</td><td>13</td></tr></table>
Node number up to 3 digits with unused digits filled with “0”.
In case of Node No.=”01” as shown in the example below:
Enter “0”.
1.3.6.1.4.1.119 2.3.76.2.4.21.1.13.[10.1.][0]
Node No.
Response
Response data is an integer.</td><td>-</td><td>-</td><td>-</td></tr>
<tr><td></td><td>Node No. (1st Digit)</td><td>1.3.6.1.4.1.119.2.3.76.2.4.21.1.1.n0.n1.n2
(“GET” function is not available for this item)</td><td>x</td><td>-</td><td>-</td></tr>
<tr><td></td><td>Node No. (2nd Digit)</td><td>1.3.6.1.4.1.119.2.3.76.2.4.21.1.2.n0.n1.n2
(“GET” function is not available for this item)</td><td>x</td><td>-</td><td>-</td></tr>
<tr><td></td><td>Node No. (3rd Digit)</td><td>1.3.6.1.4.1.119.2.3.76.2.4.21.1.3.n0.n1.n2
(“GET” function is not available for this item)</td><td>x</td><td>-</td><td>-</td></tr>
<tr><td></td><td>Number of attempts</td><td>1.3.6.1.4.1.119.2.3.76.2.4.21.1.4.n0.n1.n2
Name for the MIB entry: ipSvPegCntAttempts</td><td>x</td><td>-</td><td>-</td></tr>
<tr><td></td><td>Number of Blocks</td><td>1.3.6.1.4.1.119.2.3.76.2.4.21.1.5.n0.n1.n2
Name for the MIB entry: ipSvPegCntBlock</td><td>x</td><td>-</td><td>-</td></tr>
<tr><td></td><td>Number of Outgoing calls</td><td>1.3.6.1.4.1.119.2.3.76.2.4.21.1.6.n0.n1.n2
Name for the MIB entry: ipSvPegCntOutgoing</td><td>x</td><td>-</td><td>-</td></tr>
</table>

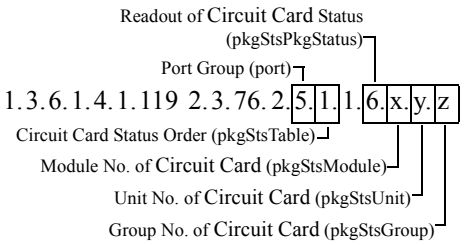
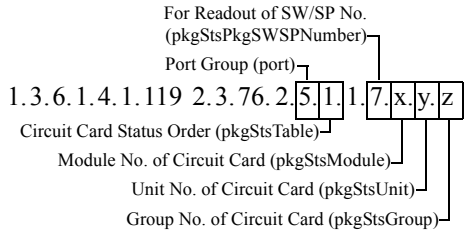
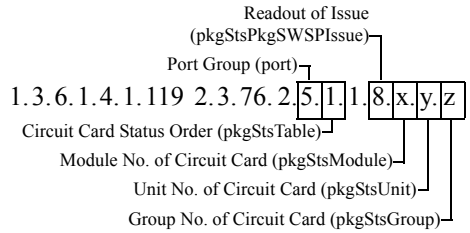
CONTENTS			OBJECT ID	GET	SET	TRAP
Traffic Group	Readout the Number of IP Service Peg Counts	Outgoing Uncompleted 1 (Reason: Missing number, Invalid number)	1.3.6.1.4.1.119.2.3.76.2.4.21.1.7.n0.n1.n2 Name for the MIB entry: ipSvPegCntOutUncomplete1	x	-	-
		Outgoing Uncompleted 2 (Reason: Receive station is busy.)	1.3.6.1.4.1.119.2.3.76.2.4.21.1.8.n0.n1.n2 Name for the MIB entry: ipSvPegCntOutUncomplete2	x	-	-
		Outgoing Uncompleted 3 (Reason: No answer from Receive station.)	1.3.6.1.4.1.119.2.3.76.2.4.21.1.9.n0.n1.n2 Name for the MIB entry: ipSvPegCntOutUncomplete3	x	-	-
		Outgoing Uncompleted Others (Reason: Others)	1.3.6.1.4.1.119.2.3.76.2.4.21.1.10.n0.n1.n2 Name for the MIB entry: ipSvPegCntOutUncomplete4	x	-	-
		Number of Incoming Calls	1.3.6.1.4.1.119.2.3.76.2.4.21.1.11.n0.n1.n2 Name for the MIB entry: ipSvPegCntIncoming	x	-	-
		Incoming Uncompleted 1 (Reason: Missing number, Invalid number)	1.3.6.1.4.1.119.2.3.76.2.4.21.1.12.n0.n1.n2 Name for the MIB entry: ipSvPegCntInUncomplete1	x	-	-
		Incoming Uncompleted 2 (Reason: Receive station is busy.)	1.3.6.1.4.1.119.2.3.76.2.4.21.1.13.n0.n1.n2 Name for the MIB entry: ipSvPegCntInUncomplete2	x	-	-
		Incoming Uncompleted 3 (Reason: Receive station is restricted)	1.3.6.1.4.1.119.2.3.76.2.4.21.1.14.n0.n1.n2 Name for the MIB entry: ipSvPegCntInUncomplete3	x	-	-

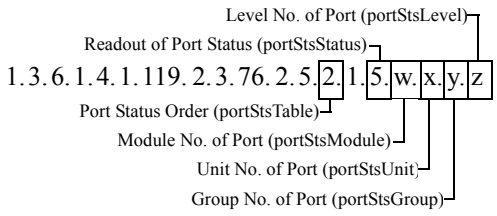
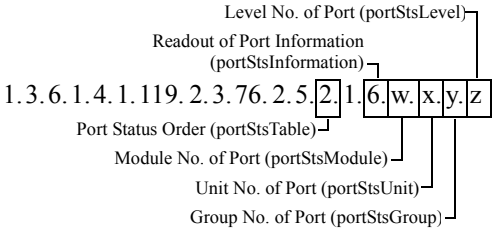
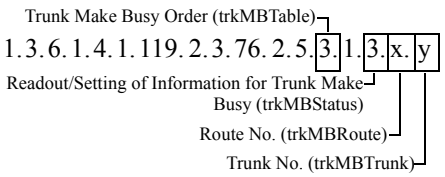
<table>
<tr><th colspan="3">CONTENTS</th><th>OBJECT ID</th><th>GET</th><th>SET</th><th>TRAP</th></tr>
<tr><td rowspan="7">Traffic Group</td><td rowspan="2">Readout the Number of IP Service Peg Counts</td><td>Incoming Uncom-pleted Others (Reason: Others)</td><td>1.3.6.1.4.1.119.2.3.76.2.4.21.1.15.n0.n1.n2
Name for the MIB entry: ipSvPegCntInUncomplete4</td><td>x</td><td>-</td><td>-</td></tr>
<tr><td colspan="2">Readout the IP Service Traffic</td><td>1.3.6.1.4.1.119.2.3.76.2.4.22.1</td><td>-</td><td>-</td><td>-</td></tr>
<tr><td rowspan="4"></td><td>Node No. (1st Digit)</td><td>1.3.6.1.4.1.119.2.3.76.2.4.22.1.1.n0.n1.n2
(“GET” function is not available for this item)</td><td>x</td><td>-</td><td>-</td></tr>
<tr><td>Node No. (2nd Digit)</td><td>1.3.6.1.4.1.119.2.3.76.2.4.22.1.2.n0.n1.n2
(“GET” function is not available for this item)</td><td>x</td><td>-</td><td>-</td></tr>
<tr><td>Node No. (3rd Digit)</td><td>1.3.6.1.4.1.119.2.3.76.2.4.22.1.3.n0.n1.n2
(“GET” function is not available for this item)</td><td>x</td><td>-</td><td>-</td></tr>
<tr><td>Read the IP Service Traffic</td><td>Request
IP Service Traffic Order (crtTrafTable)
1.3.6.1.4.1.119 2.3.76.2.4.22.1.4.n0.n1.n2
Specify a Readout the Traffic (ipSvTrafData)
Node No. (3 digits) Note 2 (ipSvPegCntNode0~ipSvPegCnt-Node2)

Note:2 Digits 0~13 can be entered to the Node No. as follows.
<table><tr><td>Station No.</td><td>no data</td><td>1</td><td>2</td><td>3</td><td>4</td><td>5</td><td>6</td><td>7</td><td>8</td><td>9</td><td>0</td><td>*</td><td>#</td><td>P</td></tr><tr><td>Input value</td><td>0</td><td>1</td><td>2</td><td>3</td><td>4</td><td>5</td><td>6</td><td>7</td><td>8</td><td>9</td><td>10</td><td>11</td><td>12</td><td>13</td></tr></table>
Node number up to 3 digits with unused digits filled with “0”.
In case of Node No.=”01” as shown in the example below:
Enter “0”.
1.3.6.1.4.1.119 2.3.76.2.4.22.1.4. 10.1. 0
Node No.

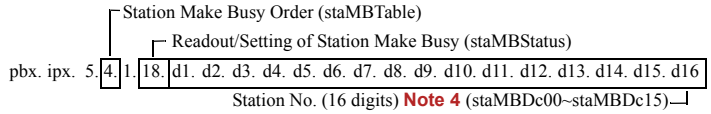
Response
Response data is coded as ASCII string.
<Example>
0.123 [Unit:Erlang]</td><td>x</td><td>-</td><td>-</td></tr>
</table>

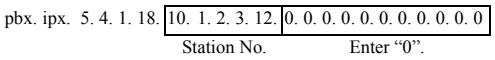
CONTENTS			OBJECT ID	GET	SET	TRAP
Traffic Group	Traffic Measurement Ready Notification		There is no specific operating instructions for this trap. However, it more than one of checking items specified by traffic measuring order becomes available to be checked, the object ID, “1.3.6.1.4.1.119.2.3.76.2.4.23.0”, is sent to SNMP Manager. The notification is indicated in integral number. Notified number “1” means, the measuring data is collectable.	-	-	X
	Entity of Traffic Measurement Ready Notification		1.3.6.1.4.1.119.2.3.76.2.4.24.0.0	-	-	-
Port Group			1.3.6.1.4.1.119.2.3.76.2.5	-	-	-
	Readout the status of Circuit Card		1.3.6.1.4.1.119.2.3.76.2.5.1.1	-	-	-
		Module No. of Circuit Card	1.3.6.1.4.1.119.2.3.76.2.5.1.1.1.x.y.z (“GET” function is not available for this item)	x	-	-
		Unit No. of Circuit Card	1.3.6.1.4.1.119.2.3.76.2.5.1.1.2.x.y.z (“GET” function is not available for this item)	x	-	-
		Group No. of Circuit Card	1.3.6.1.4.1.119.2.3.76.2.5.1.1.3.x.y.z (“GET” function is not available for this item)	x	-	-
		Readout of Firmware Infomation	**Request** Readout of Firmware Information (pkgStsFirmInformation) Port Group (port) 1.3.6.1.4.1.119 2.3.76. 2.5.1.1.4.x.y.z Circuit Card Status Order (pkgStsTable) Module No. of Circuit Card (pkgStsModule) Unit No. of Circuit Card (pkgStsUnit) Group No. of Circuit Card (pkgStsGroup) **Response** Response data is coded as ASCII string (a maximum of 32 characters). And nothing is displayed when there is no Firmware Information. <Example> SP-3295 V.07	x	-	-
		Readout of Circuit Card Type	**Request** Readout of Circuit Card type (pkgStsPkgType) Port Group (port) 1.3.6.1.4.1.119 2.3.76. 2.5.1.1.5.x.y.z Circui Card Status Order (pkgStsTable) Module No. of Circuit Card (pkgStsModule) Unit No. of Circuit Card (pkgStsUnit) Group No. of Circuit Card (pkgStsGroup) **Response** See 1. About the Circuit Card Status Readout on page 2346.	x	-	-

CONTENTS			OBJECT ID	GET	SET	TRAP
Port Group	Readout the status of Circuit Card	Readout of Circuit Card Status	**Request**  **Response** See 1. About the Circuit Card Status Readout on page 2346.	x	-	-
		Readout of SW/SP No.	**Request**  **Response** Response data is coded as ASCII string (Maximum 16 characters). **<Example>** 3295	x	-	-
		Readout of Issue	**Request**  **Response** Response data is coded as ASCII string (a maximum of 16 characters). **<Example>** 7	x	-	-
	Readout of Port Status		1.3.6.1.4.1.119.2.3.76.2.5.2.1	-	-	-
		Module No. of Port	1.3.6.1.4.1.119.2.3.76.2.5.2.1.1.w.x.y.z (“GET” function is not available for this item)	x	-	-
		Unit No. of Port	1.3.6.1.4.1.119.2.3.76.2.5.2.1.2.w.x.y.z (“GET” function is not available for this item)	x	-	-
		Group No. of Port	1.3.6.1.4.1.119.2.3.76.2.5.2.1.3.w.x.y.z (“GET” function is not available for this item)	x	-	-
		Level No. of Port	1.3.6.1.4.1.119.2.3.76.2.5.2.1.4.w.x.y.z (“GET” function is not available for this item)	x	-	-

CONTENTS			OBJECT ID	GET	SET	TRAP
Port Group	Readout of Port Status	Readout of Port Status	**Request**  **Response** See 2. About the Port Status Readout on page 2347.	x	-	-
		Readout of Port Information	**Request**  **Response** See 2. About the Port Status Readout on page 2347.	x	-	-
	Readout/Setting of Information for Trunk Make Busy (GET/SET)		1.3.6.1.4.1.119.2.3.76.2.5.3.1	-	-	-
		Route No.	1.3.6.1.4.1.119.2.3.76.2.5.3.1.1.x.y ("GET" function is not available for this item)	x	-	-
		Trunk No.	1.3.6.1.4.1.119.2.3.76.2.5.3.1.2.x.y ("GET" function is not available for this item)	x	-	-
		Readout/Setting of Information for Trunk Make Busy	**Request**  **<Input data for SET>** Only Integer can be entered to the Setting of Make Busy. 0=Idle1 1=Make Busy 2=Incoming/Outgoing Make Busy (only for Physical Route) **Response** **<Output data for GET>** Output data is an integer. 0=Idle1 1=Make Busy 2=Incoming/Outgoing Make Busy (only for Physical Route)	x	x	-

CONTENTS			OBJECT ID	GET	SET	TRAP
Port Group	Readout/Setting of Information for Trunk Make Busy (GET/SET)		1.3.6.1.4.1.119.2.3.76.2.5.4.1 **Note:** "pbx.ipx" indicates "1.3.6.1.4.1.119.2.3.76.2".	-	-	-
		Station No. [1st Digit]	pbx.ipx.5.4.1.1.d1.d2.d3.d4.d5.d6.d7.d8.d9.d10.d11.d12.d13.d14.d15.d16 ("GET" function is not available for this item)	x	-	-
		Station No. [2nd Digit]	pbx.ipx.5.4.1.2.d1.d2.d3.d4.d5.d6.d7.d8.d9.d10.d11.d12.d13.d14.d15.d16 ("GET" function is not available for this item)	x	-	-
		Station No. [3rd Digit]	pbx.ipx.5.4.1.3.d1.d2.d3.d4.d5.d6.d7.d8.d9.d10.d11.d12.d13.d14.d15.d16 ("GET" function is not available for this item)	x	-	-
		Station No. [4th Digit]	pbx.ipx.5.4.1.4.d1.d2.d3.d4.d5.d6.d7.d8.d9.d10.d11.d12.d13.d14.d15.d16 ("GET" function is not available for this item)	x	-	-
		Station No. [5th Digit]	pbx.ipx.5.4.1.5.d1.d2.d3.d4.d5.d6.d7.d8.d9.d10.d11.d12.d13.d14.d15.d16 ("GET" function is not available for this item)	x	-	-
		Station No. [6th Digit]	pbx.ipx.5.4.1.6.d1.d2.d3.d4.d5.d6.d7.d8.d9.d10.d11.d12.d13.d14.d15.d16 ("GET" function is not available for this item)	x	-	-
		Station No. [7th Digit]	pbx.ipx.5.4.1.7.d1.d2.d3.d4.d5.d6.d7.d8.d9.d10.d11.d12.d13.d14.d15.d16 ("GET" function is not available for this item)	x	-	-
		Station No. [8th Digit]	pbx.ipx.5.4.1.8.d1.d2.d3.d4.d5.d6.d7.d8.d9.d10.d11.d12.d13.d14.d15.d16 ("GET" function is not available for this item)	x	-	-
		Station No. [9th Digit]	pbx.ipx.5.4.1.9.d1.d2.d3.d4.d5.d6.d7.d8.d9.d10.d11.d12.d13.d14.d15.d16 ("GET" function is not available for this item)	x	-	-
		Station No. [10th Digit]	pbx.ipx.5.4.1.10.d1.d2.d3.d4.d5.d6.d7.d8.d9.d10.d11.d12.d13.d14.d15.d16 ("GET" function is not available for this item)	x	-	-
		Station No. [11th Digit]	pbx.ipx.5.4.1.11.d1.d2.d3.d4.d5.d6.d7.d8.d9.d10.d11.d12.d13.d14.d15.d16 ("GET" function is not available for this item)	x	-	-
		Station No. [12th Digit]	pbx.ipx.5.4.1.12.d1.d2.d3.d4.d5.d6.d7.d8.d9.d10.d11.d12.d13.d14.d15.d16 ("GET" function is not available for this item)	x	-	-
		Station No. [13th Digit]	pbx.ipx.5.4.1.13.d1.d2.d3.d4.d5.d6.d7.d8.d9.d10.d11.d12.d13.d14.d15.d16 ("GET" function is not available for this item)	x	-	-

<table>
<tr><th colspan="3">CONTENTS</th><th>OBJECT ID</th><th>GET</th><th>SET</th><th>TRAP</th></tr>
<tr><td rowspan="5">Port Group</td><td rowspan="4">Readout/ Setting of Information for Trunk Make Busy (GET/ SET)</td><td>Station No. [14th Digit]</td><td>pbx.ipx.5.4.1.14.d1.d2.d3.d4.d5.d6.d7.d8.d9.d10.d11.d12.d13.d14.d15.d16
(“GET” function is not available for this item)</td><td>x</td><td>-</td><td>-</td></tr>
<tr><td>Station No. [15th Digit]</td><td>pbx.ipx.5.4.1.15.d1.d2.d3.d4.d5.d6.d7.d8.d9.d10.d11.d12.d13.d14.d15.d16
(“GET” function is not available for this item)</td><td>x</td><td>-</td><td>-</td></tr>
<tr><td>Station No. [16th Digit]</td><td>pbx.ipx.5.4.1.16.d1.d2.d3.d4.d5.d6.d7.d8.d9.d10.d11.d12.d13.d14.d15.d16
(“GET” function is not available for this item)</td><td>x</td><td>-</td><td>-</td></tr>
<tr><td>Tenant No.</td><td>pbx.ipx.5.4.1.17.d1.d2.d3.d4.d5.d6.d7.d8.d9.d10.d11.d12.d13.d14.d15.d16</td><td>x</td><td>-</td><td>-</td></tr>
<tr><td>Readout/ Setting of Station Make Busy</td><td>Request
 

Note: “pbx.ipx” indicates “1.3.6.1.4.1.119.2.3.76.2”.
Note:4 Digits 0~12 can be entered to the Station No. as follows.
<table><tr><td>Station No.</td><td>no data</td><td>1</td><td>2</td><td>3</td><td>4</td><td>5</td><td>6</td><td>7</td><td>8</td><td>9</td><td>0</td><td>*</td><td>#</td></tr><tr><td>Input value</td><td>0</td><td>1</td><td>2</td><td>3</td><td>4</td><td>5</td><td>6</td><td>7</td><td>8</td><td>9</td><td>10</td><td>11</td><td>12</td></tr></table>Station number up to 16 digits with unused digits filled with “0”.
In case of Station No.=”0123#” as shown in the example below:
 

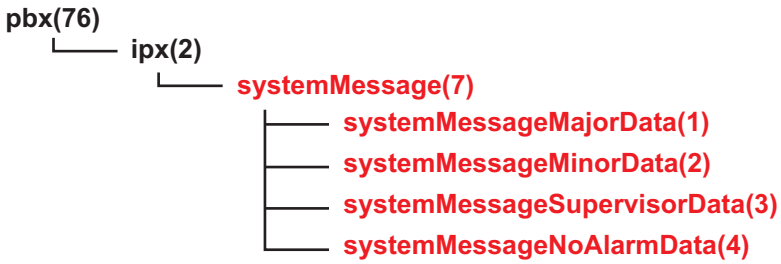
Response
<Output data for GET>
Output data is an integer.
0=Idle
1=Make Busy
2=Incoming/Outgoing Make Busy</td><td>x</td><td>x</td><td>-</td></tr>
</table>

CONTENTS		OBJECT ID	GET	SET	TRAP
Terminal Group		1.3.6.1.4.1.119.2.3.76.2.6	-	-	-
	RTP statistics Information Notification	There is no specific operating instructions for this trap. However, when RTP statistics Information Notification is requested, the object ID, "1.3.6.1.4.1.119.2.3.76.2.6.1.0", is sent to SNMP Manager. The notification is coded as ASCII string (a maximum of 127 characters).	-	-	x
	Entity of RTP statistics Information Notification	1.3.6.1.4.1.119.2.3.76.2.6.2.0.0	-	-	-
	Terminal Claim Handling Notification	There is no specific operating instructions for this trap. However, when Terminal Claim Handling Notification is executed, the object ID, "1.3.6.1.4.1.119.2.3.76.2.6.3.0", is sent to SNMP Manager. The notification is coded as ASCII string (a maximum of 127 characters).	-	-	x
	Entity of Terminal Claim Handling Notification	1.3.6.1.4.1.119.2.3.76.2.6.4.0.0	-	-	-

[ipx(2) group]

MIB tree and description of [systemMessage(7)]

System Message



	systemMessage(7)	
	systemMessageMajorData(1)	Trap of system message (major alarm)
	systemMessageMinorData(2)	Trap of system message (minor alarm)
	systemMessageSupervisorData(3)	Trap of system message (supervisory alarm)
	systemMessageNoAlarmData(4)	Trap of system message (no alarm)

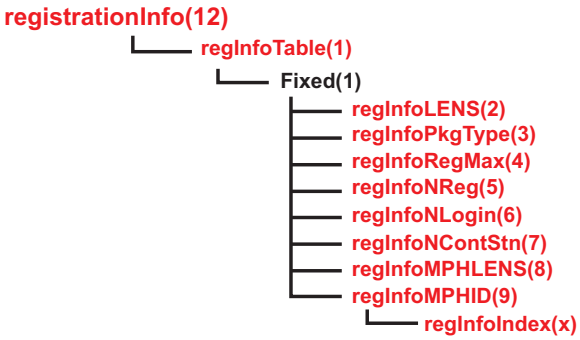
[About the Specific Number within the Trap Message.]

Please refer to for this information.

Note: “pbx.ipx” indicates the object ID “1.3.6.1.4.1.119.2.3.76.2”.

MIB tree and description of [registrationInfo(12)]

Registration information group



	registrationInfo(12)	Registration information group
	regInfoTable(1)	Registration information table
	regInfoLENS(2)	LENS readout
	regInfoPkgType(3)	Circuit card type readout
	regInfoRegMax(4)	Number of maximum registration readout
	regInfoNReg(5)	Number of current registration readout
	regInfoNLogin(6)	Number of current Login readout
	regInfoNContStn(7)	Specified number of control station readout
	regInfoMPHLENS(8)	MPH accommodated location readout
	regInfoMPHID(9)	MPH ID readout
	regInfoIndex(x)	Index number

[Request]: pbx.ipx.12.1.1.2.x

[Response]:
Output the LENS related to index number in ASCII strings. Value “NULL” will be output when no LENS is related to index.
<Output Example>:

02 . 0 . 12 . 0
MG(dot)U(dot)G(dot)L

[Request]: pbx.ipx.12.1.1.3.x

[Response]:
Circuit card type will be output in ASCII strings. Value “noSuchName” will be output when no LENS is related to index.
<Output Example>:

IPPAD
Internal PHI
Internal PHE

[Request]: pbx.ipx.12.1.1.4.x
[Response]:
Number of maximum registration will be output in integer.
Value “noSuchName” will be output when no LENS is related to index.

[Request]: pbx.ipx.12.1.1.5.x
[Response]:
Number of current registration will be output in integer.
Value “noSuchName” will be output when no LENS is related to index.

[Request]: pbx.ipx.12.1.1.6.x
[Response]:
Number of current Login will be output in integer.
Value “noSuchName” will be output when no LENS is related to index.

[Request]: pbx.ipx.12.1.1.7.x
[Response]:
Specified number of control station will be output in integer.
Value “noSuchName” will be output when no LENS is related to index.

[Request]: pbx.ipx.12.1.1.8.x
[Response]:
MPH accommodated location related to index number will be output in ASCII strings.
Value “noSuchName” will be output when no LENS is related to index.

[Request]: pbx.ipx.12.1.1.9.x
[Response]:
MPH ID will be output in integer.
Value “noSuchName” will be output when no LENS is related to index.

Note: “pbx.ipx” indicates the object ID “1.3.6.1.4.1.119.2.3.76.2”.

1. About the Circuit Card Status Readout

a.) Response Information of the Circuit Card Type Readout

The following circuit card type information is output in ASCII format (a maximum of 32 characters). When the circuit card type cannot be identified, "-"(hyphen) is displayed.

ASCII Character	Description
LC	Analog Line circuit card
TRK	Trunk circuit card
RS	Register/Sender circuit card
ATT	Attendant Console circuit card
DLC	DLC circuit card
PWR	7U-PIR Power Unit/8U-PIR Power circuit card
DCFT	DCFT circuit card
MFTL	MFTL (MLCB) circuit card
SMI	SMI circuit card
MDMT	MDMT circuit card
DLMX	DLMX circuit card

b.) Response Information of the Circuit Card Status Readout

The following circuit card status information is output in ASCII format (a maximum of 16 characters).

ASCII Character	Description
Idle	One or more than one port in the group is idle.
OffLine	No circuit card is mounted or circuit card is make-busy.
MB	Circuit card in the group is make-busy or soft make-busy.
Busy	All the ports in the group are in use.

2. About the Port Status Readout

a.) Response Information of the Port Status Readout

The following port status information is output in ASCII format (a maximum of 16 characters).

ASCII Character	Description
Idle	Idle
MB	Make-busy
Busy	In use.
LockOut	Locked out

b.) Response Information of the Port Data Readout

The following port information is output in ASCII format (a maximum of 256 characters). The output information varies according to route (internal/external), connection route, and station. The port information includes the multiple parameters, and each parameter is separated with a comma.

<External Trunk>

First Parameter: "Trunk"
Second Parameter: External Route Number (1-899)

Note: Displays the Logical route number assigned towards the outer office (external) route. Displays the Physical route when no Logical route is assigned.

Third Parameter: Trunk Number (1-255)
Fourth Parameter: Trunk/Signal Type

The following Trunk/Signal Type information is output in ASCII format.

Note: When the Trunk/Signal Type cannot be identified, "unknown" is displayed.

ASCII Character	Description
Loop Start COT	Loop COT
Ground Start COT	Ground Start (North America)
CDH COT	CDH (North America)
Caller ID COT	Call ID (Loop)
Loop Start Tie Line	Loop TIE
E&M Tie Line	E&M TIE
DX Tie Line	DX TIE
24 V4 Tie Line	24V TIE
Loop DID	Loop DID
NTT Loop DID	Central Office Line, NTT Loop DID
EMD (Iran)	EMD (Iran)
Russia	Russia Trunk
Digital Voice	Digital Trunk (Voice)
Digital Signal	Digital Trunk (Signal)
Nurse Call	Nurse Call Trunk
Toll	Terminal Toll
CAS	CAS Trunk
Paging	Paging Trunk
Radio Paging	Pager Paging
Paging Transfer	TIE Line Paging
Emergency Recording	Emergency Call

<Internal Trunk>

First Parameter: Internal Trunk Type

Note: Outputs the strings according to list of internal office route in ASCII format. When the Internal Trunk type cannot be identified, “unknown” is displayed.

Second Parameter: Route Number (901-947)
Third Parameter: Trunk Number (1-255)

The following Internal Trunk Type information is output in ASCII format.

ASCII Character	Description
Attendant Console	Attendant Console
Originating Register	Originating Register Trunk
Incoming Register	Incoming Register Trunk
MF Register	Register for MF
Sender	Sender Trunk DP/PB
PB Receiver	PB Receiver for Automated Attendant
Amp Pool	Amplifier Pool Trunk
Register/Sender	Sender/Register Trunk
Three-way Conference	Three-party Conference Trunk
Six-way Conference	Six-party Conference Trunk
Extended Internal Device	Other Station Device
Originating Register for ATTCON	Originating Register Trunk (Attendant Console)
Three-way Conference for ATTCON	Three-party Conference Trunk (Attendant Console)
Announcement	Announcement Trunk
Night Attendant Console	Night Attendant Console
MFC Receiver	MFC Receiver
MFC Sender	MFC Sender
Master Telephone	Master Telephone
Modem19	Modem19
Modem20	Modem20
Modem21	Modem21
Modem22	Modem22
Modem23	Modem23
Modem24	Modem24
Modem25	Modem25
Modem26	Modem26
CCS Sender	CCS Sender
CCS Register	CCS Register
Data Signaling	Data Signal
Rate Adaptation	Rate Adaptation
terminal of conference trunks	Conference Trunk Terminal

<Connection Trunk>

First Parameter: "C-Trunk"
Second Parameter: Connection Route Number (1-1023)
Third Parameter: Connection Trunk Number (1-4095)
Fourth Parameter: Trunk/Signal Type

Note: Outputs the strings according to list of Trunk/Signal kind in ASCII format. When the Trunk/Signal kind cannot be identified, "unknown" is displayed.

<Station>

First Parameter: Terminal Type
Second Parameter: User Group Number (1-255)
Third Parameter: Parameter Tenant Number (1-255)
Forth Parameter: Station Number [1-16 digits (0-9, *, #)]

Note: Outputs the strings according to list of Terminal kind in ASCII format. When the Terminal kind cannot be identified, “unknown” is displayed.

The following Terminal Type information is output in ASCII format.

ASCII Character	Description
DP10	Single Line Telephone DP (10pps)
PB	Single Line Telephone PB
DP/PB	Single Line Telephone DP/PB
DP20	Single Line Telephone DP (20pps)
DP10(Room)	Not Used
PB(Room)	Not Used
DP/PB(Room)	Not Used
DP20(Room)	Not Used
House Phone (Admin)	Not Used
House Phone (Guest)	Not Used
Voice Mail	Not Used
Digital Phone	Dterm (Dterm: Voice Port)
Digital Phone (Data)	Data Terminal (Dterm: Data Port)
Hotline	Hotline
CAS Line	CAS Line
Data Module	Data Terminal via Data Module
Bridge Phone	Not Used
Virtual	Virtual Line Appearance (for Dterm Multi-Line)
Text Mail	Text Mail
Packet Switching	Packet Switching
ACD	ACD
Emulator	Emulator
ISDN Terminal (TEI Auto)	ISDN Terminal
ISDN Terminal (TEI Fixed)	Not Used
VPS	Not Used
8-party Conference	Eight Conference Equipment
PHS	Not Used

<PH Kind>

First Parameter: PH Kind

Note: Outputs the strings according to list of PH kind in ASCII format. When the PH kind cannot be identified, “invalid PH” is displayed.

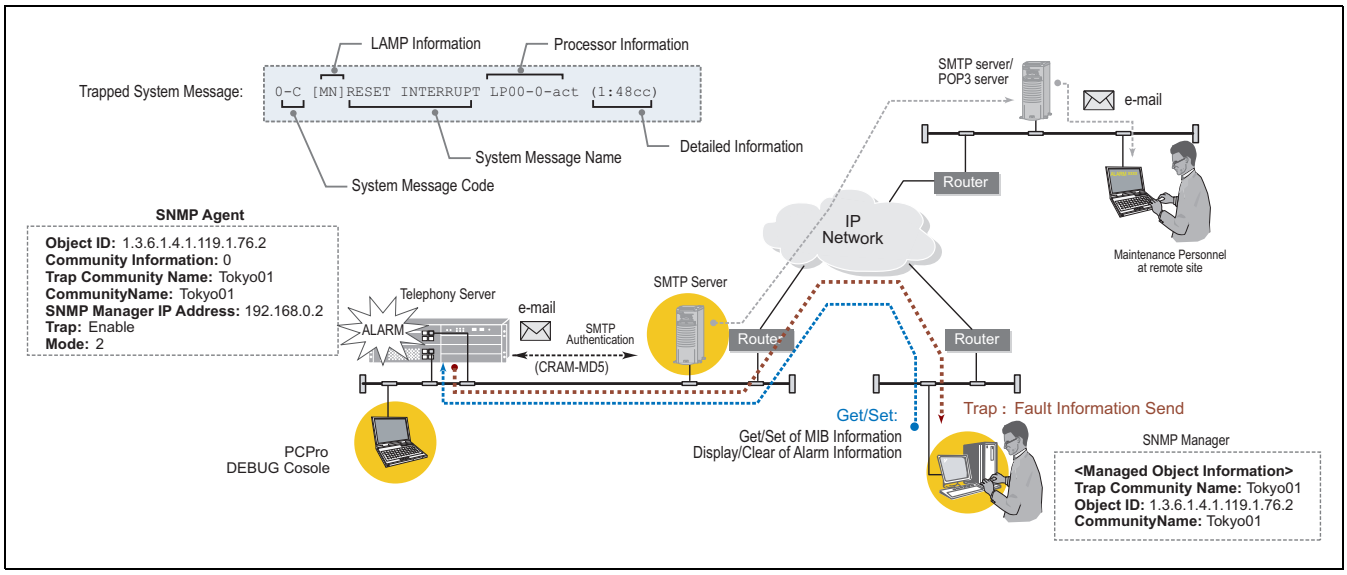
The following Terminal Type information is output in ASCII format.

ASCII Character	Description
IPPAD	IPPAD
Internal PHE	Internal PHE
Internal PHC	Internal PHC

S-150 System Message Report

General Description

This feature allows the Telephony Server to send system messages that are created in the system by an e-mail notification to the e-mail address specified in advance.
By using this feature, maintenance personnel can obtain fault information of the system from the remote site. SMTP authentication (by CRAM-MD5) is also supported for e-mail notification. If required, *message filtering* function can be used. This filter can select system messages to be output according to the alarm lamp grade. When not specified, all system messages will be output. In addition, system messages can be sent not only to SMTP Server but to PCPro and SNMP Manager as well. For more detailed information on System Message Report by SNMP manager, refer to [S-148] SNMP of this manual.



Operating Procedure

No manual operation is required.

Service Conditions

Note: Refer to the service conditions for [S-148] SNMP of this manual as well.

1. Buffer size for message output is 4 k bytes. Therefore a maximum of 4 k bytes messages (approximately, 100 messages) can be stored even if mail server fails. However, if the buffer overflow occurs, the oldest message will be deleted first.
2. DFTD command can read out both new and old messages.
3. As an SMTP authentication algorithm, Challenge/Response Authentication Mechanism-MD5 (CRAM-MD5) only is provided. Other authentication algorithm such as Digest-MD5, PLAIN are not supported.
4. When message filtering has been set, the following restriction will be placed for 6-A (System Failure) message: 6-A turns on MN lamp of the system, however it will be output as a Non-Alarm.
5. TCP port number of SMTP Server used for e-mail notification is set to 25. Assign the System Data to modify the port number for SMTP Server.

6. System messages cannot be output to System Message Printer when this service is enabled. When both services are enabled by the system data assignment, the data assignment for [S-150] System Message Report takes precedence.
7. It takes a maximum of 64 seconds for changes in the settings of System Message Report to take effect after clicking the SET button on the ADTM command window. Until then, the old settings are used to send system messages via e-mail.

Programming

STEP 1: **ASYDL** - Assign the following data:

[SMTP Authentication for E-mail]

When SMTP authentication is required, assign the following.

SYS 1, Index 963 Bit 0=1 (CRAM-MD5 Authentication is performed to communicate with mail server.)

[Message Filtering]

When using System Message Report Filtering function, set the following data. The default data output all system messages.

SYS 1, Index 964 Bit 0~5

System Message Report Filtering function for SMTP Server (e-mail):

Bit 1	Bit 0		
0	0	=	All messages will be output
0	1	=	Supervisory lamp messages or higher will be output
1	0	=	Minor lamp messages or higher will be output
1	1	=	Major lamp messages will be output

System Message Report Filtering function for SNMP Manager:

Bit 3	Bit 2		
0	0	=	All messages will be output
0	1	=	Supervisory lamp messages or higher will be output
1	0	=	Minor lamp messages or higher will be output
1	1	=	Major lamp messages will be output

System Message Report Filtering function for EVLOG command Console:

Bit 5	Bit 4		
0	0	=	All messages will be output
0	1	=	Supervisory lamp messages or higher will be output
1	0	=	Minor lamp messages or higher will be output
1	1	=	Major lamp messages will be output

[E-mail Notification to Message Submission Agent]

When sending system messages to an E-mail reception server (Message Submission Agent) with an authentication function, TCP port number for SMTP Server must be assigned with hexadecimal in the following system data.

SYS 1, Index 1309 (the upper bytes for TCP port number in HEX)
SYS 1, Index 1310 (the lower bytes for TCP port number in HEX)

<Example>

The data assignment shown below is an example for setting the port number as 587 (in decimal) which is recommended by RFC 2476.

SYS 1, Index 1309 = 02
SYS 1, Index 1310 = 4B

STEP 2: **ADTM - Assign data for TCP/IP module data.**

[SNMP Manager]

Assign the required data referring to Programming of SNMP [S-148] STEP 3.

[SMTP Server (e-mail)]

Select “LAN2” from “Select LAN Interface”.
Select “Options” tab → “System Message E-mail Output”.
Select “Mail Server/Address to Reply” from “Data to Assign” and assign the following data.

IP Address of Mail Server: Enter the IP address of the mail server.

Host Name of MGC: Enter the host name of MGC (A maximum of eight characters).

Address to Reply: Enter the mail address to replay (A maximum of 64 characters).

Maximum Text Size [0 - 1024]: Specify the maximum size of sending data.

Difference from GMT: Select the time difference from GMT in the combo box.

E-Mail Fault Message: Select this to send fault messages.

When SMTP Authentication is required, also assign the following:

User Name (A maximum of 15 characters)

Password (A maximum of 15 characters)

S-151 SIP Instant Message

General Description

This feature enables to exchange Instant Messages between a Soft Phone (SIP) and a WLAN Handset (MH Series).

Operating Procedure

Refer to User's Guide of each terminal.

Service Conditions

1. Instant Message can be sent/received between the following terminals:
   - Soft Phone (SIP)
   - WLAN Handset (MH Series)

   Note: Soft Phone (Proprietary Protocol) is not available for this feature.

2. A maximum number of characters that can be sent in an Instant Message is as shown below. For more information on other terminals, refer to the related manuals.

Soft Phone (SIP)	300 (single-byte alphanumeric)
WLAN Handset (MH Series)	300 (single-byte alphanumeric)

3. Instant Message can be sent/received via FCCS/CCIS network.
4. Instant Message cannot be sent/received via Central Office Line including MG(SIP).
5. Instant Message can be sent/received regardless the state of the terminal (idle or busy), but it depends on the operating condition of each terminal.
6. Instant Message cannot be sent to the terminal that has not performed REGISTRATION.
7. Sending/receiving of Instant Messages cannot be restricted by the office data assignment.
8. The following shows the conditions when using SOFT PHONE WLAN HANDSET [S-159]/SOFT PHONE WIRED-PHONE HANDSET [S-147]:

   - Instant Message can be sent to the Softphone by specifying the Station Number of the PS/wired telephone. **Note 1**

     Note 1: When using SOFT PHONE WLAN HANDSET [S-159]/SOFT PHONE WIRED-PHONE HANDSET [S-147], Instant Message cannot be sent by specifying the Station Number of the Softphone.

   - If Instant Message is sent from the Softphone to other SIP terminal, the Station Number of the PS/wired telephone is notified as the Calling Party Number.

9. The following shows the conditions for sending Instant Messages to both SIP terminal and Softphone when using WLAN Handset (MH Series) as the handset of Soft Phone (SIP):

- By assigning system data (ASYDL, SYS1, Index 691, Bit 2 = 1), Instant Message can be sent both Soft Phone (SIP) and WLAN Handset (MH Series). **Note 2**

  Note 2: When using WLAN Handset (MH Series) as the handset of Soft Phone (SIP), Instant Message cannot be sent by specifying the Station Number of the Soft-phone.

- When each of receiving station (WLAN Handset (MH Series) or Soft Phone (SIP)) can receive Instant Message, "Receive OK" message is displayed on originating station.

- If both of receiving stations (WLAN Handset (MH Series) and Soft Phone (SIP)) cannot receive Instant Message, the error reason will be displayed on the originating station. **Note 3**

  Note 3: There are some terminals which cannot display error reason.

- If Instant Message is sent from Soft Phone (SIP) to the other SIP terminals, the station number of WLAN Handset (MH Series) will be displayed as the originating station number. Therefore, when the other SIP terminal sends Instant Message, the destination will be WLAN Handset (MH Series). **Note 4**

  Note 4: By assigning system data (ASYDL, SYS1, Index 691, Bit 2 = 1), Instant Message will be sent to both of call originating stations (WLAN Handset (MH Series) and Soft Phone (SIP)).

10. The following shows the conditions when using DIAL PLAN TO SUPPORT IP CENTREX FUNCTIONALITY [D-166]:

    - When sending Instant Message to a Station in other UGN, specify "Private Number + Telephone Number" as the destination number.

    - When sending Instant Message to a Station in the same UGN, specify the Station Number or Telephone Number only as the destination number.

    - When sending Instant Message, "Private Number + Telephone Number" is notified as the Calling Party Number.

11. If Instant Message is sent to the Station that sets Call Forwarding - All Calls or Call Forwarding PS Incoming Call Incomplete, the Instant Message cannot be transferred. The Instant Message is sent to the specified Station only.
12. Even if Instant Message is sent to a Station within the Hunting Group, it can be sent to the specified Station only.
13. Even if Station sets DO NOT DISTURB [D-11], the Station cannot deny the incoming Instant Message. DO NOT DISTURB [D-11] is not available for Instant Message.
14. NUMBER SHARING [N-45] is not available for Instant Message.

    - To send Instant Massage to a main station, Station Number of the main station must be specified.

    - To send Instant Message to a sub-station, Station Number of the sub-station must be specified.

    - If Instant Message is sent from a sub-station, the other party's LCD displays not main station number, but the sub-station number.

15. The following shows the conditions when this feature is activated via CCIS link:
    a.) To exchange an Instant Message via CCIS network, the UDP Port Number "3456" and "5060" are used. Therefore, it is necessary to unblock these UDP ports over the network.
    b.) Even If an Instant Message is exchanged via the existing CCIS network (non-peer-to-peer connection), IP connection between each node is required.
    c.) When FCCS link and CCIS link coexist in a network, the following condition must be considered:
        - A maximum of 64 nodes can be included in the network.
        - When connecting multiple FCCS networks to a CCIS link, a unique FPC must be assigned to each node.
    d.) When using this feature in the CCIS Opened Numbering network, the following conditions must be considered:
        - CCIS route to a counter office must be selected for the primary route.
        - Instant Message destination number must include Access Code, Office Number and Station Number.
        - Do not assign Change Digit Code to CCIS route on the called side by using the ACDD command. Leave the data intact and use its result for analysis.
    e.) This feature uses User Datagram Protocol (UDP) communication to exchange an Instant Message via CCIS network (not CCIS message). Therefore, even if CCIS route is blocked, Instant Message can be sent/received via CCIS network.
    f.) By assigning the system data (ASYDL, SYS 1, Index 691, Bit 2 = 1), Instant Message can be sent both WLAN Handset (MH Series) and Soft Phone (SIP) in these conditions as follows.
        - Peer to Peer connection of CCIS network
        - Legacy connection of CCIS network
        - Peer to Peer connection of FCCS network

Programming

ASYDL - System Data 1, Index 691, Bit 2
Assign the destination of Instant Message.
0: WLAN Handset (MH Series) only.
1: WLAN Handset (MH Series) and Soft Phone (SIP).

When using this feature via CCIS link, the specific data assignment is required for any SIP terminals (Soft Phone (SIP) and WLAN Handset (MH Series)). For details on the necessary data, see Soft Phone [S-155].

Note: The specific data described in "Softphone Internodes IP Address Resolution over CCIS/FCCS" is not required when using this feature via FCCS link only.

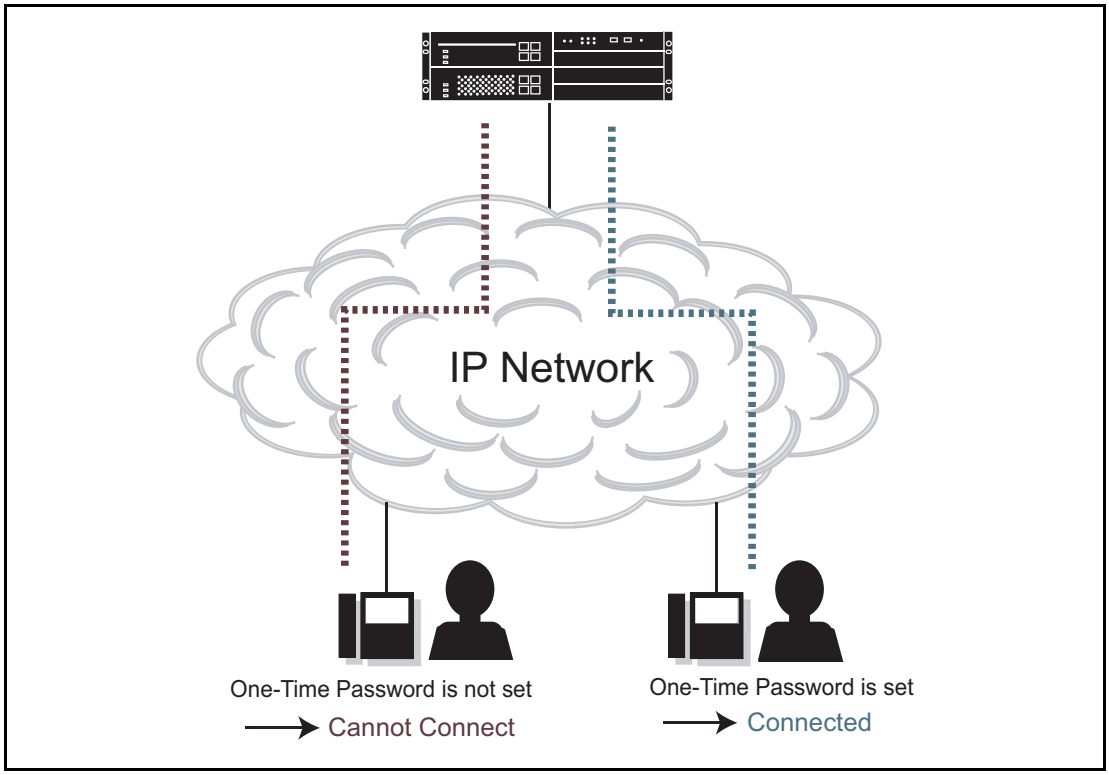
S-152 SIP Terminal Authentication

General Description

This feature provides SIP terminals/equipment with authentic method using one-time password in order not to allow malicious access. Because the one-time password is a disposable password that can be used only one time, using the password makes it difficult for unauthorized person to sneak into the network. A one-time password in the system becomes invalid when the available period is expired. For example, with terminals, the procedure for this feature proceeds with the system in the following steps:

- Assign a one-time password on the system.
- Assign a one-time password on the terminal.
- Start up the terminal, which transmits a one-time password.
- Terminal Authentication, which verifies the one-time password and the MAC Address information, is performed between the system and the terminal.
- If those procedures are completed successfully, the system allows the terminal to log on to the network.

If this feature is used with “VoIP Encryption” feature, and then VoIP Encryptions for RTP data and signaling data are initiated with Advanced Encryption Standard (AES) modes (128-bit length), it makes data more secured, preventing third party from tapping data flowing over the IP network. For more information about the SIP settings of Media Gateways, refer to the appropriate MG part of the Peripheral Equipment Description.



Operating Procedure

In the case of DtermIP (SIP)

1. Select “2.SIP Setting” → “7.Encryption” → “1.Authentication Mode”.
2. Select “2.Enable” and press the soft key below OK indication.
3. Select “2.SIP Setting” → “7.Encryption” → “2.One Time Password”.
4. Enter a one-time password and press the soft key below OK indication.
5. Back to main menu, and press the soft key below Save indication.
6. Select “2.SIP Setting” → “7.Encryption” → “3.Authentication Status”.
7. Make sure the status indicates Acceptable on the screen.

Service Conditions

1. After Terminal Authentication using a one-time password has been completed normally, back up the data of Data Memory by the MEM_HDD command using Data Type for One Touch Speed Calling for DtermIP.
2. When SR-MGC is used in the system, copy the office data into the SR-MGC (such as using the CMNT command) after backing up the data with the MEM_HDD command. For more information, refer to “Survivable Remote MGC [S-141]”. Thereby, SIP terminals that have completed SIP terminal Authentication become able to be rescued by the SR-MGC.
3. If the authentication data saved in an TP and a terminal are different, SIP Terminal Authentication needs to be performed again.
4. If you switch an TP to logged on, SIP Terminal Authentication has to be completed with the target TP before you switch it.
5. When specifying the Server IP Address for SIP terminals/equipment to connect to the network through SIP Terminal Authentication, use the IP address of the Telephony Server configured for the authentication.
6. To make one-time password valid, the system routinely verifies its data. While the verification is being performed, assigning or deleting Terminal Authentication cannot be executed. If the proceeding is failed because operations are executed during the verification, wait for a minute until the verification is completed.
7. When using SIP Terminal Authentication in the Telephony Server, accommodate PHDs/PHIs compatible with the feature into the system. Non-compatible PHDs/PHIs and compatible PHDs/PHIs cannot be mounted into the same system.
8. When this feature is performed, VoIP Encryption [V-27] is automatically enabled for security. In this case, if NAT is located between Terminals and SP, the encryption should be applied to a part of data. (ASYDL SYS1 Index 831 Bit 3=0)
9. When using SIP Terminal Authentication, use the terminal compatible with the feature.

10. When using MAC Authentication mode, SIP terminals perform SIP Terminal Authentication with SP; therefore, you have to specify a one-time password. For more information, see the following:

MAC Authentication Mode

	ASYDL INDEX 831 SYS1 Bit 4=1		ASYDL INDEX 831 SYS1 Bit 4=0	
	Terminal Registration Mode ON	**Terminal Registration Mode OFF**	**Terminal Registration Mode ON**	**Terminal Registration Mode OFF**
One-time password is matched between terminals and systems.	Registration is OK. →VoIP Encryption is enabled. **Note 1**	Login screen displays and asks a user to register in the system. →VoIP Encryption is disabled.	Registration is OK. →VoIP Encryption is enabled. **Note 1**	Registration is OK. →VoIP Encryption is disabled.
One-time password is Not matched between terminals and systems.	Registration NG Authentication is failed.	Login screen displays and asks a user to register in the system. →VoIP Encryption is disabled.	Registration NG Authentication is failed.	Registration is OK. →VoIP Encryption is disabled.
One-time password is Not assigned by ASECL.	Registration NG Register Timeout	Login screen displays and asks a user to register in the system. →VoIP Encryption is disabled.	Registration NG Register Timeout	Registration is OK. →VoIP Encryption is disabled.

Note 1: "Encryption is enabled" means when it is listed as Certified Terminal in the ASECL command.

11. When using Login mode, SIP terminals do not perform SIP Terminal Authentication with SP; therefore, you don't have to specify a one-time password. For more information, see the following:

Login Mode

	ASYDL INDEX 831 SYS1 Bit 4=1		ASYDL INDEX 831 SYS1 Bit 4=0	
	Terminal Registration Mode ON	**Terminal Registration Mode OFF**	**Terminal Registration Mode ON**	**Terminal Registration Mode OFF**
One-time password is matched between terminals and systems.	Registration is OK. →VoIP Encryption is enabled. **Note 2**	Login screen displays and asks a user to register in the system. →VoIP Encryption is disabled.	Registration is OK. →VoIP Encryption is enabled. **Note 2**	Registration is OK. →VoIP Encryption is disabled.

Login Mode

	ASYDL INDEX 831 SYS1 Bit 4=1		ASYDL INDEX 831 SYS1 Bit 4=0	
	Terminal Registration Mode ON	Terminal Registration Mode OFF	Terminal Registration Mode ON	Terminal Registration Mode OFF
One-time password is Not matched between terminals and systems.	Registration NG Authentication is failed.	Login screen displays and asks a user to register in the system. →VoIP Encryption is disabled.	Registration NG Authentication is failed.	Registration is OK. →VoIP Encryption is disabled.
One-time password is Not assigned by ASECL.	Registration NG Register Timeout	Login screen displays and asks a user to register in the system. →VoIP Encryption is disabled.	Registration NG Register Timeout	Registration is OK. →VoIP Encryption is disabled.

Note 2: "Encryption is enabled" means when it is listed as Certified Terminal in the ASECL command.

12. The following equipment are compatible with the Terminal Authentication.

AVAILABLE EQUIPMENT	AVAILABLE FIRMWARE VERSION
MG(BRI) Card [SCA-2BRIA] MG(BRI) Box [MG-2BRIA]	SP-3826 Issue 1.0 or later
MG(PRI) Card [SCA-24PRIA] MG(PRI) Box [MG-24PRIA] **Note 3**	SP-3884 Issue 5A or later
MG(PRI) Card [SCA-30PRIA] MG(PRI) Box [MG-30PRIA] **Note 3**	
UG50 (PRT 1.5M) [SIP]	Any version
UG50 (PRT 2M) [SIP]	
UG50-24PRIA [SIP]	
UG50-30PRIA [SIP]	
MG-COT [SCA-6COTB] **Note 3**	SP-3948 6MGCOT PROG-A Issue 1 or later
MG-COT [SCA-6COTC] **Note 3**	SP-3948 6MGCOT PROG-A Issue 1 or later
UG50 (COT-TYPE2) [SIP]	Any version
UG50-6COT [SIP]	

AVAILABLE EQUIPMENT	AVAILABLE FIRMWARE VERSION
UG50 (PGT-TYPE2) [SIP]	Software version 2.00 or later
MG-T1(SIP) Card [SCA-24DTIA]	SP-3926 MG DTI PROG-A Issue 1A or later
8LC Card [SCA-8LCA/SCA-8LCA-EMEA]	SP-3893 8MC PROG-A Issue 1 or later
8LC Card [SCA-8LCC/SCA-8LCC-EMEA]	SP-3983 8MC PROG-B Issue 1 or later
UG50 (LC) [SIP]	Any version
UG50-8LC	
DtermIP (SIP)	1.4.3.0 or later
DT700 Series	2.2.0.0 or later
DT800 Series	3.0.0.0 or later

Note: To save ASECL data to Flash Card, use MEM_HDD command. Select “One-Touch Speed Calling for Dterm IP”.

Note 3: This feature can be performed when the equipment described are in SIP mode.

13. When this feature is used, Terminal Remote Login [T-66] cannot be performed.
14. Refer to the UG50 Installation Guide for detailed conditions for UG50.

Programming

STEP 1: **ASYDL** - System Data 1, Index 831, Bit 3: Assign the data to determine whether NAT is used/NAT is used or not.

System Data 1, Index 831, Bit 4: 0/1 (Terminal Authentication is disabled/enabled)

Note: Regardless of this bit, the Terminal Authentication procedure by SIP terminal will be required if VoIP encryption [V-27] is enabled on the SIP terminal.

STEP 2: **ASECL** - Assignment of Security Data (LDM)

Assign one-time password and its available period. Click the Authentication Registration tab to assign the following data:

One-Time Password: Assign one-time password for authentication registration with 4 to 10 digits. (0-9, *, # can be used).

Available Period: Specify an available period for one-time password (indefinite/a minute to 7days).

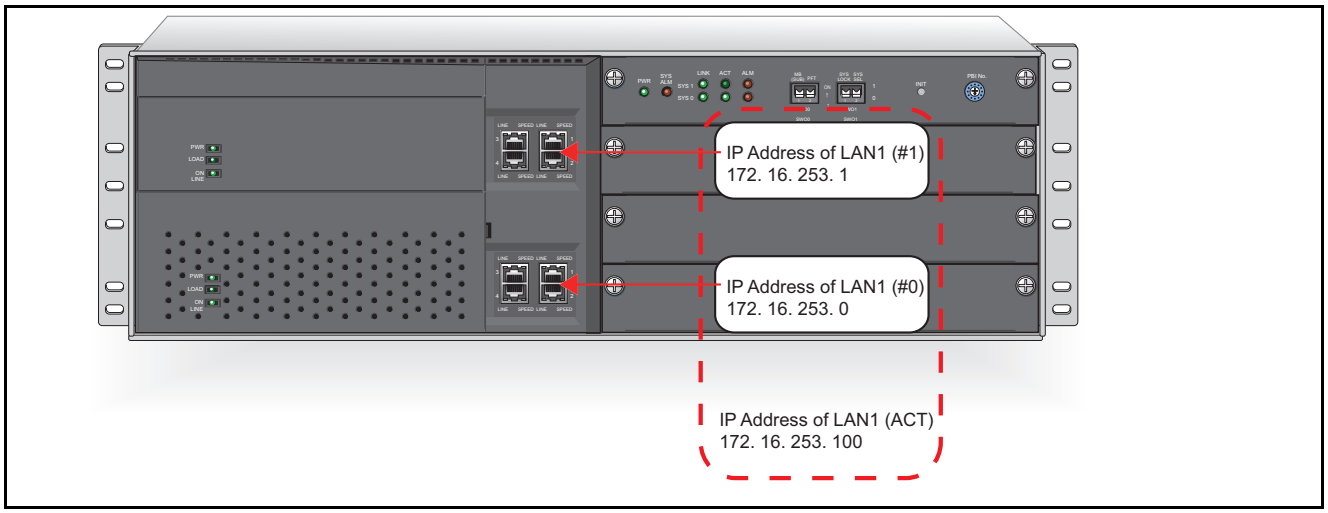
Note: Note the following conditions for a one-time password:

- A one-time password becomes invalid when an available period is expired or the system is initialized.
- For security purpose, you are recommended to specify an available period for the one-time password.
- When a one-time password and its available period are set, backup them by selecting “One Touch Speed Calling for DtermIP” in the Data Type Select parameter of the MEM_HDD/MEM_HDD_N command.

S-154 SIP Wireless Terminal Backup

General Description

When CPU card is set in redundancy configuration, the IP address of both LAN1 (#0) and LAN1 (#1) can be integrated. By specifying the IP address of LAN1 (ACT) as the IP address of REGISTER destination from each SIP terminals, in normal operation, LAN1 (#0) controls the SIP terminals. If a failure occurs in the LAN1 (#0), SIP terminals can originate or receive a call by using LAN1 (#1). Therefore, it increases the reliability of communication.



Operating Procedure

No manual operation is required.

Service Conditions

1. This feature is available under the conditions of each SIP Terminal in addition to the following conditions. For details on the conditions of each SIP Terminal, refer to SP CONTROLLED SIP TERMINAL [S-168].
2. This feature supports the following terminals:
   - WLAN Handset (MH Series)
   - SP Controlled Standard SIP terminal
3. Be sure to assign the pair SIP server ID assigned for the SIP redundancy configuration, by using ASSDL/ ASSDN command.
4. When using this feature, the IP address of LAN1 (ACT) must be specified as REGISTER destination IP address from each terminal. Do not use the IP address of LAN1 (#0)/LAN1 (#1).
5. The IP address of LAN1 (ACT) can be shared between two LAN1s only [LAN1 (#0)/LAN1 (#1)].
6. LAN1 (#0)/LAN1 (#1)/LAN1 (ACT) need to be placed in the same segment (broadcast domain).
7. LAN1 (STBY) does not accept the SIP Request from each SIP Terminals.

8. All REGISTER information on LAN1 (#0) and LAN1 (#1) is copied from LAN1 (ACT) to LAN1 (STBY). All REGISTER information will be held after the system changeover.
   a.) When LAN1 (ACT) is shifted to Standby mode, all REGISTER information of new LAN1 (ACT) is copied to new LAN1 (STBY).
   b.) When system changeover occurs during the copying process, REGISTER information may not be copied properly (REGISTER information copying will be completed within about 2 minutes).
   c.) When a failure occurs in both LAN1 (ACT) and LAN1 (STBY), REGISTER information will be cleared. If the information is cleared, each SIP Terminals must REGISTER again.
9. The same data must be assigned on the PRESENCE SERVER tab of ASSDL/ASSDN command for both LAN1 (#0) and LAN1 (#1). If the different PRESENCE SERVER data is assigned between the LAN1s, the operation of the Presence server after the system changeover is not ensured.
10. The same domain cannot be set to the both LAN1s. However, the domain set to either of LAN1 (#0) or LAN1 (#1) affects both of them.
11. Addition/Deletion of the domain data on ASSDL/ASSDN command affects both LAN1 (#0) and LAN1 (#1).
12. The same number of speech channels must be set to LAN1 (#0) and LAN1 (#1).
13. If a failure occurs in the LAN1 (ACT), the LAN1 (STBY) starts operating in Active mode within 60 seconds.
14. If “Pair SIP Server ID” assigned on ASSDL/ASSDN command differs from “The other side SIP server ID”, “501 Not Implemented” is sent to the SIP terminal as a reply to the REGISTER request.
15. “Pair SIP Server ID” on ASSDL/ASSDN command cannot be changed during operation.
16. The following explains how to change “Pair SIP Server ID” on ASSDL/ASSDN command and the information of the redundant partner.
    (1) Select “SIP Server” in “Function” box by the ASPC command, and press Get button.
    (2) Blank “SIP Server ID” box and press Set button.
    (3) Change “Pair SIP Server ID” on ASSDL/ASSDN command.
    (4) Set your SIP server ID to “SIP Server ID” on the ASPC command.
17. Both LAN1 (#0) and LAN1 (#1) require the domain setting on ASSDL/ASSDN. If all actual domains are assigned for one LAN1, at least one dummy domain must be assigned for the other LAN1.
18. The same DETAIL data on BASE DATA tab of ASSDL/ASSDN command must be assigned in LAN1 (#0) and LAN1 (#1).
19. When SIP terminal data is changed or deleted and either of LAN1 [LAN1 (#0) or LAN1 (#1)] does not respond it, a warning message is displayed.
20. When the system changeover occurs, new LAN1 (ACT) sends “Gratuitous APR Packet” message to SIP terminals. Some SIP terminals cannot respond to this message immediately. These SIP terminals are required to be reset.
21. Manual system changeover can be executed by using the CMOD command. (“SP mode Change” on the COSSL/COSSN command is unavailable.)
22. Even if a failure occurs in the LAN1 (ACT) and the system changeover is executed, the calls in progress are maintained, except in the following cases: **Note 1**, **Note 2**

    - SIP Terminal releases the call.
    - Button operation such as Switch Hook Flash is performed.
    - The party on the other end releases the call.

- Call Holding Timer expires.

Note 1: Only when a call is in 2-party talking, the call will be held.

Note 2: The conversation between the SIP terminals accommodated in the same LAN1 is maintained even if the timer expires or the timer is disabled.

23. If the called party is using a terminal not available for this feature, the call will not be maintained if a failure occurs and the system changeover is executed again.
24. If a failure occurs in the LAN1 (ACT) while a SIP Terminal is engaged in a call and the SIP Terminal hands over after the system changeover is accomplished, the call is disconnected.

Programming

STEP 1: Office Data Setting (Setting in the LAN1)

(1) **ASDIL/ASDIN** - SIP Domain Name Index Data

Assign the domain name to be used for SIP service. For details, refer to Initial data setting for WLAN Handset (MH Series) and SP Controlled SIP terminal in Chapter 5 of the Appliance Model Installation Manual.

(2) **ASSDL/ASSDN** - SIP Server Operation Data

Assign SIP Server ID of the redundant partner LAN1. For details, refer to Initial data setting for WLAN Handset (MH Series) and SP Controlled SIP terminal in the Appliance Model Installation Manual.

<REDUNDANCY tab>
Pair SIP Server ID: Assign SIP Server ID of the redundant partner LAN1 (1-127).

(3) **ASYDL/ASYDN** - System Data

System Data1, Index701, Bit 0~5: SIP telephone call holding timer **Note 3**
01 Hex~3d Hex (Setting value x 9 minutes)
00 Hex (Default) = 63 minutes
Bit 6~7: SIP telephone call holding canceling
01 Hex = Invalid the call holding
00 Hex (Default) = Valid the call holding

Note 3: Timer range: Setting time + 9 minutes

STEP 2: Setting on Each Terminal

When using this feature, set the IP address of LAN1 (ACT) as REGISTER destination IP address from each terminal. For details on setting of REGISTER destination IP address, refer to the related User's Guide.

S-155 Soft Phone

General Description

A Softphone is an application enables the user's personal computer to be used as a multi function telephone set. This section provides information on following service features using a Softphone. For details on a Softphone itself, refer to the User Guide.

- SOFT PHONE WLAN HANDSET [S-159]/SOFT PHONE - PS [S-158]

  This feature allows Softphone users to use the PS/WLAN Handset (MH Series) as the handset of the Softphone. For details on this feature, SOFT PHONE WLAN HANDSET [S-159]/SOFT PHONE - PS [S-158].

- SOFT PHONE WIRED-PHONE HANDSET [S-147]

  This feature allows Softphone users to use a wired telephone (analog wired telephone/Desktop terminal) as a handset of a Softphone. For details on this feature, SOFT PHONE WIRED-PHONE HANDSET [S-147].

- Softphone Password Change

  This feature allows Softphone users to change the Login Password in a Softphone screen.

- Softphone Internodes IP Address Resolution over CCIS/FCCS

  A Softphone can obtain an IP address of a Softphone accommodated in another FCCS/CCIS network so that it can send an Instant Message and perform the application sharing function such as Video Conference.

- Softphone - Other Party Information Readout

  This feature allows the system to send the information about the connected party (IP address and telephone number) as a response to inquiry from the Softphone. By using this feature, Softphone can receive the information of the following connected parties.

  Party using Sub Line

  Party engaged in a Three-party Conference

  Party engaged in a 8/16/32-party Conference

Operating Procedure

<Softphone Password Change>

1. Register to the system.
2. Start up the Password Change tool, and then enter the current password and new password. For more detailed information, see the User Guide.

<Softphone Inter-node IP Address Resolution over CCIS/FCCS>

To send an Instant Message

- In the case of FCCS or CCIS closed numbering network

  Enter a Station Number on the Short Message Destination, which is described as "To:". When the IP address of the destination Softphone is retrieved, an Instant Message is transmitted.

- In the case of CCIS opened numbering network

  Enter "Access Code" + "Office Number" + "Station Number" on the Short Message Destination, which is described as "To:". When the IP address of the destination Softphone is retrieved, an Instant Message is transmitted.

To perform the application sharing function

While keeping connection alive on the Softphone or the terminal in conjunction with the Softphone. Click the Application Sharing button.

<Softphone - Other Party Information Readout>

None.

Service Conditions

<Common Conditions>

1. Be sure to assign the following key data to a Programmable Feature key/Flex key of a Softphone.
   - SPEAKER key (When the Flex key data has been changed)
   - LOGOUT key
2. When using Softphone Wireless Handset/Softphone Wired-Phone Handset, the line of the terminal to be used as a handset of Softphone must be assigned as Prime Line. When using ACD service, ACD line must be used as Prime Line. At all other times, My Line must be assigned as Prime Line.
3. When using a Softphone, the following data needs to be assigned.
   a.) Before using a Softphone, be sure to assign the following data by ADKS command.

   LKP=3 (ALL Line/Feature buttons)

   b.) Before using Name Display service, be sure to assign the following system data.

   ASYD, System Data 1, Index 241, Bit 1 = 1 (The Number of digits displayed in Name Display Service is 16.)

c.) To use the Call Forward Settings Panel of a Softphone, the following data needs to be assigned by AKYD command.

FKY 1 (Call Forwarding - Busy Line), 2 (Call Forwarding - All Calls), 22 (Call Forwarding - Don't Answer) and 148 (Call Forwarding - Logout)

4. Distinctive Ringing [D-10] is not available for Softphones. Softphones support only the standard ringing pattern.
5. When using a Softphone as an ACD terminal:
   a.) The RELEASE key must be assigned to a programmable Feature key/Flex key of Softphone.
   b.) To get Calling Party Number of ACD incoming call, the received ANI Number (Calling Party Number) needs to be displayed at the beginning of the Connection Displays - ACD [C-199A].

Note: Soft Client SP350 in R3 or earlier cannot be used as an ACD terminal.

<Softphone Password Change>

1. This feature is available for SP30 with Software Version 13.0.0.0 or later.
2. Enter Station Number as password for first time login.
3. The number of significant digits for the password is 4 through 10. The password can includes "0-9", "*" and "#". Also, it is necessary to make the number of digits for the password coincide with MND assigned on AMND/AMNDL/AMNDN command.
4. Login Password can be changed when a Softphone has registered to the System.
5. Registered Login Passwords are stored in the memory of the system.
6. Login Password cannot be deleted from a Softphone. The RDPD command is used to delete the registered password.
7. The following shows the conditions concerning synchronization of Login Password between the nodes:

   In the case of FCCS Cluster: Login Passwords are synchronized when the office data is updated.

   In the case of SR-MGC: Login Passwords are synchronized when the office data is copied after the periodical diagnostic.

<Softphone Inter-node IP Address Resolution over CCIS/FCCS>

1. This feature uses a 3456 port (UDP). Therefore, it is necessary to unblock the port over the network.
2. In DtermSP30, self-node's office number in Numbering Plan must be correctly assigned by using DtermSP30Config.
3. This feature can be performed other than peer-to-peer connection of FCCS/CCIS network. However, its network must include IP connection for the feature.
4. This feature requires the following firmware:
   - DtermSP30: Version 9 or later
5. On the condition of opened numbering network set up in Common Channel Signaling system, the restrictions are as follows:
   a.) CCIS route to the destination office must be programmed as the first choice of the AOPR/AOPRL/AOPRN table.
   b.) The called party number transmitting to the destination office must include all digits dialed (an Access Code, an Office Number, and a Station Number).
   c.) A uniform numbering must be assigned to the destination office.
   d.) For originating office, a uniform numbering including the self-node identification code for CCIS

(RT=0 in the ARNP command) must be assigned correctly.

e.) Change Digit Code to CCIS route on the called party (the ACDD command) cannot be programmed. Leave the data intact and use the result for analysis.

f.) Communication with CCIS message is not used for this feature, however, a UDP (Port Number:3456) connection between MGCs is needed for this feature to perform.

<Softphone - Other Party Information Readout>

1. This feature is available when ASYDL/ASYDN, System Data 1, Index 863, Bit 2 = 1.
2. This feature supports only the connected parties engaged in a Two-party Call or a Conference Call.
3. This feature is available through FCCS or CCIS network.
4. This feature is not available in the UMGi system.
5. The conditions concerning to 8/16/32-party Conference are shown below.
   a.) When using CFT card, this feature supports 8-Party Conference only. When using VS32 Conference Server, this feature supports all the conferences (8/16/32-Party Conference).
   b.) When 8/16/32-party Conference (MEET-ME) includes a conferee transferred by hooking operation through 3-party Conference:
      - the parties on the 8/16/32-party Conference cannot receive the information about the party connected with the transferred party of the 3-party Conference.
      - intermediate party and transferred party of 3-party Conference can only receive the information about the party connected with the member of the 3-party Conference.
   c.) When performing 8-Party Conference with CFT card in the FCCS network, assign "SFI 72 = 1" (ASFC/ASFCN command) at all the connected nodes. **Note 1**

      Note 1: It is recommended to use ASFCN command.

6. To activate this feature, firmware 15.0.0.0 or later version is required to be equipped on DtermSP30.

<Conditions Specific to UMGi/RNIP System>

1. In a UMGi system, this feature cannot be used in a RN in survival mode that has soft phones registered.
2. In a RNIP/UMGi system, this feature cannot be used when a failure occurs in the CCN.
3. In UMG system, this feature cannot be used when a failure occurs in CMG.

Programming

For details on the Station data setting of a Softphone, see “IP Station Data” in Chapter 2 of this manual.

<Softphone Password Change>

- Assignment of Login Password Data

STEP 1: **ASYD** - System Data 1, Index 93, Bit 6 = 0 (Separate tables)
STEP 2: **ASFC/ASFCN** - Assign a Service Feature Restriction Class that allows SFI 186, to the Stations used for “DtermIP Login function”. Data must be “0”.
STEP 3: **AMND/AMNDL/AMNDN** - Assign the MND data for the passwords.
TN: 0
DC: Destination Code (Assign the 1st digit (“0-9”, “#”, “*”) of all passwords.)
MND: Maximum Necessary Digits (Assign the MND of passwords.)
TOLL: Local Code
AN: Out of service
RATE: 0
A/D: Analog

- Deletion of Login Password

STEP 1: **RDPD** - Delete the registered password if necessary.
TYPE: Choose either STN or TELN.

- When TYPE = STN:

TN: Tenant Number
STN: Station Number (Physical Number)

- When TYPE = TELN:

UGN: User Group Number
TELN: Telephone Number

- Back up Login Password to the hard disk

STEP 2: **MEM_HDD** - Back up the passwords to the hard disk.
Direction Select: Choose “Memory to Hard Disk”.
Data Type Select: Choose “Call Forwarding.”
LP Select: Specify the LP Number.

<Softphone Inter-node IP Address Resolution over CCIS/FCCS>

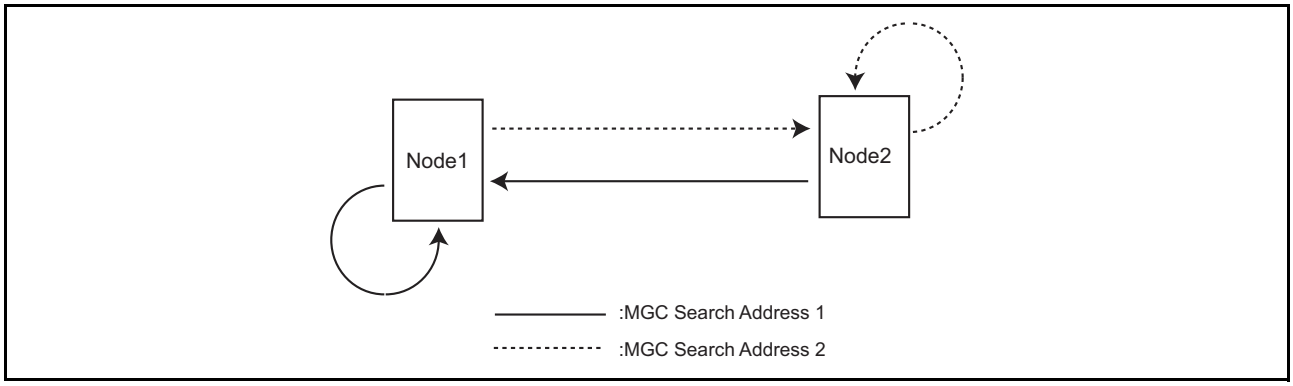
STEP 1: **ADTM** - Assign this MGC Search Address. **Note 2**, **Note 3**

Select the Option tab and the Registration Service, and then click the Execute button. On the screen, Enter MGC Search Address 1 and MGC Search Address 2. These items are assigned in the following:

Note 2: Assign the same IP address (ACT) that has been assigned in the “REG” parameter of “Configure Application Port” under “Options” tab in the same node.

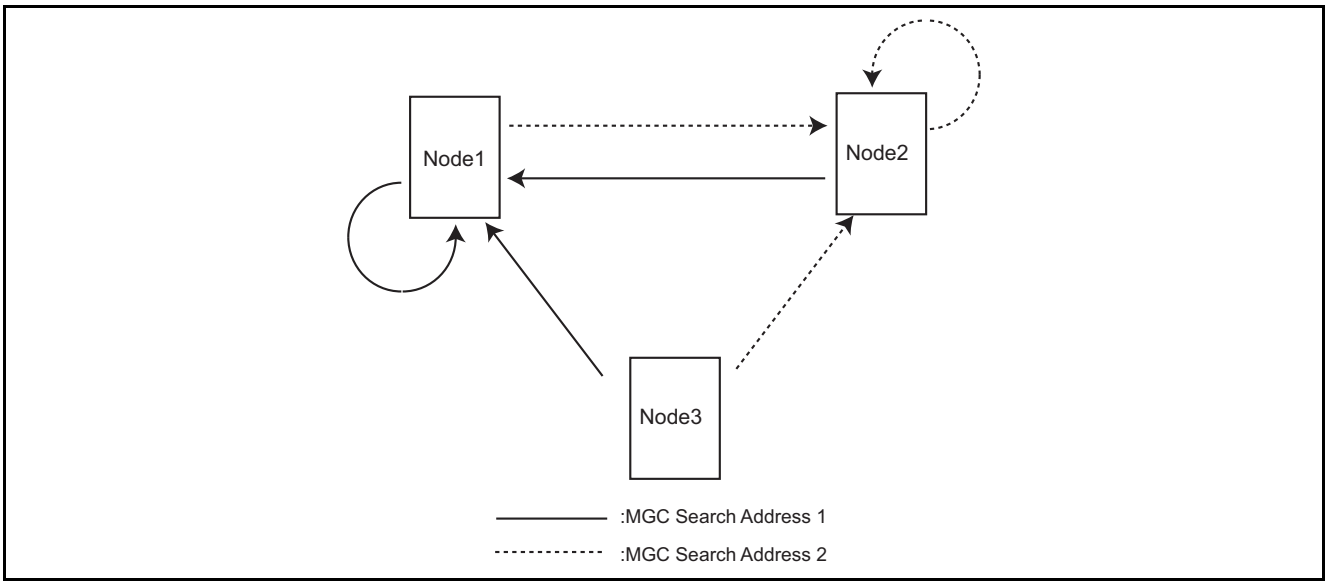
Note 3: Either the IP address of LAN1 (ACT) or LAN2 (ACT) can be chosen as the MGC Search Address. Note, however, that LAN2 is used when the existing IPX/SV7000 network is migrated to the SV9500 network. For more details about migration, refer to “Migration Guide”. In the following examples, LAN1 is used.

- Two nodes constitute a network



NODE	MGC SERCH ADDRESS1	MGC SERCH ADDRESS2
Node 1	IP Address of Node 1 (self node) (IP Address of LAN1 (ACT))	IP Address of Node 2 (IP Address of LAN1 (ACT))
Node 2	IP Address of Node 1 (IP Address of LAN1 (ACT))	IP Address of Node 2 (self node) (IP Address of LAN1 (ACT))

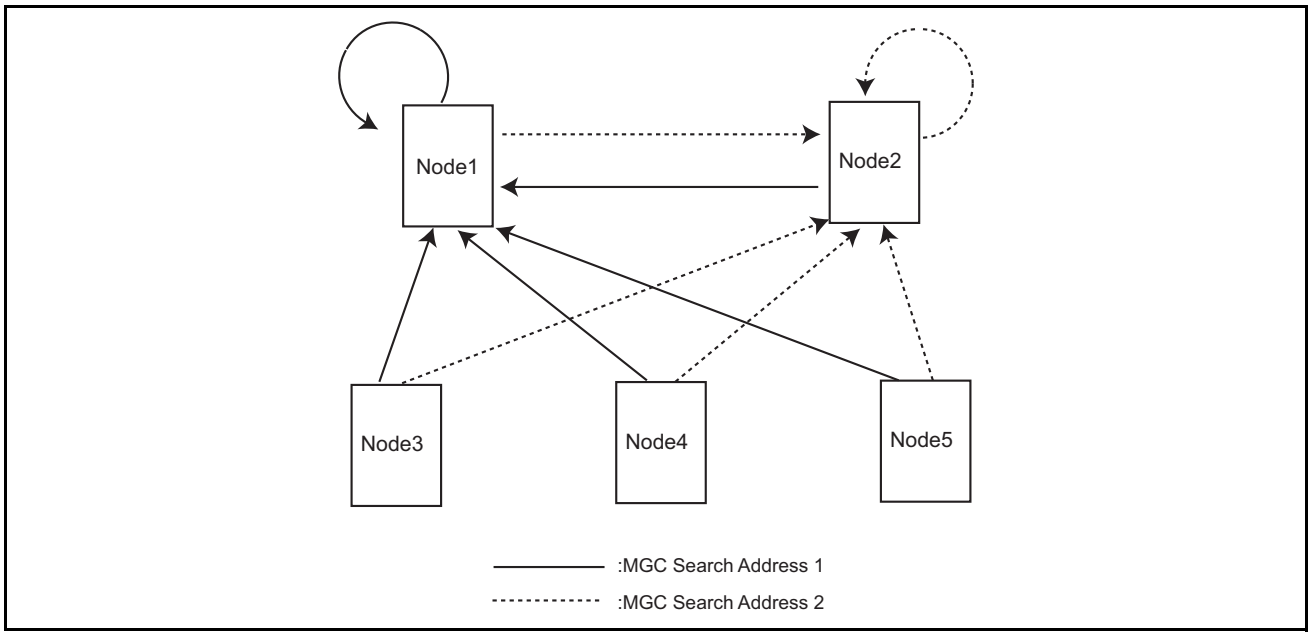
- Three nodes constitute a network



NODE	MGC SERCH ADDRESS1	MGC SERCH ADDRESS2
Node 1	IP Address of Node 1 (self node) (IP Address of LAN1 (ACT))	IP Address of Node 2 (IP Address of LAN1 (ACT))
Node 2	IP Address of Node 1 (IP Address of LAN1 (ACT))	IP Address of Node 2 (self node) (IP Address of LAN1 (ACT))
Node 3	IP Address of Node 1 (IP Address of LAN1 (ACT))	IP Address of Node 2 (IP Address of LAN1 (ACT))

- More than three nodes constitute a network

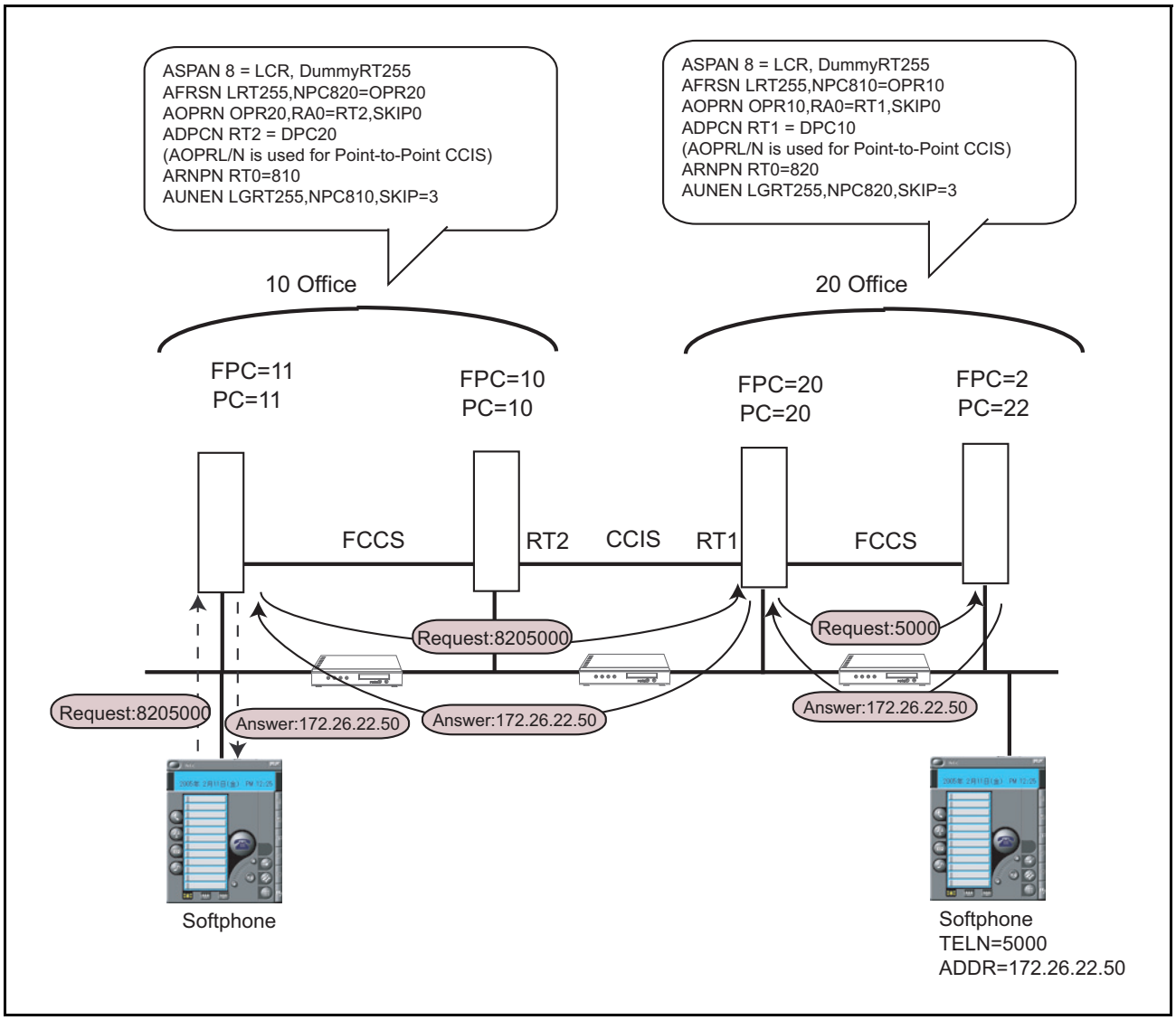
In the case that more than three nodes constitute a network, the same pattern can be applied.



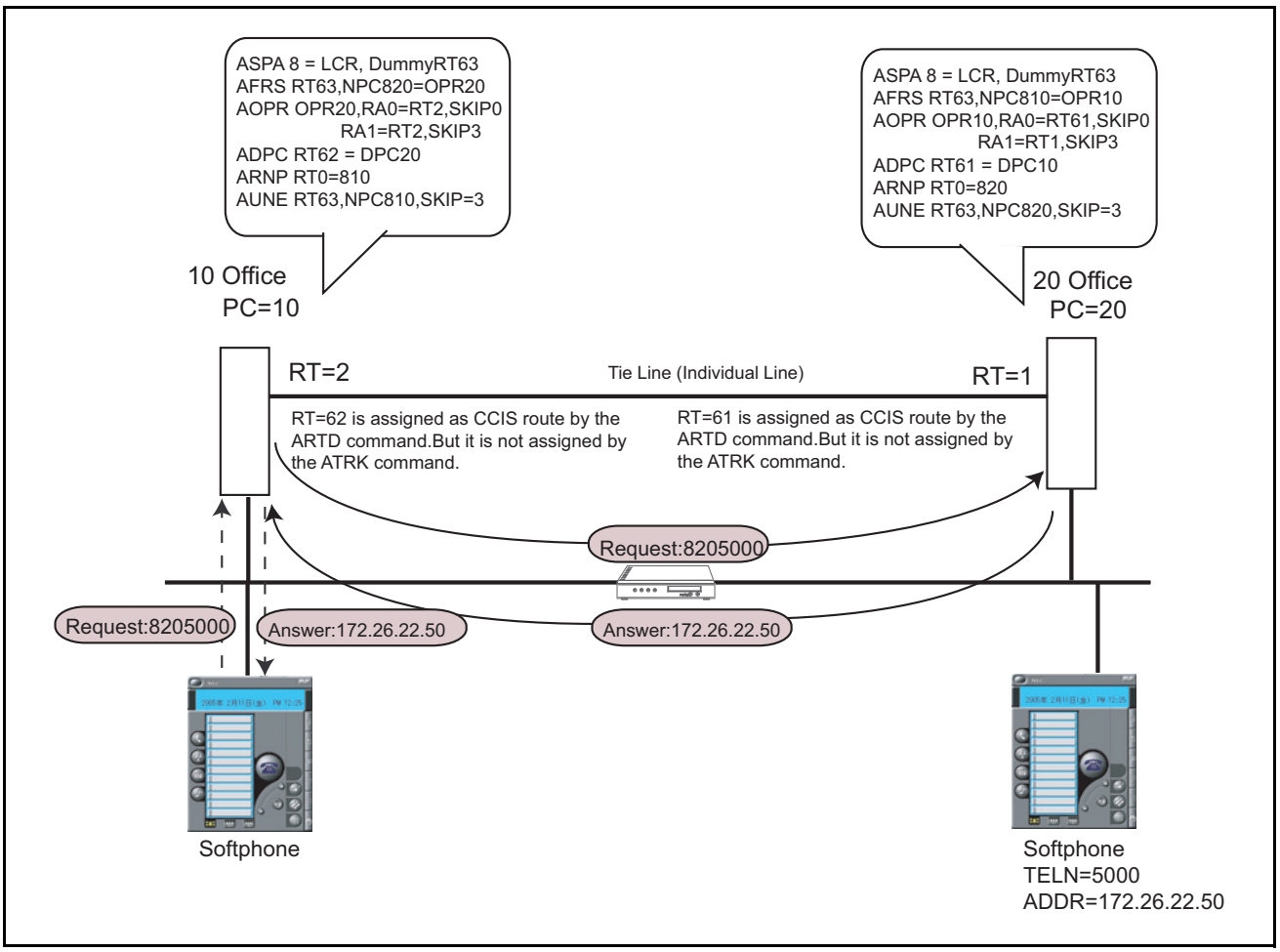
NODE	MGC SERCH ADDRESS1	MGC SERCH ADDRESS2
Node 1	IP Address of Node 1 (self node) (IP Address of LAN1 (ACT))	IP Address of Node 2 (IP Address of LAN1 (ACT))
Node 2	IP Address of Node 1 (IP Address of LAN1 (ACT))	IP Address of Node 2 (self node) (IP Address of LAN1 (ACT))
Node 3	IP Address of Node 1 (IP Address of LAN1 (ACT))	IP Address of Node 2 (IP Address of LAN1 (ACT))
Node 4	IP Address of Node 1 (IP Address of LAN1 (ACT))	IP Address of Node 2 (IP Address of LAN1 (ACT))
Node 5	IP Address of Node 1 (IP Address of LAN1 (ACT))	IP Address of Node 2 (IP Address of LAN1 (ACT))

The following is examples of the network configuration using this feature.

- When FCCS and CCIS networks coexist



- When Tie line using individual line is used



<Softphone - Other Party Information Readout>

STEP 1: **ASYDL/ASYDN** - System Data 1, Index 863, Bit 2. Softphone - Other Party Information Readout. 0/1 = Out of service/In service

STEP 2: **ASFC/ASFCN** - When performing 8-Party Conference with CFT card in the FCCS network, the data setting shown below is required at all the nodes. In this case, it is recommended to assign the data by ASFCN command.

SFI = 72 (Called Station Switch Hook Flash Restriction)

RES = 1 (Allowed)

S-157 Selective Disconnect 8 Party Conference

General Description

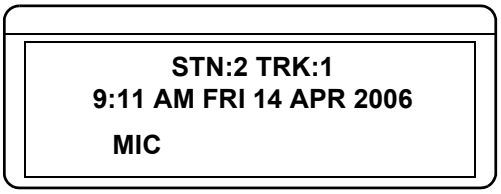
This feature will allow a conference attendee (or leader) to use a Feature key to display each party in the conference. Once the party is displayed the user can press the DISC key to drop the party or the CONFERENCE DISPLAY Feature key to move to the next party.
This feature functions with the following conference types:
ADD-ON CONFERENCE 8 PARTY [A-121]
CALL HOLD CONFERENCE 8 PARTY
ASSISTED CONFERENCE 8 PARTY [A-143]

This feature works in conjunction with the CONSULTATION HOLD RELEASE [C-169] function, the last party displayed using the CONFERENCE DISPLAY Feature key becomes the party to be dropped by the DISC key.

Note: This feature is available in North America only.

Operating Procedure

During a normal 8 party Add-on Conference the display of the Desktop terminal will show the user the number of station and trunks currently connected in the conference.
Example:



When the CONFERENCE DISPLAY Feature key is pressed the LCD will display a conference attendee. Each subsequent press of the CONFERENCE DISPLAY Feature key will display a different attendee, until all attendees have been displayed. (See Fig 1-1)

In Fig 1-1, an illustration of Dterm series *i* is used as an example.

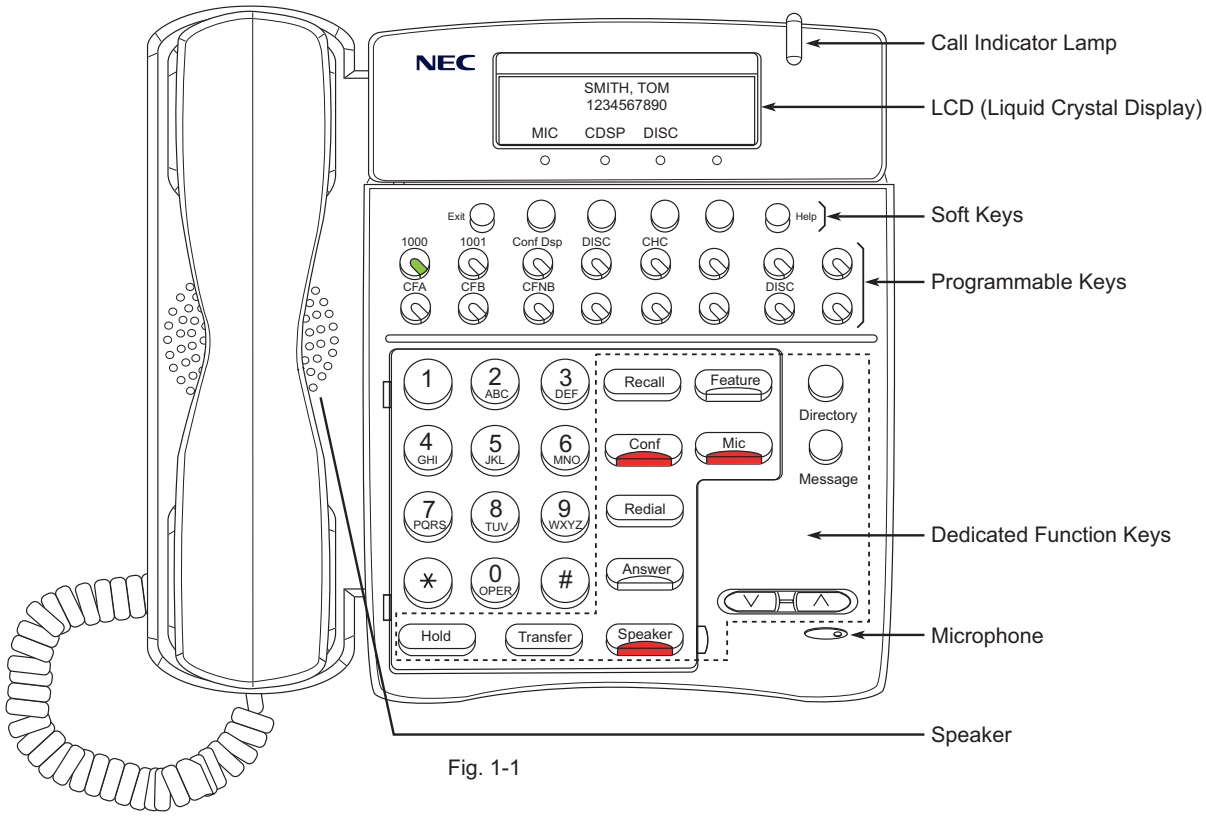


Fig. 1-1

To Disconnect a Party During a Conference. (See Fig. 1-1)

1. Press the CONFERENCE DISPLAY Feature key, until the party to be dropped is displayed.
2. Press the DISC key to drop the party.

Service Conditions

1. All the other Add-on Conference 8 Party [A-121] service conditions apply.
2. This feature is not available for Meet Me conference or Progressive conference.
   a.) It functions on Add-on Conference 8 Party [A-121], Call Hold Conference 8 Party and Assisted Conference 8 Party [A-143].
3. The CONFERENCE DISPLAY Feature key in not available for 3 party Conferences. (RT 909)
4. The disconnect key will only disconnect a single party.
   a.) Once the disconnect key is used it can not be used again by that user until the user:
   - Adds a new party to the Conference. (Last party Drop)
   - Uses the CONFERENCE DISPLAY Feature Key Again. (Selective Drop)
5. All latest Firmware and Flash for the 8CFT and 32CFTV card must be used.
   a.) SP-3654 must be 04.00 or above.

b.) SP-1114 must be 2A or above.

6. The CONFERENCE DISPLAY Feature Key will only function in an Add-on Conference 8 Party [A-121], Assisted Conference 8 Party [A-143], or Call Hold Conference 8 Party.
7. The conference leader (Port 0) can not be dropped by this feature.
   a.) The leader will be displayed in the conference, but will not be dropped.
   b.) A user can use the DISC key to disconnect its own port if they are not the leader of the conference.
   c.) If a trunk becomes the conference leader because the original station released this trunk will not be able to be released by the DISC key from any other station on the conference.
8. During Conference Display the conference attendee is displayed for 10 seconds.
   a.) After 10 seconds the display is changed back to normal.
   b.) If the DISC key is pressed the last party displayed will be dropped. (Even if the display has timed out)
9. The number / name displayed for the conference attendee is based off availability.
   a.) For Outgoing calls

   - The dialed number is displayed. (Example: 91234567890 9 is LCR access code - 1234567890 is the number)
   - ACID can be assigned to associate an outgoing number to name, but the access code must be included in the assignment. (New and only for this feature)
   - For internal Station calls the ANDD name and the station number will be displayed.
   - If no name is available by ACID command UNKNOWN NAME is displayed for the name.

   b.) For incoming calls

   - The number is based off ANI of the Route and Trunk. (Example: 1234567890 would be displayed if ANI was received off an ISDN Route.)
   - If no ANI is available the number display will be blank.
   - If ISDN Incoming Name is available it will be displayed. (See ISDN Programming Manual)
   - ACID can be used to assign a name to an incoming number if ISDN name service is not available. (see ISDN Programming manual)
   - If no name is available UNKNOWN NAME will be displayed.

Interactions

1. OAI SCF and SMFR/SMFN is not supported on the 8 party conferences.
2. The CONFERENCE DISPLAY Feature key can be assigned to a Soft key pattern, but must be assigned in the normal connected state. The Soft key may not be available in all states displayed.
3. PHS/WLAN Handset (MH Series)/SIP Handler Controlled Standard SIP terminals can attend a conference.
4. Disconnecting a party during a conference is available by operations only from Desktop terminals.
5. SP Controlled Standard SIP terminals are not supported with this feature.

Programming

All normal Programming for Add-on Conference 8 Party [A-121] apply plus the below steps.

STEP 1: **ASYDL** - System Data 1, Index 803, Bit 1 = 1 Consultation Hold Release in-service
System Data 1, Index 803, Bit 2 = 1 Operations are valid for stations in an 8-Party Add On Conference.

STEP 2: **ASFC -** SFI 131 Add-On Conference - 8-Party (must be assigned to stations that are allowed to use the DISC key when in an 8-Party state).

STEP 3: **AKYD** - FKY 131 Disconnect Key, FKY 179 Conference Display Key

STEP 4: **ACID** Only needed to assign a name to an outgoing number if required.
The Access code must be included in the CID.
Type = 3, CID = Dialed Number including Access code (9), SRV3 = 3, NAME = Name to display.

S-159 Soft Phone WLAN Handset

General Description

This feature allows a Soft Phone user to use the WLAN Handset (MH Series) as the handset of the Soft Phone. The user can select a handset (Soft Phone Handset or WLAN Handset (MH Series)) by pressing the Handset Switch key on the Soft Phone.

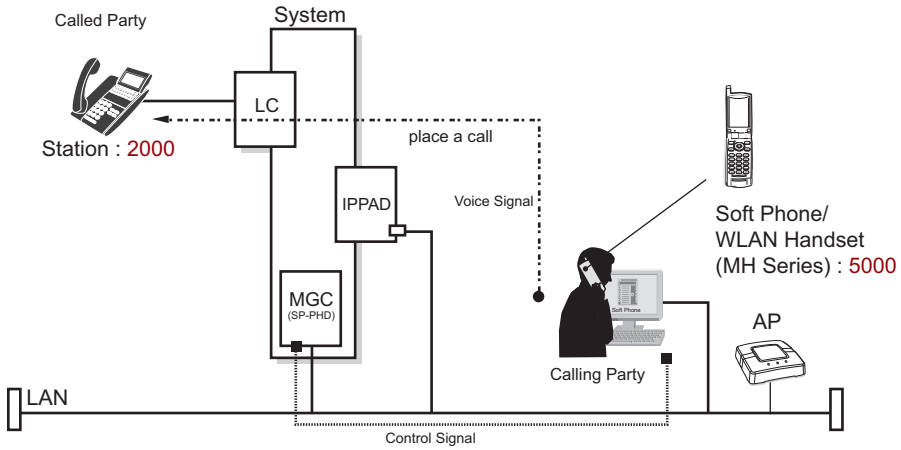
Example:



Operating Procedure

Operating procedures for the following configurations are explained here.

Example:



Call Origination

- Call Origination (Goes off-hook from the WLAN Handset (MH Series) before dialing)
  - (1) Go off-hook from the WLAN Handset (MH Series).
  - (2) Dial the destination number from the WLAN Handset (MH Series) or Soft Phone. - Address Book also can be used.
- Call Origination (Dials before going off-hook from the WLAN Handset (MH Series))
  - (1) Press the Speaker button on the Soft Phone.
  - (2) Dial the destination number from the Soft Phone.- Address Book also can be used.
  - (3) Go off-hook from the WLAN Handset (MH Series).

Call Termination

- Call Termination to the WLAN Handset (MH Series) Line on the Soft Phone
  (1) WLAN Handset (MH Series) or Soft Phone rings.
  (2) Go off-hook from the WLAN Handset (MH Series).
- Call Termination to Sub Line on the Soft Phone
  (1) Sub Line on the Soft Phone rings.
  (2) Click the Sub Line on the Soft Phone, and then go off-hook from the WLAN Handset (MH Series).
  The call can be picked up.

  Note: The call can be picked up also by going off-hook from the WLAN Handset (MH Series) before clicking the Sub Line on the Soft Phone.

Call Park from the Soft Phone

- To Park a Call
  (1) Press the Hold button on the Soft Phone while talking with a destination.
  (2) The call is parked and the Call Park lamp flashes on the Function Key.
  (3) Hear Service Set Tone.
  (4) Go off-hook from the WLAN Handset (MH Series).
- To Retrieve the Parked Call
  (1) Go off-hook from the WLAN Handset (MH Series).
  (2) Press the Call Park button on the Soft Phone.

  Note: The Parked Call can be retrieved also by pressing the Call Park button on the Soft Phone before going off-hook from the WLAN Handset (MH Series).

Service Conditions

1. The terminal that can be used as a handset varies depending on the version of Soft Phone.

	DtermSP30 (Ver.2) (Proprietary Protocol)	DtermSP30 (Ver.4) (Proprietary Protocol)	DtermSP30 (Ver.7 or later) (Proprietary Protocol/SIP) Soft Client SP350
Analog Telephone	N	N	×
Dterm/DtermIP/IP Enabled Dterm	N	N	×
DtermIP(SIP)	N	N	×
WLAN Handset (MH Series)	N	N	×

2. In principle, when assigning WLAN Handset (MH Series) data, use the NDM commands (FPC=1).

3. Call origination is available from the WLAN Handset (MH Series) or Soft Phone by dialing a desired number. Call origination by the Address Book on Soft Phone is also available.

4. This feature can be set/canceled by pressing the Handset Switch key provided on the Soft Phone. Note that the Handset Switch key is invalid while the Soft Phone or WLAN Handset (MH Series) is engaged (Call Termination / Call Origination / Talking).

5. When a Soft Phone using this feature receives an incoming call, the call is answered by the WLAN Handset (MH Series). If the WLAN Handset (MH Series) moves out of zone, Automatic Announcement - Called PS out of Zone/Call Forwarding - PS Incoming Incomplete is applied or ROT is heard.
   If a Soft Phone received an ISDN incoming call, the caller hears RBT.
   If the WLAN Handset (MH Series) moves out of zone, the above features will be applied or ROT will be heard.
   When PS out of zone occurs while the Handset Switch is applied, the calling party hears RBT.

6. While the WLAN Handset (MH Series) is talking by this feature, HOLD, TRANSFER, CONF can be set from the WLAN Handset (MH Series) or Soft Phone.

7. When a call terminates on a Sub Line, it can be answered from the WLAN Handset (MH Series). If this is required, set SFI 29 (Direct Call Pickup) to 1 by ASFC for WLAN Handset (MH Series) and Soft Phone.

8. When this feature has been set the following is restricted:

   - Using the handset of PC in which Soft Phone is installed while the WLAN Handset (MH Series) is engaged in call.

   - Call Origination from/Call Termination to My Line of Soft Phone.

9. When this feature has been set, the WLAN Handset (MH Series) is automatically specified as the handset whenever registration is performed. Even if the Handset Switch is applied, the WLAN Handset (MH Series) becomes the handset of Soft Phone when registration is performed. If Soft Phone registration is per-

formed while the WLAN Handset (MH Series) is placing a call or talking, key operation will not be accepted from the Soft Phone. (**Note 1**) The key operation will be available when the WLAN Handset (MH Series) releases the call.

Note 1: The unacceptable key status is same as that of the WLAN Handset (MH Series) placing a call or talking. However, following keys are excepted:

Function key of Last Number Call: When this key is pressed, the number is displayed on the LCD. However the dial number is unavailable to place a call.

Function key of Logout: This key is acceptable, except pressing after following keys;
- Function key of Last Number Call
- Function key of Speaker
- Function key of Prime Line

10. When this feature is used, the WLAN Handset (MH Series) user requires to go off-hook to place a call.

11. A Soft Phone user cannot retrieve the parked call by pressing the Park key when the user is engaged in call.

12. When assigning Call Pickup Group / Call Pickup Expand data not using Telephone Number, be sure to assign the WLAN Handset (MH Series) Station Number and My Line Number of the Soft Phone by using ACPG / ACPE commands.

13. In the following cases, when the user goes off-hook from the WLAN Handset (MH Series), ROT is heard:

    - WLAN Handset (MH Series) user presses the Call Park Key - the call is parked.

    - The parked call is retrieved by other terminal or the held party releases the call.

14. When assigning the APGN/ADGN command for Call Park Group while this feature is enabled, WLAN Handset (MH Series) Station Number and My Line Number of the Soft Phone must be assigned to the same group. Also, when ASYDL/ASYDN, System Data 1, Index 679, Bit 4 is assigned as “0”, The MG of WLAN Handset (MH Series) Station Number and MG of My Line Number of the Soft Phone must be the same.

15. This feature cannot be used for Hotel / OAI / ACD system.

16. There may be cases where operations while using this feature cannot be recorded in Call Logs. This is also the case when using Redial or Last Number Called -D.

17. When Soft Phone WLAN Handset is used, Call Forwarding - All Calls is not canceled even if Call Forwarding - All Calls Automatic Canceled has been assigned by ASYD command.

18. If Application Sharing function such as White Board is activated while in conversation, the Hold button of Soft Phone cannot be used until the conversation is completed.

19. When using this feature, it is available to retrieve only the held call via Call Park Group -D.

20. This feature is not available for a call via PHC.

21. When a WLAN Handset (MH Series) uses this feature, the WLAN Handset (MH Series) user can place a call on hold by using Call Park access codes. The data programming is required for Call Park.

22. When the handset is WLAN Handset (MH Series), the WLAN Handset (MH Series) cannot answer the held call on the Sub Line on Soft Phone.

23. Soft Phone WLAN Handset is available when the talking WLAN Handset (MH Series) moves among FCCS offices.

24. Both of WLAN Handset (MH Series) and Soft Phone can release the call when WLAN Handset (MH Series) is used as a handset.

25. When PRESET DIALING from WLAN Handset (MH Series) is executed in the state that number is left on the LCD of Soft Phone connected with WLAN Handset (MH Series), WLAN Handset (MH Series) will receive ROT or Dead Number Announcement.

26. A maximum of 256 units per MG can be assigned for MG (Module Group) of WLAN Handset (MH Series) Line Appearance data (APLAML/APLAMN).

27. When a call is transferred by switch hook operation on a WLAN Handset (MH Series), the transferred call must be retrieved by switch hook flash operation on the WLAN Handset (MH Series). On this occasion, if the transferred call is retrieved by switch hook flash operation on the Soft Phone, a difference of connection status indications will occur between the WLAN Handset (MH Series) and the Soft Phone. **Note 2**

28. When a call is transferred by switch hook operation on a Soft Phone, the transferred call must be retrieved by switch hook flash operation on the Soft Phone. On this occasion, if the transferred call is retrieved by switch hook flash operation on the WLAN Handset (MH Series), a difference of connection status indications will occur between the WLAN Handset (MH Series) and the Soft Phone. **Note 2**

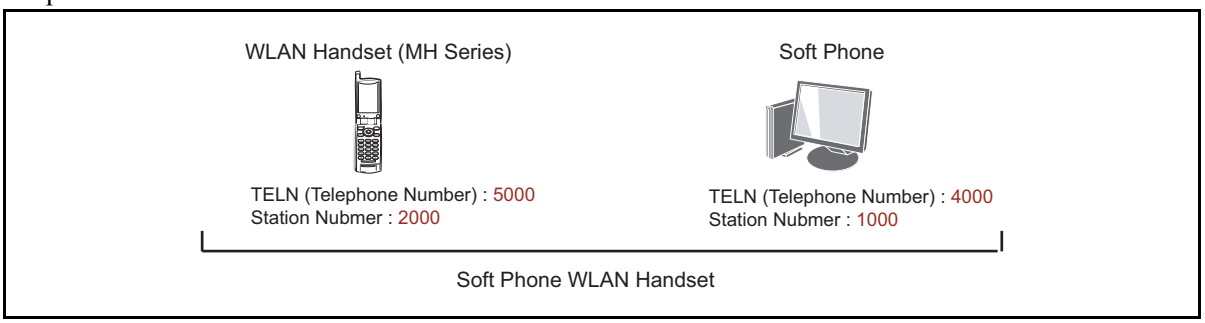
29. Call Park number cannot be displayed on the LCD of WLAN Handset (MH Series).

30. In this feature, when assigning Call Pickup-Group and Expanded Call Pickup-Group data, you only need to assign station number of WLAN Handset.

Note 2: The LCD on WLAN Handset (MH Series) may indicate that it is in the process of transferring while in a conversation, and vice versa.

The following shows the example condition for each terminal. Assign the data referring to the example condition. In case of this example, Station Number used for call origination or termination will be “5000”.

Example condition:



When using standalone configuration

STEP 1: **ASYD** (Assignment of System Data)
System data 1, Index 390, bit 2 = 1. Allow the Caller ID display on the bottom line of Dterm LCD in service.

STEP 2: **ASCL** (Station Class Data) - Assign the same Station Class to both WLAN Handset and Soft Phone.

Note: When the Station Classes of the WLAN Handset and the Soft Phone are different, some operations of this feature may not be performed from the Soft Phone.

Read Station Class Data of WLAN Handset.
TN: Tenant Number
STN: 2000 (Station Number of WLAN Handset)
TEC=28 (PS)
RSC: Route Restriction Class [0-15]
SFC: Service Feature Restriction Class [0-15]

For Soft Phone, assign the same Station Class as that of WLAN Handset.
TN: Tenant Number
STN: 1000 (Station Number of Soft Phone)
TEC=12 (Dterm)
RSC: Route Restriction Class [0-15] (Same as RSC of WLAN Handset.)
SFC: Service Feature Restriction Class [0-15] (Same as SFC of the WLAN Handset)

STEP 3: **APLAMN** (Assignment of PS Line Appearance Module - Group for NDM)

Assign the MG of Soft Phone that accommodates WLAN Handset (MH Series) Line as a Sub Line to Telephone Number of the WLAN Handset (MH Series).
UGN: User Group Number
TELN: Telephone Number of the WLAN Handset (MH Series)
Local Module Group

[In case of the example condition]
UGN: User Group Number
TELN: 5000
Local Module Group

Note: Although this parameter allows the system to register an MG automatically by pressing the Set button without specifying the MG, do not register in this way here. It is because MG assigned in this way conflicts with MG registered Soft Phone for this feature.

Note: When the number of unit is more than 256 per MG, assign the unit also to the MG Number 02-07.

STEP 4: **AKYD** (Assignment of Key Data for Dterm)

Assign the WLAN Handset (MH Series) Station Number to the Prime Line of Soft Phone.
PL TN: Prime Line Tenant Number
PL STN: Prime Line Station Number (Station Number of WLAN Handset (MH Series))
S = 0 (Without OFF-HOOK suppression)
KYN: Key Number
KYI = 2 (Multiline)
Multi Line TN: Multi Line Tenant Number
Multi Line STN: Multi Line Station Number (Station Number of WLAN Handset (MH Series))
KYN: Key Number
KYI = 2 (Multiline)
Multi Line TN: Multi Line Tenant Number
Multi Line STN: Multi Line Station Number (Station Number of Soft Phone)

STEP 5: **APJSN** (Assignment of PS Join Soft Phone for NDM)

Assign the relationship between the My Line of Soft Phone and WLAN Handset (MH Series).
UGN : WLAN Handset (MH Series) User Group Number
TELN : WLAN Handset (MH Series) Telephone Number
S_UGN : Soft Phone User Group Number
S_TELN : Soft Phone Telephone Number

[In case of the example condition]
UGN : WLAN Handset (MH Series) User Group Number
TELN : 5000 (Telephone Number of WLAN Handset (MH Series))
S_UGN : Soft Phone User Group Number
S_TELN : 4000 (Telephone Number of Soft Phone)

When using FCCS

STEP 1: **ASYD** (Assignment of System Data)
System data 1, Index 390, bit 2 = 1. Allow the Caller ID display on the bottom line of Dterm LCD in service.

STEP 2: **ASCL/ASCL_T** (Station Class Data)

Assign the same Station Class to both WLAN Handset and Soft Phone.

Note: When the Station Classes of the Soft Phone and the WLAN Handset are different, some operations of this feature may not be performed from Soft Phone.

Read Station Class Data of WLAN Handset.
UGN: User Group Number
TELN: 5000 (Telephone Number of WLAN Handset)
FPC: Fusion Point Code
TN: Tenant Number
STN: 2000 (Station Number of WLAN Handset)
TEC=28 (PS)
RSC: Route Restriction Class [0-15]
SFC: Service Feature Restriction Class [0-15]

For Soft phone, assign the same Station Class as that of WLAN Handset.
UGN: User Group Number
TELN: 4000 (Telephone Number of Soft Phone)
TN: Tenant Number
STN: 1000 (Station Number of Soft Phone)
TEC=12 (Dterm)
RSC: Route Restriction Class [0-15] (Same as RSC of WLAN Handset.)
SFC: Service Feature Restriction Class [0-15] (Same as SFC of WLAN Handset.)

STEP 3: **APLAMN** (Assignment of PS Line Appearance Module - Group for NDM)

Assign the MG of Soft Phone that accommodates WLAN Handset (MH Series) Line as a Sub Line to Telephone Number of the WLAN Handset (MH Series).
UGN: User Group Number
TELN: Telephone Number of the WLAN Handset (MH Series)
Local Module Group

[In case of the example condition]
UGN: User Group Number
TELN: 5000
Local Module Group

Note: Although this parameter allows the system to register an MG automatically by pressing the Set button without specifying the MG, do not register in this way here. It is because MG assigned in this way conflicts with MG registered Soft Phone for this feature.

Note: When the number of unit is more than 256 per MG, assign the unit also to the MG Number 02-07.

STEP 4: **AKYD** (Assignment of Key Data for Dterm)

Assign the WLAN Handset (MH Series) Station Number to the Prime Line of Soft Phone.
PL TN: Prime Line Tenant Number
PL STN: Prime Line Station Number (Station Number of WLAN Handset (MH Series))
S = 0 (Without OFF-HOOK suppression)
KYN: Key Number

KYI = 2 (Multiline)
Multi Line TN: Multi Line Tenant Number
Multi Line STN: Multi Line Station Number (Station Number of WLAN Handset (MH Series))
KYN: Key Number
KYI = 2 (Multiline)
Multi Line TN: Multi Line Tenant Number
Multi Line STN: Multi Line Station Number (Station Number of Soft Phone)

[In case of the example condition]
PL TN: Prime Line Tenant Number
PL STN: 2000 (Station Number of WLAN Handset (MH Series))
S = 0 (Without OFF-HOOK suppression)
KYN: 1-32
KYI = 2 (Multiline)
Multi Line TN: Multi Line Tenant Number
Multi Line STN: 2000 (Station Number of WLAN Handset (MH Series))
KYN: 1-32
KYI = 2 (Multiline)
Multi Line TN: Multi Line Tenant Number
Multi Line STN: 1000 (Station Number of Soft Phone)

Assign Handset Switch button to a function button of Soft Phone when required.
KYN: Key Number
KYI = 1 (Function Key)
FKY = 157 (Handset Switch)

STEP 5: **APJSN** (Assignment of PS Join Soft Phone for NDM)

Assign the relationship between the My Line of Soft Phone and WLAN Handset (MH Series).

UGN	: WLAN Handset (MH Series) User Group Number
TELN	: WLAN Handset (MH Series) Telephone Number
S_UGN	: Soft Phone User Group Number
S_TELN	: Soft Phone Telephone Number

[In case of the example condition]

UGN	: WLAN Handset (MH Series) User Group Number
TELN	: 5000 (Telephone Number of WLAN Handset (MH Series))
S_UGN	: Soft Phone User Group Number
S_TELN	: 4000 (Telephone Number of Soft Phone)

Data Assignments for Other Services

There are two types of services: services that need to be assigned for only WLAN Handset, and services that need to be assigned for both WLAN Handset and Soft Phone.

(a) Services that need to be assigned for only WLAN Handset.
- Incoming Call services:
  A variety of Station Hunting services or UNIFORM CALL DISTRIBUTION (UCD) [U-1]
- Call Pickup services:
  CALL PICKUP-GROUP [C-7] and Expand Call Pickup-Group
- Setting/Canceling of CF services:
  CALL FOWRARDINFG-ALL CALLS [C-5], CALL FORWARDING -BUSY LINE [C-2], CALL FORWARDING-DON'T ANSWER [M-25] and LOGGED OUT IP STATION-CALL DESTINATION [L-53]
- SP Controlled SIP Terminal On Hold (Assignment of ASPA/ASPAL/ASPAN SRV=SSCA SIDA=119 is needed.)

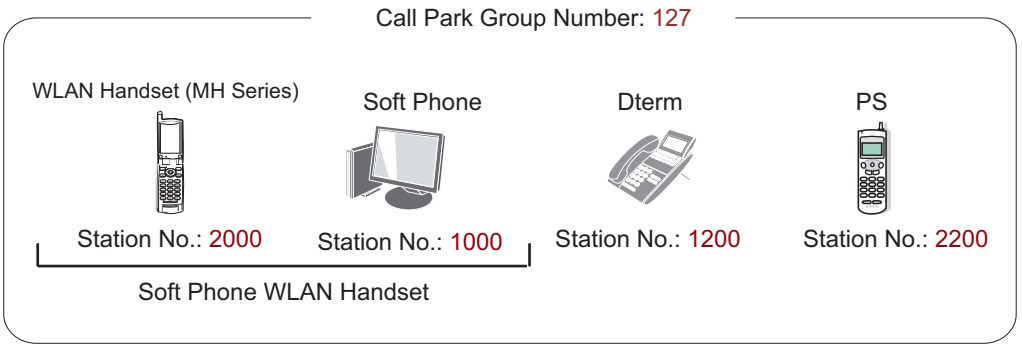
(b) Services that need to be assigned for both WLAN Handset and Soft Phone. **Note 3**
- CALL PARK GROUP-D [C-176D]:
  WLAN Handset and Soft Phone must be assigned to the same group of the same MG.
- LEAST COST ROUTING FOR MG-PRI [L-54] (MA-ID) (Assignment of AMAA is needed for Soft Phone.)
- Other services:
  Assign required data for WLAN Handset and Soft Phone.

Note 3: When the service data is not assigned for Soft Phone, you may not operate the services from Soft Phone (even while WLAN Handset is not used as the handset of Soft Phone).

- When using Call Park

  Following shows the example condition when using Call Park. Assign the data referring to the example condition. For the basic data programming and conditions of Call Park, see Call Park in this guide and Call Park [C-29] in Data Programming Manual - Business.

  Example condition:



Note: Suppose that Softphone (Station No.: 1000) and Dterm (Station No.: 1200) are registered to MG (Module Group) 3.

Parameter	Terminal				
	Soft Phone WLAN Handset		Dterm (1200)	WLAN Hand-set (MH Series) (2200)	GROUP NO (127)
	WLAN Hand-set (MH Series) (2000)	Soft Phone (1000)			
ADGN	-	GROUP NO: 127	GROUP NO: 127	-	-
APGN	MG: 03 GROUP NO: 127 STN: 2000	-	-	MG: 03 GROUP NO: 127 STN: 2200	-
APGS	-	-	-	-	MG: 03 GROUP NO: 127 STN: 1000

STEP 1: **ASFC** (Assignment of Service Feature Restriction Class)
SFI=67: Call Park-Access and Answer
SFI=68: Call Park-Called
RES=1 (Allowed)

STEP 2: **AKYD** (Assignment of Key Data for Dterm)

Assign Call Park Group -D Key. Assign FKY 75-82 to the Soft Phone key.

Note: When only one Call Park Group Key is assigned, assign FKY 75 to the Dterm Key. When more than one Call Park Group Key is assigned, FKY 75 must be assigned first, and then FKY 76, FKY 77, and continuing through the remaining FKY 78 - FKY 85.

STEP 3: **ASYDL/ASYDN** (Assignment of System Data for LDM/NDM)

The following data setting is required when enabling Call Park Group across MGs/IMGs.

System Data 1, Index 679, bit 4 = 1 (Enable Call Park Group -D across MGs/IMGs)

Note: Delete all existing data assigned through the APGS, APGN and ADGN commands in advance.

STEP 4: **ADGN** (Assignment of Dterm Group Number)

Assign a group number to a station number of the Soft Phone using this feature and to other station numbers to be assigned to the same group. The group number must be the same as the one assigned by the APGN command.

STN: Station Number
GROUP NO: Group Number (127-255)

[In case of the example condition]
STN: 1000, 1200
GROUP NO: 127

Note: When ASYDL/ASYDN, System Data 1, Index 679, Bit 4 is assigned as “0”, each group must consist of stations belonging to the same Module Group (MG).

STEP 5: **APGN** (Assignment of PHS Group Number)

Assign MG and a group number to the WLAN Handset (MH Series) Station Numbers to be assigned to the same group. MG must be the same as the that of Soft Phone using this feature. The group number must be the same as the one assigned by the APGN command.

TN: Tenant number
PS STN: WLAN Handset (MH Series) Station Number
MG: Module Group Number (00-07)
GROUP NO: Group Number (127-255)

[In case of the example condition]
TN: Tenant number
PS STN: 2000, 2200
MG: 03
GROUP NO: 127

STEP 6: **APGS** (Assignment of PHS Group Number Control Station)

Assign a station number of the Desktop terminal that are desired to be added to the same MG and GROUP NO as the PS station numbers assigned by the APGN command.

MG: Module Group Number (00-07)

GROUP NO: Group Number (127-255)
TN: Tenant number
STN: Station Number

[In case of the example condition]
MG: 03
GROUP NO: 127
TN: Tenant number
STN: 1000

FKY	Service Feature	Soft Phone WLAN Handset	
		is used	not used
1	Call Forwarding - Busy Line	×	×
2	Call Forwarding - All Calls	×	×
3	Executive Right of Way	–	×
4	Call Waiting	–	×
5	Call Back	–	–
6	Message Reminder	–	–
7	Save and Repeat	–	×
8	Last Number Call	–	×
9	Boss-Secretary - Message Waiting Lamp Control; Set	–	–
10	Call Hold	–	–
11	Boss-Secretary - Message Waiting Lamp Control; Cancel	–	–
12	Call Pickup - Group	×	×
13	Three-way Calling	×	×
14	Voice Call	–	× Note 4
15	Speed Calling - System	–	×
16	Function	–	–
17	CAS SHF	–	×
18	Call Transfer - All Calls	×	×
19	UCD Busy Out	–	–
20	Non-Exclusive Hold / Exclusive Hold (Hold)	× Note 5	×
21	Flash Button	×	×

FKY	Service Feature	Soft Phone WLAN Handset	
		is used	not used
22	Call Forwarding - Don't Answer	×	×
23 - 24	-		
25	ISDN Malicious Call (Australia and U.A.E. only)	–	×
26	Speed Calling - Station	–	–
27	Originate (ORG)	–	×
28	Answer (ANS)	–	×
29	Data (DATA)	–	–
30	Display / Auto (DSPY / AUTO)	–	–
31	Data Transfer (DAX)	–	–
32	-		
33	Message Waiting Lamp (MW)	–	–
34	OAI Key Code 1 (Break)	–	–
35	OAI Key Code 2 (Auto / Man)	–	–
36	OAI Key Code 3	–	–
37	OAI Key Code 4 (Work)	–	–
38	OAI Key Code 5 (Night)	–	–
39	OAI Key Code 6 (Trunk Trouble)	–	–
40	OAI Key Code 7 (Tally)	–	–
41	OAI Key Code 8 (Assistance Key)	–	–
42	OAI Key Code 9 (Emergency Key)	–	–
43	OAI Key Code 10 (Log ON / OFF)	–	–
44	OAI Key Code 11 (Answer)	–	–
45	OAI Key Code 12 (Monitor / Barge)	–	–
46	OAI Key Code 13	–	–
47	UCD Call Waiting Lamp / OAI Key Code 14	–	–
48	Privacy Release	–	–
49	Speed Calling - One Touch	–	–
50	Do Not Disturb	–	–

FKY	Service Feature	Soft Phone WLAN Handset	
		is used	not used
51	Microphone	–	–
52	Speaker	×	×
53 - 54			
55	PAGE	–	×
56	-		
57	Recall	–	–
58	-		
59	Release Key	–	×
60	Manual Signaling Key	–	–
61 - 74	-		
75	Call Park Group -D Key 1	×	×
76	Call Park Group -D Key 2	×	×
77	Call Park Group -D Key 3	×	×
78	Call Park Group -D Key 4	×	×
79	Call Park Group -D Key 5	×	×
80	Call Park Group -D Key 6	×	×
81	Call Park Group -D Key 7	×	×
82	Call Park Group -D Key 8	×	×
83	Retain Conference	–	–
84	Serial Call	–	–
85	Internal Zone Paging	–	–
86	-		
87	Wake Up Set	–	–
88	Wake Up Reset	–	–
89	Do Not Disturb Set	–	–
90	Do Not Disturb Reset	–	–
91	Room Cut Off Set	–	–
92	Room Cut Off Reset	–	–

FKY	Service Feature	Soft Phone WLAN Handset	
		is used	not used
93	Message Waiting Set	–	–
94	Message Waiting Reset	–	–
95	Check In	–	–
96	Check Out	–	–
97	Status	–	–
98	Audit	–	–
99	Enter	–	–
100	Clear Enter	–	–
101	END	–	–
102 - 112	-		
113	Call ID Change Display	–	–
114	-		
115	Call Pickup Expand	×	×
116	-		
117	Ringer Mute	–	×
118	Call Return	–	×
119	-		
120	Next Page of Software Keys	–	–
121	Previous Page of Software Keys	–	–
122	Soft Key Assignment	–	–
123	Off Hook Ringing (OFR)	–	–
124	Mute (MUTE)	–	–
125	Headset / Handset	–	–
126	PS Key	–	–
127	Boss Move	–	–
128	Call Block	–	–
129	Call Trace	–	×
130	Calling Party CLIR / CLIP Control (Australia only)	–	–

FKY	Service Feature	Soft Phone WLAN Handset	
		is used	not used
131	Disconnect Key	–	–
132	Call Redirect (North America only)	–	–
133	Remote Call Hold (North America only)	–	–
134 - 135	-		
136	Selectable Station Call Pickup	×	×
137	Selectable Station Call Pickup Group	×	×
138 - 141	-		
142	LOGOUT (for IP Enabled Dterm)	×	×
143	Voice Coding Mode Change	–	–
144	Voice Coding Select	–	–
145 - 147	-		
148	Call Forwarding - Logout	–	–
149	Message Audible Alert (North America only)	–	–
150	-		
151	MA-ID Setting	–	–
152	-		
153	Day/Night Mode (North America only)	–	–
154	Multi-Internal Zone Paging (North America only)	–	–
155	-		
156	Visitor Key	–	–
157	Handset Switch	×	×
158	Call Park Deluxe Hold/Retrieve-1	–	×
159	Call Park Deluxe Hold/Retrieve-2	–	×
160	Call Park Deluxe Hold/Retrieve-3	–	×
161	Call Park Deluxe Hold/Retrieve-4	–	×
162	Call Park Deluxe Hold/Retrieve-5	–	×
163	Privacy Key	–	-
164-184	-		

FKY	Service Feature	Soft Phone WLAN Handset	
		is used	not used
185	Call Pickup Key Group-1 **Note 6**	–	–
186	Call Pickup Key Group-2 **Note 6**	–	–
187	Call Pickup Key Group-3 **Note 6**	–	–
188	Call Pickup Key Group-4 **Note 6**	–	–
189	Call Pickup Key Group-5 **Note 6**	–	–
190 - 320	-		

Note 4: Available only when Soft Phone is a calling party

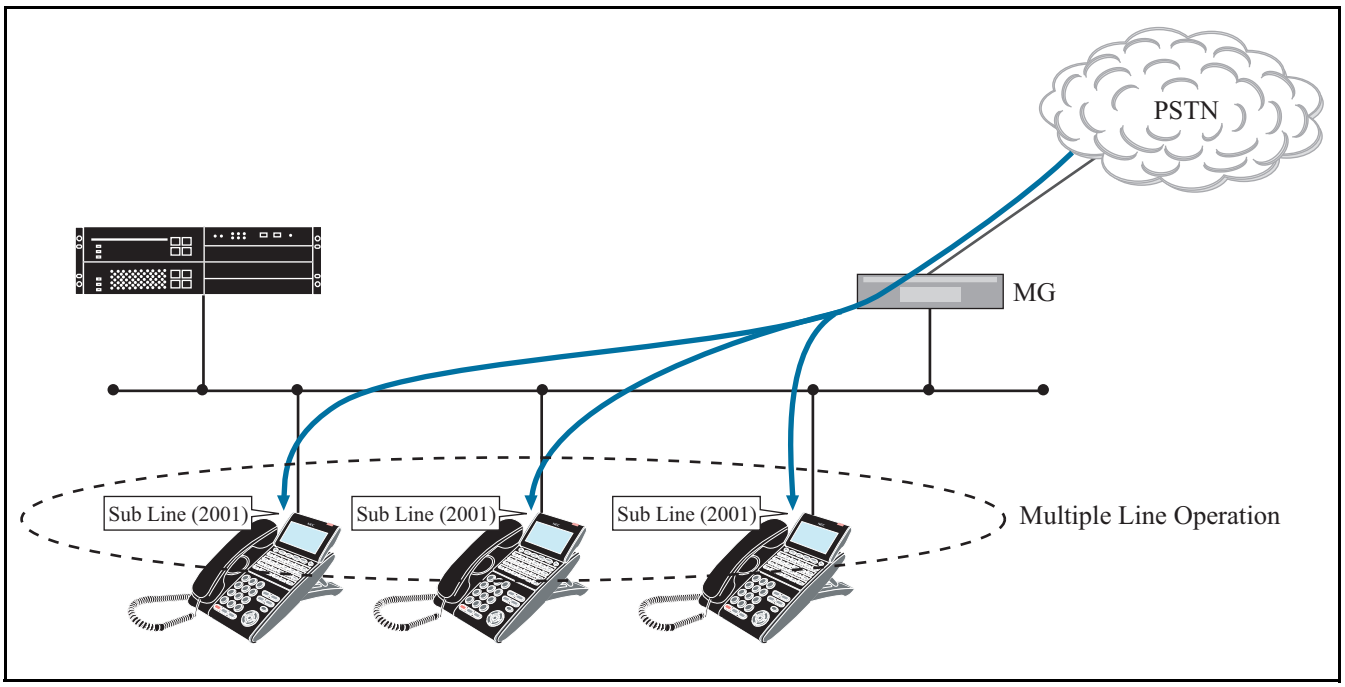
Note 5: Available when Call Park access codes have been assigned except Exclusive Hold.

Note 6: Call Trace is not available for WLAN Handset (MH Series).

S-160 SIP Multiple Line Operation

General Description

This feature allows SIP Multiple Line terminals to accommodate multiple lines other than My Line (i.e. Sub Lines) on the programmable Line/Feature keys. In Multiple Line group, a station can answer the call directed to other member's My Line by using Sub Line.



Operating Procedure

Operation procedure on each service feature is the same as the DtermIP terminal's.

Service Conditions

1. The time information assigned by ATIM/ATIMN/ANTPL is applied to SIP Multiple Line terminals.
2. Conditions for using SIP Multiple Line terminals in OAI/ACD system are:
   - OAI system supports SIP Multiple Line terminals.
   - ACD system supports DT800/DT700 Series only.
3. See "Payload Setting Available for each IP Equipment" in Chapter 3 IP Equipment of System Description. for the applicable payload type on SIP Multiple Line terminal.
4. Feature List shows the service features supported on SIP Multiple Line terminal.
5. For the configuration settings on SIP Multiple Line terminal, see Peripheral Equipment Description (Multi-Line Telephone).

6. Do not set User ID and password for login from the configuration menu of SIP Multiple Line terminal when you use MAC Authentication to log in to the SIP Multiple Line terminal. If both the AMAC command data and the configuration menu data (User ID and password for login) of SIP Multiple Line terminal are set, the setting of the configuration menu has higher priority than that of the AMAC command when you log in to the SIP Multiple Line terminal.
7. The following table shows the availability of functions by pressing the Feature key and the digit keys on SIP Multiple Line terminals. **Note 1**

Digit Key	Function	Availability
0	Sidetone ON ⇔ OFF	X
1	Microphone ON ⇔ OFF **Note 2**	X
2	Handset receiving volume setting **Note 3**	N
3	Ringing Tone setting **Note 3**	N
4	Sending/Receiving volume control	N
5	Hands-Free ON **Note 4**	N
6	Hands-Free OFF **Note 4**	N
7	Call Indicator Lamp ON ⇔ OFF	X
8	Dynamic Dial Pad/Preset Dial ON ⇔ OFF **Note 5**, **Note 6**	X

Note 1: From FP95-114 V4 Issue 2.0, these functions can be set with the ATIFD command instead of using the Feature key and the digit keys. For details, refer to PREFERENCE SETTING - MULTI-LINE TELEPHONE WITH PCPRO [P-90].

Note 2: For DT800/DT700 Series and DtermIP (SIP), “MIC: ON ⇔ OFF” function by pressing the Feature key + the digit key 1 also switches “MIC ON” and “MIC OFF” display on the LCD in addition to the MIC LED indication ON/OFF switching.

Note 3: The function equivalent to this can be set from the configuration settings on terminal.

Note 4: For DT800/DT700 Series and DtermIP (SIP), Hands Free setting (used/not used) is in conjunction with the setting of “MIC: ON ⇔ OFF” function by pressing the Feature key + the digit key 1.

Note 5: Dynamic Dial Pad and Preset Dial ON cannot be activated simultaneously.

Note 6: For using this function, the assignment of system data is required. For the details, refer to Selective Dialing [S-162].

- For DT800/DT700 Series, the setting procedure on terminal is as follows.

  [Config] → [3. Maintenance Settings] → [4. Adjust] → [1. Audio] → [3. Receive Bias]

  Note: For the details, refer to Peripheral Equipment Description (Multi-Line Telephone) - Chapter 3 IP Station - Configuration Setup.

- For DtermIP(SIP), the setting procedure on terminal is as follows.

  [3.Audio&Visual Settings] → [31.Audio Settings] → [315.Receive Bias]

  Note: For the details, refer to Peripheral Equipment Description (Multi-Line Telephone) - Chapter 3 IP Station - Configuration Setup.

- For DT800/DT700 Series, the setting procedure on terminal is as follows.

  [Config] → [5. Setting] → [1. User Setting] → [1. Incoming Call]

  Note: For the details, refer to Peripheral Equipment Description (Multi-Line Telephone) - Chapter 3 IP Station - Configuration Setup.

- For DtermIP(SIP), the setting procedure on terminal is as follows.

  [3.Audio&Visual Settings] → [31.Audio Settings] → [317.Ring Settings]

  Note: For the details, refer to Peripheral Equipment Description (Multi-Line Telephone) - Chapter 3 IP Station - Configuration Setup.

8. When the PH Fixed Hunting feature is enabled (ASYDL, System Data 1, Index 824, Bit 0 = 1), SIP Multiple Line terminals are brought under the control of the same PHD every time they log in. In this case, the last settings of the following PHD-managed features are retained on the SIP Multiple Line terminals even after the re-registration to the PHD: **Note 7**
   a.) Lamp indication on SIP Multiple Line terminals
   b.) Call features below:

   - Message Reminder [M-19]
   - Last Number Call -D [L-6D]
   - Save and Repeat -D [S-24D]
   - Call Return [C-151]

   Note 7: The last settings are not retained when system data is added or changed with the following commands before re-registration:
   ADKS/AKYD/ASCL/ASCL_T/ASTN/MBLE/MBST/MBST_T

<Conditions for DtermIP (SIP)>

1. DtermIP (SIP) cannot assign alternative IP address as a subordinate connection in the case of DHCP connection failure on starting up.
2. DtermIP (SIP) does not support the payload type G.723.1.
3. Online Monitoring feature supports the speech information display on LCD by holding down the HELP key for over one second. For more details, refer to "Online Monitoring Feature of DtermIP (SIP)" in Peripheral Equipment Description.

<Conditions for DT800/DT700 Series>

1. As DT800/DT700 Series terminal does not have a REDIAL and CONF keys, allocate these functions to any Line/Feature key or Soft key if required.
2. Service features available to DT800/DT700 Series terminal are equivalent to DtermIP (SIP).
3. Online Monitoring feature supports the speech information display on LCD by holding down the HELP key for over one second. For more details, refer to "Online Monitoring" for DT800/DT700 Series in Peripheral Equipment Description.

Interactions

Multiple Line can accommodate stations and virtual lines connected beyond different Module Groups (Expanded Multiple Line Operation [E-26D]). Up to 512 stations can accommodate one SIP Multiple Line station on their Multiple Line. In the same LP (ex. MG00 and MG01), the number of accommodation is not limited. For more details, see Expanded Multiple Line Operation [E-26D].**Note 8**

Note 8: This feature is available when using two or more PIRs and mounting them into different LPs.

Programming

Before starting the station data programming below, initial data setting for connecting TP and SP (MPH setting) is required. See Appliance Model Installation Manual - Chapter 6 System Startup - Additional information on System Startup - Initial Data Assignment.

STEP 1: **ASYD** - Assign the system data.

SYS1 Index 321 Bit 0: Multiple Line operation beyond different Module Groups.
0/1 = Restricted/Allowed

Note: Assign this bit to 1 (Allowed) to use Expanded Multiple Line Operation [E-26D].

STEP 2: **ASYDL** - Assign the system data (LDM).

SYS1, Index 1248, Bit 2: Auto-login through MAC Authentication when the login screen of SIP Multi Line terminal remains displayed for five minutes. **Note 9**, **Note 10**
0/1: Unavailable/Available

Note 9: If you assign "0" to this bit, SIP Multi Line terminal continues to display the login screen until you perform the login operation manually.

Note 10: If you assign "1" to this bit, SIP Multi Line terminal whose station number is assigned to AMAC command automatically logs in to the system through MAC Authentication after the login screen remains displayed for five minutes assigned by AMAC command when you do not operate for 5 minutes on login screen in SIP Multiple Line.

Note: System initialization is required when you change this system data.

STEP 3: **ANPD** - Assign the 1st DC of Station Number and the total digits of Station Number.

STEP 4: **ASPA** - Assign the Access Code of Station Number using the 1stDC specified by ANPD.

TN:	Tenant Number
ACC:	Access Code (maximum 6 digits)
CI:	"N: Normal"/"H: Hooking"
SRV:	"STN: Station Number"

STEP 5: **AISTL** - Assign the IP Station data of each SIP Multiple Line station.

TN:	Tenant Number (1-63)
IP STN:	Station Number of SIP Multiple Line terminal (maximum 5 digits)
KIND:	SIP MultiLine TEL
TEC:	Telephone Equipment Class (12 = Dterm)
RSC:	Route Restriction Class (0-15)
SFC:	Service Feature Restriction Class (0-15)
LENS:	Allocate the SIP Multiple Line terminal to vacant virtual LENS on which station/trunk is not mounted. (SIP Multiple Line terminal cannot overwrite the already registered station/trunk.)

Note: When accommodating the Multiple Line on a SIP Multiple Line terminal's Line/Feature Key by using a software line, first assign the software line data by ASDT (assign SIP Multiple Line terminal's My Line data by AISTL), then allocate the software line to the SIP Multiple Line terminal's Line/Feature key by AKYD.

STEP 6: **AKYD** - Assign My Line, other Multiple Line, and desired function to each Line/Feature key on SIP Multiple Line terminal.

S-162 Selective Dialing

General Description

This feature allows a user to make an outgoing call from a Desktop terminal with the several patterns of operating procedure.

- Dynamic Dial Pad

  Allows a user to make an outgoing call by dialing the desired station number (Digit keys 0 - 9, *, and #) without pressing the SPEAKER key or going off-hook.

- Preset Dial

  Allows a user to make an outgoing call by pressing the SPEAKER key or going off-hook after dialing the desired station number (Digit keys 0 - 9, *, and #).

To Set This Feature From Terminal

- When only Dynamic Dial Pad is activated;
  (ASYD, System Data 1, Index 384 Bit 6=1/ASYDL, System Data 1, Index 1256 Bit 1=0)

1. Press the FEATURE key + 8.
2. Dynamic Dial Pad is set when “DirectDialPad ON” appears on the top of LCD.

- When only Preset Dial is activated;
  (ASYD, System Data 1, Index 384 Bit 6=0/ASYDL, System Data 1, Index 1256 Bit 1=1)

1. Press the FEATURE key + 8.
2. Preset Dial is set when “Preset Dial ON” appears on the top of LCD.

- When both Dynamic Dial Pad and Preset Dial are activated;
  (ASYD, System Data 1, Index 384 Bit 6=1/ASYDL, System Data 1, Index 1256 Bit 1=1)

1. Press the FEATURE key + 8.
2. Dynamic Dial Pad is set when “Dial Mode:Direct” appears on the top of LCD. **Note 1**
3. Press the FEATURE key + 8 again.
4. Preset Dial is set when “Dial Mode:Preset” appears on the top of LCD. **Note 1**

Note 1: Every time the FEATURE key + 8 are pressed, Dial Mode on the top of LCD switches in the order of “Dial Mode:Normal” → “Dial Mode:Direct” → “Dial Mode:Preset”.

To Originate a Call

- Origination with Dynamic Dial Pad

  Dial the desired station number by pressing the Digit keys. The dialed numbers appear on the top of LCD. Pressing the SPEAKER key or going off-hook is not necessary.

- Origination with Preset Dial (ASYD, System Data 1, Index 384, Bit 0 =1)

1. Dial the desired station number by pressing the Digit keys. The dialed numbers appear on the top of LCD.
2. Press the SPEAKER key or go off-hook.

- Origination with Preset Dial (ASYD, System Data 1, Index 384, Bit 0 =0)

1. Dial the desired station number by pressing the Digit keys. The dialed numbers appear on the top of LCD.
2. Press the originating Line key.
3. Go off-hook.

To Cancel Originating Call with Preset Dial

Originating call operation is canceled by the following Soft key operation;

- Press EXIT key or CANCEL key. **Note 2**
- Press CLEAR key when only one digit is dialed. **Note 3**, **Note 4**

Note 2: By pressing the HOLD key on the Bluetooth Handset, you can cancel the call.

Note 3: When the dialed number is two digits or more, dialed number is deleted one digit every time CLEAR key (Soft key or Flex key) is pressed.

Note 4: By pressing the TRANSFER key on the Bluetooth Handset of the Cordless Terminal (DT330 (12BT)), you can cancel the call, or delete one digit from dialed numbers.

To Cancel This Feature From Terminal

- When only Dynamic Dial Pad is activated;
  (ASYD, System Data 1, Index 384 Bit 6=1/ASYDL, System Data 1, Index 1256 Bit 1=0)

1. Press the FEATURE key + 8.
2. Dynamic Dial Pad is canceled when “DirectDialPadOFF” appears on the top of LCD.

- When only Preset Dial is activated;
  (ASYD, System Data 1, Index 384 Bit 6= 0/ASYDL, System Data 1, Index 1256 Bit 1=1).

1. Press the FEATURE key + 8.
2. Preset Dial is canceled when “Preset Dial OFF” appears on the top of LCD.

- When both Dynamic Dial Pad and Preset Dial are activated;
  (ASYD, System Data 1, Index 384 Bit 6 =1/ASYDL, System Data 1, Index 1256 Bit 1=1)

1. Press the FEATURE key + 8 until “Dial Mode:Normal” appears on the top of LCD.
2. Both Dynamic Dial Pad and Preset Dial are canceled when “Dial Mode:Normal” appears on the top of LCD. **Note 5**

Note 5: Every time the FEATURE key + 8 are pressed, Dial Mode on the top of LCD switches in the order of “Dial Mode:Normal” → “Dial Mode:Direct” → “Dial Mode:Preset”.

Service Conditions

1. Once this feature is set at each terminal by pressing the FEATURE key + 8, the setting is maintained even

if the insertion and extraction of Modular Cable, re-registration of IP Station, ELC circuit card initialization and system initialization are performed.

2. When System is updated, this feature setting from each terminal is cleared.
3. To activate this feature with Digital Terminal, the following firmware is required to be equipped on the ELC circuit card.

Circuit Card	Firmware	Remark
PA-16 ELCJ-B	SP-3655 16ELCJB PROG-D	-
	SP-3656 16ELCJB PROG-E	
	SP-3752 16ELCJB PROG-F	
CH-16ELCA	SP-4035 16ELCA PROG-A Issue 1 or later	-
PA-16ELCN	SP-3977 16ELCN PROG-B Issue 1 or later	To enable Preset Dial
PA-16ELCN-A **Note 6**	SP-4013 16ELCN-A PROG-A Issue 1 or later	

Note 6: Cordless Terminal (DT330 (12BT)) is available when SP-4034 16ELCN-A PROG-B Issue1 or later is used.

4. Preset Dial only works when the Terminal Mode Setting of Switch 11 (SW11 5-6) on ELC circuit card is set to ON/ON (Standard Setting). For details on the switch settings of the circuit cards, see sections related to ELC cards in Chapter 3 of the Circuit Card Description.

Conditions applied to Dynamic Dial Pad only

1. The following operations are required when the System Data (ASYD, System Data 1, Index 384 Bit 6=1) is assigned while the system is operating.

   - Digital Terminal: Circuit card initialization of the appropriate ELC circuit card
   - IP Stations: Re-registration by any one IP Station accommodated in SP-PHD

2. This feature is available when the Prime Line is idle, whether or not a call is terminating to the Sub Line.
3. This feature is not available when:
   - a call terminates to a Prime Line.
   - the station presses the SPEAKER key or goes off-hook.
   - the line LED for Prime Line lights or flashes.
4. This feature is not available from an Analog Terminal connected to an APR.
5. When ASYD, System Data 1, Index 384, Bit 0 =1, the following conditions must be considered.
   a.) In case both Prime Line and Sub Line are idle:
   - When dialing the desired station number, this feature call is originated from Prime Line.
   - When dialing the desired station number after pressing a Prime Line key, this feature call is originated from Prime Line.
   - When dialing the desired station number after pressing a Sub Line key, this feature call is originated from Sub Line.

   b.) In case a call terminates to Prime Line and Sub Line is idle:
   - When dialing the desired station number, this feature cannot be activated.
   - When dialing the desired station number after pressing a Prime Line key, this feature cannot be activated.

- When dialing the desired station number after pressing a Sub Line key, this feature cannot be activated.

c.) In case Prime Line is idle and a call terminates to Sub Line:
- When dialing the desired station number, this feature call is originated from Prime Line.
- When dialing the desired station number after pressing a Prime Line key, this feature call is originated from Prime Line.
- When dialing the desired station number after pressing a Sub Line key, Incoming call to Sub Line is answered.

6. IP Stations cannot be placed in Configuration Mode while this feature is set. To enter this mode, clear the Dynamic Dial Pad setting by pressing the FEATURE key + 8.
7. The conditions of this feature when using the Cordless Terminal (DT330 (12BT)) are as follows.

- When this feature is executed from the Bluetooth Handset, the call is originated from the speakerphone on the Bluetooth Cradle.

- While the Bluetooth Cradle displays Incoming/Outgoing call history, because dialing numbers cannot be entered on the Bluetooth Handset, Dynamic Dial Pad cannot be executed from the Bluetooth Handset.

Conditions applied to Preset Dial only

1. The following operations are required when the System Data (ASYDL, System Data 1, Index 1256 Bit 1=1) is assigned while the system is operating.

- Digital Terminal: Circuit card initialization of the appropriate ELC circuit card

- IP Stations: System Initialization

2. Preset Dial cannot be used when SOFT PHONE WIRED-PHONE HANDSET [S-147] is in active.
3. In the case of re-registration by IP stations when LOCATION DIVERSITY [L-55], SR-MGC [S-144] or Remote Node over IP is in active, the latest backup of One-Touch Speed Calling for DtermIP data will be maintained as the data set to each terminal by pressing the FEATURE key + 8.
4. This feature cannot be used with terminals that do not support Soft keys such as DT300 Series (DT310) 2/12-button type and DT700 Series (DT710) 2-button type and Softphone.
5. LCD does not display the name stored in the database of PHONEBOOK [P-83] during the Preset Dial operation. The name will be displayed after the call is originated.
6. Preset Dial operation cannot be made a shift to the phone book registration.
7. DIALED NUMBER PRIVACY MODE [D-165] cannot be used when Preset Dial is in active.
8. If Preset Dial is used to dial digits after Service Code (SSC), such as setting Call Forwarding feature with access code, only SSC can be sent. Even if SSC + digits are dialed, digits after SSC cannot be sent.
9. When station data that is deleted once is assigned again to the newly-mounted terminal without changing the mounting location, data set to each terminal by pressing the FEATURE key + 8 will be carried on to new terminal in each mounting location.
10. Preset Dial operation is available only when the terminal is in idle state. The operation is not available if the terminal is engaged in a call.
11. Pauses cannot be used, so digits sent after the called party answers cannot be assigned.
12. Dialed number should consist of up to 32 digits. The dialed number is invalid if the digits of number exceed 32 digits.

13. When a call terminates to the line on the terminal engaged in Preset Dial operation, the operation will not be interrupted until the call is answered. When the call is answered, the line will be engaged in the call.
14. During Preset Dial operation, key operations, such as a Feature key operation, will be invalid if they are not related to Preset Dial. In the case of pressing the REDIAL key, redialed number will be displayed until the called party is rung.
15. When call origination is not performed normally such as because seizing the register is failed, call origination history may not be stored.
16. When the number of digits of the dialed number is less than the maximum necessary digits (values assigned to MND parameter of AMND/AMNDL/AMNDN), the call is originated after a predetermined prepause timer assigned by ASYD, SYS 1, Index 128 bits 0-3 (default: 12 seconds) expires.
17. Outgoing call from Incoming/Outgoing call history is not available while using Preset Dial from the Bluetooth Cradle of Cordless Terminal (DT330 (12BT)).

Programming

STEP 1: **ASYD** - System Data 1, Index 384, Bit 6. Assign this bit as data "1" to enable Dynamic Dial Pad.
Note 7, **Note 8**

STEP 2: **ASYDL** - System Data 1, Index 1256, Bit 1. Assign this bit as data "1" to enable Preset Dial.
Note 8

Note 7: In the case of Dynamic Dial Pad, any one IP Station accommodated in SP-PHD must perform registration for this data change to take effect.

Note 8: When "Office Data Load & System Initialize" or "Program & Office Data Load & System Initialize" is selected on SINZ command, the following operations are required to retain the office data:
1. Boot MEM_HDD command.
2. Select "One-Touch Speed Calling for DtermIP."
3. Execute MEM_HDD command and retain the office data.

S-165 Security Control Setup

General Description

This feature allows you to enable/disable the Security button (**Note 1**). This button starts up the Security mode (**Note 2**) with one touch. The following three schemes are available for the control of the Security button with System Data assignment.

Note 1: On the DT700 Series (DT750/DT730G/DT730) terminals, the Security button is located next to the upper left corner of the LCD. On the DT800 Series (DT830G/DT830) terminals, a Line/Feature key can be programmed as the Security button. For details of the Line/Feature key data, see "Programming."

Note 2: Security mode restricts access to service features and configurations provided by the Telephony Server. For further details, see Peripheral Equipment Description (Multi-Line Telephone) - Chapter 3 IP Station - Configuration Setup - DT800/DT700 Series - Configuration Data Setting Procedure - Security Mode.

(a) Security button invalidation on a system basis **Note 4**

Disables the Security button on all DT800/DT700 Series terminals under the control of the system.

(b) Security button validation on a system basis **Note 4**

Enables the Security button on all DT800/DT700 Series terminals under the control of the system.

(c) Security button control on an individual terminal basis

Enables/Disables the Security button on each terminal by using IP Phone Manager. **Note 5**, **Note 6**, **Note 7**

Note 4: See "Programming" of this feature for details on the data setting of Security button control on a system basis.

Note 5: IP Phone Manager is an application tool for configuration of Desktop terminals. To enable/disable the Security button, set the Lock Mode according to "4.5.3 Security Lock" in the manual of IP Phone Manager. When connecting a Desktop terminal with IP Phone Manager, make sure the port numbers match. You can set the port number on the phone side by selecting [0. Config] - [6. Application Setting] - [2. IPM Self Port].

Note 6: The Security button cannot be enabled by using the IP Phone Manager when the Security button invalidation is set on a system basis. If you try to do this, incorrect information on Security Lock status may be temporarily displayed on the terminal LCD, but it disappears in several seconds.

Note 7: If the Security button is disabled through the IP Phone Manager while the Security button validation is set on a system basis, it will remain disabled until the restart of terminal. After the restart, the Security button will be enabled.

[Notes on Terminal Behavior during the Security Lock]

- In the Security mode, the Security Lock disables the following operation on the terminal: call origination, answer, handset's on-hook/off-hook, and Line/Feature key operation.
- Although an incoming call can terminate to the terminal when the Security Lock is on, the call cannot be answered without releasing the Security Lock.

Operating Procedure

None on the terminal. For more information on operation in the Security mode, see Peripheral Equipment Description (Multi-Line Telephone) - Chapter 3 IP Station - Configuration Setup - DT800/DT700 Series - Configuration Data Setting Procedure - Security Mode.

Service Conditions

<Important Notices>

1. When the security control setting is changed by system data assignment, a restart is required for each terminal; if not, the data change will not be reflected. For details on the system data, see the Programming section in this service.
2. The security status (i.e. Security Lock ON/OFF) on a terminal is maintained after the terminal is restarted.
3. When setting the Security button through the IP Phone Manager, use "Security button control on an individual terminal basis" mode. While in "Security button invalidation on a system basis" mode, the setting to enable the Security button via IP Phone Manager is invalidated. While in "Security button validation on a system basis" mode, the setting to disable the Security button via IP Phone Manager is validated after the re-registration of the terminal.
4. To enable the disabled Security button, use one of the following methods:

   - From [Config] menu on a terminal

     Administrator's menu [Config] → "3.Maintenance Settings" → "3.Data Clear" → "1.Factory Value" or "2.Terminal Data" **Note 8**

   - By Web Programming

     [Maintenance Settings] → [Data Clear] → [Factory Value] or [Terminal Data] **Note 8**

   - By IP Phone Manager **Note 9**

     [DataReset] → [IP Phone Setting]

Note 8: When "1. Factory Value" is selected, all of the data will be reset to the factory default value.
When "2. Terminal Data" is selected, terminal data setting such as IP address, Server Address, data assignment for incoming call, ongoing call, and display will be set to the default data.

Note 9: See manuals for IP Phone Manager for more information.

<Interactions with Other Features>

1. When DT800/DT700 Series terminals re-register to the SR-MGC or the Backup Node of Location Diversity [L-55], the system data setting related to the Security control setting in Primary Node should be the same with the data in Backup Node. If not, the terminals act differently after the node switching because terminal behavior depends on the System Data setting on the node which they re-register to.
2. Even if the Security Lock is enabled, terminal user can originate an emergency call to a pre-assigned number by off-hooking the handset. The Security Lock is released after the emergency call origination.
3. It is possible to make a VOICE CALL [V-2] if the destination terminal is in the Security Lock status. In this case, you can talk over the microphone (MIC) if the called party has set the Security Lock with the MIC turned on. However, other operations (such as making or answering a call, using Feature keys, and going on/off hook with a handset) are restricted on the locked terminal.

<Hardware/Software>

This feature is available for the following terminals and firmware.

DT830/DT830G: FW3.0.0.0 or later

DT750/DT730: FW3.0.0.0 or later

DT730G: FW1.0.0.0 or later

Note: This feature is not available for DT710 terminals because they do not have the Security button.

<Interactions with Networks>

None

<Conditions Specific to This Feature>

1. Security control status is not managed in relation to each station number by the system, but stored on an individual DT800/DT700 Series terminal. When moving DT800/DT700 Series terminals to another system, initialize them in advance.
2. Once the Security Lock is enabled on a DT800/DT700 Series terminal, an incoming call can terminate to the terminal, but users cannot answer the call because the terminal is locked. To answer the call, release the Security Lock.

<Other Conditions>

None

Programming

Security button invalidation on a system basis

STEP 1: **ASYDL/ASYDN** - System Data 1, Index 1256, Bit 3 = 1 (Security button invalidation on a system basis)

Note: Changes made with this command are only effective after restarting the terminal.

STEP 2: **AKYD/AKYD_T** - Assign the following data to the Line/Feature Keys of the Desktop terminal.

Note: This step is only required for DT800 Series terminals.

Note: Most of the parameters for the AKYD command are the same as those for the AKYD_T command, with some exceptions such as TN and STN. Parameters TN and STN of the AKYD command correspond to UGN and TELN of the AKYD_T command, respectively.

TN: Tenant Number (1-63)
STN: Station Number (max. 5 digits)
KYN: Key Number (1-24)
KYI = 0 (Key is not used)
FKY = 197 (Security)

Note: Changes made with this command are only effective after restarting the terminal.

Security button validation on a system basis

STEP 1: **ASYDL/ASYDN** - System Data 1, Index 1256, Bit 3 = 0 (Security button validation in accordance with the setting of ASYDL/ASYDN SYS1 Index 1257 Bit 1)
- System Data 1, Index 1257, Bit 1 = 1 (Security button validation on a system basis)

Note: Changes made with this command are only effective after restarting the terminal.

STEP 2: **AKYD/AKYD_T** - Assign the following data to the Line/Feature Keys of the Desktop terminal.

Note: This step is only required for DT800 Series terminals.

Note: Most of the parameters for the AKYD command are the same as those for the AKYD_T command, with some exceptions such as TN and STN. Parameters TN and STN of the AKYD command correspond to UGN and TELN of the AKYD_T command, respectively.

TN: Tenant Number (1-63)
STN: Station Number (max. 5 digits)
KYN: Key Number (1-24)
KYI = 1 (Function Key)
FKY = 197 (Security)

Note: Changes made with this command are only effective after restarting the terminal.

Security button control on an individual terminal basis

ASYDL/ASYDN - System Data 1, Index 1256, Bit 3 = 0 (Security button validation in accordance with the setting of ASYDL/ASYDN SYS1 Index 1257 Bit 1)
- System Data 1, Index 1257, Bit 1 = 0 (Security button control on an individual terminal basis)

Note: Changes made with this command are only effective after restarting the terminal.

S-166 Software Digital Tone Generator

General Description

Software Digital Tone Generator (hereinafter called "IP-DTG") enables SIP Handler Controlled SIP terminals to receive PBX tones without using an external Digital Tone Generator.

WARNING **After changing the system data (ASYDL, SYS1, Index 1353, Bit 1) for Software Digital Tone Generator (IP-DTG), System Initialization is required.**

Operating Procedure

None

Service Conditions

<Important Notices>

None

<Interactions with Other Features>

1. Connections between the IP-DTG and SIP Handler Controlled SIP terminals cannot be restricted by PEER TO PEER BANDWIDTH CONTROL [P-75]. Adjust the value for MAX_CONN: Maximum Number of Concurrent Connections through the ALIDL command.
2. Collecting Communication Log and Fault Log are not available using RTP INFORMATION OUTPUT [R-58].

<Hardware/Software>

None

<Interactions with Networks>

None

<Conditions Specific to This Feature>

1. A SIP Handler Controlled SIP terminal cannot receive a short tone.
2. 256 terminals can be connected concurrently per system by default.

Note: When the number of the concurrent connection exceeds the limit in each node, local tones generated by the terminal are received or the terminal is released depending on services.

3. Available payload type and size are as follows.

Payload Type	Payload Size
G.711μ-law	20 ms
	40 ms
G.711A-law	20 ms
	40 ms
G.729a **Note 1**	20 ms
	40 ms

Note 1: Available since FP95-114 V4.

Note: For G.711, the codec is determined according to the system data (ASYD, SYS1, Index 64, Bit 0: μ-law/A-law for TSW).
SIP Handler Controlled SIP terminal also needs to support G.711μ-law and A-law. When the codec of IP-DTG does not match with the payload type used by the SIP Handler Controlled SIP terminal, the SIP Handler Controlled SIP terminal is released.

4. Tones assigned with Tone Code (ASYD, SYS1, Index 811) is used for IP-DTG. When a Tone Number is assigned to a Station Number by the ASLG command, tones for each Station Number can be specified on a country basis.
5. DTMF signal detection is available by RFC2833 but In-Band DTMF signal detection is not supported. When SIP Handler Controlled SIP terminal is connected to IP-DTG, while receiving a Special Dial Tone (SPDT) or a Busy Tone (BT), the Telephony Server services become available by detecting DTMF signals of RFC2833 received from the SIP terminal and sending them to the TP part through SIP Handler.
6. When the system is a dual-CPU system, IP-DTG in the ACT side system is used.
7. System Changeover is available while SIP Handler Controlled SIP terminal's receiving tones from IP-DTG. After System Changeover, SIP Handler Controlled SIP terminal keeps receiving the tone from IP-DTG in the former ACT side system. From the next tone, the SIP Handler Controlled SIP terminal receives from IP-DTG in the new ACT side system.
8. Generating a music resource is available as Music-on-Hold. When an external music resource is set, it is preferentially generated.

   Note: Using external music resources is recommended. Hold tones tends to be heard longer, and then it can weigh on CPU occupancy in the Telephony Server and bandwidth of LAN.

9. Minuet is generated as Music-on-Hold.

<Conditions Specific to UMGi/RNIP System>

1. 256 terminals can be connected concurrently per node by default.

   Note: When the number of the concurrent connections for each node (CCN/RNs) exceeds the limit, no tone is received depending on services.

2. IP-DTG data assignment is required per node which accommodates the SIP Handler Controlled SIP terminal.
3. A SIP Handler Controlled SIP terminal and controlling SIP Handler must belong to the same node, then the SIP Handler Controlled SIP terminal will receive tones.
   When the SIP Handler Controlled SIP terminal and controlling SIP Handler belong to different nodes, the SIP Handler Controlled SIP terminal does not receive tones.

<Other Conditions>

IP-DTG performs the registration to TP part and then it is controlled through the Internal PHE.

Programming

To generate the internal music resource for Music-on-Hold, the data assignment for User Agent Class Data is required. For details, refer to SIP HANDLER - STANDARD SIP TERMINAL [S-167].

STEP 1: **ASYD** - System Data 1, Index 64, Bit 0: 0/1=G.711μ-law/G.711A-law

STEP 2: **ASYDL** - Assign the system data.
System Data 1
Index 811, Bit 0-7: Assign Tone Code by countries.
Index 1353, Bit 1=1: Enable Software Digital Tone Generator (IP-DTG).

STEP 3: **ALIDL** - Specify the LENS of IP-DTG.
PH LENS: Specify an even-numbered Group Number. **Note 2**
PH TYPE: Select IP-DTG.
DTG-ID: IP-DTG Identification Number [1-16]
Single-CPU System: Assign DTG-ID = 1 (fixed)
Dual-CPU System: Assign DTG-ID = 1 and 2 (fixed)
For UMGi/RNIP System, refer to the following table:

IP-DTG to be Assigned

PBI No. (LP No.)	CPU Side	IP-DTG
0x30 (0)	#0	1
	#1	2
0x38(4)	#0	3
	#1	4
0x40 (8)	#0	5
	#1	6
0x48(12)	#0	7
	#1	8
0x50(16)	#0	9
	#1	10
0x58(20)	#0	11
	#1	12

IP-DTG to be Assigned

PBI No. (LP No.)	CPU Side	IP-DTG
0x60(24)	#0	13
	#1	14
0x68(28)	#0	15
	#1	16

MAX_CONN: Maximum Number of Concurrent Connection [1-4095 (default: 256)] **Note 3**, **Note 4**

Note: To change the PH LENS or DTG-ID value, delete the existing PH LENS or DTG-ID value first.

Note: If the existing PH LENS or DTG-ID value is deleted, tones from the deleted IP-DTG stops and the SIP Handler Controlled SIP terminal will not receive tones.

Note 2: The LENS can be allocated on either a Virtual PIR or a Physical PIR except for LENS with Group No. 24-31. As for physical PIR, set the data to a vacant slot.
For the PH LENS parameter in UMGi/RNIP System, specify the available Group Number which belongs to LENS in the node corresponding to DTG-ID.

Note 3: Assign the Maximum Number of Concurrent Connections per IP-DTG. For this data assignment, consider the available bandwidth between the Telephony Server (LAN1) and the SIP Handler Controlled SIP terminal.

Payload Size		Transmission Capacity
G.711	20 ms	88 Kbps
	40 ms	76 Kbps
G.729a **Note 5**	20 ms	32 Kbps
	40 ms	20 Kbps

Note 4: The MAX_CONN value can be changed without deleting the existing value. When the changed value is lower than the existing number of concurrent connections, concurrent connections are retained but new connections are restricted until concurrent connections become fewer than the changed value.

Note 5: Available since FP95-114 V4.

STEP 4: **ALOCL** - Assign the Location ID data.

To assign the payload data for RTP packets sent from the IP-DTG, allocate Location ID to an IP address (**Note 6**) used by the IP-DTG.

Note 6: Assign the following IP addresses:
Single-CPU System:Assign LAN1 IP address (ACT)
Dual-CPU System:Assign LAN1 IP address (System-0)/LAN1 IP address (System-1)

Note: Setting of Type of Service (Signaling Packet) data is not applied to the IP-DTG.

STEP 5: **AIVCL** - Assign the IP service voice control data.

Assign the payload data used between the IP-DTG and the SIP Handler Controlled SIP terminal.

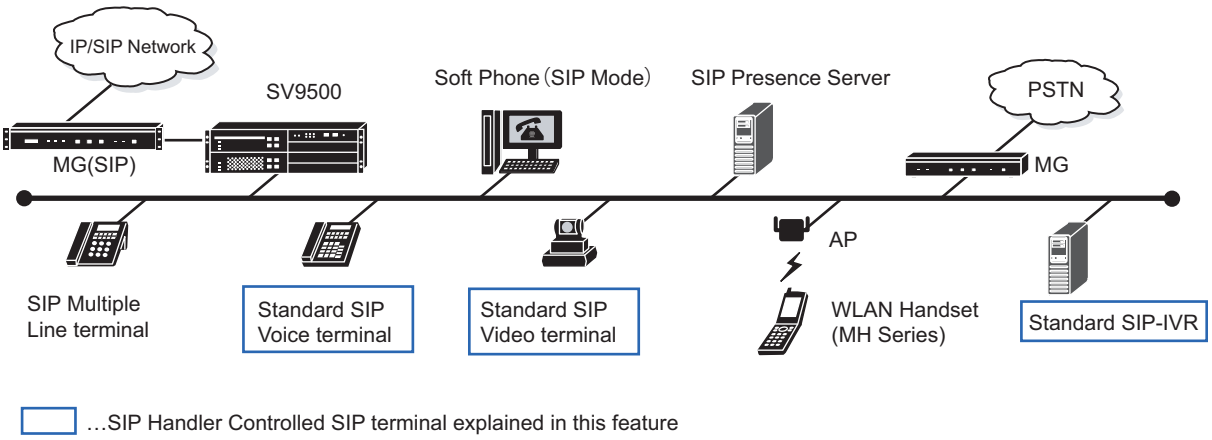
Note: Note the following:

- When assigning the payload data, be sure to include payload type: G.711/G.729a and payload size: 20 ms or 40 ms. If no data is set for this command, payload type: G.711 and payload size: 40 ms are applied.
- Data settings other than payload data are not applied to the SIP Handler Controlled SIP terminal.

S-167 SIP Handler - SIP Terminal

General Description

This feature allows the system to accommodate the SIP terminals mentioned below as stations by controlling them through the SIP Handler inside the Telephony Server. The SIP terminals can establish an interconnection with another Telephony Server Station such as a Desktop terminal, and use various Telephony Server services.



- **Standard SIP Voice Terminal (Note 1, Note 2)**
  Standard SIP voice terminal (third-party product) is based on IETF RFC3261 and Internet-Drafts. Standard SIP voice terminals support basic call services and PBX supplementary services.
- **Standard SIP Video Terminal (Note 1, Note 2)**
  Standard SIP video terminal (third-party product) is based on IETF RFC3261 and Internet-Drafts.
- **Standard SIP-IVR**
  Standard SIP-IVR is based on IETF RFC3261 and controlled by SIP Handler. This device supports typical Telephony Server features and can be used under the UCD and ACD environment.

Note 1: Note that a connection test is required when using a Standard SIP Voice/Video terminal (3rd party product).

Note 2: When using a Standard SIP terminal with LCD (3rd party product), the information displayed on the LCD depends on the specifications of terminal being used.

Control Method of SIP Terminal

SIP terminals can be controlled with the methods listed below.

- SP Controlled **Note 3**
- SIP Handler Controlled

When using SIP terminals, the available services differ according to the control method to be used. Decide which method to use referring to the Feature List included at the beginning of this manual.
SIP Handler Controlled SIP terminal is recommended if SP Controlled SIP terminal is not necessary.

Advantages of SP Controlled Standard SIP Terminal	Advantages of SIP Handler Controlled SIP Terminal
The following features are available with SP Controlled SIP terminal. • Call Forwarding-PS Incoming Incomplete [C-155] • Automatic Announcement-Called PS Out of Zone [A-129] • SIP Instant Messages **Note:** The above features are unavailable with SIP Handler Controlled SIP terminals.	The following features are available with SIP Handler Controlled SIP terminal. These are enhanced features in comparison to SP Controlled SIP terminal features. • Various PBX services which have been enriched by enabling PBX tones to be received through IP-DTG • Enriched OAI service • Enriched Hotel service • Improved maintainability (Detailed data setting by User Agent/ the reduction of the number of occupied LENS) • Ongoing call retention during the CPU changeover

Note 3: For details of this method, refer to SP CONTROLLED SIP TERMINAL [S-168].

User Agent Class

SIP messages and sequences vary depending on the Standard SIP terminal type with each manufacturer. To adjust those differences, detailed operation data can be assigned per Standard SIP terminal type by using the User Agent Class data assigned with the AUACL/AUACN command.

When User Agent information notified by a SIP Handler Controlled SIP terminal contains User Agent Information assigned with the AUACL/AUACN command (partial match), the Standard SIP terminal is controlled according to the detailed operation data as a terminal that belongs to a User Agent Class corresponding to the SIP Handler Controlled SIP terminal.

By assigning detailed operation data per User Agent Information, each SIP Handler Controlled SIP terminal can be controlled without changing the office data even if a terminal user logs in to a different type of the SIP Handler Controlled SIP terminal through the same login Telephone Number.

Also the User Agent Class data can be assigned to Telephone Number with the ADSDL/ADSDN command. The User Agent Class is decided according to a login Telephone Number regardless of the User Agent Information given by the SIP Handler Controlled SIP terminal.

Single Line Mode and Multi Line Mode

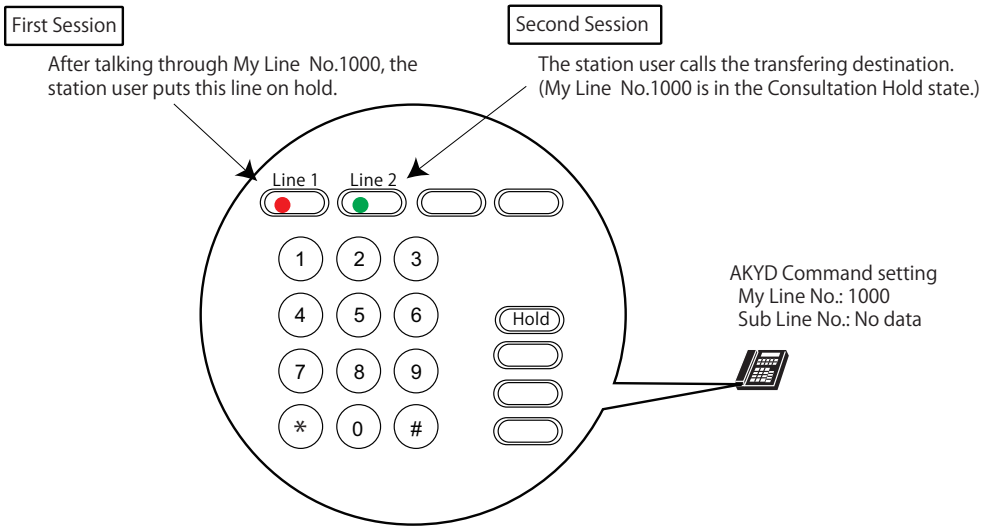
When SIP Handler Controlled SIP terminal allows multiple sessions of calls, the Standard SIP terminal can work on Single Line Mode or Multi Line Mode. Either mode can be selected with the AUACL/AUACN command.

Note: Multi Line Mode is only available for guest station in the Hotel system.

- **Single Line Mode**

  SIP Handler controls the Standard SIP terminal only using My Line. When a Standard SIP terminal user talks with a destination station and puts the call on hold in the first session, and then the Standard SIP terminal user places another call in the second session, the second session is processed as an event of Call Transfer operation.

Example 1: When the station is engaged in a call on the Line 1, the station puts the line on hold, and then the station places another call from the Line 2.

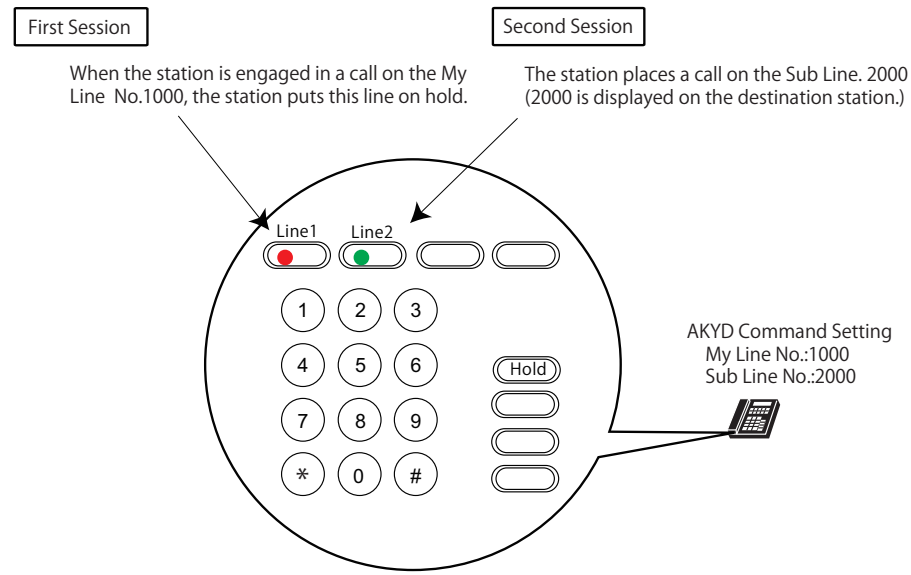


- **Multi Line Mode**

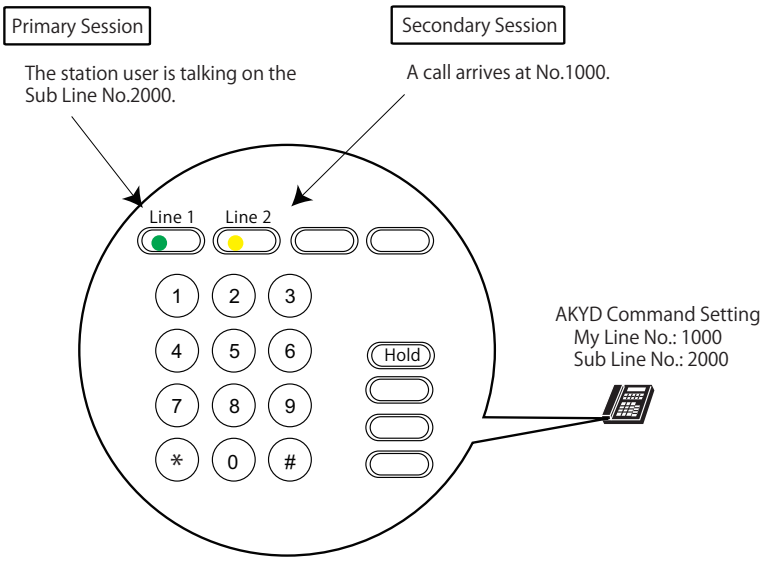
A Standard SIP terminal can accommodate the lines of the other Standard SIP terminals or Desktop terminals as Sub Line. On Multi Line Mode, multiple sessions are in progress, the SIP Handler controls the Standard SIP terminal by binding each session and line state. When a Standard SIP terminal user puts an ongoing line (the first session) on hold, another call arrival at a different line is treated as the second session which is different from the ongoing call in the first session.

Multi Line Mode also refers to a state in which the Standard SIP terminal performs registration through a Telephone Number and performs multiple sessions. The following cases are not included: A Standard SIP terminal performs multiple sessions by registrations through multiple Telephone Numbers.

Example 1: When the station is engaged in a call on the Line 1, the station puts the line on hold, and then the station places another call from the Line 2.



Example 2: While the station user is answering a call on the Sub Line No. 2000, another call arrives at the My Line No.1000. (From Example 1, My Line key and Sub Line key are switched.)



Operating Procedure

None

Service Conditions

<Important Notices>

1. The following typical SIP services are not supported:
   - Presence
   - Instant Message
2. A maximum of 4000 Standard SIP terminals are controllable per system.

<Interactions with Other Features>

1. SR-MGC is not supported.
2. RTP cannot be encrypted.
3. SIP messages can use UDP or TCP.

   Note: TCP is available from FP95-114 V4.

4. Control signal (SIP messages) can be encrypted. For the detail of SIP messages encryption, refer to SIP Handler - SIP Terminal Support for TLS Encryption [S-169].

   Note: Encryption of Control signal (SIP messages) is supported since FP95-114 V4.

5. In case of using TCP for communication with the Internal SIP Handler, the following applies:
   a.) Besides SIP messages, TCP is used for the session information exchange of terminals (proprietary protocol) (**Note 4**) as used in VIDEO CALL [V-31] and IP CONTROL-FAX [I-44].
   The maximum number of simultaneous connections of both SIP messages and session information exchanges is 1000 connections (including two or more connections between the same devices). When 1000 connections are established and a new connection is going to be established, the oldest connection is disconnected and the new connection is established.
   b.) A TCP connection is disconnected if the following time passes without packets being sent or received.
   - SIP Signaling (client connection): 185 seconds
   - SIP Signaling (server connection): 190 seconds
   - Session information exchanges (client connection): 25 seconds
   - Session information exchanges (server connection): 30 seconds

   Note: Depending of the condition “a.)” a connection may be disconnected before the maximum time of inactivity.

   c.) The only Protocol Handler that can use TCP is the Internal SIP Handler. Other PHs as SP-PHD, SP-PHI, SP controller, etc cannot use TCP.
   d.) When TCP is used for SIP signaling, set the terminal to TCP use before performing a REGISTER.

e.) If the transport protocol is defined in the REGISTER, that protocol is used regardless of the setting of AUACN/L (CDN:22).

Note: Transport protocol as defined in the transport parameter of the Contact header of the REGISTER request.

f.) For a CPU using dual configuration, the TCP connections are disconnected when there is a system changeover. However, since the connections are reconnected where needed after the system changeover, call control is not affected.

g.) When using TCP, make sure that all the terminals employed support TCP. Refer to the manual of each terminal for the TCP settings.

Note 4: Session information exchange of terminals (proprietary protocol) means the media information exchange (SDP information) between the terminals and the SIP Handler on the Telephony Server when using VIDEO CALL [V-31] and IP Control-FAX [I-44].

6. T.38 Fax Communication is supported from FP95-114 V4. See IP CONTROL-FAX [I-44] for an explanation.
7. The following are the conditions when a SIP Handler Controlled SIP terminal performs registration to the Telephony Server which is equipped with SIP Handler.
   a.) Log in to the terminal with a Telephone Number. Station Number cannot be used to log in to the terminal.
   b.) The terminal cannot be logged in with MAC Authentication Mode.
   c.) Login Password is used as the password for Digest Authentication during REGISTER process.
   d.) Assign Login Password with the ASPW command.
   e.) A maximum of ten digits can be entered for Login Password. A password can contain 0-9, * and #.
   f.) A SIP Handler Controlled SIP terminal cannot perform registration to the system with a station number which has already been used by another terminal such as an IP terminal, a SIP Multiline terminal or another SIP Handler Controlled SIP terminal.
   The SIP Handler Controlled SIP terminal overrides the registration of the existing terminal using the same station number and then the overridden terminal becomes inoperative. The SIP Handler Controlled SIP terminal can be overridden by the other terminal, but the SIP Handler Controlled SIP terminal overrides the terminal again after the periodical REGISTER information update.
   g.) Do not set a password from the SIP Handler Controlled SIP terminal using the access code.
   h.) Data assignment of a login password is in common with that of an IP terminal or a SIP Multiline terminal. To enable a SIP Handler Controlled SIP terminal to log in to the system without changing the login station number after the IP terminal or the SIP Multiline terminal modifies the login password with an access code, use the modified password.
8. When an IP address of a Standard SIP terminal is changed through DHCP, the Telephony Server treats the REGISTER with the changed IP address as Overriding.
9. In the following conditions, the SIP Handler Controlled SIP terminal goes into a logout state in which the terminal state cannot be displayed in the IPAN or DISD command window. In the logout state, the SIP Handler Controlled SIP terminal cannot place or receive calls until it performs registration to the system.
   - The Telephony Server does not receive REGISTER for updating Expire from the SIP Handler Controlled SIP terminal within the time of register expires (**Note 5**).
   - The Telephony Server receives Expire=0 as Expiration of REGISTER from the SIP Handler Controlled SIP terminal.

- The SIP Handler Controlled SIP terminal is reset with the DISD command **Note 6**
- The SIP Handler is initialized with the ADPM command **Note 6**
- The system is initialized with the SINZ command **Note 6**

Note: System Message 33-S: IP PKG Failure is not output in any of above conditions.

Note 5: 3600 seconds is set for the default. This data can be changed with the Register Expires parameter in the AUACL/AUACN command.

Note 6: To go into the login state, a Standard SIP terminal needs to perform registration. The Standard SIP terminal performs registration at the timing of its REGISTER update. So the Standard SIP terminal cannot be used until the REGISTER update after reset or initialization is executed by using the command.

10. Wireless Standard SIP terminal cannot be used for Out of Zone features such as Call Forwarding PS Incoming Call Incomplete [C-155] and Automatic Announcement - Called PS Out of Zone [A-129].
11. When a Standard SIP terminal puts a call on hold while receiving a Special Dial Tone (SPDT) or a Busy Tone (BT), the state in which a call normally cannot be held, the call turns into a hold state. After cancelling the call hold state, the terminal receives the tone again.
12. When a Standard SIP terminal puts a call on hold while receiving a Special Dial Tone (SPDT) or a Busy Tone (BT), the terminal, depending on its type, may go into a talking state such as displaying the duration of the call.
13. The following are conditions of Caller ID and Name Display when a Standard SIP terminal receives a call.
    a.) In the INVITE request message, Name Display information is given in the display-name field of the From header.
    b.) When there is no calling party number information, “Anonymous” is given as the calling party number information.
    c.) For a standard SIP terminal which supports RFC3325 and RFC5876, the connected party name and number can be displayed with the setting of the “Notice of Connected Number” parameter of the AUACL/AUACN command. For other terminals, when the connected party is changed by the transferring operation or shifting from a three-way call into a two-way call due to the disconnect of one party, the destination name and number cannot be shown.
    d.) Only name display information containing symbols and alphanumeric characters can be displayed.

    Note: To enable calling name and number display, the Name Display parameter setting is required through the AUACL/AUACN command.

14. When a SIP Handler Controlled SIP terminal forwards a call and the Call Forwarded destination does not answer yet, including while hearing a Busy Tone (BT) from the IP-DTG or a Ringback Tone (ROT), the LCD on the Standard SIP terminal may become a talking state such as displaying the duration of the call.
15. SIP Handler Controlled SIP terminals support Three-Way Calling features using CFT or VS32 but they cannot receive Three-Way Calling Tones (Add-On Ticker) for Three-Way Calling features.

    Standard SIP terminals do not support Three-Way Calling features which are established using the audio mixing function of Standard SIP terminals.

16. The following are conditions for a call on hold.
    a.) When a Standard SIP terminal puts a call on hold, the system behaves the same as pressing the HOLD key on a Desktop terminal.
    b.) An incoming call to a SIP Handler Controlled SIP terminal can be answered by a Desktop terminal when it accommodates a line of the Standard SIP terminal as Multiple Line, but the Standard SIP terminal cannot answer a call held by the Desktop terminal even if the call is an incoming call to My Line of the Standard SIP terminal.
    c.) When a SIP Handler Controlled SIP terminal does not answer a call on hold and the destination side recall timer expires, the Standard SIP terminal can be recalled automatically with the following system data assignment (the data assignment is in common with that of a Desktop terminal).
        - ASYD, SYS1, Index 144: Automatic Hold Recall Timer
        - ASYD, SYS2, Index 12 Bit 0=0: Automatic Recall Ringer - Non-Exclusive Hold is required
    d.) Do not set the Hold Fixation Mode (ASFC, SFI 254) to Allowed (RES=1). If it is allowed, Standard SIP terminals cannot answer calls after putting them on hold. Keep the Hold Fixation Mode restricted (RES=0).
    e.) When a SIP Handler Controlled SIP terminal releases a call after it puts the call on hold, the destination station does not call back the SIP Handler Controlled SIP terminal. The destination stations is released.
17. When a destination station releases a call while talking, the SIP Handler Controlled SIP terminal receives a Reorder Tone (ROT) generated by IP-DTG. After a certain period of time (**Note 7**), the SIP Handler Controlled SIP terminal is released or the receiving tone changes to a Busy Tone (BT) generated by the local side.

    Note 7: The period of time is based on Miscellaneous Timer Counter set by ASYD, SYS1, Index 401, Bit 0-3 (the data assignment is in common with that of Desktop terminal). A Standard SIP terminal does not depend on the value set by ASFC, SFI 136: Automatic Idle Return.

18. The IP-DTG cannot be used in the following reasons;
    - There is no data assignment for IP-DTG.
    - The maximum IP-DTG concurrent connections is exceeded.
    - Registration of IP-DTG is not completed.
    - IP-DTG tones are disabled by the AUACL/AUACN command.

Standard SIP terminals receive the following tones;

IP-DTG Tones	When IP-DTG Cannot Be Used
• Special Dial Tone (SPDT) • Service Set Tone (SST)	Standard SIP terminals receive no tone.
• Ring the destination station • the destination station releases the call • the destination puts the call on hold	Standard SIP terminals receive tones generated by the local side.

Behaviors when the destination station releases the call depends on the terminal specification.

19. The following table shows conditions for Call Hold/Call Transfer/Three Way Call.

<table>
<tr><th colspan="2" rowspan="2">Feature/Terminal Behavior</th><th colspan="2">In the Consultation Hold state, Second Call is on hold/Intermediate Station is not released (AUACL/N: Hold 2nd Call = 0)</th><th colspan="2">In the Consultation Hold state, Second Call is on hold/Intermediate Station is released (AUACL/N: Hold 2nd Call = 1)</th></tr>
<tr><th>Consultation Hold Release is disabled (AKYD: No DISCONNECT Feature key is set).</th><th>Consultation Hold Release is enabled (AKYD: DISCONNECT Feature key is set).</th><th>Consultation Hold Release is disabled (AKYD: No DISCONNECT Feature key is set).</th><th>Consultation Hold Release is enabled (AKYD: DISCONNECT Feature key is set).</th></tr>
<tr><td colspan="2">Attempting to put a call on hold even if it is unavailable.</td><td colspan="2">• A call is assumed to be on hold.
• Available to response the call again.
• Receiving RTP is continued.
Note 8</td><td colspan="2">• A call is assumed to be on hold.
• Available to response the call again.
• Receiving RTP is continued.
Note 8</td></tr>
<tr><td colspan="2">When the first call is on hold, put the ongoing second call on hold.</td><td>• A call is assumed to be on hold.
• Destination of the second call receives no tone or RTP warning tone.
Note 8</td><td>• The second call is released.
• The first call is on hold (Due to responding the first call, behavior is equivalent of the Consultation Hold Release state)</td><td colspan="2">Call Transfer - All Calls [C-11] / Consultation Hold - All Calls [C-17]</td></tr>
<tr><td rowspan="3">Call Transfer - All Calls [C-11] / Consultation Hold - All Calls [C-17]</td><td>Through the first call being on hold, BYE (Disconnection) message is received.</td><td colspan="2">Call Transfer - All Calls [C-11] / Consultation Hold - All Calls [C-17]</td><td colspan="2">Call Transfer - All Calls [C-11] / Consultation Hold - All Calls [C-17]</td></tr>
<tr><td>While talking through the second call, BYE (Disconnection) message is received.</td><td>Call Transfer - All Calls [C-11] / Consultation Hold - All Calls [C-17]</td><td>• The second call is released.
• The first call is on hold (Behavior is equivalent of the Consultation Hold Release state due to responding the first call)</td><td>Call Transfer - All Calls [C-11] / Consultation Hold - All Calls [C-17]</td><td>• The second call is released.
• The first call is on hold (Behavior is equivalent of the Consultation Hold Release state due to responding the first call)</td></tr>
<tr><td>While calling through the second call, REFER message is received.</td><td colspan="2">Call Transfer - All Calls [C-11] / Consultation Hold - All Calls [C-17]</td><td colspan="2">Call Transfer - All Calls [C-11] / Consultation Hold - All Calls [C-17]</td></tr>
<tr><td rowspan="2">Blind Transfer To Station [B-18] / Blind Transfer To Attendant [B-17]</td><td>While calling the second call, BYE (Disconnection) message is received through the first call being on hold.</td><td colspan="2">Blind Transfer To Station [B-18]
Blind Transfer To Attendant [B-17]</td><td colspan="2">Blind Transfer To Station [B-18]
Blind Transfer To Attendant [B-17]</td></tr>
<tr><td>After putting the first call on hold, REFER message is received.</td><td colspan="2">Blind Transfer To Station [B-18]
Blind Transfer To Attendant [B-17]</td><td colspan="2">Blind Transfer To Station [B-18]
Blind Transfer To Attendant [B-17]</td></tr>
</table>

Feature/Terminal Behavior		In the Consultation Hold state, Second Call is on hold/Intermediate Station is not released (AUACL/N: Hold 2nd Call = 0)		In the Consultation Hold state, Second Call is on hold/Intermediate Station is released (AUACL/N: Hold 2nd Call = 1)	
		Consultation Hold Release is disabled (AKYD: No DISCONNECT Feature key is set).	Consultation Hold Release is enabled (AKYD: DISCONNECT Feature key is set).	Consultation Hold Release is disabled (AKYD: No DISCONNECT Feature key is set).	Consultation Hold Release is enabled (AKYD: DISCONNECT Feature key is set).
Blind Transfer To Station [B-18] / Blind Transfer To Attendant [B-17]	While calling the second call, REFER message is received	Blind Transfer To Station [B-18] Blind Transfer To Attendant [B-17]		Blind Transfer To Station [B-18] Blind Transfer To Attendant [B-17]	
Three Way Calling with CFT trunk/VS32		Three Way Calling		Three Way Calling	
Three Way Calling (Failed)	The second call is released. The first call is ongoing. Seizing Conference trunks (CFT/VS32) is failed.	• The second call is on hold. • The first call is ongoing. (Behavior is equivalent of switching the destination during the Consultation Hold state. The state in the terminal side may not be matched)	• The second call is released. • The first call is ongoing. (Behavior is equivalent of switching the destination during the Consultation Hold Release)	• The second call is on hold. • The first call is ongoing. (Behavior is equivalent of switching the destination during the Consultation Hold state. The state in the terminal side may not be matched)	• The second call is released. • The first call is ongoing. (Behavior is equivalent of switching the destination during the Consultation Hold Release)

Note 8: When CDN9: Reject Unacceptable reINVITE=1 (Reject (488)) is assigned with the AUACL/AUACN command, the state goes into:
Send 488 response, No state transition. Sending/Receiving RTP is continued.

20. Connected party number and Name display information (including Guest name of Hotel Service) can be provided for a standard SIP terminal which supports RFC3325 and RFC5876. The conditions for the notification of Destination number information are as follows:

    a.) The call number and name of the connected party are set in the P-Asserted-Identity header and sent to a standard SIP terminal. The following table shows the timing for the notification of connected party information.

Terminal Operation/ Events	Server Provides Connected Party Information When	Frequency
• Outgoing call from a standard SIP terminal • Hold/transfer and other services operation	• 18x Provisional response (180 Ringing/183 Session Progress etc.) is sent for initial INVITE sent from a standard SIP terminal	Each time (Resends the 18x response to the standard SIP terminal and updates the connected party information when the call is connected to another terminal before its destination answers.)
	• 200 OK Successful response is sent for INVITE/re-INVITE sent from a standard SIP terminal	Each time
	• UPDATE request is sent after sending 200 OK Successful response for INVITE request sent from a standard SIP terminal	When the connected terminal is changed

Terminal Operation/ Events	Server Provides Connected Party Information When	Frequency
• Incoming call to a standard SIP terminal • The connected terminal is changed by hold/transfer and other services operation	• Initial INVITE is sent to standard SIP terminal	Each time
	• UPDATE request after receiving 200 OK/18x Successful response is sent for INVITE request sent to standard SIP terminal	When the connected terminal is changed
	• reINVITE request is sent to standard SIP terminal	Each time

b.) When a call is placed or transferred to a standard SIP terminal, if there is no calling party number information, "Anonymous" is given as the calling party number information.

c.) Only Name display information containing symbols and alphanumeric characters can be notified.

d.) When a call is placed from a standard SIP terminal to a destination which is set to transfer calls to another terminal, Number information and Name display information on the LCD of the standard SIP terminal is not changed until it hears RBT.

e.) When a standard SIP terminal user is transferred to a terminal which denies UPDATE request in INVITE session, the connected party information is not available to the user. In this case, the connected party number and name on the LCD are not updated.

f.) A change in the state of a call affects the connected party information displayed on the LCD.

1) The information about the last party you talked with is displayed on the LCD of your standard SIP terminal when:

• Another party is added to the call, establishing a three-way call.

• You are connected to Attendant console, Announcement Trunk, Analog C.O. line, or Analog Tie Line and ACIS, while ringing or talking on the phone.

2) The information about the opposite party you are talking is displayed on the LCD of your standard SIP terminal when one of the parties in a three-way call is disconnected.

g.) When a caller is blind-transferred to a standard SIP terminal, caller number and name information are displayed when the intermediate party releases the call if ASYDL, SYS1, Index 1382, Bit 5 is set to 1. If Bit 5 is set to 0, the information is displayed when the standard SIP terminal answers the call. (data setting available from FP95-114 V4. Prior to that, the information is displayed only when the standard SIP terminal answer the call.)

Note: To provide Destination number information, Notice of Connected Number parameter setting is required through the AUACL/AUACN command.

Note: To notify the connected party of Name display information, Name Display parameter setting is required through the AUACL/AUACN command.

Multi Line Mode (only for guest station in the Hotel system)

Feature/Terminal Behavior		Sending 200OK (AUACL/N command: Reject Unacceptable reINVITE=0)
Attempting to put a call on hold even if it is unavailable.		• A call is assumed to be on hold. • Available to response the call again. • Receiving RTP is continued. **Note 9**
While the first call being on hold, putting the second call on hold.		Both of the first call and the second call are on hold.
Call Transfer - All Calls [C-11] / Consultation Hold - All Calls [C-17]	Through the first call being on hold, BYE (Disconnection) message is received.	All calls are released.
	While talking through the second call, BYE (Disconnection) message is received.	• The second call is released. • The first call is on hold. (Behavior is equivalent of Consultation Hold Release due to the first call is answered)
	While calling through the second call, REFER message is received.	Send 488 response to REFER message. Subordinate behaviors are depending on each terminal specification. (When the terminal sends a CANCEL message to the second call and then answers the first call, the behavior is equivalent of Consultation Hold Release)
Blind Transfer To Station [B-18] / Blind Transfer To Attendant [B-17]	While calling the second call, BYE (Disconnection) message is received through the first call being on hold.	All calls are released.
	After putting the first call on hold, REFER message is received.	Blind Transfer To Station [B-18] / Blind Transfer To Attendant [B-17]
Blind Transfer To Station [B-18] / Blind Transfer To Attendant [B-17] (REFER method)	While calling the second call, REFER message is received	Send 488 response to REFER message. Subordinate behaviors are depending on each terminal specification. (When the terminal sends a CANCEL message to the second call and then answers the first call, the behavior is equivalent of Consultation Hold Release)
Three Way Calling	With CFT trunk/VS32	• If Call Hold Conference set with the AKYD command, Three Way Calling is in service. • If not, the second call is released and the first call is ongoing. (Behavior is equivalent of Consultation Hold Release)
Three Way Calling (Failed)	The second call is released. The first call is ongoing. Seizing Conference trunks (CFT/VS32) is failed.	• If Call Hold Conference set with the AKYD command, the first call is ongoing. At the same time, the second call is released and the destination of second call calls back. • If not, the second call is released and the first call is ongoing. (Behavior is equivalent of Consultation Hold Release)

Note 9: When CDN9: Reject Unacceptable reINVITE=1 (Reject (488)) is assigned with the AUACL/AUACN command, the state goes into:
Send 488 response, No state transition. Sending/Receiving RTP is continued.

21. To use Busy Line features, it is necessary for the Standard SIP terminal to receive a Busy Tone (BT) from IP-DTG.
22. When a SIP Handler Controlled SIP terminal in Single Line mode performs CALL TRANSFER - All CALLS [C-11] and the connection with the transfer destination is released by the following operations, the SIP Handler Controlled SIP terminal (intermediate station) cannot automatically return to the original call with the first party (transferred party):

    - the transfer destination answers the transferred call and then releases the call
    - Call Transfer is canceled by the SIP Handler Controlled SIP terminal while the transfer destination is ringing

    In this case, the first party is remains on hold. The SIP Handler Controlled SIP terminal (intermediate station) can return to the call with the first party by canceling the hold state. (Depending on the type of the SIP Handler Controlled SIP terminal, the call hold state may be canceled automatically.)

23. When calling to an external line where parameter 2nd DT is set to other than 0 with the ASPA/ASPAL/ASPAN command, the dial tone (DT) will be heard momentarily. To avoid this, execute the AUACL/AUACN command as follows:

    - When 2nd DT is set to 1: Execute the AUACL/AUACN command with the parameter CDN12 TONE[0] bit 7=0 (Disable SDT).

    Note: If changing the setting from the default value, change to 7E from FE.

24. When the SIP Handler Controlled SIP terminal on the system assigned one digit dialing instead of Switch Hook Flash (ASYD, SYS1, Index 70, Bit 4=1) is used, it is available to use only while the terminal is receiving a busy tone. To use this feature, the setting for receiving digits operation (ASYDL, SYS1, Index 1382, Bit 0=1) is required.

    Note: The one digit dialing feature is not available while talking on the SIP Handler Controlled SIP terminal.

<Hardware/Software>

1. The following table shows conditions and notification method of available DTMF signals.

DTMF Signal Notification Method	Conditions
SIP INFO Method	1. DTMF signals can be sent to equipment connecting to the Telephony Server except for the following: • Standard SIP terminals controlled by SP CONTROLLED SIP TERMINAL [S-168] • SIP Handler Controlled SIP terminals which do not support SIP-INFO method. 2. Operations of various features provided by the Telephony Server are available. 3. 0-9, * and # of DTMF signals can be made transparent (A-D, Flash cannot be made). 4. When DTMF signals in NEC Proprietary Protocol Mode are received from a destination, DTMF signals are converted with the SIP-INFO method as shown below and sent to SIP Handler Controlled SIP terminals. 0 - 9 → "0"- "9" * → "10" # → "11" **Note:** * and # can be changed by the AUACL/AUACN command data assignment. * → "*" # → "#" 5. When DTMF signals in NEC Proprietary Protocol Mode are received from a destination, DTMF signals are converted as shown below and sent to SIP Handler Controlled SIP terminals. "0" - "9" → 0 - 9 "10" → * "*" → * "11" → # "#" → #
RFC2833	1. DTMF transparency is available only between terminals corresponding to RFC2833. 2. Telephony Server services are available by using IP-DTG.
In-band Signaling	1. DTMF transparency is available between all IP equipment. 2. Telephony Server services are unavailable.

Note: To enable Telephony Sever services, using SIP INFO Method or RFC2833 is needed. Confirm the specifications of your Standard SIP terminal and specify DTMF Signal Notification Method on the terminal side data setting if required (SIP INFO Method is recommended due to equipment which allows sending and receiving DTMF and conditions for operating features). At this time, use only one DTMF Signal Notification Method. Do not mix up several DTMF Signal Notification Methods. If multiple types of methods are sent, it may cause malfunction.

2. For an IP equipment/Control Module that performs calls with a SIP Handler Controlled SIP terminal, use the latest firmware (or more specifically, software) available. With old firmware (IP equipment with its firmware not updated as in use of the SV7000/NEAX2400), audio connections may not be possible. In case of trying to connect IP equipment that has an old firmware with a SIP Handler Controlled SIP terminal, the call will be released at the start of the Call Time with the IP equipment. IP equipment whose firmware does not allow connectivity, the firmware revisions that do are as listed below.

Equipment Type	No.	Equipment	Sales Name	Firmware	Issue
Terminal	1	Dterm Series i with IP adapter	IP-R UNIT		2.01
	2	DtermIP	ITR-32D-1x		2.00
	3		ITR-8D-1x		2.06
	4	DtermIP INA-SET	ITR-LC-1x		1.10
	5	DtermSP30	DTERM SP30 CE	SC-3299 DTERM SP30-R4	4.0.0.0
	6		DTERM SP30 AE	SC-3196 DTERM SP30	2.1.0.0
MC/MG	1	Analog MC **Note 10**	SN8029 MC2A-A	SP-3629 2AMC PROG-A	Unsupported
	2		SN8029 MC2A-B	SP-3753 2AMC PROG-B	1A
	3		SN8029 MC2A-C	SP-3753 2AMC PROG-B	25A
	4	MG(BRI)	MG-2BRI-AA	SP-3709 BRI MG BSC PROG-A	1
	5	MG(PRI) **Note 10**	SN8023 MG2M-xx	SW-437 PRI MG PROGRAM-B	2B
	6		SN8032 MG15M-xx	SW-423 PRI MG PROGRAM-A	2B
IP-PAD	1	IP-PAD Card	SPA-32IPPAD	SW-425 IPPAD PROGRAM-A	7
	2		SPA-32IPPAD-B	SW-428 IPPAD PROGRAM-B	8
	3		SPA-32IPPAD-C	SW-435 IPPAD PROGRAM-C	5
	4		SPA-32IPPAD-D	SW-439 IPPAD PROGRAM-D	1

Note 10: UG50 is not included.

3. When a call is placed from a SIP Handler Controlled SIP terminal, the information given in the display-name field of the From header cannot be used as the Name Display information.
4. Controlling SIP Handler Controlled SIP terminal is unavailable through SIP Proxy Server.
5. A SIP Handler Controlled SIP terminal corresponding to session timer (RFC4028) cannot make a Call Retention if it is not able to communicate with the Telephony Server during the call. The call will be released from the terminal side with its session timer monitor.

6. In case of using MWI (Message Waiting Lamp control (RFC3842)), the expire value SUBSCRIBE that notifies the SIP Handler Controlled SIP terminal to the Telephony Server will need more than 120 seconds. If needed, setup the terminal.
7. When you lift the handset (before originating the call) of the SIP Handler Controlled SIP terminal, the terminal state enters into an idle mode in the Telephony Server side. In such mode, its behaviors when an incoming call arrives at a SIP Handler Controlled Standard terminal are determined by the specifications of the terminal.
8. In the case of a Standard SIP terminal with a local call hold tone, when the audio connections are switched, the local call hold tone will be audible for a moment.

<Interactions with Networks>

SIP Handler Controlled Standard SIP terminals cannot be used in a network where Peer-to-Peer CCIS and DTI-FCCS coexist and each node is connected in tandem.

<Conditions Specific to This Feature>

1. SIP Handler Controlled SIP terminals can receive tones through the IP-DTG without using external devices.

   Note: The IP-DTG does not support short tones (Call Waiting tone).

2. SP Controlled SIP terminals and SIP Handler Controlled SIP terminals can be mixed in a system.
3. The SIP Handler uses LAN1 IP address2 (ACT) by default (**Note 11**); however, you can implement Standard SIP terminals without setting the IP Address2 (ACT) if you use the ADTM command to configure SIP Handler's LAN interface to use IP address (ACT).
   Despite this, it is recommended using IP Address2 (ACT) for this feature.

   Note 11: If the settings of Internal SIP Handler's LAN port, Application port or IP address2 (ACT), are changed with the ADTM command while the Internal SIP Handler is in operation, TCP/IP Module must be initialized. (In the case of IP address2 (ACT), TCP/IP Module is automatically initialized.)
   After initialization of TCP/IP Module is completed, use the ADPM command to initialize the Internal SIP Handler.

   Note: If a SIP Handler Controlled SIP terminal uses LAN1 IP Address2 (ACT), it cannot share SIP Server port used for sending or receiving SIP messages with a SP Controlled Standard SIP terminal. Destination Telephony Server port number needs to be changed to a port number that differs from 5060 (for example, 55060) in the SIP Handler Controlled SIP terminal side.

4. A maximum of 16 digits can be used as a Station Number (Telephone Number).
5. To use the SIP terminal, the client license for the SIP terminal is required. The number of client licenses is shared by SP Controlled and SIP Handler Controlled SIP terminals. When the system accommodates both types of SIP terminals, licenses are required for the total sum of the SIP terminals.
6. SIP terminals cannot use characters to select a destination for dialing (0-9, #, * only).
7. DTMF tones may not be sent, depending on the combination of terminal.

8. When DTMF is sent from the terminal as voice data, a call using compressed codec except for G.711 is not guaranteed.
9. Each Station Number (or Telephone Number) is set as the default password of the SIP terminal in the system. The password can be modified with the ASPW command. The RDPD command can be also used to delete the password. After modification of the password, modify the password in the SIP terminal configuration data and perform the registration to the system such as turning the SIP terminal OFF/ON.
10. A SIP terminal can accommodate other Desktop terminal's line as Sub Line.
11. Desired values of Type of Service (ToS) can be set to SIP messages sent from the system with the ALOCL command. The ToS values apply to LOC-ID=0 (System).
12. The registration status of SIP Handler Controlled SIP terminals can be checked through the DISD command. Also registration information of a specified SIP Handler Controlled SIP terminal can be cleared but soft RST packets cannot be sent to the SIP terminals in the registration state. For more information about the DISD command, refer to Operations and Maintenance Manual.
13. During the CPU changeover, ongoing calls are maintained and Telephony Server services are available.
14. Standard SIP terminals can be accommodated in Virtual/Physical PIRs.
15. The following are conditions for SIP Handler Controlled SIP terminals in Single Line Mode.

    (a) The SIP Handler controls Standard SIP terminals using only the My Line. When a Standard SIP terminal user talks with a destination station and puts the call on hold in the first session, and then the Standard SIP terminal user places another call in the second session, the second session is processed as an event of Call Transfer operation.

    (b) Even if another station's line is set to Login Telephone Number of a Standard SIP terminal as Sub Line using the AKYD command, placing and receiving a call are unavailable through Sub Line.

    (c) Another Standard SIP terminal or a Desktop terminal can accommodate My Line of a Standard SIP terminal in Single Line Mode as Sub Line.
16. The following are conditions for Standard SIP terminals in Multiline Mode.

    (a) Set the Sub Line data to the Login Telephone Number of Standard SIP terminal using the AKYD command. When Sub Line data is not set with the AKYD command, the Standard SIP terminal behaves as a terminal in Single Line Mode even if Multiline Mode is specified with the AUACL/AUACN command.

    (b) A secondary line used for placing or receiving a call cannot be selected. My Line takes priority to be seized when a call is placed. Incoming calls give the appearance that they arrive at the same line on the Standard SIP terminal regardless of arriving at My Line or Sub Line.

    (c) An idle line on the Standard SIP terminal is selected automatically and the station number of the seized line is displayed on the destination station in the following cases:

    - After talking with a destination station, the Standard SIP terminal user puts the call on hold in the first session, and then places another call in the second session.
    - When the Standard SIP terminal user places a call while a line is used by the other terminal due to Multiple Line Operation.

    (d) Up to two lines including My Line can be controlled by the Telephony Server. It is available to use the lowest number of line assigned for the KYN parameter of the AKYD command except for the My Line. The Standard SIP terminal needs to support concurrent multiple sessions.

(e) Multiple Line Mode enables a Standard SIP terminal to accept multiple calls. Single Line Mode enables Standard SIP terminals to use more features than Multiple Line Mode.

(f) To replace a SSG released in the North American market, select Multiple Line Mode to make the Standard SIP terminals compatible.

(g) In case of a conference that uses Call Hold Conference processing, when the 1st call is on hold and its calling timer goes out, this 1st call, which is called without being able to carry out any conference action, becomes the 2nd call which is maintaining the conversation. Because of that, to carry out a conference operation, after putting the 1st call on hold, it is necessary to carry out such conference operation within the time frame specified with the Automatic Hold Recall Timer (ASYD, SYS1, Index 144) for the Desktop terminal (only for guest station in the Hotel system).

Note: When the Standard SIP terminal does not support concurrent multiple sessions, set the Standard SIP terminal in Single Line Mode (default setting).

17. Telephony Server does not check the domain name given by a Standard SIP terminal. Set any domain name to the Standard SIP terminal (normally, set the IP Address for the destination of registration).
18. The conditions of incoming calls to a SIP Handler Controlled SIP terminal are as follows:

(a) In case of an incoming call to a SIP Handler Controlled SIP terminal that has not been registered at the Telephony Server, the caller hears a Ringback Tone (RBT).

(b) If the communication between the Telephony Server and the SIP Handler Controlled SIP terminal is lost while receiving a call, the caller hears a Ringback Tone (RBT) continuously.

(c) If the SIP Handler Controlled SIP terminal continues on a receiving call state for 3 minutes and during that time the message for updating the receiving call state (180 Ringing) is not notified from the SIP Handler Controlled SIP terminal, the incoming call is canceled. However, the caller hears a Ringback Tone (RBT) continuously.

(d) Even if an error response is sent from a SIP Handler Controlled SIP terminal, the caller hears a Ringback Tone (RBT) continuously.

(e) If a SIP Handler Controlled SIP terminal is released while receiving a call, it will not receive the incoming call again after the releasing. The caller can make an incoming call to the SIP Handler Controlled SIP terminal by releasing and sending again the call. While the caller continues calling, callings from the SIP Handler Controlled SIP terminal will be restricted.

Note: If a SIP Handler Controlled SIP terminal is released while receiving a call, the following may have occurred:

- A SIP Handler Controlled SIP terminal is started with the POWER ON button or is restarted while receiving a call
- A SIP Handler Controlled SIP terminal is released after a temporary communication disconnection with the Telephony Server while receiving a call.
- If the incoming call is canceled based on the receiving call state not being updated (without a 180 Ringing notification).
- The SIP Handler Controlled SIP terminal sent back an Error Reply to the incoming call.

(f) When a caller releases a call while calling a SIP Handler Controlled SIP terminal, if the Telephony Server cannot receive a response from the SIP Handler Controlled SIP terminal due to communication failures, the SIP Handler Controlled SIP terminal rings continuously and does not become an idle state. However, when the response is not notified from the SIP Handler Controlled SIP terminal for 3 minutes or more, the receiving call state is canceled and then the SIP Handler Controlled SIP terminal becomes an idle state.

19. When a call is placed to a Standard SIP terminal which does not perform registration to the Telephony Server, the caller receives a Ringback Tone (RBT).
20. QoS data cannot be set to packets sent from a Standard SIP terminal such as SIP messages and RTP.
21. When a line allowing a Standard SIP terminal to use is accommodated to another terminal and in use, the Standard SIP terminal cannot place a call. The Standard SIP terminal is released or receives a Busy Tone (BT).
22. A call put on hold from the SIP Handler Controlled SIP terminal using the Call Hold operation cannot be retrieved by the Telephony Server.
    Because of that, when the SIP Handler Controlled SIP terminal puts the call on hold using the call hold operation (usually, the call hold operation is disabled) after the opposite party of the SIP Handler Controlled SIP terminal puts the call on hold, due to the AUACL/N Command (CDN9: Reject Unacceptable reINVITE=0: Not Reject (200OK)) data setting, the SIP Handler Controlled SIP terminal goes into an assumed on hold state. In such state, if the calling party tries to maintain a call using the switch hook flash operation, it will not be possible.
    The opposite party which performs the switch hook flash hears no tones, RTP Warning Tone or Reorder Tone (ROT). With the retrieval operation of the SIP Handler Controlled SIP terminal, it will be possible to maintain a call again.
23. The call will be released if all the payloads do not match (payload type supported by the SIP Handler Controlled SIP terminal, payload type supported by intended party and payload type specified with the ALOCL/AIVCL command).
24. Handover (the movement of a phone engaged in communication between WI-FI access points), which requires IP address modifications, is not supported.

<Conditions Specific to UMGi/RNIP System>

1. In an UMGi system, all the nodes can accommodate Standard SIP terminals.
2. In a RNIP system, from FP95-112 V2, only the CCN can accommodate Standard SIP terminals.
3. The maximum number of speech channels is 4000 channels per CCN/RN. As for the UMGi/RNIP system designed for high-traffic networks, the maximum number of speech channels is 3000 channels per node. The maximum number of the speech channels may not be used depending on the assignment conditions of other terminals' station data or trunk data.
4. The following data assignment is required for each node to which Login Telephone Number of Standard SIP terminals are assigned:
   SIP Handler, LAN1 IP Address 2 (ACT), and IP-DTG
5. As a Registration Server, assign the LAN1 IP Address 2 of the node which includes the LENS applicable to the Telephone Number used for the Standard SIP terminal's login.

<Conditions Specific to Video Support>

This enhancement is to enable Video calls only between SIP phones under same SIP handler.

Note: This conditions are for Video calls until FP95-113 V3. For Video calls from FP95-114 V4, see VIDEO CALL [V-31].

When voice path is connected, voice payload information is exchanged by a proprietary method with the existing SIP handler. However, Video media information is referred directly from the received SDP information from the opposite terminal under same SIP handler in the exchange SDP.

When AUACL/N-CDN3 (INVITE with SDP) is 1 (on), SV9500 inserts Video media information that originator offers on the dummy SDP of INVITE/reINVITE request.

1. Video codec cannot be setup/changed by Office data. Please refer to (6) about how Video code is selected.
2. Conditions for AUACL/N CDN3 (INVITE with SDP) setting
   a.) In case CDN3 is 0 (off):
   Terminal receiving a call cannot know if it is a video call since INVITE/reINVITE doesn't include video SDP.
   b.) In case CDN3 is 1 (On):
   Terminal receiving a call receives Video SDP of INVITE/reINVITE request.
   c.) When the originator is under the same SIP handler, the SV9500 inserts only payload information the SV9500 supports (Refer to (6)) in the incoming SDP among payload information that the originator offers. If the originator offers multiple video payload information, the SV9500 inserts multiple codecs and the received terminal selects one among them. So, terminal receiving a call can know if originator wants to have video or not.

   However, there is condition for CDN3 is ON.

   "C=" IP address field is set a dummy IP (=SV9500 IP) in the session description.
   Voice is G.711 only.

   Note: If AVICL sets any codec except G.711, when the terminal answers a call, the call is released.

   RTP port of Video media description is a dummy port like voice case.

   Note: It is not the port number that originator offers.

   d.) After the terminal answers a call, reINVITE is sent to change connection to correct destination from dummy IP/Port. If terminal changes RTP port number when it receives a call or when it sends in reINVITE, connection cannot be done. This is the same as Voice - Cannot accept port-number change within same INVITE session.
3. It is only possible to have video between SIP terminals under same SIP handler.

   Video is not supported between FCCS systems, between UMGi's nodes, between P2P-CCIS, between external trunks like MG-SIP, between SP-controlled SIP or between any combinations of these.
4. Supported video codecs are as follows:

   - H261
   - H263
   - H264

- MP4V-ES

If the terminal doesn't send any of the above codecs in the SDP offer, the SDP answer from the SV9500 won't include video payload. Video payload information exchange cannot be guaranteed.

5. Internal SIP handler uses the below method for the reply SDP answer after receiving SDP offer from terminal in both cases of incoming, outgoing call. So, Internal SIP handler picks one Video payload type & codec by below logic and sends it in the SDP answer to the originator & destination. (SIP handler picks actual payload type will be used)

   Note: In case AUACL/N CDN3 is 0. (In case CDN3 is 1, destination side picks codec)

   Payload type priority follows the order of the list of video payload ("m" line of video media) in SDP which originator offers.

   When re-connection by reINVITE, order of list of video payload that originator offers within reINVITE transaction will get priority.

   However, if the SV9500 doesn't know the originator or destination (NOTE), prioritize video codec using the list of the terminal that has higher extension-LENS location request.

   Note: By Recall, etc., both source and destination will become UAC transaction of initial INVITE.

6. Cannot control video payload bandwidth. Only voice bandwidth can be controlled.
7. During a call, media information change cannot be supported. If the terminal does, the Internal SIP Handler cannot support.

   However, if process is relegated to the terminal side, there won't be any issue.

   Example: Switch video call to Voice only call

   If the terminal just stops sending/receiving Video UDP, then it is okay without processing reINVITE or UPDATE.
8. C= connection IP, a= received mode (sendonly, etc) are set as session description above "m" line. These are NOT written under media description ("m" line).
9. Terminal proprietary media control information cannot be transparent.

   Content-Type: application/media_control+xml

   >> INFO message by Polycom-VVX1500
   Content-Type: application/media_control+xml cannot be transferred.

10. Video conferencing is not available without external Video conferencing equipment.

<Other Conditions>

1. To use Standard SIP-IVR in the ACD system, refer to Hot Split - ACD [H-31A] in the Data Programming Manual - ACD.
2. To use SIP Handler Controlled SIP terminals in the OAI system, refer to SIP Handler Controlled SIP terminal in OAI System in the Data Programming Manual - OAI.

Programming

Note: The following shows assignment for Station Data and Password for Digest Authentication. Assign the following data after the initial data assignment and IP-DTG data assignment.

- For the initial data assignment, refer to 2.1.3 “If using SIP Handler Controlled SIP Voice Terminal and SIP Video Terminal” in Chapter 6 System Startup - Additional Information on System Startup - Initial Data Assignment in Appliance Model Installation Manual.
- For the IP-DTG data assignment, refer to SOFTWARE DIGITAL TONE GENERATOR [S-166].

<Station Data Assignment>

To register SIP terminal data

Before this data assignment, complete the Numbering Plan Data assignment for Station Numbers and Telephone Numbers with the ANPD/ASPA/ASPAN commands.

STEP 1: **ASYDL** - Assign the following system data.

System Data 1

Index 803, Bit 1: 1 (Consultation Hold Release is in service.)

Index 880, Bit 5: 1 (Internal SIP Handler enabled.)

Index 1382, Bit 0: Receiving digits operation of the SIP Handler Controlled SIP terminal when a called station is busy (while receiving a Busy Tone (BT) generated by IP-DTG).

0=Step Call

1=Going switch hook flash (Receiving a Special Dial Tone (SPDT))

Bit 1: Digest Authentication when the SIP Handler Controlled SIP terminal performs registration

0=In Service (default)

1=Out of Service

Note: When Bit 1=1 (Out of Service) is selected, pressing any digit key while hearing a Busy Tone (BT) enables operations of features in the Busy state.

Bit 5: Change of the Number and Name display information while the SIP Handler Controlled SIP terminal is being called **Note 7**

0=Out of Service

1=In Service

Note 7: Note the following:

- This setting is for terminals that are set to 1 (Enable) in Notice of Connected Number of the AUACL/N command.
- When a caller is blind-transferred to a SIP Handler Controlled SIP Terminal, the timing for displaying the caller information can be set to the moment when the intermediate party is disconnected from the call by setting this data to 1. (In the case of setting to 0, the caller information is displayed when the call is answered by the SIP Handler Controlled SIP Terminal.)
- Available from FP95-114 V4.

Index 1394, Bit 0 and 1: Select transport protocol for session information exchange of SIP devices. **Note 8 Note 9**

Transport Protocol	Bit 0	Bit 1
AUTO **Note 10**	0	0
TCP **Note 11**	1	0
UDP	0	1

Note 8: If FCCS system is used, set these bits in the ASYDN to ensure that all nodes have the same settings.

Note 9: Regarding systems where there are session information exchanges between the SIP handlers of more than one Telephony Server, this setting is valid only for FCCS system and UMGi system. In other combinations, UDP is used.

Note 10: With this setting the packets are sent using UDP, but if the size exceeds 1400 bytes TCP is automatically used.

Note 11: If there is the risk of packet fragmentation when using UDP, set to TCP.

Index 1408, Bit 0: Use TCP for SIP Messages.
0=Disabled
1=Enabled

Note: If this setting is changed, Internal SIP Handler needs to be initialized with the ADPM, ADTM or SINZ command.

Note: Available since FP95-114 V4.

STEP 2: **AUACL/AUACN** - Assignment of the User Agent Class Data for LDM/NDM
Assign User Agent Classes and their class data corresponding to User Agents for the SIP Handler-controlled Standard SIP terminals.

STEP 3: **AISTL** - Assign IP Station data.
TN: Tenant Number
IP STN: IP Station Number (Maximum 5 digits)
KIND: DtermIP
TEC=12: Dterm
RSC: Route Restriction Class (0-15)
SFC: Service Feature Restriction Class (0-15)
LENS: Line Equipment Numbers

STEP 4: **AKYD** - Assign IP Station data for the SIP Handler Controlled SIP terminal.

To use a Standard SIP terminal assigned as Multi Line Mode through the AUACL/AUACN command (only for guest station in the Hotel system).

TN: Tenant Number
STN: Station Number (Maximum 5 digits)
PRI=0
PL TN: Prime Line Tenant Number
PL STN: Prime Line Station Number
S=0: OG from Prime Line is not restricted
MWD=0: MW data display on the top line
LN PRE=0: Standard data of Line Preference
TP=0: 4/8/16-button type Desktop terminal
KYN1
    KYI=2: Multi-line
    KD=0: Line
    TN: My Line Tenant Number
    STN: My Line Station Number
    RG: Ring Information for My Line
KYN2
    KYI=2: Multi-line
    KD=0: Line
    TN: Multi-line (such as Virtual Line) Tenant Number
    STN: Multi-line (such as Virtual Line) Station Number
    RG: Ring Information for Each Line

To use a Standard SIP terminal assigned as Single Line Mode through the AUACL/AUACN command

TN: Tenant Number
STN: Station Number (Maximum 5 digits)
PRI=0
PL TN: Prime Line Tenant Number
PL STN: Prime Line Station Number
S=0: OG from Prime Line is not restricted
MWD=0: MW data display on the top line
LN PRE=0: Standard data of Line Preference
TP=0: 4/8/16-button type Desktop terminal
KYN1
    KYI=2: Multi-line
    KD=0: Line
    TN: My Line Tenant Number
    STN: My Line Station Number
    RG: Ring Information for My Line
KYN2
    KYI=1: Function Key
    FKY=131: Disconnect Key

Note: To use a Standard SIP terminal in Multi Line Mode, the DISCONNECT Feature key is not required.

To use Attended Transfer and Three-Way Conference with a Standard SIP terminal assigned as Multiline Mode through the AUACL/AUACN command (only for guest station in the Hotel system).
KYN3
KYI=1: Function Key
FKY=116: Call Hold Conference
To use Call Redirect [C-173] (For North America Only)
KYN4
KYI=1: Function Key
FKY=132: Redirect

Note: When INVITE is sent to the Standard SIP terminal and 603 (Decline) is received from the Standard SIP terminal, CALL REDIRECT [C-173] is activated.

Note: Data assignment with this command is used for internal processing of the Telephony Server. This data assignment is not the key data assignment for Standard SIP terminals.

STEP 5: **ALGSL/ALGSN** - Assign a Telephone Number to a Station Number used for the Standard SIP terminal.

Note: Telephone Number assignment is required for SIP Handler Controlled SIP terminals.

Assign the following data if required.

STEP 6: **ASFC** - For Stations allowed Attended Transfer and Three-Way Conference, assign a Class of Service that allows SFI=75: Call Hold Conference (only for guest station in the Hotel system).
D/N
SFI=75: Call Hold Conference
SFC: Service Restriction Class
RES: Restriction Data

STEP 7: **ASLG** - Specify the Tone Data used per system.
[FIRST] tab - Basic Data Setting
IP-DTG generates tones compliant with the Country Code assigned with the system data.
[TERMINAL] tab - Terminal Data Setting
By the following data assignment, tones from IP-DTG to Standard SIP terminal can be specified on a Tenant Number/Station Number basis.
Change Data Type: Check the check box for “TONE”.
KIND: Specify terminal kind.

Note: SIP Handler Controlled SIP terminal can use the language data compliant with specified tones only.

STEP 8: **ADSDL/ADSDN** - Assign detailed data for the Telephone Number used for the Standard SIP terminal.
UGN: User Group Number
TELN: Telephone Number
KIND: Kind of User Agent (User Agent of Terminal/User Agent of System)
TYPE: Type of User Agent (System Default/User Agent Class)
UAN: User Agent Class Number

Note: Assign this data when you use the fixed User Agent Class setting to the login Telephone Number regardless of User Agent Information given by SIP Handler Controlled SIP terminal.
This data setting is not required when you use the User Agent Class according to the User Agent Information given by SIP Handler Controlled SIP terminal.

STEP 9: **ALOCL** - Assign the IP location ID data
Assign Location ID data of the Standard SIP terminal when Peer to Peer Bandwidth Control [P-75] is used.

Note: Data setting for Setting of Type of Service (Signaling Packet) is not applied to a SIP Handler Controlled SIP terminal.

STEP 10: **AIVCL** - Assign the Payload data between IP location IDs of the SIP Handler Controlled SIP terminal and connecting IP equipment. Confirm the specification of Standard SIP terminal before this data assignment.

Note: Setting of Jitter Buffer and Setting Type of Voice Packet Service are not applied to a SIP Handler Controlled SIP terminal.

Note: Data other than Payload data is not applied to a SIP Handler Controlled SIP terminal.

Note: For Setting of Payload, including the settings of Type=G.711 and Size=20ms is recommended.

STEP 11: **ALIDL** - Assign the mounting location of the Internal SIP Handler and IP-DTG.
PH LENS: Assign the MG, Unit, LG of an open line group for use by Internal SIP Handler
PH Type=Internal SIP Handler
REGMAX=4095

To delete SIP terminal data

The deletion procedure is the reverse order of assigning the feature.

<Setting Password for Digest Authentication>

STEP 1: **ASPW** - Assign the password for Standard SIP terminal.
TYPE: Select the type of data memory (either LDM or NDM) storing the Telephone Number.
UGN: User Group Number
TELN: Assign Telephone Number of SIP terminal.
Current Password: Asterisks (*) are always displayed if the old password data exists. **Note 12**
New Password: Enter the new password (maximum 10 digits). **Note 13**
New Password Confirmation: Enter the new password again.

Note: The following need to be noted:
- When this data is not set, the Telephone Number of each terminal is used by default.
- After setting the password, back up the office data by using the MEM_HDD command Data Type: Call Forwarding.
- The assigned password can be deleted by using the RDPD command.

Note 12: The assigned password is indicated with 16 asterisks (*) regardless of the actual password digits.

Note 13: The TELN (telephone number) of IP terminals/SIP Multiple Line terminals/SIP Handler Controlled SIP terminal can also be assigned up to 10 digits.

<Video Data Assignment>

Note: This video data assignment is for Video calls until FP95-113 V3. For Video calls from FP95-114 V4, see VIDEO CALL [V-31].

STEP 1: **ASYDL** - Assign the following system data.
System Data 1
Index 880, Bit 5: 1 (Internal SIP Handler enabled)
Index 1353, Bit 1: 1 (Software Digital Tone Generator (IP-DTG) enabled) **Note 14**
Index 1382, Bit 1: Digest Authentication when the SIP Handler Controlled SIP terminal performs registration
0=In Service (default)
1=Out of Service
Index 1382, Bit 2: 1 (Internal SIP Handler Video enabled)

Note 14: System initialization is necessary when this data is set.

STEP 2: **ADTM** - Assign IP address 2.
Select Lan1
Add IP address 2

STEP 3: **ALIDL** - Assign PH lens.
Add PH lens for PH type Internal SIP handler
Add PH lens for PH type IP-DTG

STEP 4: **AISTL -** Assign SIP stations with Video capability.

STEP 5: **ALGSL -** Assign Stations programmed in AISTL**.**

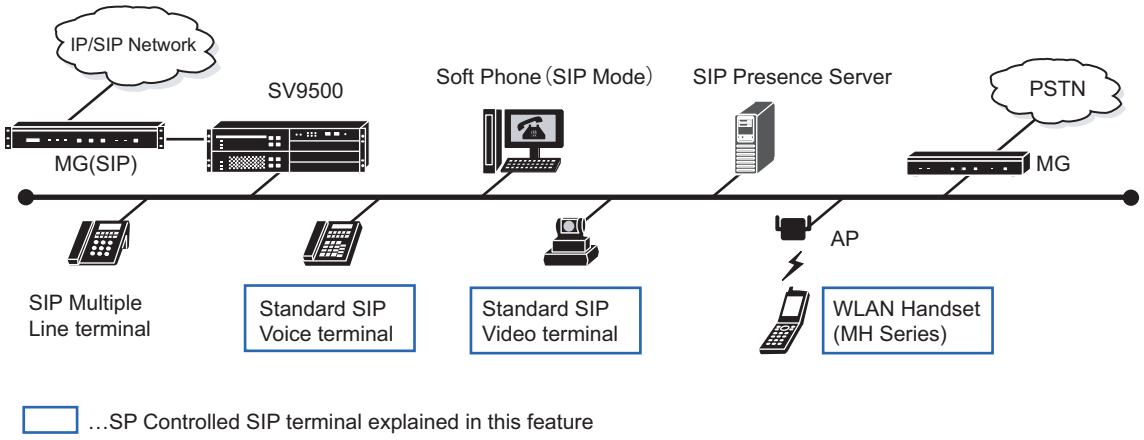
STEP 6: **AUACL -** Assign the following.
Function: User Agent Class Setting
Type: System default
Kind: System default setting

CDN 3: Send INVITE with SDP (In case CDN3 is 1, destination side picks codec)
CDN 4: Hold Method
CDN 12: Tone [2] = FE
CDN 13: Tone [1] = 3F

S-168 SP Controlled SIP Terminal

General Description

This feature allows the system to accommodate the SIP terminals mentioned below as stations by controlling them through the SP part and the Internal PHE inside of the Telephony Server. The SIP terminals can establish an interconnection with another Telephony Server Station such as a Desktop terminal, and use various Telephony Server services.



- **Standard SIP Voice Terminal (Note 1, Note 2)**
  Standard SIP voice terminal (third-party product) is based on IETF RFC3261 and Internet-Drafts. Standard SIP voice terminal supports basic call services and PBX supplementary services.

- **Standard SIP Video Terminal (Note 1, Note 2)**
  Standard SIP video terminal (third-party product) is based on IETF RFC3261 and Internet-Drafts. Standard SIP video terminal supports video codecs, and performs the video conferences.

- **WLAN Handset (MH Series)**
  Based on IETF RFC3261 and an Internet-Draft, it has the original functions of NEC and also the functions of a built-in PHS terminal. It supports various Telephony Server features equivalent to those of a built-in PHS terminal.

Note 1: Note that a connection test is required when using a Standard SIP Voice/Video terminal (third-party product).

Note 2: When using a Standard SIP terminal with LCD (third-party product), the information displayed on the LCD depends on the specifications of the terminal being used.

Note: Although the Call Forwarding feature can be set using Preset Dialing (Special Access Code + Call Forwarding Destination Number), the user cannot hear Service Set Tone (SST).

Control Method of SIP Terminal

SIP terminals can be controlled with the methods listed below.

- SP Controlled
- SIP Handler Controlled

When using SIP terminals, the available services differ according to the control method to be used. Decide which method to use referring to the Feature List included at the beginning of this manual.
SIP Handler Controlled SIP terminal is recommended if SP Controlled SIP terminal is not necessary.

Advantages of SP Controlled SIP Terminal	Advantages of SIP Handler Controlled SIP Terminal
The following features are available with SP Controlled SIP terminal. • CALL FORWARDING-PS INCOMING INCOMPLETE [C-155] • AUTOMATIC ANNOUNCEMENT-CALLED PS OUT OF ZONE [A-129] • SIP Instant Messages **Note:** The above features are unavailable with SIP Handler Controlled SIP terminals.	The following features are available with a SIP Handler Controlled SIP terminal. These are enhanced features in comparison to an SP Controlled SIP terminal features. • Various PBX services which have been enriched by enabling PBX tones to be received through IP-DTG • Enriched OAI service • Enriched Hotel service • Improved maintainability (Detailed data setting by User Agent/reduction of the number of occupied LENS) • Ongoing call retention during the CPU changeover

Operating Procedure

None.

Service Conditions

1. SP Controlled SIP terminals and SIP Handler Controlled SIP terminals can be mixed in a system.
2. An SP Controlled SIP terminal uses LAN1 IP address (ACT).
3. The Station Number (Telephone Number) of a SIP terminal is up to 12 digits in default setting ("*" and "#" cannot be used).

   Note: An up to 16 digits Station Number is available for a SIP terminal depending on the system data setting (ASYDL/ASYDN SYS1 Index 702 Bit 0).

4. The maximum number of speech channels (1200 channels) may not be available depending on the assignment conditions of other terminals' station data or trunk data. 1200 SIP speech channels will require 1600 vacant LENS positions in the system.

   Note: The maximum number of SIP terminal speech channels that can be assigned to each LP (Local Partition) is 402. Therefore, if you assign 1200 SIP terminal speech channels, 3 LPs (Virtual MG=02-07) are used.
   Only when using SIP WIRELESS TERMINAL BACKUP [S-154], 804 channels can be assigned in each LP per SP.
   Single: 402 channels per LP (1200 channels per system)
   Dual: 804 channels per LP (2400 channels per system)

   Note: If any of the following are accommodated in a LP, the maximum number of SIP terminal speech channels per LP (402 channels) becomes smaller.

   - CS
   - ISDN terminal
   - PRI/BRI (including MG)

   SIP Speech channels share common memory with CS, ISDN terminal, and PRI/BRI (including MG). So to prevent capacity related issues, try to avoid assigning these devices in LP's containing large amounts of SIP Speech channels.

   Note: When assigning station data of SIP terminal and PS, Virtual LENS (MG=26-31) are automatically used as SIP terminal and PS accommodated locations. Therefore, if a system accommodates both SIP terminals and PSs, their total number must be 4000 or less.

5. To use a SIP terminal, the client license for the SIP terminal is required. The number of client licenses is shared by SP Controlled and SIP Handler Controlled SIP terminals. When the system accommodates both types of SIP terminals, licenses are required for the total sum of the SIP terminals.
6. SIP terminals cannot use characters to select a destination for dialing (0-9, #, * only).
7. Depending on the combination of the terminal, DTMF tones may not be sent.
8. When DTMF is sent from the terminal as voice data, a call using compressed codec (except for G.711) cannot be guaranteed.
9. Each Station Number (or Telephone Number) is set as the default password of a SIP terminal in the system. The password can be modified with the ASPW command. The RDPD command can be also used to delete the password. After modification of the password, modify the password in the SIP terminal configuration data and re-register the SIP terminal by turning off/on the power.

10. A SIP terminal can accommodate other Desktop terminal's line as Sub Line.
11. The desired values of Type of Service (ToS) can be set to SIP messages sent from the system with the ALOCL command. The ToS values apply to LOC-ID = 0 (System).
12. The registration status of SIP terminals can be checked through the DISD command. Also soft RST packets can be sent to SIP terminals in registered state. For more information about the DISD command, refer to Operations and Maintenance Manual.
13. Standard SIP terminals support only Preset Dialing. Overlap Dialing is not supported.
14. A SIP terminal uses Digest Authentication under the following conditions:

Setting on the PCPro		Authentication	
The terminal authentication per call is omitted (ASSDL/ASSDN)	NO DIGEST AUTHENTICATION (ASPTL/ASPTN)	REGISTER Method	For Initial INVITE Method
No check	No check	Supported	Supported
Check	No check	Supported	Not Supported
No check	Check	Not Supported	Not Supported
Check	Check	Not Supported	Not Supported

15. If the REGISTER information becomes invalid due to an un-REGISTER from the terminal or the REGISTER expiration, the SIP terminal can not be used. For example, the calls in progress are released on a mandatory basis. In this case, it is necessary to re-register the SIP terminal by turning off/on the power.
16. When the Telephony Server is turned off/on, restarting the SIP terminal is also required.
17. When the REGISTER request from the SIP terminal is accepted by the SP, the internal clock of the SP is sent from the SP to the SIP terminal. Some SIP terminals synchronize their internal clocks to the SP's internal clock. Thus, it is necessary to keep the SP's internal clock accurately.
18. This feature is not available for:
    - Hotel service
    - ACD service
    - OAI service **Note 3**

    Note 3: Only WLAN Handsets (MH Series) can be used in the OAI system.

19. The SP does not have Redirect Server function.
20. When a SIP terminal performs a switchhook flash, the prepause timer value is approximately three minutes as a standard.
21. When a SIP terminal or SIP server is in the following state, it is treated as in the case of Out of Zone.
    - SIP terminal is turned off.
    - Communication between a SIP terminal and SIP server is down due to a network error or other reasons (including in the case where a wireless terminal moves out of an AP's area).
    - Virtual Speech channels of SIP server in ACT mode are all busy.
22. To use the SIP telephone, the client license for the SIP terminal is required.
23. Each SIP terminal uses its Telephone Number (TELN) as a password by default.
24. Even if the system is rebooted, the SP does not notify SIP terminal any SIP terminal under its control. Therefore, the SIP terminal cannot be used until it is registered again.

25. When a SIP terminal's line on Desktop terminal is held, held SIP terminal can retrieve it by performing Preset Dialing or Overlap Dialing. In the case of a SIP terminal which supports only Preset Dialing dummy number, it can retrieve the held call. The Telephony Server does not need to assign this dummy number. (i.e. Telephony Server ignores any numbers which are dialed by preset dialing.)
26. SP Controlled Standard SIP terminals cannot be used in a network where Peer-to-Peer CCIS and DTI-FCCS coexist and each node is connected in tandem.
27. In an FCCS network, perform the required setting to use the virtual speech channels of the node to which the SIP terminal is registered.

<Conditions for WLAN Handset (MH Series)>

1. Instant Messages can be sent/received among multiple domains.
2. Instant Messages can be sent/received between WLAN Handsets (MH Series) by specifying each SIP address.

<Conditions for SP Controlled Standard SIP (Voice/Video) Terminal>

1. A Standard SIP terminal is disconnected from the system in the following cases:
   - Connecting party disconnects during a call
   - Dialed party is busy
   - Dialed number is restricted

   In these cases, the Standard SIP terminal hears a local tone such as Busy Tone (BT) or Reorder Tone (RT), or becomes idle.
2. A Standard SIP terminal cannot hear tones such as Second Dial Tone (SDT), Special Dial Tone (SPDT) or Service Set Tone (SST). Therefore, depending on the service, Standard SIP terminal is not allowed to dial. In this case, the terminal is disconnected immediately. **Note 4**

   Note 4: Call Back (Calling Party), Executive Right-of-Way (Calling Party), Privacy (Called Party), Call Waiting (Calling Party), Call Waiting (Called Party), Call Park Set and Answer
3. When a party connected to a Standard SIP terminal performs a Blind Transfer, the Standard SIP terminal continues to hear local hold tone until the third party answers.
4. Even if a Standard SIP terminal is transferred and the connecting party is changed, the system does not display the new connecting party's number on the Standard SIP terminal's LCD.
5. When a Standard SIP terminal transfers a call and goes on-hook before the forwarding destination answers, the held party itself calls the forwarding destination. In this case, if the held party is a Standard SIP terminal, the held Standard SIP terminal continues to hear a local hold tone until the destination answers.
6. A Standard SIP terminal enables Call Hold features supporting RFC3264 a=sendonly method and supporting RFC2543 c=0 method. You can select either of those methods on a terminal basis with the ASPTL/ASPTN commands.

<Conditions for SP Controlled Standard SIP Video Terminal only>

The following describes additional conditions particular to a Standard SIP Video terminal.

1. Standard SIP Video terminal user can answer the incoming call by voice only.
2. The following shows the codec to be used for the video connection.

G.711μ-law64K	G.711A-law64K	G.723.1 (6.3K)	G.722-64K	G.728
G.729	Unrestricted Digital	MP4A-LATM	G.722.1	H.263 **Note 5**
H.261 **Note 5**	H.264 **Note 5**	MP4V-ES **Note 5**		

Note 5: Codec for Video Call

3. Video Call can be placed on hold and transferred (Attended Transfer only). Blind Transfer is not available for Video Call.
4. If a Standard SIP Video terminal transfers an original Video Call to other Video terminal and goes on-hook before the transferred call (second call) is answered, both original call and second call are released.
5. If a Standard SIP Video terminal in the Video connection is recalled, the terminal is temporarily placed in the voice connection.
6. When a Standard SIP Video terminal attempts to transfer a call, the Video terminal should complete the transferring operation within 10 seconds after holding the call.
7. If a Standard SIP Video terminal fails to transfer a held call, retrieve the held call and then perform holding and transferring operation again.
8. If a Standard SIP Video terminal makes a call to the terminal that does not support a video function, voice connection is established between the two parties.
9. The line of a Standard SIP Video terminal can be accommodated to other telephones' Sub Line (Desktop terminal). In this case, the video connection cannot be performed via Sub Line.
10. The Video terminal supporting multipoint conference must send REGISTER information of all the conferees' Station Number to the Telephony Server.
11. Bandwidth Control is not available for the Video call.

<Conditions Specific to UMGi/RNIP System>

1. Only the CCN can accommodate Standard SIP terminals.

Programming

In addition to the following data settings, the initial data for the system (system configuration data, Internal PHE data, etc.) is required to be set. For details on the initial data settings, refer to "Initial Data Assignment" in Chapter 6 System Startup of Appliance Model Installation Manual.

Note: In an FCCS network, assign the following station data for the node to which the speech channels are set.

STEP 1: **ASYD** - Assign the following system data.
System Data 1
Index 30 = 06 Hex (Assign the mounting capacity of Data Memory (DM))
Index 186, Bit 6 = 1 (CCIS is enabled) **Note 6**
Index 220, Bit 6 = 0 (ISDN is enabled) **Note 6**

Note 6: Although CCIS and ISDN services must be enabled by the system data here, the optional software for CCIS and ISDN is not required when CCIS and ISDN services are not used.

System Data, Index 420, Bit 0 = 1 (Access code using pre-set dialing is enabled)

STEP 2: **ASYDL/ASYDN** - Assign the following system data.

ASYDL
- System Data 1, Index 700

Assign the SIP Terminal SHF Prepause Timer.
The timer value setting is [MTC × TC] seconds. **Note 7**

Note 7: When this data is 00 HEX, the timer value is automatically set to about 3 minutes.

bit 0-3 (MTC: 1-F)
bit 4-6 (TC)

b6	b5	b4	
0	0	0	: Not used
0	0	1	: Not used
0	1	0	: Not used
0	1	1	: Not used
1	0	0	: 30 seconds
1	0	1	: Not used
1	1	0	: Not used
1	1	1	: Not used

bit 7 = 0/1 (SIP Terminal SHF Prepause Timer: In service/Out of service)

ASYDL/ASYDN

- System Data 1, Index 702, Bit 0=1 (More than 12 digits (16 digits at the maximum) of Telephone Number is available for a SIP terminal) **Note 8**

- System Data 1, Index 702, Bit 5=1 (Digest Authentication is in service when a response from a Standard SIP terminal does not include "qop" parameter.) **Note 9**, **Note 10**, **Note 11**

Note 8: Assign this system data to register more than 12 digits of Telephone Number to SIP terminals. Leave this bit to 0 (default setting) when it is not required. Use ASYDN command in FCCS network.

Note 9: This system data is necessary for Standard SIP terminals based on RFC2617.
When the system sends a 401 or a 407 request during a Standard SIP terminal's registration or call origination and receives a response from the terminal without the "qop" parameter which is used to decide the Digest Authentication method, the terminal fails Digest Authentication.
To avoid this situation, this system data enables the Standard SIP terminal to use Digest Authentication if the system receives a response without "qop" parameter from the terminal.

Note 10: This system data is available when you select TYPE=1 in the TERMINAL (Terminal Data) field of ASPTL/ASPTN command.

Note 11: This system data becomes valid after you reset the SP with the COSSN command. Follow this procedure to reset the SP:
[1] Click the REDUNDANCY tab. Then type SIP server ID and FPC of the target SP.
[2] Click the Get button to get the information of the specified SP.
[3] Click the Reset button to reset the specified SP.
[4] Click OK.

STEP 3: **ASDIL/ASDIN** - Assign SIP domain name index data. For more details on this command, refer to Chapter 5 of Appliance Model Installation Manual and Command Manual.

SIP Server ID: display only (SIP server ID assigned by ASSDL/ASSDN is displayed).

Domain Name: Enter a desired domain name [Max 128 characters (alphanumeric characters, a dot, a hyphen)].

STEP 4: **ASSDL/ASSDN** - Assign the SIP server operating data. For more details on this command, refer to Chapter 5 of Appliance Model Installation Manual and Command Manual.

FPC: Enter the Fusion Point Code (1-253). (ASSDN only)

SIP Server ID: Enter the SIP Server ID within the range of 1-127. Note that this ID must be assigned as *SP SIP-ID* when performing setup of SP.
Note 12

Note 12: If the value 128 or more has been assigned as SIP Server IDs, it is necessary to read out the assigned SIP Server IDs and delete them. Assign SIP Sever IDs again, within the range of 1-127.

<BASE DATA tab>

CHANNEL: Assign the number of SIP terminal speech channels (3-1200)

NAME: Assign the SIP Server Name (Max. 16 characters). Note that this name must be assigned again as *SP hostname* when performing setup of SP.
TN: Tenant Number
LENS: Assign the range of LENS of SP. **Note 13**

Note 13: Specify the virtual speech channels (physical PIR or virtual PIR) having no station or trunk data. It is not available to overwrite the existing station or trunk data.

<BASE tab>
(DETAIL): Click each checkbox if required.
The terminal authentication per call is omitted: Default is OFF (this setting blocks connections attempted by unauthorized terminal.)
Change in SIP Server port number: Only the default value (5060) can be used. **Note 14**

Note 14: Do not change the default port number.

<DOMAIN tab>
This tab is used for assigning domain names that are managed by a specified SP. Select the domain name(s) assigned by ASDIL/ASDIN. Although the SP can manage multiple domains, a domain cannot be managed by multiple SPs.

EDIT DOMAIN: Select the domain name to add.
DOMAIN LIST: All the domains allowed to connect are listed.

<REDUNDANCY tab>
This tab is used for assigning the redundant partner SP ID. For more details, refer to SIP WIRELESS TERMINAL BACKUP [S-154]

Pair SIP Server ID: Assign SIP Server ID of the redundant partner SP (1-127).

STEP 5: **ASPO** - Assign SIP terminal operation data.
TN: Tenant Number
STN: Physical Station Number (a maximum of 5 digits)
RSC: Route Restriction Class, 1- 15 (class grouping on Outgoing/Incoming Call Restriction, specified by ARSC command)
SFC: Service Feature Restriction Class, 1 - 15 (class grouping on allowed/denied service features, specified by ASFC)
PHIN: Virtual LENS of the SIP Terminal is displayed.
PS NO: Display internal identification number (HEX)
**(NEC engineer use)*

STEP 6: **ASPTL/ASPTN** - Assign SIP Terminal Data.
UGN: User Group Number **Note 15**
TELN: Telephone Number (a maximum of 12 digits) **Note 16**

Note 15: When using DIAL PLAN TO SUPPORT IP CENTREX FUNCTIONALITY [D-166], User Group Number (UGN) can be assigned in the range of 1 to 255. SIP terminals (WLAN Handset (MH Series), Standard SIP Voice Terminal and Standard SIP Video terminal) can use DIAL PLAN TO SUPPORT IP CENTREX FUNCTIONALITY [D-166].

Note 16: An up to 16 digits Telephone Number can be registered when ASYDL/ASYDN SYS1 Index 702 bit 0=1 is assigned. To turn off the data of ASYDL/ASYDN SYS1 Index 702 bit 0=1, you need to delete the SIP terminal data assigned when ASYDL/ASYDN SYS1 Index 702 bit 0=1 is set.
When this data is changed, re-registration of SIP terminals is required. If SIP WIRELESS TERMINAL BACKUP [S-154] is activated, re-registration is not required.

BASE (Base Data)

FPC:	Fusion Point Code (1-253) - only for ASPTN
TN:	Tenant Number
STN:	Physical Station Number assigned by ASPO
P-NO:	"Access Code + Office Code" assigned by AUPVN + "Telephone Number" of SIP Terminal is displayed. **Note 17**, **Note 18**

Note 17: This parameter is displayed on ASPTN only. This parameter is used for DIAL PLAN TO SUPPORT IP CENTREX FUNCTIONALITY [D-166]. For a WLAN Handset (MH Series), it is necessary to set "Private Number" + "Domain" as SIP address from each terminal.

Note 18: Access Code + Office Code + Telephone Number assigned by AUPVN is not displayed in the following conditions:
- When ASYDL/ASYDN SYS1 Index 702 bit 0=1 (More than 12 digits of Telephone Number is available for SIP terminal) is assigned.
- When reading out the data assigned before enabling DIAL PLAN TO SUPPORT IP CENTREX FUNCTIONALITY [D-166].

TERMINAL (Terminal Data)

TYPE:	For Standard SIP Voice Terminal or Standard SIP Video terminal, select "TYPE1". For Wireless Handset (MH Series), select "TYPE3".
DOMAIN INDEX: DOMAIN NAME:	Select a domain index and a domain name which the SIP terminal belongs to. In an FCCS network, select the domain name of SIP server of the node to which the ASPO command data is set. When the domain name is not input, only index of domain name is displayed.

[DETAIL]
Selectable parameters in each TYPE will appear in DETAIL field as shown below when TYPE is set:

When TYPE=TYPE 1 is set	When TYPE=TYPE 2 is set	When TYPE=TYPE 3 is set
• NO DIGEST AUTHENTICATION • WIRELESS • SOFTPHONE • VIDEO TERMINAL • INVITE with SDP • When secondary call is disconnected, held call remains on hold • RTP Relay BOX (*Not used) • MWI (RFC3842) • SESSION TIMER • SENDONLY • Nurse Call Display • Distinctive Ringing • When the notification error occurs in SET or DEL process, then continue processing without displaying a confirmation dialog.	• NO DIGEST AUTHENTICATION • WIRELESS • SOFTPHONE • VIDEO TERMINAL • Nurse Call Display • IP-IVR • DNIS on FROM field of INVITE • When the notification error occurs in SET or DEL process, then continue processing without displaying a confirmation dialog.	• NO DIGEST AUTHENTICATION • WIRELESS • SOFTPHONE • VIDEO TERMINAL • MWI (Extension header method) • When the notification error occurs in SET or DEL process, then continue processing without displaying a confirmation dialog.

NO DIGEST AUTHENTICATION: Mark the checkbox if the terminal does not support Digest Authentication.

WIRELESS: Mark the checkbox when a WLAN Handset (MH Series) is used.

SOFTPHONE: Mark the checkbox when using SOFT PHONE WLAN HANDSET [S-159] or SOFT PHONE WIRED-PHONE HANDSET [S-147].

VIDEO TERMINAL: Mark the checkbox when Standard SIP Video terminal is used.

MWI (Extension header method): Mark the checkbox when a WLAN Handset (MH Series) is used. (This parameter is valid only when TYPE = TYPE3.)

MWI (RFC3842): Mark the checkbox when Standard SIP Voice Terminal or Standard SIP Video terminal is used.

INVITE with SDP: Select the check box if the terminal does not support INVITE without SDP. (The terminal must support re-INVITE without SDP.) When the check box is selected, a dummy SDP will be added to the initial INVITE from the Telephony Server. (This parameter is valid only when TYPE=TYPE1.)

When secondary call is disconnected, held call remains on hold:	Select the check box if CONSULTATION HOLD RELEASE [C-169] (when terminal A transfers a held call (terminal B) to terminal C, terminal B remains in hold state even when terminal A and terminal C are disconnected) is used with Standard SIP terminal. (This parameter is valid only when TYPE=TYPE1.)
SENDONLY:	Select the checkbox when your Standard SIP terminal enables Call Hold features supporting RFC3264 a=sendonly method. When this parameter is not selected, Standard SIP terminal performs Call Hold features supporting RFC2543 c=0 method. (This parameter is valid only when TYPE=TYPE1.)
Nurse Call Display	Not used. (This parameter is valid when TYPE=TYPE1 or TYPE2.)
IP-IVR	Not used. (This parameter is valid when TYPE=TYPE2.)
DNIS on FROM field of INVITE	Not used. (This parameter is valid when TYPE=TYPE2.)
Distinctive Ringing:	Select the check box if Distinctive Ringing is used. (This parameter is valid only when TYPE=TYPE1.)
When the notification error occurs in SET or DEL process, then continue processing without displaying a confirmation dialog.:	When this check box is selected, the SET or DEL process continues without displaying a confirmation dialog even if the notification error is received from the SIP server. (This parameter is valid when TYPE=TYPE1, TYPE2 or TYPE3.)

Note: Normal operation of the terminal is not guaranteed if the Type and Detail parameters are incorrectly assigned.

Note: If the data on the TERMINAL tab is changed, the SP receives notification about the data change. If the SP cannot receive the notification for some reason, the disparity between the actual Office data and information on the SP is created. At this time, the terminal is required to re-register.

STEP 7: **ASFC** - Assign a Service Feature Restriction Class that restricts (RES=0) the following SFIs to the Standard SIP terminal Station.

SFI 4: Executive Right-of-Way (Called)
SFI 5: Call Waiting-Originating (Calling)
SFI 6: Call Waiting-Originating (Called)
SFI 67: Call Park-Access and Answer

The following explains how to assign the passwords for SP Controlled SIP terminals.

ASPW - Assign the password for SIP terminal.

TYPE:	Select the type of data memory (either LDM or NDM) storing the Telephone Number.
UGN:	User Group Number
TELN:	Assign Telephone Number of SIP terminal.
Current Password:	Asterisks (*) are always displayed if the old password data exists **Note 19**.
New Password:	Enter the new password (maximum 16 digits) **Note 20**.
New Password Confirmation:	Enter the new password again.

Note 19: The assigned password is indicated with 16 asterisks (*) regardless of the actual password digits.

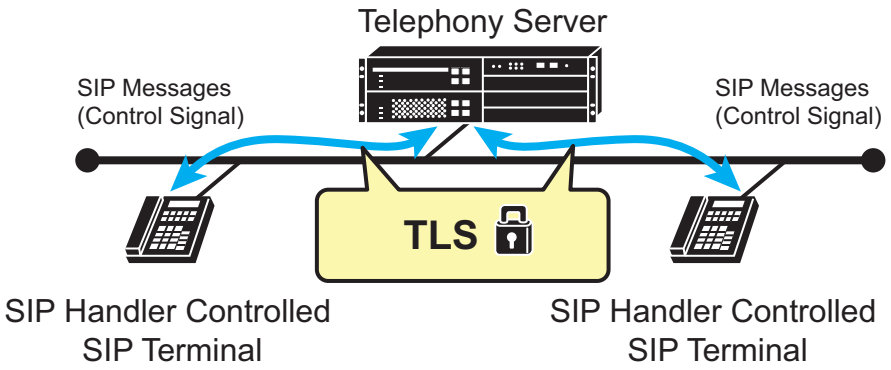
Note 20: The TELN (telephone number) of IP terminals/SIP Multiple Line terminals can be assigned up to 10 digits.

S-169 SIP Handler - SIP Terminal Support for TLS Encryption

General Description

This feature provides the encryption feature for SIP messages on the network between SIP Handler Controlled SIP terminal and Telephony Server. This way, telephone tapping and SIP message falsification by a third person can be prevented.

Note: This feature is available since FP95-114 V4.



Operating Procedure

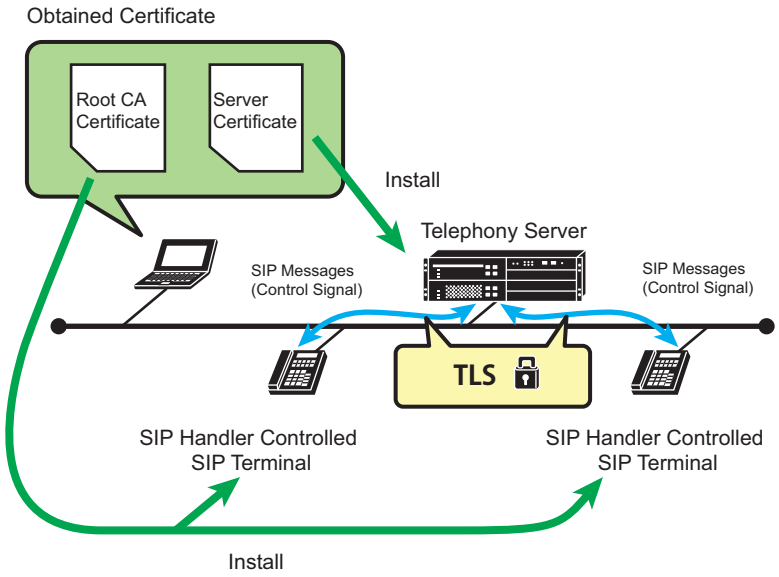
When SIP Handler Controlled SIP terminal accesses the Telephony Server, a certificate file is required to confirm an identification of the Telephony Server. For using the certificate file, the user requires a public key certificate (the certificate issued by the certificate authority) or a self signed certificate (the certificate created by the user.) The maintenance personnel is required to install the certificate the user obtained.

The installation directories for each certificate are as follows:

- Server certificate: Install the certificate to the Telephony Server.
- Root CA certificate: Install the certificate to the SIP Handler Controlled SIP terminal. **Note 1**

Note 1: When the certificate is installed on the SIP Handler Controlled SIP terminal beforehand, this installation is not required.

With the installation of the certificate, data going through the network between Telephony Server and SIP Handler Controlled SIP terminal is encrypted by TLS and the network information will be protected.



1. Precaution for obtaining the certificate

This procedure is done by the user. The precautions when the user obtains a certificate are as follows.

(a) Public Key Certificate and Self Signed Certificate

- Available file format for server certificate is PEM format only.
- Be sure to prepare the server certificate in combination with server secret key. **Note 2**

Note 2: To combine the server certificate and server secret key, edit the certificate file as follows using text editor.

```
-----BEGIN RSA PRIVATE KEY-----
(The contents of server secret key)
-----END RSA PRIVATE KEY-----

-----BEGIN CERTIFICATE-----
(The contents of server certificate)
-----END CERTIFICATE-----
```

- A server secret key encrypted with a password is not available. When creating a server secret key, do not set password. If the password is set to the server secret key, decrypt the password and combine the server secret key with the server certificate.

- When the certificate has expired, server identification is disabled and the communication between Telephony Server and SIP Handler Controlled SIP terminal is disabled. In that case, the following procedure is required.

Expired Certificate	Counter Measures
Root CA Certificate	• Obtain the client CA certificate and install the certificate file to the SIP Handler Controlled SIP terminal. • Reissue the server certificate and install the certificate file to Telephony Server.
Server Certificate	• Reissue the server certificate and install the certificate file to Telephony Server.

- When creating server certificate, assign the IP address of SIP Handler Controlled SIP terminal to the Common Name.

(b) Public Key Certificate

When you order the issue of server certificate to the public certification authority, intermediate certificate may be issued with the server certificate depending on the certification authority. In that case, be sure to prepare the server certificate which is combined with intermediate certificate. **Note 3**

Note 3: To combine the server certificate and intermediate certificate, edit the certificate file as follows using text editor. The order of certificate is placed from low assurance level to high assurance level on certificate chain.

```
-----BEGIN CERTIFICATE-----
(The contents of server certificate)
-----END CERTIFICATE-----

-----BEGIN CERTIFICATE-----
(The contents of intermediate certificate)
-----END CERTIFICATE-----
```

(c) Self Signed Certificate

- When using self signed certificate, specify the validity date of certificate while creating self signed certificate and server certificate.
- When creating self signed certificate, you can decide to set or not set a password of secret key for client authentication. If the password is set to the self signed certificate, enter the password when you make a signature to the self signed certificate or Certificate Signing Request (CSR).

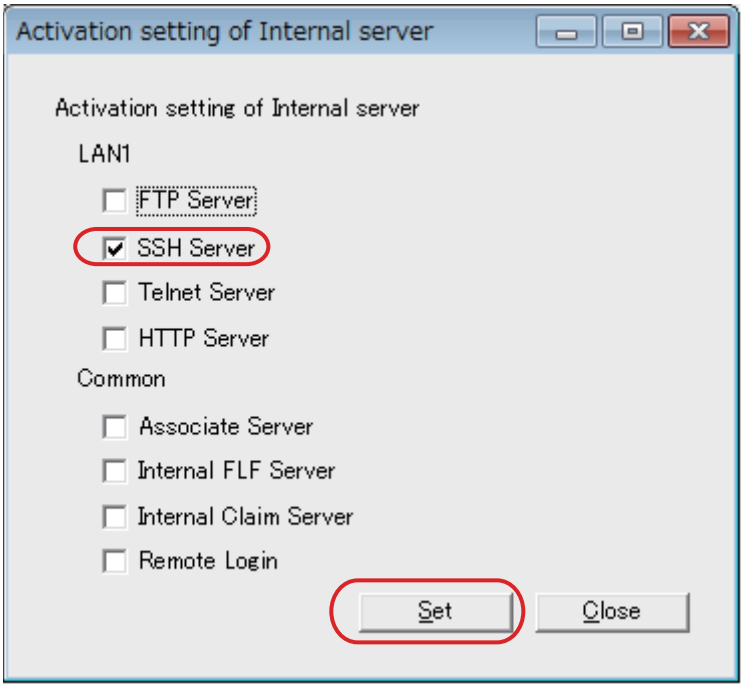
2. Installation Procedure of Server Certificate

This procedure is done by maintenance personnel.

STEP 1: Startup SSH server.

Activate the internal SSH server in the Telephony server.

1. Execute ADTM command on PCPro.
2. Choose “LAN1” on “Select LAN Interface” and Click “Get”.
3. Select the Options tab and then select “Activation setting of Internal server” from maintenance tab, and click “Execute”.
4. Select the checkbox of “SSH Server”, and click “Set”.

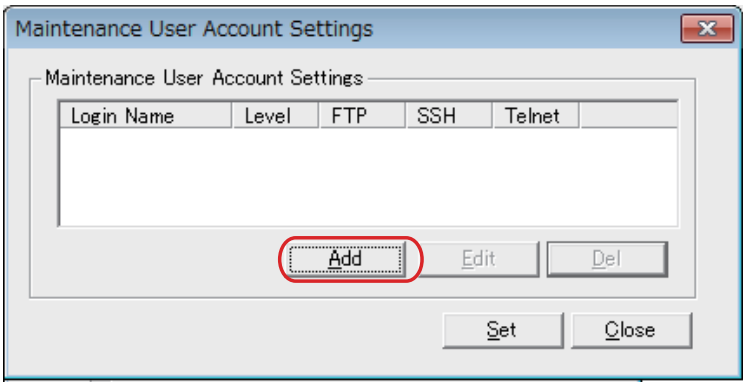


STEP 2: Set the account of maintenance personnel of Telephony Server.

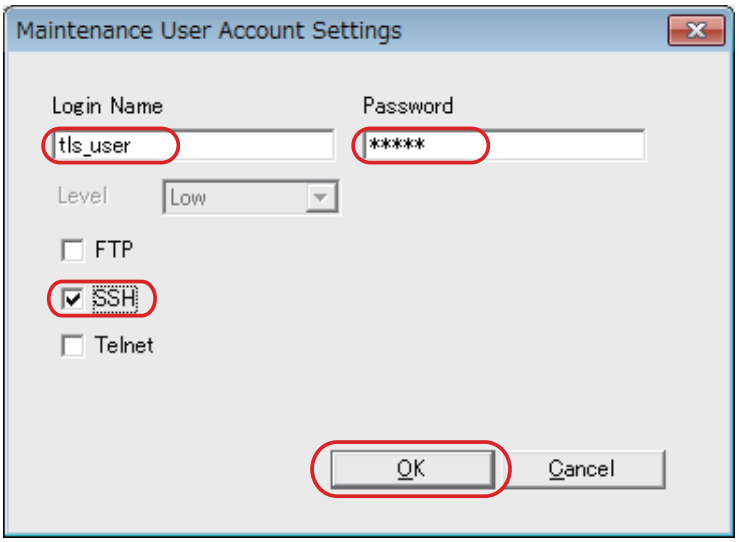
Create the Maintenance User Account for SSH.

1. Execute ADTM command on PCPro.
2. Select “LAN1” on “Select LAN Interface” and Click “Get”.
3. Select the Options tab and then select “Maintenance User Account Settings” from maintenance tab, and click “Execute”.

4. Click “Add”. When using the account already created, click “Edit”.



5. Enter login name and password, and select the checkbox of “SSH”. Then click “OK”.



6. Click “Set”.

STEP 3: Install the server certificate from the maintenance PC to the Telephony Server.

Install the server certificate saved on the local folder of maintenance PC to the Telephony Server using SCP client or SFTP client.

1. On the maintenance PC, login to the Telephony Server using SCP client or SFTP client. The login name and password are same as the login name and password of maintenance personnel on STEP 2.

2. Change the file name of server certificate saved on the local folder of maintenance PC to “sv9500certificate.pem” and put the certificate files (/mnt/CF/tls/certs) on the folder of Telephony Server.

3. Installation Procedure of Root CA Certificate

Install root CA certificate to SIP Handler Controlled SIP terminal. This procedure is for maintenance personnel. For the details of Installation procedure, refer to the each manual of SIP Handler Controlled SIP terminal.

Note: When you obtained public key certificate from a reliable certificate authority, when the root CA certificate is already preinstalled on the SIP Handler Controlled SIP terminal, this installation is not required.

Service Conditions

<Important Notices>

1. When using this feature, SIP Handler Controlled SIP terminal is required to support TCP. For the details, see SIP HANDLER - SIP TERMINAL [S-167] in this manual.

<Interactions with Other Features>

1. This feature cannot be used in collaboration with VoIP Encryption [V-27].
2. This feature does not support Location Diversity [L-55].

<Hardware/Software>

1. This feature is available only for SIP Handler Controlled SIP terminal with the following conditions:
   - The terminal supports the feature of TLS 1.2.
   - TLS feature is set to “enabled” on the configuration setting of the terminal.
2. The applicable TLS protocol version is TLS 1.2. Also, the following cipher suites are available for the Telephony Server.

Cipher Suites
TLS_RSA_WITH_AES_128_CBC_SHA **Note 4**
TLS_RSA_WITH_AES_256_CBC_SHA
TLS_RSA_WITH_AES_128_CBC_SHA256
TLS_RSA_WITH_AES_256_CBC_SHA256

Note 4: The required cipher suites of TLS1.2. defined in RFC5246.

3. The feature does not support SRTP. Therefore, be sure not to set SRTP to enable on the configuration setting of the terminal.
4. SR-MGC is not supported for this feature.

<Interactions with Networks>

None.

<Conditions Specific to This Feature>

1. For using this feature, install a server certificate in the Telephony Server. Also, a root CA certificate synchronized with server certificate is required to be installed in SIP Handler Controlled SIP terminal.
2. After the server certificate is installed in the Telephony Server again, execute program Non-Load Initialization of Internal SIP Handler. If Non-Load Initialization is not executed, the old server certificate (before installed) will be used.

3. When server certificate has expired, the Telephony Server does not restore the disconnection of TLS and conversation. However, registration of SIP terminal will be disabled. When validity date is updated, install the server certificate one more time.
4. For dual configuration, install a server certificate to the Telephony Server both system #0 and system #1. For the other following features and systems, is required to be installed different server certificate for each node.

   - Geographic Redundancy feature
   - Geographic Redundancy-enabled UMGi system
   - UMGi system

5. For TLS connection, IP address of LAN1 ACT2 on the Telephony Server and port number 5061 are used as default setting. Also, IP address of LAN1 on the Telephony Server and port number 55061 are used. The LAN interface and port number can be changed by "Configure Application Port" of ADTM command. When the data setting is changed, initialize the SIP Handler Controlled SIP terminal.
6. In this feature, only server side is authenticated by TLS connection. The TLS handshake protocol is established by TLS connection request (Client Hello (**Note 5**)) from SIP Handler Controlled SIP terminal to Telephony Server side. From the Telephony Server, a certificate request will never be sent to the terminal side.

   Note 5: TLS handshake protocol signal is defined by RFC2246.
7. The maximum number of TCP connections between Telephony Server and SIP Handler Controlled SIP terminal are 1000 by adding up the number of TLS connection and non TLS connection.
8. Change the maximum number of TLS connection (ASYDL, SYS1, Index 1418, Bit 0-3) after installing the server certificate to the Telephony Server.
9. When the number of TLS connection reaches the maximum number of TLS connections, Telephony Server will not accept a new TLS connection request in spite of the setting of maximum number of new TCP connection (ASYDL, SYS1, Index 1418, Bit 0-3).
10. When the maximum number of TLS connection is over, the registration of SIP Handler Controlled SIP terminal for TLS connection is disabled.
11. The specified port (TLS) by ADTM command is used for TLS connection only. Non TLS connection is not available.
12. When a no-communication state continues beyond the time which is assigned by the system data (ASYDL, SYS 1, Index 1421, Bit 0-3), Telephony Server sends TCP Keep Alive to the SIP Handler Controlled SIP terminal to monitor the state of TLS. When the Telephony Server does not receive the response of TCP Keep Alive from SIP Handler Controlled SIP terminal, sending retry is executed to the terminal up to 9 times until the response returns for the time interval set in the system data (ASYDL, SYS 1, Index 1421, Bit 4-7). If the terminal does not send the response after 9 times sending retry, the Telephony Server disconnects the TLS connection (When default value is assigned, the Telephony Server will be disconnected after 69 seconds (60 seconds + (one second x 9 seconds)).
13. When the TLS connection of SIP Handler Controlled SIP terminal has been disconnected, the terminal cannot originate/receive a call until the terminal performs reregistration to the Internal SIP Handler of the node that accommodates the terminal. The conditions required for reregistration are as follows:
    a.) On state of monitoring of TLS, when the response from SIP Handler Controlled SIP terminal has been lost.
    b.) By the Service Conditions of SIP Handler - SIP Terminal [S-167], when the terminal reregistration is required or the TCP connection has been disconnected.

Note 6: For details, refer to SIP Handler - SIP terminal [S-167] in this manual.

c.) By the Service Conditions of UMGi system and Geographic Redundancy feature, when the terminal reregistration is required or the TCP connection has been disconnected.

Note 7: For details of Service Conditions of Geographic Redundancy, refer to 8. Geographic-Redundancy-Enabled UMGi System in Chapter 2 Service Conditions of Geographic Redundancy Configuration Manual. For details of Service Conditions of UMGi, refer to 6. Conditions in Chapter 1 General of UMGi Configuration and Maintenance Guide.

14. When the CPU changeover is executed, TLS connections will be disconnected. However, the control signal from the Telephony Server cannot be transmitted. Then, the Telephony Server does not disconnect the conversation. the operation of SIP terminal after TLS disconnection depends on the specification of the terminal.
15. When the maximum number of TCP connection (ASYDL, SYS1, Index 1418, Bit 0-3) is changed while the TLS connection or non TLS connection is established, the already established session of TLS connection or non TLS connection will not be disconnected. If the rate of connection number has been changed, the initialization of Internal SIP Handler is required.
16. Only control signals between Telephony Server and SIP Handler Controlled SIP terminal supporting TLS are encrypted by TLS connection.
17. When Internal SIP Handler has been initialized, the connection of all terminals which are controlled by Internal SIP Handler will be disconnected.

<Others>

None.

Programming

STEP 1: **ASYDL** - Assign the following system data.

System Data 1

Index 1408, Bit 0=1 (TCP connection for SIP message is enabled)

Note: When this data is changed, each of following procedures is required:

- Initialization of Internal SIP Handler with the ADPM command
- Initialization of TCP/IP Module with ADTM command
- System Initialization with SINZ command

Index 1418, Bit 0-3: In the maximum number of TCP connection for SIP message, set the maximum number of TLS connection by percentage.

0-10=Setting value x 10% (0-100%)

11-15=100%

Note: When assigning the data, the following conditions are required:

- Install server certificate before changing this data.
- Assign a rate number which is larger than the number of the terminals supported TLS for TLS connection.

- When 0% is assigned, TLS connection is not available. When 100% is assigned, only TLS connection is available.
- The maximum number of TCP connection is 1000. Therefore, when the value is assigned as “1 (10%)”, the maximum number of TLS connections is 100.

Index 1421, Bit 0-3: Assign the time to send TCP Keep Alive for the first time on the TLS connection in 10 seconds units.

0=60 seconds (default)

1-15=Setting value × 10 seconds (10-150 seconds)

As an example, when the value “1” is assigned, TCP Keep Alive will be sent to SIP Handler Controlled SIP terminal after a no-communication state continues for 10 seconds.

Note: Default setting is recommended.

Bit 4-7: Assign the sending interval of TCP Keep Alive on the TLS connection in one seconds units.

0=1 second (default)

1-15=Setting value × 1 seconds (1-15 seconds)

Note: When assigning the data, the following conditions are required:

- TCP Keep Alive sending retry is executed to the terminal up to 9 times until a response is returned for the time interval assigned in this command. The number of retry times cannot be changed.
- Default setting is recommended.

STEP 2: **ADTM** - Assignment of Data for TCP/IP Module

Assign the port number of TLS (TCP Server) for Internal SIP Handler, when using default port number, this setting is not required. When “0” is assigned, the setting value will be as follows.

<Options Tab>

Configure Application Port

Internal SIP Handler

TLS (TCP Server): 55061 (LAN1)/5061 (LAN1 (ACT2))

Note: The port number assigned in ADTM command will be set to the setting of system data (ASYDL, SYS1, Index 1419 and Index 1420).

T-1 Tandem Switching of TIE Trunks - 2/4-Wire

General Description

This feature allows Trunk-to-Trunk Connections through the switching system, without the need for any Attendant assistance or control. The major use of this feature is in association with the Dial Tandem TIE Line Network to allow TIE Line Connections and Incoming TIE Line Calls automatic access to, and completion of, local Central Office Calls.

Operating Procedure

To operate

1. Lift the handset. Receive Dial Tone from the distant office.
2. Dial the TIE Trunk Access Code. Receive Dial Tone from the System.
3. Dial another TIE Trunk Access Code.

OR

1. Dial the Central Office Access Code. Receive Dial Tone from the other distant System or local Central Office.
2. Dial an extension number of the distant System.

OR

1. Dial the Central Office exchange number (local or toll).

Service Conditions

1. When using a Two-wire application, there may be an appreciable decrease in transmission decibel Levels.
2. When using a 4-wire application, an TLT circuit card is required for every four TIE Lines. This circuit card provides amplification/attenuation so that the desired transmitting and receiving Levels can be maintained, providing the overall Tandem system with transparency.
3. When TIE Trunks are busy, the Calling Station will receive Busy Tone.
4. Each Incoming TIE Trunk may have various restrictions for TIE-to-Central Office Calling.
5. Careful consideration should be given to Access Code Numbering Plans to avoid unnecessary loss of Access Codes and Code duplication within the same system.
6. Tandem switching can also be accomplished via the ATTENDANT CONSOLE [A-3].
7. There is no limitation to the allowable number of Tandem Switching Connections.
8. This feature can also be activated when using a Trunk in the Remote Node via FCCS link. For more detailed information, see the Data Programming Manual - FCCS.

Interactions

1. The Connection described is a user Outside the system connected to an Incoming TIE Line to an Outgoing TIE Line. TRUNK-TO-TRUNK CONNECTION [T-10] is established by either the Attendant Console or a Station user and is connected to two Outgoing Trunks.
2. TOLL DENIAL [T-6]: If TOLL DENIAL is applied to an Incoming TIE Trunk Group, the Calling Station user is routed to Reorder Tone if an attempt is made to dial-access a Toll Call.
3. Incoming dial repeating TIE Trunks (Second Dial Tone) can connect to the following types of Outgoing Trunks:
   a.) Dial repeating TIE Trunk (Second Dial Tone)
   b.) Senderized TIE Trunk

c.) Central Office Trunk
d.) FX Trunk
e.) WATS Trunk
f.) CCSA Trunk

4. STATION MESSAGE DETAIL RECORDING (SMDR) [S-10]: On Tandem Connections, the Incoming TIE Trunk Group Route Number and Outgoing Trunk Number are recorded.
5. Two Outgoing Trunk Group restriction tables are provided per system. The Direct Dial Access table is indexed when a distant Station user dials the Trunk Group Access Code, and the ATTENDANT CONSOLE [A-3] table is indexed on Attendant completed calls.

Programming

STEP 1: **ARTD** - Assign the TIE Line routes as shown below:

BOTHWAY TIE-LINES:
RT: 1

1-OSGS	:2	2-ONSG	:3	3-ISGS	:2	4-INSG	:3	5-TF	:3
6-TCL	:4	7-L/T	:1	8-RLP	:2	15-LSG	:5		

Note: To change the data for existing route (RT), initialize or reboot the trunk device such as circuit card, MG, or VS32.

STEP 2: **ATRK** - Assign the LENS, TIE Line Route Number, Trunk Number, and Tenant Number.

STEP 3: **MBTK** - Assign the Make Idle status to the TIE Line Trunks.

STEP 4: **ANPD** - Reserve a Number Level for Trunk Access. Select Normal Service (N) or Hooking Service (H) from the combo box for Connection Indexes. Assign NND in accordance with a predetermined Numbering Plan. Busy Lamp Field is not activated.

STEP 5: **ASPA** - Assign the Access Code as assigned in ANPD above. Select “N: Normal” Service or “H: Hooking” Service from the combo box for Connection Indexes. Type of Service, SRV = “OGC: Outgoing Call” for Outgoing Trunk. Assign the Route Number associated with this Access Code.

STEP 6: **ARRC** - Assign to the system Trunk-to-Trunk Connection. Use Alternative Route Index (ARI-D-RES) for Direct Connection. Three possible Restrictions may be assigned; data “0”, Connection is Restricted; data “1”, Connection is Allowed; or data “2”, Toll Restriction is required. For TANDEM SWITCHING OF TIE TRUNKS-2/4 WIRE, TOLL RESTRICTION is not required.

STEP 7: **APAD** - Assign any Pad data that may be required for desired quality of service. (Optional, see TIE LINE CONNECTION WITH PAD CONTROL [T-5]).

FCCS Programming

This data setting describes only the additional or replacement commands used to activate FCCS Services. When used as an FCCS feature, these commands replace the non-network commands (for example, AAED is replaced by AAEDN). Note that the data setting for FCCS Service varies depending on the Node.

STEP 1: **ARTDN** - Assign the TIE Line routes as shown below:

BOTHWAY TIE-LINES:

RT: 1

1-OSGS	:2	2-ONSG	:3	3-ISGS	:2	4-INSG	:3	5-TF	:3
6-TCL	:4	7-L/T	:1	8-RLP	:2	15-LSG	:5		

Note: To change the data for existing route (RT), initialize or reboot the trunk device such as circuit card, MG, or VS32.

STEP 2: **ANPDL** - Reserve a Number Level for Trunk Access. Select Normal Service (N) or Hooking Service (H) from the combo box for Connection Indexes.Assign NND in accordance with a predetermined Numbering Plan. Busy Lamp Field is not activated.

STEP 3: **ASPAL** - Assign the Access Code as assigned in ANPD above. Assign Connection Indexes, CI = “N: Normal” and “H: Hooking”. Type of Service, SRV = “OGC: Outgoing Call” for Outgoing Trunk. Assign the Route Number associated with this Access Code.

STEP 4: **ARRCN** - Assign to the system Trunk-to-Trunk Connection. Use Alternative Route Index (ARI-D-RES) for Direct Connection. Three possible Restrictions may be assigned; data “0”, Connection is Restricted; data “1”, Connection is Allowed; or data “2”, Toll Restriction is required. For TANDEM SWITCHING OF TIE TRUNKS-2/4 WIRE, TOLL RESTRICTION is not required.

STEP 5: **APADN** - Assign any Pad data that may be required for desired quality of service. (Optional, see TIE LINE CONNECTION WITH PAD CONTROL [T-5]).

T-2 Three-Way Calling

General Description

This feature enables any Station user, including Desktop terminal user, to establish a Three-Way Call by connecting an additional party to an already existing conversation.

Even if the additional party (Desktop terminal) holds the call, Three-Way Call may be established. (This service is called Consultation Hold.)

Operating Procedure

Using switch hook flash

Procedure 1

1. Perform switch hook flash. First party is placed on Consultation Hold. Receive Special Dial Tone.
2. Dial the desired number. Receive answer.
3. Announce the Conference and perform switch hook flash again.
4. Three-Way Call is now established.

Procedure 2

1. While connected with the first party, perform switch hook flash. Receive Special Dial Tone. The first party is placed on hold.
2. Dial the additional party (Desktop terminal). Receive Ringback Tone.
3. The additional party answers. Consultation Hold-All Calls state has been entered.
4. The additional party presses the HOLD key. The call in progress is held.
5. Perform switch hook flash and return to the first party.
6. The additional party or another Desktop terminal in the same Multi Line Group answers the Held Call.
7. Three-Way Call is established.

<The following procedures are applied for Desktop terminals only.>

Using CONF key

Procedure 1

1. Press the TRANSFER key. Receive Special Dial Tone. The first party is placed on hold.
2. Dial the desired number. Receive answer.
3. Announce the Conference and press the CONF key.
4. Three-Way Call is now established. The LCD displays:



Procedure 2 (Hold on Consultation Hold)

1. While connected with the first party, press the TRANSFER key (Desktop terminal). Receive Special Dial Tone. The first party is placed on hold.
2. Dial the additional party (Desktop terminal). Receive Ringback Tone.
3. The additional party answers. Consultation Hold-All Calls state has been entered.

4. The additional party presses the HOLD key (Desktop terminal). The call in progress is held.
5. Press the CONF key and return to the first party.
6. The additional party or another Desktop terminal in the same Multi Line Group answers the Held Call.
7. Three-Way Call is established.

Service Conditions

1. The originating party may hang up at any time. The remaining two parties will not be disconnected if assigned as such in system programming.
2. This feature may be activated during any call.
3. This feature is available for the following combination of parties:
   Station*-Station-Station
   Station*-Station-Central Office Line
   Station*-Station-TIE Line
   Station*-Central Office Line-TIE Line
   Station*-Central Office Line-Central Office Line **Note 1**
   Station*-TIE Line-TIE Line
   * The originating party for this feature must be a Station (including a PS).

   Note 1: This combination is only available when using a Trunk that can detect the Release Signal from the Central Office Line.

4. The maximum number of simultaneous Three-Way Calls is dependent on the quantity of Digital Conference Trunks within the system. One Digital Conference Trunk circuit card can handle eight simultaneous Three-Way Calls.
5. Three-Way Calling feature is not restricted by the line class of service assigned to the user's line; thus, a fully restricted Station can transfer a call to another Station or the Attendant.
6. This feature can be activated via PHC (Internal PHC) under the following conditions:
   a.) Any operation other than Call Hold and Call Release cannot be performed during Three-Way Calling via PHC.
   b.) If there are not enough available ports on IPPAD when the CONF key is pressed, Three-Way Calling cannot be established. (For example, up to three ports are required when using an IP Station.) **Note 2**
   c.) An additional party cannot perform switch hook flash during Three-Way Calling.
   d.) On a Three-Way Call where all parties are connected to each other by CCIS Networking via IP through PHC, if one of the parties places the call on hold, the other two will hear no Music On Hold tone. When the party returns to the original call, the other two will hear an Override tone.
   e.) On a Three-Way Call among users A, B, and C, where user A is connected to users B and C via DTI FCCS and users B and C are connected to each other by CCIS Networking via IP through PHC, if user A places the call on hold, the other two will hear Music On Hold tone.
   f.) When Call Hold is released during Three-Way Calling via VS32, Override tone is not provided.

   Note 2: IPPAD is not required for a communication among only IP equipment (without the intervention of Legacy equipment).

7. By assigning system data (ASYDL, SYS 1, Index 830, Bit 5 = 1), Three-Way Calling Tone (Add-On Ticker) can be sent when Three-Way Calling is established.

Note: This system data (ASYDL, SYS 1, Index 830, Bit 5 = 0) works in a different way, depending on following countries:
Austria
: Three-Way Calling Tone (Add-On Ticker) is sent at a cycle of four seconds.
EMEA except for Austria
: Three-Way Calling Tone (Add-On Ticker) is sent at a cycle of 32 seconds.
North America and Australia
: Three-Way Calling Tone is not sent.

The following are the service conditions of Three-Way Calling Tone (Add-On Ticker).

a.) Three-Way Calling Tone (Add-On Ticker) can be sent when Three-Way Calling is established by the following operations.

- Performing switch hook flash
- Pressing the CONF key of Dterm/DtermIP/DT Series
- Pressing the CONF key of Attendant Console

b.) Three-Way Calling Tone (Add-On Ticker) can be sent to all the parties in Three-Way Calling.
c.) This feature does not support Three-Way Calling created by OAI or Hotel feature.
d.) Three-Way Calling Tone (Add-On Ticker) supports the following MGs.

Product Name	F/W Product Name	Remarks
MG(BRI) [MG-2BRIA]	SP-3826 BRI MG SIP BSC PROG-A Issue 2 or later	–
MG(BRI) [SCA-2BRIA]		
MG(PRI) (1.5M) [MG-24PRIA]	SP-3884 MG PRI PROG-A Issue 3 or later	SIP is supported for 5A or later.
MG(PRI) (1.5M) [SCA-24PRIA]		
MG(PRI) (2M) [MG-30PRIA]		
MG(PRI) (2M) [SCA-30PRIA]		
MCMG [MG-4LC2COTA]	SP-3778 MCMG PROG-A Issue 1 or later SP-3927 MCMG PROG-B Issue 1 or later **Note 3**	–
MCMG [SCA-4LC2COTA]		
MG-COT Card [SCA-6COTA]	SP-3875 6MGCOT PROG-A Issue 1 or later **Note 3**	North America is supported.
MG-COT Card [SCA-6COTB]	SP-3948 6MGCOT PROG-A Issue 1 or later	EMEA is supported.
MG-COT Card [SCA-6COTC]	SP-3948 6MGCOT PROG-A Issue 1 or later **Note 3**	North America is supported.

Product Name	F/W Product Name	Remarks
UG50 (PRT 1.5M)	Any version	–
UG50 (PRT 2M)		
UG50-24PRIA		
UG50-30PRIA		
UG50 (COT-TYPE1)		
UG50 (COT-TYPE2)		
UG50-4LC2COTA (COT)		
UG50-6COT		

Note 3: EMEA is not supported.

e.) Three-Way Calling Tone (Add-On Ticker) is not transmitted to the operator and terminal connected via PHC.

f.) This feature is not available for MG(BRI) Card [SCA-2BRIA]/MG(BRI) Box [MG-2BRIA]/ MG(BRI) [SCA-2BRIA-B].

g.) The following table shows the specifications of Three-Way Calling Tone (Add-On Ticker).

Country	Frequencies	Cadence	Time Slot
America	425 Hz-3 dBm	80ms-on/80ms-off/80ms-on/off	DTG TS = 3
Australia	425 Hz-3 dBm	80ms-on/80ms-off/80ms-on/off	DTG TS = 3
Netherlands	425 Hz-3 dBm	10 times 32ms-on/480ms-off: Repeating every 32s	DTG TS = 3
Belgium	425 Hz-3 dBm	5 times 32ms-on/480ms-off: Repeating every 32s	DTG TS = 3
Sweden	1400 Hz-27 dBm	352ms-on,off: Repeating every 32s	DTG TS = 3
UK	425 Hz-3 dBm	2 times 32ms-on/352ms-off: Repeating every 4s	DTG TS = 3
Germany	425 Hz-3 dBm	10 times 32ms-on/480ms-off: Repeating every 32s	DTG TS = 3
Austria	425 Hz-3 dBm	4 times 32ms-on/960ms-off: Repeating every 4s	DTG TS = 3
Denmark	425 Hz-3 dBm	2 times 32ms-on/480ms-off: Repeating every 32s	DTG TS = 3
Swiss	425 Hz-23 dBm	448ms-on: Repeating every 32s	DTG TS = 3
Spain	425 Hz-3 dBm	5 times 32ms-on/480ms-off: Repeating every 32s	DTG TS = 3
Greece	950 Hz-13 dBm	480ms-on, off: Repeating every 32s	DTG TS = 3
Italy	425 Hz-3dBm	10 times 32ms-on/480ms-off: Repeating every 32s	DTG TS = 3

8. If any one party places a Three-Way Call in progress on Call Hold, the other two parties cannot activate Call Hold.
9. When a Three-Party Conference includes an external trunk (C.O. Line or TIE Line), the station which has changed from hold status to Three-Party Conference status cannot be placed on hold again.

10. When a Three-Party Conference Trunk (CFT) has no vacancy, a Three-Party Conference cannot be established. In this case, the connected party switches every time the CONF key is pressed, and the other party will be on hold.

When the Consultation Hold service is used:

1. Service Feature Restriction Class of Stations may be assigned at each Station.
2. This service is not available when Three-Way Calling has been established.
3. After the originating party performs switch hook flash, the party can only go on-hook and perform switch hook flash. The party in conversation can only operate on-hook.
4. This service is not available when the Transferred Call is held by CALL HOLD and so on.
5. When the additional party answers the Held Call, whether to send Override Tone or not depends on the originating party's Service Feature Restriction Class.
6. The Conference Trunk (CFT) function is required for the Node which accommodates the originating party of the Three-Way Call within the FCCS Network.
7. Three-Way Calling can also be established when using the Trunk belonging to the Remote Node within the FCCS Network.

<The following conditions are applied for Desktop terminals only.>

1. The following LCD display patterns exist when a Three-Way Call is established on a Desktop terminal:

   - Pattern 1: Originating party is in conversation with the additional party before a Three-Way Call is established.

     a.) Originating party (Station A) is in conversation with the first party (Station B).
     b.) Station A performs switch hook flash. Additional party (Station C) is dialed. Conversation is established with Station C.
     c.) Station A presses CONF key. Station A/B/C displays "CONF" on its LCD.

   - Pattern 2: Originating party is in conversation with the first party before a Three-Way Call is established.

     a.) Originating party (Station A) is in conversation with the first party (Station B).
     b.) Station A performs switch hook flash. Additional party (Station C) is dialed. Conversation is established with Station C.
     c.) Station A presses TRANSFER key. Station C is placed on hold. Station A and Station B is in conversation.
     d.) Station A presses CONF key. Station A displays "CONF" on its LCD. Station B/C displays "CNF" on its LCD.

Interactions

1. A Station in an established Three-Way Call cannot perform switch hook flash to activate a feature. The flash disconnects the last party connected.
2. All Three-Way Calls are protected by privacy. No more than three ports can be bridged together at one time, unless PRIVACY RELEASE [P-18] is being used.
3. If a Three-Way Call includes two Trunks and the Station hangs up, the Trunks will be released, unless the Trunks are allowed TRUNK-TO-TRUNK CONNECTION [T-10] via the ARRC command, ARI D-RES.
4. SP Controlled SIP terminals do not support Three-Way Calling features using CFT or VS32 unless brought into the Three-way call by another station.

   SIP Handler Controlled SIP terminals support Three-Way Calling features using CFT or VS32 but they cannot receive Three-Way Calling Tones (Add-On Ticker) for Three-Way Calling features.
   SIP Handler Controlled SIP terminals do not support Three-Way Calling features which is established using the audio mixing function of the SIP Handler Controlled SIP terminals.

Programming

STEP 1: **ASYD**-SYS 2, Index 1, Bits 0 and 1. Consultation Hold allowed or denied. 00: Denied; 01: Originating and Terminating Calls allowed, Tandem Call Denied; 10: Only Terminating Calls Allowed; 11: All calls allowed.

SYS 2, Index 6, Bit 2. This Bit remains as data "0." CALL TRANSFER-ALL CALLS [C-11].

SYS 2, Index 6, Bit 3. Enable add-on ability. This Bit remains as data "0" to allow Three-Way Calling.

SYS 3, Index 2. Assign data 91H for Normal (N), Hooking (H) values.

STEP 2: **ASYDL**-Assign System Data

SYS 1, Index 830, Bit 5: This system data works in a different way, depending on following countries:

- When Bit 5 = 0

  Austria
  : Three-Way Calling Tone (Add-On Ticker) is sent at a cycle of four seconds.
  EMEA except for Austria
  : Three-Way Calling Tone (Add-On Ticker) is sent at a cycle of 32 seconds.
  North America and Australia
  : Three-Way Calling Tone is not sent.

- When Bit 5 = 1

  Three-Way Calling Tone (Add-On Ticker) is sent only once when Three-Way Calling is established.

SYS 1, Index 811, Bits 0-7: Specify the country code for IP terminal Tone. For more information about the country code, refer to Command Manual.

STEP 3: **ATRK**-Assign LENS as Route 909, Digital Conference Trunk. The LENS consists of the following:

XX	X	XX	X
Module Group	Unit	Group	Level

Group and Level numbers for the Digital Conference Trunk-210, 213, 216, 221, 224, 227, 232, 235.

STEP 4: **MBTK**-Make idle the Digital Conference Trunk

STEP 5: **ARRC**-Allow trunk-to-trunk connection between any two routes that may be used in Three-Way Calling. Assign these routes for ARI A-RES: This Connection is for a Station and two Trunks ARI D-RES. If the originating party hangs up, the remaining two parties will stay connected.

IC RT: Incoming External Route Number

OG RT: Outgoing External Route Number

ARI A-RES = 1: Alternate routing is Allowed

ARI D-RES = 1: Alternate routing is Allowed

STEP 6: **ARSC**-The Route Restriction Class must be created to allow the Station access to the involved routes.

STEP 7: **ASFC**-When Consultation Hold Enhancement is used, assign each Station Service Feature Restriction Class (SFC) that allows SFI 138 (Consultation Hold Enhancement) and SFI 140 [Send Warning Tone (Three-Way Calling)].

<**For Desktop terminals**>

STEP 8: **AKYD**-Assign a Feature key that provides CAS SHF on the Desktop terminal.
KYI: 1
FKY: 17

FCCS Programming

This data setting describes only the additional or replacement commands used to activate FCCS Services. When used as an FCCS feature, these commands replace the non-network commands (for example, AAED is replaced by AAEDN). Note that the data setting for FCCS Service varies depending on the Node. **Note 1**

STEP 1: **ALRTN**-Assign this command to connect the Logical Route to the Physical Route.

STEP 2: **ARRCN**-Allow for Trunk-to-Trunk Connections between any two routes that may be used in Three-Way Calling. Assign these routes for ARI A-RES: This Connection is for a Station and two Trunks ARI D-RES. If the originating party hangs up, the remaining two parties will stay connected.

STEP 3: **ARSCN**-The Route Restriction Class must be constructed to allow the Station access to the involved routes.

Note 1: When using ASYDL or ASYDN to program an FCCS feature, the basic programming of ASYD is required in addition to ASYDL or ASYDN.

T-3 TIE Line Access

General Description

This feature allows any Station user dial access to a TIE Line.

Operating Procedure

1. Lift the handset. Receive Dial Tone.
2. Dial the TIE LINE ACCESS Code. Receive Dial Tone from the distant System.
3. Dial the desired extension number.

Service Conditions

1. TIE LINE ACCESS may be denied to individual Stations via CLASS OF SERVICE-INDIVIDUAL [C-15].
2. The System is equipped with either automatic or dial-repeating TIE Trunks.
3. When a power failure occurs at the local System (without reserve power backup), all TIE Trunk Connections are lost.
4. Each TIE Line Group can be programmed for DP or DTMF address signaling Incoming and/or Outgoing.
5. TOLL DENIAL [T-6]: The Calling Station user is routed to Reorder Tone if an attempt is made to dial-access a call.
6. The System can be equipped with Senderized TIE Trunks. Immediate start, delay, or wink start signaling is available.
7. When a Trunk Group Access Code is dialed by a user, the TIE Trunk Group is used to index a Trunk Group restriction table to determine if the call attempt is allowed. For each outgoing Trunk Group, the following restriction options can be assigned per RSC:
   a.) Indicates the TIE Trunk is not allowed access to the Trunk Group dialed.
   b.) Indicates the TIE Trunk is allowed access (unrestricted).
   c.) Indicates the Station is allowed access but is restricted from making 0/1 toll calls or is Code (three-digit, six-digit) restricted.
8. Two Outgoing Trunk Group restriction tables are provided per system. The Direct Dial Access table is indexed when a Station user dials the Trunk Group Access Code and the ATTENDANT CONSOLE [A-3] table is indexed on Attendant completed calls. The restriction options per RSC can differ per table, allowing Attendant completion of Outgoing Calls for restricted Stations.
9. DTMF cannot be received on an Incoming TIE Line Call when the Signal Interface for Incoming Route Data is set for Immediate Start.

Interactions

1. For Senderized TIE Lines, see UNIVERSAL SENDER [U-3]. The ANND command must be programmed for Senderized TIE Lines (normally for 4(OGC)).
2. If other services, such as STATION MESSAGE DETAIL RECORDING (SMDR) [S-10] and LEAST-COST ROUTING-3/6 DIGIT [L-5] are to be used in conjunction with this feature, the AMND command must be programmed. Go to the ASTP and AOSP command, if digit translation is required.
3. MISCELLANEOUS TRUNK RESTRICTION [M-3]: Restrictions to TIE Trunks are determined by the Route Service Class (RSC) assigned to the Station.

Programming

STEP 1: **ARTD** - Assign the TIE Line routes as shown below:

BOTHWAY LINES:

RT: 1

1-OSGS	:2/7	2-ONSG	:3	3-ISGS	:2/7	4-INSG	:3	5-TF	:3
6-TCL	:4	7-L/T	:1	8-RLP	:2	13-AC	:1	15-LSG	:5
28-ANS	:1	30-PAD	:7						

Note: To change the data for existing route (RT), initialize the circuit card.

STEP 2: **ATRK** - Assign the LENS, TIE Line Route Number, Trunk Number, and Tenant Number.

STEP 3: **MBTK** - Assign the Make Idle status to the TIE Line Trunks.

STEP 4: **ANPD** - Assign a Number Level for Trunk Access. Select Normal Service (N) or Hooking Service (H) from the combo box for Connection Indexes. Number of Necessary Digits is usually one or two. Busy Lamp Field is not activated.

STEP 5: **ASPA** - Assign the Access Code as assigned in ANPD above. Assign Connection Indexes, CI = "N* Normal" or "H: Hooking". Type of Service, SRV = "OGC: Outgoing Call" for Outgoing Trunk. Assign the Route Number associated with this Access Code.

STEP 6: **ARSC** - The Route Restriction Class must be constructed to allow the Station access to the involved routes.

FCCS Programming

This data setting describes only the additional or replacement commands used to activate FCCS Services. When used as an FCCS feature, these commands replace the non-network commands (for example, AAED is replaced by AAEDN). Note that the data setting for FCCS Service varies depending on the Node.

STEP 1: **ARTDN** - Assign the TIE Line routes as shown below:

BOTHWAY LINES:

RT: 1

1-OSGS	:2/7	2-ONSG	:3	3-ISGS	:2/7	4-INSG	:3	5-TF	:3
6-TCL	:4	7-L/T	:1	8-RLP	:2	13-AC	:1	15-LSG	:5
28-ANS	:1	30-PAD	:7						

STEP 2: **ANPDL** - Assign a Number Level for Trunk Access. Select Normal Service (N) or Hooking Service (H) from the combo box for Connection Indexes.Number of Necessary Digits is usually one or two. Busy Lamp Field is not activated.

STEP 3: **ASPAL** - Assign the Access Code as assigned in ANPD above. Assign Connection Indexes, CI = "N* Normal" or "H: Hooking". Type of Service, SRV = "OGC: Outgoing Call" for Outgoing Trunk. Assign the Route Number associated with this Access Code.

STEP 4: **ARSCN** - The Route Restriction Class must be constructed to allow the Station access to the involved routes.

T-5 TIE Line Connection With Pad Control

General Description

This feature provides a switchable transmission pad on TIE Trunks that allows Tandem Connections. Necessary pad control is activated to protect against echo.

Note: For PAD value settings toward IP equipment, see the PAD/EC Control IP/PHC Support section in Chapter 2 of this manual.

Operating Procedure

No manual operation is required.

Service Conditions

1. Pad Control (2dB) will automatically be switched in for the following Connections:
   a.) TIE Line-to-Station
   b.) Station-to-TIE Trunk/CCSA
   c.) CCSA-to-Station
   d.) TIE Line-to-Attendant
   e.) CCSA-to-Attendant
   f.) Attendant-to-TIE Trunk/CCSA
2. Pad Control (2dB) will not be switched in for the following Connections:
   a.) TIE Line-to-Central Office Trunk/TIE/CCSA
   b.) CCSA-to-Central Office Trunk/TIE/CCSA
   c.) Central Office Trunk-to-TIE/CCSA
   d.) THREE-WAY CALLING [T-2] used in conjunction with TRUNK-TO-TRUNK CONNECTION [T-10]
3. The 4W E&M Trunk/DTI [D-16] provides PAD CONTROL (0,2,4,8,12,16dB) so that the desired transmitting and receiving Levels can be maintained, providing the overall Tandem system with transparency.

Interactions

1. If volume on the DTI is too loud, assign CDN 30:Pad (ARTD) as data “2”.
2. The TLT card provides four Two-wire DID, CCSA, and TIE Line Circuit Access. These circuits may be set for Loop or E&M signaling depending on the specific card. The card contains switchable transmission pads that are switched in or out via software command.
3. The TLT card provides four 4-wire TIE Line Circuits. These circuits may be set for E&M signaling. The card contains switchable transmission pads that are switched in or out via software.

Programming

STEP 1: **ARTD** - Assign the TIE Line routes as shown below:

BOTHWAY LINES:

RT: 1

1-OSGS	:2	2-ONSG	:3	3-ISGS	:2	4-INSG	:3	5-TF	:3
6-TCL	:4	7-L/T	:1	8-RLP	:2	15-LSG	:5		

Pad control is programmed in CDN 30:Pad. For DTI circuit cards, assign data "7", otherwise assign data "0".

Note: To change the data for existing route (RT), initialize the circuit card.

STEP 2: **APAD** - Assign the Pad data required for the Tandem Connection. Assign the following:

KIND: Kind of Connection
T: Tandem Connection
ICRT: Incoming Route Number
OGRT: Outgoing Route Number
SFC: Service Feature Restriction Class ⟶ [Not Used]
IC PAD: Incoming PAD Data (1-15)
OG PAD: Outgoing PAD Data (1-15)
RT: Route Number ⟶ [Not Used]

PAD DATA: See PAD DATA configuration for each card in the Circuit Card Description manual.

FCCS Programming

This data setting describes only the additional or replacement commands used to activate FCCS Services. When used as an FCCS feature, these commands replace the non-network commands (for example, AAED is replaced by AAEDN). Note that the data setting for FCCS Service varies depending on the Node.

STEP 1: **ARTDN** - Assign the TIE Line routes as shown below:

BOTHWAY LINES:

RT: 1

1-OSGS	:2	2-ONSG	:3	3-ISGS	:2	4-INSG	:3	5-TF	:3
6-TCL	:4	7-L/T	:1	8-RLP	:2	15-LSG	:5		

Pad control is programmed in CDN 30:Pad. For DTI circuit cards, assign data "7", otherwise assign data "0".

STEP 2: **APADN** - Assign the Pad data required for the Tandem Connection. Assign the following:

KIND: Kind of Connection
T: Tandem Connection
ICRT: Incoming Route Number
OGRT: Outgoing Route Number
SFC: Service Feature Restriction Class ⟶ [Not Used]
IC PAD: Incoming PAD Data (1-15)
OG PAD: Outgoing PAD Data (1-15)
RT: Route Number ⟶ [Not Used]

PAD DATA: See PAD DATA configuration for each card in the Circuit Card Description manual.

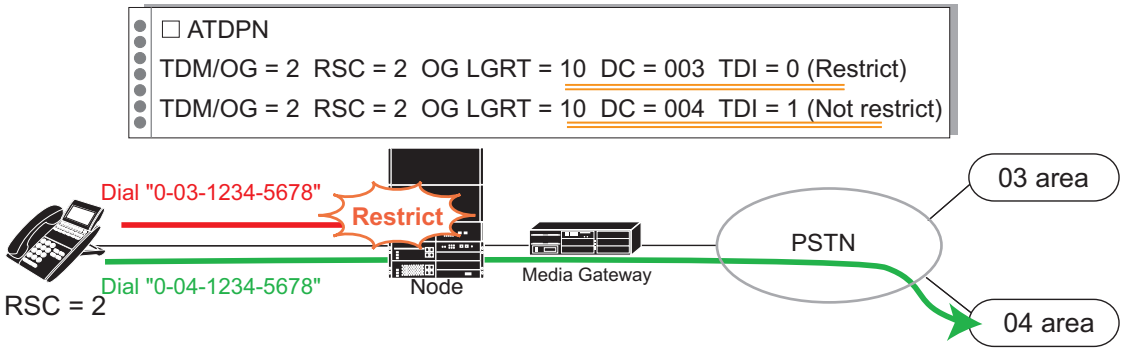
T-6 Toll Denial/Toll Diversion

General Description

This feature prohibits Station users from placing Long Distance Calls over a specific group of Trunks. Restricted calls are routed to either an ATTENDANT CONSOLE [A-3] (diversion) or to an intercept tone (denial).
Toll calls can be restricted on the basis of Tenant Number (TN) of originating station/incoming trunk by ATDPEL/ATDPEN command, combining the Route Restriction Class (RSC) setting. See the examples below:

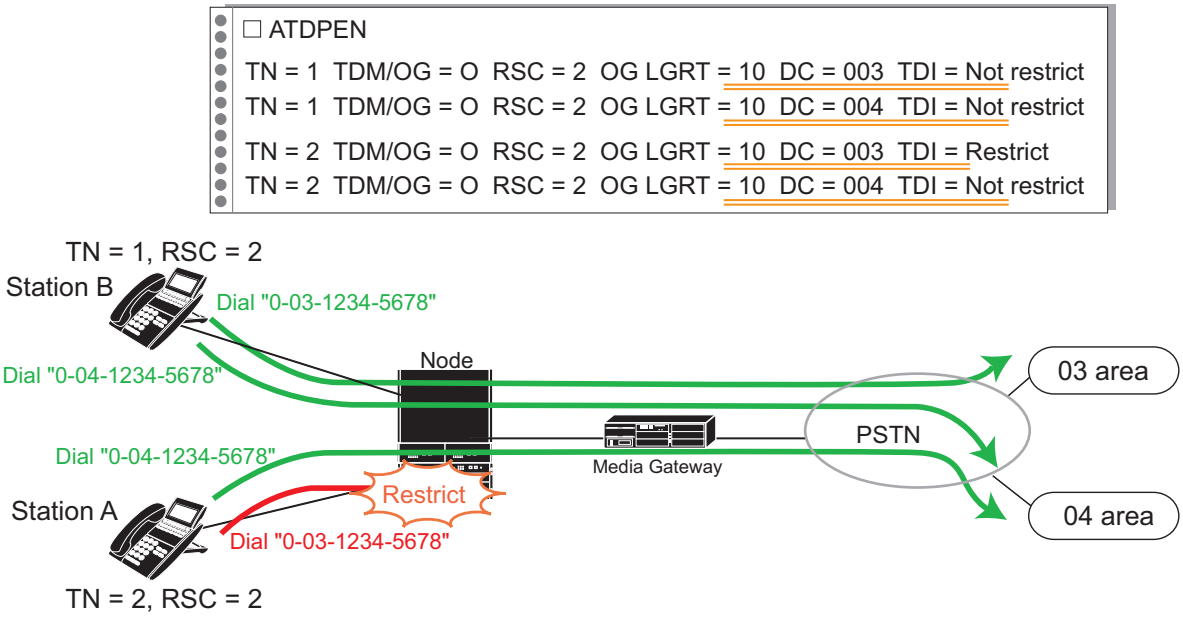
<e.g.1> Toll call restriction by RSC (with ATDP/ATDPL/ATDPN command)

When restricting outside calls to 03 area from the Station with RSC = 2:



<e.g.2> Toll call restriction by RSC for each TN (with ATDPEL/ATDPEN command)

When restricting outside calls to 03 area from the station with RSC = 2 and TN = 2 (Station A):



When the stations with the same Route Restriction Class (RSC = 2 is set to both Station A and Station B) dial the same destination, Station B (TN = 1) can originate outside calls to 03 area while Station A (TN = 2) is restricted.

Operating Procedure

No manual operation is required after this feature is programmed into office data.

Service Conditions

1. TOLL DENIAL/TOLL DIVERSION is the industry-standard 0/1 type.
2. Immediately after dialing any Area Code with a second digit of "0" or "1", the user is restricted.
3. Restriction may be programmed to give the Station user a busy signal or to divert the user to the Attendant Console.
4. This feature may be applied to all types of Trunks (DDD, FX, WATS), provided the same restrictions apply to all Trunks of the same type.
5. This feature is applied to individual Stations on a route-restriction basis.
6. The 0/1 Toll Denial and Toll Diversion feature cannot be used together for the same Tenant.
7. The 0/1 Toll Denial and Toll Diversion feature is assigned on a per line (RSC) Class of Service and/or per-Incoming TIE Trunk Group basis. The feature, when provided, is applicable on a per Trunk Group basis.

Conditions for Tenant Number (TN) based TOLL DENIAL/TOLL DIVERSION

1. TOLL DENIAL/TOLL DIVERSION is applied to individual stations by the setting of Tenant Number (TN), Route Restriction Class (RSC), and outgoing trunk.
2. When toll call from station is restricted by Route Restriction Class (RSC),
   - Toll call restriction can be released on the basis of Tenant Number (TN) and Destination Code (DC) by using ATDPEL/ATDPEN command.
   - Toll call by Speed Dialing is restricted/allowed by Service Feature Restriction Class (SFC).
3. Toll call in tandem connection is restricted on the basis of Tenant Number (TN) and Destination Code (DC) by using ATDPEL/ATDPEN command.
4. This feature supports the outgoing calls of LCR/LCRS. Therefore, toll call from the restricted station is denied after dialing all the digits.
5. Toll call from Sub Line is restricted by the TN of My Line.
6. The ACD and Hotel systems do not support this feature.
7. The OAI system supports this feature only for station-to-outgoing-trunk calls. To configure the OAI system to restrict specific outgoing calls (**Note 1**), use the ARSC and ATDP commands (or ARSCN and ATDPL/ADTPN). For details, see "SCF Restriction" in Data Programming Manual - OAI.

   Note 1: The ARRC/ARRCN or ATDPEL/ATDPEN command cannot be used for this purpose.

8. Toll call from visitor PS (roaming in other node) is restricted by the TN and RSC assigned for visitor PS number on the visitor node.
9. Be sure to assign a same value to TN and UGN when Dial Plan to Support IP Centrex Functionality [D-166] is used.
10. ATDPEL/ATDPEN command does not support Toll Restriction - 3/6-Digit [T-7] and operator call to outside.

11. The following commands work in the priority order below:

ATDPEL/ATDPEN	1	high
ATDPN	2	↑
ATDPL	3	
ATDP	4	low

12. When toll call is restricted to each tenant of originating station/incoming trunk, use ATDPEL command for the nodes without FPC setting, and use ATDPEN command for the nodes with FPC setting.
13. When a trunk call is manually transferred from a station to an outside party using trunk tandem connection, parameters used to implement a routing restriction policy depend on the System Data (ASYDL/ASYDN, SYS1, Index 1393, Bit 5).

Note: The System Data (ASYDL/ASYDN, SYS1, Index 1393, Bit 5) is available since FP95-113 V3.

Interactions

1. TOLL DENIAL/DIVERSION is applicable only to RES 2 when assigning routes via Route Restriction Index (RRI) 3.
2. LEAST-COST ROUTING-3/6 DIGIT [L-5]: For LCR(S) applications, allow the LCR Flag Route. The route chosen via AOPR will then be governed by the RSC of the Station. If the Station is toll-restricted from the route, TOLL DENIAL/DIVERSION will be applied.
3. TOLL DENIAL and TOLL DIVERSION cannot be used simultaneously in the same Tenant.
4. ROUTE ADVANCE [R-6]: Station A dials a Toll Call by accessing route 1. Route 1 is busy and advances to route 2. Station A is restricted from making this call on route 2.
5. PASSING DIAL TONE [P-2]: The Attendant Console cannot use the PASSING DIAL TONE [P-2] feature to allow a toll restricted Station to dial a restricted number.
6. In System Data 1, Index 135, a seizure guard timer exists to protect the system from individuals who attempt to defeat TOLL RESTRICTION/TOLL DIVERSION with rapid dialing. Leave as data “00” for a default value of one second (RAM Data is 1FH).
7. A Station uses the OGC direct dial access to a route. The Station is restricted via RES 0: Connection Restricted. The Station receives Reorder Tone.
8. This feature is not available for the Attendant Console.

Programming

STEP 1: **ASYD** - System Data 1, Index 75. Should the following commands be developed for Day/Night: 0/1: No/Yes. See DAY/NIGHT CLASS OF SERVICE [D-15] for details.

Bit 0- Develop ARSC/ARSCN command for Day/Night?

Bit 3- Develop ATDP/ATDPL/ATDPN command for Day/Night?

System Data 2, Index 1, Bits 4 and 5. If TOLL DENIAL is required, assign these Bits as data “00”. If TOLL DIVERSION is required, assign these Bits as “01”. Calls are then diverted to the Attendant Console. Assigned on a per Tenant basis.

STEP 2: **ARSC** - Assign Route Restriction Classes. When these are assigned to Stations, classes will determine whether or not a Station will be allowed access to a specific route or specific dialing sequence. See TOLL RESTRICTION-3/6 DIGIT [T-7]. Assign the following:

DAY/NIGHT: Day/Night Mode

D: Day Mode
N: Night Mode

TN: Tenant Number
RT: Route Number
RSC: Route Restriction Class
RRI: Route Restriction Index
0: IC Restriction via Attendant Console (DDD, etc.)
1: IC Restriction Direct Connection (DID, etc.)
2: OG Restriction via Attendant Console
3: OG Restriction Direct Connection (DDD, etc.)

RES: Restriction Data
0: Connection is Restricted
1: Connection is Allowed
2: Connection is Toll Restricted (See TOLL RESTRICTION-3/6-DIGIT [T-7])

STEP 3: **ASDT** - When assigning the data to specific Stations, assign the correct RSC to each Station.

FCCS Programming

This data setting describes only the additional or replacement commands used to activate FCCS Services. When used as an FCCS feature, these commands replace the non-network commands (for example, AAED is replaced by AAEDN). Note that the data setting for FCCS Service varies depending on the Node.

Note: When using ASYDL or ASYDN to program an FCCS feature, the basic programming of ASYD is required in addition to ASYDL or ASYDN.

STEP 1: **ALRTN** - Assign this command to connect the Logical Route to the Physical Route.

STEP 2: **ASYDL** - Index 640, assign FPC of the Attendant Console for Toll Diversion

STEP 3: **ARSCN** - Assign Route Restriction Classes. When these are assigned to Stations, classes will determine whether or not a Station will be allowed access to a specific route or specific dialing sequence. See TOLL RESTRICTION-3/6 DIGIT [T-7]. Assign the following:

DAY/NIGHT: Day/Night Mode
D: Day Mode
N: Night Mode

TN: Tenant Number
RT: Route Number
RSC: Route Restriction Class
RRI: Route Restriction Index
0: IC Restriction via Attendant Console (DDD, etc.)
1: IC Restriction Direct Connection (DID, etc.)
2: OG Restriction via Attendant Console
3: OG Restriction Direct Connection (DDD, etc.)

RES: Restriction Data
0: Connection is Restricted
1: Connection is Allowed
2: Connection is Toll Restricted (See TOLL RESTRICTION-3/6-DIGIT [T-7])

<Programming for Tenant based TOLL DENIAL/TOLL DIVERSION>

STEP 1: **ASYD** - System Data 2

Index 1 b4 and b5: Connection pattern for Toll Restriction
b5 = 0, b4 = 0: TOLL DENIAL (Reorder Tone connection)
b5 = 0, b4 = 1: TOLL DIVERSION (Terminated to Attendant Console)
b6: Speed Calling overrides Toll Restriction (on SFC base).
0/1 = Out of service/In service **Note 2**

Note 2: When this system data is assigned to 1 = In service, stations with TOLL DENIAL/TOLL DIVERSION can originate toll calls using Speed Calling assigned by AABD command.

STEP 2: **ASYDL/ASYDN** - System Data 1

Index 802 b1: Tenant data table development for ATDPEL/ATDPEN. 0/1 = Tenant base/Common **Note 3**, **Note 4**

Index 1393 b5: Tandem connection control for manual trunk-to-trunk call transfer via station **Note 5**
0 = Control based on ARI A-RES, RRI = 3 and ARI D-RES
- For manual call transfer to another trunk:
Restriction Data for Alternative Route Index (ARI A-RES) of the ARRC/ARRCN command and Route Restriction Index 3 (RRI = 3) of the ARSC/ARSCN command
- For drop of the transferring party from the call:
Restriction Data for Alternative Route Index (ARI D-RES) of the ARRC/ARRCN command

1 = Control based on ARI A-RES
- For manual call transfer to another trunk:
Restriction Data for Alternative Route Index (ARI A-RES) of the ARRC/ARRCN command
- For drop of the transferring party from the call:
Restriction Data for Alternative Route Index (ARI A-RES) of the ARRC/ARRCN command

b6: Tandem connection for manual trunk-to-trunk call transfer when the ARI A-RES parameter of the ARRCN command is set to 2 (Toll Restriction is required) in the following cases: **Note 5**
- ASYDL/ASYDN, SYS1, Index 1393, b5 = 0
- Manual trunk-to-trunk call transfer via an attendant

0/1 = Allowed/Restricted

Note 3: When assigning toll call restriction data by ATDPEN command, be sure to assign the same data on Network Control Node (NCN) and Local Node (LN).

Note 4: When ASYDL/ASYDN, SYS1, Index 802, Bit 1 = 1 (Tenant data table is common) is assigned, TN = 1 is always applied in ATDPEL/ATDPEN command even when other Tenant is selected.

Note 5: Available since FP95-113 V3.

STEP 3: **ASYDL** - System Data 1

Index 1121 b0: Day/Night data table for ATDPEL/ATDPEN.
0/1 = Common/Separate **Note 6**, **Note 7**
b1: Day/Night data table for ATDPEL/ATDPEN in tandem connection.
0/1 = Common/Separate **Note 6**, **Note 7**

Note 6: When assigning toll call restriction data by ATDPEN command, be sure to assign the same data on Network Control Node (NCN) and Local Node (LN).

Note 7: When Day/Night data table is assigned to 0 = Common, D/N = D (Day) is always applied in ATDPEL/ATDPEN command even when N (Night) is selected.

STEP 4: **ASYDN** - System Data 1

Index 800 b6: Tenant data table for ARSCN command in NDM.
0/1 = Tenant base/Common

STEP 5: **AMND/AMNDL/AMNDN** - Maximum Necessary Digits Data. Assign MND to Destination Code. **Assign on each node.*

TN: Tenant Number
DC: Destination Code (= Access Code + Area/Office Code)
MND: Maximum Necessary Digits to DC
TOLL: Local Code/Toll Code
AN: Out of service/Account Number
RATE: 0 (fixed)
A/D: Type of destination party: Analog/Digital

STEP 6: **ARSC/ARSCN** - Route Restriction Class Data. Whether to restrict toll call is determined for RSC and TN of station.

DAY/NIGHT: D/N = Day Mode/Night Mode
TN: Tenant Number
RT: Route Number
RSC: Route Restriction Class
RRI: Route Restriction Index
0: IC Restriction via Attendant Console (DDD, etc.)
1: IC Restriction Direct Connection (DID, etc.)
2: OG Restriction via Attendant Console
3: OG Restriction Direct Connection (DDD, etc.)
RES: Restriction Data
0: Connection is restricted
1: Connection is allowed
2: Toll connection is restricted **Note 8**

Note 8: When RES = 2 is assigned, toll calls from the stations with the specified RSC are restricted.

STEP 7: **ARRC/ARRCN** - Alternative Route Restriction. Assign toll call restriction data between incoming route and outgoing route. **Assign to configure tandem connection.*

Note: In the ARRCN command, use "IC LGRT" and "OG LGRT" in place of "IC RT" and "OG RT" respectively.

IC RT / IC LGRT: External route number of incoming route
OG RT / OG LGRT: External route number of outgoing route
ARI A-RES: 2 = Toll Restriction is required **Note 9**, **Note 10**
ARI D-RES: 2 = Toll Restriction is required **Note 11**

Note 9: This option of the ARI A-RES parameter is only available in the ARRCN command. Select the option when the System Data (ASYDL/ASYDN, SYS1, Index 1393, Bit 5) is set to 1.

Note 10: This option is available since FP95-113 V3.

Note 11: Even when ARI D-RES = 2 is assigned, toll call in tandem connection is not restricted unless restriction data is assigned by ATDPEL/ATDPEN command.

STEP 8: **ATDPEL/ATDPEN** - Expanded Toll Call Restriction data. Toll call is to be restricted/allowed depending on Destination Code and Tenant Number.

TN: Tenant Number of originating station/incoming trunk [1-63]
TDM/OG: TDM = Tandem Connection **Note 12**, **Note 13**
OG = Outgoing Connection **Note 14**
D/N: Day Mode/Night Mode
RSC: Route Restriction Class of originating station [1-15] **Required when OG is selected.*
IC RT/LGRT: Incoming Route Number [1-255/1-899] **Required when TDM is selected.* **Note 15**
OG RT/LGRT: Outgoing Route Number [1-255/1-899] **Note 15**
DC: Destination Code [Maximum 24 digits] **Note 16**
TDI: Restriction Data [Connection is restricted/allowed] **Select either of the radio button.*

Note 12: To make Tandem Connection effective, assign the alternative route restriction to ARI D-RES = 2 (Toll Connection is restricted) by ARRC command.

Note 13: When TDM (Tandem Connection) is selected, D/N mode cannot be separately assigned. Irrespective of D/N mode setting, restriction data for tandem connection is registered as common data.

Note 14: To make OG (Outgoing Connection) effective, assign RES = 2 (Toll Connection is restricted) by ARSC command.

Note 15: Assign IC LGRT and OG LGRT when ATDPEN command is used. Use ATDPEL command for the nodes without FPC setting, and use ATDPEN command for the nodes with FPC setting.

Note 16: When toll call is to be restricted in the node where all digits are sent out for LCR/LCRS dialing, in DC parameter of ATDPEL/ATDPEN command, add the ACC of ARNP command before the sending digits (the ACC corresponds to the NND of ANPD). Be sure to assign equal or more digits of NND for the ACC by ARNP command.

STEP 9: **AABD** - Speed Calling Restriction Data. Release restriction of outgoing toll call origination by dialing Speed Calling on a SFC basis.

ADC: Abbreviated Digit Code of Speed Calling [Maximum 4 digits]

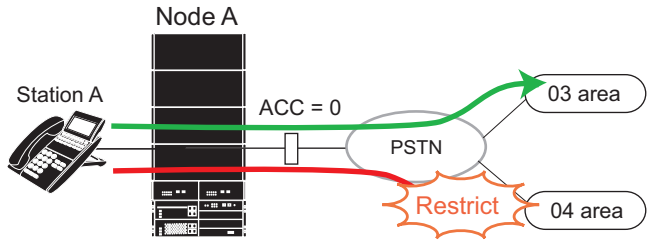
SFC: Service Feature Restriction Class [1-15]

RES: Restriction Data. 0/1 = Connection is restricted/allowed

<Data Programming examples>

a. TOLL DENIAL/TOLL DIVERSION to Outgoing Connection

The following example shows the data programming on Node A to allow Station A (RSC = 2) to originate a toll call only to 03 area:

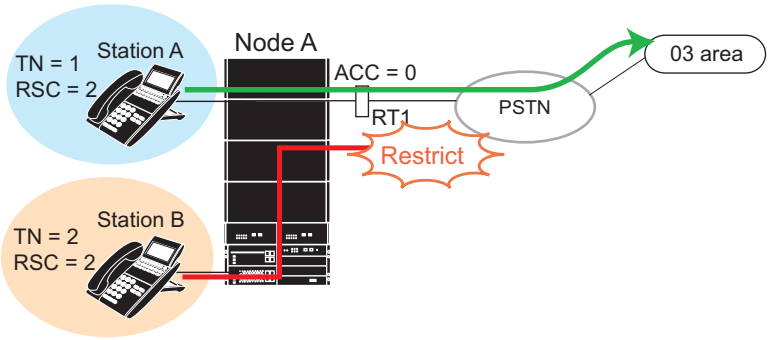


[Data assignment on Node A]

ANPD	1st = 0 NND = 1
ASPA	ACC = 0 SRV = "LCR: Least Cost Routing"
ARNP	RT = RT1 ACC = 0
ARSC	RT = RT1 RSC = 2 RES = 2: Toll connection is restricted
ATDP	TDM/OG = O: Outgoing Connection RSC = 2 OG RT = RT1 DC = 003: Reverse number of external route 0 + Area Code 03 TDI = 1: Connection is Allowed **Note:** Enter the information such as Destination Area Code (DC) about outgoing Toll connections that are allowed. Other outgoing Toll connections are restricted.

*When ATDPEL/ATDPEN is used

The following example shows the data programming on Node A to restrict toll calls from Station A (TN = 1, RSC = 2) to 03 area only and block all toll calls from Station B (TN = 2, RSC = 2):

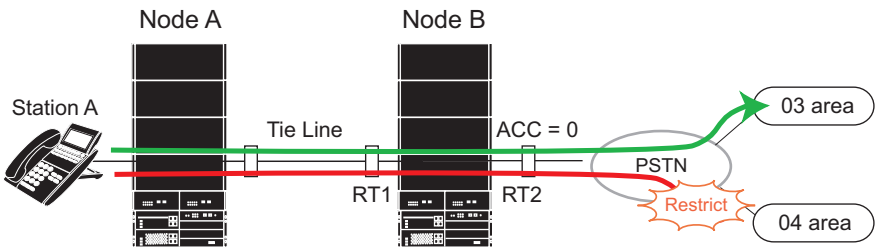


[Data assignment on Node A]

ATDPEL/ATDPEN	TN = 1: Tenant Number of originating trunk TDM/OG = OG: Outgoing Connection RSC = 2 OG LGRT = RT1 DC = 003: Reverse number of external route 0 + Area Code 03 TDI = Connection is Allowed **Note:** Enter the information such as Destination Area Code (DC) about outgoing Toll connections that are allowed. Other outgoing Toll connections are restricted.

b. TOLL DENIAL/TOLL DIVERSION on Intermediate Node in Tandem Connection

The following example shows the data programming on Node B (intermediate node in tandem connection) to prohibit Station A from placing toll calls to 04 area via a tie line. In this case, use the ATDP command to enter the information such as Destination Area Code (DC) about outgoing Toll connections that are restricted.

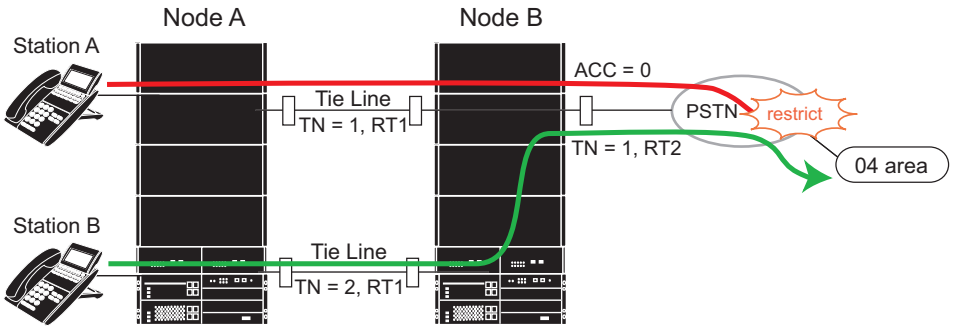


[Data assignment on Node B]

ANPD	1st = 0 NND = 1
ASPA	ACC = 0 SRV = "LCR: Least Cost Routing"
ARNP	RT = RT2 ACC = 0
ARRC	IC RT = RT1 OG RT = RT2 ARI D-RES = 2: Toll Restriction is required
ATDP	TDM/OG = T: Tandem Connection IC RT = RT1 OG RT = RT2 DC = 004: Reverse number of external route 0 + Area Code 04 TDI = 0: Connection is Restricted **Note:** Enter the information such as Destination Area Code (DC) about tandem Toll connections that are restricted. Other tandem Toll connections are allowed.

*When ATDPEL/ATDPEN is used

The following example shows the data programming on Node B (intermediate node in tandem connection) to block outgoing toll calls from Station A to 04 area via a tie line (TN = 1, RT = 1) only.



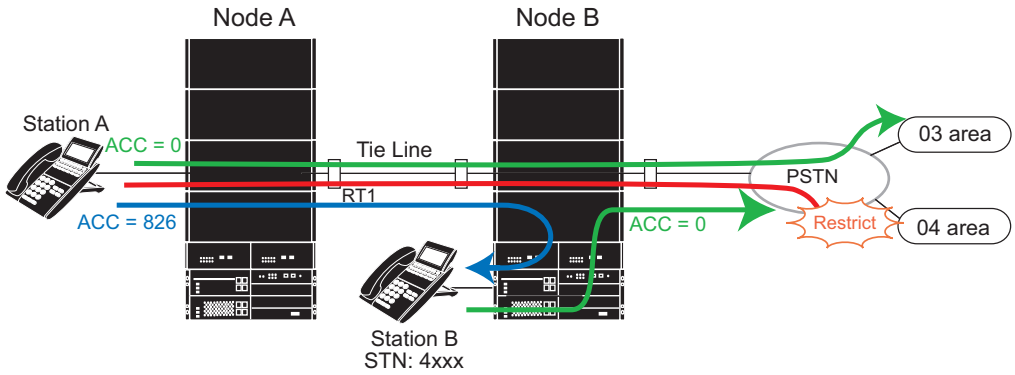
FROM/TO	DIALING NUMBER	TOLL CALL
From stations in Node A to 04 area via Node B using the trunk of TN = 1 and RT = RT1	0-04-xxxxxxxx	Restricted
From stations in Node A to 04 area via Node B using the trunk of TN = 2 and RT = RT1	0-04-xxxxxxxx	Allowed

[Data assignment on Node B]

ATDPEL/ATDPEN	TN = 1: Tenant Number of incoming trunk TDM/OG = TDM: Tandem Connection IC LGRT = RT1 OG LGRT = RT2 DC = 004: Reverse number of external route 0 + Area Code 04 TDI = Connection is Restricted **Note:** Enter the information such as Destination Area Code (DC) about tandem Toll connections that are restricted. Other tandem Toll connections are allowed.

c. TOLL DENIAL/TOLL DIVERSION on Originating Node in Tandem Connection

The following example shows the data programming on Node A (originating node in tandem connection) to prohibit Station A from placing toll calls via a tie line, except to 03 area. In this case, use the ARSC command to enter restriction data for each Route Restriction Class and use the ATDP command to enter the information such as Destination Area Code (DC) about outgoing Toll connections that are allowed.



FROM/TO	DIALING NUMBER	CALL ORIGINATION
From stations in Node A to stations in Node B	826-4xxxx	Allowed
From stations in Node A to 03 area via Node B	0-03-xxxxxxxx	Allowed
From stations in Node A to 04 area via Node B	0-04-xxxxxxxx	Restricted
From stations in Node B to PSTN line	0 + toll code + xxxxxxxx	Allowed

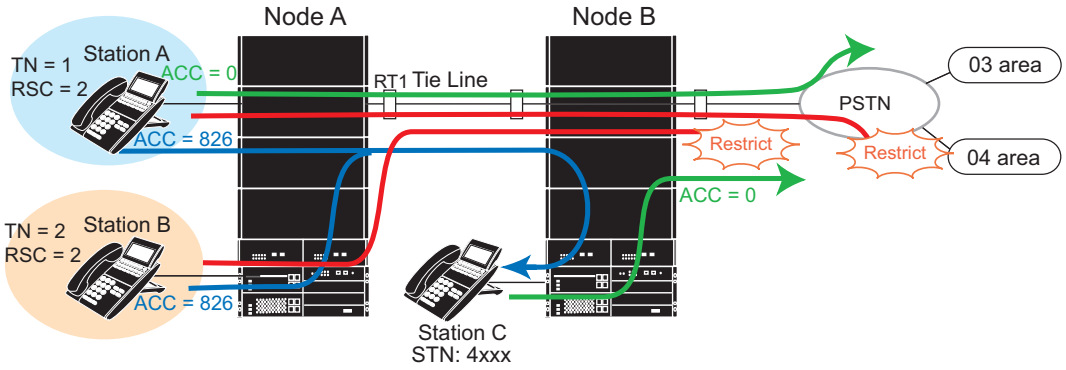
[Data assignment on Node A]

ANPD	1st = 8, NND = 3 1st = 0, NND = 1
ASPA	ACC = 826, SRV = "LCR: Least Cost Routing" ACC = 0, SRV = "LCR: Least Cost Routing"
AFRS	NPC = 826, OPR = 826 NPC = 0, OPR = 0
AOPR	OPR = 826, SKIP = 3 OPR = 0, SKIP = 0
ARNP	<To set ACC of C.O. line> RT = RT1, ACC = 0 <To set ACC of tie line> RT = RT1, ACC = 826

ARSC	RT = RT1 RSC = 2 RES = 2: Toll connection is restricted
ATDP	TDM/OG = O: Outgoing Connection RSC = 2 OG RT = RT1 <When ACC to C.O. line is set in ARNP> DC = 0003: Reverse Number 0+ ACC 0 + Area Code 03 <When ACC to tie line is set in ARNP> DC = 826003: Reverse Number 826 + ACC 0 + Area Code 03 **Underlined numbers are the digits to be sent.* TDI = 1: Connection is Allowed **Note:** Enter the information such as Destination Area Code (DC) about outgoing Toll connections that are allowed. Other outgoing Toll connections are restricted.

*When ATDPEL/ATDPEN is used

The following example shows the data programming on Node A (originating node in tandem connection) to block outgoing toll calls from stations in Node A via a tie line, except for toll calls from Station A (TN = 1, RSC = 2) via a tie line to 03 area.



FROM/TO	DIALING NUMBER	CALL ORIGINATION
From Station A (TN = 1, RSC = 2) in Node A to 03 area via Node B	0-03-xxxxxxxx	Allowed
From Station B (TN = 2, RSC = 2) in Node A to 03 area via Node B	0-03-xxxxxxxx	Restricted
From Stations in Node A to 04 area via Node B	0-04-xxxxxxxx	Restricted
From stations in Node A to stations in Node B	826-4xxxx	Allowed
From stations in Node B to PSTN line	0 + toll code + xxxxxxxx	Allowed

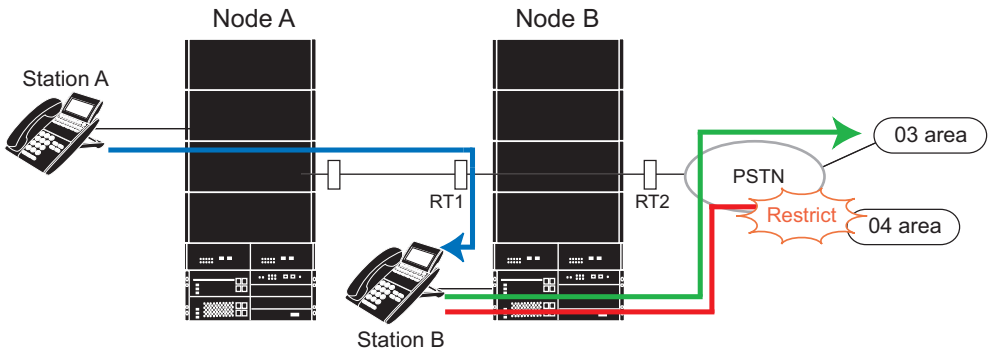
[Data assignment on Node A]

ATDPEL/ATDPEN	TN = 1 TDM/OG = OG: Outgoing Connection RSC = 2 OG LGRT = RT1 <When ACC to C.O. line is set in ARNP> DC = 0003: Reverse Number 0+ ACC 0 + Area Code 03 <When ACC to tie line is set in ARNP> DC = 826003: Reverse Number 826 + ACC 0 + Area Code 03 **Underlined numbers are the digits to be sent.* TDI = Connection is Allowed **Note:** Enter the information such as Destination Area Code (DC) about outgoing Toll connections that are allowed. Other outgoing Toll connections are restricted.

d. TOLL DENIAL/TOLL DIVERSION of Trunk to Trunk (Tandem) Connections via Attendant Console or Station

- When ASYDL/ASYDN, SYS1, Index 1393, Bit 5 = 0:
  Tandem connection control based on ARI A-RES, RRI = 3 and ARI D-RES for manual trunk-to-trunk call transfer via station

The following example shows the data programming on Node B (intermediate node in tandem connection) to prohibit Station B (RSC = 2) from transferring calls from Station A to 04 area. In this case, use the ATDP command to enter the information such as Destination Area Code (DC) about outgoing Toll connections that are allowed.

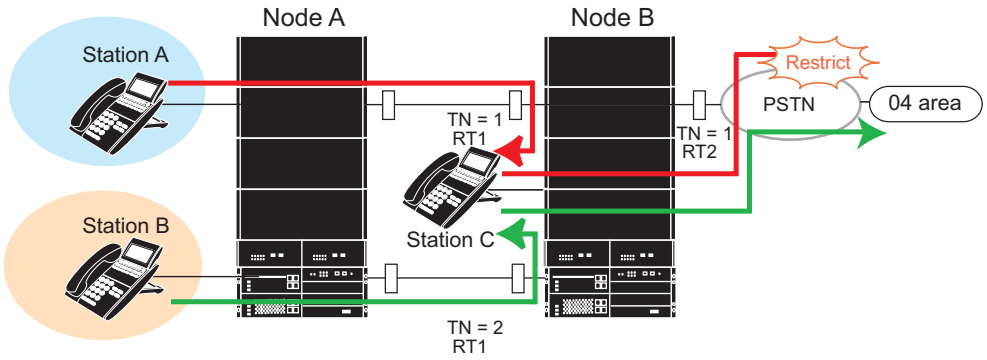


[Data assignment on Node B]

ASYDL/ASYDN	SYS 1, Index 1393, Bit 5 = 0
ANPD	1st = 0 NND = 1
ASPA	ACC = 0 SRV = "LCR: Least Cost Routing"
ARNP	RT = RT2 ACC = 0
ARSC	RT = RT2 RSC = 2 RES = 1: Toll connection is allowed
ARRC	IC RT = RT1 OG RT = RT2 ARI A-RES = 1: Alternate routing is Allowed ARI D-RES = 1: Alternate routing is Allowed
ATDP	TDM/OG = O: Outgoing Connection RSC = 2 OG RT = RT1 DC = 003: Reverse number of external route 0 + Area Code 03 TDI = 1: Connection is Allowed **Note:** Enter the information such as Destination Area Code (DC) about outgoing Toll connections that are allowed. Other outgoing Toll connections are restricted.

*When ATDPEL/ATDPEN is used

The following example shows the data programming on Node B (intermediate node in tandem connection) to prohibit Station C (RSC = 2) from transferring calls from Station A via Tenant 1 (TN = 1) of Route 1 (RT = 1) to 04 area, while allowing Station C to transfer calls from Station B via Tenant 2 (TN = 2) of Route 1 (RT = 1) to the area.



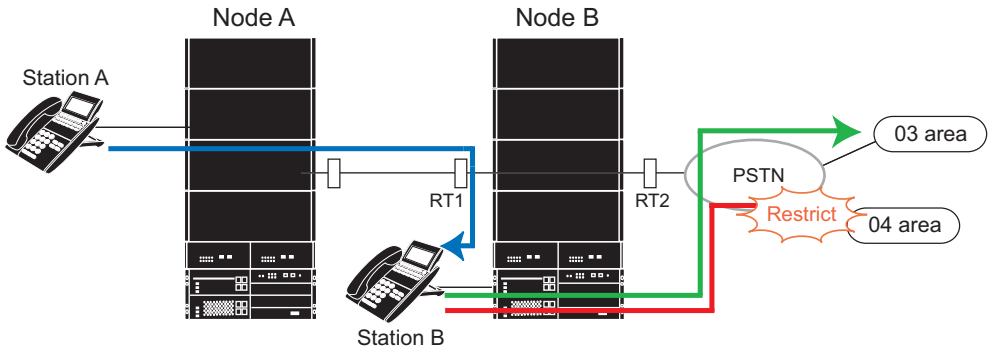
FROM/TO	DIALING NUMBER	TOLL CALL
From stations in Node A to 04 area via Node B using the trunk of TN = 1 and RT = RT1	0-04-xxxxxxxx	Restricted
From stations in Node A to 04 area via Node B using the trunk of TN = 2 and RT = RT1	0-04-xxxxxxxx	Allowed

[Data assignment on Node B]

ASYDL/ASYDN	SYS 1, Index 1393, Bit 5 = 0
ATDPEL/ATDPEN	TN = 1: Tenant Number of incoming trunk TDM/OG = OG: Outgoing Connection RSC = 2 OG LGRT = RT2 DC = 004: Reverse number of external route 0 + Area Code 04 TDI = Connection is Allowed **Note:** Enter the information such as Destination Area Code (DC) about tandem Toll connections that are allowed. Other tandem Toll connections are restricted.

- When ASYDL/ASYDN, SYS1, Index 1393, Bit 5 = 1: **Note 17**
  Tandem connection control based on ARI A-RES for manual trunk-to-trunk call transfer via station

The following example shows the data programming on Node B (intermediate node in tandem connection) to prohibit Station B from transferring calls from Station A to 04 area. In this case, use the ATDP command to enter the information such as Destination Area Code (DC) about outgoing Toll connections that are restricted.



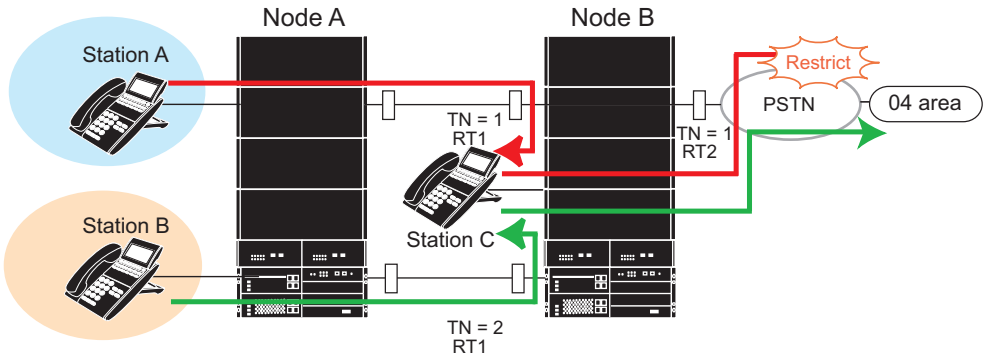
[Data assignment on Node B]

ASYDL/ASYDN	SYS 1, Index 1393, Bit 5 = 1
ANPD	1st = 0 NND = 1
ASPA	ACC = 0 SRV = "LCR: Least Cost Routing"
ARNP	RT = RT2 ACC = 0
ARRCN	IC LGRT = RT1 OG LGRT = RT2 ARI A-RES = 2: Toll Restriction is required.
ATDP	TDM/OG = T: Tandem Connection IC RT = RT1 OG RT = RT2 DC = 004: Reverse number of external route 0 + Area Code 04 TDI = 0: Connection is Restricted **Note:** Enter the information such as Destination Area Code (DC) about outgoing Toll connections that are restricted. Other outgoing Toll connections are allowed.

Note 17: Available since FP95-113 V3.

*When ATDPEL/ATDPEN is used

The following example shows the data programming on Node B (intermediate node in tandem connection) to prohibit Station C from transferring calls from Station A via Tenant 1 (TN = 1) of Route 1 (RT = 1) to 04 area, while allowing Station C to transfer calls from Station B via Tenant 2 (TN = 2) of Route 1 (RT = 1) to the area.



FROM/TO	DIALING NUMBER	TOLL CALL
From stations in Node A to 04 area via Node B using the trunk of TN = 1 and RT = RT1	0-04-xxxxxxxx	Restricted
From stations in Node A to 04 area via Node B using the trunk of TN = 2 and RT = RT1	0-04-xxxxxxxx	Allowed

[Data assignment on Node B]

ASYDL/ASYDN	SYS 1, Index 1393, Bit 5 = 1
ATDPEL/ATDPEN	TN = 1: Tenant Number of incoming trunk TDM/OG = TDM: Tandem Connection IC LGRT = RT1 OG LGRT = RT2 DC = 004: Reverse number of external route 0 + Area Code 04 TDI = Connection is Restricted **Note:** Enter the information such as Destination Area Code (DC) about tandem Toll connections that are restricted. Other tandem Toll connections are allowed.

T-7 Toll Restriction - 3/6-Digit

General Description

This feature allows the System to be programmed to restrict Outgoing Calls according to specific area and/or central office codes. This restriction is controlled on the basis of a three-digit Area Code or a six-digit combination area and Office Code Numbering Plan.

Operating Procedure

No manual operation is required.

Service Conditions

1. The programmed restriction pattern may consist of either codes to be allowed or codes that are to be denied.
2. This feature is assigned on a CLASS OF SERVICE-INDIVIDUAL [C-15] basis.
3. When a Toll Call is denied, Reorder Tone will be returned to the Calling Party. If desired, the Calling Party can be routed to the Attendant Console. See TOLL DENIAL/TOLL DIVERSION [T-6] for further details.
4. The Outgoing Call using a Trunk in the Remote Node via FCCS link is available. Therefore, Route Restriction class data must be identical in the FCCS Network. For more detailed information, see the Data Programming Manual - FCCS.

Interactions

1. When Toll Restriction is specified in ARRC, all the possible dialing-patterns must be assigned in ATDP. If office codes within an Area Code do not need to be restricted, allow the Area Code only. Always provide the system with the most economical configuration of digits when constructing TOLL RESTRICTION tables.
2. In the ARRC command, ARI-A-RES must be programmed to allow two Trunks and a Station in a Conference. ARI-D-RES is programmed if the two Trunks are to remain connected after the Station drops off, or if TOLL RESTRICTION applies to a direct Trunk-to-Trunk Connection, such as TIE Line to DDD.
3. ROUTE ADVANCE [R-6]: Station A dials a Toll Call by accessing route 1. Route 1 is busy and advances to route 2. Station A is restricted from making this call on route 2. The call will be restricted.
4. PASSING DIAL TONE [P-2]: The Attendant cannot use PASSING DIAL TONE [P-2] to allow a toll-restricted Station from using a restricted route.
5. OUTGOING TRUNK QUEUING [O-2]: When a Station is restricted from all other routes, a Station user may queue a Trunk to complete a call over unrestricted routes.
6. LEAST-COST ROUTING-3/6-DIGIT [L-5]: Translations in AFRS must be such that the digits dialed in the Trunk-to-Trunk Connection would be recognizable in terms of choosing a route. If a route cannot be chosen, TOLL RESTRICTION 3/6 DIGIT cannot be applied.
7. When using the FCCS Network, the Outgoing Call using a Trunk in the Remote Node is available. Therefore, Route Restriction class data must be identical in the FCCS Network. For more detailed information, see the Data Programming Manual - FCCS.

Programming

N + Dialing				
Call Type	**Digits Dialed**	**Digits Needed**	**MND**	**Toll**
Local Calls	9-NNX-XXXX	9-N	8	Local Code
Near Distance Calls	9-1-NNX-XXXX	9-1-NN	9	Toll Code
Long Distance Calls	9-1-NPA-NXX-XXXX	9-1-NP	12	Toll Code

Without N + Dialing				
Call Type	**Digits Dialed**	**Digits Needed**	**MND**	**Toll**
Local Calls	9-NNX-XXXX	9-N	8	Local Code
Long Distance Calls	9-NPA-NXX-XXXX	9-NP	11	Toll Code

Note: The digits underlined are to be listed in the DC (Digit Code) parameter.
Overseas or Operator-assisted Calls should be listed with an MND equal to the calculated number of required digits. Special dialing sequences, such as, “911” with an MND = 4 should also be listed.
N = 2-9, P = 0 or 1, X = 0-9

STEP 1: **ATDP** - This command is used with the AARP command. It designates the digit codes that will be allowed, denied, or toll-restricted. These codes are designated by Route Restriction Class. The Number of Necessary Digits data is used to specify where a 3 or 6-digit translation should begin. The digits to be examined for 3 or 6-digit translation are listed in AARP. Assign the following:

D/N: Day/Night Mode
- 1: Day Mode
- 2: Night Mode

TDM/OG: Tandem Connection/Outgoing Connection
- 1: T (Tandem Connection) **Note 1**
- 2: O (Outgoing Connection)

IC RT: Incoming Route Number (Tandem Connection Only)

RSC: Restriction Class (0-15)

OG RT: Outgoing Route Number

DC: Destination (Area) Code (Maximum of 11 digits)

TDI: Restriction Data *
- 0: Connection is Restricted
- 1: Connection is Allowed
- 2: 3/6 Digit Toll Restriction [Go to AARP]
- 3: Central Office Operator Call 9 + 0.

NND: Number of Necessary Digits (Used only when TDI = 2).

Note 1: If assigning TDM/OG=T (Tandem Connection), data cannot be assigned for the Day and Night modes separately. In this case, the assigned data is used for both Day and Night modes regardless of the setting of DAY/NIGHT parameter.

Note: *TDI 4 may appear when the digits input under the DC parameter are incorrect. The switch is looking for more digits.

STEP 2: **ASYD** - System Data 1, Index 75. Should the following commands be developed for Day/Night? 0/1: No/Yes. See DAY/NIGHT CLASS OF SERVICE [D-15] for more details.

Bit 0- Develop ARSC command for Day/Night?

Bit 3- Develop ATDP command for Day/Night?

System Data 2, Index 1, Bit 3. Does TOLL RESTRICTION apply to SPEED CALLING-STATION [S-21]? 0/1: Yes/No.

System Data 2, Index 1, Bits 4 and 5. If TOLL DENIAL (Busy Tone) is required, assign these Bits as data "00". If TOLL DIVERSION (to Attendant) is required, assign these Bits as "01".

System Data 2, Index 1, Bit 7. Does TOLL RESTRICTION apply to SPEED CALLING-SYSTEM [S-3]? 0/1: Yes/No.

STEP 3: **ARSC** - Assign Route Restriction Classes. When these are assigned to Stations, classes will determine whether or not a Station will be allowed access to a specific route or specific dialing sequence. Assign the following:

DAY/NIGHT: Day/Night Mode
- D: Day Mode
- N: Night Mode

TN: Tenant Number

RT: Route Number

RSC: Route Restriction Class

RRI: Route Restriction Index
- 0: IC Restriction via Attendant Console (DDD, etc.)
- 1: IC Restriction Direct Connection (DID, etc.)
- 2: OG Restriction via Attendant Console
- 3: OG Restriction Direct Connection (DDD, etc.)

RES: Restriction Data
- 0: Connection is Restricted
- 1: Connection is Allowed
- 2: Connection is Toll Restricted

STEP 4: **AMND** - This command indicates the total number of digits to expect, based on the first digits dialed. A "9" + a Local Call will equal 8 digits; "9" + "1" + an Area Code Call will equal 12 digits. Assign the dialed digits information and the corresponding Maximum Necessary Digits (MND) data. Choose either Local Code or Toll Code for the Toll Code Identification (TOLL) parameter.

STEP 5: **AARP** - This command is used to designate digits translated for three digit and six digit restriction. In ATDP, a digit code is listed. The system will examine these digits and if the Restriction Code TDI is assigned as data "2" Toll Restriction, the program will do two operations. The first operation will examine the NND value. If NND is "6", then the AARP information will apply to the sixth digit dialed. For example, assume 9-1-301-555-1212 is dialed. AARP will look at the sixth digit dialed for translation. Translation will begin with 555.

Assign the following:

OGRT: Outgoing Route Number

RSC: Route Restriction Class

FLAG:
- 1: Three Digits
- 2: Six Digits

DC: 555
RES: Restriction Data
0: Restricted
1: Allowed

Note: Only three digits or six digits may be entered for the DC parameter of this command.

TOLL RESTRICTION EXAMPLES:

Several examples of the use of TOLL RESTRICTION are listed below:

RSC 1: Allow local only.
RSC 2: Allow local plus Area Codes 718, 212, 201, and 914.
RSC 3: Allow local plus Area Codes 718, 212, and 201; and office codes 239, 465, and 798 only within Area Code 914.
RSC 4: Allow local plus all Area Codes except 808, 213, 408, and 714.
RSC 5: Allow local plus all Area Codes, but restrict Office Code 976 within each Area Code.

To simplify the examples, only one route, (route 1, DDD) will be used. There will be no difference between Day and Night restrictions. 1 + dialing is assumed with an LCR Access Code of "9".

Example 1 - RSC 1 Assignment (Allow local only)

ATDP Command-
TDM/OG: 2 (O: Outgoing Connection)
D/N: 1 (Day Mode)
OG RT: 1 RSC: 1
DC: 92 TDI: 1 (Connection is Allowed)

Repeat these same assignments for Digit Codes "93" through "99" to allow all local office codes.

Notice that assignments for digit codes "90" and "91", which cover Operator Calls and all 1 + Area Code Calls, are not assigned. Digit Codes not assigned are automatically restricted.

AARP Command-
Not Required. This command is only required when TDI = 2 is assigned in ATDP.

Example 2 - RSC 2 Assignment
(Allow local plus Area Codes 718, 212, 201, and 914)

ATDP Command-
TDM/OG: 2 (O: Outgoing Connection)
D/N: 1 (Day Mode)
OG RT: 1 RSC: 2
DC: 92 TDI: 1 (Connection is Allowed)

Repeat these same assignments for Digit Codes "93" through "99" to allow all local office codes.

To allow Area Codes 718, 212, 201, and 914, repeat the ATDP command for the following Digit Codes:

DC: 91718 TDI: 1 (Connection is Allowed)
DC: 91212 TDI: 1 (Connection is Allowed)
DC: 91201 TDI: 1 (Connection is Allowed)
DC: 91914 TDI: 1 (Connection is Allowed)

This allows these specific Area Codes.

Area Codes not assigned are automatically restricted.

AARP Command-
Not Required.

Example 3 - RSC 3 Assignment
(Allowed local plus Area Codes 718, 212, and 201; and office codes 239, 465, and 798 only within Area Code 914.)
ATDP Command-
TDM/OG: 2 (O: Outgoing Connection)
D/N: 1 (Day Mode)
OG RT: 1 RSC: 3
DC: 92 TDI: 1(Connection is Allowed)
Repeat these same assignments for Digit Codes "93" through "99" to allow all local office codes.
To allow Area Codes 718, 212, and 201, repeat the ATDP command for the following Digit Codes:
DC: 91718 TDI: 1 (Connection is Allowed)
DC: 91212 TDI: 1 (Connection is Allowed)
DC: 91201 TDI: 1 (Connection is Allowed)
To allow office codes 239, 465, and 798 within Area Code 914, assign the ATDP command as follows:
TDM/OG: 2 (O: Outgoing Connection)
D/N: 1 (Day Mode)
OG RT: 1 RSC: 3
DC: 91914 TDI: 2 (Go to AARP Command)
NND: 6 (Begin restriction after 91914, i.e; the 6th digit dialed.)
AARP Command-
OGRT: 1 RSC: 3
FLAG: 1 DC: 239 RES: 1 (Connection is Allowed)
Repeat these same assignments for the following Digit Codes:
OGRT: 1 RSC: 3 FLAG: 1 DC: 465 RES: 1
OGRT: 1 RSC: 3 FLAG: 1 DC: 798 RES: 1
This allows office codes 239, 465, and 798 within Area Code 914. All other office codes are automatically restricted.

Note: This method of assignment allows office codes 239, 465, and 798 to any Area Code that is assigned as TDI = 2 in ATDP. For example, if the assignment for Area Code 718 was changed from TDI = 1 (allowed) to TDI = 2, then the office codes 239, 465, and 798 would also be allowed for Area Code 718.

If different Office Codes are to be allowed in the 718 Area Code (for example 623 and 567), then the assignments for Area Codes 914 and 718 must be made as follows:
ATDP Command-
TDM/OG: 2 (O: Outgoing Connection)
D/N: 1 (Day Mode)
RSC: 3 OG RT:1
DC: 919 TDI: 2 (Go to AARP Command)
NND: 3
TDM/OG: 2 (O: Outgoing Connection)
D/N: 1 (Day Mode)
RSC: 3 OGRT: 1
DC: 917 TDI: 2 (Go to AARP Command)

NND: 3

This tells the system to check AARP tables for restrictions for all Area Codes beginning with 9 and 7.

AARP Command-

OGRT: 1 RSC: 3 FLAG: 2
DC: 914239 RES: 1 (Connection is Allowed)
DC: 914465 RES: 1 (Connection is Allowed)
DC: 914798 RES: 1 (Connection is Allowed)
OGRT: 1 RSC: 3 FLAG: 2
DC: 718623 RES: 1 (Connection is Allowed)
DC: 718567 RES: 1 (Connection is Allowed)

This allows just these office codes within these particular Area Codes. All other area codes beginning with "7" or "9" are also restricted. At this point, if allowing Area Code 714 is desired, assign AARP as follows:

AARP Command-

OGRT: 1 RSC: 3 FLAG: 1
DC: 714 RES: 1 (Connection is Allowed)

Example 4 - RSC 4 Assignment

(Allow local plus all Area Codes except 808, 213, 408, and 714)

ATDP Command-

TDM/OG: 2 (O: Outgoing Connection)
D/N: 1 (Day Mode)
OG RT: 1 RSC: 4
DC: 92 TDI: 1 (Connection is Allowed)

Repeat these same assignments for Digit Codes "93" through "99" to allow all local office codes.

TDM/OG: 2 (O: Outgoing Connection)
D/N: 1 (Day Mode)
OG RT: 1 RSC: 4
DC: 912 TDI: 2 (Go to AARP Command)
NND: 3
DC: 913 TDI: 2 (Go to AARP Command)
DC: 914 TDI: 2 (Go to AARP Command)
\\\ (Through)
DC: 919 TDI: 2 (Go to AARP Command)

This tells the system to check AARP tables for all restrictions for Area Codes.

AARP Command-

OGRT: 1 RSC: 4 FLAG: 1 DC: 808 RES: 0*
OGRT: 1 RSC: 4 FLAG: 1 DC: 213 RES: 0*
OGRT: 1 RSC: 4 FLAG: 1 DC: 408 RES: 0*
OGRT: 1 RSC: 4 FLAG: 1 DC: 714 RES: 0*
OGRT: 1 RSC: 4 FLAG: 2 DC: 000000 RES: 1**

*This assignment restricts the Area Codes listed.

**DC: 000000" indicates all other Destination Codes except those already assigned Destination Codes.

RES: Restriction Data: (0/1)

0: Not assigned Area Codes are restricted
1: Not assigned Area Codes are allowed.

The above assignment allows all other dialing patterns (Area Codes) except 808, 213, 408 and 714.

Example 5 - RSC 5 Assignment
(Allow local plus all Area Codes, but restricted from Office Code 976 within each Area Code)

ATDP Command-

TDM/OG: 2 (O: Outgoing Connection)
D/N: 1 (Day Mode)
OG RT: 1 RSC: 5
DC: 92 TDI: 1 (Connection is Allowed)

Repeat these same assignments for Digit Codes "93" through "99" to allow all local office codes.

TDM/OG: 2 (O: Outgoing Connection)
D/N: 1 (Day Mode)
OG RT: 1 RSC: 5
DC: 912 TDI: 2 (Go to AARP Command)
NND: 6
DC: 913 TDI: 2 NND: 6
(Go to AARP Command)
DC: 914 TDI: 2 NND: 6
(Go to AARP Command)
\\\ (Through) \\\
DC: 919 TDI: 2 NND: 6
(Go to AARP Command)

This tells the system to check AARP tables for all restrictions for Area Codes.

AARP Command-

OGRT: 1 RSC: 5 FLAG: 1 DC: 976
RES: 0 (Restricted)
OGRT: 1 RSC: 5 FLAG: 1 DC: 000000
RES: 1 (Allowed)

This will restrict Office Code 976 and allow every other Office Code.

STEP 6: **ARRC** - Provide Trunk-to-Trunk Connection capabilities according to the requirements of the system. The applicable Trunk-to-Trunk Connections will be those provided by parameter ARI-D-RES Direct Connection; for example, REMOTE ACCESS TO THE SYSTEM [R-2] and TANDEM SWITCHING OF TIE TRUNKS-2/4 WIRE [T-1].

ICRT: Incoming Route Number
OGRT: Outgoing Route Number
ARI-A-RES: This allows/restricts Trunk-to-Trunk and Tandem Connection via the Attendant Console.

0: Alternate routing is Restricted
1: Alternate routing is Allowed

ARI-D-RES: This allows/restricts Trunk-to-Trunk and Tandem Connections via direct-dial access from a Station.

0: Alternate routing is Restricted
1: Alternate routing is Allowed

2: Toll Restriction is Required

STEP 7: **ATDP** - When TOLL RESTRICTION is required for Trunk-to-Trunk Connections, ATDP will provide all the digit information. Therefore, up to 11 digits may be listed as being allowed for a specific Trunk-to-Trunk Connection. TDI = 2 should not be assigned because AARP cannot be used for Tandem Connections.

Example 6 - Trunk-to-Trunk Connection

Allow all Local Calls

ATDP Command-

TDM/OG: 1 (T: Tandem Connection)

IC RT: Route 2 OG RT: 1

DC: 92 TDI: 1 (Connection is Allowed)

Repeat these same assignments for Digit Codes “93” through “99” to allow all local office codes.

Notice that assignments for digit codes “90” and “91”, which cover Operator Calls and all 1 + Area Code Calls, are not assigned. Digit Codes not assigned are automatically restricted.

Example 7 - Trunk-to-Trunk Connection

Allow Limited Area Codes

ATDP Command-

TDM/OG: 1 (T: Tandem Connection)

IC RT: Route 2 OG RT:1

DC: 92 TDI: 1 (Connection is Allowed)

Repeat these same assignments for Digit Codes “93” through “99” to allow all local office codes.

To allow Area Codes 718, 212, 201, and 914 repeat the ATDP command for the following Digit Codes:

DC: 91718 TDI: 1(Allow)

DC: 91212 TDI: 1(Allow)

DC: 91201 TDI: 1(Allow)

DC: 91914 TDI: 1(Allow)

This allows these specific Area Codes.

All the Area Codes which are not assigned are automatically restricted.

FCCS Programming

This data setting describes only the additional or replacement commands used to activate FCCS Services. When used as an FCCS feature, these commands replace the non-network commands (for example, AAED is replaced by AAEDN). Note that the data setting for FCCS Service varies depending on the Node.

Note: When using ASYDL or ASYDN to program an FCCS feature, the basic programming of ASYD is required in addition to ASYDL or ASYDN.

Note: For more detailed information, see the Data Programming Manual - FCCS.

STEP 1: **ARSCN** - Assign the route restriction class.

STEP 2: **ATDPL/ATDPN** - Assign the digit codes to be allowed or denied access, or toll-restricted.

T-8 Trunk Answer from Any Station (TAS)

General Description

This feature allows any Station (except one with incoming restrictions) to answer Incoming Calls when the system is in Night Mode. Incoming Exchange Network Calls will activate a common alert (TAS) signal at the customer's premises. By dialing a specified Code, any Station may answer the call and then extend it to any other Station using CALL TRANSFER-ALL CALLS [C-11].

Operating Procedure

To operate

1. TAS signal (bell, buzzer, etc.) sounds.
2. Lift the handset. Receive Dial Tone.
3. Dial the specified TAS Access Code.
4. Connection to Incoming Call is completed.

Service Conditions

1. An Incoming Call that terminates in the Night mode will have the following priority:
   a.) NIGHT CONNECTION-FLEXIBLE [N-2]
   b.) NIGHT CONNECTION-FIXED [N-1]
   c.) TRUNK ANSWER FROM ANY STATION (TAS) [T-8]
2. TAS indication will be activated for the following Incoming Calls:
   a.) Listed Directory Number
   b.) Foreign Exchange
   c.) WATS
   d.) CCSA
3. All Consoles assigned to handle calls for a particular group of Stations (whether or not they belong to the same TENANT [T-12]) must have NIGHT CONNECTION [N-1,2] set before this feature will be activated.
4. The TAS indicator can be provided on a per Tenant basis.
5. The TAS indicator must be locally-provided.
6. This feature is sometimes referred to as Universal Night Answer (UNA).
7. Individual Stations may be denied access to this feature via CLASS OF SERVICE-INDIVIDUAL [C-15].
8. If the system has changed to Day mode while a TAS Call is being indicated, the ATTENDANT CONSOLE [A-3] can be connected to this call.
9. Only one TAS per Tenant can be assigned as the TAS signaling port.

Interactions

1. For TAS to operate, an ATI circuit card is required, and the system must be in the Night mode, depending on the type of Attendant Console system that is in service. Exercise care when setting the resistance value via switch settings on the ATI circuit card.
2. TENANT SERVICE [T-12]: Only one TAS Circuit may be assigned per Tenant.
3. NIGHT CONNECTION-FIXED/FLEXIBLE [N-1, 2]: TAS can be used in conjunction with NIGHT CONNECTION-FIXED/FLEXIBLE [N-1, 2]. When the Night Connection Station(s) is busy, Incoming Calls are directed to TAS.

4. CALL TRANSFER-ALL CALLS [C-11]: A Station answering a call via the TAS feature will have access to the transfer feature.
5. CENTRALIZED ATTENDANT SERVICE (CAS) [C-20]: TAS may be installed at either the Main site only or at both the Main and Satellite sites. Operations at either site are the same as TAS installed in a non-CAS system. The Satellite system must have a means of placing the system into Night mode. This would require an ATI circuit card connected to a Night switch.
6. CALL FORWARDING - ALL CALLS [C-5] / BUSY LINE [C-2]: When a Direct Inward Dialing (DID) Call terminates to a Station setting CALL FORWARDING - ALL CALLS [C-5] or BUSY LINE [C-2] to the Attendant, the call can be answered using the TAS feature.
7. CLASS OF SERVICE-INDIVIDUAL [C-15]: Stations must be allowed access to TAS via Service Feature Restriction Class. The Station Route Restriction Class will either allow or deny Connection to Incoming Central Office Calls via TAS.
8. DAY/NIGHT CLASS OF SERVICE [D-15]: A CLASS OF SERVICE-INDIVIDUAL [C-15] change that occurs when the system goes into Night mode may allow a Station to answer Incoming Trunk Calls that are restricted to that Station in Day mode.
9. HOUSE PHONE [H-2], INDIVIDUAL ATTENDANT ACCESS [I-6], STATION-TO-STATION CALLING-OPERATOR ASSISTANCE [S-12], and ATTENDANT NIGHT TRANSFER [A-19]: These calls will not be directed to TAS when the system is in Night mode.
10. CONSULTATION HOLD [C-17]: TAS cannot be accessed via CONSULTATION HOLD.
11. TRUNK ANSWER ANY STATION will not default if other provisions are not made to answer Incoming Calls when the system is in Night mode. See the ACFR command.
12. The type of Attendant Console system used determines how the system will enter into the Night mode. See System Data 2, Index 2, Bits 2 and 3.
13. A Station user receives a Reorder Tone after dialing the TAS Access Code if one or more of the following applies:
    a.) Station Line is inward restricted.
    b.) Station Line is fully restricted.
    c.) Station is restricted from the TAS Access Code.
    d.) Station has a party on Hold.
    e.) Another Station has answered the TAS Call and no other calls are in the TAS queue.
14. Incoming Calls remain in the TAS queue until answered from a Station or until abandoned by the far end.
15. When this feature is in service and NIGHT CONNECTION - FIXED [N-1] is activated in the 4-IMG system, a Night Station needs to be set using an external key box connected to an NCU card.

Programming

STEP 1: **ASYD** - System Data 1, Index 68, Bit 2. Type of TAS ringing; 0/1: Interrupted/Continuous.

System Data 1, Index 77, Bit 0. Is Day/Night change accomplished by removing Attendant Console handset? 0/1: No/Yes.

System Data 1, Index 229, Bit 3. Day/night change system message printout. 0/1: No/Yes.

STEP 2: **ANPD** - Reserve a Number Level for Feature Access. Select Normal Service (N), Hooking Service (H) or Busy Service (B) from the combo box for Connection Indexes. (applicable to any feature assigned to this Level). Assign NND in accordance with a predetermined Numbering Plan.

STEP 3: **ASPA** - Assign an Access Code to TRUNK ANSWER FROM ANY STATION, SRV = "SSC: Service Code", SID = 16 TAS. Connection Index of CI = “N: Normal”.

STEP 4: **ASFC** - Assign a Service Feature Restriction Class that allows SFI 13 for the Station to receive TRUNK ANSWER FROM ANY STATION.

STEP 5: **ACFR** - Allow a Transfer Service Feature Index of 5, TAS to the listed types of Incoming Calls to be answered by TAS. The Call Class Indexes (CCI) will correspond to ARTD, CDN 6:TCL; Trunk Class parameter.

CCI: 1-LDN
2-DDD2
3-DDD3
4-TIE LINE
5-DDD4

STEP 6: **ATAS** - Assign the TAS Circuit to the TN, MG, PIR, and the Circuit Number: 0/1; based on the Circuit Location. TAS CKT # = Attendant Console CKT #.

T-9 Trunk Group Busy Lamp

General Description

This feature provides the ATTENDANT CONSOLE [A-3] with a visual indication when all Trunks in a particular Trunk Group are busy (LDN, WATS, FX, DOD, CCSA, TIE or Special Trunks). Assignment of the TGBLs is made using the PCPro.

Operating Procedure

No manual operation is required.

Service Conditions

1. The ATTENDANT CONSOLE [A-3] must have a designated TRUNK GROUP BUSY LAMP (TGBL) section.
2. This feature may be used on Trunk Groups consisting of WATS, TIE, FX, and special Trunks.
3. Ten TGBLs are available on each Attendant Console.
4. Trunk Group Warning to indicate when a preset number of Trunks in a group is busy is not provided.
5. Trunk Groups cannot be combined to light a common TGBL.
6. A maximum of 10 Trunks including LD and TIE are displayed on an Attendant Console.
7. Desk Console cannot be used to activate this feature.

Interactions

1. ATTENDANT CONSOLE [A-3] number is the equivalent to the Trunk Number when assigning the Attendant Console Route 901 in ATRK.
2. The ATRK command must be assigned before the ATGL command. If ATRK is not assigned first, ERROR: 0245, “ATTENDANT CONSOLE [A-3] not assigned yet”, will be indicated.

Programming

STEP 1: **ASYD** - System Data 1, Index 9. Assign the number of ATTENDANT CONSOLE [A-3] to be used in the system.

STEP 2: **ATRK** - Assign Route 901 to the Attendant Console.

STEP 3: **ARTD** - Assign Route Data for each Trunk Group that will be represented by a Trunk Group Busy Lamp.

Note: To change the data for existing route (RT), initialize or reboot the trunk device such as a circuit card or MG.

STEP 4: **ATRK** - Assign Trunks to the routes assigned in ATRK.

STEP 5: **MBTK** - Make idle the Trunks that are assigned, including the ATTENDANT CONSOLE [A-3].

STEP 6: **ASYD** - System Data 1, Index 15, Bits 0-3. In hexadecimal, assign how many TGBLs are in service, to a maximum of 10 lamps. Assign 0AH.

STEP 7: **ATGL** - Assign the ATTENDANT CONSOLE [A-3] number, the TGBL number (1-10), and the Route Number(s) to be associated with the TGBL.

T-10 Trunk-to-Trunk Connection

General Description

This feature provides any Station user with the ability to Conference together either two Central Office Calls or a Central Office and TIE Line Call.

Operating Procedure

To operate

1. Place Incoming Caller on hold.
2. Receive Special Dial Tone.
3. Dial the desired Central Office or TIE Access Code.
4. Receive Second Dial Tone.
5. Dial the desired number. Receive answer.
6. Announce the Conference and momentarily press the switch hook again.
7. The original party is now included in Connection.

Service Conditions

1. The originating party may hang up at any time. The additional two parties will not be disconnected.
2. This feature may be restricted to individual Stations via System Data.
3. In the case of Outgoing Calls on Trunks with no Answer Supervision, this feature may be activated only after 18 seconds have elapsed since the last digit was dialed. This timer is programmable in System Data.
4. In the case of Outgoing Calls on Trunks with Answer Supervision, this feature may be activated as soon as the Called Party answers.
5. A Station user may hang up after completing a TRUNK-TO-TRUNK CONNECTION without affecting the Connection of the two Outside Parties, provided that:
   a.) One Trunk is incoming and the other is outgoing.
   b.) Both Trunks are Ground Start.
   c.) Station users wait a programmed time interval after dialing an Outside Line before hanging up.
6. If the originating party encounters a no-answer or busy signal when trying to Conference a Third Party, the originating party switch hook flashes and is reconnected to the Held Call.
7. This feature is inoperative when Release Signal is not provided from the distant end.

Interactions

1. For direct Trunk-to-Trunk Connection, see TANDEM SWITCHING OF TIE TRUNKS-2/4-WIRE [T-1].
2. If a THREE-WAY CALLING [T-2] Connection includes two Trunks, then when the Station hangs up, the Trunk will release, unless the Trunks are allowed TRUNK-TO-TRUNK CONNECTION via the ARRC command, ARI-D-RES.

Programming

STEP 1: **ASYD** - System Data 1, Index 156. The timer value for non-Senderized Trunks between seizure and when a switch hook flash is possible. Default is 30 seconds.Without SMDR, it is recommended to change it to two seconds. Program 31H.

System Data 1, Index 157. The timer value for Senderized Trunks between seizure and when a switch

hook flash is possible. Default is 18 seconds. Without SMDR, it is recommended to change it to two seconds. Program 31H. (See STATION MESSAGE DETAIL RECORDING [S-10]).

System Data 2, Index 1, Bits 0 and 1. CONSULTATION HOLD. Assign data "11" for CONSULTATION HOLD service on Originating, Terminating, and Tandem Calls.

System Data 2, Index 6, Bit 2. This Bit remains as data "0". CALL TRANSFER-ALL CALLS [C-11].

System Data 2, Index 6, Bit 3. Enable add-on Station. This Bit remains as data "0" for THREE-WAY CALLING [T-2].

System Data 3, Index 2. Assign data 91H for normal switch hook flash values.

STEP 2: **ARSC** - Stations allowed to establish TRUNK-TO-TRUNK CONNECTION must be provided access to routes that have been allowed TRUNK-TO-TRUNK CONNECTION in the ARRC command.

STEP 3: **ASFC** - Assign a Service Feature Restriction Class that allows SFI 10 to Stations that will have CALL HOLD [C-6].

STEP 4: **ATRK** - Assign these LENS as Route 909, Digital Conference Trunk. LENS consist of the following:

XX	X	XX	X
Module Group	Unit	Group	Level

Group and Level numbers for the Digital Conference Trunk are 210, 213, 216, 221, 224, 227, 232, and 235.

STEP 5: **ARRC** - Routes that will be allowed TRUNK-TO-TRUNK CONNECTION must be allowed here for ARI-A-RES.

ICRT: Incoming Route Number

OGRT: Outgoing Route Number

ARI A-RES: This allows/restricts Trunk-to-Trunk (Tandem) Connection via the Attendant Console and Stations.

ARI D-RES: This allows/restricts Trunk-to-Trunk (Tandem) Connections via Direct Dial access from a Station and Trunks.

0: Alternate routing is restricted

1: Alternate routing is allowed

2: Toll Restricted

When ARI-D-RES is needed for a direct dial TRUNK-TO-TRUNK CONNECTION, refer to the specific application within this document for the programming steps. Such applications include REMOTE ACCESS TO SYSTEM [P-75], TANDEM SWITCHING OF TIE TRUNKS-2/4-WIRE [T-1], INCOMING CENTRAL OFFICE TO TIE LINE CONNECTION [I-3], and TOLL RESTRICTION-3/6-DIGIT [T-7].

FCCS Programming

This data setting describes only the additional or replacement commands used to activate FCCS Services. When used as an FCCS feature, these commands replace the non-network commands (for example, AAED is replaced by AAEDN). Note that the data setting for FCCS Service varies depending on the Node.

Note: When using ASYDL or ASYDN to program an FCCS feature, the basic programming of ASYD is required in addition to ASYDL or ASYDN.

STEP 1: **ARSCN** - Stations allowed to establish TRUNK-TO-TRUNK CONNECTION must be provided access to routes that have been allowed TRUNK-TO-TRUNK CONNECTION in the ARRC command.

STEP 2: **ARRCN** - Routes that will be allowed TRUNK-TO-TRUNK CONNECTION must be allowed here for ARI-A-RES.

ICRT: Incoming Route Number

OGRT: Outgoing Route Number

ARI A-RES: This allows/restricts Trunk-to-Trunk Tandem) Connection via the Attendant Console and Stations.

ARI D-RES: This allows/restricts Trunk-to-Trunk (Tandem) Connections via Direct Dial access from a Station and Trunks.

0: Alternate routing is restricted

1: Alternate routing is allowed

2: Toll Restricted

When ARI-D-RES is needed for a direct dial TRUNK-TO-TRUNK CONNECTION, refer to the specific application within this document for the programming steps. Such applications include REMOTE ACCESS TO SYSTEM [P-75], TANDEM SWITCHING OF TIE TRUNKS-2/4-WIRE [T-1], INCOMING CENTRAL OFFICE TO TIE LINE CONNECTION [I-3], and TOLL RESTRICTION-3/6-DIGIT [T-7].

T-12 Tenant Service

General Description

This feature provides for more than one organization (Tenant) to share the same System. Through system programming, each organization may be restricted to its own Central Office Trunks, ATTENDANT CONSOLEs [A-3], and extension links. In addition, Incoming Calls are directed to the specific organization (Tenant).

Operating Procedure

No manual operation is required.

Service Conditions

1. An Attendant Console is normally provided for each Tenant. However, when required, a single common Console may be shared by two or more Tenants.
2. Inter-Office Calling between Tenants may or may not be restricted, depending on System Data.
3. Different Tenants may share a common group of Trunks, where required.
4. When a Tenant Station dials “0” (Operator Call Code), it will be connected to that Tenant’s Console.
5. Different Tenants cannot use the same Numbering Plan.
6. The same Telephone Numbers cannot be assigned among different Tenants (for example, Stations numbered 2XXX can be associated with only one Tenant).
7. STATION-TO-STATION [S-11], CALL TRANSFER [C-11] THREE-WAY CALLING [T-2], TAS [T-8] and CALL FORWARDING [C-2, 3, 5] Calls between Tenants can be restricted or allowed.
8. SMDR [S-10] includes Tenant identification.
9. The same Access Code can be assigned among Tenants for service features (for example, CALL HOLD [C-6], CALL BACK [C-1], CALL PICKUP - GROUP [C-7], CALL FORWARDING - ALL CALLS [C-5], SPEED CALLING [S-3], TAS [T-8], etc.).
10. The Telephony Server can provide TENANT SERVICE [T-12] to a maximum of 63 Tenants.
11. Tenant Numbers should be allocated on a network basis for the FCCS Network.

Interactions

1. To provide TENANT SERVICE, features need not be separated. Whenever possible, keep features common to use data memory as economically as possible. When commands are separated, they must be programmed for each Tenant accommodated in System Data 2.
2. FORCED ACCOUNT CODES [F-7] and AUTHORIZATION CODES [A-20] do not require separate Tenant service for AMND, System Data 1, Index 93, Bit 6.
3. Each Tenant may have its own Attendant Console(s), unless that number exceeds system capacity; for example, five Tenants in a two-PIR configuration. The number of Consoles per Tenant is not limited by programming but is limited by the system configuration.
4. Different Tenants may or may not share the same Central Office Trunks. Termination of a Trunk is a function of the ARSC command. DID Trunks can be assigned to any Tenant. However, the Listed Directory Number, as assigned in the ALDN command, determines where the call terminates. If the Incoming LDN digits are assigned to Tenant 2, the call terminates to the Tenant 2 Attendant Console, even if the digits come on a Tenant 1 Trunk.

5. CALL FORWARDING [C-2, 3, 5]: When a Station Call forwards to the Attendant Console, the call will go to the originating Station's Attendant Console, as assigned by the Tenant parameter of the ATRK command.
6. CLASS OF SERVICE-INDIVIDUAL [C-15]: Refer to the programming section. This feature affects the construction of all classes of service.

Programming

STEP 1: **ASYD** - System Data 1, Index 8. Assign the number of Tenants that the system will serve, in hexadecimal format.

System Data 1, Indexes 92, 93, and 94. The Bits of these Indexes allow for either common or separate Tenant assignments for other specific commands. The following Bits represent the following commands. Assign as follows; 0/1: Separate/Common.

Index 92
Bit 0, All of ASYD, System Data 2
Bit 1, ASPA, AASP, AGSP
Bit 2, ANPD, AANP, AGNP
Bit 3, ALDN, ASDT and ASAT (Must be assigned as data "1".)
Bit 4, ARSC
Bit 5, ACFR
Index 93
Bit 0, ASFC
Bit 1, ACFO
Bit 2, ATAS
Bit 3, ASPD
Bit 4, ASTP and AFRS (O.G.Connection)
Bit 5, ASTP and AFRS (I.C.Connection)
Bit 6, AMND
Bit 7, AAED
Index 94
Bit 0, ATNR
Bit 1, AEFR (EPN only)
Bit 2, APCR
Bit 3, AATC, ASPS
Bit 4, ASPF
Bit 5, ADNR
Bit 6, ASCR
Bit 7, ASID, ATCR
Bit 2, APCR. (EPN Only)

STEP 2: **ATNR** - Assign the appropriate Tenant restrictions.

D/N: Day/Night Mode
- Day Mode
- Night Mode

OGTN: Originating Tenant
TMTN: Terminating Tenant

TRI: Tenant Restriction Index (0-15) **Note 1**
RES: Restriction Data (0/1)
0: Restricted
1: Allowed

Note: Data can be entered on a Day/Night basis if specified in ASYD, System Data 1, Index 75, Bit 2 = 1.

Note 1: Assigning data "1" (allow) to a TRI Index is not all that is required for a call to be completed between Tenants. Each Tenant must also have a Numbering Plan defined that allows for the Station Numbers in the other "allowed" Tenants to be dialed.

Assignment of Tenant Restriction Class
TRI: Tenant Restriction Index
0: Allows intra and inter-Tenant Station Calls. Allow calls within Self-Tenant.
1: Allows CALL FORWARDING-ALL CALLS, DON'T ANSWER, BUSY LINE on an intra and inter-Tenant basis. Allow this feature within Self-Tenant.
2: Allows or restricts Incoming Calls, via the Night Attendant Console, from Tenant to Tenant. This allows the Tenant, as assigned to a Trunk, to be terminated to a Tenant, as assigned to a Station.
3: Allows the CALL FORWARDING-ALL CALLS, DON'T ANSWER, and BUSY LINE on an intra and inter-Tenant basis as when set by the Attendant Console.
4: Allows or restricts Incoming Calls to the Attendant Console on a per Tenant basis. Thus, an Attendant Console may handle calls only for a specific Tenant.
5: Day/Night Mode Change Restriction is not used in North America.
6: Not Used.
7: Incoming Call of the Trunk within OGTN terminates to a Station within TMTN (in the case of MFC signaling only).
8-15: Not Used.

STEP 3: **ATRK** - Assign the Attendant Consoles to receive Incoming Trunks according to Tenant. The TN parameter will determine which specific Tenants this Attendant Console will handle. Example: In a three-Tenant system, the Attendant Console will handle Tenants 1 and 3. Assign the following:

RT: 901 TK: 1
LENS: 000200
CNT: 2 1 TN: 1 RSC: 0
CNT: 2 2 TN: 3 SFC: 0

STEP 4: **MBTK** - Make idle the Attendant Console Trunks.

FCCS Programming

This data setting describes only the additional or replacement commands used to activate FCCS Services. When used as an FCCS feature, these commands replace the non-network commands (for example, AAED is replaced by AAEDN). Note that the data setting for FCCS Service varies depending on the Node.

ASYDL/ASYDN - The Bits of these Indexes allow for either common or Separate Tenant assignments for other specific commands. 0/1 = Separate/Common

System Data 1, Index 800

Bit 0: ALGNL/ALGNN, ALGSL/ALGSN
Bit 1: ASPAL/ASPAN **Note 2**
Bit 2: ANPDL/ANPDN
Bit 3: APCNL/APCNN
Bit 4: AFRSL/AFRSN, AUNEL/AUNEN
Bit 5: ASTPL/ASTPN
Bit 6: ARSCN **Note 3**
Bit 7: AAEDL/AAEDN

Note 2: This data cannot be changed when any data is assigned in ASPAL or ASPAN command.

Note 3: Valid with ASYDN command only.

Index 801

Bit 1: AMNDL

Index 802

Bit 0: AHSPL/AHSPN
Bit 1: ATDPEL/ATDPWN
Bit 2: AINDL

T-18 Time Display

General Description

This feature provides a digital time display on the Attendant Console.

Operating Procedure

The time is constantly displayed in the time display area of the Attendant Console.

Service Conditions

1. The time is displayed according to the built-in clock of the Attendant Console.
2. The built-in clock of the Attendant Console is independent of the clock in the Telephony Server.

Programming

Above the time display is an area marked ADJUST, with two openings: One labeled “H” (hours) and the other labeled “M” (minutes). Using a pointed object that will fit into the opening, gently press the appropriate ADJUST button (H or M) until the correct Hour or Minute is displayed.

Interactions

Upon initial installation, it may take up to 30 minutes for clock back-up power reserve to charge. Once the charging process is complete, the clock adjustments can be made.

T-18D Time Display -D

General Description

This feature provides a digital time display on a Desktop terminal's LCD. Also, the time display pattern can be changed by system data assignment. The following table shows an example of time display pattern.

Time Display Pattern	
12 hour	**24 hour**
10:30 PM WED 6 NOV 2013	22:30 WED 6 NOV

Operating Procedure

No manual operation is required.

Service Conditions

1. The time display pattern can be changed for DT800/DT700/DT400/DT300 Series only. For other terminals, the time display pattern cannot be changed (fixed to default display pattern).
2. When the DT400/DT300 terminals are accommodated to IPG (Digital) or UG50, time display pattern cannot be changed (fixed to default display pattern).
3. The following ELC card with firmware is required for changing the time display pattern of DT400/DT300 Series.

ELC Card	Firmware Requirements
CH-16ELCA	SP-4035 16ELCA PROG-A Version 1 or later

4. In FCCS network, when you change the time indication format (AM12:00 or AM0:00) for LCD, log in to the system and assign the system data (ASYDL, System Data 1, Index 1256, Bit 0) to each node. For details, refer to Programming.

   Note: ASYDN command is not supported for the setting of time indication format.

5. The time is displayed according to the clock in the system and is set via the PCPro.
6. The time will not be displayed when a Desktop terminal is set for a second line MESSAGE WAITING INDICATION -D [M-85D] and MW/VMM is displayed.
7. While NAME DISPLAY - SYSTEM [N-28], NAME DISPLAY - SYSTEM - CCIS [N-37], or NAME DISPLAY WITHOUT - OAI - CCIS [N-37] 16-digit display is activated on the Desktop terminal with Two-line LCD display, the time is not displayed. When both the Calling/Called Station are released from a Station-to-Station Call, the LCD of the Calling/Called Station returns to the TIME DISPLAY.

Interactions

1. Only the Desktop terminal with display has the TIME DISPLAY feature capability.
2. The calendar and time must be adjust via ATIM command.

Programming

STEP 1: **ASYDL/ASYDN** - System Data 1, Index 1249, Bit 0-3. When you assign the other parameters than the parameters shown below, the Display Pattern of the terminal will be 0.

Note: The data assignment of Desktop terminal is also required for this feature. For details, refer to 5. Desktop Terminal Data in Chapter 2 of this manual.

Pattern Number	Bit				Time Display Format	Example Display
	b3	b2	b1	b0		
0 (Default)	0	0	0	0	12 hour	10:30 AM WED 6 NOV 2013
1	0	0	0	1		6 NOV WED 10:30 AM
2	0	0	1	0		10:30 AM NOV 6 WED
3	0	0	1	1		11-6 WED 10:30 AM
4	0	1	0	0		10:30 AM WED 6 NOV
5	0	1	0	1	24 hour	6 NOV WED 10:30
6	0	1	1	0		10:30 NOV 6 WED
7	0	1	1	1		11-6 WED 10:30
8	1	0	0	0		10:30 WED 6 NOV

Note: When you assign the parameters except the above table, the pattern number is specified as “0 (Default)”.

Note: Available for DT800/DT700/DT400/DT300 Series only.

Note: After this system data assignment, the following operations are required.
- Digital terminal: Initializing a corresponding ELC card.
- SIP terminal: Restarting the terminal after 30 second is elapsed.

STEP 2: **ASYDL** - System Data 1, Index 1256, Bit 0.
Assign this command to specify the Time Display Format.
0/1 = AM12:00 (midnight)/AM0:00 (midnight).

Bit	12 hour	24 hour
0	AM 12:00	24:00
1	AM 0:00	0:00

Note: Available for DT800/DT700/DT400/DT300 Series only.

Note: After this system data assignment, the following operations are required.
- Digital terminal: Initializing corresponding ELC card.
- SIP terminal: Restarting the terminal after 30 second is elapsed.

Note: In FCCS network, be sure to assign this system data to each node.

T-23 Trunk Line Appearance

General Description

This feature allows a Desktop terminal user to directly access a Central Office Trunk or TIE Line without dialing an Access Code. Incoming Calls can also be answered at the TRUNK LINE APPEARANCE [T-23].

Operating Procedure

To access an Outgoing Trunk

1. Press the TRUNK LINE APPEARANCE Feature key.
2. Lift the handset or press the SPEAKER key. Central Office or distant System Dial Tone will be heard.
3. Dial the destination number.

To answer an Incoming Call from the TRUNK LINE APPEARANCE

1. Ringing and an LED indication alert the Desktop terminal user of the Incoming Call.
2. Press the TRUNK LINE APPEARANCE Feature key.
3. Lift the handset or press the SPEAKER key.
4. Answer the Incoming Call.

Service Conditions

1. A TRUNK LINE APPEARANCE can be accommodated on Desktop terminals, and 24/40-button Add-On Modules.
2. A TRUNK LINE APPEARANCE can be assigned as a SOFTWARE LINE APPEARANCE [S-33].
3. A TRUNK LINE APPEARANCE can be assigned in a Circular or Pilot Hunt Group.
4. A TRUNK LINE APPEARANCE is available for destination offices that can receive DP/PB signals.
5. CALL FORWARDING-ALL CALLS [C-5], CALL FORWARDING-BUSY LINE [C-2] and CALL FORWARDING-DON'T ANSWER [C-3] can be set at the Attendant Console for a TRUNK LINE APPEARANCE assigned to a Virtual Line Appearance or an unused data port.
6. A TRUNK LINE APPEARANCE can receive Incoming Calls from internal Stations, DID, DIT, or TIE Line Trunks.
7. When a call is originated with the RECALL key and the Trunk is busy, the call is in the queue waiting to seize the idle Trunk. If no Dial Tone has been found for two seconds, the Calling Party hears a Busy Tone (if the idle Trunk is found within two seconds, the Station is connected to the idle Trunk).
8. When TRUNK LINE APPEARANCE is used for the DP line, program two seconds or less for outgoing inhibit timing after Trunk release (ARTD CND 34: GUARD = 1-3).
9. This feature is not supported with MC&MG-COT/MG-COT/UG50.

Interactions

1. For Incoming Calls, the Trunk must be assigned as a DIRECT-IN TERMINATION [D-7] Trunk.
2. The Trunk may still be seized by other Stations or the ATTENDANT CONSOLE if the route is not isolated from other features such as LEAST-COST ROUTING-3/6 DIGIT [L-6] or ROUTE ADVANCE [R-6].

3. The Trunk may be selected by INDIVIDUAL Trunk Access [I-4]. If accessed, the Desktop terminal will not have a lit LED for busy condition, as would occur when a Trunk is seized in a 1A2 Key system. If the user of TRUNK LINE APPEARANCE goes off hook to seize the Trunk, Busy Tone is returned. The LCD displays:

BUSY
(Time Display)

4. TRUNK LINE APPEARANCE and HOTLINE-OUTSIDE [H-9] are mutually exclusive. Both features require the AHLS command, and the same Station cannot be provided with two Hotline features.

Programming

STEP 1: **ASDT** - Assign the Station with the TRUNK LINE APPEARANCE as a Virtual Line (TEC = 18).

TN: Tenant Number
STN: Station Number
LENS: Line Equipment Numbers (6 digits)
TEC: Telephone Class (1-31)
- 1: DP (10 PPS)
- 2: PB
- 3: DP/PB
- 12: Dterm
- 13: Data Terminal Via Dterm
- 14: Hotline
- 15: CAS Line
- 16: Data Terminal Via Data Module
- 18: Virtual Circuit
- 23: ISDN Terminal
- 27: Eight Conference Equipment

RSC: Route Restriction Class (0-15)
For assignment of RSC, ARSC command.
SFC: Service Feature Restriction Class (0-15)
For assignment of SFC, ASFC command.

STEP 2: **ARTD** - Assign the route of the TRUNK LINE APPEARANCE as a ground-start, ring-down Trunk.

RT: 1

1-OSGS	:2	2-ONSG	:3	3-ISGS	:1	4-INSG	:3	5-TF	:1/3
6-TCL	:1	7-L/T	:1	8-RLP	:2	15-LSG	:1	44-PRV	:1

Note: To change the data for existing route (RT), initialize or reboot the trunk device such as a circuit card or MG.

STEP 3: **ATRK** - Assign the LENS, Route Number, Trunk Number, and Tenant Number. Assign one or more Trunks to a specific route used to be connected as a TRUNK LINE APPEARANCE.

STEP 4: **MBTK** - Assign the make idle status to the Trunks.

STEP 5: **ASFC** - Assign a Service Feature Restriction Class that allows Brokerage Hotline (SFI 36) to the Stations to receive TRUNK LINE APPEARANCE.

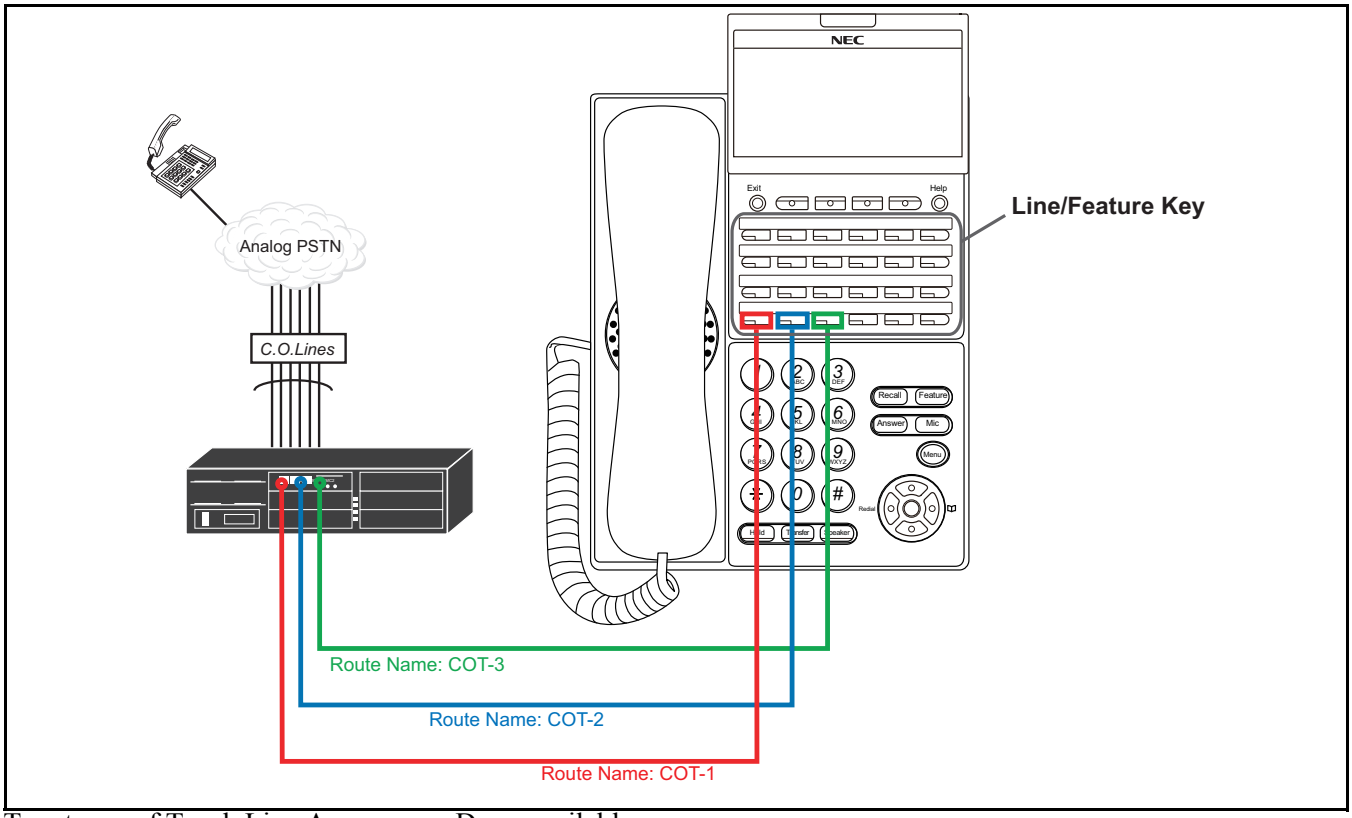
STEP 6: **AHLS** - Assign the following parameters for the Station with the TRUNK LINE APPEARANCE:

HOT TN: Hotline Tenant Number.
HOT STN: Hotline Station Number with the TRUNK LINE APPEARANCE.
TYPE: For TRUNK LINE APPEARANCE, assign Type 3.
CON RT: Assign the TRUNK LINE APPEARANCE Route.
CON TK: Assign the TRUNK LINE APPEARANCE Trunk.

T-23D Trunk Line Appearance -D

General Description

This feature allows a IP Station user to answer incoming calls from a C.O. line by pressing the corresponding Line key.



Two types of Trunk Line Appearance-D are available:

<1-C.O. line to 1-Line key on IP Station>
Connecting a C.O. line to a Line key of IP Station one-to-one (shown in the example figure above).

<1-C.O. line to Line key on Multiple IP Stations>
Sharing a C.O. line allocated to multiple IP Stations. User can originate/answer a call with SPEAKER/ANSWER key control on IP Station.

Operating Procedure

To answer an Incoming Call on IP Station

1. Message Waiting Lamp indication informs the IP Station user of the Incoming Call.
2. Lift the handset or press the SPEAKER key.
3. Press the TRUNK LINE APPEARANCE-D Feature key.
4. Answer the Incoming Call from C.O. line.

Service Conditions

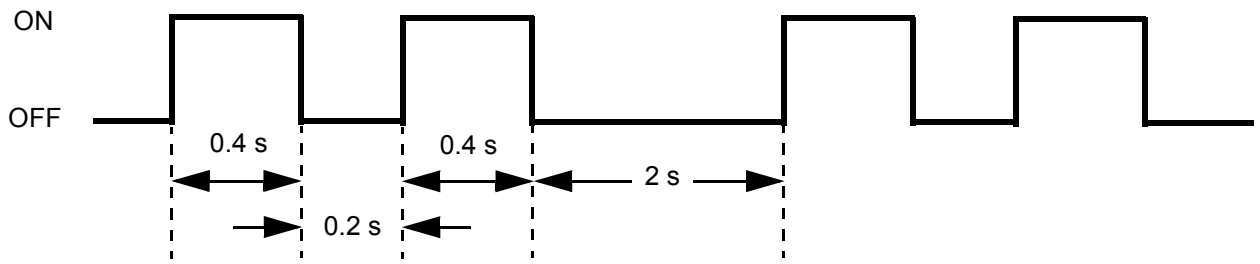
1. MC&MG-COT/MG-COT/UG50 is required for this feature. Available firmware version is:
   - MG-COT [SCA-6COTA]: SP-3875 6MGCOT PROG-A
   - MG-COT [SCA-6COTB] (Proprietary Protocol): SP-3948 6MGCOT PROG-A **Note 1**
   - MG-COT [SCA-6COTC] (Proprietary Protocol): SP-3948 6MGCOT PROG-A **Note 1**
   - MG-COT [SCA-6COTB] (SIP): SP-3948 6MGCOT PROG-A **Note 1**
   - MG-COT [SCA-6COTC] (SIP): SP-3948 6MGCOT PROG-A **Note 1**
   - MCMG [SCA-4LC2COTA/MG-4LC2COTA]: SP-3778 MCMG PROG-A
     SP-3927 MCMG PROG-B
   - UG50 (COT-TYPE1): No condition
   - UG50 (COT-TYPE2): No condition
   - UG50-4LC2COTA (COT): No condition
   - UG50-6COT: No condition

   Refer to Peripheral Equipment Description (IP Devices/UG50) for the data assignment of each MG.

   Note 1: Firmware SP-3948 6MGCOT PROG-A Issue 1.0.0.0 version or later is required.

2. When using MC&MG-COT for this feature, assign the highest priority to a route through MC&MG-COT for LCR.
3. System data assignment for Trunk Answer from Any Station (TAS) is required for this feature, but the TAS service is not activated.
4. Whether the Trunk Answer from Any Station (TAS) feature is available or not is determined by the Service Feature Restriction Class (SFC: 34) setting to each station.
5. When Trunk Line Appearance-D station does not answer the incoming call and Call Forwarding-Don't Answer is activated, the LCD on the IP Station does not display to inform the execution of call forwarding.
6. Analog C.O. line corresponding to PB signal is only applicable.
7. IP Stations must assign a My Line to a Line/Feature key beforehand.
8. All the virtual lines of C.O. trunk, except My Line, can be assigned as Trunk Line Appearance-D lines on one IP station. (e.g. On 16-button IP station, a maximum of 15 lines can be accommodated as Trunk Line Appearance-D. The remaining one line is the My Line.)
9. Up to 128 of C.O. lines can be accommodated as Trunk Line Appearance-D per system.
10. Distinctive Ringing is not activated when incoming call terminates. A Ringer Pattern is only one below.

Ringing



11. Analog Telephone cannot answer the incoming call of Trunk Line Appearance-D.
12. When IP Station answers and holds the incoming call of Trunk Line Appearance-D, Analog Telephone station cannot answer the held call. IP stations belong to multiple-line group can answer the held call.
13. Call termination is informed to the user only by the Message Waiting Lamp indication on IP Station.
14. This feature is not applied to the outside call via ISDN line.
15. To send out Hooking signal to the destination party, assign the Route Class data of COT route to HKS=1.
16. When originating a call, IP station hears dummy tone from its' own system. Assign the dummy tone by the 2nd DT parameter of ASPA command.

    ASPA 2nd DT: 0: -
    Second DT is not sent.
    1: 2nd DT
    Second DT is sent.
17. Numbering plan data for the call origination via COT must be assigned by the LCR setting of ASPA command.
18. Trunk Line Appearance-D station and the corresponding trunk are basically accommodated in the same module group. When Trunk Line Appearance-D station and the trunk are accommodated in the different module groups (**Note 2**), set ASYD, SYS 1, Index 321, Bit 0 = 1 (available for Desktop terminal only).

Note 2: In that case, the module groups must be in the same LP.

Interactions

When conventional COT circuit card is used, Trunk Line Appearance [T-23] is applied.

Programming

Make sure the data setting for MC&MG-COT/MG-COT/UG50 is completed before starting the following data assignment. When MC&MG-COT is used, perform steps outlined in <MC&MG-COT Configuration for LCR>. Otherwise, proceed with Step 8 in <Settings Common to MC&MG-COT, MG-COT and UG50>.

<MC&MG-COT Configuration for LCR>
For details about the parameters of each command, see the Programming section of LEAST COST ROUTING - 3/6 DIGIT [L-5].

STEP 1: **ANPD/ANPDL/ANPDN** - Assign an access code for LCR service, defining the first digit and the number of digits.

STEP 2: **ASPA/ASPAL/ASPAN** - Enter a dummy route number in the External Route Number (RT) field of the External Route Number Data for the LCR access code.

STEP 3: **AFRS/AFRSL/AFRSN** - Define a Number Pattern Code (NPC **Note 3**) and an Outgoing Route Selection Number (OPR) for the dummy route you specified in the previous step. OPR is used to map a dialed NPC to the route pattern for LCR.

Note 3: Make sure that NPC includes the LCR access code.

STEP 4: **AOPR/AOPRL/AOPRN** - For each OPR you defined in the previous step, specify availability hours by using the Time of Day Change Pattern Number (TDPTN) parameter and priority by using the Route Advancing Order (RA) parameter.

Note: Assign the highest priority to a route through MC&MG-COT for LCR.

STEP 5: **AADC/AADCL/AADCN** - If additional digit information is required, assign a DC additional number (not including the access code for LCR) to each Pattern Number Location (PNL).

STEP 6: **AMND/AMNDL** - Specify the maximum number of digits to be received by the Originating Register Trunk (ORT) circuit for each Destination Code (DC).

STEP 7: **ARNP/ARNPL** - Assign an access code for each individual route.

<Settings Common to MC&MG-COT, MG-COT and UG50>

STEP 8: **ASYD** - Assign the system data related to Trunk Answer from Any Station (TAS).
SYS1, Index 72, Bit 2. Tone pattern to be sent out to stations on Exclusive Hold/Non-Exclusive Hold.
0/1=Dial Tone (DT)/Service Set Tone (SST)
SYS2, Index 11, Bit 6. 0/1=Trunk Answer from Any Station (TAS) is Out of service/In Service.
STS2, Index 12, Bit 0. 0/1=Automatic Recall Ringer is required for Non-Exclusive Hold/Not required.
STS2, Index 12, Bit 1. 0/1=Automatic Recall Ringer is required for Exclusive Hold/Not required.

STEP 9: **ASFC** - Assign a Service Feature Restriction Class to allow the following feature to the stations of Trunk Line Appearance-D.
SFI 34=1 (Trunk Answer from Any Station (TAS))
SFI 35=1 (Exclusive Hold, Non-Exclusive Hold)

STEP 10: **ACFR** - Assign Call Forwarding Restriction data for Trunk Line Appearance-D.
TSFI=2 (Direct in Termination (Night only))
CCI=1 (C.O. Line)
RES=1 (Transfer service is allowed)

STEP 11: **ARTD** - Assign the route class data of the Trunk Line Appearance-D trunk.
RT: C.O. trunk of MG-COT/UG50 Card/MCMG Card
37-CLD=1 (Trunk Answer from Any Station (TAS) is in service)
67-HKS=1 (Hooking Service is in service) **Required to send out Hooking signal to outside party.*
114-DCH:=1 (Channel Integration is ON)
125-MG COT=1 (Available)

Note: To change the data for existing route (RT), reboot the MG.

STEP 12: **ARTI** - Assign Trunk Application Data.
68-VIR=2 (MG Connection)

Note: To change the data for existing route (RT), reboot the MG.

STEP 13: **AMGIL** - Assignment of Media Gateway Information.
Refer to the Peripheral Equipment Description for details.

STEP 14: **ATRK** - Assign the LENS, Route Number, and Trunk Number. Assign one or more Trunks to a specific route used to be connected as Trunk Line Appearance-D.

STEP 15: **MBTK** - Release the make busy state of Trunk Line Appearance-D trunks.

STEP 16: **ACOC** - Assign the C.O. line code (COC) to the External Route Number and Trunk Number of each Trunk Line Appearance-D trunk. This code is dialed to answer the held C.O. line.

STEP 17: **ARSC** - Assign the Route Restriction Class (RSC) to allow the connection between the C.O. line route and the Route Class of Trunk Line Appearance-D stations.

STEP 18: **ASDT** - Assign the Station Number of Trunk Line Appearance-D as a Virtual Line (TEC=18).
STN: Station Number of Virtual station

LENS: Line Equipment Numbers (six digits). Virtual Line can be accommodated to Group 24-31.
TEC: Telephone Class. TEC=18 (Virtual Circuit)
Assign the My Line of DtermIP station.
TEC: Telephone Class. TEC=12 (Dterm)
*RSC and SFC data for Virtual Line can be skipped.

STEP 19: **AKYD** - Assign the Virtual Line and My Line on the Line/Feature keys of DtermIP.
KYI=2 (Multiple Line) for both Virtual Line and My Line.
RG (Ringing information)=7 for Virtual Line.

Note: Be sure to assign the My Line as “Prime Line.”

STEP 20: **ACSI** - Assign the Virtual Line (Virtual Station Number) to be connected to when incoming call terminates from the C.O. line.
CSI=2 (Virtual Line Termination)

STEP 21: **AMKD** - Assign the C.O. trunk route and the Trunk Number to the Virtual Line (Virtual Station Number) of Trunk Line Appearance-D.
STN: Station Number of Virtual station
KYN: Key Number of Trunk Line Appearance-D. Fixed to KYN=1.
MTN: Trunk Line Appearance Number. Fixed to MTN=0.
KYI: Select 1=C.O. trunk.

Note: To originate a call from a station by this feature using MCMG/MG-COT/UG50, LCR/LCRS data (ANPD, ASPA, AFRS, AOPR, AMND, ARNP) needs to be set to MCMG/MG-COT/UG50.

T-28 Trunk-to-Trunk Third-Party Cancellation

General Description

This feature allows a Station to cancel an active Trunk-to-Trunk Connection (during a Three-way-Connection) and return to the Original Call.

Operating Procedure

If a Station is presently engaged in a call with another Station or with a Trunk Party

1. Switch hook flash. Receive Special Dial Tone.
2. Dial a Trunk Access Code and the number for the desired Third Party. The Called Party is rung.
3. Switch hook flash to establish a Three-way Connection.
4. Switch hook flash to release the Trunk. The Station returns to the original call.

Service Conditions

1. The Third Party that can be canceled by this feature is an Outgoing Trunk seized by an outgoing access from CONSULTATION HOLD-ALL CALLS [C-17].
2. If a Broker's Call has been activated by a Dterm, the following restrictions apply:
   a.) In the case of CONSULTATION HOLD-ALL CALLS [C-17] outgoing (Station hold), the Third Party can be canceled as follows: Engaged in Station-to-Station Call ⟶ Three-Way Connection ⟶ Switch hook.
   b.) The Third Party cannot be canceled by the following sequence: Engaged in Station-to-Trunk Call ⟶ Three-Way Connection ⟶ Switch hook.
   c.) This restriction does not apply to CONSULTATION HOLD-ALL CALLS [C-17] outgoing (Trunk hold).

Programming

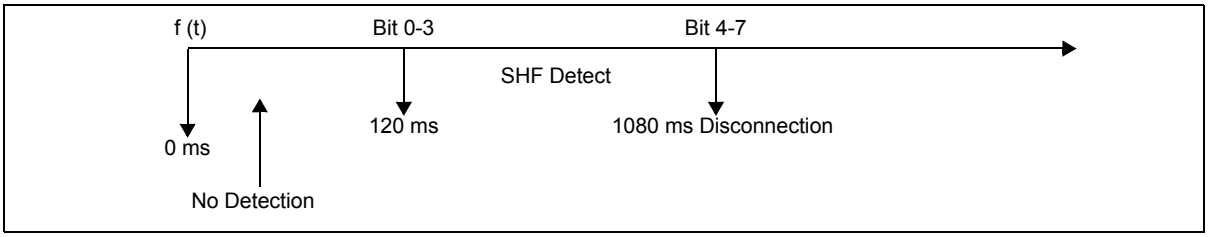
ASYD - System Data 1, Index 156. Trunk Soft Hold Timer A, used with Second Dial Tone Trunks. Assign data 00H for 30 seconds. (Data 31H may be used for a value of two seconds if CALL TRANSFER-ALL CALLS [C-11] is to be used immediately after connecting to the Trunk.) This is the time between Trunk Connection and when switch hook flash is allowed on a non-Senderized Trunk Connection.

System Data 1, Index 157. Trunk Soft Hold Timer A, used with Senderized Trunks. Assign data 00H for a value of 18 seconds. (Data 31H may be used for a value of two seconds, if CALL TRANSFER-ALL CALLS [C-11] is to be used immediately after connecting to the Trunk.). This is the time between Trunk Connection and when switch hook flash is allowed on a Senderized Trunk Connection. It is also used to determine when SMDR billing will begin.

System Data 2, Index 1, Bits 0 and 1. CONSULTATION HOLD allowed or denied? 00: Denied; 01: Originating and Terminating Calls allowed, Tandem Call denied; 11: All calls allowed. Normally assign data "11".

System Data 3, Index 2, Bits 0-3. switch hook flash starts timer. To calculate this value: (1 ~ FH) x 120 msec = Timer.

System Data 3, Index 2, Bits 4-7. Switch hook flash ends timer. To calculate this value: (1 ~ FH) x 120 msec = Timer. For both beginning and ending timers, assign 91H for 120 - 1080 msec.



T-62 Timed Reminder

General Description

This feature allows a Station user to set a Timed Reminder Call from the Station. The system calls up the Station at the designated time. When the Station answers the call, the Station user hears a message or Music-On-Hold.

Operating Procedure

To set a Timed Reminder Call

1. Lift a handset. Receive Dial Tone.
2. Dial the Timed Reminder (SET) Access Code.
3. Designate your desired time by dialing four digits. (for example, 16:30 → 1630)
4. Hear Service Set Tone or a recorded message which indicates the Timed Reminder Call has been set.
5. Hang up.

To cancel a preset Timed Reminder Call

1. Lift the hand set. Receive Dial Tone.
2. Dial the Timed Reminder (CANCEL) Access Code.
3. Hear Service Set Tone or a recorded message which indicates the Timed Reminder Call has been canceled.
4. Hang up.

To answer a Timed Reminder Call

1. A Station which has been set Timed Reminder starts ringing at the predetermined time.
2. Lift the handset.
3. Hear Music-On-Hold or a recorded message which indicates this call is a Timed Reminder Call.
4. Hang up.

Service Conditions

1. In the following cases, the Station user attempting to set/cancel this service receives Reorder Tone:
   a.) when SFI 41 of the service class is 0,
   b.) when CALL FORWARDING - ALL CALLS [C-5] is set.
2. The wake-up times are to be set at five-minute intervals. For example, if the user dials 1004, the wake-up time will be set at 10:00 a.m.
3. 500 Stations per Local Processor can be set for a single wake-up period. If the 500 Stations maximum is exceeded, the system will automatically set the wake-up time five minutes earlier (to the maximum of 75 minutes).
4. The wake-up time entered latest has the priority over the ones entered previously.
5. The no answer timer for the Wake-up Call is 30 seconds. After the no answer timer expires, no retry is performed.

   Note: No Answer Time is variable.(ASYD SYS 1 Index 139)

6. If the link block, etc. occurs while being recalled, no retry is performed.
7. When the Station is busy or lockout while being recalled, the Station is recalled three minutes after. This retry is repeated three times.

8. When CALL FORWARDING - BUSY LINE is set to the Station setting wake-up, the destination is recalled (except when the destination is not the Station Line, the Station setting wake-up is recalled). When the destination is busy, no retry is performed.
9. When the Announcement Trunk is blocked, the user receives music.
10. When the Timed Reminder Cancel Code is dialed, the user hears Service Set Tone or the announcement even when the service is not set.

Programming

STEP 1: **ASYD** - System Data 1, Index 139. No Answer Timer for Call Back. Assign 00H. (RAM Data is 3FH = 30 seconds.)

System Data 1, Index 162, Bits 6 and 7 tone type when a Timed Reminder Call is answered.

b7	b6	b7	b6
0	0 = Music-On-Hold	0	1 = Announcement

Note: When Music-On-Hold is selected, STEPs 4 through 8 are not required.

System Data 1, Index 163, Bit 7. Start time for Timed Reminder/Wake Up Call 0/1 = On time / 5min. earlier the appointed time.

System Data 1, Index 169, Bits 4. through 7. Maximum advancing time for Timed Reminder Calls, which becomes effective when the number of Stations which have appointed the same time exceeds the threshold value (maximum 500 STNs per LP or CPU). This value is set as follows.

5x N (min.) N ranges 0 to F (Hex).

STEP 2: **ASFC** - Allow this feature to proper Service Feature Restriction Classes (SFI 41). 0/1 = Restricted / Allowed

STEP 3: **ASPA** - Assign the following two Access Codes for this feature.

Timed Reminder (SET):SRV = "SSC: Service Code", SID = 48

Timed Reminder (CANCEL):SRV = "SSC: Service Code", SID = 49

STEP 4: **ARTD** - Assign Route Data for Announcement Trunk(s).

Note: To change the data for existing route (RT), initialize or reboot the trunk device such as circuit card or VS32.

STEP 5: **ATRK** - Assign Trunk data for the Announcement Trunk(s).

STEP 6: **MBTK** - Make idle the assigned Trunk(s).

STEP 7: **ARSC** - Allow the Connection between the assigned Trunk(s) and Stations.

STEP 8: **AAED** - Assign Announcement Equipment data.

EQP = 16 (Timed Reminder Set-up Message)

EQP = 18 (Timed Reminder Cancel Message)

EQP = 22 (Timed Reminder/Wake Up Call Message for Announcement Trunk)

C: Duration of Connection

0/1: Disconnect in 30 seconds / The connection is held until the station releases.

R: Sending Ringback Tone

0/1: To be sent / Not to be sent

A: Answer Signal Sending

0/1: No answer from in coming Trunk / Answer from Incoming Trunk

M: Multiple Connection

0/1: Single / Multiple Connection

RT: Route Number TK: Trunk Number

Standard programming is as follows:

C = 1, R = 1, A = 0, M = 1

T-63 Tone Block

General Description

This feature allows the user of a Station (via analog adapter) or an Analog Station (PB) to restrict any call. This prevents interruption of data transmission, before the Connection.

Note: This feature is not available in Australia.

Operating Procedure

To set the service

1. Lift the handset. Receive the Dial Tone.
2. Dial the Access Code of "Data Privacy on Demand; Entry".
3. Hear the Special Dial Tone or the Dial Tone (depending on the System Data).
4. Dial the desired phone number.
5. The Called Party answers.

Note: This service may also be set during communication by performing a switch hook flash.

To cancel the service
When the operating user cancels this service during the communication in progress

1. Perform a switch hook flash (the Called Party is held).
2. Hear the Special Dial Tone.
3. Dial the Access Code of DATA PRIVACY ON DEMAND [D-17] cancel.
4. Hear the Service Set Tone.
5. Press the switch hook button to reconnect the Held Party.
6. Tone Block is canceled.

Service Conditions

1. This service is available for the Analog Station or the Desktop terminal with analog adapter.
2. If the Station user stops to transfer a call in the middle of the operation, this service is not canceled.
3. This service is canceled when the Station, which has been set Tone Block, is released from the Connection.
4. Services to be restricted by setting Tone Block service are shown below.

ATTENDANT CAMP-ON WITH TONE INDICATION	[A-1]
BUSY VERIFICATION	[B-3]
EXECUTIVE RIGHT-OF- WAY	[E-1, 1D]
CALL WAITING TERMINATING	[C-12]
CALL WAITING - ORIGINATING	[C-31]

Interactions

1. When TONE BLOCK is set to the Desktop terminal which is assigned the DO NOT DISTURB (DND) Feature key, the lamp of the DND Feature key flashes. When the DND Feature key is pressed, the lamp goes off and the TONE BLOCK service is canceled.
2. A user can activate the current service for restricting a call (Data Privacy On Demand) as usual by:
   a.) dialing the Access Code for entry or cancel the service during communication, or
   b.) pressing the DND Feature key for entry or cancel the service.

Programming

STEP 1: **ANPD** - Reserve a Number Level for Feature Access and cancel. Select Normal Service (N) or Hooking Service (H) from the combo box for Connection Indexes. Assign NND in accordance with a predetermined Numbering Plan.

STEP 2: **ASPA** - Assign the Access Code for Tone Block as shown below:

CI = “H: Hooking”, SRV = "SSCA: Service Code Appendix", SIDA = 48 (Data Privacy on Demand; Entry)

CI = “H: Hooking”, SRV = "SSCA: Service Code Appendix", SIDA = 49 (Data Privacy on Demand; Cancel)

CI = “N: Normal”, SRV = "SSCA: Service Code Appendix", SIDA = 48 (Data Privacy on Demand; Entry)

STEP 3: **ASYD** - Assign the System Data.

System Data 1, Index322, Bit 6

0 = SPDT is sent after dialing the Access Code for this service.

1 = DT is sent after dialing the Access Code for this service.

T-64 Terminal Login via NAT

General Description

This feature is to translate a local IP address used within one network to a global IP address known within global network.

Service Conditions

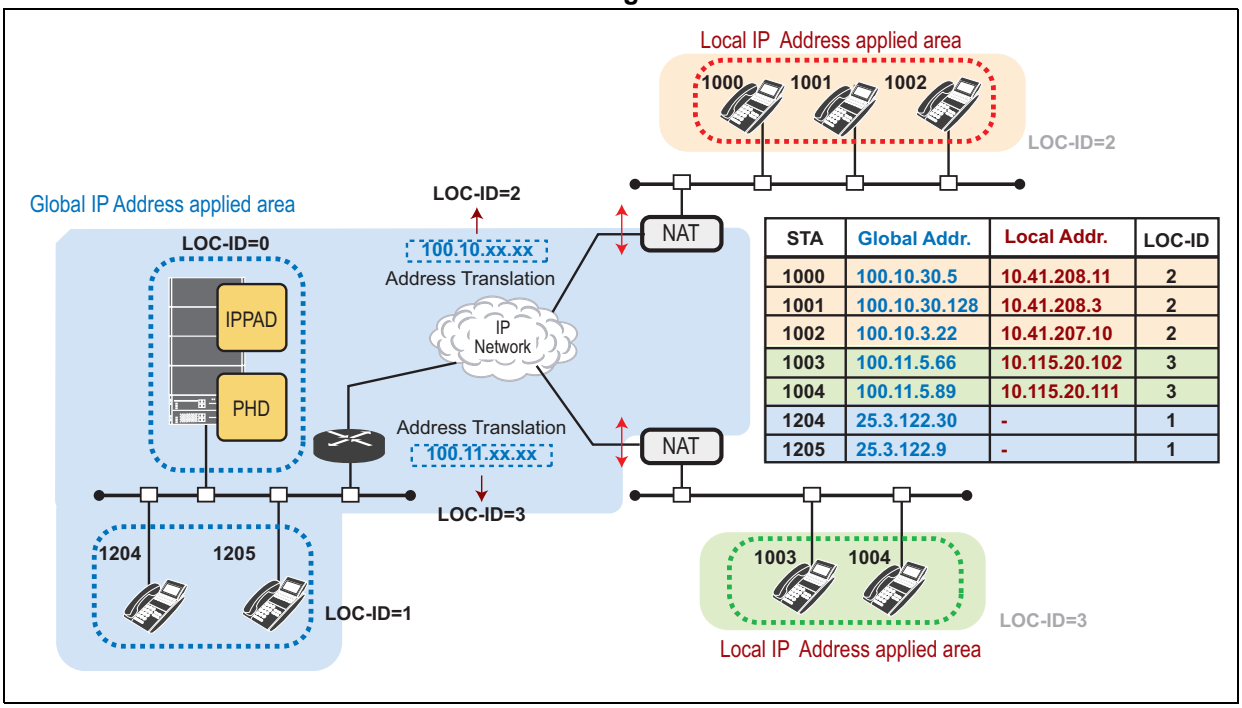
IP terminals can be connected over NAT when the network is under the following conditions:

- ASYDL, SYS 1, Index 824, Bit 6 = 1 is set.
- Global IP Address is assigned to MGC and the related devices such as PH, IPPAD circuit cards. (private IP Address can be assigned to IP terminals only.)
- As an address translation method, Basic NAT is used. (IP Masquerade is not supported.) Global IP address is assigned to an IP terminal placed in Local Network when NAT is performed. Only one single local IP address can be translated to one single global IP address.
- Global IP Address that is assigned to IP terminal does not change when the terminal has logged in.
- All IP terminals under a NAT device are assigned an identical LOC-ID. (Service features based on LOC-ID such as Bandwidth Control by Location ID cannot be used under the NAT.) LOC-ID is assigned to Global IP Address.

MGC can recognize if IP Address Translation was performed or not by comparing the Own IP Address of Initial Setup Request message with the source IP Address of the IP header. The terminal recognizes the global IP Address by Registered Address that was enclosed in the registration transaction.

The need of Address Translation is determined by comparing the two party's LOC-IDs. If the LOC-IDs are not the same, the Global IP Address is used. Otherwise, Local IP Address is used.

Terminal Login via NAT



STA	Global Addr.	Local Addr.	LOC-ID
1000	100.10.30.5	10.41.208.11	2
1001	100.10.30.128	10.41.208.3	2
1002	100.10.3.22	10.41.207.10	2
1003	100.11.5.66	10.115.20.102	3
1004	100.11.5.89	10.115.20.111	3
1204	25.3.122.30	-	1
1205	25.3.122.9	-	1

Programming

STEP 1: **ASYDL** - System Data 1, Index 824, Bit 6
1 = Peer-to-Peer Connection under NAT environment is in service **Note 1**, **Note 2**

When Peer-to-Peer connection within a private network is used, set b6 to 1.

Note 1: This bit allows Peer-to-Peer connection within the private network that is located under NAT environment, where MGC is located in global address side. Set this bit to 0 in the following cases:
- When unsupported network configuration such as Virtual Private Network (VPN) is used
- When NAT method other than Basic NAT (i.e. IP Masquerade) is used

Note 2: When Bit 6 is set to 0, global address is specified at the time of registration.

STEP 2: **ALOCL** - Assign LOC-IDs to Global IP Addresses. **Note 3**

Note 3: When NAT is used in the network, assign an identical LOC-ID to IP terminals under the NAT device.

T-65 Terminal Claim Handling

General Description

This feature allows users to send a claim notification (such as one against voice quality) by using the Desktop terminal. The user claim is output to SNMP manager and/or the EVLOG command to notify the maintenance personnel.

Operating Procedure

To operate Terminal Information/Terminal Claim Handling Feature

1. Push the Soft key or the Feature key assigned as a CLAIM or TERMINAL INFO Feature key to display a screen, which shows terminal's IP address (Terminal Info) or which requests user input for terminal information and Terminal Claim Handling information (Claim Menu).
2. If the Claim Menu is selected, enter "1" (Yes) or "0" (No) to select an item. To return to terminal mode, push the Soft key assigned as a CLAIM Feature key.

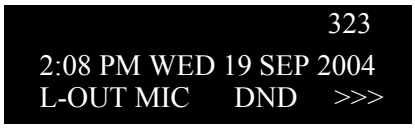
   Note: For Softphone, pressing "1" from keyboard does not send a trap. It works from the display keypad.

3. By pressing the Soft key or the Feature key assigned as a TERMINAL INFO key, the claim is sent to the SNMP manager as the TRAP packet and EVLOG command.

For more information about the way of outputting to SNMP manager, refer to SNMP [S-148] in this manual.

To display the terminal communicating status

1. Dial a destination number.
2. When opposite party answers the call, press the CALL INFORMATION DISPLAY Feature key assigned as FKY or a Soft key. Otherwise, press the TERMINAL INFORMATION SELECTION Feature key, then press "1".



3. LCD display on DtermIP changes from the regular display to Call Information Display.
   - Displayed information is automatically updated every five seconds.
   - A displayed item can appear on the first line by pressing its corresponding Digit key.



Lost Packet (0-99999999)
: If the value is over eight digits, "99999999" is displayed.
Codec
: G711/G711a/G723/G729a

Note: During conversation, the current packet loss is displayed dynamically every 5 seconds. When on-hook and press the CALL INFORMATION DISPLAY Feature key, the total number of packets lost is displayed.

4. On the third line, Up/Down/Exit of a Soft key is displayed. If Down is pressed, the third item (Payload) as follows is displayed.

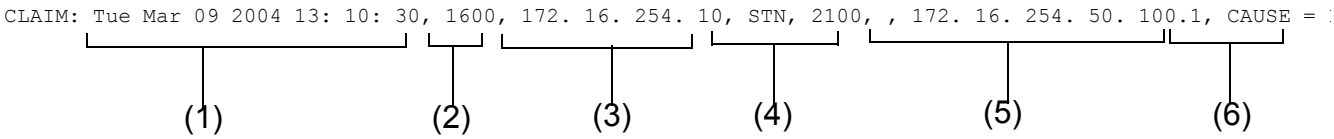
```
2. Codec        : G711
3. Payload      : XXXX
UP    DOWN        Exit
```

Payload Size (0-60ms)
: The range of the number is displayed in 10ms units.

5. Press the Exit key in the Soft key to change back to the Regular Display.

[Detailed information of SNMP Manager and EVLOG command]

The output format of claim information is in the following:



The claim message indicates that Station Number 1600 (IP address: 172.16.254.10) claimed "cut-off" while talking with another Station Number 2100 via IPPAD (C_RT=100, C_TK=1, IP Address: 172.16.254.50).

(1) Time when a claim notification has been received is indicated.

(2) Station Number that has sent a claim notification is indicated.

(3) IP Address of Station that has sent a claim notification is indicated.

(4) In the case of Extension-to-Extension connection, station number of opponent is indicated.
In the case of Trunk connection, route/trunk number is indicated.

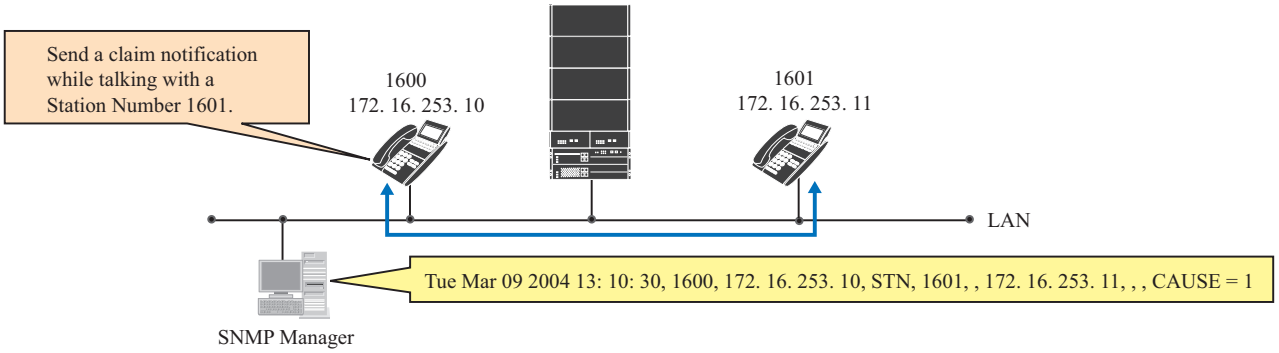
(5) IP Address of opposite system connected on the IP connection. In the case of speech via IPPAD, connection route and trunk number of the IPPAD are indicated.

(6) Contents of Claim are indicated. (Such as No Tone, Echo, Voice Cut, Delay, Noise, Low Volume, High Volume, One-way Speech, and Distortion)

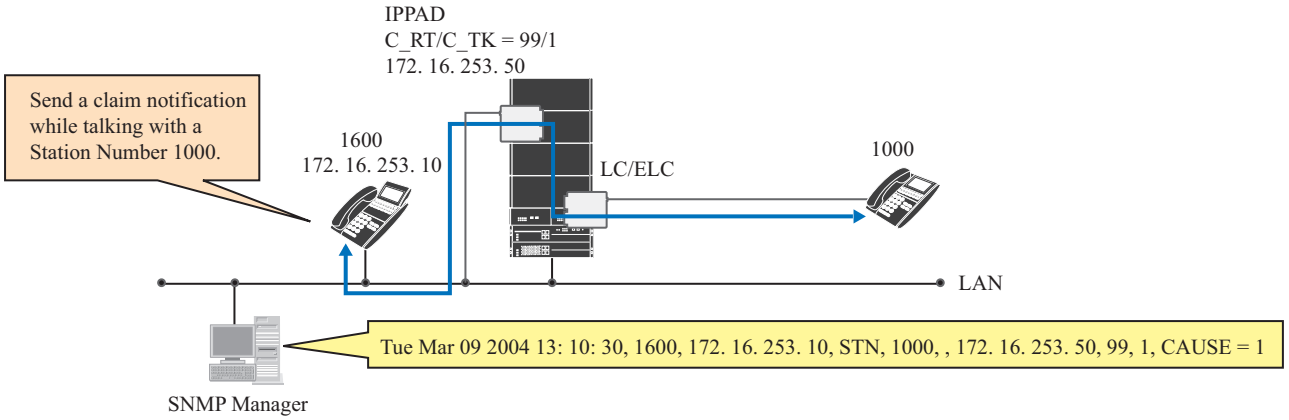
Available characters on SNMP Trap Display are up to 128 characters in ASCII format. Each parameter is separated by comma (,). Detailed information of EVLOG command are followed after “CLAIMS:”.

[OUTPUT IMAGE]

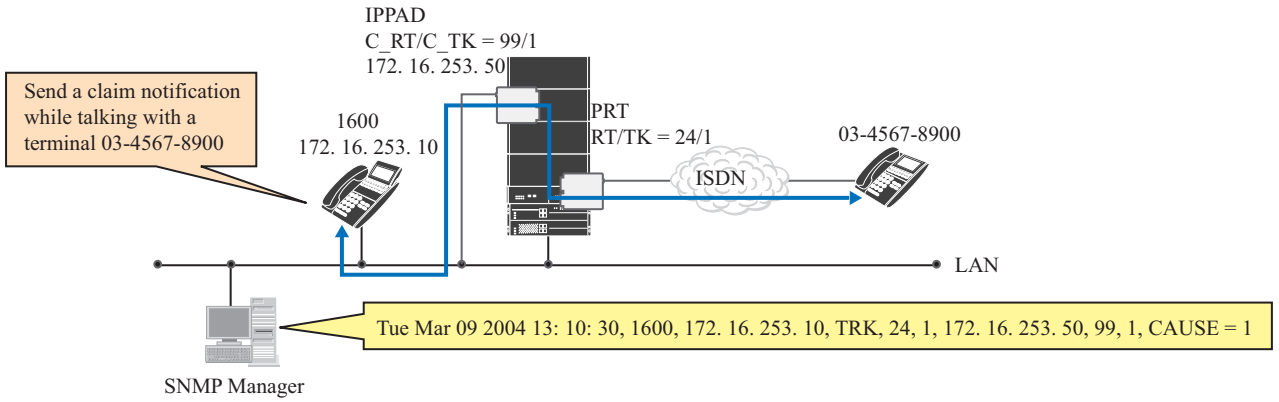
Example 1) When IP Station and IP Station are used



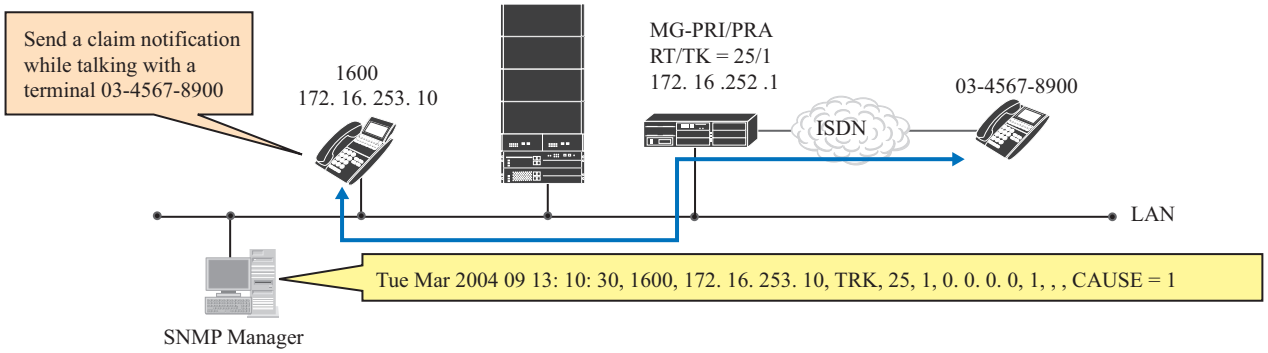
Example 2) When IP Station and Digital Terminal accommodated in a Node



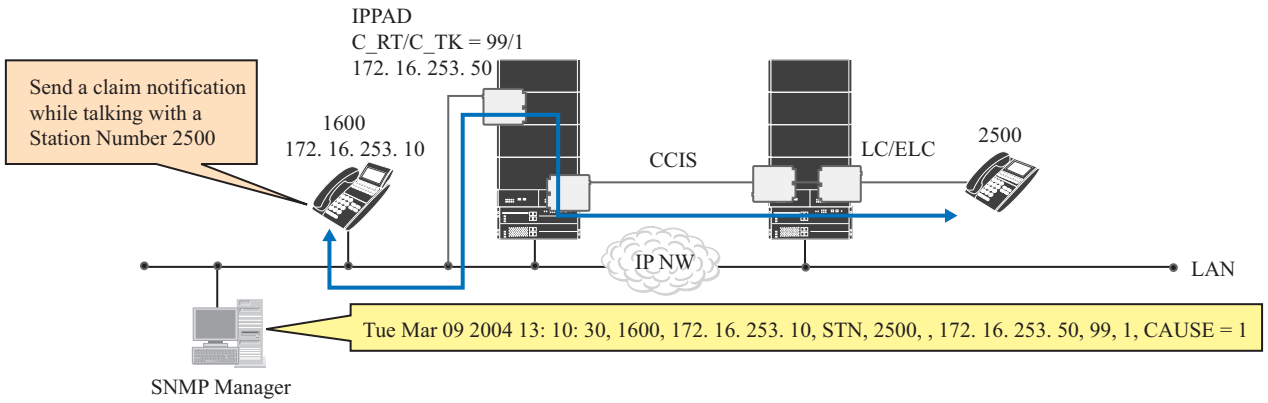
Example 3) When IP Station and Telephone connected to ISDN via Trunk



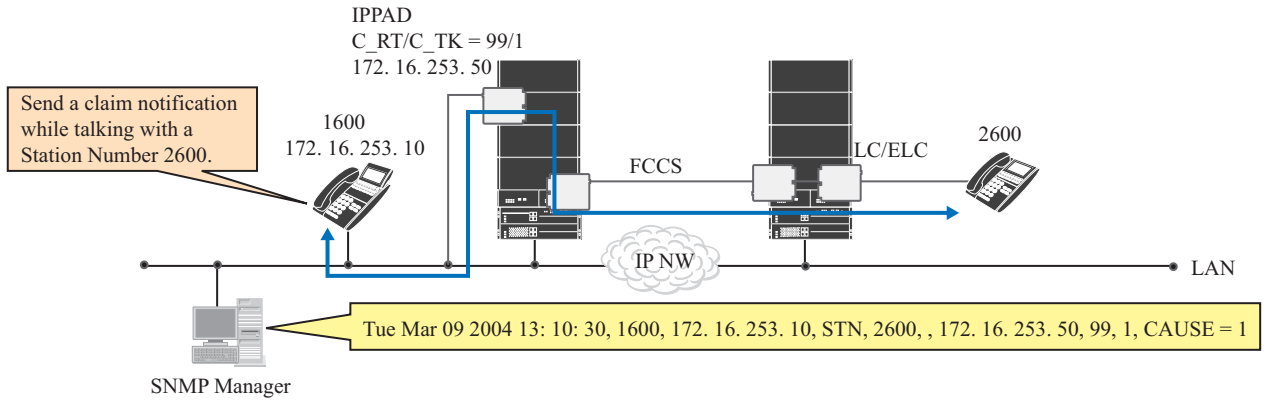
Example 4) When IP Station and Telephone connected to ISDN via MG-PRI



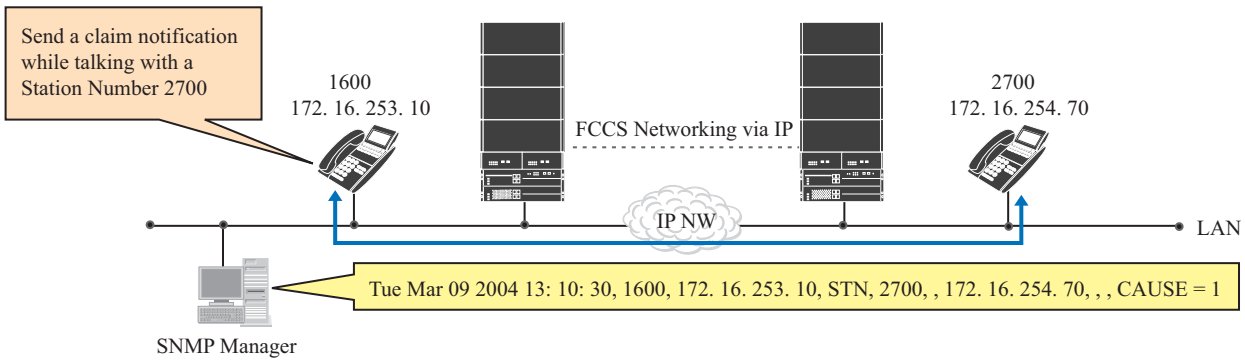
Example 5) When IP Station and Digital Terminal connected via CCIS



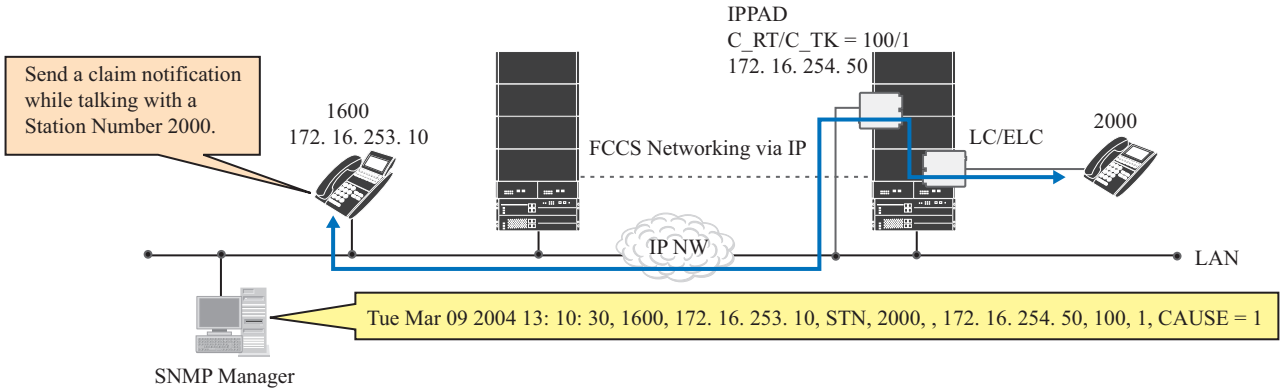
Example 6) When IP Station and Digital Terminal connected via FCCS



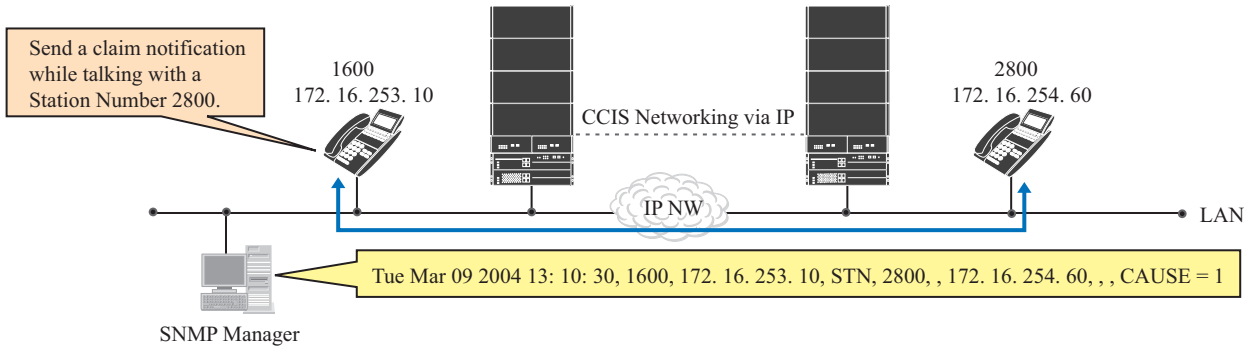
Example 7) When IP Station and IP Station are connected via FCCS Networking via IP



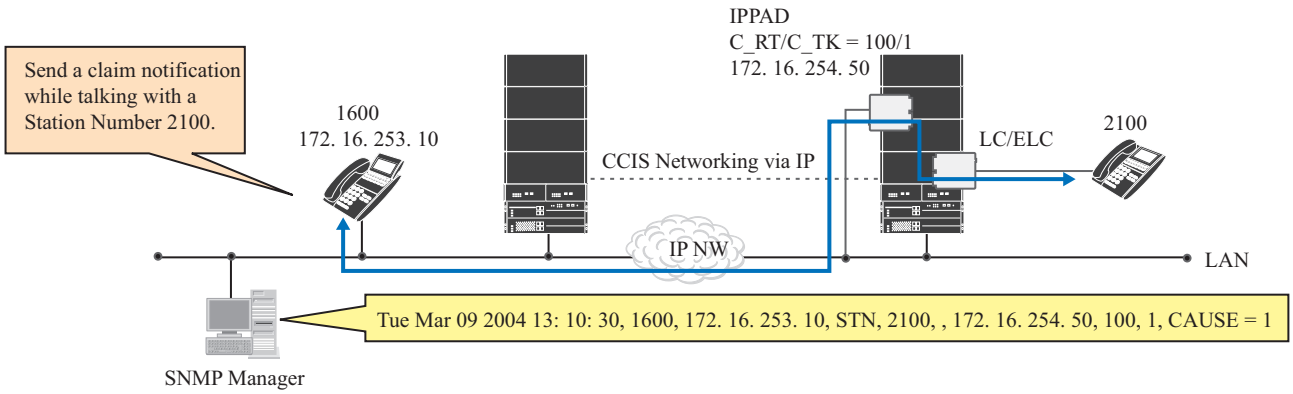
Example 8) When IP Station and Digital Terminal are connected via FCCS Networking via IP



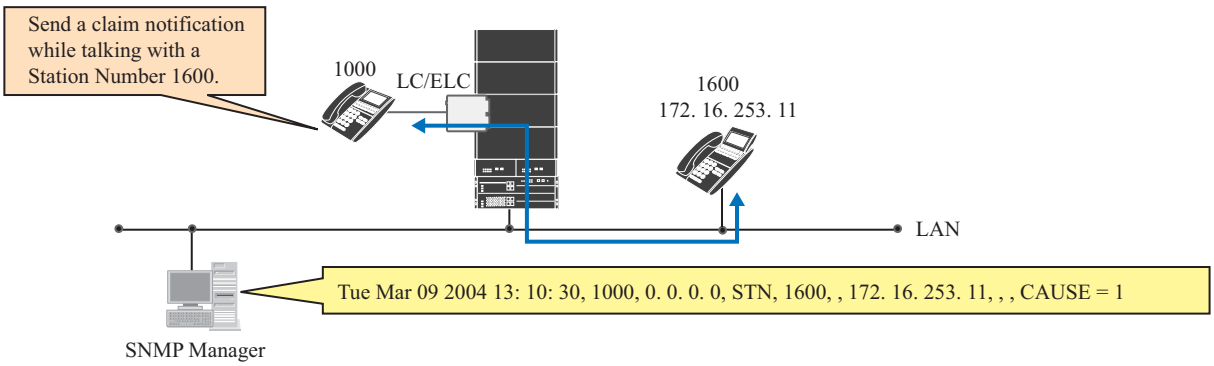
Example 9) When IP Station and IP Station are connected via CCIS Networking via IP



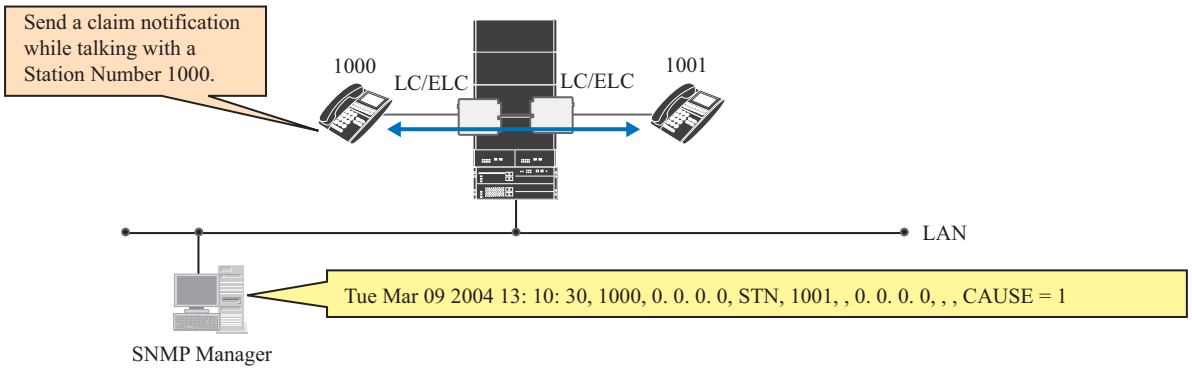
Example 10) When IP Station and Digital Terminal are connected via CCIS Networking via IP



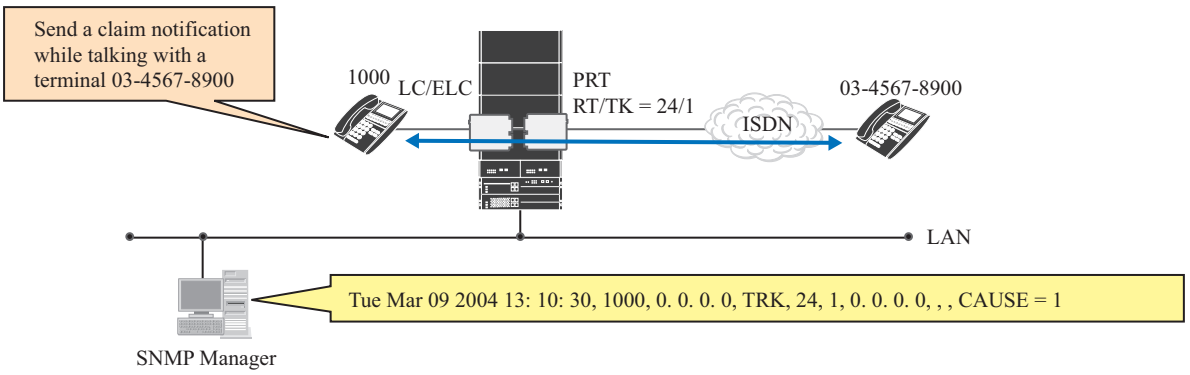
Example 11) When IP Station and Digital Terminal accommodated in a Node are connected



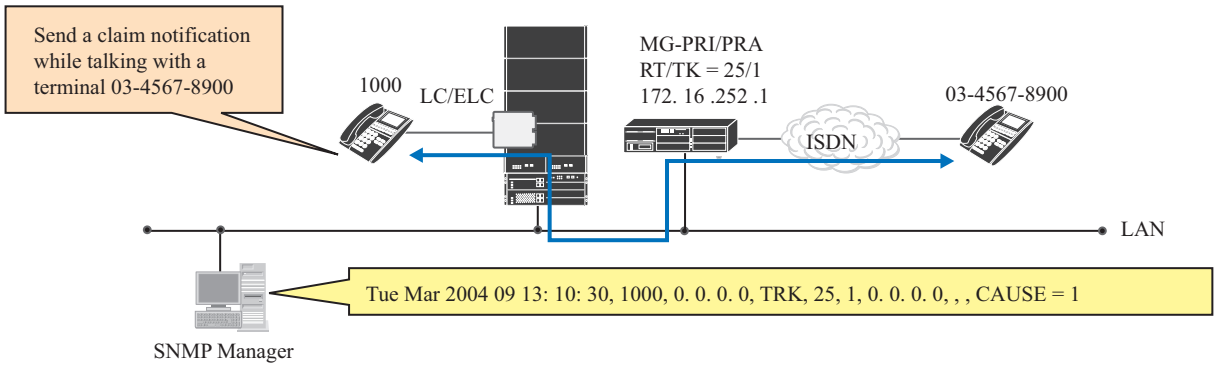
Example 12) When Digital Terminals accommodated in a Node are connected



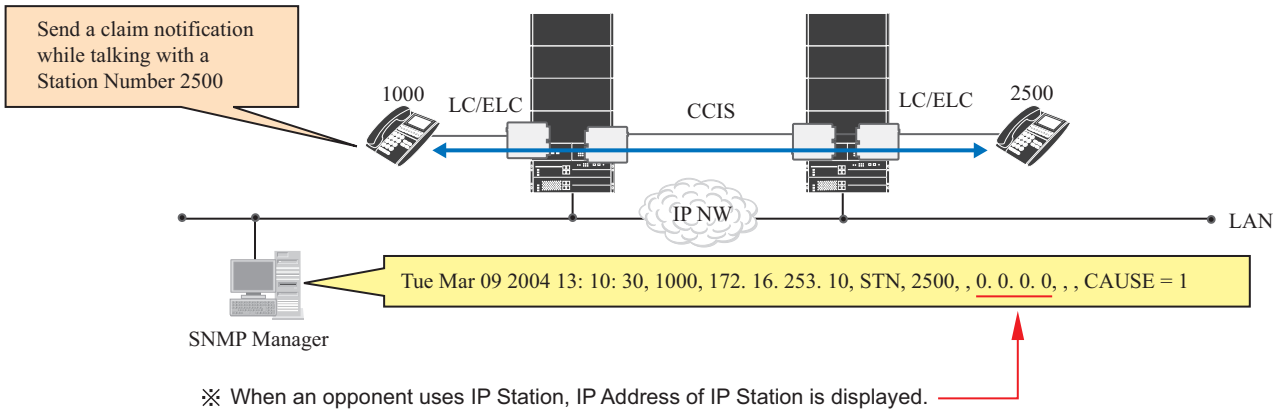
Example 13) When Digital Terminal and Telephone connected to ISDN via Trunk are connected



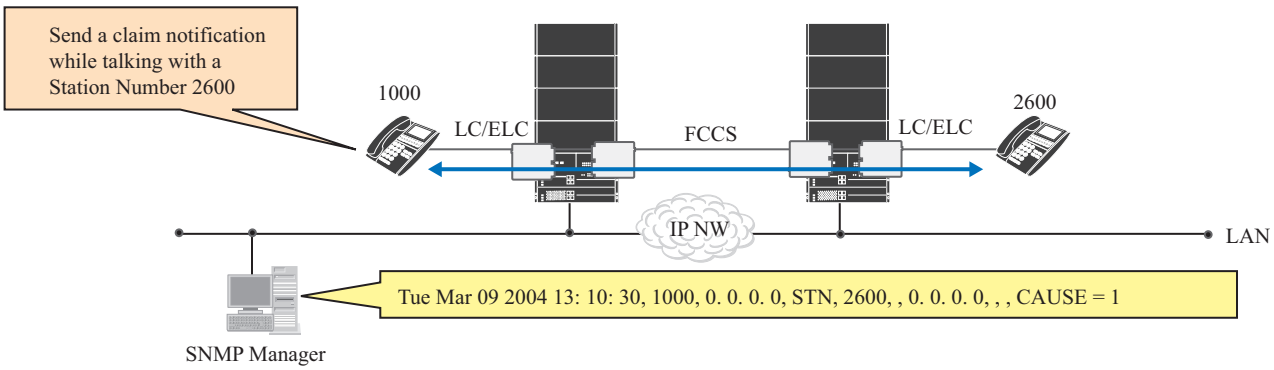
Example 14) When Digital Terminal and Telephone connected to ISDN via MG-PRI are connected



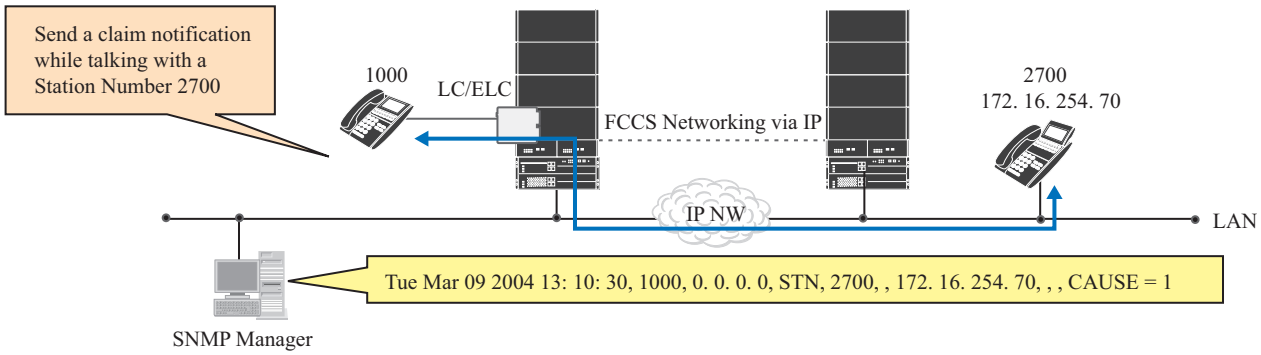
Example 15) When Digital Terminals via CCIS are connected



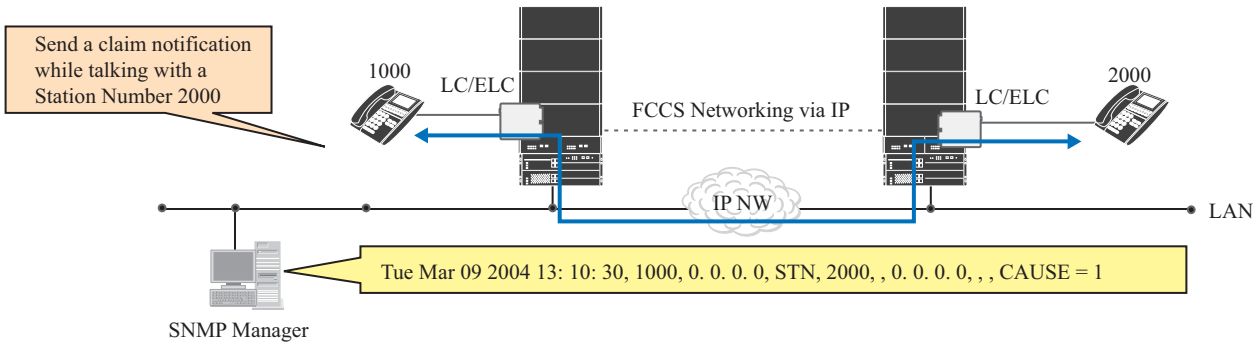
Example 16) When Digital Terminals via FCCS are connected



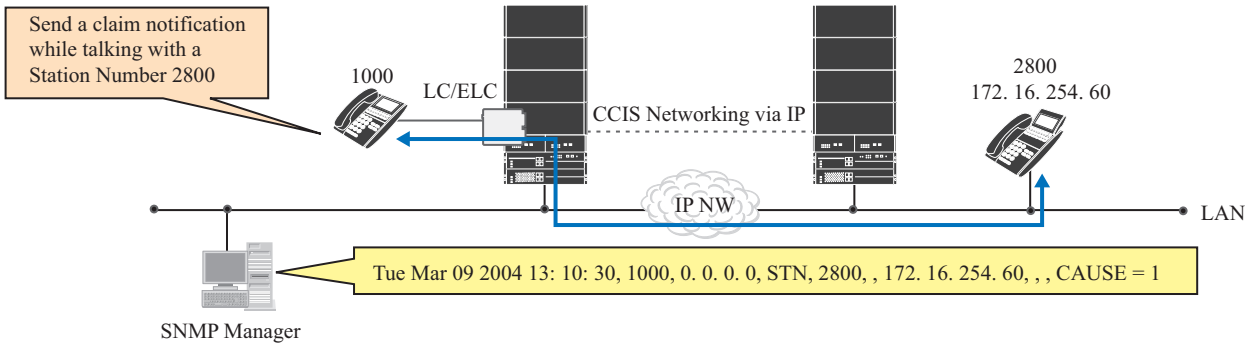
Example 17) When Digital Terminal and IP Station are connected via FCCS Networking via IP



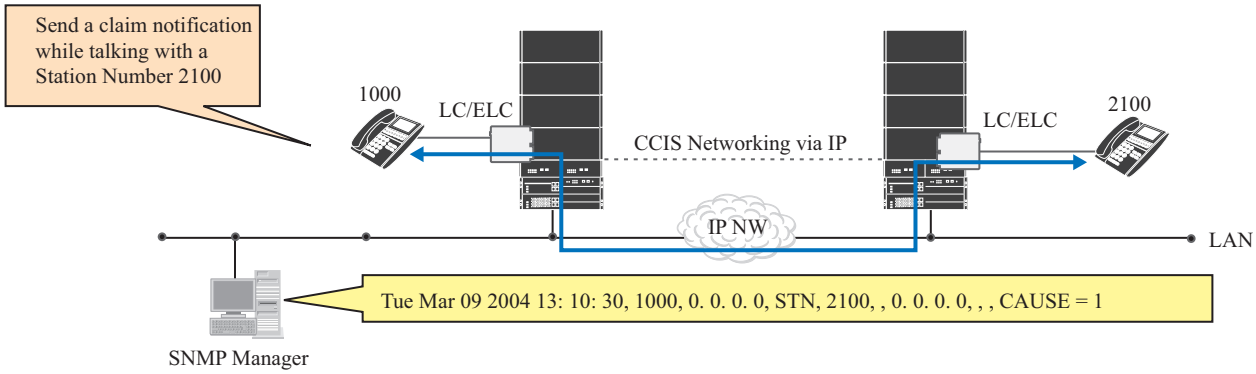
Example 18) When Digital Terminals are connected via FCCS Networking via IP



Example 19) When Digital Terminal and IP Station are connected via CCIS Networking via IP



Example 20) When Digital Terminals are connected via CCIS Networking via IP



Service Conditions

1. Since this service utilizes OAI functionality, refer to Data Programming Manual - OAI.
2. The number of external OAI connections is normally up to 16. However, when one internal OAI connection is used, the number of external connections is limited to 15.
3. This feature is available on the Desktop terminal with LCDs.
4. After the system has been initialized, first-push of CLAIM Feature key or INFO Feature key is inactive due to OAI MSF specifications.
5. When the Digital Terminal sends a claim notification, IP address is not displayed.
6. In the case of ISDN Line with MG-PRI, IP address of MG-PRI is not displayed.
7. IP Station information call display cannot be performed during speech. However, the information from the system is stored in the terminal. Pushing the Soft key replaces it with the latest information.
8. If a Station in a standby status activates the IP Station information call display, the previous information is displayed. When Station activating the IP Station information call display starts to speech, the previous information is cleared, then the information on progress call starts to be collected.
9. When network congestion causes the communication trouble, Network Busy is displayed on the second line.
10. SP-PHD is required for using IP Station information call display.
11. When the system is switched to SR-MGC, this feature depends on the firmware version of the SR-MGC.
12. While IP Station information call display is displayed, if assigned IP address is expired or a file is being downloaded, call information display is active but the call information display is overwritten with "IP Address Expired" or "Downloading".
13. The number of packet loss will be counted at that time speech is started or finished, depending on network status.
14. While a terminal is displaying the terminal communicating status, terminal information, and terminal claim handling, because the terminal is switched to terminal-control status, other features such as Transfer on the terminal cannot be performed.
15. The only 10-key panel can be used in idle status in Softphone for the selection of the Claim Menu. The numeric keypad of the PC cannot be used.
16. Use the Online Monitoring Feature when using this feature for DtermIP (SIP) or DT800/DT700 Series.

Programming

STEP 1: **ASYD** - Assign data as shown below:
SYS1,
Index 31, Bit 0-3: Mounted Capacity of Common Memory
04H: For Telephony Server with OAI name display
05H: FLF FCCS service is used in OAI features
Index 79, Bit 6=0 (Enable OAI/ACD Service)
Index 207, Bit 0=1(ACDP/IP is mounted), Bit 1=fixed to 0

STEP 2: **ASYDL** - Assign data as shown below:
SYS1,
Index 513 (01H is fixed at Local Data Memory)
Index 864,
Bit 0=1 (Built-in IP or ACDP is used)
Bit 4=1 [Assign eight ports of Status Monitoring for terminal (SMFN)]
Bit 5=0 (Clear SMFN when the FCCS Link down)

Note: The following Index 965 must be assigned after the data described above has been assigned.

Index 965,
Bit 3=1 (Internal Claim Handling Server is enabled)

- When assigning a Feature key as a CLAIM Feature key

STEP 1: **AOKC** - Assign the OAI Key code as follows:
KEY-CODE: OAI Key Code
F-KIND=1 [Terminal Mode Set Facility (MSF)]
C-TONE=0/1 (Chime Control on Normal End of MSF, TMF, and OAI are controlled/not controlled)
OP-CODE: Operation Code (See the table below.)

STEP 2: **AKYD** - Assign the data when the Feature key is assigned as a CLAIM Feature key.

Table 0-1 AKYD and AOKC Compatible Sheet

AKYD	AOKC	
Function Key	**OAI Key Code (KEY-CODE)**	**Operation Code (OP-CODE)**
34	1	
35	2	
36	3	
37	4	
38	5	
39	6	
40	7	
41	8	
42	9	
43	10	
44	11	
45	12	
46	13	
47	14	

OP-CODE	OPERATION	DISPLAY
170	Top Menu	Push 1-9, #=Menu
171	Terminal Information	IP Address Display → Local Mode
172	Claim Menu	**Claim Menu**

To delete the attributes, take the opposite way described in the above.

- When assigning the Soft key as a CLAIM Feature key

STEP 1: **AOKC** - Assign the data as shown below:
KEY-CODE: OAI Key Code
F-KIND=1 [Terminal Mode Set Facility (MSF)]
C-TONE=0/1 (Chime Control on Normal End of MSF, TMF, and OAI are controlled/not controlled)
OP-CODE: Operation Code

STEP 2: **AFDD** - Assign the data as shown below:
FKY: Assign the Function Key Code specified by the AOKC command.
DSP: Display Message on LCD

Operation Code (OP-CODE)	LCD Display (DIS)
170/173	INFO
171/174	STAT
172/175	CLM

Note: Up to six characters can be used per Soft key. However, if six characters are determined, it might be difficult to distinguish one from the next one.

STEP 3: **ADKS** - Assign the data as shown below:
The following are assigned if Soft Key Pattern other than default value (0) is used.
TN: Assign Tenant Number to be used for Terminal Claim Handling
STN: Assign Station Number to be used for Terminal Claim Handling
SKP: Assign Soft Key Pattern to be used for Terminal Claim Handling
LKP: Assign Line Key Pattern to be used for Terminal Claim Handling
PG: Enable/Disable LCD Page (PG) to be used for Terminal Claim Handling

STEP 4: **ADSL** - Assign the data as shown below:
SKP: Assign Soft Key Pattern specified by the ADKS command.
SN: Assign 0 (Empty) or 1 (Busy) in Status Number.
SKN: Assign Soft Key Number corresponding with OAI Key Code. The key number specified here should be the same key number assigned by the AOKC command.

T-66 Terminal Remote Login

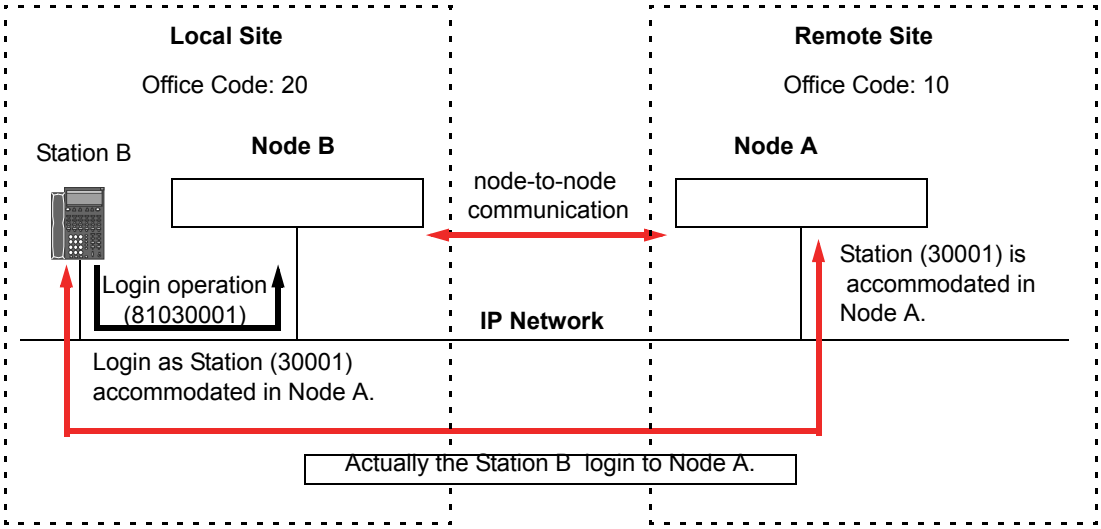
General Description

This feature allows a DtermIP to log in to the Remote Node without any change of the terminal settings.

Operating Procedure

There are two types of Login operations from DtermIP.

(a) CCIS over IP Network adopting the Open Numbering Plan
<Example>
- Access Code for outgoing: 8
- Station Number: 30001 (Accommodated in Node A)

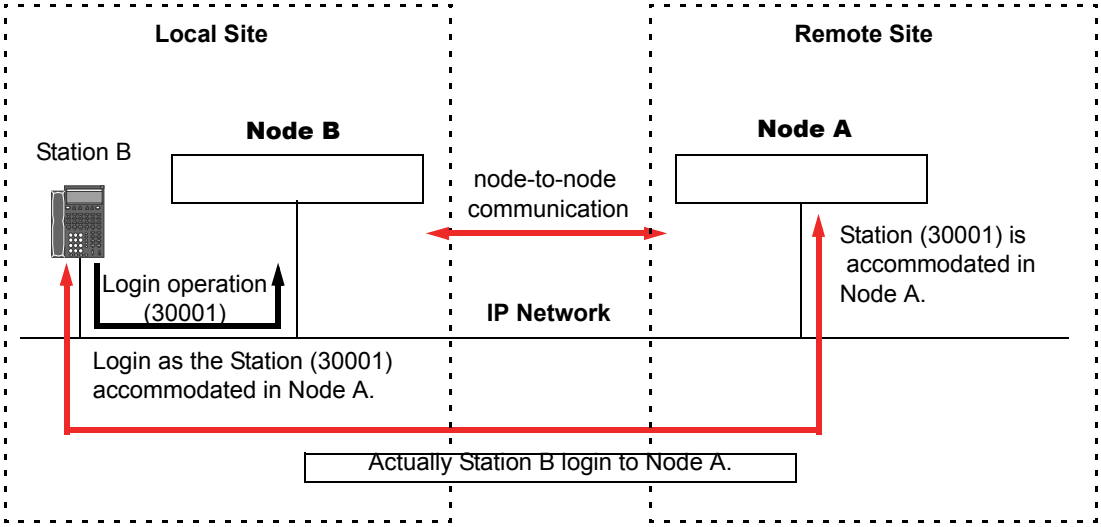


Enter the Access Code + Office Code + Station Number at Login prompt. And enter the password that is for the Station "30001" of Node A. Example of display is shown below.

```
Login    :  81030001
Passwd   :       *****

Cancel  BK  Set  OK
```

(b) FCCS over IP Network adopting the Closed Numbering Plan
<Example>
- Station Number: 30001 (Accommodated in Node A)



Enter the Station Number at Login prompt. And enter the password that is for Station "30001" of Node A. Example of display is shown below.

```
Login   :    30001
Passwd  :    *****

Cancel  BK  Set  OK
```

Service Conditions

1. In the case of CCIS over IP network, assign the numbering (LCR) data (Maximum 16 digits) to access to Remote Site from Local site.
2. In the case of tandem connection, the number of hops is up to 16.
3. This feature is available when FCCS over IP network or CCIS over IP network is established between the Local site and the Remote site. If the maximum number of nodes is 64, CCIS - FCCS Co-existing Network is available.
4. If the number of offices installed is fewer than 64 nodes, networks with office numbers and networks without office number can coexists in the same network.
5. The following should be specified before this feature is performed.
   - Registration Ports of DtermIP must be available on two-way communication
   - All the Registration Ports of the Nodes performing this feature should be the same port number.

6. When CCIS over IP is used between nodes, LCR setting is required for login to the nodes.
7. The Local and Remote Node must be connected through the IP Network. Certain port numbers used by DtermIP must be open or free among sites. (Those ports should not be blocked by firewalls.)
8. This feature is not available via SV9300/SV8300/NEAX 2400 IPS.
9. When Q-sig is the first choice and CCIS over IP is the second choice, this feature does not work. To avoid this, program another CCIS over IP route as the first choice and make busy all trunks in the route.
10. If the terminal key data is changed after the login through this feature, the terminal is automatically reset (After the reset, Login prompt is displayed on the LCD of the terminal). To log in to the same Station after the reset, press the Yes key when "OVERRIDE?" is displayed on the LCD of the terminal.
11. For "Login", enter the number that can be used when originating a call to the login destination from the operating terminal.
12. When the user logs out with the LOGOUT Feature key during Remote Login operation, the first log-in operation for the same node will end in failure within three minutes of the log-out operation.
13. This feature is available for DtermIP(SIP) with firmware v.2.0.4.0 or later.
14. When SIP TERMINAL AUTHENTICATION [S-152] is used, this feature cannot be performed.
15. Login Code feature, which you can log in with an attribute number instead of Station Number, is available for an FCCS over IP network. For more details, refer to LOGIN CODE [L-56].
16. This feature does not work in case of CCIS over IP Network adopting the Closed Numbering Plan.

Programming

To call up the stations in the Self Node (where the particular station is accommodated) from any Local Nodes, LCR or FCCS over IP data setting is required. Before proceed the data programming for this feature, be sure the following setting has been completed:

- LCR data setting (see the LCR/LCRS section in Chapter 2 of this manual)
- FCCS over IP data setting (see Data Programming Manual - FCCS)

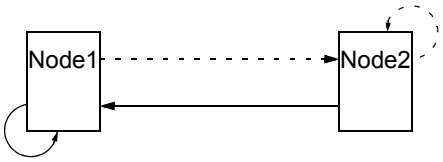
STEP 1: **ADTM -** Activate Remote Login Handler
Options - Activation Setting of Internal Server
Check box for Remote Login

STEP 2: **ADTM** - Assign of Data for TCP/IP Module. **Note 3, Note 4**
Options - Registration Service
MGC Search Address 1: Enter the IP address 1 according to the examples mentioned below
MGC Search Address 2: Enter the IP address 2 according to the examples mentioned below
Consecutive Login Failure Threshold [0-15]
Password Encryption [MD5(Default)/Proprietary]

Note 3: Assign the same IP address (ACT) that has been assigned in the "REG" parameter of "Configure Application Port" under "Options" tab in the same node.

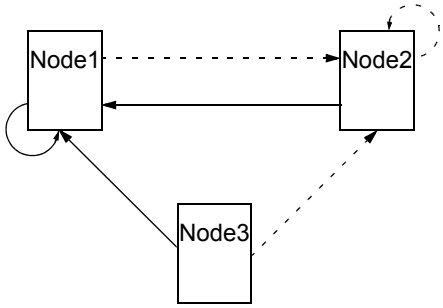
Note 4: Either the IP address of LAN1 (ACT) or LAN2 (ACT) can be chosen as the MGC Search Address. Note, however, that LAN2 is used when the existing IPX or SV7000 network is migrated to the SV9500 network. For more details about migration, refer to "Migration Guide". In the following examples, LAN1 is used.

- In case of two nodes in the network



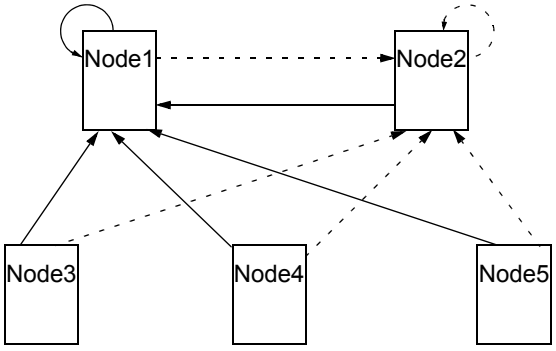
Node	IP Address1	IP Address2
Node1	IP Address of Node1 (Self Node) (IP Address of LAN1 (ACT))	IP Address of Node2 (IP Address of LAN1 (ACT))
Node2	IP Address of Node1 (IP Address of LAN1 (ACT))	IP Address of Node2 (Self Node) (IP Address of LAN1 (ACT))

- In case of three nodes in the network



Node	IP Address1	IP Address2
Node1	Address of Node1 (Self Node) (IP Address of LAN1 (ACT))	Address of Node2 (IP Address of LAN1 (ACT))
Node2	Address of Node1 (IP Address of LAN1 (ACT))	Address of Node2 (Self Node) (IP Address of LAN1 (ACT))
Node3	Address of Node1 (IP Address of LAN1 (ACT))	Address of Node2 (IP Address of LAN1 (ACT))

- In case of more than four nodes in the network



Node	IP Address1	IP Address2
Node1	Address of Node1 (self node) (IP Address of LAN1 (ACT))	Address of Node2 (IP Address of LAN1 (ACT))
Node2	Address of Node1 (IP Address of LAN1 (ACT))	Address of Node2 (Self Node) (IP Address of LAN1 (ACT))
Node3	Address of Node1 (IP Address of LAN1 (ACT))	Address of Node2 (IP Address of LAN1 (ACT))
Node4	Address of Node1 (IP Address of LAN1 (ACT))	Address of Node2 (IP Address of LAN1 (ACT))
Node5	Address of Node1 (IP Address of LAN1 (ACT))	Address of Node2 (IP Address of LAN1 (ACT))

T-67 Timed Forced Disconnect Trunk

General Description

This feature will disconnect a trunk from a Station after a predetermined time elapses.

Note: This feature is available in North America and Asia.

Operating Procedure

No Station user operation is required for this feature to function.
When a Station is connected to a trunk for the predetermined (by ASYD timer) amount of time, the caller will be sent a warning tone (800hz for 500msec). After 30 seconds, the caller is then disconnected.

Service Conditions

1. SMDR must be in-service on the route for this service to function.
   a.) For incoming SMDR, set SMDR2 = 1 in ARTD.
   b.) For outgoing SMDR, set SMDR =1 in ARTD.
   c.) ASYD, System Data 2, Index 3 must also be set for SMDR in-service.
2. Emergency (911) routes should not have the service set.
   a.) ARTI data for FDISC should be set = 0.
3. The warning tone will be sent to the Station in any state of connection or tone.
   a.) This may interrupt some other service, such as Dial tone in a Consultation hold dialing state.
4. This feature timer is based on trunk connection so if the call is transferred or forwarded the timer starts from the first answer supervision for the trunk.
5. This feature is not available for PS or BRI Stations.
6. When the disconnection occurs the Stations state will respond like the trunk released.
7. The timer for this service will run every 30 seconds. Because of this the time before the warning tone is sent can very up to 30 seconds.
8. This feature is available for the following lines.

   - Analog COT via COT circuit card or MG-COT
   - MFC-R2
   - ISDN via PRT circuit card or MG (PRI)
   - CCIS over IP (Internal PHC)

Programming

STEP 1: **ARTD/ARTDN** - For routes using this service.
Assign SMDR = 1 for SMDR service.
Assign SMDR2 = 1 for incoming SMDR service.

Note: To change the data for existing route (RT), initialize or reboot the trunk device such as a circuit card or MG.

STEP 2: **ASYD** - System Data 1, Index 32, Bit 7 = 1 for SMDR in service.

STEP 3: **ARTI** - For each route using this service.

FDISC= 0 for no Forced Disconnect on this route.
1 for Forced Disconnect only on outgoing calls.
2 for Forced Disconnect only on incoming calls.
3 for Forced Disconnect on incoming and outgoing calls.

Note: To change the data for existing route (RT), initialize or reboot the trunk device such as a circuit card or MG.

STEP 4: **ASFC** - For Service Feature Restriction Classes using this service.

SFI 226 = 0 Forced Disconnect is out of service.
1 Forced Disconnect is in service.

STEP 5: **ASYDL** - System Data 1, Index 552, Bit 6: 0/1 = Forced Disconnect out of service/in service.
System Data 1, Index 552, Bit 7: 0/1 = Do not register 11-A alarm/Register 11-A for Forced Disconnect.
System Data 1, Index 662 Timer for Forced Disconnect. **Note 1**

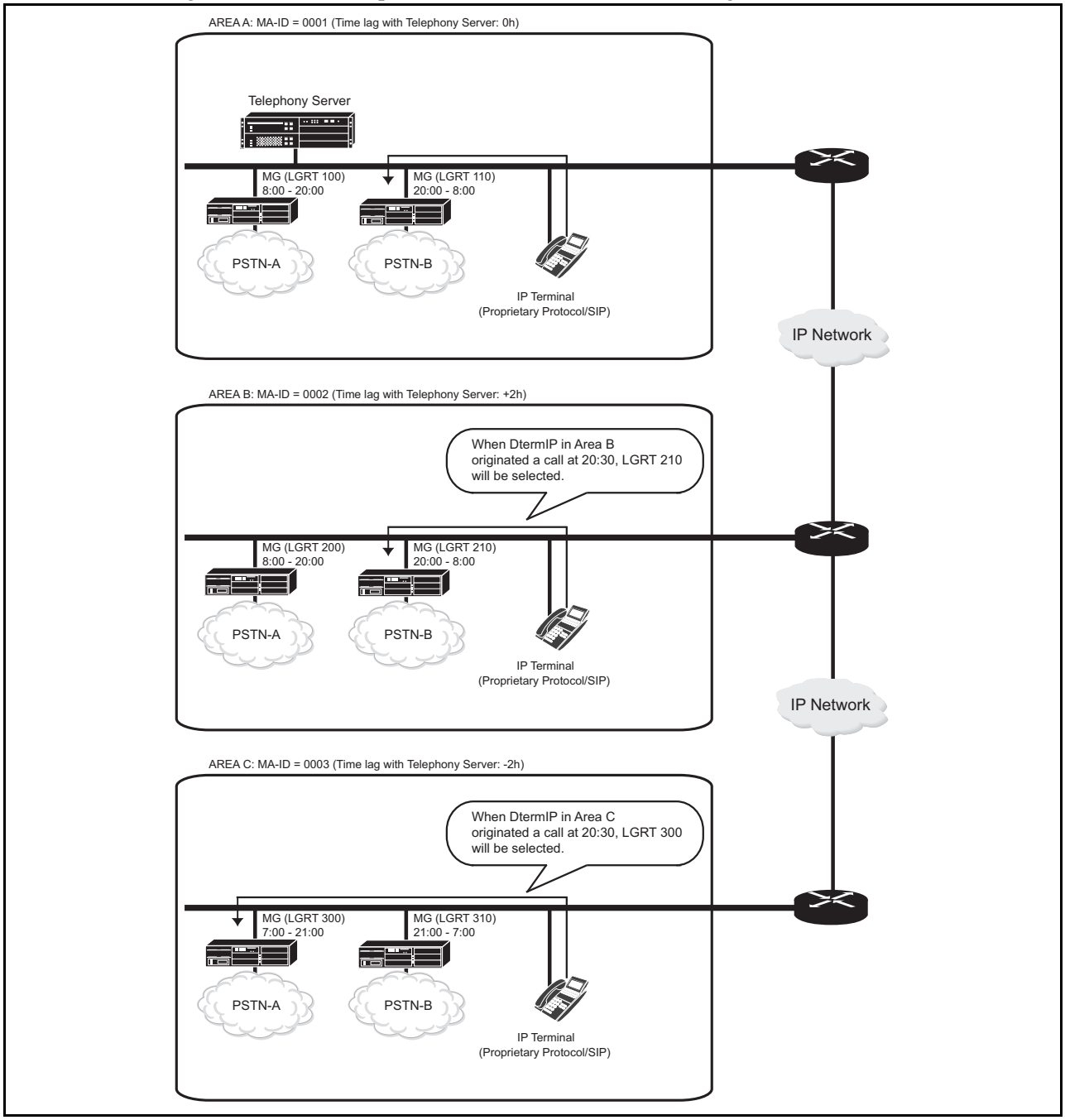
Note 1: Index 662 is MTC of the timer. Timer = MTC * 30 sec.
Ex:01 = 30 seconds, 02 = 60 seconds.

T-68 Time-based Routing for Multiple Time Zones

General Description

This feature enables the system to select outgoing route for each time zone that originating party belongs to by assigning the time of day change pattern and route selecting pattern for MA-ID of each area and corresponding to Multiple Time Zone.

- The below figure shows an example the route is selected when making a call at 20:30 as a time of each area.



Operating Procedure

No operating procedure is required.

Service Conditions

1. To specify the time zone of each area, assign MA-ID to the network (the assigned MA-ID to network address must be used for time changing pattern). For more information about the assignment of MA-ID, refer to Multiple Time Zones [M-114].
2. This feature is available when the originating system is LCR or LCRS.
3. For using this feature, assign ATCPL/ATCPN command and specify MA-ID for the terminal that originates a call.
4. The time change pattern for each MA-ID can be specified at 30 minute intervals (ex. 8:30 / 9:00).
5. The time change pattern can be assigned up to eight patterns including default pattern.

   Note: Maximum of eight patterns can be separated by dividing 1 day (24 hours) into 16.
6. When ATCPL/ATCPN command is assigned, this feature will be given a priority over the following related features.

   - LCR-Time of Day Routing-PRI/PRA
   - LCR-Attendant Manual Override-PRI/PRA
   - LCR-Clocked Manual Override-PRI/PRA

   Note: These above features will be used when a call is originated from the terminal that MA-ID is not assigned.
7. This feature is available only for IP terminal. When originating calls from other than an IP terminal, a conventional Outgoing Route pattern (which does not use this feature) will be selected.

   Note: A terminal accommodated in Analog MC uses MA-ID of network address assigned to the Analog MC when changing the time change pattern.
8. This feature is not available for SR-MGC.
9. This feature is not supported for Hotel/OAI/ACD system.
10. If this feature is used at backup node by FCCS Cluster, LCR follows the setting at the backup node.

Interactions

1. For more information about multiple time zone, refer to Multiple Time Zones [M-114].
2. This feature can be used with Least Cost Routing for MG-PRI [L-54]. For more information about it, refer to Least Cost Routing for MG-PRI [L-54].
3. The following are the service conditions when Call Forwarding - All Calls - Outside [C-28] is assigned:

- When a terminal (Intermediate Party) assigned with Call Forwarding - All Calls, Call Forwarding - Don't Answer [C-74] or Call Forwarding - Busy Line - Outside [C-60] is in the same Node as the Calling Party, Outgoing Route pattern for the MA-ID assigned to the Intermediate Party will be used. If the Intermediate Party is in a different Node (including other node in FCCS Network), a conventional Outgoing Route pattern (which does not use this feature) will be selected.

  Note: If PS is the Intermediate Party, when using Call Forwarding - All Calls - Outside [C-28], Call Forwarding - Busy Line - Outside [C-60], a conventional Outgoing Route pattern (which does not use this feature) will be selected.

- Regardless of the Node in which the Intermediate Party is registered, conventional Outgoing Route pattern will be selected when Logged Out IP Station - Call Destination or Call Forwarding - PS Incoming Incomplete is assigned.

Programming

STEP 1: **AMAIL/AMAIN** - Specify MA-ID and the area information.
MA-ID: Message Area ID (4digits) within the range from 0000 to 9999.
AREA NAME: Enter an area name within 16 characters.

STEP 2: **AMAA** - Add MA-IDs to terminals and trunks.
In case of SIP station, select KIND=Network to assign MA-ID to corresponding network address.

STEP 3: **ATCPL/ATCPN** - Specify the Time-of-Day change pattern information for each MA-ID.
MA-ID: Enter the Message Area ID (4digits [0001-9999])
FROM: Enter the start time of TDPTN (00:00-23:30)
TO: Enter the end time of TDPTN (00:00-23:30)
TDPTN: Time of day change pattern (0-7)

STEP 4: **ANPDL/ANPDN** - Assign the first digit of an Access Code for LCR/LCRS and digit number of the Access Code.
TN: Tenant Number
1stDC: The first digit of service Access Code. (0-9, *, #)
CI: Select Normal Service (N), Hooking Service (H), or Busy Service (B) from the combo box for Connection Indexes.
NND: Number of Necessary Digits for service Access Code (max. 6-digit)
BLF: Busy Lamp Field Display Activated on ATT (0/1 = Out of Service/In Service)

STEP 5: **ASPAL/ASPAN** - Assign the Access Code for LCR/LCRS.
ACC: Access Code for LCR/LCRS (0-9, *, #)
CI: Connection Status Index (“N: Normal”/”H: Hooking”)
SRV: Kind of Service (“LCR: Least Cost Routing”/”LCRS: Register Sender LCR”)
LGRT: External Logical Route Number (Dummy Route Number)

STEP 6: **AMNDL/AMNDN** - Specify a required maximum of the digits for MA-ID and DC.

Note: When the necessary data such as Destination Code and Maximum Necessary Digit Code has been already assigned, these data will be read out.

STEP 7: **ARNPL/ARNPN** - Specify the Reverse Numbering Plan Data for LDM/NDM.
LGRT: Logical Route Number (1-899)
ACC: Access Code (0-9, *, #)

STEP 8: **AFRSL/AFRSN** - Specify an Outgoing Route Selection Number (OPR). **Note 1**
TN: Tenant Number (1-899)
NPC: Number Pattern Code
MA-ID: Message Area ID (Max. 4digits [0-9999])
AREA NAME: Area Name
OPR: Outgoing Route Selection Number (0-4000)

Note 1: By changing the system data, you can reduce the amount of registered data for the AFRSN command, and the office data consumption. For details, refer to the Data Programming Manual - FCCS.

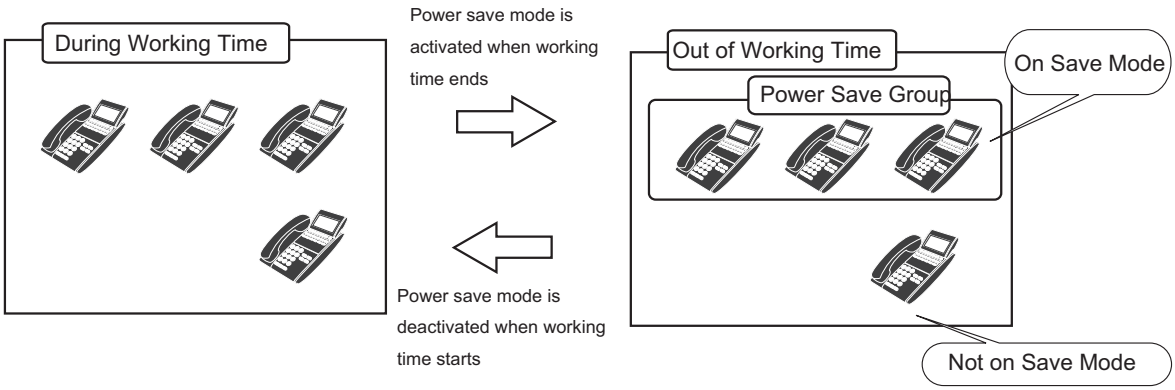
STEP 9: **AOPRL/AOPRN** - Specify routes and the condition for using the outgoing route selective pattern.
TDPTN: Time of Day Change Pattern Number (0-7)
OPR: Outgoing Route Selection Pattern Number (0-4000)
RA: Route Advancing Order (0-7)
E: Route Advancing End Indication (0/1 = Ended/-)
LGRT: Logical Route Number (1-899)
SKIP: Number of digits to be skipped (0-24)
PNL: Pattern Number Location (0-999)
OVFT: Overflow Tone (0/1 = -/Overflow Tone)
PRSC: Priority Restriction Class (0-15)

T-69 Terminal Power Save Mode

General Description

Electricity consumption can be reduced for a group of multiline telephones not in use by setting the terminals to a power save mode. The change to set power save mode and its cancellation are performed per each power save group as follows (all the methods can be used in combination):

- Automatic change based on a time schedule
- Change based on a terminal key operation
- Change with PCPro



Operating Procedure

1. In the case of an automatic change based on a time schedule, since the change to power save mode and its cancellation are performed based on the settings of such time schedule, it is not necessary to perform any operation on any of the terminals.
2. In the case of a change based on a terminal key operation, the procedure is as follows:

   (a) Power Save Mode Setting

   [1] Press the POWER SAVE Feature key.

   [2] A red light will flash on the POWER SAVE Feature key and a message indicating that the power save mode is off will be displayed on the LCD screen.

   [3] Press again the POWER SAVE Feature key.

   [4] A message indicating that the power save mode has been activated will be displayed on the LCD.

   [5] The stations of the power save group will change to power save mode.

Note: Read the point 4 of Important Notice on Service Conditions. In the power save group, a steady red light will light on the terminals that do not change to power save mode.

(b) Power Save Mode Cancellation

[1] Press the POWER SAVE Feature key.

[2] A red light will flash on the POWER SAVE Feature key and a message indicating that the power save mode is on will be displayed on the LCD screen.

[3] Press again the POWER SAVE Feature key.

[4] A message indicating that the power save mode has been disabled will be displayed on the LCD.

[5] The POWER SAVE Feature key that is lit on all the stations in the power save group will extinguish and the power save mode will be canceled for the group.

3. In the case of a change from PCPro, the procedure is as follows.

(a) Power Save Mode Setting

[1] Run the APSMCL command from PCPro.

[2] Choose [Immediate Switching] from the TYPE parameter and enter in the G-ID parameter the number of the power save group that you want to set to power save mode.

[3] Click the [GET] button.

[4] Current Status and the Switching Mode parameter will be displayed. Select [Power save].

[5] Click the [EXECUTE] button.

[6] The stations of the power save group will change to power save mode.

Note: In the power save group, a steady red light will light on the terminals that do not change to power save mode.

(b) Power Save Mode Cancellation

[1] Run the APSMCL command from PCPro.

[2] Choose [Immediate Switching] from the TYPE parameter and enter in the G-ID parameter the number of the power save group that you want to set to power save mode.

[3] Click the [GET] button.

[4] Current Status and the Switching Mode parameter will be displayed. Select [Power feed].

[5] Click the [EXECUTE] button.

[6] The POWER SAVE Feature key that is lit on all the stations in the power save group will extinguish and the power save mode will be canceled for the group.

Service Conditions

[Important Notice]

1. The terminals that can use this service are as follows.
    - Digital terminal **Note 1**
    - IP terminal (DT800/DT700 Series) **Note 2 Note 3**

    Note 1: Digital terminals accommodated in IPG and UG50 cannot use the service.

    Note 2: The valid versions of firmware for DT700 series are from version 5.0.4.0

    Note 3: DT830G/DT730G cannot use this service.

2. In case of using power save mode for Digital terminals, the following ELC card is required.
    - CH-16ELCA
3. Power save mode change is performed on a group basis. You can create a power save group by adding physical station numbers to it. The maximum number of groups available is 255. The only stations that can be registered in a group are terminals (TEC=12). The maximum number per group is the same as the maximum number of lines per system. Only stations that are in the same node can be registered in the same group.
4. When all the terminals in the group are digital, in the case of a change based on the terminal key operation, at least one of the group terminals that can perform changes must be left out of the power save mode. After the group is changed to power save mode, the terminal will remain in an active state.
   If you do not set a station that can perform changes to remain in normal state after the changeover (with the APSGL command settings), because all stations will get into a state of no power feeding, it will not be possible to cancel the power save mode with a terminal key operation.

    Note: Even if all terminals in a group are in a state of no power feeding, it is possible to cancel such state with PCPro or setting a time schedule.

5. To perform a change based on a terminal key operation, you can only use a terminal with Prime Line set to My Line.

[Interaction with other features]

1. In the case of SOFT PHONE WIRED PHONE HANDSET [S-147] changes can be done only with the wired telephone. Also, the soft phone and the wired telephone must have the same power save group number data (APSGL command G-ID parameter).

[Hardware/Software]

1. The POWER SAVE Feature key can be registered as a Soft key.
2. Regarding Digital terminals the following conditions apply.

    (a) During power save mode, since the power feeding of Digital terminals is stopped, no operation, including the reception and sending of calls, is possible. For this reason, it is recommended to let one terminal for emergency use out of the power save group or let it in an active state after the change (with the APSGL command settings).

(b) When being on power save mode, the outgoing/incoming call history is not saved. If you want to maintain the records use the following methods.

- Outgoing calls history: Last Number Called -D
- Incoming calls history: Call History

(c) If power supply is not available or any operation that requires ELC card initialization is performed (from PCPro with the AKYD, ADKS, MBST commands, etc), call control information (Call Indicator Display, call display, BLF function, etc) will be cleared.

(d) If a line stops being power fed, the BL lamp at the front of the ELC card will flash a green light.

(e) When power save mode is canceled, because of the power feeding control based on the ELC card mounting location information, the boot time can be different for each terminal (with no more than a 30-second duration).

(f) Digital terminals behave as terminals with a PFT adapter after changing to power save mode. After power save mode cancellation, the terminals behave again as Digital terminals.

3. In a power save mode, DT800/DT700 Series users have some limited functionality for the following features (note that other features are available in full functionality).

Power Save Function	Processing Content
Volume	Maximum volume restricted -20db
LED **Note 4**	Brightness restricted to half the usual
Hands-free function	Blocked
Backlight	Backlighting deactivated

Note 4: The LED of 8LK UNIT or DSS Console connected to DT800/DT700 series is not restricted by power save mode.

4. For DT800/DT700 Series, the volume level during power save mode will not be maintained after change the mode. If the power save mode is disabled, it will go back to the volume level before change to power save mode.

[Interaction with network services]

-

[Conditions of this service]

1. In the case of a change to save power mode based on a terminal key operation, there is no limitation on the type of terminal used for such operation. Also, DT800/DT700 Series and Digital terminals can be in the same group.
2. In the case of a call to a digital terminal on power save mode, if no transfer service is set, the caller will continually hear a ringing sound. If a transfer service is set, such service will be activated.

Note: Even on power save mode it is possible for DT800/DT700 series to perform incoming call and response operations.

3. The time needed to change to power save mode in all the terminals of the group will vary according to the number of terminals. For example, 100 terminals in 1 LP will need around 25 seconds and 200 terminals will require around 50 seconds.
4. On a terminal that is not in an idle state, changeover to power save mode is inhibited until the terminal becomes idle**.** The time needed for the turning into an idle state will be affected by the same rule stated on the point (3.). However, if during the waiting time one of the terminals enters into an active state, time count will be reset. **Note 5**, **Note 6**

   Note 5: A terminal that is not in an idle state means the following:
   - Being in the middle of a call.
   - Making an operation with the phone (putting on off-hook state, turning on the speaker, etc) or in the middle of an incoming call to My Line (however, in case of performing a local operation like using the operation menu, the power save mode change will be performed).
   - Other station is using as Sub Line the My Line of a terminal of the power save group (even if it is on hold).
   - There is a call held with CALL HOLD [C-6], EXCLUSIVE HOLD -D [E-4] or similar functions.
   - CALL BACK [C-1], CALL BACK - CCIS [C-44] or similar functions are set (if they are in the middle of being set the power save mode change will be performed).

   Note 6: In case of CALL PARK GROUP -D [C-176D] being set, the power save mode change will be performed.

5. If just after the system initialization of the Telephony Server, or during the reset of a terminal, one of the things mentioned below happens, all the terminals will boot in a non power save mode. The terminals will continue being in such non power save mode until a change is performed with one of the methods already explained.

   [Digital Terminal]

   - ELC card initialization

   [DT800/DT700 Series]

   - Terminal Log out/ Log in
   - Health Check error
   - A Power discontinuity of the terminal
6. If any data related to power save mode is changed while the terminals are in active state, the changes will be reflected the next time there is a change of the power save mode. However, if the power save group is erased the change will be reflected immediately and the concerned terminals will have their power save mode canceled **Note 7**.

   Note 7: Terminals on power save mode belonging to other groups will not be affected.

7. During power save mode, a red light will be lit steady on the POWER SAVE Feature key for terminals of the group that are not on such power save mode. The turn on/off of the light will happen at the same time that there is a change.

8. Regarding the automatic change based on a time schedule, the activation and cancellation of the mode can be registered to happen to each group on each day on intervals no lower than 15 minutes. It is also possible to only set its activation or cancellation.

   Examples of a setting

   (Example 1) From Monday to Friday change to power save mode from 19:00 to 8:00 and on weekend days stay all day on power save mode.

Day	Power save mode cancellation	Power save mode activation
Monday	8:00	19:00
Tuesday	8:00	19:00
Wednesday	8:00	19:00
Thursday	8:00	19:00
Friday	8:00	19:00
Saturday	-	-
Sunday	-	-

   (Example 2) Power save mode set to all day only on Wednesday, other days cancellation set with a time schedule to 8:30 and activation left to PCPro or performing a terminal keys operation.

Day	Power save mode cancellation	Power save mode activation
Monday	8:30	-
Tuesday	8:30	-
Wednesday	-	0:00
Thursday	8:30	-
Friday	8:30	-
Saturday	8:30	-
Sunday	8:30	-

9. The time scheduler is based on the clock of the Telephony Server. When the Daylight Saving Time option is enabled, the scheduler automatically accounts for one hour of Daylight Saving Time. This scheduler is not supported for multiple time zones.

   Note: If you have offices in multiple time zones, group them by time zone and create a separate schedule for each group, factoring in a time offset relative to the system clock.

10. If the time system of the Telephony Server is changed and exceeds the change time of the time schedule, the power save mode change will not be executed.
11. With the system data (ASYDL/N - SYS 1, Index 1262, Bit 5) it is possible to set a temporary deactivation of the time schedule function. For example you can keep the Desktop terminals in your office in power save mode on a holiday by deactivating the time schedule function a day before the holiday.

    Note: You can switch between power save and active modes with PCPro or a terminal key operation when time schedule function is not available.

12. A change performed with a terminal key operation will only affect the power save group to which the terminal belongs.
13. A POWER SAVE Feature key can be registered for more than one terminal of the power save group.
14. It is possible to perform a change based on a terminal key operation using a terminal that is in an idle state.
15. After pressing a POWER SAVE Feature key once, if one of the conditions listed below happens before pressing the key for the second time, the mode change does not take effect.
    - Operate with other key **Note 8**
    - An off-hook operation
    - 6 seconds pass after pressing the POWER SAVE Feature key for the first time (cancellation due to time-out)

    Note 8: The following buttons are excluded:
    - FEATURE key
    - Dial key, ANSWER key.
    - Cursor key (excluded the cursor on the left “redial”).

16. If more than one user presses the POWER SAVE Feature key simultaneously in an attempt to switch between power save and active modes:
    - If the timing of both the first and second key press is exactly the same across all those users, the change will happen as normal.
    - If the timing of the first key press is exactly the same across all those users, the change will happen at the timing when the next first POWER SAVE Feature key press is detected. A second POWER SAVE Feature key press by another user will be ignored. **Note 9**

    Note 9: If the second POWER SAVE Feature key press by another user is done more than 6 seconds after the first press, since the operation is canceled, this second press is detected as again a first press.

    - If another user presses the POWER SAVE Feature key for the first time after the power save mode activation, the key press is interpreted as the first key event for the power save mode cancellation. Thus, the second press of the POWER SAVE Feature key cancels the power save mode.
    - If another user presses the POWER SAVE Feature key for the first time after the power save mode deactivation, the key press is interpreted as the first key event for the power save mode activation. Thus, the second press of the POWER SAVE Feature key activates the power save mode.
17. The following conditions apply to a change done with PCPro.

    (a) If a mode checking is done with the APSMCL command, the actual mode at the time of performing a change and at the time the checking was done may be different.

    (b) The time required for the all terminals within the group to enter into the power save mode due to an immediate switchover is the same as with other change methods.

[Other]

-

Programming

1. APSGL (Assignment of Power Save Group ID for LDM)

   TN: Tenant Number
   STN: Station Number (Maximum 6 digits)
   G-ID: Power save group ID (1-255)
   No Power Saving:
   In case of setting group terminals that do not enter into power save mode, select the check box.

[Time Schedule Change]

1. ASYDL/ASYDN (System Data (LDM/NDM))

   SYS1 Index 1262

   Bit 5: Terminal power save mode change based on a time schedule.

   0/1: Enabled/Disabled

2. APSMCL (Assignment of Power Save Mode Change Data for LDM)

G-ID: Power save group ID (1-255)
TYPE: Time Schedule

Power Save Mode

From Sunday to Saturday set the following parameters.

Disable:

Select the check box of the days when power save mode is not used

Hour (0-23)

Minute (0/15/30/45)

Choose the time for power save cancellation

Enable:

Select the check box of the days when power save mode is used

Hour (0-23)

Minute (0/15/30/45)

Choose the starting time for power save mode

[Change based on a terminal key operation]

1. AKYD (Assignment of Key Data for Desktop Terminal)

   KYN: Key number
   KYI=1 (function key)
   FKY=184 (Power Saving)

   • In case of a Soft key

2. ADSL (Assignment of Desktop Terminal's Soft Key on LCD)

   SKP (0-15)
   SN=0 (idle)
   SKN (0-5)
   FKY=184 (Power Saving)

T-86 Timing Start

General Description

This feature enables the system to recognize the end of dialing by using not the digits of the dialed number but a specified timing. By pressing "#" or "*" key after dialing the station number, the station user will not have to wait for ring start.

Operating Procedure

1. Lift handset or press the SPEAKER key. Hear Dial Tone.
2. Dial the desired station.
3. Wait for inter-digit time-out or dial "#" or "*". Access to the desired service or station is completed.

Service Conditions

1. The Interdigit Timer (ASYD, SYS1, Index129, Bits 0 to 3) will be used for TIMING START if "#" or "*" is not assigned as a TIMING START character with the system data (ASYD, SYS1, Index161, Bit 6 or 7), or if the station users forgets to press the TIMING START character. In that case, ISDN Sub Address Dialing is not available.
2. This feature is available only when station numbers are dialed.
3. When "*" or "#" is used for TIMING START, it cannot be used as the second or third digit of any service feature access code or station number.
4. This feature is not available for Standard SIP telephone and Attendant Console.

Programming

STEP 1: **ASYD** - System Data 1, Index 129. Register Inter-digit Timer. Miscellaneous Timer Counter (MTC) is to be assigned a value from 0 Hex to F Hex (0-15). Timer Value Setting is MTC × 2 seconds.

Note: When this data is 00 Hexadecimal, the default data is automatically set to 6 seconds.

System Data 1, Index 161, Bit 6. Timing Start using "#" Code.
0/1 = Ineffective/Effective
System Data 1, Index 161, Bit 7. Timing Start using "*" Code.
0/1 = Ineffective/Effective
System Data 1, Index 170, Bit 4. Timing Start
0/1 = Not required/Required

U-1 Uniform Call Distribution (UCD)

General Description

This feature distributes Incoming Calls to a UCD Group of up to 100 Stations. Calls are distributed to Idle Stations in a circular pattern, in the order in which they arrive.

Operating Procedure

To transfer a call to a busy UCD Group

1. All the Stations in a UCD Group are busy.
2. A Station user not in the group answers an Incoming Call (Trunk or Station).
3. The Station user dial the UCD primary number after pressing the switch hook or the TRANSFER key:
   a.) The Station user hears Busy Tone.
   b.) The Calling Party is held.
4. The Station goes on-hook.
   a.) The Calling Party hears Ringback Tone (during queuing).
5. The UCD Group becomes idle:
   a.) The call terminates to an Idle Station in the UCD Group.
   b.) The Calling Party hears Ringback Tone. (during ringing)
6. The Idle Station answers the call:
   a.) The Calling Party is connected to the UCD Station.

To set Busy Out using Access Code from idle status

1. Lift the handset and Receive DT (Dial Tone).
2. Dial the Access Code for setting Busy Out.
3. Receive SST (Service Set Tone).
4. Replace the handset.

To cancel Busy Out using Access Code from idle status

1. Lift the handset and Receive DT (Dial Tone).
2. Dial the Access Code for canceling Busy Out.
3. Receive SST (Service Set Tone).
4. Replace the handset.

To set Busy Out using Access Code from busy status

1. Perform a switch hook flash and Receive SPDT (Special Dial Tone).
2. Dial the Access Code for setting Busy Out.
3. Receive SST (Service Set Tone). **Note 1**, **Note 2**
4. Perform a switch hook flash again to back to the conversation.

To cancel Busy Out using Access Code from busy status

1. Perform a switch hook flash and Receive SPDT (Special Dial Tone).
2. Dial the Access Code for canceling Busy Out.
3. Receive SST (Service Set Tone). **Note 1**, **Note 2**
4. Perform a switch hook flash again to back to the conversation.

Note 1: You would return to the conversation when 30 seconds has been elapsed after you have started to hear SST.

Note 2: You will be called back if you release the call while you hear SST.

To set Busy Out using Line/Feature key from idle status

1. Press the BUSY OUT Feature key.
2. The LED of the Feature key lights ("BYO SET" is displayed on LCD for 5 seconds).

To cancel Busy Out using Multi Line Feature key of Desktop terminal from idle status

1. Press the BUSY OUT Feature key.
2. The LED of the Feature key is extinguished ("BYO CNCL" is displayed on LCD for 5 seconds).

To set Busy Out using Multi Line Feature key of Desktop terminal from Busy Status

1. Press the BUSY OUT Feature key.
2. The LED of the Feature key lights ("BYO SET" is displayed on LCD for 5 seconds).
3. Back to the conversation. **Note 3**

To cancel Busy Out using Multi Line Feature key of Desktop terminal from Busy Status

1. Press the Busy Out Feature key.
2. The LED of the Feature key is extinguished ("BYO CNCL" is displayed on LCD for 5 seconds).
3. Back to the conversation. **Note 3**

Note 3: You can talk with the calling party even during a setting/canceling operation.

Service Conditions

1. A station belongs to a UCD Group and its station number is a UCD number is called "controlling station." If a caller dials the station number of the non-controlling station in UCD Group, the line is connected to the dialed station directly. When the dialed station is in busy state, the caller hears Busy Tone. (That is Uniform Call Distribution (UCD) [U-1] and UCD hunting are not performed.)
2. The maximum number of UCD Groups is 63.
3. A UCD Group may have 100 Stations.
4. The Route Restriction class (RSC) of Stations in an Uniform Call Distribution (UCD) Group must be identical.
5. UCD Groups are assigned through the PCPro.
6. Stations are configured in a circular-type Hunt Group. A call to the group starts at the first Idle Station, after the last Station that received a call.
7. If all of the group's Stations are busy, the call waits in a queue. The caller hears Ringback Tone. The first call in the queue will be answered.
8. If a Sub Line of a Station belongs to an UCD group where one or more incoming calls are waiting in the queue, the station is not called back immediately even though its My Line becomes available. The timing of the call back is; about 4 seconds if DELAY ANNOUNCEMENT - UCD [D-31] is not in service, and about 8 seconds if it is in service.
9. CALL WAITING LAMP-UCD [C-76] will light (if programmed) after calls are in queue and will flash when the threshold value for that UCD Group is reached.

10. A Station user can block UCD Calls by dialing a Busy Out (BYO) Code (one to three digits), or if it is a Dterm, by pressing the DND or BYO Feature key. The Station user can still originate calls from and receive calls to its own Telephone Number. The Station user can dial the BYO Code from a normal Dial Tone Connection or from a CONSULTATION HOLD-ALL CALLS [C-17] condition.
11. Each UCD Group is assigned a UCD Group Number, allowing DIRECT INWARD DIALING (DID) [D-8] Calls to be routed to selected groups.
12. If the Attendant or the Station user transfers a call to a busy UCD Group, the call will wait in queue until a Station in the group becomes idle.
13. Incoming Calls to individual Stations in the group are not distributed by UCD hunting. Only calls to the UCD Group Number follow the circular distribution.
14. Each UCD Group has a controlling Station. Only the controlling Station or the Attendant can activate or deactivate CALL FORWARDING-ALL CALLS [C-5] when the UCD Group Number is not a phantom number.
15. If CALL FORWARDING-ALL CALLS [C-5] is set on a controlling Station and the Destination Station is idle, a call to the UCD Group is transferred. If the designated Station is busy, the user will hear Busy Tone.
16. If CALL FORWARDING-ALL CALLS [C-5] is set on a non-controlling Station, that Station is skipped in any UCD hunt.
17. The Attendant can use BUSY VERIFICATION [B-3] to check the busy/idle condition of the UCD Stations. If the BYO Code or key has been used, the Attendant will receive a Reorder Tone.
18. CALL WAITING-ORIGINATING [C-31] can be activated by an individual UCD Station user towards another Station within the system. Also, another Station user can activate this feature towards an individual UCD Group Station. CALL WAITING-ORIGINATING [C-31] can be activated toward a UCD Group Number when all the Stations and the queue are busy. Waiting is set to the controlling Station.
19. CALL WAITING-TERMINATING [C-12], when assigned to an individual UCD Group Station, allows the Calling Station user to wait on the Called Station's line when it is busy. The Group Station user can be connected to the Waiting Call by hanging up. The Station rings and the call is connected when the caller goes off-hook. These calls have priority over UCD Calls in the queue.
20. CALL BACK [C-1] can be activated by an individual UCD Group Station user towards another Station. Also, another Station user can activate this feature towards an individual UCD Station. CALL BACK [C-1] can be activated toward a UCD Group Number when all the Stations and UCD queue are busy. CALL BACK [C-1] is set to the controlling Station.
21. A UCD Group Number can be used as the Destination Station of DIRECT-IN TERMINATION [D-7] or of NIGHT CONNECTION-FIXED/FLEXIBLE [N-1/N-2].
22. A UCD Group Number can be assigned as the Destination Station of OFF-HOOK ALARM [O-6], PRIORITY CALL [P-9], ATTENDANT NIGHT TRANSFER [A-19], and LDN NIGHT CONNECTION [L-28, 29].
23. If a Phantom Number is assigned to a controlling Station, UCD hunting may or may not occur (depending on the APHN command setting), when the Phantom Number is dialed.

24. Three types of traffic measurements can be provided for the UCD feature:
    a.) UCD route peg count:
    - Count of Incoming Calls from each route to each UCD Group
    - Count of calls answered by UCD
    - Count of other calls (abandoned, restricted, blocked)

    Example:

From 07/24 08:00 to 07/24 09:00				
Route	**UCD Group**	**Incoming**	**Answer**	**Abandon**
10	1	9	7	2
	2	15	15	0
Route Total	24	22	2	

    b.) UCD Group Peg Count:
    - Count of Incoming Calls to each UCD Group
    - Count of answered calls
    - Count of other calls (abandoned, restricted, blocked)

Example:

From 07/24 09:00 to 07/24 10:00			
UCD Group	**Incoming**	**Answer**	**Abandon**
1	12	6	6
2	11	8	3
Total	23	14	9

    c.) UCD Station peg count:
    - Count of Incoming Calls to each Station in each group
    - Count of answered calls
    - Count of other calls (abandoned, restricted, blocked)

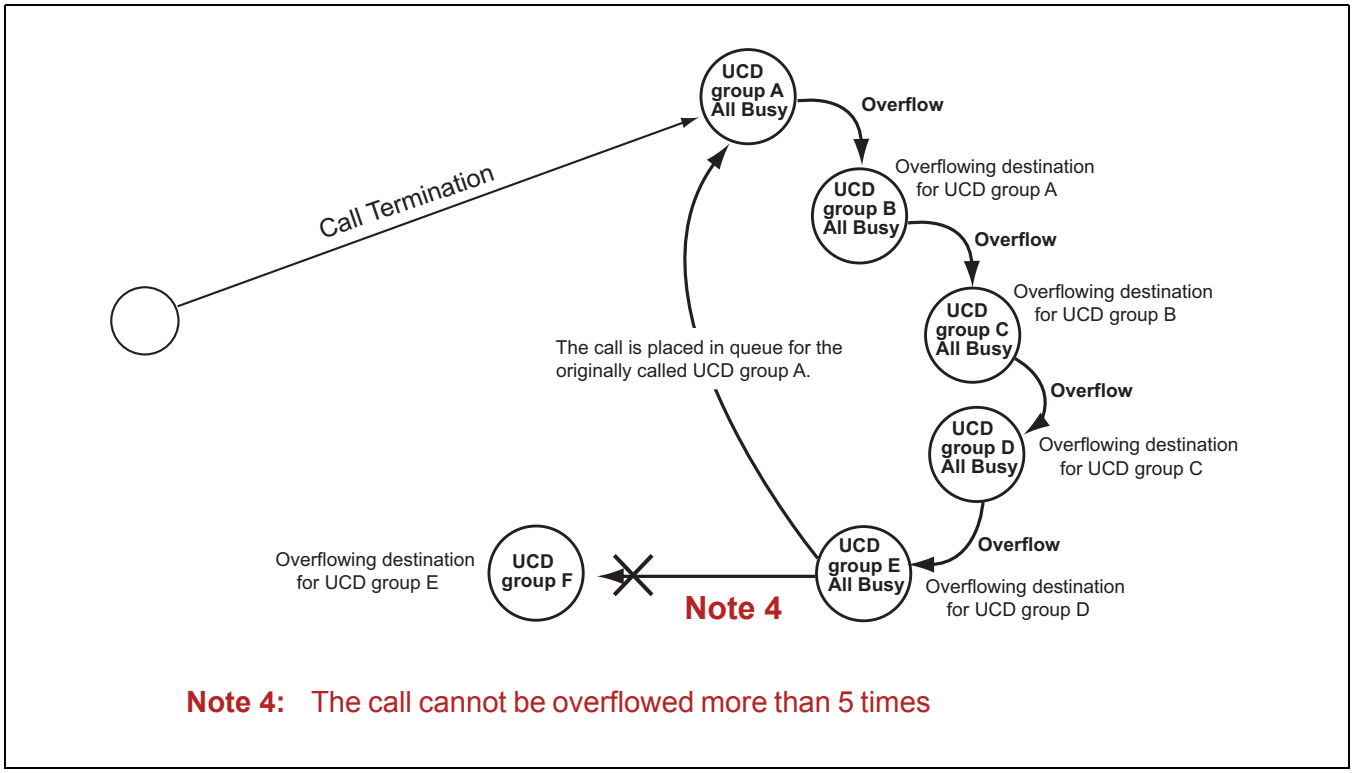
Example:

From 07/24 9:00 to 07/24 10:00					
UCD Group	**TN**	**STN**	**Ringing**	**Answer**	**Abandon**
1	1	1000	4	4	0
		1001	3	2	1
Group Total			7	6	1

25. UCD Peg counts are available for calls transferred to the UCD Group by the Attendant and for calls transferred to the UCD Group from a Station.
26. The total numbers of Incoming Calls for which the UCD queueing feature is available, and other Incoming Calls such as Intra-office, TIE Line, etc., are 384 calls per System.

    Calls above this limit are terminated by a Busy Tone Connection.
27. During queueing block, the ATTENDANT CAMP-ON WITH TONE INDICATION [A-1] is valid. The Attendant hears Reorder Tone when the ATTENDANT CAMP-ON WITH TONE INDICATION [A-1] is restricted.
28. OVERFLOW-UCD [O-13] is available.

29. When a UCD controlling Station transfers the in-progress Station Call or Trunk Call to a Station which has been set CALL FORWARDING-ALL CALLS [C-5] or CALL FORWARDING-BUSY LINE [C-2] whose forwarding destination is the UCD controlling Station, these features are valid.
30. When UCD queueing is not available and the System Data ASYD System Data 1 Index 60 Bit 3 = 1 assigned, UCD peg count is not activated.
31. When using IPPAD, UCD queueing is not available (Assign ASYD System Data 1 Index 60 Bit 3 = 1.)
32. When using IPPAD (i.e. when calling from IP station), if link block occurs such as the IPPAD is all busy, the call is queued. Even if any IPPAD becomes available, the caller remains in queuing status. When any station of the UCD group releases the call, the station is connected to the caller. When Delay Announcement-UCD is set, it is performed on connecting to the announcement. When the service is not set, it takes less than 6 seconds to perform the action.
33. When the controlling Station transfers the Station/Trunk to another Station that has set CALL FORWARDING - ALL CALLS [C-5] or CALL FORWARDING - BUSY LINE [C-2] (to the same controlling Station), Call Forwarding is activated and the MCI message is sent out.
34. Whether UCD Incoming Call information is to be sent to a Message Center Interface (MCI) or not can be specified on a UCD Group basis using the AUCD command. This feature can be assigned to each UCD Group. In the case of UCD Incoming Calls overflowing to a UCD Group, the judgement whether the information is to be sent to MCI or not depends on the terminated UCD Group.
35. When OVERFLOW - UCD [O-13] is activated, the transferred destination's data decide whether UCD Incoming Call information is sent to MCI or not.
36. The UCD Group can be programmed to the Local Data Memory (LDM). The conditions are as shown below.
    a.) Only Telephone Number and PS Number can be used when programming UCD Group to LDM.
    b.) "UCD Group assigned by ASHU command", "UCD Group assigned by ASHUL command" and "UCD Group assigned by ASHUN command" cannot be used together in the system. If assigning different kind of UCD Groups in the system, the "UCD Group assigned by ASHUN command" has top priority, and the "UCD Group assigned by ASHUL command" takes precedence over "UCD Group assigned by ASHU command".
    c.) The maximum number of UCD Groups which can be programmed in LDM per Node are 63 groups.
    d.) A UCD Group, programmed in a LDM, can accommodate 100 Stations at the maximum.
    e.) The overflowing destination for UCD Group, programmed in LDM, is UCD Group, programmed in LDM only.
    f.) Only one overflowing destination can be assigned to an UCD Group.
    g.) A call can be overflowed 4 times at the maximum by using OVERFLOW-UCD [O-13]. If overflowing destinations are all busy, the call is queued into original group (depending on System Data). See the figure below.



h.) The user can set or cancel Busy Out using UCD Stations programmed in LDM.

37. This enhancement enables a Station within UCD Group to deny hunting function for an Incoming Call originated from Station (including ATT). The service conditions are as follows:
    a.) Incoming DDD (TCL = 1)/FX (TCL = 2)/WATS (TCL = 3) Line Call are not available.
    b.) When CALL FORWARDING-DON'T ANSWER or CALL FORWARDING-BUSY LINE is set to a Station within UCD Group, a call to the Station is forwarded according to their service condition.
    c.) To use this enhancement, allow SFI 181 of a Station which denies hunting function.
38. This feature is available via FCCS.
39. A UCD Group for FCCS Service is not available for the destination of the Dead Level or the dead number.
40. The following number is assignable to a Station of the UCD Group.
    a.) Physical Station Number [Physical Station Number, Monitored Number (Pilot Number), DTE Station]
    b.) Telephone Number
41. When activating this feature via the FCCS Network, only the Telephone Number is available for FCCS Service.
42. When activating this feature via the FCCS Network, the maximum number of the UCD Groups in a Node is 63.
43. When activating this feature via the FCCS Network, the Station assigned by using the Physical Station Number and the Station assigned by using the Telephone Number cannot be accommodated in the same group and the same Node.

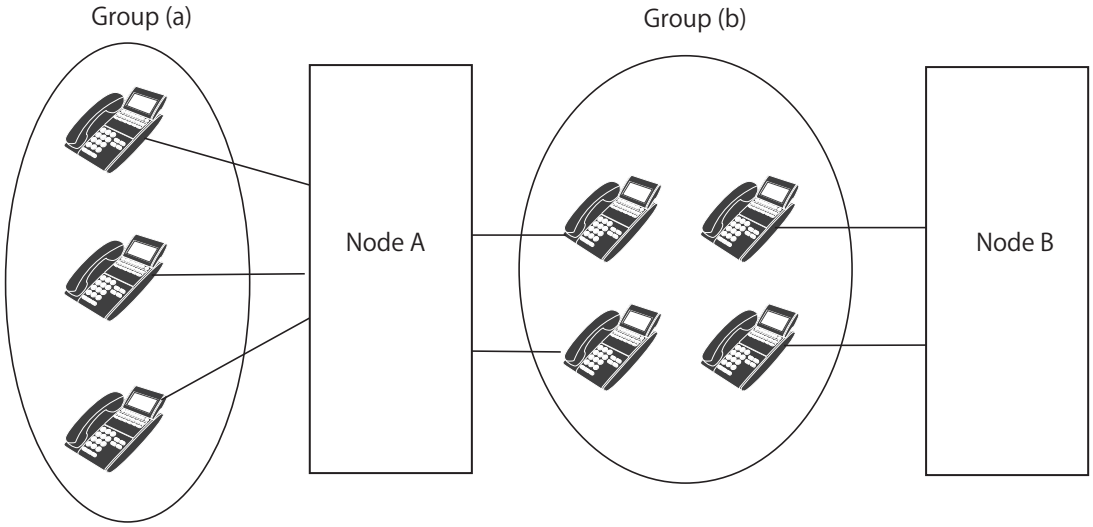
44. When activating this feature via the FCCS Network, the maximum number of Stations per group are as shown below:

    Physical Station Number: 100 Stations
    Telephone Number: 100 Stations

45. When activating this feature via the FCCS Network, two types of UCD Groups are available:

    Group (a): All the Stations belonging to one group are accommodated in the same Node.
    Group (b): Stations belonging to one group are accommodated in different Nodes.

    Group (a) and Group (b) can be used together within a FCCS Network.

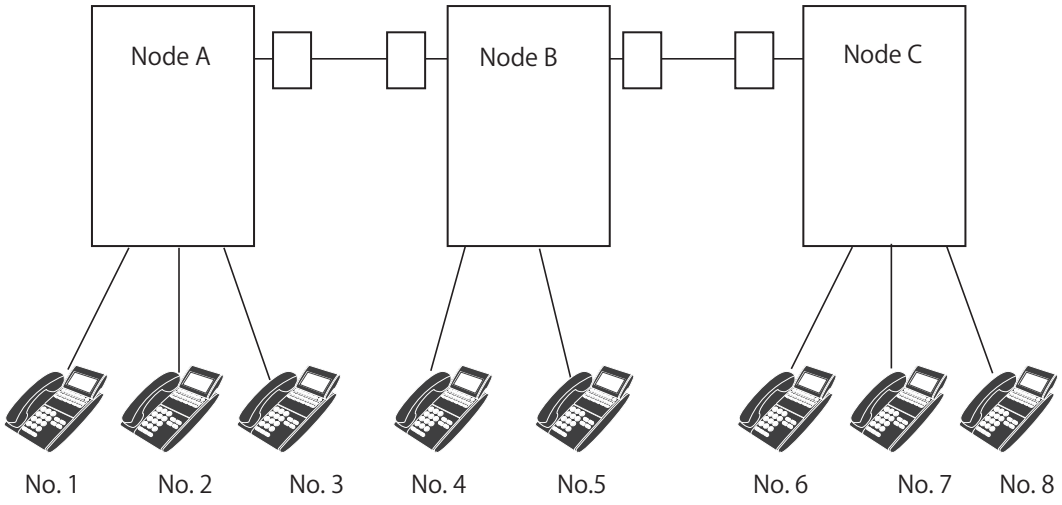


46. When using the FCH card in conjunction with FCCS:
    a.) Stations in four different Nodes can be assigned in a UCD Group.
    b.) When Stations belonging to different Nodes are assigned to a UCD Group, the Station hunt is not activated in assigned order if the Stations are not assigned in continuous order within a Node. First, the call is shifted to the Stations within the Node of the UCD controlling Station, even though it is not in consecutive order (in the Node, Station hunt is activated in assigned order). After all the Stations in the Node are hunted, the Station hunt is shifted to the next assigned

Station and the Stations in the Node are hunted in assigned order. See the figure below.

Example:

Station No.1 is the UCD controlling Station. When the call encounters busy, the call in hunted in order of No. 3, No. 4 (hunted in the Node of the controlling Station) to No. 2, No. 6, No.8 (hunted in the Node C that accommodates the next Station in assigned order) to No. 5, No .7.



Note: No.1 to No.8 indicate the order being assigned.

47. Phantom Telephone Number (PH TELN) can also be assigned to each Telephone Number (a maximum of 16 PH TELNs per TELN, TELNs + PH TELNs in total = 60,000 or less). The assigned PH TELN may or may not be the UCD Group Number by the APHNN command setting (see Interactions).
48. When a call terminates to the UCD Station through FCCS, MCI data can be output. MCI text format is designated via the ASYDL command.
49. When activating this feature via the FCCS Network, when PEG COUNT [P-7] service is provided, UCD Route Peg Count/UCD Group Peg Count/UCD Station Peg Count is available.
50. When activating this feature via the FCCS Network, in regard to CALL WAITING LAMP - UCD [C-76] service, if a UCD Incoming Call is terminated when the queue slot is full, the lamps of the Stations in the same Node that accommodates the controlling Station, start to flash.
51. When activating this feature via the FCCS Network, a UCD Station assigned by the Telephone Number cannot be a Destination Station for DIRECT-IN TERMINATION (DIT) [D-7] or NIGHT CONNECTION FIXED/FLEXIBLE [N-1/2].
52. A UCD Group may accommodate members from four different Fusion Point Codes.
53. When all FCCS links are busy:
    a.) When the FCCS links between the UCD member Station Node and the Calling Station Node (or Trunk termination Node) are all busy, the UCD Stations are not hunted and they are treated as busy Stations.

b.) When the FCCS links between the Calling Station Node and the UCD controlling Station Node are all busy, UCD queuing function cannot be provided for the Calling Station and the Calling Station hears Busy Tone.
c.) When the CCIS line is used as the alternative routing, SERVICE DISPLAY is not activated and the hunting function is completed at the terminated Station.

54. When using a Phantom Telephone Number (PH TELN) for a UCD controlling Station via FCCS, consider the following:
    a.) Whether to use the PH TELN to start the UCD hunt or not can be selected by the command APHNN. When using the PH TELN, assign "1" in the "PH" parameter. If not, assign "0".
    b.) When "PH = 1": UCD hunt starts by dialing either the Telephone Number of the controlling Station or its Phantom Telephone Number. Both numbers can be the UCD Group Number.
    c.) When "PH = 0": UCD hunt starts by dialing the Telephone Number of the controlling Station only. The feature is not provided when the PH TELN is dialed.
55. When a WLAN Handset (MH Series)/SP Controlled SIP terminal or PS is assigned as a member of UCD group, UCD queueing is not available.
56. When using Location Diversity [L-55], the Telephone Number of User Profile Data must be assigned to the station except the controlling station by using ASHUN command.

Note: Controlling stations are not available in Location Diversity [L-55] (only non-controlling stations are available).

57. When registering UCD group by using ASHUEN command, be sure not to check the "Multi Boss Secretary Group" checkbox.
58. When using listup command, the following conditions are applied.

- When listing up by LSHUEN command, the all of UCD group and Boss-Secretary group are output, and ON (Boss-Secretary group)/OFF (UCD group) is displayed per groups.
- When listing up by LSHUN command, the all of UCD group and Boss-Secretary group are output. However, ON (Boss-Secretary group)/OFF I(UCD group) is not displayed per groups.

59. Busy Out is available under the following conditions.
    a.) A station which has set Busy Out can be temporarily removed from a UCD group. Thus, any incoming calls through UCD are not terminated to the station while Busy Out is being set (the station can receive/originate non-UCD calls as usual). After canceling Busy Out, UCD incoming calls are routed to the station again.
    b.) For Desktop terminals, setting/canceling of Busy Out is only possible from My Line.
    c.) Setting/canceling of Busy Out through use of Access Code is only possible while an operating station hears DT or is engaged in a two-party call.
    d.) System initialization cancels Busy Out in progress.
    e.) Busy Out cannot be set/canceled from other stations.
    f.) Attendant Console cannot set/cancel Busy Out to a station.
    g.) Only the UCD group member stations can set/cancel Busy Out.
    h.) Busy Out cannot be set/cancel from a Desktop terminal Sub Line.
    i.) When activating this feature via the FCCS Network, Busy Out is available for the Station assigned by the Telephone Number.
60. CALL FORWARDING - DON'T ANSWER [C-3] can be set up on every station in a UCD group. On a station where CALL FORWARDING - DON'T ANSWER is enabled, an incoming call, if unanswered, is forwarded to the designated destination accordingly.

61. When a UCD group is specified as the forwarding destination of CALL FORWARDING - DON'T ANSWER [C-3], unanswered calls are handled as follows:

    - When there is at least one idle station in the UCD group, unanswered calls are distributed to idle stations in the group evenly in a circular shape.
    - If all stations in the UCD group are busy and UCD Queuing is available (ASYD, System Data 1, Index 60, Bit 3 = 0),
        - a station call is not queued to the group, and it rings the original destination station.
        - a trunk call (including a call from a TIE line) is queued to the group.

<Conditions Specific to UMGi/RNIP System>

The total numbers of Incoming Calls for which the UCD queueing feature is available, and other Incoming Calls such as Intra-office, TIE Line, etc., are 640 calls per System.

Interactions

1. UCD PEG COUNT information is available using Traffic commands. Refer to PEG COUNT [P-7] for more information.
2. Do not set ASFC, SFI 252=1 (Multi-Boss/Multi-Secretary is in service) to stations belonging to UCD groups when using the following features.
    - CALL WAITING LAMP-UCD [C-76]
    - DELAY ANNOUNCEMENT-UCD [D-31]
    - OVERFLOW-UCD [O-13]
    - PEG COUNT [P-7] - UCD PEG COUNT
3. CALL FORWARDING-ALL CALLS [C-5]: Only the controlling Station or the Attendant Console can activate this feature. When activated, calls to the UCD Group will be diverted. If the Destination Station is busy, Busy Tone is returned to the Calling Party. If the UCD Group Number is a Phantom Number, CALL FORWARDING-ALL CALLS [C-5] cannot be activated.
4. CALL FORWARDING ALL CALLS/BUSY LINE [C-2, 5]: When a member Station in a UCD Group activates CALL FORWARDING-ALL CALLS [C-5] or CALL FORWARDING-BUSY LINE [C-2], only calls directed to that Station are forwarded. UCD Group Calls are not affected. UCD Calls will bypass the Call Forwarded Station.
5. CALL FORWARDING-DON'T ANSWER [C-3]: All calls terminating at a UCD Station will forward if the call is not answered before the Call Forwarding timer expires.
6. There is no peg count for calls directed to a UCD Group which terminate to a Station Outside the UCD Group due to CALL FORWARDING [C-2, 3, 5].
7. ATTENDANT CAMP-ON WITH TONE INDICATION [A-1]: UCD Calls extended by the Attendant to the controlling Station will not Camp-On unless all of the non-controlling Stations are either busy or in the Busy Out state, and the controlling Station is engaged in a two-way conversation. Then the call will Camp-On to the controlling Station and will terminate at the first available Station.
8. In the event of a non-UCD Call extended to a member Station of a UCD Group, ATTENDANT CAMP-ON WITH TONE INDICATION [A-1] will operate normally.

9. If the UCD controlling Station answers an ATTENDANT CAMP-ON WITH TONE INDICATION [A-1] Call using CONSULTATION HOLD-ALL CALLS [C-17], there is no peg count.
10. BUSY VERIFICATION [B-3]: This feature can be used by the Attendant to check the busy/idle condition of an individual Station assigned to a UCD Group. When this feature is used on an individual Station that is idle but has set Busy Out, the Attendant will receive Reorder Tone.
11. BUSY VERIFICATION [B-3] will activate the normal hunting sequence, and the next Station in the UCD Group scheduled to receive the call will be rung. BUSY VERIFICATION [B-3] will operate only when all the Stations in the group are busy, and only if the Station is engaged in an actual call.
12. A Dterm user can set or cancel Busy Out at any time and from any condition (Idle, Busy, Three-way Conference, PRIVACY RELEASE [P-18]), simply by pressing the BUSY OUT Feature key.
13. Single Line Station users can set Busy Out by going off-hook and dialing the BUSY OUT Access Code, or from the switch hook flash, by dialing the BUSY OUT Access Code when engaged in a normal two-way conversation.
14. The DND Feature key functions in the same manner as the BUSY OUT Feature key for UCD Calls. However, while the BUSY OUT Feature key allows non-controlling UCD Stations to receive calls dialed directly, use of the DND Feature key restricts all Incoming Calls.
15. If OVERFLOW-UCD [O-13] is in service, overflow will take place if all the Stations in the first group are busy or in the Busy Out state.
16. If the OVERFLOW-UCD [O-13] Group is also busy, queuing to the first group takes place.
17. Once BUSY OUT Cancel Code is activated, any UCD Call in queue will automatically be directed to and ring the cancelling UCD Station.
18. BUSY OUT set/cancel cannot be activated by a Dterm user from the idle (on-hook) condition if the My Line is in use by another Dterm user. The Dterm user must first go off-hook or press the SPEAKER key (even though no line will actually be seized), before pressing the BUSY OUT set/cancel key.
19. If the system is reset, traffic measurement requests are no longer in service.
20. When using a Phantom Number for a UCD controlling Station, consider the following:
    a.) Whether to use the Phantom Number to start the UCD hunt or not can be selected by the command APHN. When using the Phantom Number, assign "1" in the "PH" parameter. If not, assign "0".
    b.) When "PH = 1": UCD hunt starts by dialing either the Station Number of the controlling Station or its Phantom Number Both numbers can be the UCD Group Number.
    c.) When "PH = 0": UCD hunt starts by dialing the Station Number of the controlling Station only. The feature is not provided when the Phantom Number is dialed.

Programming

STEP 1: **ASYD** - System Data 1, Index 60, Bit 3. Selection of UCD Queuing:
0/1 = UCD Queuing is provided/Not provided
System Data 1, Index 70, Bit 6. Designation of Announcement Trunk used for Delay Announcement-UCD service: 0/1 = Common/Per UCD Group

STEP 2: **ASYDL** - System Data System Data 1, Index 804, Bit 0
0/1 = BT/RBT is sent to a caller when an Attendant Console transfers the call to UCD Group that is all busy.

STEP 3: **ASHU/ASHUL** - Assign the required data for the UCD Group:

TN/UGN:	Tenant Number/User Group Number
STN/TELN:	Station Number/Telephone Number

[Station List in Group]
STN/TELN: Station Number/Telephone Number
(The order in which Stations are programmed is the order in which they will be hunted.)

Note: Do not enter Phantom Number (PHSTN/PH TELN).

STEP 4: **ANPD** - Reserve a Number Level for Feature Access. Select Normal Service (N), Hooking Service (H) or Busy Service (B) from the combo box for Connection Indexes. Assign NND in accordance with a predetermined Numbering Plan.

STEP 5: **ASPA** - Assign an Access Code to UCD BUSY OUT. SRV = "SSCA: Service Code Appendix", SIDA = 50 BUSY OUT set, SRV = "SSCA: Service Code Appendix", SIDA = 51 BUSY OUT cancel. Connection Status Indexes CI = "N: Normal" and CI = "H: Hooking".

STEP 6: **AKYD** - Assign BUSY OUT set/cancel key (same key does both; toggles function on/off). Assign Type:Function:UCD BUSY OUT [19]. Assign UCD CALL WAITING INDICATION Feature key. Assign Type:Function:CALL WAITING LAMP [47].

STEP 7: **ADSL** - Assign UCDBO Feature key; Assign SN = 0 (Idle), 3 (Called Station Busy) and FKY = 19 (UCD Busy Out).
This command is used for assigning a function to each Soft key.

STEP 8: **ATRF** - Assign Traffic Measurement Order.
TYPE: Type of Traffic Measurement
8: UCD Route Peg Count
9: UCD Group Peg Count
10: UCD Station Peg Count
INTERVAL(min):
Output Interval must be assigned in units of ten minutes (range is from 30-120 minutes), or data "0" must be assigned, which is the instruction for assigning output time.
Start TIME, End TIME:
Assign the traffic measurement time period between Start TIME and End TIME longer than an hour. Also, if the measurement should be executed throughout a day, assign the same data in both Start TIME and End TIME parameters.
Start RT, End RT:
Start/End External Route Number (available when TYPE 8 is selected)
Output TIME: This data entry is valid only when INTERVAL(min)=0.

STEP 9: **DTFD** - Display of Traffic Data
UCD Route Peg Count: DTF301
UCD Group Peg Count: DTF302
UCD Station Peg Count: DTF303

STEP 10: **AUCD/AUCDL** - Assign the related data for the UCD Group, if necessary. Assign the Tenant Number/User Group Number and the Station Number/Telephone Number of the UCD controlling Station, QTH ACT, QTH for the UCD PEG COUNT, CWT for CALL WAITING LAMP UCD [C-76], and MCI for MESSAGE CENTER INTERFACE [M-26].

STEP 11: **APHN** - Assign Phantom Number to each UCD controlling Station (if necessary).
MODE: Read Mode
Choose the data retrieval mode between the following:
1: REAL STN (Real station number)

2: PH STN (Phantom station number)
TN: Tenant Number
REAL STN: Real Station Number of the controlling Station
PH STN: Phantom Station Number **Note 5**
[PH LIST]
PH: Kind of Phantom Station Number
0 = Individual Phantom Number
1 = Phantom Number in a Hunting Group (UCD hunting to be activated)
PH STN: Phantom Station Number

Note 5: This parameter is displayed when "2: PH STN" is specified for the parameter MODE.

By using the following command, a Station within a UCD Group can deny hunting function for Incoming Call from a Station.
ASFC - Allow the service feature restriction class (SFI 181) for UCD Group Station which denies hunting function for a Incoming Call from Station.

FCCS Programming

This data setting describes only the additional or replacement commands used to activate FCCS Services. When used as an FCCS feature, these commands replace the non-network commands (for example, AAED is replaced by AAEDN). Note that the data setting for FCCS Service varies depending on the Node.

Note: When using ASYDL or ASYDN to program an FCCS feature, the basic programming of ASYD is required in addition to ASYDL or ASYDN.

STEP 1: **ASYD** - This data must be assigned to the NCN.
System Data 1, Index 60, Bit 3: 0/1 = UCD Queuing is used/not used
STEP 2: **ANPDN** - This data must be set to the NCN.
Reserve a Number Level for Feature Access.
STEP 3: **ASPAN** - This data must be set to the NCN.
Assign an Access Code of UCD BUSY OUT.
STEP 4: **ASHUN** - This data must be assigned to the NCN.
UGN: User Group Number
1st TELN: Telephone Number of UCD Controlling Station **Note 6**
TELN: Telephone Number of the UCD Station **Note 6**

Note 6: Do not enter Phantom Number (PHSTN/PH TELN).

STEP 5: **AUCDN** - This data must be assigned to the NCN, Assign the related data for UCD Group, if necessary.
UGN: User Group Number
TELN: Telephone Number of UCD Controlling Station
QTH ACT: UCD Queuing Peg Count is Invalid/Valid
QTH: UCD Queuing Peg Count Value
CWT: Threshold value of Call Waiting Lamp (1-255)

MCI: 0/1 = MCI data for the call terminated to the UCD Group can be output/cannot be output

STEP 6: **APHNN** - Assign Phantom Number to each UCD controlling Station (if necessary).

MODE: Read Mode
Choose the data retrieval mode between the following:
1: TELN (Telephone Number)
2: PH TELN (Phantom Telephone Number)

UGN: User Group Number

TELN: Telephone Number of the Circular Hunt Group

PH UGN: Phantom User Group Number **Note 7**

PH TELN: Phantom Telephone Number **Note 7**

[PH LIST]

PH: Kind of Phantom Telephone Number
0 = Individual Phantom Number
1 = Phantom Number in a Hunting Group (UCD hunting to be activated)

PH UGN: Phantom User Group Number

PH TELN: Phantom Telephone Number **Note 8**

Note 7: This parameter is displayed when "2: PH STN" is specified for the parameter MODE.

Note 8: Multiple PH TELNs (a maximum of 16) can be assigned per TELN. Select whether to use each PH TELN for UCD hunt or not by the "PH" parameter. When the BUSY OUT Feature key is assigned to the Feature key, assign the data to all the Nodes by the AKYD/ADSL command.

U-3 Universal Sender

General Description

This feature permits Dialed Numbers to be transmitted via a Sender, a device that will automatically add or delete the necessary number of digits or perform any necessary conversions.

Operating Procedure

No manual operation is required.

Service Conditions

1. For Incoming Connections, digits may be deleted according to the route involved. A maximum of four digits may be added (including the Access Code).
2. For Outgoing Connections, digits may be added/deleted according to the route involved. A maximum of 24 digits may be deleted, and a maximum of 4 may be added (including the Access Code).
3. LEAST-COST ROUTING [L-5]: Digits may be added/deleted according to the route involved and the dialed digits. A maximum of 24 digits may be deleted, and a maximum of 24 may be added, consisting of either digits or about 3 second pauses.

Interactions

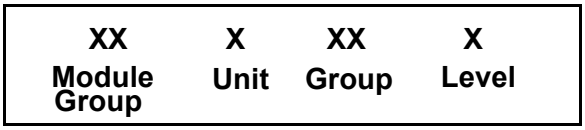
1. For features requiring digit translation, refer to the specific feature programming description.
2. The ASTP, AOSP, and AISP commands operate to translate Incoming TIE Line digits that are received by the Register and then sent to the opposite System via UNIVERSAL SENDER. The ANND command is used like the AMND command. It tells the system how many digits the Register is to expect for a specific route. The UNIVERSAL SENDER then sends the digits to the opposite System.

   Features requiring the UNIVERSAL SENDER are:

ATTENDANT CONSOLE	[A-3]
ATTENDANT KEY PAD	[A-4]
CALL FORWARDING-OUTSIDE	[C-28]
LEAST-COST ROUTING-3/6 DIGIT	[L-5]
OUTGOING TRUNK QUEUING	[O-2]
OFF-HOOK QUEUING	[O-7]
PUSHBUTTON CALLING	[P-4]
PUSHBUTTON-TO-ROTARY CONVERSION	[P-6]
SPEED CALLING	[S-3, 21, 23]
TOLL RESTRICTION-3/6-DIGIT	[T-7]

Programming

STEP 1: **ATRK** - Assign these LENS as Sender Route 905, Trunk. LENS consists of the following:



Level numbers for the Sender Trunks are 4-7.

STEP 2: **MBTK** - Assign the Make Idle status to the Sender Trunks.

V-1 Variable Timing Parameters

General Description

The Variable Timing Parameters feature gives the System the versatility to change timing parameters, using the PCPro. All timing parameters for a specific timing sequence are set initially within the generic program. These timing parameters can be changed to accommodate user requirements.

Operating Procedure

No manual operations required.

Service Conditions

The following list describes the major VARIABLE TIMING PARAMETERS:

1. Automatic Recall Timer: AUTOMATIC RECALL [A-8]:
   a.) standard timing: 30 seconds
   b.) variable timing: 2-30 seconds (two-second increments)
2. No Answer Timer for Station Call: CALL FORWARDING-DON'T ANSWER [C-3]:
   a.) standard timing: 30 seconds
   b.) variable timing: 2-30 seconds (two-second increments)
3. No Answer Timer for Station Call via an Attendant Console: CALL FORWARDING-DON'T ANSWER [C-3]:
   a.) standard timing: 10 seconds
   b.) variable timing: 2-30 seconds (two-second increments)
4. No Answer Timer for Call Transfer: CALL TRANSFER-ALL CALLS [C-11]:
   a.) standard timing: 30 seconds
   b.) variable timing: 2-30 seconds (two-second increments)
5. Periodic Time Indication Tone Timer: PERIODIC TIME INDICATION TONE [P-11]:
   a.) standard timing: first period - 3 minutes; other periods - 3 minutes
   b.) variable timing: first period - 0.5 to 8 minutes (0.5-minute increments), other periods - 0.5 to 8 minutes (0.5-minute increments)
6. Recall Timer for unanswered paging: PAGING TRANSFER [P-10] and RADIO PAGING [R-16]:
   a.) standard timing: 30 seconds
   b.) variable timing: 30 to 480 seconds (30-second increments).

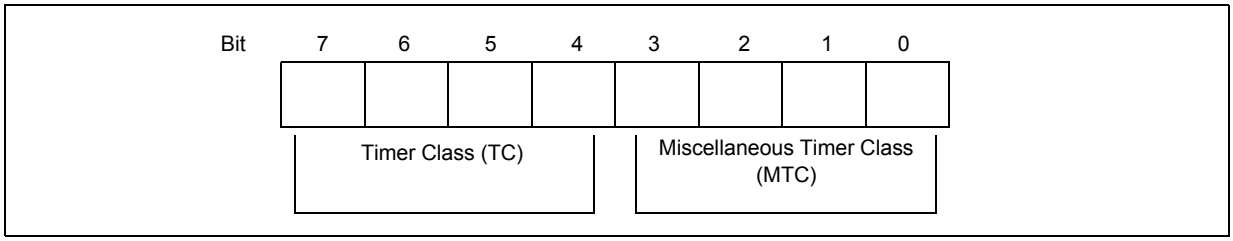
All other timers are variable (e.g., Register prepause timer, Sender prepause timer, and switch hook timer).

Note: If no data is entered, the system operates properly using Default data (RAM Data). ASYD Indexes are not usually reprogrammed, with the possible exception of System Data 1, Indexes 156 and 157. All the hexadecimal data mentioned is Default data.

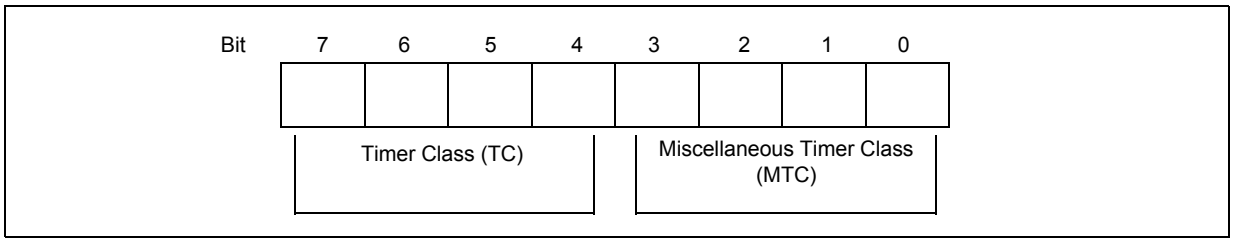
Interactions

1. To calculate VARIABLE TIMING PARAMETERS, multiply the Timer Class by the Miscellaneous Timer Class, TC x MTC = VARIABLE TIMING PARAMETER. TC is a set value. MTC is any value from 1-15, assigned in hexadecimal. For the PERIODIC TIME INDICATION TONE Timer, a TC of 3 equaling two seconds is not available, but a TC of 4 equaling 30 seconds is available.

   Timer Index:



   TC values for Indexes 136, 147, 150, 155, 156, 157, 158
   TC 0 = -
   1 = -
   2 = -
   3 = two seconds.
   TC values for Indexes 134, 139, 140, 141, 152
   TC 0 = -
   1 = -
   2 = -
   3 = two seconds
   7 = eight seconds.
   TC values for Indexes 137, 146
   TC 0 = -
   1 = 64 milliseconds
   2 = -
   3 = two seconds
   4 = 30 seconds.
   TC values for Indexes 138
   TC 0 = -
   1 = 64 milliseconds
   2 = -
   3 = two seconds
   4 = 30 seconds
   5 = five minutes
   6 = one second.
   Timer Index:



TC values for Indexes 142, 143, 144
TC 0 = -
1 = -
2 = -
3 = two seconds
4 = 30 seconds
7 = eight seconds.
TC values for Indexes 135
TC 0 = -
1 = 64 msec.
TC values for Indexes 149
TC 0 = -
1 = -
2 = -
3 = two seconds
4 = 30 seconds.
TC values for Indexes 145
TC 0 = -
1 = two seconds.

Programming

ASYD - System Data 1. All VARIABLE TIMING PARAMETERS are located between Indexes 128 and 159, with the exception of Paging.

AUTOMATIC RECALL [A-8]

System Data 1, Index 145. Assign "9F" (in hex) for 30 seconds.

CALL FORWARDING-DON'T ANSWER [C-3]

System Data 1, Index 141. Assign 35H for 10 seconds. This timer works in conjunction with Index 139, which is assigned "3F" (in hex) for 30 seconds. After ringing for 10 seconds at the first Station and 30 seconds at the CALL FORWARDING Destination Station, the call returns to the Attendant Console. CALL TRANSFER-ALL CALLS [C-11].

System Data 1, Index 140. Assign 3FH for 30 seconds.

PERIODIC TIME INDICATION TONE [P-11]

System Data 1, Indexes 137 and 138. Timer Class of data "4" for 30 seconds is available for both timers. Index 137 sets when the first tone will be heard. Assign "46" (in hex) for 3 minutes. Index 138 sets the interval between the first tone and all subsequent tones. Assign "46" (in hex) for 3 minutes. This assignment works with ARTD, CDN 29:TELP.

PAGING TRANSFER [P-10]

System Data 1, Index 74, Bits 0-3. Set the Paging No-Answer timer for Recall on Transfer. Set to "00" (in hex) for 30 seconds. (0-F + 1) x 30 = Recall Value RADIO PAGING [R-16].

System Data 1, Index 73, Bit 4. Not used. System Data 1, Index 73, Bit 5. The Recall timer is doubled, as set in System Data 1, Index 74, Bits 4-7. After the first half of the timer, recall of the radio page occurs via the System. The system grabs a Register and the Radio Page Trunk and resends the two or three-digit Answer Code. Is this operation wanted? 0/1: Yes/No.

System Data 1, Index 74, Bits 4-7. Recall timer for radio page. (N + 1) x 30 = Time Out, where N = value set in hex in Bits 4-7. If the paged party doesn't respond, the page is canceled.

V-2 Voice Call

General Description

This feature enables a Single Line Telephone to make a VOICE CALL [V-2] when the Called Party is a Desktop terminal. This feature is also available for calls between two Desktop terminals. This path exists from the Calling Party to the Called Party's built-in speaker. If the Called Party's MIC is ON, the Called Party can converse hands-free. This feature is also available when the Called Station is in the headset mode.

Operating Procedure

From a Single Line Station to a Desktop terminal

1. Dial the Telephone Number of the desired Desktop terminal and receive Ringback Tone.
2. Perform a Switchhook flash.
3. Dial the VOICE CALL Access Code. A signal tone (Two-bursts) is transmitted over the Called Party's speaker.
4. Called Desktop terminal displays:

VOICE	**XXXX**
(Time Display)	

5. Announce the call over the built-in speaker.

From one Desktop terminal to another Desktop terminal

1. The originating Desktop terminal user dials the desired Telephone Number and receives a Ringback Tone.
2. The Calling Party presses the VOICE CALL Feature key. A signal tone (Two-bursts) is transmitted over the Called Party's speaker. The Calling Party's LCD displays:

Called Number

VOICE	**XXXX**
(Time Display)	

Service Conditions

1. This feature is available when the Called Station is a Desktop terminal.

   Note: Even if the Called Station is a Desktop terminal, Voice Calls can not be made to Sub Lines and Software Lines.

2. When the Called Station is not a Desktop terminal, Reorder Tone will be sent to the Calling Party for a pre-determined time, followed by Ringback Tone.
3. Once a VOICE CALL has been activated, the Called Station will appear busy on Multi Line sets displaying the Called Station.
4. VOICE CALL will follow CALL FORWARDING - ALL CALLS [C-5] and CALL FORWARDING - BUSY LINE/DON'T ANSWER [C-2, 3].

5. VOICE CALL will terminate at the Station whose My Line is the same as the extension called.
6. Only the Called Station can respond HANDS-FREE ANSWER BACK [H-4D].
7. STATION MESSAGE DETAIL RECORDING (SMDR) [S-10] is not available for this service.
8. If the Called Station is restricted from VOICE CALL SERVICE by Class of Service, status of the Calling/Called Party is as follows:
   a.) Calling Party
   Single Line: Receives Ringback Tone after hearing Reorder Tone for 30 seconds or by pressing the switch hook during Reorder Tone.
   Desktop Terminal: When using the VOICE key: Receives Ringback Tone, and LCD displays VOICE REST for five seconds.
   When using an Access Code: Receives Ringback Tone after hearing Reorder Tone for 30 seconds or by pressing the switch hook during Reorder Tone. During the Reorder Tone, LCD displays VOICE REST.
   b.) Called Party's Desktop terminal continues to ring. **Note 1**

   Note 1: To restrict Voice Call access to a called party, refer to "To restrict Voice Call access to a called party" in Programming.

9. This feature is also available when a called party is using a headset (in the headset mode). Consider the following:
   a.) To enable a VOICE CALL to a Desktop terminal in the headset mode, additional system data setting is required: ASYDL/ASYDN, SYS1, Index1144, Bit 5=1 (Voice Call to a Desktop terminal in the headset mode).
   b.) When a called party is in the headset mode, a VOICE CALL over CCIS is not available.
   c.) When you receive a VOICE CALL with a headset on (in the headset mode), note the following:
   - A special signal tone, one or two brief beeps different from the signal tone through the speaker, comes over the headset.
   - On the DT800/DT700 Series terminal, the VOICE CALL is heard over the headset. On other series of Desktop terminals, the Voice Call is heard from the speaker.
   - In the headset mode, pressing the HEADSET Feature key switches to the handset mode, in which the VOICE CALL is provided over the speaker with the indication "VCL" or "VOICE" on the called Desktop terminal (and calling one, if available).
   - To answer the VOICE CALL, press the SPEAKER key, instead of the ANSWER key.

Interactions

A Voice Call cannot be sent to a Single Line Station.

Programming

Operating with the Access Code:

STEP 1: **ASYD** - System Data 1, Index 17, Bit 0 = 0 no Voice Call from Analog Phone/Bit 0 = 1 Voice Call from Analog Phone. Hooking (H) service for single-line set. This is used for the MESSAGE REMINDER [M-19] and VOICE CALL features of a single-line set.

System Data 1, Index 321, Bit 7 = 0 Four beeps of Voice Call Chime (**Note 2**) /Bit 7 = 1 One beep of Voice Call Chime

Note 2: A Called Party in the headset mode will hear two beeps of Voice Call Chime.

STEP 2: **ASYDL/ASYDN** - System Data 1, Index 1144, Bit 5 = 0 no Voice Call to a Desktop terminal in the headset mode/Bit 5 = 1 Voice Call to a Desktop terminal in the headset mode.

STEP 3: **ANPD** - Reserve a Number Level for Feature Access. Select Normal Service (N), Hooking Service (H) or Busy Service (B) from the combo box for Connection Indexes (applicable to any feature assigned to this Level). Assign NND in accordance with a predetermined Numbering Plan.

STEP 4: **ASPA** - Assign an Access Code to VOICE CALL, SRV = "SSCA: Service Code Appendix", SIDA = 41.

Connection Status, CI = “H: Hooking”.

STEP 5: **ASFC** - Assign a Service Feature Restriction Class that allows SFI 52 (Voice Call Using an Access Code) to Single Line Stations which will be allowed to initiate Voice Calls.

To refuse VOICE CALL:

Assign a Service Feature Restriction Class allowing SFI 93 (Voice Call Restrict) to Desktop terminals that do not allow access by VOICE CALL.

Operating with the Feature key:

STEP 1: **AKYD** - Assign the Station according to the following parameters:

TN: 1

STN: Station Number

TP: Type of Desktop Terminal (0-3)

RING:

0: No Ringing
1: Ringing All Lines
2: Ringing on Prime Line Only
3: Flexible Ring Assignment, enables RG parameter.

KYN: Key Number (1-40)

KYI: Service Index

0: Key Not Used
1: Feature Key
2: Multi Line Key

KD: Not assigned for Feature Key

FKY: Feature Key Number = 14, VOICE CALL

STEP 2: **ADSL** - Assign VOICE key; Assign SN = 2 (Originating; RBT Connection) and FKY = 14 (Voice Call).

This command is used for assigning a function to each Soft key.

FCCS Programming

This data setting describes only the additional or replacement commands used to activate FCCS Services. When used as an FCCS feature, these commands replace the non-network commands (for example, AAED is replaced by AAEDN). Note that the data setting for FCCS Service varies depending on the Node.

Note: When using ASYDL or ASYDN to program an FCCS feature, the basic programming of ASYD is required in addition to ASYDL or ASYDN.

STEP 1: **ANPDL** - Reserve a Number Level for Feature Access. Select Normal Service (N), Hooking Service (H), or Busy Service (B) from the combo box for Connection Indexes (applicable to any feature assigned to this Level). Assign NND in accordance with a predetermined Numbering Plan.

STEP 2: **ASPAL** - Assign an Access Code to VOICE CALL, SRV = “SSCA: Service Code Appendix”, SIDA = 41.
Connection Status, CI = “H: Hooking”.

V-3D Volume Control -D

General Description

This feature is used for controlling the volume of a Desktop terminal set's built-in speaker and receiver.

Operating Procedure

1. The Desktop terminal set is equipped with a sliding volume control knob located in the front, beneath the dial pad. Volume can be increased by moving the control to the right, and decreased by moving it to the left.
2. The ringing volume is controlled by a switch on the underside of the Desktop terminal. It is marked RING and located directly below the LCD Display Panel. The ringing volume is increased by moving the switch to the right and decreased by moving the switch to the left.
3. The ringing tone can be set to one of three distinct tones, using switch settings "1", "2", or "3", depending on user preference. Each switch setting produces a separate distinct ringing tone.

   Setting "1" produces the lowest.
   Setting "2" produces a medium.
   Setting "3" produces the highest.

   To control the ringing tone, there is a switch located underneath the Desktop terminal. It is opposite the TRANSFER key, and is labeled RINGING TONE. It has switch settings marked 1, 2 and 3.

For Desktop terminal

1. The Desktop terminal is equipped with two control buttons located on the top front of the phone. Volume can be increased by pushing the right control button while off-hook. Volume can be decreased by pushing the left control button while off-hook.
2. The ringing volume is also adjusted by the same two control buttons. Ringing volume by pushing the right control button while the Desktop terminal is ringing. Decrease the ringing volume by pushing the left control button while the Desktop terminal is ringing.
3. The ringing tone can be set to one of four distinct ringing tones by performing the following: Press the FEATURE key + 3. Each time this is performed, one of the four patterns is selected.

Service Conditions

Ringing, Recall Ringing, and Voice Call Signal Tone are not controlled by VOLUME CONTROL.

Programming

No programming is required.

V-19 Voice Mail Password Display Elimination

General Description

This feature eliminates Additional Digit Display on the Desktop terminal for security purposes.

Note: This feature is only available in North America.

Operating Procedure

1. Dial the station number; for example, 2000. The station number flashes.
2. When the called station answers, the station number (2000) becomes steady.
3. Dial additional digits. The display shows 2000XXXX; Xis the replacement for the authorization code.

Service Conditions

1. When the station-to-station call is established, all additional dial codes are replaced by X.
2. LAST NUMBER REDIAL and SAVE AND REPEAT features keep dialed information, but display X's when these Feature keys are pressed.
3. SPEED CALLING SYSTEM -D feature keeps dialed digits, and all digits are displayed at the same time when this Feature key is pressed.
4. This feature applies only to:
   - Station-to-Station calls (internal).
   - Station-to-Station calls over CCIS
5. This feature applies only to the Desktop terminal of the calling party. (The called party's Desktop terminal does not display X's.)
6. This function starts after the called party answers and ends by SHF (Transfer), Hang-up or Hold. If the calling party executes SHF, dialed digits are displayed (even if it comes back to the Station-to-Station call again).
7. This feature does not apply to Call Transfer or Consultation Hold to Station Call (Station/Trunk Hold) by station or Attendant, because these operations are for leaving messages. Example:
   - Station A talks to Station B.
   - Station A executes SHF and calls Station C, and is forwarded to Station D (Voice Mail).
   - Station A hangs up before or after Station D answers.
   - Station B leaves a message in Voice Mail.
8. This feature does not apply to a station call through a non-CCIS trunk.
9. This feature is allowed on a system-wide basis.

Programming

STEP 1: **ASFC** - Assign a Service Feature Restriction Class that allows SFI 23 to the Stations for this feature.

V-26 VS32 Conference Server

General Description

The VS32 is a voice server with the Multiple-Party Conference, Three-Way Calling, Announcement, and Music On Hold features. These features can be performed over the networks in conjunction with the Telephony Server. Communication cost can be reduced drastically by using these features over the IP network. The VS32 Conference Server can perform these features without any circuit card the previous system has.

Note: Term “Conference leader” used here is the party who hosts the conference, and term “Conferee” used here is the party who attends the conference upon conference leader's request.

Note: “VS32” refers to the following devices.

DEVICE TYPE	FIRMWARE VERSION
VS-32(Dual) Box [MG-VS32VA]	SP-3891 CA-VS32 PROG-A Issue 3A or later
VS-32(Dual) Card [SCA-VS32VA-B/SCA-VS32VA] **Note 3**	SP-3891 CA-VS32 PROG-A Issue 2A or earlier
Software-based VS32 **Note 1 Note 2 Note 3**	SP-4094 VS32VM PROG-A

Note 1: Available from FP95-114 V4.

Note 2: Only provides Announcement service.

Note 3: It only operates in proprietary protocol mode.

Note: When VS32 having no Voice Compression feature is used, “G.711” needs to be assigned as payload type in the data programming.
For payload type of the VS32 having Voice Compression feature, refer to the System Description.

Operating Procedure

1. Announcement

1. Lift the handset. Receive Dial Tone.
2. Dial the Announcement Service Access Code.
3. Listen to the message from the Announcement Machine.

2. IP External Music On Hold

None.

3. Music On Hold Per Line

None.

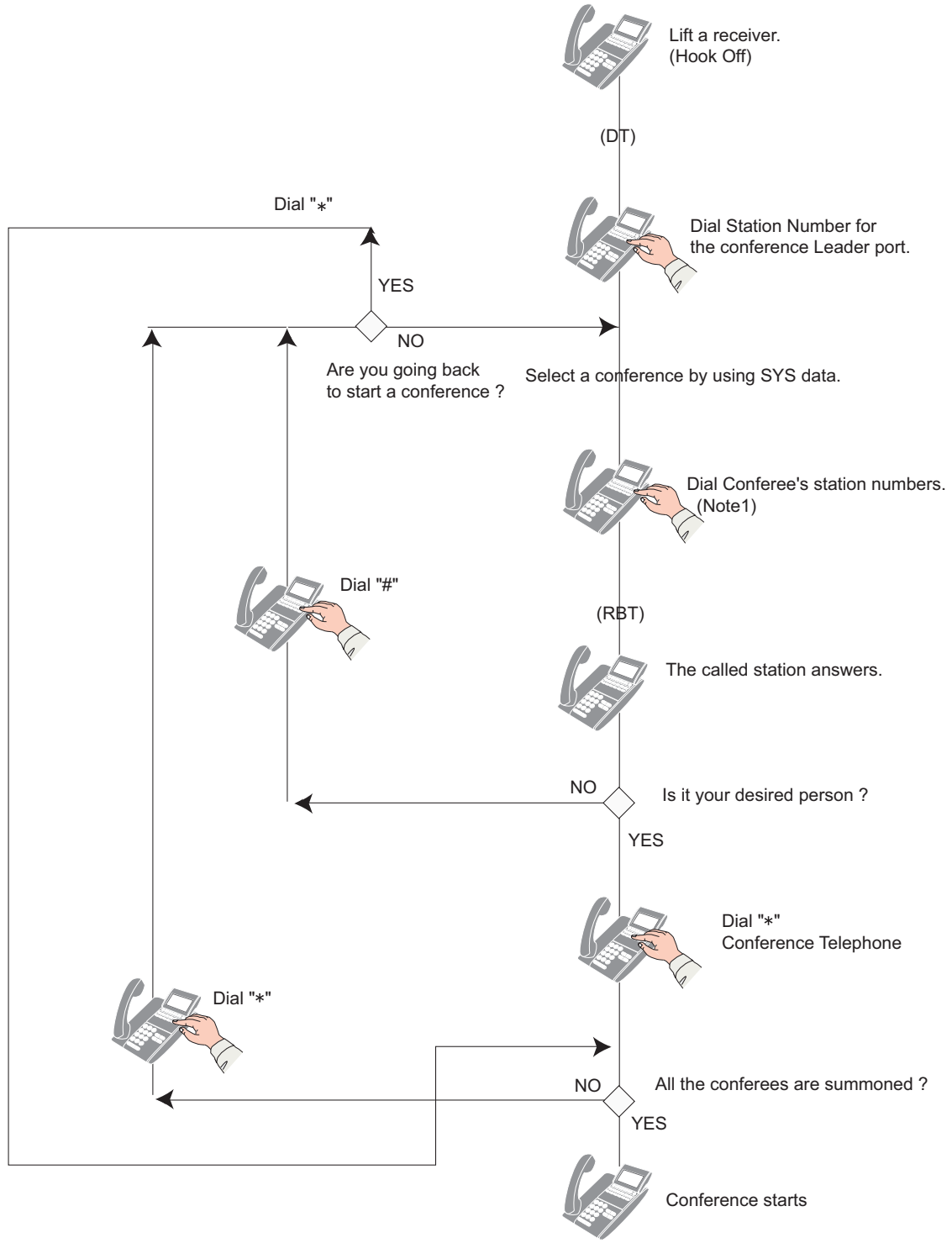
4. Multiple-Party Conference

4.1 PRESET

- Calling a conference
  1. Dial the Feature Access Code (One to Five digits) for PRESET CONFERENCE.
  2. Predesignated conferees (a maximum of 7 to 31, depending on the setup number of participant) will be called simultaneously and automatically. Receive Ringback Tone during that time.
  3. When the first conferee answers, Ringback Tone stops and the conference call mode is entered. Ringing will continue to the conferees who do not answer.
  4. The conferees who answer will enter the conference call one by one.
- Release of the conference call
  1. When one conferee who is in conference mode releases, the remaining conference leader will continue in the conference mode.
  2. When the conference leader releases from the conference mode, all the conferees will be released.

When all the conferees release, the conference leader will be released.

4.2 PROGRESSIVE



Note: Conferees is set in no-tone status and the person must wait with receiver hooked off.

4.3 MEET-ME

- Attendant-Controlled Conference
  1. Station (Conference Leader) calls Attendant to request Conference.
  2. Attendant dials the Station on Port #0 of the VS32, then presses RELEASE.
  3. Attendant sequentially calls all desired parties, presses START, dials the Station on Port #1 of the VS32, then presses RELEASE. Once all parties are connected, the Conference Leader with a Push button Telephone may dial "0" to block additional participants there by assuring privacy.
  4. If the Conference Leader dials "#", all Conferees will be disconnected. Otherwise, the conference is terminated only after all parties have hung up.
  5. To release from a Conference Circuit individually, a user must hang up. The remaining Conference may continue without interruption.
- Station-Controlled Conference
  - Procedure 1
    1. Station user wishing to establish the Conference (Conference Leader) calls each party desired and advises them of the Conference.
    2. All Stations then call the Conference Telephone Number, and the Conference Leader calls the special Conference number for the leader.
    3. Conference Connection is automatic.
    4. The Conference Leader may dial "0" to lock out additional parties.
    5. If the Conference Leader connected to the Conference Circuit dials digit "#", all Conferees will be disconnected. Otherwise, the Conference is terminated only after all parties have hung up.
    6. To release from a Conference Circuit individually, a Station user hangs up. The remaining Conferees may continue without interruption.
  - Procedure 2
    1. The Conference Leader dials the Telephone Number of conferee; receive Ringback Tone
    2. The Conferee answers.
    3. The Conference Leader makes a Switch Hook Flash (SHF).
    4. Conference Leader dials the Conference Telephone Number on the Conference port #1; Conference port automatically answers.
    5. The Conference Leader hangs up; the conferee is connected to the Conference port #1.
    6. The Conference Leader sequentially calls all desired parties (up to 7/15/31 parties) and connects them to the other Conference port (2 to 7/15/31).
    7. The Conference Leader dials the Conference Telephone Number on the Conference port #0.
    8. The Conference port automatically answers.
    9. The Conference Leader may dial "0" to lock out additional parties.
    10. If the Conference Leader connected to the Conference Circuit dials digit "#", all Conferees will be disconnected. Otherwise, the Conference is terminated only after all parties have hung up.
    11. To release from a Conference Circuit individually, a Station user hangs up. The remaining Conferees may continue without interruption.

4.4 Add-On

Refer to ADD-ON CONFERENCE - 8 PARTY [A-121].

5. Three-Way Calling

- Procedure 1
  1. Press switch hook. First party is placed on Consultation Hold. Receive Special Dial Tone.
  2. Dial the desired number. Receive answer.
  3. Announce the Conference and press switch hook again.
  4. The THREE-WAY CALLING is now established.
- Procedure 2
  1. While connected with the first party, press the hookswitch. Receive Special Dial Tone. The first party is placed on hold.
  2. Dial the Second Party (Desktop terminal). Receive Ringback Tone.
  3. The Second Party answers. The CONSULTATION HOLD - ALL CALLS state has been entered.
  4. The Second Party presses the HOLD key and the call in progress is held.
  5. Press the hookswitch and return to the Original Call.
  6. The Second Party or another Desktop terminal in the same Multi Line Group answers the Held Call.
  7. THREE-WAY CALLING is established.

Service Conditions

1. Service Conditions for VS32 (Common to Proprietary Protocol mode and SIP mode)

1. Accommodation Location Conditions and Hunt Selection Conditions are in the following:

<table>
<tr><th></th><th>ANNOUNCEMENT</th><th>IP EXTERNAL MUSIC ON HOLD</th><th>MULTIPLE-PARTY CONFERENCE</th><th>THREE-WAY CALLING</th></tr>
<tr><td rowspan="3">Accommodated Location Conditions
Note 1</td><td colspan="4">Accommodated in the same CPU</td></tr>
<tr><td colspan="2" rowspan="2">Accommodated in the same location with existing DAT Card</td><td>Accommodated in the same location with existing CFT Card</td><td>Free to assign the data except for Groups 24 to 31</td></tr>
<tr><td colspan="2">Register them in Consecutive Accommodated Locations</td></tr>
<tr><td>Payload</td><td colspan="4">For payload type/payload size available for VS32, refer to the System Description</td></tr>
</table>

Note 1: Accommodated Location can be assigned with Virtual PIR and Physical PIR.

2. Internal PHE is used in control of the same CPU with VS32 Conference Server.
3. Bandwidth Control is managed between IP terminals accommodated in MGC and VS32 Conference Server.
4. VS32 Conference Server can work with UMG type.
5. If VS32 having no VCT is in the network, because it supports only G.711, add G.711 as a CODEC used in Peer to Peer Bandwidth Control [P-75].
6. Individual trunk access with the Access Code and one assigned by PCPro are available.

7. Announcement Pattern (ANP) and Repeat Pattern (REP) positioned at Detailed Announcement Information of the AIVSL command are assigned per channel of VS32, which will determine the music source file on the channel.
8. The following Hotel features can be activated via VS32 Conference Server:

<Announcement feature>

a.) The following Hotel features can be activated via VS32 Conference Server:

- •Automatic Wake-Up [A-10/A-27]
- •Alert Service [A-57]
- •Announcement Service - Guest Station [A-141]
- •Do Not Disturb [D-11]/Do Not Disturb-D [D-11]
- •Emergency Call through PMS [E-31]
- •Group Service through PMS [G-4]
- •Group Restriction [G-21]
- •Maid Status [M-22]
- •Maid Status - Answerback [M-68]
- •Room Cutoff [R-9]

b.) Hotel Announcement features via VS32 Conference Server can be activated by single connection only. Hotel Announcement features provided by multiple connection are not available

c.) When same Announcement Equipment Number (EQP) can be set to both VS32 Conference Server (VS) and DAT card (DAT Announcement Machine) by AAEDL/AAEDN command. In this case, the destination equipment (VS32 Conference Server or DAT card) varies depending on the setting of LANG (Kind of Language) parameter.

(Example)
When checking into Guest Station with “English” from the host computer, the destination equipment is as follows:

Setting of LANG parameter (AAEDL/AAEDN)		**VS32 Conference Server**		
		Common	English	Other Language
DAT Card	Common	VS (DAT)	VS (VS)	DAT (DAT)
	English	DAT (DAT)	VS (DAT)	DAT (DAT)
	Other Language	VS (VS)	VS (VS)	-

VS: Connecting to VS32 Conference Server
DAT: Connecting to DAT card
-: None
( ): When destination is Legacy Equipment

<IP External Music On Hold feature>

a.) The following Hotel features can be activated via VS32 Conference Server:

•Automatic Wake-Up [A-10/A-27]

•Group Service through PMS [G-4]

•Maid Status [M-22]

<Three-Way Calling feature>

a.) The following Hotel feature can be activated via VS32 Conference Server:

•Overtime Call [O-9]

<Multiple-Party Conference>

a.) The following Hotel feature can be activated via VS32 Conference Server:

•Multiple-Party Conference (VS32) for Hotel [M-121]

2. Service Conditions for VS32 in Proprietary Protocol mode

1. VS32 Conference Server makes registration by concentrated PHE method. Therefore, if the PHE is not hunted due to a fault such as invalid internal PHE, VS32 Conference Server will not be able to work.
2. Internal PHE and VS32 Conference Server must be installed on the same CPU.
3. Health Check is performed between internal PHE and VS32 Conference Server. If Health Check NG occurs due to no response from VS32 Conference Server, the registration information will be deleted in MGC and VS32 Conference Server will restart.
4. Use the DISD command to confirm the status of VS32 Conference Server registration.

3. Service Conditions for VS32 in SIP mode

1. VS32 operating in SIP mode is available.
2. When the setting of ASYDL, SYS1, Index 880, Bit 1 (Internal PHI-PRI/BRI)/Bit 3 (Internal PHE) is modified, initialize the Internal PHI-BRI/Internal PHE module with the ADPM command.
3. VS32 requests registration to SP-PHI, not to Internal PHE. To accommodate VS32, activate SP-PHI referring to the data programming procedure.
4. VS32 makes registration by concentrated SP-PHI method. Therefore, when the SP-PHI is not hunted due to a fault such as invalid SP-PHI, VS32 will not be able to work.
5. Up to 4095 ports can be controlled by SP-PHI simultaneously (assigned by REGMAX to SP-PHI in ALIDL command).
6. Conditions on fault monitoring:

   - Health Check is performed between SP-PHI and VS32. If Health Check NG occurs due to no response from VS32, the registration information will be deleted in MGC and VS32 will restart.
   - SP-PHI and VS32 is connected via UDP/IP. So periodic diagnosis is activated in proprietary method. When any fault is detected in periodic diagnosis, System Message [33-S] IP PKG Warning: Health Check Error is output on the LENS of SP-PHI.

- When communication error is detected in sending control packets between Internal PHE and SP-PHI, System Message [33-R] IP PKG Warning: Sending Time-out is output on the LENS of Internal PHE. The System Message is output when control packet sending time-out occurs continuously 8 times or more.

7. Use the following maintenance command to display the operating state related to VS32:

- IPAN: to display the connection status between the Telephony Server and VS32.
  <IP terminal tab> Select "Voice Server (SIP)" under the category of SIP Multi-Function.
  <PH tab> Specify the LENS of SP-PHI. The number of registration/login on SP-PHI is displayed.

- DIPI: to display IP equipment information. Select "VS32 (SIP)" under the category of SIP Media Converter.

- DISD: to display the registration status of IP stations.
  <View Type: Station> SP-PHI's LENS is displayed on PH LENS, and PHI is displayed as PH TYPE.
  <View Type: PH Control> Specify the LENS of SP-PHI to display the registration status on SP-PHI.
  **Other type of IP equipment (MG) is not displayed.*
  <View Type: Voice Server> Specify the MAC address of VS32 to VS-ID.

- DPHL: to display the registration status on each PH, counted on the number of registration/login on SP-PHI.

4. Service Conditions for Announcement

Note: For additional conditions of the VS32 installed as a virtual machine in a server, see conditions section of Software-based VS32 in the Software Model Peripheral Equipment Description manual.

Service conditions of VS32 Conference Server Announcement Service Feature is same as those of Announcement Service. The following describes additional service conditions peculiar to VS32 Conference Server Announcement feature.

1. This feature is provided only in single connection.
2. When multiple VS32 Conference Servers are installed in a system, VS32 Conference Server sound file must be one source in the system.
3. Be sure to specify the music source. If any sound is not selected, 1kHz test tone will be set on the line.
4. Make sure that the routes of DAT circuit card and route of announcement machine have been divided.
5. Only Announcement Service defined by the AAED/AAEDL/AAEDN command is supported.
6. Multi-channel recording is not supported.

5. Service Conditions for IP External Music On Hold

1. This service is available when ASYD:SYS1 Index 64 Bit 7 = 1.
2. This feature is exclusively for IP stations.
3. The Music On Hold used in this service will be replayed from the beginning.
4. In the case of MC-MG, Hold Tone feature will be given higher priority than this service.
5. This feature is provided by single connection (a maximum of 255) only.
6. Make sure to specify the music source. If any sound is not selected, 1kHz test tone will be set on the line.

7. When using this service, if a resource for VS32 IP External Music On Hold cannot be used because VS32 Conference Server has not been registered, it cannot hunt a call, or other reason, the hold tone will depend on the ASYDL setting of SYS1 Index 810 Bit 2 = 0/1.

   - VS32 External Hold Tone cannot be used.

ASYDL		Alternative Music On Hold
SYS1, Index 810, Bit 2	SYS1, Index 818, Bit 2	
0	0	Terminal Internal Hold Tone
1	0	MGC External Hold Tone
0	1	Terminal Internal Hold Tone
1	1	MGC External Hold Tone

   - VS32 External Hold Tone can be used.

ASYDL		Alternative Music On Hold
SYS1, Index 810, Bit 2	SYS1, Index 818, Bit 2	
0	0	Terminal Internal Hold Tone
1	0	MGC External Hold Tone
0	1	VS32 IP External Music On Hold
1	1	VS32 IP External Music On Hold

8. For using this feature, it is required to register two or more channels for IP External Music On Hold.
9. For FCCS, this feature is available only in extensions belonging to the system; it is not available in extensions belonging to the other nodes.

6. Service Conditions for Music On Hold Per Line

1. This feature is available since FP95-114 V4.
2. This feature plays music when the user places a call on hold during conversation on a line (My Line or Sub Lines) configured for this feature with the ASADL or ASADN command.
3. This feature is able to send music on hold only to IP devices.
4. This feature is available on lines for Trunk Line Appearance [T-23] or Trunk Line Appearance -D [T-23D].
5. This feature is available during an outgoing call to a Phantom Number (**Note 1**) if a Telephone Number of the line being used is associated with this feature (through the ASADL or ASADN command).

   Note 1: Phantom Numbers cannot be assigned to the TELN parameter of the ASADL or ASADN command.

6. When activated, this feature always plays music on hold from the beginning.
7. This feature does not work with the following features:

   - Location Diversity [L-55]
   - Software Multiple My Line Appearance [S-143]

- Handover **Note 2**

Note 2: Handover during a PS call will switch the Music On Hold connection from a source for Music On Hold Per Line to the one for another Music On Hold feature for the system.

8. More than one channel (VS32 MOH: IP External Music On Hold) must be configured with the AIVSL command.
9. When configuring an ACD terminal for this feature, use the ASADL or ASADN command to specify sound sources to the ACD line (Virtual Line) number and the Non-ACD line (My Line) number, individually.
10. In case that this feature is not available for any reason (such as unavailability of the data for the ASADL or ASADN command), alternative Music On Hold features can be used by setting the following System Data:

ASYDL		Alternative Music On Hold
SYS1, Index 810, Bit 2	SYS1, Index 818, Bit 2	
0	0	Terminal Internal Hold Tone
1	0	MGC External Hold Tone
0	1	VS32 IP External Music On Hold
1	1	VS32 IP External Music On Hold

11. In configurations where this feature is used in conjunction with the IP External Music On Hold feature, these two features share the same resources of VS32.
12. The following settings override the Music On Hold Per Line settings:

(a) MG External Music-On-Hold [M-125] with the AMGIL command

For the MGs (Kind of MG Trunk) listed below, the setting of Music On Hold per MG (the Music On Hold Exists parameter in the AMGIL command) takes precedence over the setting of Music On Hold Per Line:

- ISDN Trunk (BRI)
- MC-MG (COT)
- ISDN Trunk (BRI) [SIP]
- MG (6COT)
- ISDN Trunk (PRI 1.5M) [SIP]
- ISDN Trunk (PRI 2M) [SIP]
- MG (6COT) [SIP]
- MG-T1 [SIP]

(b) Sound source and pattern of announcement for each channel using the AIVSL command

The setting of the ANP parameter in the AIVSL command takes precedence over the setting of the ANP parameters of the ASADL and ASADN commands.

(c) Suppression of music on hold during 8/16/32-Party Conference call with the ASFC or ASFCN command (SFI 221 = 1)

Setting the RES parameter of SFI 221 to 1 disables the Music On Hold Per Line feature during 8/16/32-Party Conference call.

13. Music On Hold Per Line uses the sound sources of VS32.
14. This feature supports the following hardware:

X: Supported, N: Not Supported, C: Conditionally Supported

Terminals/Devices	Holding Party	Held Party	Remarks
DT800 Series DT700 Series	X	X	
DT400 Series DT300 Series	X	N	Terminals directly connected to ELC
Soft Client SP350, DtermSP30	X	X	
UG50 (DLC), UG50-IPG (Digital), IPG Digital	X	X	
Analog MC, UG50-2MC, UG50-8LC	X	X	
Analog station	X	N	Terminals directly connected to LC
UG50 (LC), UG50-IPG (Analog), IPG Analog	X	X	
Standard SIP terminal (SP Controlled SIP terminal)	X	X	
Standard SIP terminal (SIP Handler Controlled SIP terminal)	X	X	
Attendant Console	N	N	
PS	N	N	
MC&MG-COT **Note 4**	N	X	
MG(BRI)	N	X	
MG(PRI) **Note 5**	N	X	
MG(SIP)	N	X	
MG-COT (6COT) **Note 6**	N	X	

Terminals/Devices	Holding Party	Held Party	Remarks
MG-T1	N	X	
COT, PRT, BRT	N	N	Trunks on cards mounted in PIR
VS32 (Multiple-Party Conference, Three-Way Calling)	N	C*	*No music on hold is played when Suppression of music on hold during 8/16/32-Party Conference call is enabled with the ASFC or ASFCN command (SFI 221 = 1).

Note 4: This includes UG50 (COT-TYPE1), UG50 (PGT-TYPE1), UG50-4LC2COTA (COT) and UG50-4LC2COTA (PGT).

Note 5: This includes UG50 (PRT), UG50-24PRIA and UG50-30PRIA.

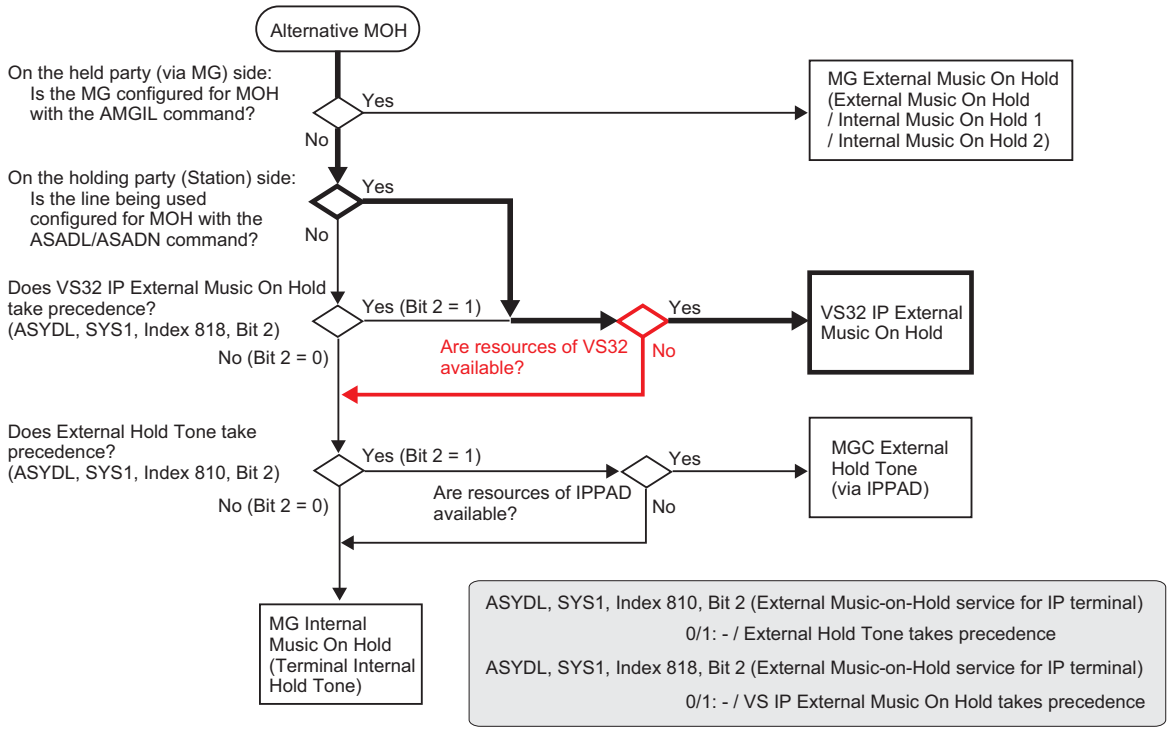
Note 6: This includes UG50 (COT-TYPE2), UG50 (PGT-TYPE2) and UG50-6COT.

15. In an FCCS system, the VS32 in the node serving the held party provides music on hold through an external connection. This requires that each node of the system have its own VS32 and all the VS32 have the same WAVE files.
16. Music On Hold Per Line works in FCCS Networking via IP (Peer-to-Peer), but not in DTI-FCCS.
17. Music On Hold Per Line is not available for calls via CCIS over IP.
18. When using multiple VS32 as a sound source, assign the same WAVE files to each of them.
19. A maximum of 249 music source files (WAVE files 001 to 249) can be specified. Furthermore, you can play the audio material on an external sound source connected to the MUSIC Connector (auxiliary input interface) of VS32. For WAVE files, there is a limit to the capacity of VS32. See “VS-32(Dual) [SCA-VS32VA/MG-VS32VA/SCA-VS32VA-B]” in Peripheral Equipment Description (IP Devices) for details.

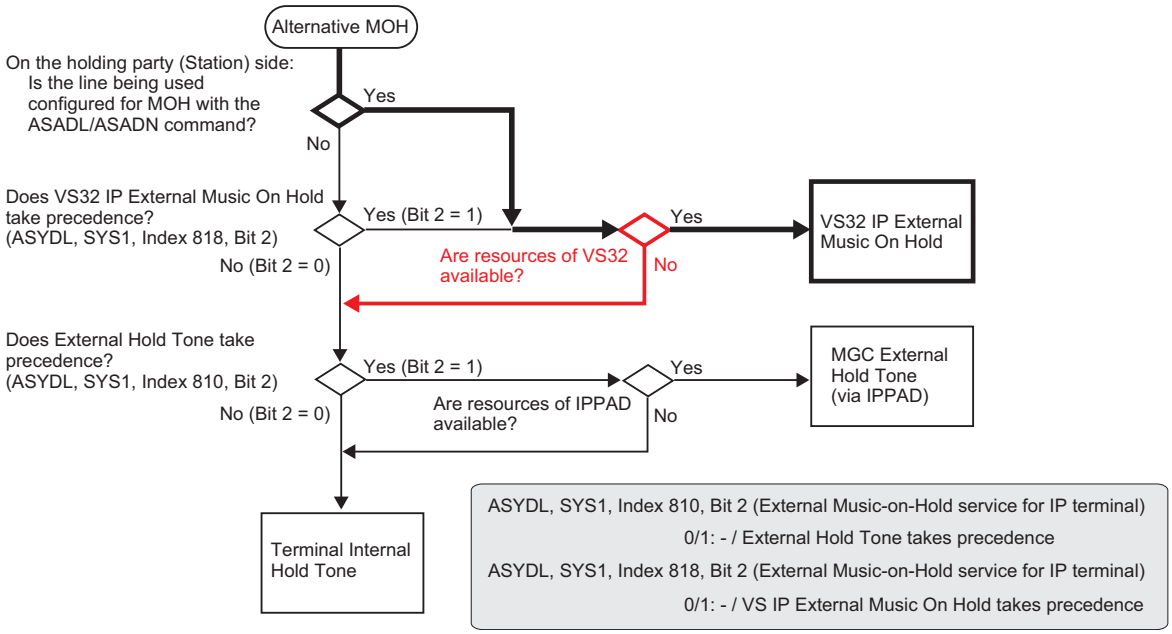
    Note: If no sound source is specified, a test tone of 1 kHz is generated.

20. If the resources of VS32 are not available (for example, because of non-registration of the VS32 or line all busy), an alternative Music On Hold feature is determined through the process shown in the following flow chart:

(a) Case where the held party is connected to an MG, such as MG(BRI), MG(PRI) and MC-MG (COT)



(b) Case where the held party is an IP station



7. Service Conditions for External Multiple Music-On-Hold Source

The following describes service conditions peculiar to External Multiple Music-On-Hold Source Feature.

1. This feature requires a port on IPPAD circuit card.
2. LENS (PIR) needs to be accommodated for SPM (Speech Path Memory). System capacity license software for more than three PIR is required.
3. Tone is not supported.
4. RSC of ARSC command cannot be used to restrict the connection to Announce equipment.
5. The following take approximately five minutes to establish connections between SPM and IPPAD, IPPAD and VS32. The External Multiple Music-On-Hold Source is unavailable until the connection has been re-established.

    - System initialization

      During this operation, the audible tone is retrieved from Tone ROM in TSW and the hold tone is retrieved from MUSIC ROM in EMA/PLO.

    - IPPAD Initialization

      Silent status during this operation.

    - Insert/remove connector into/from IPPAD

      Silent status during this operation.

    - VS32 re-registration

      Silent status during this operation.

    The connections between SPM and IPPAD, IPPAD and VS32 cannot be established under the following cases. During this operation, the audible tone is retrieved from Tone ROM in TSW and the hold tone is retrieved from MUSIC ROM in EMA/PLO.

    a.) VS32 cannot be used for some cases such as VS32 failed registration or MOH cannot be seized.
    b.) IPPAD cannot be used for some cases such as the system accommodates no IPPAD or the IPPAD is being used or busy.
6. Announcement head start is not supported.
7. In 160 and UMG types, non-IP terminals can take External Multiple Music-On-Hold Source only if they belong to the same LP with SPM
8. This feature is not supported in Multiple Music on Hold.
9. If IP External Music On Hold and this feature are assigned, IP terminal will use hold tone from IP External Music On Hold feature.
10. This feature is not available in ISDN trunks, ISDN terminals, and PSs.
11. This feature is not available in FCCS.

8. Service Conditions for Multiple-Party Conference

Service conditions of VS32 Conference Server Multiple-Party Conference Feature is same as those of Three-way Calling Feature. The following describes additional service conditions peculiar to VS32 Conference Server Multiple-Party Conference Feature.

8.1 PRESET

1. In the case of 8-party and 16-party features, CH15 and CH16 cannot be assigned as the same conference group by the AIVSL command.
2. The first port number in each group is used as Conference leader port.
3. When Extension is entered the HOLD key while participating in Multiple-Conference, the other party on the same conference can stop sounding the Hold Tone. When the Hold Tone is set to no sound (SFI 221 = 1 on the ASFC(N) command is set), the Hold Tone will not be heard. Except for extension pushed the HOLD key, the multiple-conference will be performed without any Hold Tone.
4. Cascade connection is not supported.
5. In billing information for the PRESET feature of VS32, all of the calling parties are included as Conference Leader extension by assigning SFI: 222 = 1 in the ASFC(N) command.
6. By assigning the Hotline to Extension Number of Conferee port used in Multiple-Party Conference, dial the Extension Number of Conference leader port when making a Multiple-Party Conference.
7. It is required for the Conferee to register the destination number by the AHLS command beforehand.

   Note: If the destination number is not registered, the call will be transferred to Attendant Console.

8. If Conference Leader port gets recovered (including a case that PS is out of range) in performing this feature, this feature will be terminated.
9. When Conferee is busy or out of range, the Conferee cannot be called. The Conferee will not be able to enter the conference afterward even though the conference has a vacant channel and/or the Conferee becomes available to connect to the network.
10. If the Conferee called is busy, out of range, or roaming, Conference Leader will hear ringback tone.
11. If a Conferee already participating in the conference releases the call while the other Conferees are not able to be caught for busy or out of range, the Conference Leader will be in a silent status.
12. If a Conferee on the conference releases the call before the other Conferees are called, the conference will not be made, and the Conference Leader will hear ROT.
13. Once any attended Conferee releases the conference call, Conference Leader cannot call the Conferee again to the ongoing conference call.
14. A calling cannot be made by Blind Transfer to Extension/Attendant and Call Transfer - All Calls.
15. Call-Back settings cannot be applied to Conferee port.
16. Each port on 8/16/32-party conference feature must be specified by using other Route Restriction Class and Service Feature Restriction Class than those used by General Extensions. In the case of outgoing routes, all routes should be restricted by using the ARSC command.
17. When any of Conferee answers the phone, RBT is stopped. Even though another Conferee is called, RBT will Not be heard again.
18. In the case of PS, the conference call might not work depending on channel status of CS.
19. Conference Call cannot be performed by outgoing trunk that failed to detect any answer signal.

20. It is not available to use any outgoing trunk that once had/has a fault and cannot be detected its release signal.
21. It is not available to call a conference via Central Office Line (including ISDN) or CCIS (including DP-CCIS) by using a Conferee port.
22. In FCCS, this feature is only available within Self Node. Extension Numbers belonging to other nodes and trunks cannot be used as Conferees.
23. This feature doesn't support specified two-party call with conferee.
24. This feature doesn't support forced release of conferee.
25. When VS32 is in non-registration status, if Conference Leader port is hunted, the ROT will be sent. Pilot Number such as that for Extension Hunting feature is assigned, slide action is activated.
26. Please restrict outgoing trunk whose fault cannot be recovered.
27. Calling to conferees cannot be made via external trunks such as CCIS, Q-SIG, Dp Channel Tie line, ISDN, and MG(SIP).

8.2 PROGRESSIVE

1. In the case of 8-party and 16-party features, CH15 and CH16 cannot be assigned as the same conference group by the AIVSL command.
2. The top number of ports assigned to each group is used as Conference leader port.
3. When the HOLD key is pressed on Extension during Multiple-Conference, the other participant on the same conference can stop sounding the Hold Tone. If no sound (SFI 221 = 1 in the ASFC(N) command) is specified for the Hold Tone, the Hold Tone will not be heard. Except for extension whose the HOLD key was pressed, the multiple-conference in progressive will be performed without any Hold Tone.
4. It might take four seconds at maximum to start sending a signal from Conferee Port. This action is to check ports that were once failed has been recovered and working normally.
5. In billing information for the PROGRESSIVE feature of VS32, all of the calling parties are included as Conference Leader extension by assigning SFI: 222 = 1 in the ASFC(N) command.
6. C.O. line cannot be used by Conference Leaders.
7. Conference Leader must call all of the Conferee by himself.
8. This feature is available by ISDN terminals, PSs, and ISDN trunk.
9. A group of Extension Hunting made across multiple conference trunks by Conference Leader depends on service conditions for Extension Hunting.
10. If the called party's number includes asterisk(*) or pound sign(#), the number cannot be called by another because these signs are used as the Extension numbers for inviting Conferees.
11. Any incoming call towards #1-#7 CHs, #15CH, and #31CH (Conferee Ports) is restricted.
12. Conference Leader's telephone must use PB. (DP type is not available.)
13. If called Conferee is absent at that time or the line is busy, disconnect the line by pushing a pound sign (#). In addition, if you intend to re-call Conferees because a wrong number or non-exist dialing number was called, clear already-entered numbers by pushing a pound sign (#).
    If you intend to stop calling a new Conferee and go back to the previous conference, push an asterisk (*).
    - Asterisk (*): Conference Speech
    - Pound sign (#): Calling Control
14. When Conference Leader is calling Conferees, if the Conference Leader does not make any operation, the status transition of the line takes place from Dial Tone to Busy Tone to no tone. At this time, the Conference Leader can push a pound sign (#) or asterisks (**) to go back to the previous conference speech or status for calling Conferees.

15. While Conference Leader is calling Conferees or two-party conference (a Conference Leader and a Conferee are in the conference is being made at an early stage before transition to a conference), if a call is released, the conference will end and the Conferee participating in the conference will hear Busy Tone.
16. Conference Leader pushes an asterisk (*) to cause no tone when no Conferee participated in. Pushing asterisk (*) again changes status to a mode for calling Conferees
17. While the Conference Leader is in the conference, if the call is released, the Conference Leader will not be able to enter the conference again.
18. If participants on a conference are less than three persons, the line will be disconnected forcefully to reserve conference port. As an exception here, if the Conference Leader connects CFT#0, the line will not be disconnected.
19. For connecting to trunks, any other connection cannot be made with Tie Line and C.O. line, which cannot detect any response and its recovery. (This is restricted by Office Data.)
20. When multiple VS32s exist in a place, make Extension Hunting Groups by Conference Ports.
21. The conditions for Extension Class (#0-#7CHs, #15CH, #31CH) are as follows:
    a.) Conference Leader Port (#0CH)
    ARSC Command:
    1. Assign connected restriction data for incoming calls from RSC of Conference Leader port to C.O. line.
    2. Assign connected restriction data for originating calls from C.O. line to RSC of Conference Leader port.

    b.) Conferee Port (#1-#7CHs, #15CH, #31CH)
    ARSC command
    1. Assign connected restriction data for originating calls from RSC of Conference Leader port to C.O. line.
    2. Assign connected restriction data for incoming calls from C.O. line to RSC of Conference Leader port.
22. If Conference Leader will use ISDN terminal, select a type of ISDN telephone that has PB signaling transmission feature on speech.
23. If the VS32 ports that have not made registration due to power-off is seized by call, the ROT will be sent. If a group of Station Huntings has been already assigned across VS32s, the call will be hunted by VS32 located in the group.
24. When a conferee exits the conference call, the other members may hear Reorder Tone (ROT) for a short time.
25. While a Desktop terminal is connected to one of conference ports in a conference trunk, if a DSS key where a station number for another conference port is assigned is pressed by mistake, the call is transferred to the conference port whose station number is assigned to the DSS Key. Even after the Desktop terminal is released, the connection between these conference ports is retained, and the conference trunk becomes unavailable.
    To prevent this, do not assign station numbers for conference ports on any DSS keys.

Note: When assigning station numbers for conference ports on DSS keys is required for any operational reasons, set different tenant numbers to a Desktop terminal and a conference trunk respectively, and restrict TRI=0 (station within OGTN calls to a station within TMTN) in the ATNR command for tenants where the conference ports belong.

8.3 MEET-ME

1. In the case of 8-party and 16-party features, CH15 and CH16 cannot be assigned as the same conference group by the AIVSL command.
2. The first port number in each group is used as Conference leader port.
3. When Extension is entered the HOLD key while participating in Multiple-Conference, the other party on the same conference can stop sounding the Hold Tone. If the Hold Tone is set to no sound (SFI 221 = 1 on the ASFC command is set), the Hold Tone will not be heard. Except for extension pushed the HOLD key, the multiple-conference will be performed without any Hold Tone.
4. Extension/Desktop terminal/Attendant Console/PS can be used as a Conference Leader.
5. Because Conference Leader Port (Port #0) and Conference Port (Port#1-7/#15/#31) are used as Line Circuit, Extension Number is required.
6. Connect Conference Leader to Conference Leader Port (Port #0), and Conference to Conference Port (Port#1-#7/#15/#31).
7. Conference Unit Extension Numbers should be placed in a STATION HUNTING [S-7,8,9] Group with only the pilot numbers.
8. When calling a conference, a Conference Leader transfers the Conferees to the conference port after confirming they answer a call.
9. It is available only for Conference Leader Port to perform forced release and Privacy Settings during the conference.
10. Push “#” for forced release. Push “0” for Privacy settings (all of the Conference port are made busy).
11. Privacy makes all the Conference ports busy.
12. When Conference Leader gets released after making Privacy in the conference time, the leader cannot be back to the Conference port but the Conference Leader port.
13. Privacy settings are canceled when the conference ends.
14. Only outgoing trunk that makes a response and detects release signaling can be used for the conference.
15. It is not available for Conference Leader to recover Conferees (port).
16. Even though Conference Leader is released in the conference time, or remaining participant are two or less, the conference trunk remains its connectivity.
17. Override services, such as BUSY VERIFICATION [B-3], ATTENDANT OVERRIDE [A-7], or EXECUTIVE RIGHT-OF-WAY [E-1] are denied on a Conference Call.
18. When VS32 is in non-registration status, if Conference Leader port is hunted, the ROT will be sent. If Pilot Number such as that for Station Hunting feature is assigned, slide action is activated.
19. You can disconnect the conference call by going on-hook.
20. When a conferee exits the conference call, the other members may hear Reorder Tone (ROT) for a short time.
21. While a Desktop terminal is connected to a conference trunk, if a DSS key where a station number for a conference port is assigned is pressed by mistake, the station is transferred to the conference trunk. If the Desktop terminal is released, the connection between the conference ports is retained, and the conference trunk becomes unavailable.
    To prevent this, do not assign station numbers for conference ports on any DSS keys.

Note: When assigning station numbers for conference ports on DSS keys is required for any operational reasons, set different tenant numbers to a Desktop terminal and a conference trunk, and restrict Station-to-Station Calling between tenants with the ATNR command.

8.4 Add-On

Refer to ADD-ON CONFERENCE - 8 PARTY [A-121].

9. Service Conditions for Three-Way Calling

Service conditions of Three-Way Calling Feature is same as those of the previous Three-Way Calling Feature. The following describes additional service conditions peculiar to Three-Way Conference Feature used by VS32.

1. When IP Extension assigned Three-Way Conference Feature and VS32 Conference Server assigned Three-Way Conference Feature have been registered in the same MG, VS32 Conference Server assigned feature (RT:909) will be taken: for example, conference leader has been assigned to MG7, the system searches the trunk of RT909 in the order of MG7→ MG0 → MG1 → MG2 → MG3 → MG4 → MG5 → MG6.
2. The first channels available for Three-Way Conference Feature are CH0, CH3, CH6, CH9, CH12, CH16, CH19, CH22, CH25, and CH28. Once these channels are registered, another two sequential channels from the first channel will be occupied: for example, when CH0 is occupied as the first channel of this feature, CH1 and CH2 are occupied as well.
3. Three-Way Calling feature supports 3-Party Conference (RT909) only. It does Not support 3-Party Conference Attendant (RT913).
4. Using Least Cost Routing feature, Three-Way Calling feature can be performed effectively. For more information, refer to Least Cost Routing for MG-PRI [L-54].
5. This feature can be activated via PHC (Internal PHC) under the following conditions:
   a.) Any operation other than Call Hold and Call Release cannot be performed during Three-Way Calling via PHC.
   b.) If there are not enough available ports on IPPAD when the CONFERENCE Feature key is pressed, a Three-Way Calling cannot be established. (For example, up to three ports are required when using a DtermIP.)
   c.) Additional party cannot perform switch hook flash during Three-Way Calling.
   d.) When Call Hold is released during Three-way Calling, Override tone is not provided.
6. By assigning system data (ASYDL, SYS 1, Index 830, Bit 5=1), Three-way Calling Tone (Add-On Ticker) can be sent when a Three-way Calling is established.

   Note: This system data (ASYDL, SYS 1, Index 830, Bit 5=0) works in a different way, depending on following countries:
   Austria
   : Three-Way Calling Tone (Add-On Ticker) is sent at a cycle of four seconds.
   EMEA except for Austria
   : Three-Way Calling Tone (Add-On Ticker) is sent at a cycle of 32 seconds.
   North America and Australia
   : Three-Way Calling Tone is not sent.

   The following are the service conditions of Three-way Calling Tone (Add-On Ticker).

   a.) Three-Way Calling Tone (Add-On Ticker) can be sent when a Three-Way Calling is established by the following operations.

   - Switching hook flash
   - Pressing the CONF key of Dterm/DtermIP/DT Series
   - Pressing the CONF key of Attendant Console

b.) Three-Way Calling Tone (Add-On Ticker) can be sent to all the parties of Three-Way Calling.
c.) This feature does not support the Three-way Calling created by OAI or Hotel feature.
d.) Three-Way Calling Tone (Add-On Ticker) supports the following MGs.

Product Name	F/W Product Name	Remarks
MG(BRI) [MG-2BRIA]	SP-3826 BRI MG SIP BSC PROG-A Issue 2 or later	-
MG(BRI) [SCA-2BRIA-B/ SCA-2BRIA]		
MG(PRI) (1.5M) [MG-24PRIA]	SP-3884 MG PRI PROG-A Issue 3 or later	SIP is supported for 5A or later.
MG(PRI) (1.5M) [SCA-24PRIA]		
MG(PRI) (2M) [MG-30PRIA]		
MG(PRI) (2M) [SCA-30PRIA]		
MCMG [MG-4LC2COTA]	SP-3778 MCMG PROG-A Issue 1 or later SP-3927 MCMG PROG-B Issue 1 or later **Note 1**	-
MCMG [SCA-4LC2COTA]		
MG-COT Card [SCA-6COTA]	SP-3875 6MGCOT PROG-A Issue 1 or later **Note 1**	North America is supported.
MG-COT Card [SCA-6COTB]	SP-3948 6MGCOT PROG-A Issue 1 or later	EMEA is supported.
MG-COT Card [SCA-6COTC]	SP-3948 6MGCOT PROG-A Issue 1 or later **Note 1**	North America is supported.
UG50 (PRT 1.5M)	Any version	-
UG50 (PRT 2M)		
UG50-24PRIA		
UG50-30PRIA		
UG50 (COT-TYPE1)		
UG50-4LC2COTA (COT)		
UG50 (COT-TYPE2)		
UG50-6COT		

Note 1: EMEA is not supported.

e.) Three-Way Calling Tone (Add-On Ticker) is not transmitted to the operator and terminal connected via PHC.
f.) The following table shows the specifications of Three-Way Calling Tone (Add-On Ticker).

Country	Frequencies	Cadence	Time Slot
America	425 Hz -3 dBm	80ms-on/80ms-off/80ms-on/off	DTG TS = 3
Australia	425 Hz -3 dBm	80ms-on/80ms-off/80ms-on/off	DTG TS = 3
Netherlands	425 Hz -3 dBm	10 times 32ms-on/480ms-off - repeating every 32s	DTG TS = 3
Belgium	425 Hz -3 dBm	5 times 32ms-on/480ms-off - repeating every 32s	DTG TS = 3
Sweden	1400 Hz -27 dBm	352ms-on,off-repeating every 32s	DTG TS = 3
UK	425 Hz -3 dBm	2 times 32ms-on/352ms-off - repeating every 4s	DTG TS = 3
Germany	425 Hz -3 dBm	10 times 32ms-on/480ms-off - repeating every 32s	DTG TS = 3
Austria	425 Hz -3 dBm	4 times 32ms-on/960ms-off - repeating every 4s	DTG TS = 3
Denmark	425 Hz -3 dBm	2 times 32ms-on/480ms-off - repeating every 32s	DTG TS = 3
Swiss	425 Hz -23 dBm	448ms-on - repeating every 32s	DTG TS = 3
Spain	425 Hz -3 dBm	5 times 32ms-on/480ms-off - repeating every 32s	DTG TS = 3
Greece	950 Hz -13 dBm	480ms-on, off - repeating every 32s	DTG TS = 3
Italy	425 Hz -3dBm	10 times 32ms-on/480ms-off - repeating every 32s	DTG TS = 3

7. VS32 (SIP mode) does not support RFC2833 DTMF relay.

10. Service Conditions for Least Cost Routing Feature with VS32 Conference Server

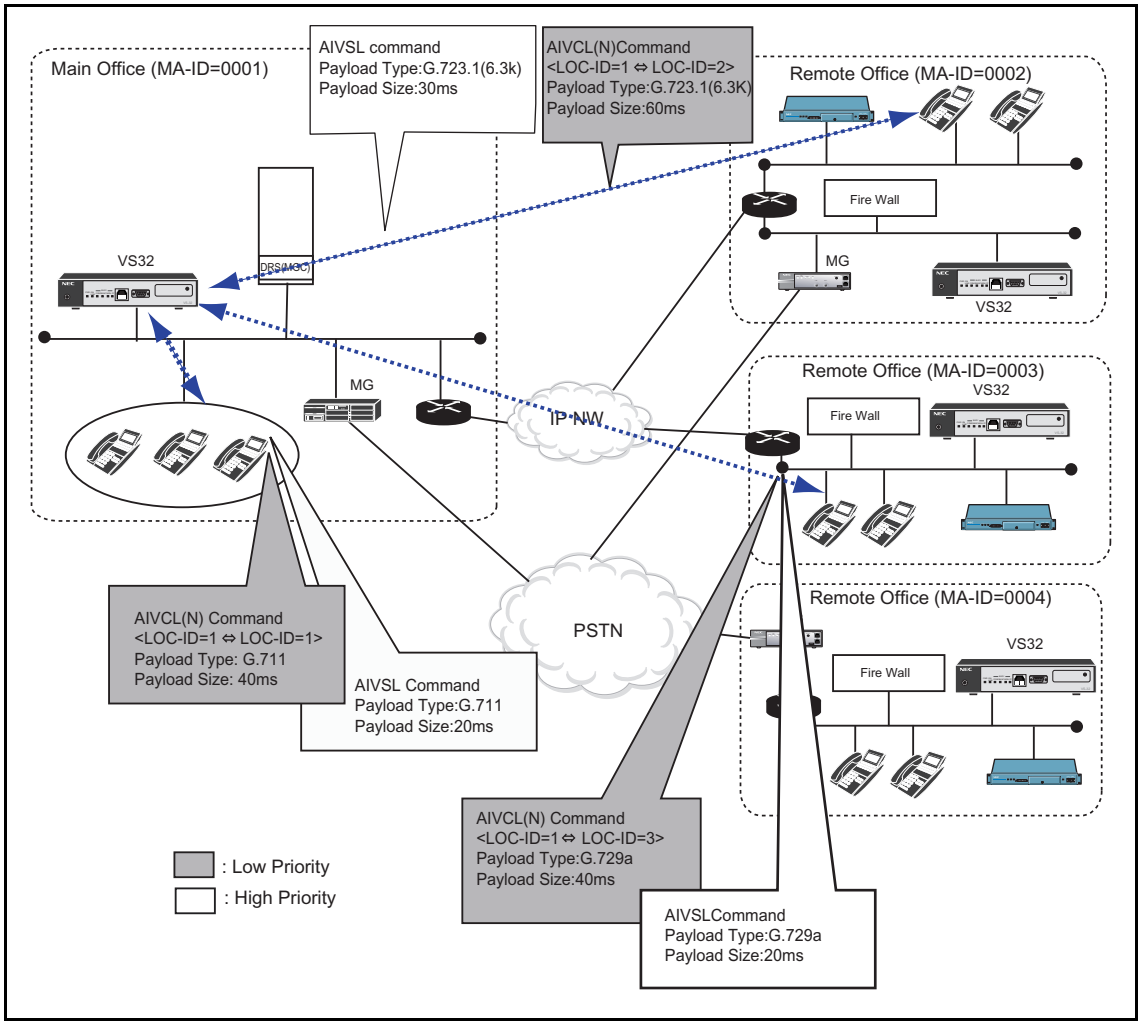
1. MA-IDs must be added to trunks related to the VS32 (such as ANNC, MOH, and 3-CONF) and any equipment to use the VS32.
2. MA-ID is fixed from 0 to 9999. If MA-ID is added to VS32, the VS32 is activated with MA-ID = 0, which means any user can free to use.
3. CFTs in TSW-BOX/MUX circuit cards need to be enabled for any user; therefore, MA-ID of the CFT must be 0.
4. Connecting to specific trunk is not available with this feature.
5. Data assignment for outgoing trunk is not necessary.
6. This feature is available in CCIS over IP, Hotel, and OAI.
7. Backup command (MEM_HDD) and Telephony Server Maintenance Menu are used to back up MA-IDs and verify the backup data. If the backup data is verified after MA-IDs has been changed by the AMAA command, verification error will occur. Therefore, it is necessary to synchronize current data with backup data. For detailed information of Office Data Backup, refer to Operations and Maintenance Manual.
8. If the MA-ID is assigned to each terminal, it can be backed up by System Data. The result of the backup data will be notified in system messages.
9. If MA-ID backup data is stored in Flash Card when system is initialized, the backup data will be loaded onto the memory.
10. MA-ID loading by the backup command (MEM_HDD) and Telephony Server Maintenance Menu will be reflected to the running system in approximately 16 minutes.
11. When all of the related VS32s are busy or not installed, DAT circuit card is not able to select any trunk.
12. In FCCS, when Three-Way Calling is performed with this feature, even if two of them are in the same location, VS32 that the last extension making the Three-Way Calling belongs to will be selected.

11. Service Conditions for Payload Change Feature

The following describes service conditions peculiar to Payload Change Feature.

1. Each command has a priority, and the payload size is determined by system in the following condition:

High Priority				Low Priority
AMGVL/AMGVN	«	AIVSL	«	AIVCL/AIVCN



2. In the case of MG-relayed connection, the payload size can be applied by the AMGVL/AMGVN command only.
3. If payload size assigned by the AIVCL/AIVCN command is below the size assigned by the AIVSL command, the payload size in the AIVSL command will be applied to the system.
4. This feature is invalid for calls via IPPAD circuit card.
5. This feature will be applied to a default payload size even though the settings are not assigned by the AIVCL/AIVCN command and the AMGVL/AMGVN command.
6. The following are recommended payload sizes on the [Detail Setting] tab:

PAYLOAD TYPE	RECOMMENDED PAYLOAD SIZE	
	3-CONF, 8/16/32-CONF	ANNC, MOH
G.711	20ms	40ms
G.723.1 (5.3K → 6.3K)	30ms	60ms
G.723.1 (6.3K → 5.3K)	30ms	60ms
G.729a	20ms	40ms

Note: In the case of VS32 that has no Voice Compression (VC) feature, set 64kbps to G.711.

Note: For more information about the command, refer to the Command Manual.

<Conditions Specific to UMGi/RNIP System>

1. VS32 Conference Server (Common to Proprietary Protocol mode and SIP mode) can work with UMGi/RNIP system.
2. In UMGi system, all the channels of one VS32 Conference Server must be accommodated in the same node (CCN/RN).

Programming

1. VS32 Conference Server Basic Data (for VS32 operating in proprietary mode)

STEP 1: **ASYDL** - Assign System Data (LDM)
SYS1, Index 825, Bit 0 = 1 and Bit 1 = 0 (indicates PAD/EC Control for IP network)
SYS1, Index 880, Bit 3 = 1 (indicates Internal PHE effective)

STEP 2: **ALIDL** - Specify the LENS (five digits) of Internal PHE.
PH LENS: Specify an even-numbered Group No. other than those 24 through 31. The LENS of can be allocated on either a Virtual PIR or a Physical PIR.
[Another sequential group following the assigned group (treated as an smaller number group) are going to be used by the Internal PHE.]
PN TYPE = Internal PHE
REGMAX: Permitted capacity of registration (1-6144)

STEP 3: **AIVSL** - Assign the VS related data
To the MAC address of VS32, assign VS feature data for each channel. **Refer to each feature setting for the assignment on BASE DATA and DETAIL SETTING tabs.*

STEP 4: **ALOCL** - Assign the IP location ID data

STEP 5: **AIVCL** - Assign the voice control data of IP service

Note: For payload type/payload size available for VS32, refer to the System Description.

Note: Path error will occur on MG-V32A, SCA-VS32A and SCA-VS32A-B when any payload type other than G.711 is assigned.

STEP 6: After data registration has been completed, reset the VS32 Conference Server by using Power Switch (OFF→ON)

2. VS32 Conference Server Basic Data (for VS32 operating in SIP mode)

Note: When switching the Proprietary Protocol mode of VS32 to SIP mode registration, make sure to delete the existing VS32 data first, and then go on to the assignment of new SIP data.

2.1 Internal PHE Data Assignment

As SIP mode of VS32 is controlled by SP-PHI, the controlling signal between TP and SP-PHI transfers via Internal PHE. The following Internal PHE data is required.

STEP 1: **ASYDL** - Assign System Data (LDM)

SYS1, Index 880, Bit 3 = 1 (Internal PHE is valid)

STEP 2: **ALOCL** - Assign the IP location ID data

Assign Location ID data of Internal PHE.

TYPE: Select "Network Address".

IP ADDRESS: Enter 0.0.0.0 for the IP Address of Internal PHE.

MASK BIT: Mask bit 32 is automatically displayed for the Network Address 0.0.0.0.

LOC-ID: 0

STEP 3: **ADTM** - Assign the TCP/IP Module data

Assign the Port Number of Internal PHE.
(60130 is used for the communication port between TP and SP in default.)

Note: Internal PHE module needs to be initialized with the ADPM command after changing location ID and port number.

STEP 4: **ALIDL** - Specify the LENS (five digits) of Internal PHE.

PH LENS: Specify an even-numbered Group No. other than those 24 through 31. The LENS can be allocated on either a Virtual PIR or a Physical PIR.

PH TYPE: Select "Internal PHE" in the combo box.

REGMAX: Capacity of registration [1-6144] **Note 1**

Note 1: Up to 4608 ports can be assigned to Virtual PIR.

2.2 SP-PHI Data Assignment

As VS32 Card requests registration to SP-PHI, MPH and SP-PHI data assignment is required.

STEP 1: **ASYDL** - Assign System Data (LDM)

SYS1, Index 880, Bit 1 = 1 (Internal PHI-PRI/BRI is available)

SYS1, Index 885, Bit 7 = 1 (SP is valid)

STEP 2: **ALOCL** - Assign the IP location ID data

Assign Location ID data of SP.

TYPE: Select "Network Address".

IP ADDRESS: Enter the IP Address of SP. **Note 1**

MASK BIT: Enter the mask bit for the IP Address.

LOC-ID: Assign any identification number.

Setting Type of Signaling Packet Service: Place a checkmark to assign the ToS for signaling packet.

IP PRECEDENCE/DIFFSERV: **Note 2**, **Note 3**

<When IP PRECEDENCE is selected>

Priority (PRECEDENCE / DSCP): Specify the precedence in the range of 0-7 (low to high).

Low Delay: Place a checkmark to set Low Delay to ON.

Throughput: Place a checkmark to set High Throughput to ON.

Reliability: Place a checkmark to set High Reliability to ON.

MIN Cost: Place a checkmark to set Cost Precedence to ON.

<When DIFFSERV is selected>

Priority (PRECEDENCE / DSCP): Specify the DiffServ Code Point in the range of 0-63.

Note 1: Internal PHE module needs to be initialized with the ADPM command after changing location ID.

Note 2: Default settings (IP PRECEDENCE, Priority = 6, Low Delay = OFF, Throughput = OFF, Reliability = OFF, Min Cost = OFF) is applied when the check box is not clicked.

Note 3: Execute the following to make the changed data effective after assigning ToS data for signaling packet.
- Internal PHE module initialization with the ADPM command or system initialization
- IPPAD card initialization (MB key ON/OFF operation or make-busy cancel by MBPM command)
- Re-registration of IP terminals

STEP 3: **ALIDL** - Assign the LAN information data (accommodated location of MPH and SP-PHI)

<MPH registration>

PH LENS — Specify an even-numbered Group No. other than those 24 through 31 (five digits). The LENS can be allocated on either a Virtual PIR or a Physical PIR. **Note 4**

PH TYPE — Select "MPH" in the combo box.

MPH ID — Identification Number of MPH [1-4095]
- Single CPU: MPH ID = 1 (Fixed)
- Dual CPU: MPH ID = 1 and 2 (Fixed)

Note: Assign MPH ID = 1 for CPU #0 and MPH ID = 2 for CPU #1.

Note: For UMGi/RNIP System MPH-ID and IP-DTG data assignment see the Initial Data section in the System Startup chapter of UMGi Configuration and Maintenance Guide and Configuration and Maintenance Guide for Remote Node over IP manuals.

<SP-PHI registration>

PH TYPE: — Select "PHI" in the combo box.

PH LENS — Line Equipment Numbers of SP-PHI (five digits) **Note 4**

REGMAX: — Capacity of registration [1-4095]

MGC PORT: — Control port number between MPH and SP-PHI [1-65535] (default: 64105) **Note 4**

TERM PORT: — Control port number in SP-PHI [1-65535] (default: 64010) **Note 4**

Note 4:
- Do not assign the same LENS to MPH and SP-PHI.
- Do not assign the same port number to MGC PORT and TERM PORT.

2.3 VS32 Data Assignment

STEP 1: **ASYDL** - Assign System Data (LDM)

SYS1, Index 825, Bit 0 = 1 (PAD/EC control for IP network is valid)

STEP 2: **AIVSL** - Assign the VS related data

VS-ID: — MAC address of VS32

Use SIP Data: — Select ON for operating VS32 in SIP mode [ON/OFF]

**Refer to each feature setting for the assignment on BASE DATA and DETAIL SETTING tabs.*

STEP 3: **ALOCL** - Assign the IP location ID data

Assign Location ID data of VS32.

TYPE: — Select "Network Address".

IP ADDRESS: Enter the IP Address of VS32. **Note 1**
MASK BIT: Enter the mask bit for the IP Address.
LOC-ID: Assign any identification number.
Setting Type of Signaling Packet Service: Place a checkmark to assign the ToS for signaling packet.
IP PRECEDENCE/DIFFSERV: **Note 2**, **Note 3**
<When IP PRECEDENCE is selected>
Priority (PRECEDENCE / DSCP): Specify the precedence in the range of 0-7 (low-high).
Low Delay: Place a checkmark to set Low Delay to ON.
Throughput: Place a checkmark to set High Throughput to ON.
Reliability: Place a checkmark to set High Reliability to ON.
MIN Cost: Place a checkmark to set Cost Precedence to ON.
<When DIFFSERV is selected>
Priority (PRECEDENCE / DSCP): Specify the DiffServ Code Point in the range of 0-63.

Note 1: Internal PHE module needs to be initialized with the ADPM command after changing location ID.

Note 2: Default settings (IP PRECEDENCE, Priority = 6, Low Delay = OFF, Throughput = OFF, Reliability = OFF, Min Cost = OFF) is applied when the check box is not clicked.

Note 3: Execute the following to make the changed data effective after assigning ToS data for signaling packet.
- Internal PHE module initialization with the ADPM command or system initialization, and VS32 initialization
- IPPAD card initialization (MB key ON/OFF operation or make-busy cancel by MBPM command)
- Re-registration of IP terminals

STEP 4: **AIVCL** - Assign the voice control data of IP service
Assign voice control data on Location ID basis. Add the payload type G.711 on the priority list for all the IP terminals and MGs using VS32 feature.

Note: Path error will occur on MG-V32A, SCA-VS32A and SCA-VS32A-B when any payload type other than G.711 is assigned.

2.4 Authentication Data Assignment

STEP 1: **ASECL** - Assign the authentication data for security
ONE TIME PASSWORD:One Time Password (OTP) to be used for authentication procedure in VS32 registration [4-10 digits with numerics, - (hyphen), . (period)] **Note 1**
AVAILABLE PERIOD:Expiration date of One Time Password [No limit/1-7 days]

Note 1: Assign the same One Time Password to VS32 by configuration command setting.

3. Data Programming for Announcement

To perform Announcement Service Feature, set the following. Note that data settings are different from the settings for FCCS.

Note: This setting is for the physical voice server. Perform the settings for the Software-based VS32 as explained in the office data settings section of Software-based VS32 in the Software Model Peripheral Equipment Description manual.

STEP 1: **ASYDL** - Assign AAEDL/AAEDN command tenant data table development
Index 800, Bit 7=0/1: Separate/Common

STEP 2: **ASTD** - Assign State Translation Data
Allow tandem connection through the COT:
STM = 0, STS = 2, ST = 1

STEP 3: **ARTD** - Assign the route class data of Announcement Trunk (VS32)
OSGS (CDN: 1) = 2 (indicates second dial tone)
ONSG (CDN: 2) = 2 (indicates PB, 60-millisecond interval)
TF (CDN: 5) = 1 (indicates outgoing trunk)
TCL (CDN: 6) = 4 (indicates Tie Line 1)
L/T (CDN: 7) = 1 (indicates Line/Trunk Identification)
LSG (CDN: 15) = 4 (indicates Loop Dial Tone [TIE Line])

Note: Make sure to divide the routes of DAT circuit card and route of announcement machine.

Note: To change the data for existing route (RT), reboot the VS32.

STEP 4: **AIVSL** - Assign the announcement trunk (VS32) data
VS-ID: Assign the MAC Address of VS32.
[BASE DATA] tab
CH: Assign the channel number of the announcement trunk (VS32).
KIND = ANNC
RT: Assign Route number
TK: Assign Trunk number
LENS: Assign Line Equipment Numbers
TN: Assign Tenant number

Note: When this data is changed for VS32 that is in fixed connection, be sure to perform re-registration of the VS32 to reflect the data change.

[DETAIL SETTING] tab
Selecting Setting of External Input Interface, assign the necessary data if other equipment is used as external announcement source.
If the payload size needs to be changed, select DETAIL SETTING tab, select Setting of Payload size, and check the box of Adjust Payload size settings of ANNC and MOH. Then, assign appropriate values in the items.

Note: Data settings will be reflected in the different ways, depending on the system. If this data is modified here while VS32 is working, the data will be reflected immediately. If some of the ports are being used, the data will be reflected to the ports when the ports are set to be idle.

STEP 5: **AAEDL** - Assign the announcement data of VS32 Conference Server

KIND = VS

TN: Assign Tenant number.

EQP: Assign Announcement Equipment Number.

Duration of Connection:

- Disconnect in 30 seconds.
- The connection is held until the extension release.

Do not Send RBT

- Not checked = Send RBT
- Checked = Do not send RBT

Send Answer Signal

- Not checked = Do not Send Answer Signal.
- Checked = Send Answer Signal

LGRT: Assign logical route number to be connected to the VS32 Conference Server (1-899).

ANP: Assign the source of announcement to be played, using announcement pattern (0-255). ANP is associated with music source file number in VS32 Conference Server.

0 - 249: Music source file 000- 249
250: Minuet (built-in)
251 - 254: Reserved
255: External Input Interface

Note: If there is no music source file number associated with ANP in VS32 Conference Server, a test music source file (test.wav/1kHz Tone) will be executed for 10 seconds.

REP: Assign Repeat pattern of the announcement (0-3).

0:Play the music source file once
1:Repeat a music source file set on the ANP parameter.
2:Play four music source files set on the ANP parameter consecutively.
(Example: 00→01→02→03 → End)
3:Play four sounds to the last, and then repeat the third and the forth by rotations.
(Example 00-01-[02-03] Repeat)
4-255: Not used.

STEP 6: **MBTK** - Release the Make Busy of the announcement trunk (VS32)

STEP 7: **ARRC** - Cancel the restriction between COT/Tie line trunk and announcement trunk (VS32)

STEP 8: **ARSC** - Cancel the restriction between the extension, announcement trunk (VS32), and connected trunk

[FCCS Data Settings]

To perform Announcement Service Feature in FCCS, take the following steps. Note that FCCS data settings are different from the data settings.

STEP 1: **ASYDN** - Assign AAEDL/AAEDN command tenant data table development
Index 800, Bit 7=0/1: Separate/Common

STEP 2: **ARTDN** - Assign the route class data of trunk to be connected to VS32 Conference Server
OSGS (CDN: 1) = 2: Second dial tone
TF: (CDN: 5) = 1: Call originating trunk (OGT)
TCL: (CDN: 6) = 4: Tie line, announcement trunk
LSG: (CDN: 15) = 4: Loop Dial Tone (TIE Line)
L/T: (CDN: 7) = 1: Line/trunk kind (Fixed to "1")

Note: Make sure to divide the route with DAT circuit card and announcement trunk.

Note: To change the data for existing route (RT), reboot the VS32.

STEP 3: **ALRTN** - Assign the logical route toward the physical route assigned in ARTD command
LGRT: Assign Logical route
FPC: Assign Fusion Point Code (FPC) of the node with trunk connected to the VS32 Conference Server
RT: Assign Physical route

STEP 4: **AAEDL/AAEDN** - Assign the announcement equipment data
KIND = VS
TN: Assign Tenant number
EQP: Assign the announcement equipment number (0-127).
Duration of Connection: Assign Connection Time
0:Disconnect in 30 seconds
1:The connection is held until the extension release
Do Not Send RBT. Check the box to enable/disable.
Send Answer Signal. Check the box to enable/disable.
LGRT: Assign Logical route number of trunk to be connected to the VS32 Conference Server (1-899).
ANP: Assign the source of announcement to be played
0 - 249: Music source file 000- 249.
250: Minuet (built-in)
251 - 254: Reserved.
255: External Input Interface
REP: Assign Repeat pattern of the announcement (0-3).
0:Play the music source file once
1:Repeat a music source file set on the ANP parameter.
2:Play four music source files set on the ANP parameter consecutively.
(Example: 00→01→02→03 → End)
3:Play four sounds to the last, and then repeat the third and the forth by rotations.
(Example 00-01-[02-03] Repeat)
4-255: Not used.

The other parameters(0,2,3) will be ignored when EQP = 64.

STEP 5: **ARRCN** - Assign Alternative Route Restriction for NDM

This command cancels the restriction between COT/Tie line trunk and VS32 Conference Server (Logical route to Logical route via the FCCS Network).

STEP 6: **ARSCN** - Assign Route Restriction Class for NDM

Cancel the restriction between extension and VS32 Conference Server.

4. Data Programming for IP External Music On Hold

To perform IP External Music On Hold Feature, take the following steps.

STEP 1: **ASYD** - Assign System Data.

SYS1, Index 64, Bit 7 = 1 (Enable the send out of hold tone)

STEP 2: **ASYDL** - Assign System Data for NDM.

SYS1, Index 800 Bit 7 0/1: Separate/Common

SYS1, Index 810, Bit 2: Specify the Bit 0/1 (-/MGC External hold tone takes precedence)(*optional) **Note 1**

SYS1, Index 818, Bit 2: Specify the Bit 0/1 (-/VS External hold tone takes precedence)

- When MGC External Hold Tone has been registered:

SYS Index 810 Bit 2	0	1	0	1
SYS Index 818 Bit 2	0	0	1	1
Hold Tone	Terminal Internal Hold Tone	MGC External Hold Tone	Terminal Internal Hold Tone	MGC External Hold Tone

- When Both MGC and VS32 Conference Server External Hold Tones have been registered:

SYS Index 810 Bit 2	0	1	0	1
SYS Index 818 Bit 2	0	0	1	1
Hold Tone	Terminal Internal Hold Tone	MGC External Hold Tone	VS32 IP External Music On Hold	VS32 IP External Music On Hold

Note 1: Instead of ASYDL command, it is available to specify this data by using **AMGIL** command. (“Music on Hold” parameter)

STEP 3: **ARTD** - Assign the Route class data of Announcement Trunk (VS32).

OSGS (CDN: 1) = 2 [indicates Second dial tone]

ONSG (CDN: 2) = 2 (indicates PB, 60 milli-second interval)

TF (CDN: 5) = 1 (indicates Call originating trunk)

TCL (CDN: 6) = 4 (indicates Tie Line1)

L/T (CDN: 7) = 1 (indicates Trunk)

LSG (CDN: 15) = 5 (indicates E&M)

Note: Make sure to divide the route with DAT circuit card and announcement machine.

Note: To change the data for existing route (RT), reboot the VS32.

STEP 4: **ARTI** - Assign application data for Announcement Trunk.
CDN79: FTRKS (Least Cost Routing)
0: VS32 is selectable. **Note 2**
1: Voice quality is taken as top priority. **Note 3, Note 5**
2: An area to which an operator belongs is given top priority. **Note 4, Note 5, Note 6**
3: Not used (same as 0)

Note 2: VS32 (ANNC and MOH) will be selected by rotation.

Note 3: Transmission of voice packets towards network can be restricted to minimum.

Note 4: Restricting VS32 use from other MA-IDs, the use of VS32 can be limited per MA-ID.

Note 5: If there is no available trunk, select a VS32 whose MA-ID is set to 0. Other than appropriate MA-IDs and MA-ID = 0 cannot be connected, therefore, VS32 is not selected by rotation. Also, to perform this feature, MA-ID must be assigned in advance to all the terminals/trunks connected to the system.

Note 6: Only RT of IP External Music On Hold feature is available. Setting it to RT of Announcement trunk enables the system to operate the same as one when“1” is defined.

Note: To change the data for existing route (RT), reboot the VS32.

STEP 5: **AIVSL** - Assign the announcement trunk (VS32) data.
VS-ID: Assign the MAC Address of VS32.
[BASE DATA] tab
CH: Assign the channel number of the announcement trunk (VS32).
KIND = MOH
RT: Assign Route number.
TK: Assign Trunk number.
LENS: Assign Line Equipment Numbers.
TN: Assign Tenant number.
Detailed Settings for Announcement Data: If Music Source and the way of Playback need to be specified, assign the appropriate ones.
[DETAIL SETTING] tab
Selecting Setting of External Input Interface, assign the necessary data if BGM equipment is used as external music source.
If the payload size needs to be changed, select DETAIL SETTING tab, select Setting of Payload size, and check the box of Adjust Payload size settings of ANNC and MOH. Then, assign appropriate values in the items.

Note: Data settings will be reflected in the different ways, depending on the system. If this data is modified here while VS32 is working, the data will be reflected immediately. If some of the ports are being used, the data will be reflected to the ports when the ports are set to be idle.

STEP 6: **MBTK** - Release the Make Busy of the announcement trunk (VS32).

STEP 7: **AAEDL** - Assign the announcement data of VS32 Conference Server.

KIND = VS

TN: Assign Tenant number.

EQP = 64 (IP External Music On Hold)

LGRT: Logical route number of trunk to be connected to the VS32 Conference Server (1-899)

ANP: Assign an Announcement Pattern (sound source for announcement message). **Note 7**

0-249: Music source files 000-249

250: Minuet (built-in)

251-254: Reserved

255: External Input Interface

REP = 1(Repeats the audio material in the source specified by the ANP parameter.) **Note 8**

Note 7: ANP numbers are associated with music source file numbers stored in VS32. For more information, refer to Peripheral Equipment Description (IP Devices).

Note 8: The other parameters(0,2,3) will be ignored.

Note: In Detailed information for Announcement Data on AIVSL command, if ANP and REP are assigned per channel on VS32, even ANP and REP on AAEDL command have been specified, settings on AIVSL command is applied as higher priority.

5. Data Programming for Music On Hold Per Line

To perform the Music On Hold Per Line feature (**Note 1**), use the following steps.

Note 1: Available since FP95-114 V4.

STEP 1: **ASYD** - Assign System Data.

SYS1, Index 64, Bit 7 = 1 (Enables Music On Hold.)

STEP 2: **ASYDL** - Assign System Data for LDM.

SYS1, Index 800, Bit 7: Tenant data loading into Local Data Memory (LDM) for the AAEDL command

0/1: Separate per tenant / Common

SYS1, Index 810, Bit 2: External Music-on-Hold service for IP terminal **Note 2**

0/1: - / MGC External Hold Tone takes precedence (*optional)

SYS1, Index 818, Bit 2: External Music-on-Hold service for IP terminal **Note 3**

0/1: - / VS IP External Music On Hold takes precedence

Note 2: The System Data for the settings of "External Music-on-Hold service for IP terminal" (ASYDL, SYS1, Index 810, Bit 2; ASYDL, SYS1, Index 818, Bit 2) together determine an alternative Music On Hold feature in the event of unavailability of Music On Hold Per Line. The System Data (ASYDL, SYS1, Index 810) is also used to select an alternative Music On Hold feature when Music On Hold Per Line is technically available but does not actually work due to link block (resource capacity shortage) on VS32.

Note 3: The System Data for the settings of "External Music-on-Hold service for IP terminal" (ASYDL, SYS1, Index 810, Bit 2; ASYDL, SYS1, Index 818, Bit 2) together determine an alternative Music On Hold feature in the event of unavailability of Music On Hold Per Line.

STEP 3: **ARTD** - Assign the Route class data of Announcement Trunk (VS32).
OSGS (CDN: 1) = 2 (Second Dial Tone)
ONSG (CDN: 2) = 2 (PB, 60 millisecond interval or No.7 CCIS)
TF (CDN: 5) = 3 (Both Way Trunk (BWT))
TCL (CDN: 6) = 4 (Tie Line / Announcement Trunk)
L/T (CDN: 7) = 1 (Trunk)
LSG (CDN: 15) = 5 (E&M)
ANS (CDN: 28) = 1 (Answer signal is given.)

Note: Reboot the VS32 after making changes to the data for an existing route (RT).

STEP 4: **ARTI** - Assign application data for Announcement Trunk. **Note 4**
CDN79: FTRKS (Flexible Trunk select)
0: Arbitrary selection. VS32 is selectable. **Note 5**
1: Voice quality priority selection. Voice quality is taken as top priority. An area to which the held party belongs is preferentially selected. **Note 6, Note 8**
2: Operation person priority selection. An area to which the holding party belongs is given top priority. **Note 7, Note 8, Note 9**
3: Not used (same as 0).

Note 4: Reboot the VS32 after making changes to the data for an existing route (RT).

Note 5: VS32 (ANNC and MOH) will be selected by rotation.

Note 6: Transmission of voice packets towards network can be restricted to minimum.

Note 7: This option restricts access to VS32 from other areas to limit the usage of VS32 per MA-ID.

Note 8: If there is no available trunk, a VS32 with MA-ID = 0 is selected. In this option, either the VS32 with MA-ID = 0 or the one in the area where the held party is located is selectable; therefore, VS32 is not selected by rotation. This option requires MA-IDs to be assigned in advance to all the terminals/trunks connected to the system.

Note 9: Only RT for IP External Music On Hold feature is available. Setting it to RT of Announcement trunk enables the system to operate in the same way as when 1 is assigned.

STEP 5: **AIVSL** - Assign trunk (VS32) data for VS32 IP External Music On Hold.

Note: Changes to the settings for an online VS32 take effect immediately without the need for reregistration. In the case of a busy channel, changes take effect immediately after it becomes idle.

VS-ID: Assign the MAC Address of the VS32.

[BASE DATA] tab

CH: Assign the channel number of the VS32.

KIND = MOH

TN: Assign Tenant number.

LENS: Assign Line Equipment Numbers.

RT: Assign Route number

TK: Assign Trunk number

Setting of announcement detail data: **Note 10**

If you want to specify an announcement source and a playback pattern for each channel, click this button and enter the data.

Note 10: Do not configure this setting for a channel to play music on hold.

[DETAIL SETTING] tab

Setting of External Input Interface:

When you connect an external BGM source to the MUSIC Connector (external input interface), you can adjust the input level. To do this, click this button. In the Setting of External Input Interface dialog box, place a check mark in the "Adjust External Input Interface setting" check box, and then select a gain pad value from the list.

Setting of Payload size: **Note 11**

Specify a payload size for each payload type.

To do this, click this button. In the Setting of Payload size dialog box, place a check mark in the "Adjust Payload size setting of ANNC and MOH" check box, and then select a payload size from the list.

For details, see "11. Service Conditions for Payload Change Feature."

Note 11: Path error will occur on MG-V32A, SCA-VS32A and SCA-VS32A-B when any payload type other than G.711 is assigned.

STEP 6: **MBTK** - Release the Make Busy of the trunk for VS32 IP External Music On Hold.

STEP 7: **AAEDL** - Assign the data of VS32 IP External Music On Hold. **Note 12**

Equipment Kind = VS

TN: Assign Tenant number.

EQP = 64 (External Music on hold of IP)

Duration of Connection = "The connection is held until the station release."

Do not Send RBT: Select this check box.

Send Answer Signal: Select this check box.
LGRT: Logical route number of the trunk of VS32 for Music On Hold (1-899)
ANP: Assign an Announcement Pattern (sound source for announcement message). **Note 13**
0-249: WAVE files 000-249
250: Minuet (built-in)
251-254: (Reserved.)
255: External Input Interface
REP = 1(Repeats the audio material in the source specified by the ANP parameter.)

Note 12: Specify a sound source for music on hold for a tenant in the system.

Note 13: ANP numbers are associated with music source file numbers stored in VS32. For more information, refer to Peripheral Equipment Description (IP Devices).

STEP 8: **ASADL/ASADN** - Assign the data of Station's ANP Data for LDM/NDM.
Define music on hold for each station number (Telephone Number).
UGN: Assign User Group Number (1-255).
TELN: Assign Telephone Number (with a maximum of 16 digits).
ANP: Assign an Announcement Pattern (1-255).
Note 14, **Note 15**
1-249: WAVE files 001-249
250: Minuet (built-in)
251-254: (Reserved.)
255: External Input Interface

Note 14: Specify a sound source for Music On Hold Per Line.

Note 15: Do not set this parameter to zero.

6. Data Programming for External Music-On-Hold Source

To perform the External Multiple Music-On-Hold Source feature, use the following steps.

STEP 1: **ASYD** - Assign System Data.
SYS1, Index 60, Bit 4 = 1 (Enable Fixed Connection)
SYS1, Index 64, Bit 7 = 1 (Enable the send out of hold tone)

STEP 2: **ASYDL** - Assign System Data for NDM.
SYS1, Index 646, Bit 0
0/1 = MUSIC ROM/Speech Path Memory (External Music on Hold)

Note: The service is currently provided only for CH00.

STEP 3: **ARTD** - Assign the Route class data of Announcement Trunk (VS32).

Note: To change the data for existing route (RT), reboot the VS32.

The following is an example.

RT:1

1-OSGS	:2	2-ONSG	:0	3-ISGS	:0	4-INSG	:0	5-TF	:0
6-TCL	:4	7-L/T	:1	8-RLP	:0	9-TQ	:0	10-SMDR	:0
11-TD	:0	12-DR	:0	13-AC	:0	14-TNT	:0	15-LSG	:0
16-SMDR2	:0	17-H/M	:0	18-MC	:0	19-EANI	:0	20-D	:0
21-MSB	:0	22-MSW	:0	23-TR	:0	24-OC	:0	25-R/L	:0
26-RSVD	:0	27-TL	:0	28-ANS	:0	29-TELP	:0	30-PAD	:0
31-OGRL	:0	32-ICRL	:0	33-HD	:0	34-GUARD	:0	31-WINK	:0
36-VAD	:0	37-CLD	:0	38-FA	:0	39-BC	:0	40-TCM	:0
41-TDMQ	:0	42-TPQ	:0	43-BT	:0	44-PRV	:0	45-A/D	:0
46-CW	:0	47-TPQ	:0	48-BL	:0	49-TRKS	:0	50-DPLY	:0
51-ACD	:0	52-2W/4W	:0	53-FAAT	:0	54-GW	:0		

STEP 4: **AIVSL** - Assign the MOH trunk (VS32) data.
VS-ID: Assign the MAC Address of VS32.
[BASE DATA] tab
VS-ID: Assign the MAC Address of the MOH trunk (VS32).
CH: Assign the channel number of the MOH trunk (VS32).
KIND = MOH
RT: Assign Route number
TK: Assign Trunk number
LENS: Assign Line Equipment Numbers
TN: Assign Tenant number
[DETAIL SETTING] tab
Selecting Setting of External Input Interface, assign the necessary data if BGM equipment is used as external music source.
If the payload size needs to be changed, select DETAIL SETTING tab, select Setting of Payload size, and check the box of Adjust Payload size settings of ANNC and MOH. Then, assign appropriate values in the items.

Note: Data settings will be reflected in the different ways, depending on the system. If this data is modified here while VS32 is working, the data will be reflected immediately. If some of the ports are being used, the data will be reflected to the ports when the ports are set to be idle.

STEP 5: **MBTK** - Release the Make Busy of the announcement trunk (VS32).

STEP 6: **AFCD** - Assign data for fixed connection between VS32 MOH and SPM.
VS-32 MOH = LENS-A
SPM = LENS-B

7. Data Programming for Multiple-Party Conference

7.1 PRESET

The following are procedures for setting the PRESET feature. Note that the procedure from 6 is separated into two ways, depending on the Conferee Settings with Extension Number or Outgoing Trunk.

STEP 1: **ASYD** - Assign System Data, System Data 1, Index 20, Bit 6 = 0 (Line Circuit Reverse Relay Control when Reorder Tone is connected to a station.)

STEP 2: **ANPD** - Determine the Extension Number Digit.
Specify the digits for the Extension Number of Conference Port.

STEP 3: **ASPA** - Assign the Access Code.
Specify the Access Code for the Extension Number of Conference Port (SRV =”STN: Station Number”).

STEP 4: **AIVSL** - Assign VS32 Conference Server Related Data.
VS-ID: Assign the MAC Address of the VS32.

In the case of 8-party conference, assign the extension number to the VS32 Conference Server channel listed below:
CH#0 (Conference leader port) and CH#1 through #7 (Conferee port).
CH#8 (Conference leader port) and CH#9 through #15 (Conferee port).
CH#16 (Conference leader port) and CH#17 through #23 (Conferee port).
CH#24 (Conference leader port) and CH#25 through #31 (Conferee port).

In the case of 16-party conference, assign the extension number to the VS32 Conference Server channel listed below.
CH#0 (Conference leader port) and CH#1 through #15 (Conferee port).
CH#16 (Conference leader port) and CH#17 through #31 (Conferee port).

In the case of 32-party conference, assign the extension number to the VS32 Conference Server channel listed below:
CH#0 (Conference leader port) and CH#1 through #31 (Conferee port).

[BASE DATA] tab
CH: Assign the VS32 Conference Server channel number.
KIND: Choose 8-party conference, 16-party conference, or 32-party conference.
CONF KIND = PRESET
TN: Tenant number
STN: Extension number
LENS: Line equipment numbers
TEC: Assign “2 (indicates PB)” to the conference leader port, and assign “14 (indicates Hot Line)” to the conferee ports.
RSC: Assign Route Restriction Class other than ordinary extensions use.
SFC: Assign Service Feature Restriction Class other than ordinary extensions use.

[DETAIL SETTING] tab

These values has been specified as proper default values for the system. Note that the values should not be changed if the system works normally. If the system fails to work, click Setting of 8/16/32-CONF and change the data such as PAD value for 8/16/32-party conference settings.

To determine conference trunk operations, click Setting of Expansion Mode and assign the operation mode for the conference trunk.

To reflect these data on VS32 Conference Server running, re-registration must be performed in the system.

Note: Data settings will be reflected in the different ways, depending on the system. If this data is modified here while VS32 is working, the data will be reflected immediately. If some of the ports are being used, the data will be reflected to the ports when the ports are set to be idle.

For the configuration of the FCCS network, go to the [FCCS Data Settings] section; otherwise continue with the next step.

STEP 5: **ASFC** - Specify the information on Service Feature Restriction for stations.

Enter a Service Feature Index (SFI) and then specify Restriction Data (RES) for the Service Feature Restriction Class (SFC) of the stations.

SFI 32: RES = 1 (Line Circuit Reverse Relay Control is available.)

SFI 50: RES = 1 [Off-Premise Extension (Long Line Telephone - No Howler, No PAD Control)]

SFI 71: RES = 1 (Distinctive Dial Tone) **Note 1**

SFI 72: RES = 1 (Called Station Switch Hook Flash is not available.)

SFI 80: RES = 1 (8-Party Conference Terminal)

SFI 221: RES = 1 (Stop the hold music during 8/16/32-Party Conference call)

Note 1: This setting is required for the Conferee ports.

- Other Service (0/1 = None/Available)
  Specify RES for the following SFI if necessary. Assign 0 to RES for SFIs other than those mentioned below.

SFI 12: RES = 1 (Speed Calling System)

SFI 14: RES = 1 (System Message Detail Recording - Trunk Basis)

SFI 19: RES = 1 [Individual Speed Calling (Speed Calling - Station/Group)]

SFI 27: RES = 1 (Account Codes/Authorization Codes/Forced Account Codes)

SFI 58: RES = 1 (Station-to-Station Station Message Detail Recording (SMDR) Call)

- When an Extension Number is specified as the Conferee:

STEP 6: **ARSC** - Specify the information on route restriction for the Route Restriction Class (RSC) of the conference leader port and conferee ports.

Restrict the conference leader port and conferee port for COT and Tie line routes.

Note: Specify restriction data “0” for RRI-0 and RRI-1 of the target RSC to restrict the incoming connection.

STEP 7: **AHLS** - Assign Route Restriction Class.
Specify a destination number to the entry of the physical station data created with the AIVSL command.

HOT TN: Hotline Tenant Number

HOT STN: Hotline Station Number

TYPE = 1 (indicates Station Number)

CON TN: Destination Tenant Number

CON STN: Destination Station Number

- When an Outgoing Trunk is specified as the Conferee:

STEP 6: **ARSC** - Specify the information on route restriction for the Route Restriction Class (RSC) of the conference leader port and conferee ports.

Note: Specify restriction data "0" for RRI-0 and RRI-1 of the target RSC to restrict incoming connection. Also specify restriction data "1" for RRI-3 of the same RSC to allow direct outgoing connection through the C.O. line or Tie line.

STEP 7: **ASPD** - Assign Abbreviated Digit Code for called party.

STEP 8: **AHLS** - Assign Route Restriction Class.
Specify a destination number to the entry of the physical station data created with the AIVSL command.
HOT TN: Hotline Tenant Number
HOT STN: Hotline Station Number
TYPE = 2 (indicates Outside party (ADC))
CON ADC: Assign Speed Calling Code assigned by ASPD.

[FCCS Data Settings]

This subsection describes only the additional or replacement commands used to implement the Multiple-Party Conference (Preset) feature in an FCCS network. When used as an FCCS feature, these commands replace the non-network commands (for example, AAED is replaced by AAEDN). Note that the data setting for FCCS Service varies depending on the node.

STEP 1: **ALGSN** - Assign a Telephone Number to the entry of the physical station data created with the AIVSL command.

STEP 2: **ASFCN** - Specify the information on Service Feature Restriction for stations.
Enter a Service Feature Index (SFI) and then specify Restriction Data (RES) for the Service Feature Restriction Class (SFC) of the stations.
SFI 32: RES = 1 (Line Circuit Reverse Relay Control is available)
SFI 50: RES = 1 [Off-Premise Extension (Long Line Telephone - No Howler, No PAD Control)]
SFI 71: RES = 1 (Distinctive Dial Tone) **Note 1**
SFI 72: RES = 1 (Called Station Switch Hook Flash is not available.)
SFI 80: RES = 1 (8-Party Conference Terminal)

SFI 221: RES = 1 (Music On Hold feature is disabled during 8/16/32-Party Conference call.)

Note 1: This setting is required for the Conferee ports.

- Other Service (0/1 = None/Available)
  Specify RES for the following SFI if necessary. Assign 0 to RES for SFIs other than those mentioned below.

  SFI 12: RES = 1 (Speed Calling System)
  SFI 14: RES = 1 (System Message Detail Recording - Trunk Basis)
  SFI 19: RES = 1 [Individual Speed Calling (Speed Calling - Station/Group)]
  SFI 27: RES = 1 (Account Codes/Authorization Codes/Forced Account Codes)
  SFI 58: RES = 1 (Station-to-Station Station Message Detail Recording (SMDR) Call)

- When an Extension Number is specified as the Conferee:

STEP 3: **ARSCN** - Specify the information on route restriction for the Route Restriction Class (RSC) of the conference leader port and conferee ports.
Restrict the conference leader port and conferee port for COT and Tie line routes.

Note: Specify restriction data "0" for RRI-0 and RRI-1 of the target RSC to restrict the incoming connection.

STEP 4: **AHLSN** - Assign Route Restriction Class.
Specify a destination number to the entry of the physical station data created with the AIVSL command.

HOT UGN: Hotline User Group Number

HOT TELN: Hotline Telephone Number

TYPE = 1 (indicates Station Number)

CON UGN: Destination Station's User Group Number

CON TELN: Destination Telephone Number

- When an Outgoing Trunk is specified as the Conferee:

STEP 3: **ARSCN** - Specify the information on route restriction for the Route Restriction Class (RSC) of the conference leader port and conferee ports.

Note: Specify restriction data "0" for RRI-0 and RRI-1 of the target RSC to restrict incoming connection. Also specify restriction data "1" for RRI-3 of the same RSC to allow direct outgoing connection through the C.O. line or Tie line.

STEP 4: **ASPDN** - Assign the Abbreviated Digit Code for called party.
STEP 5: **AHLSN** - Assign Route Restriction Class.
Specify a destination number to the entry of the physical station data created with the AIVSL command.
HOT UGN: Hotline User Group Number

HOT TELN: Hotline Telephone Number
TYPE = 2 (indicates Outside party (ADC))
CON ADC: Assign Speed Calling Code assigned by ASPD.

7.2 PROGRESSIVE

To perform PROGRESSIVE Feature, take the following steps.

STEP 1: **ASYD** - Assign System Data.

SYS1
Index 27, Bit 0-3: Tone Kind

Note that these data are available when SFI 71 = 1 in the ASFC command.

DATA	TONE	DATA	TONE
0	Dial Tone (DT)	8	-
1	Special Dial Tone (SPDT)	9	Sender Transmitting Tone (SDTT)
2	Ringing Back Tone (RBT)	A	Call Waiting Ringing Back Tone (CWRBT)
3	Consecutive Ringing Back Tone (CRBT)	B	-
4	Busy Tone (BT)	C	-
5	Reorder Tone (ROT)	D	-
6	Service Set Tone (SST)	E	-
7	Second Dial Tone (SDT)	F	-

STEP 2: **AIVSL** - Assign VS32 Conference Server Related Data.

In the case of 8-party conference, assign the extension number to the VS32 Conference Server channel listed below:

CH#0 (Conference leader port) and CH#1 through #7 (Conferee port).

CH#8 (Conference leader port) and CH#9 through #15 (Conferee port).

CH#16 (Conference leader port) and CH#17 through #23 (Conferee port).

CH#24 (Conference leader port) and CH#25 through #31 (Conferee port).

In the case of 16-party conference, assign the extension number to the VS32 Conference Server channel listed below:

CH#0 (Conference leader port) and CH#1 through #15 (Conferee port).

CH#16 (Conference leader port) and CH#17 through #31 (Conferee port).

In the case of 32-party conference, assign the extension number to the VS32 Conference Server channel listed below:

CH#0 (Conference leader port) and CH#1 through #31 (Conferee port).

[BASE DATA] tab

VS-ID: Assign the MAC Address of VS32.

CH: Assign the VS32 Conference Server channel number.

KIND: Choose 8-party conference, 16-party conference, or 32-party conference.

CONF KIND = PROGRESSIVE

TN: Tenant number

STN: Extension number

LENS: Line equipment numbers

TEC = 2 (indicates PB)

RSC: Assign Route Restriction Class other than ordinary extensions use.

SFC: Assign Service Feature Restriction Class other than ordinary extensions use.

[DETAIL SETTING] tab

Click Setting of 8/16/32-CONF and assign the necessary data such as PAD value for 8/16/32-party conference settings.

Setting of Expansion Mode
If necessary, click this and assign the operation mode for the conference trunk.

Note: Data settings will be reflected in the different ways, depending on the system. If this data is modified here while VS32 is working, the data will be reflected immediately. If some of the ports are being used, the data will be reflected to the ports when the ports are set to be idle.

For the configuration of the FCCS network, go to the section; otherwise continue with the next step.

STEP 3: **ASFC** - Specify the information on Service Feature Restriction for stations.
Enter a Service Feature Index (SFI) and then specify Restriction Data (RES) for the Service Feature Restriction Class (SFC) of the stations.

SFI 32: RES = 1 [Line Circuit Reverse Relay Control (for 16LC) is available]

SFI 71: RES = 1 (Distinctive Dial tone)

RES: 0/1 (DT/Depends on SYS Data)

SFI 72: RES = 1 (Called Extension Switch Hook Flash is not available)

SFI 221: RES = 1 (Music On Hold feature is disabled during 8/16/32-Party Conference call.)

Note: If SFI 71 and SFI 72 are set to 1, a tone specified in SYS data (ASYD Index 27 Bit 0-3) will be heard when conference number is dialed.

STEP 4: **ARSC** - Specify the information on route restriction for the Route Restriction Class (RSC) of the conference leader port and conferee ports.
Restrict the RSC of the conference leader port and conferee ports for COT and Tie line routes.

STEP 5: If multiple conference trunks exist, set the type of Station Hunting for Conference Leader port by using either of the following commands:
ASHP - Set the type of Station Hunting to Pilot.
ASHC - Set the type of Station Hunting to Circular.

[FCCS Data Settings]

This subsection describes only the additional or replacement commands used to implement the Multiple-Party Conference (Preset) feature in an FCCS network. When used as an FCCS feature, these commands replace the non-network commands (for example, AAED is replaced by AAEDN). Note that the data setting for FCCS Service varies depending on the node.

STEP 1: **ALGSN** - Assign a Telephone Number to the entry of the physical station data created with the AIVSL command.

STEP 2: **ASFCN** - Specify the information on Service Feature Restriction for stations.
Enter a Service Feature Index (SFI) and then specify Restriction Data (RES) for the Service Feature Restriction Class (SFC) of the stations.

SFI 32: RES = 1 [Line Circuit Reverse Relay Control (for 16LC) is available]

SFI 71: RES = 1 (Distinctive Dial tone)

RES: 0/1 (DT/Depends on SYS Data)

SFI 72: RES = 1 (Called Extension Switch Hook Flash is not available)

SFI 221: RES = 1 (Music On Hold feature is disabled during 8/16/32-Party Conference call.)

Note: If SFI 71 and SFI 72 are set to 1, a tone specified in SYS data (ASYD, Index 27, Bit 0-3) will be heard when conference number is dialed.

STEP 3: **ARSCN** - Specify the information on route restriction for the Route Restriction Class (RSC) of the conference leader port and conferee ports.
Restrict the RSC of the conference leader port and conferee ports for COT and Tie line routes.

STEP 4: If multiple conference trunks exist, set the type of Station Hunting for Conference Leader port by using either of the following commands:

ASHPN - Set the type of Station Hunting to Pilot for NDM.
ASHCN - Set the type of Station Hunting to Circular for NDM.

7.3 MEET-ME

To perform MEET-ME Feature, take the following steps.

STEP 1: **ASYD** - Assign System Data.
SYS1
Index 20, Bit 6 = 0 (Line Circuit Reverse Relay Control when Reorder Tone is connected to an extension)
Index 64, Bit 0: Assign the compression method (0:μ-law/1:A-law)

STEP 2: **AIVSL** - Assign VS32 Conference Server Related Data.
VS-ID: Assign the MAC Address of the VS32.

In the case of 8-party conference, assign the extension number to the VS32 Conference Server channel listed below:
CH#0 (Conference leader port) and CH#1 through #7 (Conferee port).
CH#8 (Conference leader port) and CH#9 through #15 (Conferee port).
CH#16 (Conference leader port) and CH#17 through #23 (Conferee port).
CH#24 (Conference leader port) and CH#25 through #31 (Conferee port).

In the case of 16-party conference, assign the extension number to the VS32 Conference Server channel listed below:
CH#0 (Conference leader port) and CH#1 through #15 (Conferee port).
CH#16 (Conference leader port) and CH#17 through #31 (Conferee port).

In the case of 32-party conference, assign the extension number to the VS32 Conference Server channel listed below:
CH#0 (Conference leader port) and CH#1 through #31 (Conferee port).

[BASE DATA] tab
CH: Assign the VS32 Conference Server channel number.
KIND: Choose 8-party conference, 16-party conference, or 32-party conference.
CONF KIND = MEET-ME
TN: Tenant number
STN: Extension number
LENS: Line equipment numbers
TEC = 2 (indicates PB)
RSC: Assign Route Restriction Class other than ordinary extensions use.
SFC: Assign Service Feature Restriction Class other than ordinary extensions use.

Example: When accommodated in PIR1, Group 3.

VS Channel	LENS	Exten-sion Number	TEC	SFC	RSC	Reference
0	001030	3200	2	10	10	Conference leader port
1	001031	3201	2	10	10	
2	001032	3202	2	10	10	
3	001033	3203	2	10	10	
4	001034	3204	2	10	10	Conferee port
5	001035	3205	2	10	10	
6	001036	3206	2	10	10	
7	001037	3207	2	10	10	

[DETAIL SETTING] tab

Click Setting of 8/16/32-CONF and assign the necessary data such as PAD value for 8/16/32-party conference settings.

If necessary, click Setting of Expansion Mode and assign the operation mode for the conference trunk.

Note: Data settings will be reflected in the different ways, depending on the system. If this data is modified here while VS32 is working, the data will be reflected immediately. If some of the ports are being used, the data will be reflected to the ports when the ports are set to be idle.

Note: To assign data by Logical Extension Number, use the ALGSN command.

For the configuration of the FCCS network, go to the [FCCS Data Settings] section; otherwise continue with the next step.

STEP 3: **ASFC** - Specify the information on Service Feature Restriction for stations.

Enter a Service Feature Index (SFI) and then assign 1 to Restriction Data (RES) for the Service Feature Restriction Class (SFC) of the stations.

- SFC of the conference leader port extension
  SFI 32: RES = 0 [Line Circuit Reverse Relay Control (for 16LC) is not available.]
  SFI 72: RES = 1 (Called Station Switch Hook is not available.)

- SFC of the extension in conference trunk
  SFI 221: RES = 1 (Music On Hold feature is disabled.)

STEP 4: **ARSC** - Specify the information on route restriction for the Route Restriction Class (RSC) of the conference leader port and conferee ports.

Restrict the RSC of the conference leader port and conferee ports for COT and Tie line routes.

STEP 5: Set the type of Station Hunting by using any of the following commands. Ensure that all the conferee ports are configured to be in the same Hunting Group or that the Conference Leader port and conferee ports are in the same Hunting Group.

ASHP - Set the type of Station Hunting to Pilot.
ASHC - Set the type of Station Hunting to Circular.
ASHU - Set the type of Station Hunting to UCD.

[FCCS Data Settings]

This subsection describes only the additional or replacement commands used to implement the Multiple-Party Conference (Preset) feature in an FCCS network. When used as an FCCS feature, these commands replace the non-network commands (for example, AAED is replaced by AAEDN). Note that the data setting for FCCS Service varies depending on the node.

STEP 1: **ALGSN** - Assign a Telephone Number to the entry of the physical station data created with the AIVSL command.

STEP 2: **ASFCN** - Specify the information on Service Feature Restriction for stations.

Enter a Service Feature Index (SFI) and then specify Restriction Data (RES) for the Service Feature Restriction Class (SFC) of the stations.

- SFC of the conference leader port extension
  SFI 32: RES = 0 [Line Circuit Reverse Relay Control (for 16LC) is not available.]
  SFI 72: RES = 1 (Called Station Switch Hook is not available.)

- SFC of the extension in conference trunk
  SFI 221: RES = 1 (Music On Hold feature is disabled.)

STEP 3: **ARSCN** - Specify the information on route restriction for the Route Restriction Class (RSC) of the conference leader port and conferee ports.

Restrict the RSC of the conference leader port and conferee ports for COT and Tie line routes.

STEP 4: Set the type of Station Hunting by using any of the following commands. Ensure that all the conferee ports are configured to be in the same Hunting Group or that the Conference Leader port and conferee ports are in the same Hunting Group.

ASHPN - Set the type of Station Hunting to Pilot for NDM.
ASHCN - Set the type of Station Hunting to Circular for NDM.
ASHUN - Set the type of Station Hunting to UCD for NDM.

7.4 Add-On

Refer to ADD-ON CONFERENCE - 8 PARTY [A-121].

8. Data Programming for Three-Way Calling

To perform Three-Way Conference Feature, take the following steps.

STEP 1: **ASYD** - Assign System Data
SYS2, Index 1
Bit 0,1: Transfer Service
Bit 1 = 0, Bit 0 = 1 Transfer effective (available excluding tandem)
Bit 1 = 1, Bit 0 = 1 Transfer effective (available including tandem)
SYS3, Index 2: Recovery Timer

STEP 2: **ASYDL** - Assign System Data
SYS1, Index 830, Bit 5: This system data works in a different way, depending on following countries:
- When Bit 5=0
  Austria
  : Three-Way Calling Tone (Add-On Ticker) is sent at a cycle of four seconds.
  EMEA except for Austria
  : Three-Way Calling Tone (Add-On Ticker) is sent at a cycle of 32 seconds.
  North America and Australia
  : Three-Way Calling Tone is not sent.
- When Bit 5=1
  Three-Way Calling Tone (Add-On Ticker) is sent only once when Three-Way Calling is established.

SYS1, Index 811
Bits = 0-7: Specify the Country Code for IP terminal Tone. For more information about the country code, refer to Command Manual.

STEP 3: **ASTD** - Assign State Translation Data
The following need to be assigned when Three-Party conference with COT trunk is performed.
STM = 2
STS = 0 (indicates Tandem with COT)
ST = 1 (indicates Effective)

STEP 4: **AIVSL** - Assign the system data of Three-party conference trunk
VS-ID: Assign the MAC Address of the VS32.

[BASE DATA] tab
CH: Assign the following VS32 Conference Server channel number.
CH#0-2
CH#3-5
CH#6-8
CH#9-11
CH#12-14
CH#15 (This channel cannot be assigned)
CH#16-18
CH#19-21
CH#22-24

CH#25-27
CH#28-30
CH#31 (This channel cannot be assigned)

KIND = 3-CONF

RT: Route Number (909)

TK: Trunk Number

LENS: Line equipment numbers

[DETAIL SETTING] tab

Click Setting of 3-CONF and check the Adjust 3-CONF setting box if the following data should be modified:

- Threshold value to define extension/trunk

- PAD value for adjusting conference level

- PAD value for adjusting conference value when Three-Way conference trunk is connected to UMS recording

- PAD value of UMS conversion recording

Note: Data settings will be reflected in the different ways, depending on the system. If this data is modified here while VS32 is working, the data will be reflected immediately. If some of the ports are being used, the data will be reflected to the ports when the ports are set to be idle.

STEP 5: **MBTK** - Make Busy Trunk

Release the Make Busy of the trunk specified by AIVSL.

STEP 6: **ARRC** - Assign Alternative Route Restriction

Permit the tandem connection restriction for Three-Way conference with route outside the office.

ICRT: Incoming Route Number (1-255)

OGRT: Outgoing Route Number (1-255)

ARI A-RES: Restriction Data for Alternative Route INDEX-A (0/1)

RES = 1 (indicates No Restriction)

STEP 7: **ARSC** - Assign Route Restriction Data

Specify restriction data between the extension and outgoing trunks when using Three-Way conference via the outgoing trunk.

If you will use Least Cost Routing feature, refer to Least Cost Routing for MG-PRI [L-54].

9. Data Programming for Least Cost Routing with VS32 Conference Server

STEP 1: **AMAIL** - Assign MA-IDs
MA-ID: Message Area ID within the range from 0 to 9999.
AREA NAME: Enter an area name within 16 characters.

STEP 2: **AMAA** - Assign MA-ID allocation data
Assign MA-IDs to MOHs of VS32s. To confirm the MA-ID, enter the following parameters and click the Read button to display the MA-IDs.

- FUNCTION = Trunk (When a type of terminals is determined by trunk)

  TYPE: Select a type for specifying trunk (Individual or Collective).

  RT: Route Number, this can be specified when TYPE = Individual.

  RT START-END: The range of the Route Numbers, this can be specified when TYPE = Collective.

  MA-ID: Message Area ID within the range from 0 to 9999.

- FUNCTION = LENS (When a type of terminals is determined by Accommodation Location)

  TYPE: Select a type for specifying accommodation locations (Individual or Collective).

  LENS: Accommodation location, this can be specified when TYPE = Individual.

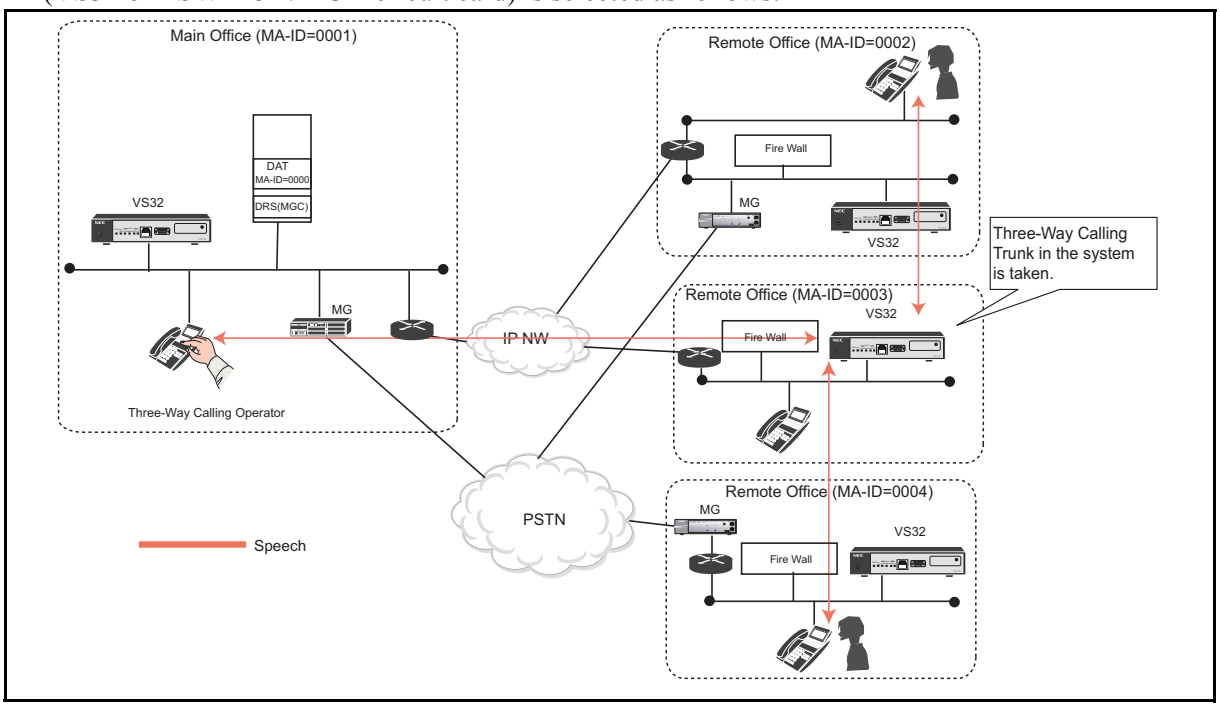
  LENS START-END: The range of the Route Numbers, this can be specified when TYPE = Collective.

  MA-ID: Message Area ID within the range from 0 to 9999.

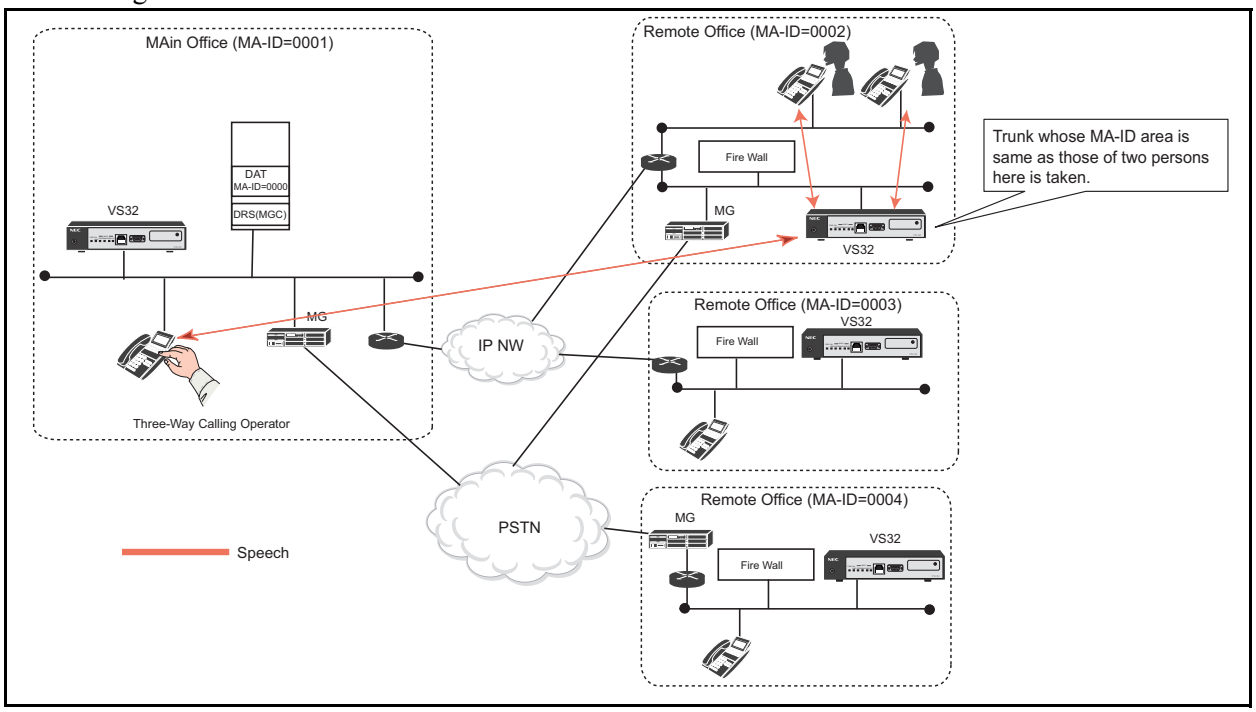
9.1 When Least Cost Routing is used with Three-Way Calling

Note: This feature is not available in FCCS.

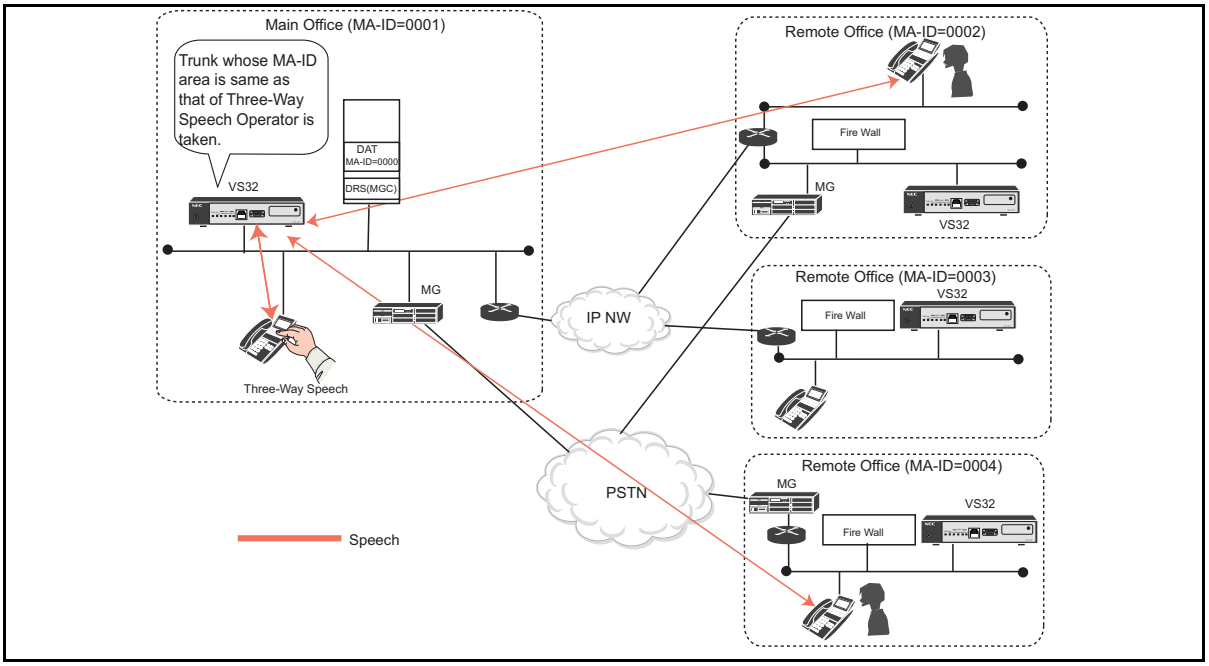
When Index 830 Bit 0 = 0 and Bit 1 = 0 on the ASYDL(N) command is assigned, an appropriate trunk (VS32 or TSW-BOX/MUX circuit card) is selected as follows:



When Index 830 Bit 0 = 1 and Bit 1 = 0 on the ASYDL(N) command (Voice quality is given top priority), if two of them are in the same location, a trunk belonging to the two people will be selected for Three-Way Calling.

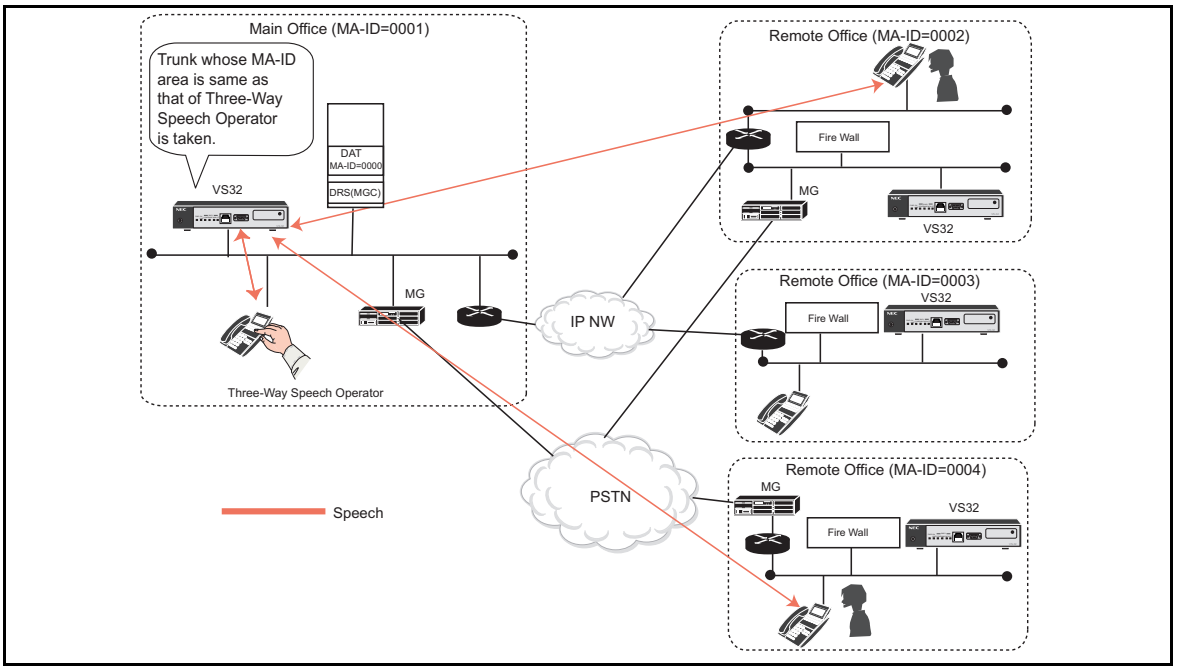


If all of three are located in different places, a trunk belonging to the Conference Leader will be selected for Three-Way Calling. Thereby, voice packets in the network can be kept to a minimum.
If the trunk cannot be used, one with MA-ID = 0 (VS32 or CFT in TSW-BOX/MUX circuit cards) will be selected.



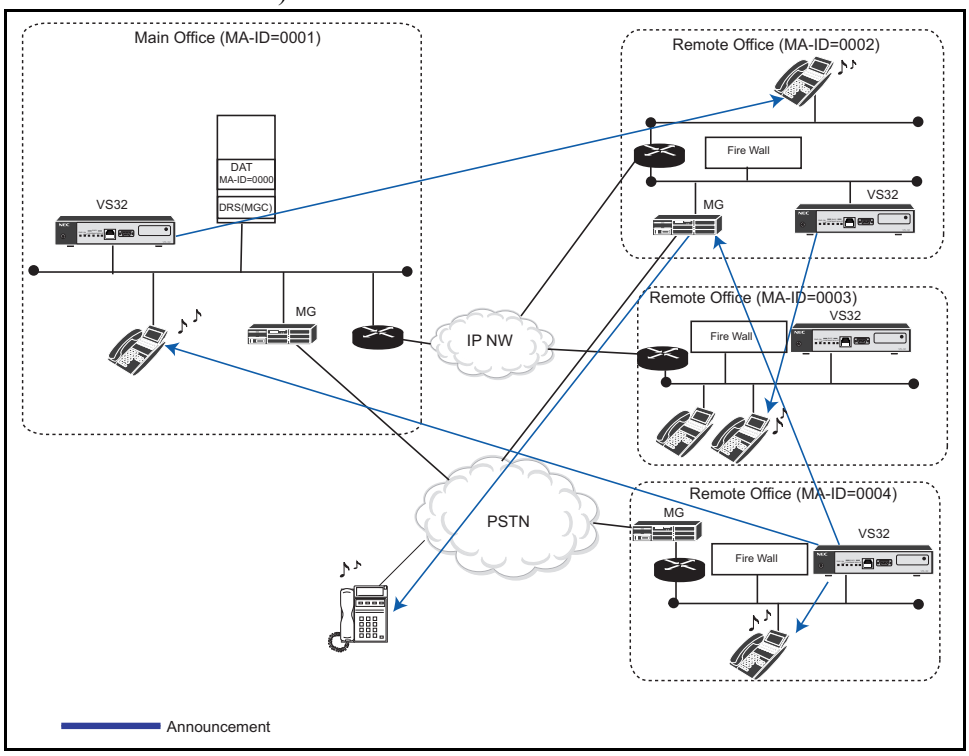
When Index 830 Bit 0 = 0 and Bit 1 = 1, an appropriate trunk (VS32 or TSW-BOX/MUX circuit card) is selected:

A trunk belonging to the Conference Leader is selected for Three-Way Calling. Thereby, terminals are limited to use trunks located in some locations.
If the trunk cannot be used, one with MA-ID = 0 (VS32 or CFT in TSW-BOX/MUX circuit cards) will be selected.



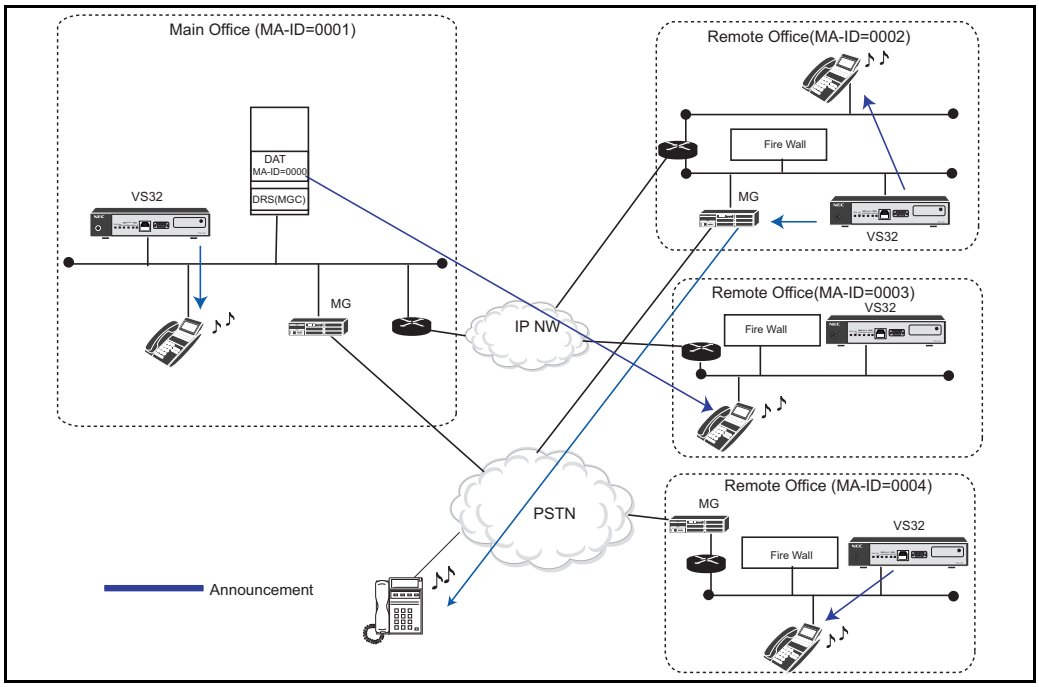
9.2 When Least Cost Routing is used with Announcement Service Feature

When CDN:79 FTRKS on the ARTI/ARTIN command is assigned “0”, an appropriate trunk (VS32 or TSW-BOX/MUX circuit cards) is selected:



When CDN:79 FTRKS on the ARTI/ARTIN command is assigned “1” (Voice quality is given top priority)

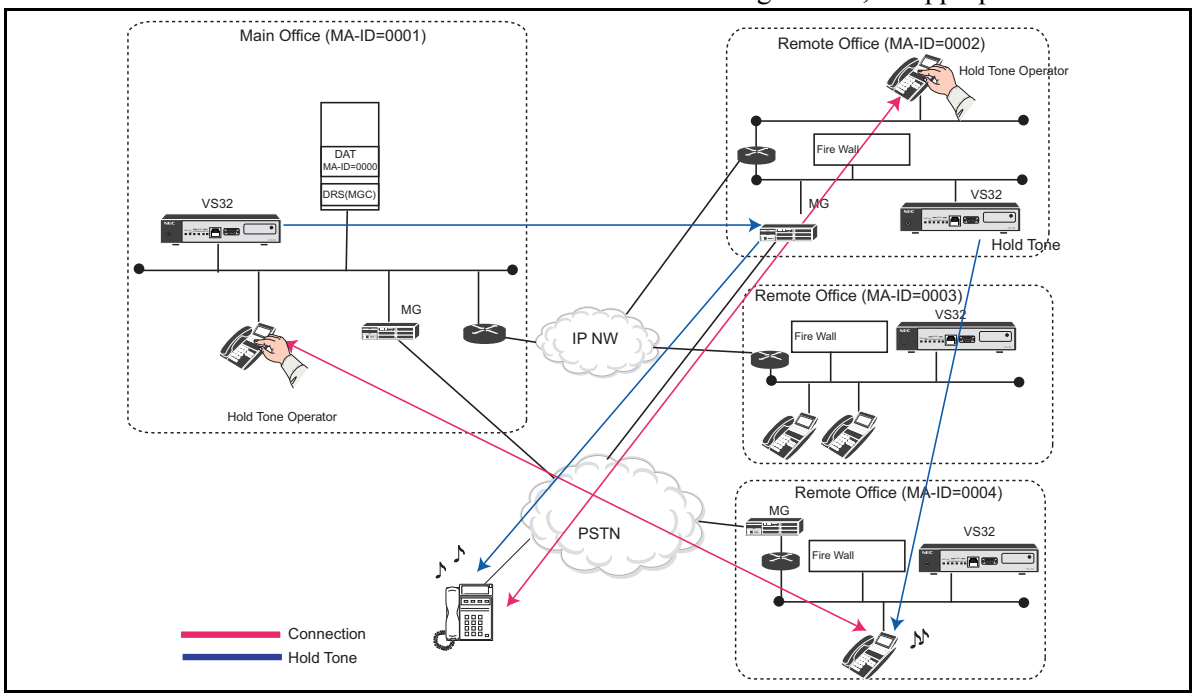
A trunk in the same location with announcement listener will be selected. If the trunk cannot be selected, a trunk with MA-ID = 0 will be selected.



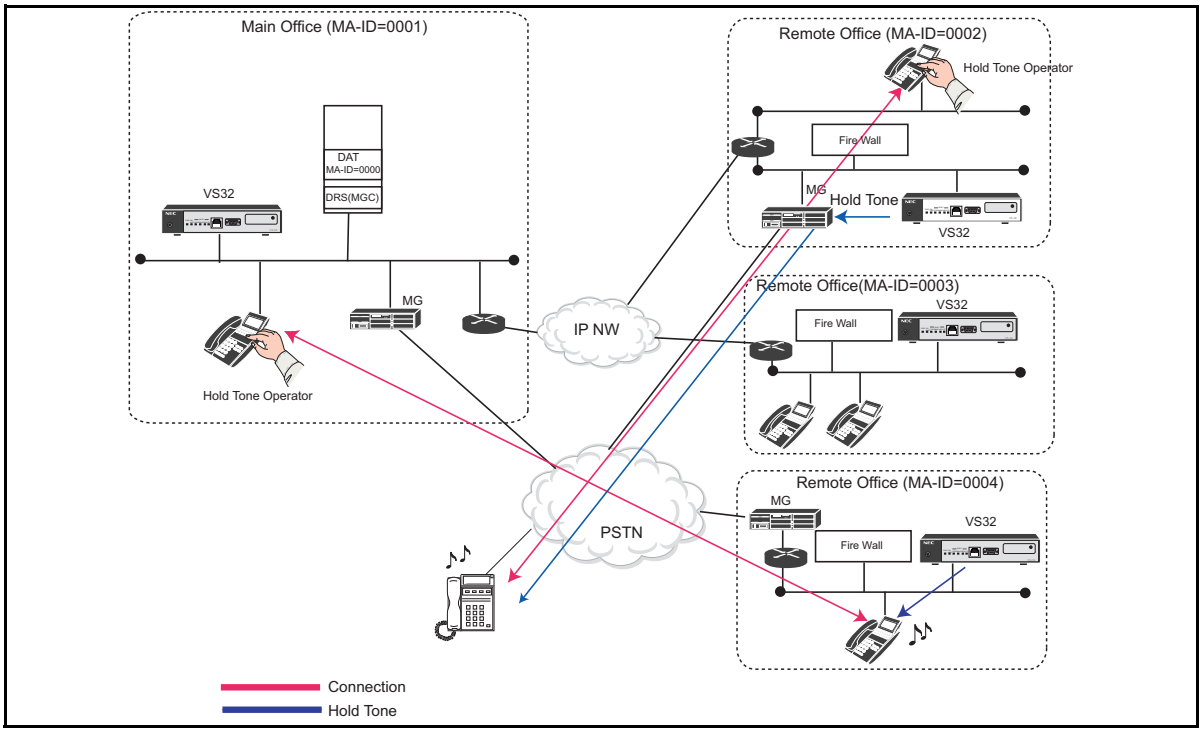
9.3 When Least Cost Routing is used with IP External Music On Hold Feature

Note: This feature is not available in FCCS.

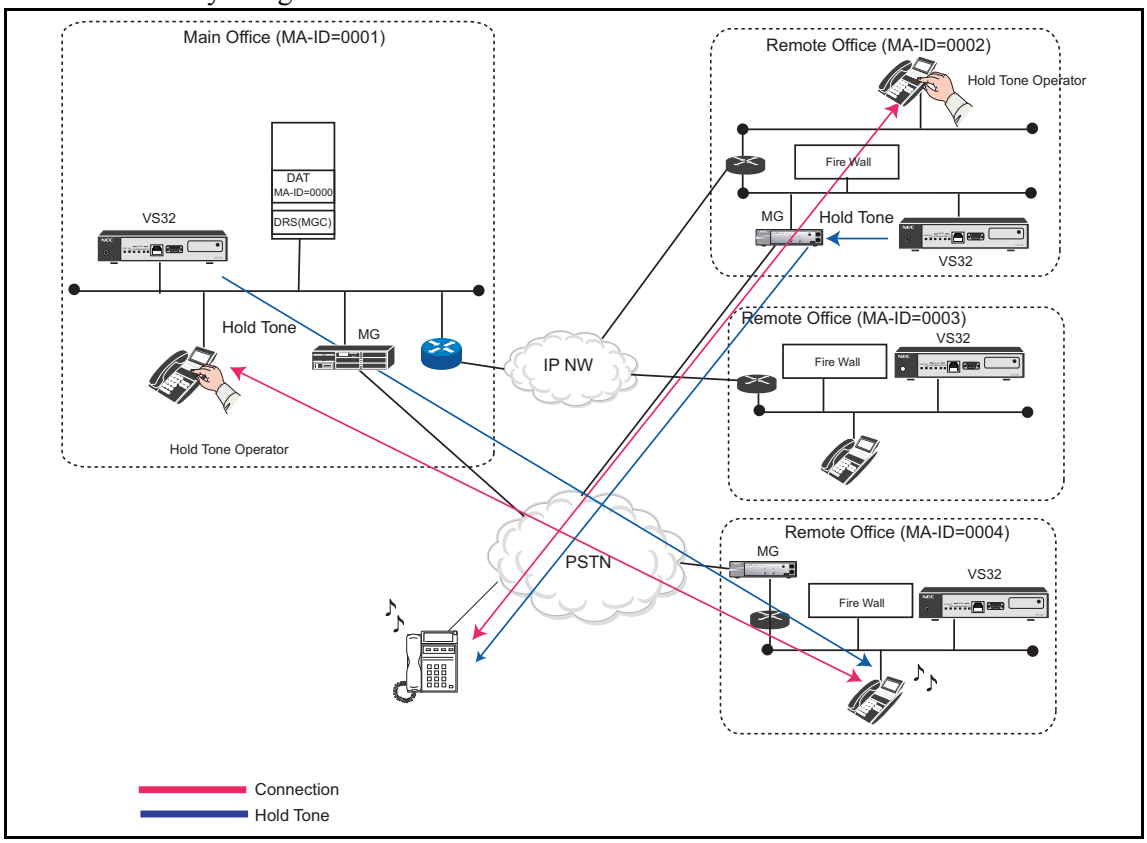
When CDN:79 FTRKS on the ARTI/ARTIN command is assigned "0", an appropriate trunk is selected:



When CDN:79 FTRKS on the ARTI/ARTIN command is assigned "1" (Voice quality is given top priority) A trunk in the same location with announcement listener will be selected. If the trunk cannot be selected, a trunk with MA-ID = 0 will be selected.



When CDN:79 FTRKS on the ARTI/ARTIN command is assigned “2” (Conference Leader is given top priority)
A trunk in the same location with Conference Leader will be selected. If the trunk cannot be selected, a trunk with MA-ID = 0 will be selected. At this time, External Hold Tone or Internal Hold Tone on terminals can be selected by using Index 810 Bit 2.

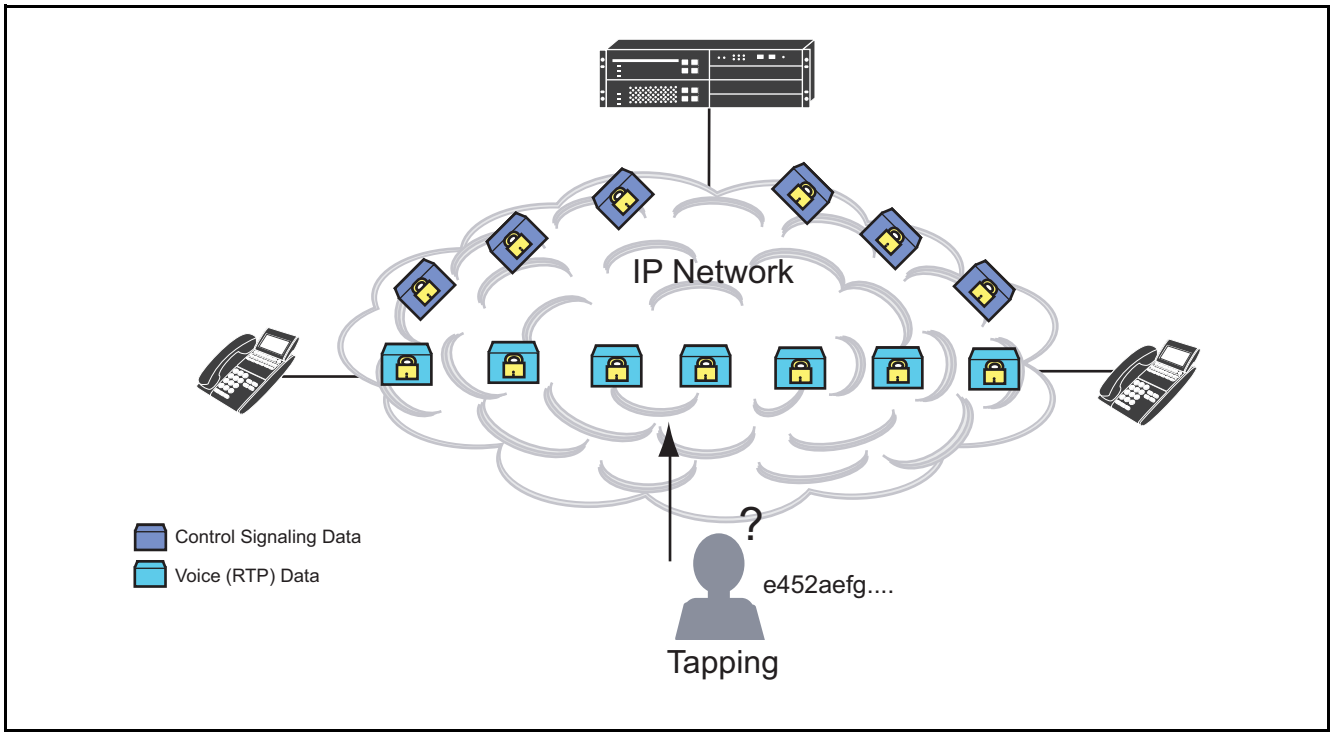


V-27 VoIP Encryption

General Description

This feature provides encrypted network system to strengthen security over IP network. This feature provides encryption feature for voice data [Real-Time Packets (RTP)] and signaling data (SIP Messages) that carries login numbers, passwords, and encryption keys. The encryption of voice data and signaling data both can be operated only in SIP-compatible equipment (hereinafter called “VoIP Encryption”). Non-SIP-compatible equipment can perform encryption only for voice data (hereinafter called “RTP Encryption”).
To use fully-featured encryption including signaling data, the verification using a one-time password (that is called “SIP Terminal Authentication [S-152]”) is required. The SIP Terminal Authentication is completed successfully, and then Encryptions for RTP data and signaling data are initiated with Advanced Encryption Standard (AES) modes (128-bit length). This secures a network by preventing third party from eavesdropping on its data.

Note: This feature is not available in China.



Operating Procedure

None.

Service Conditions

<Important Notices>

1. Using the encryption feature in the Telephony Server, accommodate PHDs/PHIs compatible with the feature (see the spread sheet for Signaling Data/Voice Data) into the system. Non-compatible PHDs/PHIs and compatible PHDs/PHIs cannot coexist in the same system.
2. Using the encryption feature for RTP in the Telephony Server, accommodate IPPAD compatible with its encryption feature into the system. Non-support IPPADs and support IPPADs cannot coexist in the same system.

<Interactions with Other Features>

1. When SIP TERMINAL AUTHENTICATION [S-152] is performed, encryption is automatically enabled for security. If the terminals/devices have not completed SIP TERMINAL AUTHENTICATION [S-152], the voice and signaling data are not encrypted. Also, although IP terminal is not supported by SIP TERMINAL AUTHENTICATION [S-152], voice data encryption is possible.
2. This feature is not available with OAI and ACD services.
3. When using VoIP Encryption on VS32(Dual)[SIP], do not mix other type of three-way conference trunks (TSW-BOX/MUX circuit card and proprietary mode of VS32(Dual)) in one system. Because only the SIP mode of VS32 supports encryption feature, encryption cannot work consistently in the system depending on the seized VS32 type. **Note 1**

Conference Trunk Type	Encryption
CFT with TSW-BOX/MUX circuit card	No Encryption
3-Way Conference with VS32(Dual) (proprietary mode)	No Encryption
3-Way Conference with VS32(Dual)(SIP mode)	Encryption

Note 1: Accommodating location (LENS) of the conference leader determines which conference trunk is to be seized in the system.

4. The devices/stations can continue to execute the encryption feature, even if they are rescued by SR-MGC.

<Hardware/Software>

1. The following are conditions for encryptions of voice data when terminals/devices are connected.

		Encrypted							Non-Encrypted
		IP terminal	SIP Multiple Line terminal Note 3	WLAN Handset (MH Series)	MG(PRI) Note 2 MG(BRI) 8LC MG-COT Note 2 MG-T1	IPPAD	VS32 Note 2	MG (SIP) Note 4	
Encrypted	IP terminal	X	X	-	X	X	X	X	-
	SIP Multiple Line terminal Note 3		X	-	X	X	X	X	-
	WLAN Handset (MH Series)			-	-	-	-	-	-
	MG(PRI) Note 2 MG(BRI) 8LC MG-COT Note 2 MG-T1				X	X	X	X	-
	IPPAD					X	X	X	-
	VS32 Note 2						X	X	-
	MG (SIP) Note 4							X	-
Non-Encrypted									

X: Available -: Not Available

Note 2: Only applies when used in SIP mode.

Note 3: The conditions of the SP350 are same as SIP Multiple Line terminal.

Note 4: Available since FP95-114 V4 Issue 2.0.

2. The following are the conditions when this feature is used with a Softphone.

   (a) When this feature is used with SP350, enables the encryption on SP350. See *UNIVERGE Soft Client SP350 Operations and Maintenance Manual* for more details.

   (b) This feature is not available for DtermSP30. Disable the encryption on DtermSP30.

   (c) SP350/DtermSP30 does not support SIP TERMINAL AUTHENTICATION [S-152].

3. If called and calling terminals/devices do not have the same encryption mode, or both/either of the terminals/devices cannot handle this feature, the voice packets will not be encrypted.
4. Using the encryption feature, use the terminal compatible with the feature.
   The following firmware is compatible with the encryptions

For Signaling Data

AVAILABLE EQUIPMENT	AVAILABLE FIRMWARE VERSION
MG(BRI) Card [SCA-2BRIA-B/SCA-2BRIA] MG(BRI) Box [MG-2BRIA]	SP-3826 Issue 1.0 or later
MG(PRI) Card [SCA-24PRIA] MG(PRI) Box [MG-24PRIA] **Note 5**	SP-3884 Issue 5A or later
MG(PRI) Card [SCA-30PRIA] MG(PRI) Box [MG-30PRIA] **Note 5**	
UG50 (PRT 1.5M) [SIP]	Any version
UG50 (PRT 2M) [SIP]	Any version
UG50-24PRIA [SIP]	Any version
UG50-30PRIA [SIP]	Any version
MG-COT [SCA-6COTB] **Note 5**	SP-3948 6MGCOT PROG-A Issue 1 or later
MG-COT [SCA-6COTC] **Note 5**	SP-3948 6MGCOT PROG-A Issue 1 or later
UG50 (COT-TYPE2) [SIP] **Note 5**	Any version
UG50 (PGT-TYPE2) [SIP] **Note 5**	Any version
UG50-6COT [SIP] **Note 5**	Any version
MG-T1(SIP) Card [SCA-24DTIA-B/SCA-24DTIA]	SP-3926 MG DTI PROG-A Issue 1A or later
8LC Card [SCA-8LCA/SCA-8LCA-EMEA]	SP-3893 8MC PROG-A Issue 2 or later
8LC Card [SCA-8LCC/SCA-8LCC-EMEA]	SP-3983 8MC PROG-B Issue 1 or later
UG50 (LC) [SIP]	Any version
UG50-8LC	Any version
DtermIP (SIP)	1.4.3.0 or later
DT700 Series	2.2.0.0 or later
DT800 Series	3.0.0.0 or later
VS-32(Dual) Card [SCA-VS32VA-B/SCA-VS32VA] VS-32(Dual) Box [MG-VS32VA] **Note 5**	SP-3891 CA-VS32 PROG-A Issue 3A or later
SP350	R1 or later

For Voice Data **Note 6**

AVAILABLE EQUIPMENT	AVAILABLE FIRMWARE VERSION
MG (SIP) [MG-128SIPMGL-A]	SP-4089 MGSIP(MM) PROG-A Issue 3.0 or later **Note 7**
MG (SIP) [MG-128SIPMGJ-B/MG-128SIPMGJ]	SP-4060 MGSIP PROG-L Issue 4.0 or later **Note 7**
Software-based MG-SIP	SP-4085 MGSIPVM PROG-E Issue 3.0 or later **Note 7**
MG(BRI) Card [SCA-2BRIA-B/SCA-2BRIA] MG(BRI) Box [MG-2BRIA]	SP-3826 Issue 1.0 or later
MG(PRI) Card [SCA-24PRIA] MG(PRI) Box [MG-24PRIA] **Note 5**	SP-3884 Issue 5A or later
MG(PRI) Card [SCA-30PRIA] MG(PRI) Box [MG-30PRIA] **Note 5**	
MG-COT [SCA-6COTB] **Note 5**	SP-3948 6MGCOT PROG-A Issue 1 or later
MG-COT [SCA-6COTC] **Note 5**	SP-3948 6MGCOT PROG-A Issue 1 or later
MG-T1(SIP) Card [SCA-24DTIA-B/SCA-24DTIA]	SP-3926 MG DTI PROG-A Issue 1A or later
8LC Card [SCA-8LCA/SCA-8LCA-EMEA]	SP-3893 8MC PROG-A Issue 2 or later
8LC Card [SCA-8LCC/SCA-8LCC-EMEA]	SP-3983 8MC PROG-B Issue 1 or later
IPPAD	PA-32IPDA/SP-3835 Issue 3.0 or later
DtermIP	V3.00 or later
DtermIP (SIP)	1.4.3.0 or later
DT700 Series	2.2.0.0 or later
DT800 Series	3.0.0.0 or later
VS-32(Dual) Card [SCA-VS32VA-B/SCA-VS32VA] VS-32(Dual) Box [MG-VS32VA] **Note 5**	SP-3891 CA-VS32 PROG-A Issue 3A or later
SP350	R1 or later

Note 5: Only for SIP mode.

Note 6: UG50 does not support this feature for Voice Data.

Note 7: Available since FP95-114 V4 Issue 2.0.

5. The following settings are necessary for the encryption of SIP Multiple Line terminal.

   (a) Enable SIP Terminal Authentication and VoIP Encryption on the terminal side. **Note 8**

(b) Assign one time password as explained in SIP TERMINAL AUTHENTICATION [S-152] in this manual.

Note 8: For DT830/DT820/DT700, see the respective <SIP Terminal Authentication and VoIP Encryption> in Security Mode/Settings section of Chapter 3 IP STATION of Peripheral Equipment Description (Multi-Line Telephone).
For DtermIP (SIP), see <Authentication Mode> and <One Time Password> in Encryption section of Chapter 3 IP STATION of Peripheral Equipment Description (Multi-Line Telephone).

6. In the following cases, the encryption of SIP Multiple Line terminal is not possible.
   - Log in without performing SIP TERMINAL AUTHENTICATION [S-152].
   - When SIP Terminal Authentication and VoIP Encryption are not enabled in the terminal.
7. If Telephony Server and Terminals communicate with each other over NAT, the encryption part of the signaling data needs to be “A part of the data is encrypted” in ASYDL SYS 1 Index 831 Bit 3=0. If the setting is changed, the key update needs to be performed using either Cipher Key Update in the ASECL command or have the operating terminals re-login to the system.
8. Refer to the Peripheral Equipment Description (UG50) for detailed conditions for UG50.

9. The number of channels that are processed depends on codec and payload cycle used by each IP component. Therefore, adjustments will be needed, such as placing unnecessary channels into the Make Busy status. The following table shows the number of available channels depending on the encryption status of each IP component.

Non-Encrypted						Encrypted					
MG(BRI) (Note 9)											
	10 ms	20 ms	30 ms	40 ms	60 ms		10 ms	20 ms	30 ms	40 ms	60 ms
G.711	4ch	4ch	4ch	4ch	-	G.711	4ch	4ch	4ch	4ch	-
G.729a	4ch	4ch	4ch	4ch	-	G.729a	4ch	4ch	4ch	4ch	-
G.723.1	-	-	4ch	-	4ch	G.723.1	-	-	4ch	-	4ch
MG(PRI) (1.5M) (Note 10) (Note 11)											
	10 ms	20 ms	30 ms	40 ms	60 ms		10 ms	20 ms	30 ms	40 ms	60 ms
G.711	20ch	23ch	23ch	23ch	-	G.711	10ch	20ch	23ch	23ch	-
G.729a	20ch	23ch	23ch	23ch	-	G.729a	10ch	20ch	23ch	23ch	-
G.723.1	-	-	23ch	-	23ch	G.723.1	-	-	23ch	-	23ch
MG(PRI) (2M) (Note 12) (Note 13)											
	10 ms	20 ms	30 ms	40 ms	60 ms		10 ms	20 ms	30 ms	40 ms	60 ms
G.711	15ch	26ch	30ch	30ch	-	G.711	10ch	17ch	22ch	27ch	-
G.729a	16ch	30ch	30ch	30ch	-	G.729a	10ch	20ch	28ch	30ch	-
G.723.1	-	-	30ch	-	30ch	G.723.1	-	-	28ch	-	30ch
8LC (Note 14) (Note 15)											
	10 ms	20 ms	30 ms	40 ms	60 ms		10 ms	20 ms	30 ms	40 ms	60 ms
G.711	8ch	8ch	8ch	8ch	-	G.711	8ch	8ch	8ch	8ch	-
G.729a	8ch	8ch	8ch	8ch	-	G.729a	8ch	8ch	8ch	8ch	-
G.723.1	-	-	8ch	-	8ch	G.723.1	-	-	8ch	-	8ch
VS32(Dual) (Note 16)											
	10 ms	20 ms	30 ms	40 ms	60 ms		10 ms	20 ms	30 ms	40 ms	60 ms
G.711	16ch	32ch	32ch	32ch	-	G.711	16ch	32ch	32ch	32ch	-
G.729a	16ch	32ch	32ch	32ch	-	G.729a	16ch	32ch	32ch	32ch	-
G.723.1	-	-	32ch	-	32ch	G.723.1	-	-	32ch	-	32ch
IPPAD Card (7U-PIR)											
	10 ms	20 ms	30 ms	40 ms	60 ms		10 ms	20 ms	30 ms	40 ms	60 ms
G.711	-	32ch	32ch	32ch	-	G.711	-	32ch	32ch	32ch	-
G.729a	-	32ch	32ch	32ch	-	G.729a	-	32ch	32ch	32ch	-
IPPAD Card (8U-PIR)											
	10 ms	20 ms	30 ms	40 ms	60 ms		10 ms	20 ms	30 ms	40 ms	60 ms
G.711	-	32ch	32ch	32ch	-	G.711	-	30ch	32ch	32ch	-
G.729a	-	32ch	32ch	32ch	-	G.729a	-	32ch	32ch	32ch	-

Non-Encrypted						Encrypted					
G.723.1	-	-	32ch	-	-	G.723.1	-	-	32ch	-	-
6COT (Note 17) (Note 18)											
	10 ms	20 ms	30 ms	40 ms	60 ms		10 ms	20 ms	30 ms	40 ms	60 ms
G.711	6ch	6ch	6ch	6ch	-	G.711	6ch	6ch	6ch	6ch	-
G.729a	6ch	6ch	6ch	6ch	-	G.729a	6ch	6ch	6ch	6ch	-
G.723.1	-	-	6ch	-	6ch	G.723.1	-	-	6ch	-	6ch
MG (SIP) (128ch) (Note 19)											
	10 ms	20 ms	30 ms	40 ms	60 ms		10 ms	20 ms	30 ms	40 ms	60 ms
G.711	128ch	128ch	128ch	128ch	-	G.711	128ch	128ch	128ch	128ch	-
G.729a	128ch	128ch	128ch	128ch	-	G.729a	128ch	128ch	128ch	128ch	-
G.723.1	-	-	128ch	-	128ch	G.723.1	-	-	128ch	-	128ch
MG (SIP) (254ch) (Note 19)											
	10 ms	20 ms	30 ms	40 ms	60 ms		10 ms	20 ms	30 ms	40 ms	60 ms
G.711	254ch	254ch	254ch	254ch	-	G.711	254ch	254ch	254ch	254ch	-
G.729a	254ch	254ch	254ch	254ch	-	G.729a	254ch	254ch	254ch	254ch	-
G.723.1	-	-	254ch	-	254ch	G.723.1	-	-	254ch	-	254ch

Note 9: MG(BRI) indicates MG(BRI) Card [SCA-2BRIA-B/SCA-2BRIA] / MG(BRI) Box [MG-2BRIA].

Note 10: MG(PRI) indicates MG(PRI) Card [SCA-24PRIA] / MG(PRI) Box [MG-24PRIA].

Note 11: UG50 (PRT 1.5M) [SIP] and UG50-24PRIA [SIP] are supported.

Note 12: MG(PRI) indicates MG(PRI) Card [SCA-30PRIA] / MG(PRI) Box [MG-30PRIA].

Note 13: UG50 (PRT 2M) [SIP] and UG50-30PRIA [SIP] are supported.

Note 14: 8LC indicates 8LC Card [SCA-8LCA/SCA-8LCA-EMEA/SCA-8LCC/SCA-8LCC-EMEA].

Note 15: UG50 (LC) [SIP] and UG50-8LC are supported.

Note 16: VS32(Dual) indicates VS-32(Dual) Card [SCA-VS32VA-B/SCA-VS32VA] / VS-32(Dual) Box [MG-VS32VA].

Note 17: 6COT indicates MG-COT [SCA-6COTB] / MG-COT [SCA-6COTC].

Note 18: UG50 (COT-TYPE2) [SIP], UG50 (PGT-TYPE2) and UG50-6COT [SIP] are supported.

Note 19: Available since FP95-114 V4 Issue 2.0.

<Interactions with Networks>

1. Encryption for RTP is applied to IP network only; it is not applied to PSTN.
2. RTP packets in CCIS over IP are not encrypted. Even though the speech via CCIS over IP is re-connected as CCIS Link Reconnection under the same node (without CCIS over IP), the RTP packets are not encrypted.

<Conditions Specific to This Feature>

1. An encryption key cannot be viewed by the maintenance personnel and the users.
2. An encryption method for RTP and signaling data is applied with AES mode using 128-bit encryption keys.
3. An encryption key for signaling data is automatically changed at 00:30 (fixed) every day. For manual operation, use the ASECL command to change the encryption key immediately.

Programming

<Data Assignment for VoIP Encryption (Encryption for Voice Data and Signaling Data)>

VoIP Encryption requires the settings for SIP Terminal Authentication [S-152]. For more information about it, refer to SIP Terminal Authentication [S-152].

STEP 1: **ASYDL** - SYS1, Index 831, bit 3: Assign the data to determine a range of encryption for signaling data. **Note 1**

Note 1: If "0" is selected in the above, a part of data is encrypted so that NAT is able to pass the data to next-hop equipment. If "1" is selected, all the data are encrypted, which NAT is not able to handle. Assign "0" when NAT is located between Telephony Server and the SIP Multiple Line terminals.

Note: If the period available for one-time password or the system is initialized, the password needs to be re-assigned when you want to use it.

Note: Assigning a one-time password with no-time-limit might risk your network security. It is recommended that a certain period should be assigned.

Note: Terminals that have completed the Terminal Authentication can be viewed by using the ASECL command.

<Data Assignment for RTP Encryption (Encryption for Voice Data only)>

None.

<Data Assignment for Updating Encryption Key >

To update the key, use the following procedure. This is necessary when the network setting is changed with NAT. For more information, refer to page 2657.

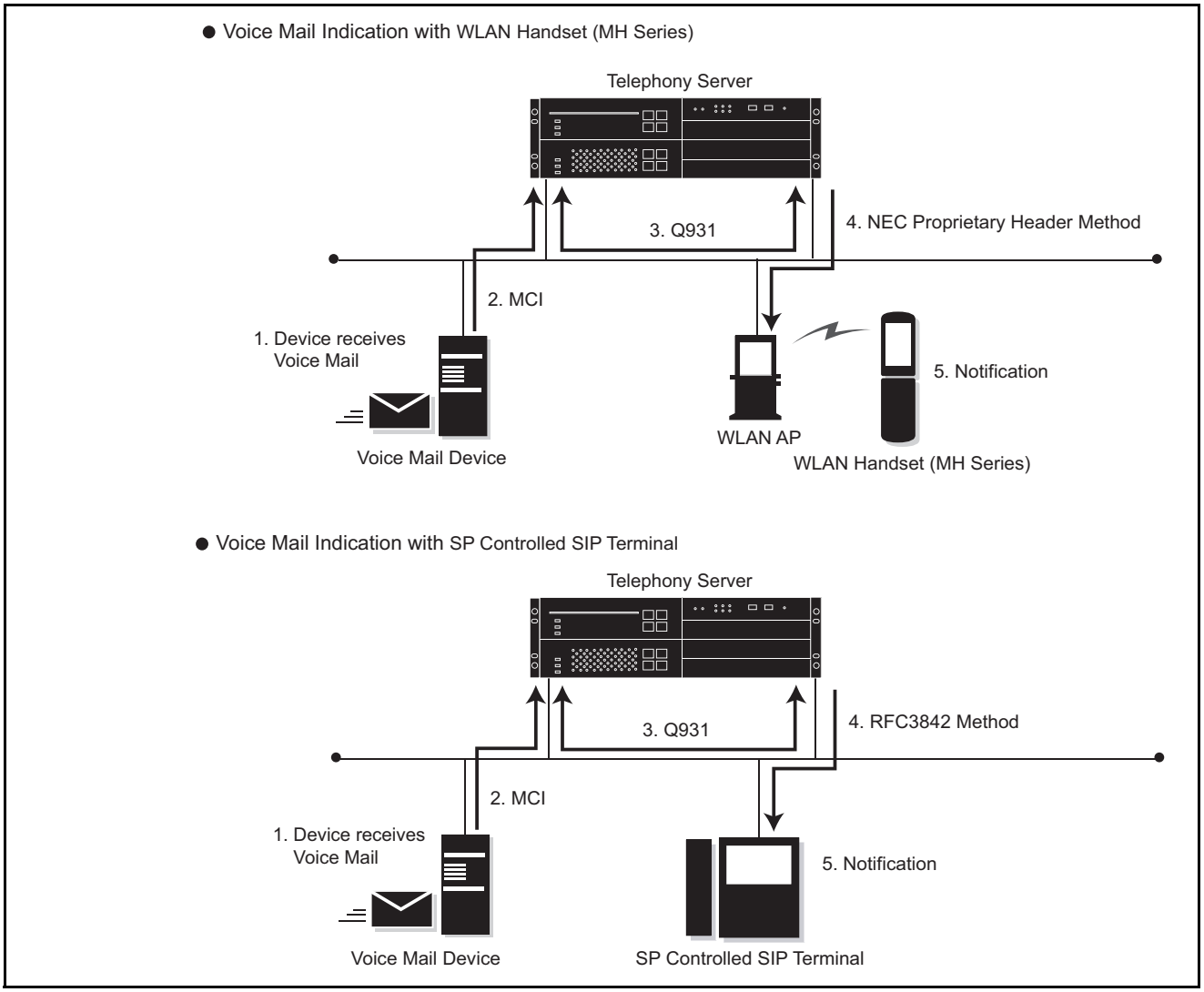
STEP 1: **ASECL** - Click the Maintenance Function tab. Select Cipher Key Update and click the Execute button.

V-29 Voice Mail Indication - SIP

General Description

This feature allows SP Controlled SIP terminals to receive notification of incoming voice mail from Message Center Interface (MCI).

Note: Voice Mail Indication-SIP with WLAN Handset (MH Series) is available.



Operating Procedure

For Voice Mail Indication-SIP with WLAN Handset (MH Series), refer to WLAN Handset (MH Series) User's Guide. For Voice Mail Indication-SIP with SP Controlled Standard SIP terminals, refer to the manual for SP Controlled Standard SIP terminals.

Service Conditions

Conditions for WLAN Handset (MH Series) and SP Controlled Standard SIP terminal

1. This feature is available for WLAN Handsets (MH Series) or SP Controlled Standard SIP terminals compliant with RFC3842, which support voice mail notification.
2. Terminals will be notified immediately after the voice mail device receives a message or all the messages not heard have been cleared.
3. Notification will not be sent to terminals without registration or terminals with expired registration time limits, when the voice mail device receives a message.
4. Notification of numbers of voice mail messages or mail types are not provided.
5. If the terminal fails to receive a MWI notification for any reason such as Out of Zone status, the notification can be provided again when the following events occur (If MWI notification request is within time limit):
   - When REGISTER process is completed
   - When MWI notification request is completed **Note 1**
   - When the terminal finishes a call
   - When the terminal abandons a call
   - When the calling party abandons a call after the call is terminated to the terminal

   Note 1: This event is applicable for Voice Mail Indication-SIP with SP Controlled Standard SIP terminals only.

Conditions for WLAN Handset (MH Series)

1. When WLAN Handset (MH Series) receives the notification of incoming voice mail, an icon appears on the terminal's LCD and LED Ring Indicator flashes green. **Note 2**

   Note 2: Ringing tone is not provided.

Conditions for SP Controlled Standard SIP terminal

1. Behaviors when the SP Controlled Standard SIP terminal receives a MWI notification such as lighting of the Message Waiting Lamp, Indicating the icon, and ringing of notification tone differ among specification of each terminal.
2. Notification will not be sent to terminals with failed MWI notification request or if a time limit of MWI notification request expires in spite of the fact that registration is within the time limit.
3. MWI notification request is canceled when registration is released or when registration time limit expires.
4. Digest Authentication for MWI notification request is not supported.
5. If the system uses SIP Wireless Terminal Backup [S-154] and SP changeover occurs, terminal will not be notified until SP that became Active finishes the MWI notification request.

   Note: "MWI notification request" stated here is the procedure that the SP Controlled Standard SIP terminal requests SP to notify the SP Controlled Standard SIP terminal about MWI information. "MWI notification request" is executed in SUBSCRIBE process.

Programming

The following shows only the data assignment pertaining to this feature. The data of this feature is usually assigned along with SIP Station data. For details on the data assignment of SIP Station, refer to SP CONTROLLED SIP TERMINAL [S-168].

STEP 1: **ASPTL/ASPTN** (SIP Terminal Data)

Note: Selectable parameters appear in DETAIL field when Type is selected.

To activate Voice Mail Indication-SIP with WLAN Handset

TERMINAL field	
TYPE:	Select “Type 3” for WLAN Handset (MH Series).
DETAIL filed	
NO DIGEST AUTHENTICATION:	Mark the checkbox if the terminal does not support.
WIRELESS:	Mark the checkbox when WLAN Handset (MH Series) is used.
SOFTPHONE:	Mark the checkbox when Soft Phone WLAN Handset [S-159] or Soft Phone Wired-Phone Handset [S-147] is activated.
VIDEO TERMINAL:	Mark the checkbox when Standard SIP Video terminal is used.
MWI (Extension header method):	Mark the checkbox to use this feature.

To activate Voice Mail Indication-SIP with SP Controlled Standard SIP Terminal

TERMINAL field	
TYPE:	Select “Type 1” for the SP Controlled Standard SIP terminal.
DETAIL filed	
NO DIGEST AUTHENTICATION:	Mark the checkbox if the terminal does not support.
WIRELESS:	Mark the checkbox when WLAN Handset (MH Series) is used.
SOFTPHONE:	Mark the checkbox when Soft Phone WLAN Handset [S-159] or Soft Phone Wired-Phone Handset [S-147] is activated.
VIDEO TERMINAL:	Mark the checkbox when Standard SIP Video terminal is used.
INVITE with SDP:	Mark the checkbox if the terminal does not support INVITE without SDP. (The terminal must support reINVITE without SDP.) When the checkbox is marked, a dummy SDP will be added to the initial INVITE from the Telephony Server. (This parameter is valid only when TYPE=1.)
When secondary call disconnected, held call remains on hold:	Mark the checkbox when using the following feature: when terminal A transfers a held call (terminal B) to terminal C, terminal B remains in hold state even when terminal A and terminal C are disconnected. (This parameter is valid only when TYPE=1.)
MWI (RFC3842):	Mark the checkbox to use this feature.
RTP Relay BOX:	Not used
Session Timer:	Not used.
Sendonly:	Select the checkbox when your SP Controlled Standard SIP terminal enables Call Hold features supporting RFC3264 a=sendonly method. When this parameter is not selected, the SP Controlled Standard SIP terminal performs Call Hold features supporting RFC2543 c=0 method.

V-30 Volume Set - Receiver

General Description

This feature allows Desktop terminal users to either retain or cancel the setting of handset/headset receiver volume changed during a telephone conversation. It should be noted that DT800 Series/DT700 Series retains the previous setting of receiver volume by default while the other terminals cancel the setting by default.

Operating Procedure

None.

Service Conditions

1. This feature is not available for DtermIP (SIP). DtermIP (SIP) always retains the changes of the receiver volume you have made during a telephone conversation.
2. When you change the setting of the system data (ASYD, SYS1, Index384, Bit 5/ASYDL, SYS1, Index678, Bit 0), you need to perform any of the following.
   - Initialization of ELC card
   - Initialization of the system
   - Initialization of DT800/DT700 Series telephone terminal
3. The setting of receiver volume is canceled when the telephone terminal is reset.
4. For DT800 Series/DT700 Series, this feature supports the control of the handset only.
5. Required system data and the default settings vary depending on telephone type.

Telephone Terminal	Data Setting	After the Changes
Dterm Series i DT400/DT300 Series DtermIP (Proprietary Protocol)	ASYD, SYS1, Index384, Bit 5 0: Previous setting is canceled 1: Previous setting is retained	For digital terminals: Initialization of ELC card For IP terminals: Initialization of the system
DtermIP (SIP)	Not available	-
DT800/DT700 Series	ASYDL, SYS1, Index678, Bit 0 0: Previous setting is retained 1: Previous setting is canceled	Initialization of telephone terminal

Programming

- For Dterm Series i, DT400 Series/DT300 Series and DtermIP (Proprietary Protocol)

  ASYD- System Data 1, Index 384, Bit 5 Retaining the setting of handset/headset receiver volume changed during a call. (0/1 = Out of Service/In Service)

  Note: The setting of receiver volume is canceled when the telephone terminal is reset.

- For DT800 Series/DT700 Series

  ASYDL- System Data 1, Index 678, Bit 0 Retaining the setting of handset receiver volume changed during a call. (0/1 = In Service/Out of Service)

  Note: When a DT800/DT700 Series terminal is reset, whether or not to retain the setting of receiver volume depends on the reset condition. For more details, refer to the User Guide.

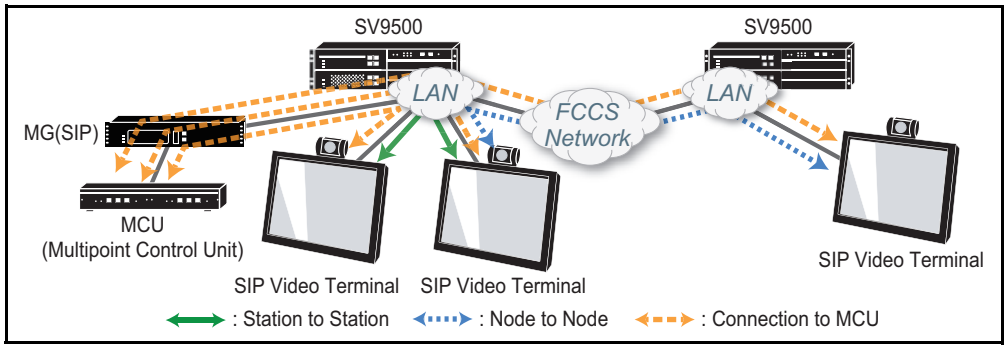
V-31 Video Call

General Description

This feature makes possible several video calls. Video calls can be transmitted using the following connections.

Note: Operations are not guaranteed for other features of SIP video terminal than video calls.

Note: The terminals used must have video transmission capabilities for a video call. A connection test performed by NEC is required before using a Standard SIP video terminal (3rd party product). Also, a test for the combination of the two specific terminals used must be performed.



Note: From FP95-114 V4 video call can be maintained by SIP Handler Controlled Standard SIP video terminals in different nodes (UMGi and FCCS).

Note: MCU (Multipoint Control Unit) allows the connection with several locations for video call.

X: Available, -: Not available

From \ To		SIP Handler Controlled Standard SIP video terminals Note 2				SP Controlled Standard SIP video terminals		MG(SIP) (MCU) Note 1	
		ASYDL SYS1 Index 1382 b2=0 Note 1		ASYDL SYS1 Index 1382 b2=1					
		Station to Station	Node to Node	Station to Station	Node to Node	Station to Station	Node to Node	Station to Station	Node to Node
SIP Handler Controlled Standard SIP video terminals Note 2	ASYDL SYS1 Index 1382 b2=0 Note 1	X	X	-	-	-	-	X	X
	ASYDL SYS1 Index 1382 b2=1	-	-	X	-	-	-	-	-
SP Controlled Standard SIP video terminals		-	-	-	-	X	X	-	-

Note 1: Available from FP95-114 V4.

Note 2: From FP95-114 V4 set ASYDL, SYS1, Index 1382, Bit 2=0.

Operating Procedure

None.

Service Conditions

<Important Notices>

1. VIDEO CALL [V-31] describes service conditions and programming for SIP Handler Controlled Standard SIP video terminals and MG(SIP) when ASYDL, SYS1, Index 1382, Bit 2= 0 (SIP Handler Controlled Standard SIP video terminal usage from SV9500 FP95-114 V4).
   See SIP HANDLER - SIP TERMINAL [S-167] for service conditions and programming for SIP Handler Controlled Standard SIP video terminals when ASYDL, SYS1, Index 1382, Bit 2= 1 (SIP Handler Controlled Standard SIP video terminal usage until SV9500 FP95-113 V3).
   See SP CONTROLLED SIP TERMINAL [S-168] for service conditions and programming for SP Controlled Standard SIP video terminals.
2. Telephony Server does not assign or control the codec for video call. Video call can be performed as long as both SIP Handler Controlled Standard SIP video terminals agree to use the same codec. SIP Handler Controlled Standard SIP video terminals operations when there is not a common codec are defined by each actual SIP Handler Controlled Standard SIP video terminal's model specifications.
3. SIP messages of video call are bigger than the messages of a voice call. If UDP is used for video call, packets may be fragmented and congestion control is not available. For this reason, if there is the risk of UDP packets fragmentation, TCP setting (**Note 3**) is recommended.

   Note 3: When using TCP, make sure that all the terminals employed for video call support TCP. Refer to the manual of each terminal for the TCP settings. See the <Interactions with Other Features> section of SIP HANDLER - SIP TERMINAL [S-167] for the conditions of TCP for Internal SIP Handler.

<Interactions with Other Features>

1. When using PEER TO PEER BANDWIDTH CONTROL [P-75], bandwidth of video call is not counted for the bandwidth calculation. Even if the video call is used as voice only and is using a codec supported by the Telephony Server, bandwidth will not be counted.
2. When MG-SIP SHARED ROUTE [M-123] is used, make sure that all the used MG(SIP) are usable for video call and have the same version. If different versions are used, even if the same access code is used, video call may become not possible depending of the seized trunk.

<Hardware/Software>

1. SIP Handler Controlled Standard SIP video terminals and devices used for video call must be compliant with the following RFCs:
   - RFC 2327
   - RFC 3261

- RFC 3262
- RFC 3264
- RFC 3311
- RFC 3323
- RFC 3515
- RFC 3550
- RFC 3891
- RFC 4028

2. MG(SIP) hardware/software conditions are as follows:
   a.) The following MG(SIP)s are supported:
      - MG-SIP128 [MG-128SIPMGL-A]
      - Software-based MG-SIP with firmware version SP-4085 Issue 1 or later
   b.) The call is connected by the first 128 channels. When all the first 128 channels are in use, a new video call will be converted to a voice call and only voice signals will be transmitted.
   c.) The maximum bandwidth for one channel is two megabytes. (beyond that bandwidth usage, video call service may deteriorate or not be possible).
   d.) When a terminal connected with MG(SIP) is put on hold, MG(SIP) does not send a hold notification to the IP network, but sends a hold tone.
   e.) When a call is sent from a terminal that supports G.711 and G.729a, MG(SIP) chooses first G.711.
   f.) When using MCU, the template pattern 8 setting is required by using "set config_template" command. For the configuration command, refer to the following manuals:
      - MG-SIP128 [MG-128SIPMGL-A]:
        CHAPTER 2 in Peripheral Equipment Description (IP Devices).
      - Software-based MG-SIP with SP-4085:
        Configuration Commands in Chapter 2 of Software Model Peripheral Equipment Description.
   g.) The following are supported on MCU connection:
      - Join a conference (Outgoing call to MCU)
      - Calling to a conference (Incoming call from MCU)
      - Video call during a conference

<Interactions with Networks>

1. In a FCCS system, video call can be used only with FCCS NETWORKING VIA IP [F-36].
2. CCIS NETWORKING VIA IP [C-163] of Data Programming Manual - CCIS cannot be used for a video call. Station-To-Station connection after the Link Reconnection of CCIS networking via IP is also not supported for the video call.
3. Connection between two SIP Handler Controlled Standard SIP video terminals or devices must be Peer-to-Peer.

<Conditions Specific to This Feature>

1. In case there is a failure in the session information inquiry, video call is possible based on the session information (proprietary protocol) (**Note 4**) exchanged during the establishing of the call. If such session information exchanged during the establishing of the call also fails, video call is not possible.

   Note 4: This session information exchange (proprietary protocol) is used for the media information (SDP information) interchange of the SIP Handler Controlled Standard SIP terminal (trough the Internal SIP Handler of the Telephony Server) and the MG(SIP). Session information inquiry is also included in the Session information exchange protocol.

2. The following are conditions for SIP Handler Controlled Standard SIP video terminal
   a.) SIP Handler Controlled Standard SIP video terminal in Multiline Mode does not perform session information inquiry for a video call received to a subline. However, video call is possible based on the session information exchanged during the establishing of the call.
   b.) Video call is not possible using the subline of a desktop terminal that has assigned the my line of a SIP Handler Controlled Standard SIP video terminal. In this case the desktop terminal can only maintain a voice communication.
   c.) According to the session information inquiry for a video call, the call sender notifies the receiver of the desired codecs for video call with an SDP included in the INVITE. However, in the following cases the INVITE message cannot include an SDP:
   - When receiving a blind transfer of REFER of a SIP Handler Controlled Standard SIP video terminal
   - In the recalling after CALL HOLD [C-6] for a long time
   - In the recalling of CALL BACK [C-1]
   - A call received to the subline when being in Multiline Mode

   Note: The reINVITE message send by the Telephony Server when switching the connection path may or may not include an SDP depending on the specific environment and other conditions.

   However, the video call is possible in the cases above with the response of the receiving side based on the session information exchanged during the establishing of the call.

   d.) The DTMF transparency conditions of a video call are the same as the conditions of a voice call. For voice call conditions see <Hardware/Software> section of SIP HANDLER - SIP TERMINAL [S-167].
   RFC2822, SIP-INFO and In-band Signaling can be used but the method is decided depending on the specifications of the actual SIP Handler Controlled Standard SIP video terminal.
   e.) In some cases, if a change in the media information using a reINVITE request is performed during a video call, the SIP Handler Controlled Standard SIP video terminal may reply with an error message regardless of the settings of CDN9 (Reject unacceptable reINVITE) in the AUACL/AUACN command.

   Note: A change in the media information means a change in the codec, etc without changing the call state.

f.) Set 2nd DT: 0 (Second DT is not sent) in the ASPA/ASPAL/ASPAN command for the connection of SIP Handler Controlled Standard SIP video terminals and MG(SIP) when using video call.

g.) When CDN20 (Multimedia Communication) of the AUACL/AUACN command is enabled, CDN3 (send INVITE with SDP) is disabled.

Programming

Note: Video call data assignment is done after finishing the basic programming for the used terminals and devices.
For SIP Handler Controlled SIP Video Terminal, see data programming in SIP HANDLER - SIP TERMINAL [S-167].
For MG(SIP), see data programming in MEDIA GATEWAY SIP [M-111].

<For SIP Handler Controlled SIP Video Terminal>

STEP 1: **ASYDL**- Assign the following.

Index 1394, Bit 0 and 1: Select transport protocol for session information exchange of SIP devices. **Note 5 Note 6**

Transport Protocol	Bit 0	Bit 1
AUTO **Note 7**	0	0
TCP **Note 8**	1	0
UDP	0	1

Note 5: If FCCS system is used, set these bits in the ASYDN to ensure that all nodes have the same settings.

Note 6: In the FCCS system and UMGi system, this setting is valid for session information exchanges of Internal SIP Handlers between nodes. In other combinations, UDP is used.

Note 7: With this setting the packets are sent using UDP, but if the size exceeds 1400 bytes TCP is automatically used.

Note 8: If there is the risk of packet fragmentation when using UDP, set to TCP.

Index 1408, Bit 0: Use TCP for SIP Messages.
0= Disabled
1= Enabled

Note: If this setting is changed, Internal SIP Handler needs to be initialized with the ADPM, ADTM or SINZ command.

Note: Since TCP for big messages is recommended, set this bit to Enabled is also recommended.

STEP 2: **AUACL/AUACN** - Assign the following.

CDN 20 (Multimedia Communication): 1 (Enable)

CDN 21 (Hold Detection at Multimedia): **Note 9**

0: Audio

1: Audio and Video

2: Non-detection

CDN 22 (Transport Layer) **Note 10**

0: AUTO

1: UDP

2: TCP

Note: Verify that ASYDL, SYS1, Index 1382, Bit 2=0.

Note: For video call until FP95-113 V3, see <Video Data Assignment> in the Programming section of SIP HANDLER - SIP TERMINAL [S-167].

Note 9: Please note the following points:

- This setting requires CDN 20 to be set to 1.
- For SIP Handler Controlled Standard SIP video terminals that notify media changes as the voice being muted while the video communication is kept, select pattern 1 or 2.
  For SIP Handler Controlled Standard SIP video terminals that do not notify media changes as the voice being muted, select pattern 0 (default).For a terminal without video capacity, even if 1 is set, operations will be the same as if 0 is set.
- A change in the parameter is applied when there is call sending/receiving.

Note 10: Please note the following points:

- This setting does not decide the transport layer for the terminals. What is set is the protocols used by the SIP Handler of the Telephony Server when assisting the communication between the terminals.
- This setting requires the TCP setting to be enabled. If TCP is not enabled (because system data settings or other reason), TCP for the sending and receiving of SIP messages is not possible.
- This setting is used only if the transport protocol is not set in the transport parameter of the Contact header in the REGISTER request. If the transport protocol is set in the REGISTER, that protocol is used regardless of the setting of CDN22.
- The default setting uses TCP for sending SIP messages of more than 1300 bytes or answering to TCP request messages (regardless of the size of the answering message).
- If UDP is selected, the SIP Handler sends the request messages using UDP (be aware that long messages may become fragmented). However, the SIP Handler is also able to process messages received with TCP and any message received with TCP is also answered with TCP regardless of this parameter setting.
- If TCP is selected, the SIP Handler uses TCP for sending messages but is able to process received messages with UDP.
- A change in the parameter is applied when there is call sending/receiving.

<For MG(SIP)>

STEP 1: **ARTI/ARTIN** - Assign the following.
CDN 110 (Multimedia Communication): 1 (In Service)

Note: Regarding the number plan (LCR/LCRS) of the MG(SIP) sending connection, set 2nd DT: 0 (Second DT is not sent) with the ASPA/ASPAL/ASPAN command.

W-1 WATS Access

General Description

This feature permits any Station user direct-dial access to outgoing Wide Area Telephone Service (WATS) lines.

Note: This feature is available in North America only.

Operating Procedure

To operate

1. Lift the handset. Receive Dial Tone.
2. Dial the WATS Access Code.
3. Receive Second Dial Tone.
4. Dial the desired number.

Service Conditions

Access to WATS lines may be denied to individual Stations via CLASS OF SERVICE-INDIVIDUAL [C-15].

Interactions

1. WATS ACCESS will operate in conjunction with the following features:

LEAST-COST ROUTING-3/6-DIGIT	[L-5]
MISCELLANEOUS Trunk Access	[M-1]
MISCELLANEOUS TRUNK RESTRICTION	[M-2]
INDIVIDUAL Trunk Access	[I-4]
OUTGOING TRUNK QUEUING	[O-2]
OFF-HOOK QUEUING	[O-7]
ROUTE ADVANCE	[R-6]
DIRECT-IN TERMINATION	[D-7]

2. A WATS Access Trunk can be used with any system feature that can be used with any other Trunk. There are no unique characteristics of a WATS Trunk of note in relation to the implementation of other features.

Programming

STEP 1: **ARTD** - Assign the WATS line routes as shown below:

INCOMING WATS LINES (GROUND START):

RT: 1

3-ISGS	:1	4-INSG	:3	5-TF	:2/3	6-TCL	:3	7-L/T	:1
8-RLP	:2	15-LSG	:1						

OUTGOING WATS LINES (GROUND START):

RT: 1

1-OSGS	:2	2-ONSG	:3	5-TF	:1/3	6-TCL	:3	7-L/T	:1
8-RLP	:2	15-LSG	:1						

STEP 2: **ATRK** - Assign the LENS, WATS Line Route Number, Trunk Number, and Tenant Number.

STEP 3: **MBTK** - Assign the Make Idle status to the WATS Line Trunks.

STEP 4: **ANPD** - Reserve a Number Level for Trunk Access. Select Normal Service (N), Hooking Service (H) or Busy Service (B) from the combo box for Connection Indexes. (applicable to any feature assigned to this Level). Number of Necessary Digits is usually one or two. Busy lamp Field is not activated.

STEP 5: **ASPA** - Assign the Access Code as assigned in ANPD above. Type of Service, SRV = “OGC: Outgoing Call” for Outgoing Trunk. Assign the Route Number associated with this Access Code.

STEP 6: **ARSC** - The Route Restriction Class must be constructed to allow the Station access to the involved routes.

W-11 WLAN Handset Name Display

General Description

This feature offers the name display function to WLAN Handset (MH250) on station-to-station calls. Destination station's user name linked to its station number is displayed on the LCD of WLAN Handset.

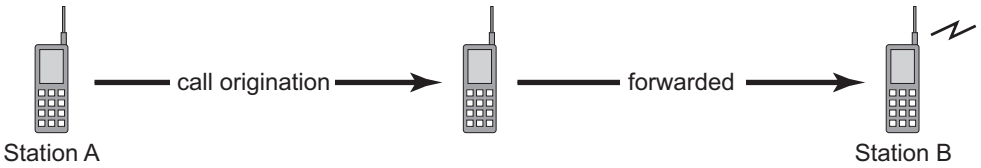
- On call origination:
  Called destination's user name is displayed on the calling WLAN Handset.
- On call termination:
  Caller station's user name is displayed on the called WLAN Handset.
- During talking:
  Connected station's user name is displayed on the WLAN Handset.
- On transferred/forwarded call:
  Transferred/Forwarded destination's user name is displayed on the calling WLAN Handset.

<CASE 1>
When Station A dials Station B's number:



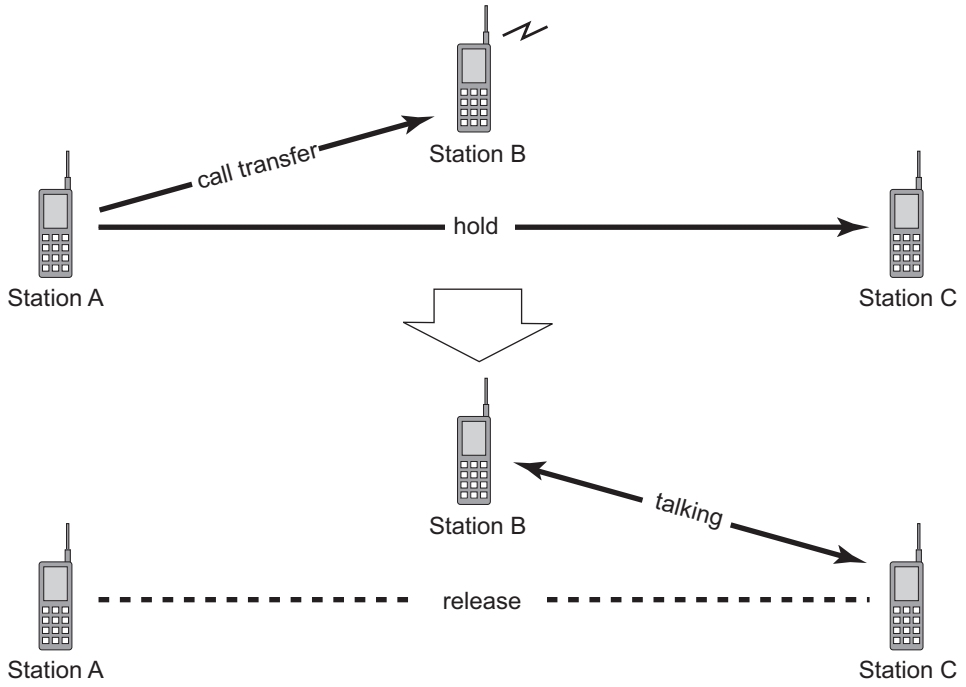
	Station A	Station B
Assigned user name	N.TARO	N.HANAKO
State	Receiving RBT	Ringing
LCD display	N.HANAKO	N.TARO

<CASE 2>
When Station A's call is forwarded to Station B (by Call Forwarding-All Calls/Busy/Don't Answer/Incoming Incomplete):



	Station A	Station B
Assigned user name	N.TARO	N.HANAKO
State	Being forwarded	Ringing
LCD display	CF:N.HANAKO	CF:N.TARO

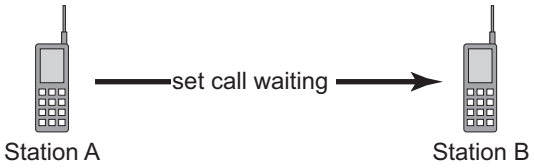
<CASE 3>
When Station A holds Station C and then transfers the call to Station B (Call Transfer/Blind Transfer):



	Station A	Station B	Station C
Assigned user name	N.TARO	N.HANAKO	F.ICHIRO
State	Receiving RBT	Ringing	Held
LCD display	N.HANAKO	N.TARO	N.TARO

	Station A	Station B	Station C
Assigned user name	N.TARO	N.HANAKO	F.ICHIRO
State	Released	Connected to Station C	Connected to Station B
LCD display	-	F.ICHIRO	N.HANAKO

<CASE 4>
When Station A sets Call Waiting to busy Station B, and then Station B answers the Call Waiting:



	Station A	Station B
Assigned user name	N.TARO	N.HANAKO
State	Receiving RBT	Answers the Call Waiting
LCD display	N.HANAKO	N.TARO

Operating Procedure

None.

Service Conditions

1. This feature is supported on MH250 stations only.

   Note: Except MH250, name display does not work on SIP terminals (Standard SIP Terminal, DtermIP (SIP)). However, these terminals can register user name, so their user name is displayed on the connected MH250 station's LCD.

2. Up to 16 digits of user name can be registered, excluding the “CF:” characters added for the forwarded calls by Call Forwarding - All Calls, Busy, Don’t Answer and Incoming Incomplete.
3. Alphanumeric characters and symbols are available to display.
4. When no user name is assigned to station number, existing Number Display features are applied.
5. This feature is not available for the calls via PHC.
6. If the last digit of user name assigned with the ANDD command is a blank space, MH250 station’s LCD will not display that blank space allocated as the last digit.

Interactions

1. This feature is supported on STATION-TO-STATION CALLS, CALL FORWARDING - ALL CALLS/ BUSY/DON’T ANSWER/INCOMING INCOMPLETE, CALL WAITING, BLIND TRANSFER, and CALL TRANSFER - ALL CALLS.
2. “CNF” is displayed on Three-Way Calling calls. When the Three-Way Calling call returns to station-to-station call, LCD displays the connected destination’s user name.
3. Name Display for OAI system takes precedence over this feature.

Programming

STEP 1: **ASYD** - System Data 1, Index 31 = 04 HEX (4 Mbyte of Common Memory (CM) mounting capacity)
System Data 1, Index 78, Bit 5 = 1 (Name display is in service)
System Data 1, Index 79, Bit 6 = 0 (OAI is in service)
System Data 1, Index 241, Bit 1. The number of characters for name display: 0/1 = 8 digits/16 digits

STEP 2: **ANDD** - Assign the name display data.
TN: Tenant Number
STN: Station Number
NAME: Name information

APPENDIX

APPENDIX A Tone Specifications for Each Country

This appendix shows the tone specifications for each country.

Note: This feature is used only for EMEA.

(1) Netherlands

Name	Signal Frequency (Hz)/ Sensitivity Level	Duration of Connection
Internal Ring Tone	425 Hz/-6 dBm	1s-on, 4s-off Repeating.
External Ring Tone	425 Hz/-6 dBm	1s-on, 4s-off Repeating.
Internal Dial Tone	425 Hz/-6 dBm	Continuous
Enquiry Dial Tone	425 Hz/-6 dBm	Continuous
First External Dial Tone	425 Hz/-6 dBm	Continuous
Busy Tone	425 Hz/-6 dBm	500ms-on, 500ms-off Repeating
Number Unobtainable Tone	425 Hz/ -6 dBm	50ms-on, 50ms-off Repeating
Congestion Tone	425 Hz/ -6 dBm	250ms-on, 250ms-off Repeating
Diversion Active Dial Tone	425 Hz/ -6 dBm	350ms-on, 25ms-off Repeating
Confirmation Tone	425 Hz/ -6 dBm	Continuous
Call Waiting Tone Burst	425 Hz/ -6 dBm	96ms-on, off
Camp-on Busy Tone to Caller	425 Hz/ -6 dBm	1s-on, 4s-off Repeating
Hold Tone	425 Hz/ -6 dBm	1s-on, 4s-off Repeating
Park Tone	425 Hz/ -6 dBm	50ms-on, 50ms-off Repeating
Routing Tone	425 Hz/ -6 dBm	5 times 75ms on/ 75ms off, followed by one time 75ms on, 425ms off - repeating
Break-in Ticker	425 Hz/ -3 dBm	10 times 32ms-on/480ms-off Repeating every 32ms **Note 1**
Add-on Ticker	425 Hz/ -3 dBm	10 times 32ms-on/480ms-off Repeating every 32ms **Note 1**
Howler Tone	2000 Hz/ +3 dBm	32ms-on, 32ms-off Repeating

Note 1: For repeating interval, time delay may occur. (maximum 4 seconds)

(2) Germany

Name	Signal Frequency (Hz)/ Sensitivity Level	Duration of Connection
Internal Ring Tone	425 Hz/-3 dBm	1s-on, 4s-off Repeating.
External Ring Tone	425 Hz/-3 dBm	1s-on, 4s-off Repeating.
Internal Dial Tone	425 Hz/-3 dBm	250ms-on, 250ms-off, 250ms-on, 250ms-off, 250ms-on, 1.25s-off Repeating.
Enquiry Dial Tone	425 Hz/-3 dBm	250ms-on, 250ms-off, 250ms-on, 250ms-off, 250ms-on, 1.25s-off Repeating.
First External Dial Tone	425 Hz/-9 dBm	Continuous
Busy Tone	425 Hz/-3 dBm	150ms-on, 475ms-off Repeating
Number Unobtainable Tone	425 Hz/-3 dBm	250ms-on, 250ms-off Repeating.
Congestion Tone	425 Hz/-3 dBm	250ms-on, 250ms-off Repeating.
Diversion Active Dial Tone	150 Hz + 450 Hz/-6 dBm + -15 dBm	250ms-on, 250ms-off, 250ms-on, 250ms-off, 250ms-on, 1.25s-off Repeating.
Confirmation Tone	425 Hz/-9 dBm	3488ms-on, off
Call Waiting Tone Burst	425 Hz/-9 dBm	992ms-on, off
Camp-on Busy Tone to Caller	425 Hz/-9 dBm	1s-on, 4s-off Repeating.
Hold Tone	425 Hz/-9 dBm	1s-on, 4s-off Repeating.
Park Tone	425 Hz/-3 dBm	250ms-on, 250ms-off Repeating.
Routing Tone	425 Hz/-6 dBm	7 times 75ms-on/75ms off, followed by one time 75ms on, 875ms - off Repeating
Break-in Ticker	425 Hz/-3 dBm	10 times 32ms-on/480ms-off-repeating every 32s **Note 2**
Add-on Ticker	425 Hz/-3 dBm	10 times 32ms-on/480ms-off-repeating every 32s **Note 2**
Howler Tone	2000 Hz/ +3 dBm	32ms-on, 32ms-off Repeating

Note 2: For repeating interval, time delay may occur. (maximum 4 seconds)

(3) Austria

Name	Signal Frequency (Hz)/ Sensitivity Level	Duration of Connection
Internal Ring Tone	425 Hz/-3 dBm	1s-on, 5s-off Repeating
External Ring Tone	425 Hz/-3 dBm	1s-on, 5s-off Repeating
Internal Dial Tone	425 Hz/-3 dBm	150ms-on, 150ms-off, 150ms-on, 150ms-off, 150ms-on, 750s-off, Repeating
Enquiry Dial Tone	425 Hz/-3 dBm	150ms-on, 150ms-off, 150ms-on, 150ms-off, 150ms-on, 750s-off, Repeating
First External Dial Tone	425 Hz/-3 dBm	Continuous
Busy Tone	425 Hz/-3 dBm	400ms-on, 400ms-off Repeating
Number Unobtainable Tone	425 Hz/-3 dBm	50ms-on, 50ms-off Repeating
Congestion Tone	425 Hz/-3 dBm	400ms-on, 400ms-off Repeating
Diversion Active Dial Tone	425 Hz/-3 dBm	375ms-on, 25ms-off Repeating
Confirmation Tone	425 Hz/-3 dBm	Continuous
Call Waiting Tone Burst	425 Hz/-3 dBm	192ms-on, 192ms-off, 192ms-on, off
Camp-on Busy Tone to Caller	150 Hz+450 Hz/-6 dBm + -15 dBm	1s-on, 5s-off Repeating.
Hold Tone	150 Hz+450 Hz/-6 dBm + -15 dBm	1s-on, 5s-off Repeating.
Park Tone	425 Hz/-3 dBm	50ms-on, 50ms-off Repeating.
Routing Tone	425 Hz/-6 dBm	7 times 75ms-on/75ms off, followed by one time 75ms on, 875ms - off Repeating
Break-in Ticker	425 Hz/-3 dBm	4 times 32ms-on/960ms-off - repeating every 4s **Note 3**
Add-on Ticker	425 Hz/-3 dBm	4 times 32ms-on/960ms-off - repeating every 4s **Note 3**
Howler Tone	2000 Hz/ +3 dBm	32ms-on, 32ms-off Repeating

Note 3: For repeating interval, time delay may occur. (maximum 4 seconds)

(4) Italy

Name	Signal Frequency (Hz)/ Sensitivity Level	Duration of Connection
Internal Ring Tone	425 Hz/-6 dBm	1s-on, 4s-off Repeating
External Ring Tone	425 Hz/-6 dBm	1s-on, 4s-off Repeating
Internal Dial Tone	350 Hz + 450 Hz/ -9 dBm + -9 dBm	Continuous
Enquiry Dial Tone	350 Hz + 450 Hz/ -9 dBm + -9 dBm	Continuous
First External Dial Tone	425 Hz/-6 dBm	600ms-on, 1.075s-off 200ms-on, 200ms-off, 600ms-on, 1.075s-off Repeating
Busy Tone	425 Hz/-6 dBm	500ms-on, 500ms-off Repeating
Number Unobtainable Tone	425 Hz/-6 dBm	100ms-on, 100ms-off Repeating
Congestion Tone	425 Hz/-6 dBm	500ms-on, 500ms-off Repeating
Diversion Active Dial Tone	350 Hz + 440 Hz/ -9 dBm + -9 dBm	350ms-on, 25ms-off Repeating
Confirmation Tone	425 Hz/-6 dBm	96ms-on, 96ms-off, 96ms-on, 96ms-off, 96ms-on, 1504 ms-off - Repeating 3 times, then silence
Call Waiting Tone Burst	425 Hz/-6 dBm	96ms-on, 4896ms-off Repeating **Note 4**
Camp-on Busy Tone to Caller	425 Hz/-6 dBm	200ms-on, 200ms-off, 200ms-on, 3.15s-off Repeating
Hold Tone	425 Hz/-6 dBm	200ms-on, 200ms-off, 200ms-on, 3.15s-off Repeating
Park Tone	425 Hz/-6 dBm	100ms-on, 100ms-off Repeating
Routing Tone	425 Hz/-6 dBm	250ms-on, 500ms-off, 250ms on, 4s-off Repeating
Break-in Ticker	425 Hz/-3 dBm	10 times 32ms-on/480ms-off - repeating every 32s **Note 4**
Add-on Ticker	425 Hz/-3 dBm	10 times 32ms-on/480ms-off - repeating every 32s **Note 4**
Howler Tone	2000 Hz/ +3 dBm	32ms-on, 32ms-off Repeating

Note 4: For repeating interval, time delay may occur. (maximum 4 seconds)

(5) Belgium

Name	Signal Frequency (Hz)/ Sensitivity Level	Duration of Connection
Internal Ring Tone	450 Hz/-4 dBm	1s-on, 3s-off Repeating
External Ring Tone	450 Hz/-4 dBm	1s-on, 3s-off Repeating
Internal Dial Tone	450 Hz/ -4 dBm	Continuous
Enquiry Dial Tone	450 Hz/ -4 dBm	Continuous
First External Dial Tone	450 Hz/-4 dBm	Continuous
Busy Tone	450 Hz/-4 dBm	500ms-on, 500ms-off Repeating
Number Unobtainable Tone	450 Hz/-4 dBm	200ms-on, 200ms-off Repeating
Congestion Tone	450 Hz/-4 dBm	200ms-on, 200ms-off Repeating
Diversion Active Dial Tone	450 Hz/ -4 dBm	1s-on, 250ms-off Repeating
Confirmation Tone	450 Hz/-4 dBm	480ms-on/480ms-off Repeating
Call Waiting Tone Burst	450 Hz/-4 dBm	256ms-on - repeating every 4s **Note 5**
Camp-on Busy Tone to Caller	450 Hz/-4 dBm	1s-on, 3s-off Repeating
Hold Tone	450 Hz/-4 dBm	1s-on, 3s-off Repeating
Park Tone	450 Hz/-4 dBm	200ms-on, 200ms-off Repeating
Routing Tone	450 Hz/-4 dBm	Continuous
Break-in Ticker	425 Hz/-3 dBm	5 times 32ms-on/480ms-off - repeating every 4s **Note 5**
Add-on Ticker	425 Hz/-3 dBm	5 times 32ms-on/480ms-off - repeating every 32s **Note 5**
Howler Tone	2000 Hz/ +3 dBm	32ms-on, 32ms-off Repeating

Note 5: For repeating interval, time delay may occur. (maximum 4 seconds)

(6) Denmark

Name	Signal Frequency (Hz)/ Sensitivity Level	Duration of Connection
Internal Ring Tone	425 Hz/-3 dBm	1s-on, 4s-off Repeating
External Ring Tone	425 Hz/-3 dBm	1s-on, 4s-off Repeating
Internal Dial Tone	400 Hz/-9 dBm	Continuous
Enquiry Dial Tone	400 Hz/-9 dBm	Continuous
First External Dial Tone	425 Hz/-3 dBm	Continuous
Busy Tone	425 Hz/-3 dBm	250ms-on, 250ms-off Repeating
Number Unobtainable Tone	425 Hz/-6 dBm	50ms-on, 50ms-off Repeating
Congestion Tone	425 Hz/-3 dBm	100ms-on, 100ms-off Repeating
Diversion Active Dial Tone	400 Hz/-9 dBm	350ms-on, 25ms-off Repeating
Confirmation Tone	425 Hz/-3 dBm	992ms-on/256ms-off Repeating
Call Waiting Tone Burst	425 Hz/-3 dBm	64ms-on, 96ms-off, 64ms-on, repeating every 4s **Note 6**
Camp-on Busy Tone to Caller	33.33 Hz x 400 Hz -17 dBm (suppressed carrier)	200ms-on, 200ms-off, 200ms-on, 200ms-off, 200ms-on, 9s-off Repeating
Hold Tone	33.33 Hz x 400 Hz -17 dBm (suppressed carrier)	200ms-on, 200ms-off, 200ms-on, 200ms-off, 200ms-on, 9s-off Repeating
Park Tone	425 Hz/-6 dBm	50ms-on, 50ms-off Repeating
Routing Tone	33.33 Hz x 400 Hz -17 dBm (suppressed carrier)	1s-on, 4s-off Repeating
Break-in Ticker	425 Hz/-3 dBm	5 times 32ms-on/480ms-off - repeating every 4s **Note 6**
Add-on Ticker	425 Hz/-3 dBm	2 times 32ms-on/480ms-off - repeating every 32s **Note 6**
Howler Tone	2000 Hz/ +3 dBm	32ms-on, 32ms-off Repeating

Note 6: For repeating interval, time delay may occur. (maximum 4 seconds)

(7) Sweden

Name	Signal Frequency (Hz)/ Sensitivity Level	Duration of Connection
Internal Ring Tone	425 Hz/-9 dBm	1s-on, 5s-off Repeating
External Ring Tone	425 Hz/-9 dBm	1s-on, 5s-off Repeating
Internal Dial Tone	425 Hz/-9 dBm	Continuous
Enquiry Dial Tone	425 Hz/-9 dBm	Continuous
First External Dial Tone	425 Hz/-9 dBm	Continuous
Busy Tone	425 Hz/-9 dBm	250ms-on, 250ms-off Repeating
Number Unobtainable Tone	425 Hz/-6 dBm	50ms-on, 50ms-off Repeating
Congestion Tone	425 Hz/-9 dBm	250ms-on, 750ms-off Repeating
Diversion Active Dial Tone	425 Hz/-9 dBm	350ms-on, 25ms-off Repeating
Confirmation Tone	425 Hz/-9 dBm	1504ms-on/512ms-off Repeating
Call Waiting Tone Burst	425 Hz/-9 dBm	192ms-on, 512ms-off, 192ms-on, off
Camp-on Busy Tone to Caller	425 Hz/-9 dBm	1s-on, 5s-off, Repeating
Hold Tone	425 Hz/-9 dBm	1s-on, 5s-off, Repeating
Park Tone	425 Hz/-6 dBm	50ms-on, 50ms-off Repeating
Routing Tone	425 Hz/-9 dBm	1s-on, 5s-off Repeating
Break-in Ticker	1400 Hz/-27 dBm	2 times 96ms-on/1504ms-off - repeating every 4s **Note 7**
Add-on Ticker	1400 Hz/-27 dBm	352ms-on, off - repeating every 32s **Note 7**
Howler Tone	2000 Hz/ +3 dBm	32ms-on, 32ms-off Repeating

Note 7: For repeating interval, time delay may occur. (maximum 4 seconds)

(8) Swiss

Name	Signal Frequency (Hz)/ Sensitivity Level	Duration of Connection
Internal Ring Tone	425 Hz/-9 dBm	1s-on, 4s-off Repeating
External Ring Tone	425 Hz/-9 dBm	1s-on, 4s-off Repeating
Internal Dial Tone	425 Hz/-9 dBm	Continuous
Enquiry Dial Tone	425 Hz/-9 dBm	Continuous
First External Dial Tone	425 Hz/-9 dBm	Continuous
Busy Tone	425 Hz/-9 dBm	500ms-on, 500ms-off Repeating
Number Unobtainable Tone	425 Hz/-9 dBm	200ms-on, 200ms-off Repeating
Congestion Tone	425 Hz/-9 dBm	200ms-on, 200ms-off Repeating
Diversion Active Dial Tone	425 Hz (-9 dBm) + 340 Hz (-14 dBm)	975ms-on, 25ms-off Repeating
Confirmation Tone	425 Hz/-19 dBm	800ms-on/992ms-off Repeating
Call Waiting Tone Burst	425 Hz/-19 dBm	192ms-on, 192ms-off, 192ms-on Repeating every 4s **Note 8**
Camp-on Busy Tone to Caller	425 Hz/-9 dBm	1s-on, 3.75s-off, 500ms-on, 500ms-off, 500ms-on, 3.75s-off Repeating
Hold Tone	425 Hz/-9 dBm	1s-on, 3.75s-off, 500ms-on, 500ms-off, 500ms-on, 3.75s-off Repeating
Park Tone	425 Hz/-9 dBm	200ms-on, 200ms-off Repeating
Routing Tone	425 Hz/-6 dBm	7 times 75ms on/off, followed by one time 75 ms on, 875ms off Repeating
Break-in Ticker	425 Hz/-23 dBm	2 times 192ms-on/1792ms-off - repeating every 4s **Note 8**
Add-on Ticker	425 Hz/-23 dBm	448ms-on - repeating every 32s **Note 8**
Howler Tone	2000 Hz/ +3 dBm	32ms-on, 32ms-off Repeating

Note 8: For repeating interval, time delay may occur. (maximum 4 seconds)

(9) UK

Name	Signal Frequency (Hz)/ Sensitivity Level	Duration of Connection
Internal Ring Tone	33.33 Hz x 400 Hz -17 dBm (suppressed carrier)	1s-on, 3s-off Repeating
External Ring Tone	33.33 Hz x 400 Hz -17 dBm (suppressed carrier)	1s-on, 3s-off Repeating
Internal Dial Tone	33.33 Hz x 400 Hz -17 dBm (suppressed carrier)	Continuous
Enquiry Dial Tone	33.33 Hz x 400 Hz -17 dBm (suppressed carrier)	Continuous
First External Dial Tone	33.33 Hz x 400 Hz -17 dBm (suppressed carrier)	Continuous
Busy Tone	400 Hz/-12 dBm	375ms-on, 375ms-off Repeating
Number Unobtainable Tone	400 Hz/-12 dBm	Continuous
Congestion Tone	400 Hz/-12 dBm	250ms-on, 250ms-off Repeating
Diversion Active Dial Tone	33.33 Hz x 400 Hz -17 dBm (suppressed carrier)	350ms-on, 25ms-off Repeating
Confirmation Tone	33.33 Hz x 400 Hz -17 dBm (suppressed carrier)	Continuous
Call Waiting Tone Burst	33.33 Hz x 400 Hz -17 dBm (suppressed carrier)	512ms-on, off
Camp-on Busy Tone to Caller	33.33 Hz x 400 Hz -17 dBm (suppressed carrier)	1s-on, 3s-off Repeating
Hold Tone	33.33 Hz x 400 Hz -17 dBm (suppressed carrier)	1s-on, 3s-off Repeating
Park Tone	400 Hz/-12 dBm	Continuous
Routing Tone	400 Hz/-12 dBm	250ms-on, 2.75ms-off Repeating
Break-in Ticker	425 Hz/-3 dBm	2 times 32ms-on/480ms-off - repeating every 4s **Note 9**
Add-on Ticker	425 Hz/-3 dBm	2 times 32ms-on/480ms-off - repeating every 4s **Note 9**
Howler Tone	2000 Hz/ +3 dBm	32ms-on, 32ms-off Repeating

Note 9: For repeating interval, time delay may occur. (maximum 4 seconds)

(10) Spain

Name	Signal Frequency (Hz)/ Sensitivity Level	Duration of Connection
Internal Ring Tone	425 Hz -12 dBm	300ms-on, 400ms-off, 300ms-on, 4s-off Repeating
External Ring Tone	425 Hz -12 dBm	300ms-on, 400ms-off, 300ms-on, 4s-off Repeating
Internal Dial Tone	425 Hz -9 dBm	Continuous
Enquiry Dial Tone	425 Hz -9 dBm	Continuous
First External Dial Tone	425 Hz x 25Hz -10 dBm (suppressed carrier)	Continuous
Busy Tone	425 Hz/-12 dBm	500ms-on, 500ms-off Repeating
Number Unobtainable Tone	425 Hz/-12 dBm	50ms-on, 50ms-off Repeating
Congestion Tone	425 Hz/-9 dBm	250ms-on, 250ms-off Repeating
Diversion Active Dial Tone	425 Hz/-9 dBm	350ms-on, 25ms-off Repeating
Confirmation Tone	425 Hz/-9 dBm	Continuous
Call Waiting Tone Burst	425 Hz/-9 dBm	2 times 256ms-on/256ms-off - repeating every 4s **Note 10**
Camp-on Busy Tone to Caller	425 Hz/-9 dBm	1s-on, 4s-off Repeating
Hold Tone	425 Hz/-9 dBm	1s-on, 4s-off Repeating
Park Tone	425 Hz/-12 dBm	50ms-on, 50ms-off Repeating
Routing Tone	425 Hz/-6 dBm	7 times 75ms on/off, followed by one time 75ms on, 875ms off - repeating
Break-in Ticker	425 Hz/-3 dBm	5 times 32ms-on/480ms-off - repeating every 4s **Note 10**
Add-on Ticker	425 Hz/-3 dBm	5 times 32ms-on/480ms-off - repeating every 32s **Note 10**
Howler Tone	2000 Hz/ +3 dBm	32ms-on, 32ms-off Repeating

Note 10: For repeating interval, time delay may occur. (maximum 4 seconds)

(11) Greece

Name	Signal Frequency (Hz)/ Sensitivity Level	Duration of Connection
Internal Ring Tone	425 Hz/-3 dBm	1s-on, 4s-off Repeating
External Ring Tone	425 Hz/-3 dBm	1s-on, 4s-off Repeating
Internal Dial Tone	425 Hz/-3 dBm	Continuous
Enquiry Dial Tone	425 Hz/-3 dBm	Continuous
First External Dial Tone	450 Hz/0 dBm	600ms-on, 1s-off 200ms-on, 200ms-off
Busy Tone	425 Hz/-3 dBm	500ms-on, 500ms-off Repeating
Number Unobtainable Tone	425 Hz/-3 dBm	200ms-on, 200ms-off Repeating
Congestion Tone	425 Hz/-3 dBm	200ms-on, 200ms-off Repeating
Diversion Active Dial Tone	25 Hz x 425 Hz/-10 dBm (suppressed carrier)	Continuous
Confirmation Tone	425 Hz/-3 dBm	3488ms-on, off
Call Waiting Tone Burst	425 Hz/-3 dBm	960ms-on, off
Camp-on Busy Tone to Caller	25 Hz x 425 Hz/-10 dBm (suppressed carrier)	200ms-on, 200ms-off 200ms-on, 1.9s-off Repeating
Hold Tone	25 Hz x 425 Hz/-10 dBm (suppressed carrier)	200ms-on, 200ms-off 200ms-on, 1.9s-off Repeating
Park Tone	425 Hz/-3 dBm	200ms-on, 200ms-off Repeating
Routing Tone	425 Hz/-6 dBm	7 times 75ms on/off, followed by one time 75ms on, 875ms off - Repeating
Break-in Ticker	950 Hz/-13 dBm	480ms-on, off - repeating every 4s **Note 11**
Add-on Ticker	950 Hz/-13 dBm	480ms-on, off - repeating every 32s **Note 11**
Howler Tone	2000 Hz/ +3 dBm	32ms-on, 32ms-off Repeating

Note 11: For repeating interval, time delay may occur. (maximum 4 seconds)

(12) South Africa

Name	Signal Frequency (Hz)/ Sensitivity Level	Duration of Connection
Internal Ring Tone	33.33 Hz x 400 Hz/ -3 dBm (suppressed carrier)	1s-on, 2s-off Repeating
External Ring Tone	33.33 Hz x 400 Hz/ -3 dBm (suppressed carrier)	1s-on, 2s-off Repeating
Internal Dial Tone	400 Hz/ -9dBm	Continuous
Enquiry Dial Tone	400 Hz/ -9dBm	Continuous
First External Dial Tone	33.33 Hz x 400 Hz	Continuous
Busy Tone	400 Hz/ -9 dBm	500ms-on, 500ms-off Repeating
Number Unobtainable Tone	400 Hz/ -9 dBm	2.5s-on, 500ms-off, Repeating
Congestion Tone	400 Hz/ -9 dBm	250ms-on, 250ms-off, Repeating
Diversion Active Dial Tone	33.33 Hz x 400 Hz/ -3 dBm (suppressed carrier)	1s-on, 200ms-off Repeating
Confirmation Tone	400 Hz/ -9dBm	608ms-on,192ms-off, Repeating
Call Waiting Tone Burst	400 Hz/ -9dBm	256ms-on, off repeating every 4s
Camp-on Busy Tone to Caller	33.33 Hz x 400 Hz/ -3 dBm (suppressed carrier)	400ms-on, 200ms-off Repeating
Hold Tone	33.33 Hz x 400 Hz/ -3 dBm (suppressed carrier)	400ms-on, 200ms-off Repeating
Park Tone	400 Hz/ -9 dBm	2.5s-on, 500ms-off, Repeating
Routing Tone	425 Hz/ -6dBm	7 times 75ms-on/75ms-off, followed by 1 time 75ms-on/875ms-off, Repeating
Break-in Ticker	425 Hz/ -3 dBm	5 times 32ms-on/480ms-off, repeating every 4s **Note 12**
Add-on Ticker	425 Hz/ -3 dBm	4 times 32ms-on/480ms-off **Note 12**
Howler Tone	<PCM Howler> 2000Hz/ +3 dBm <Power Howler> 800Hz±150Hz/ +23 dBm	<PCM Howler> 32ms-on,32ms-off Repeating <Power Howler> Continuous (Max.30s)

Note 12: For repeating interval, time delay may occur. (maximum 4 seconds)

APPENDIX B Tone Setting Differences between EMEA and Basic Tone Setting for EMEA

This appendix shows the differences of tone specification between EMEA and Basic Tone Setting for EMEA.

(a) Netherlands

Name	EMEA		Basic Tone Setting for EMEA	
	Signal Frequency (Hz)/Sensitivity Level (dBm)	Duration of Connection	Signal Frequency (Hz)/ Sensitivity Level (dBm)	Duration of Connection
Enquiry Dial Tone	425 Hz/-9 dBm	Continuous	425 Hz/- 6 dBm	Continuous
Number Unobtainable Tone	425 Hz/ -6 dBm	60ms-on, 60ms-off Repeating	425 Hz/ -6 dBm	50ms-on, 50ms-off Repeating
Diversion Active Dial Tone	425 Hz/ -6 dBm	320ms-on, 10ms-off Repeating	425 Hz/ -6 dBm	350ms-on, 25ms-off Repeating
Call Waiting Tone Burst	425 Hz/ -6 dBm	100ms-on, off	425 Hz/ -6 dBm	96ms-on, off
Routing Tone	425 Hz/ -6 dBm	5 times 80ms on/ 80ms off, followed by one time 80ms on, 500ms off - Repeating	425 Hz/ -6 dBm	5 times 75ms on/ 75ms off, followed by one time 75ms on, 425ms off - Repeating
Break-in Ticker	425 Hz/ -3 dBm	Continuous	425 Hz/ -3 dBm	10 times 32ms-on/ 480ms-off - Repeating every 32s
Add-on Ticker	425 Hz/ -3 dBm	4s-on, 30s-off Repeating	425 Hz/ -3 dBm	10 times 32ms-on/ 480ms-off Repeating every 32ms
Howler Tone	1800 Hz/ -13 dBm	250ms-on, 250ms-off Repeating	2000 Hz/ +3 dBm	32ms-on, 32ms-off Repeating

(b) Germany

Name	EMEA		Basic Tone Setting for EMEA (This feature)	
	Signal Frequency (Hz)/ Sensitivity Level (dBm)	Duration of Connection	Signal Frequency (Hz)/ Sensitivity Level (dBm)	Duration of Connection
Internal Dial Tone	425 Hz/-3 dBm	240ms-on, 240ms-off, 240ms-on, 240ms-off, 240ms-on, 1.4s-off Repeating	425 Hz/- 3 dBm	250ms-on, 250ms-off, 250ms-on, 250ms-off, 250ms-on, 1.25s-off Repeating
Enquiry Dial Tone	425 Hz/ -3 dBm	240ms-on, 240ms-off, 240ms-on, 240ms-off, 240ms-on, 1.4s-off Repeating	425 Hz/ -3 dBm	250ms-on, 250ms-off, 250ms-on, 250ms-off, 250ms-on, 1.25s-off Repeating
Busy Tone	425 Hz/ -3 dBm	160ms-on, 480ms-off Repeating	425 Hz/ -3 dBm	150ms-on, 475ms-off Repeating
Number Unobtainable Tone	425 Hz/ -3 dBm	240ms-on, 240ms-off Repeating	425 Hz/ -3 dBm	250ms-on, 250ms-off Repeating
Congestion Tone	425 Hz/ -3 dBm	240ms-on, 240ms-off Repeating	425 Hz/ -3 dBm	250ms-on, 250ms-off Repeating
Diversion Active Dial Tone	150 Hz+450 Hz -6 dBm + -15 dBm	240ms-on, 240ms-off, 240ms-on, 240ms-off, 240ms-on, 1.4s-off Repeating	150 Hz+450 Hz -6 dBm + -15 dBm	250ms-on, 250ms-off, 250ms-on, 250ms-off, 250ms-on, 1.25s-off Repeating
Confirmation Tone	425 Hz/ -3 dBm	3.5s-on, off	425 Hz/ -9 dBm	3488ms-on, off
Call Waiting Tone Burst	425 Hz/ -9 dBm	1s-on, off	425 Hz/ -9 dBm	992ms-on, off
Park Tone	425 Hz/ -3dBm	160ms-on, 480ms-off Repeating	425 Hz/ -3dBm	250ms-on, 250ms-off Repeating
Routing Tone	Not specified		425 Hz/ -6 dBm	7 times 75ms on/75ms off, followed by one time 75ms on, 875ms off - Repeating
Break-in Ticker	425 Hz/ -3dBm	Continuous	425 Hz/ -3dBm	10 times 32ms-on/ 480ms-off - repeating every 32s
Add-on Ticker	425 Hz/ -3dBm	4s-on, 30s-off Repeating	425 Hz/ -3dBm	10 times 32ms-on/ 480ms-off - repeating every 32s
Howler Tone	1800 Hz/ -13 dBm	250ms-on, 250ms-off Repeating	2000 Hz/ +3 dBm	32ms-on, 32ms-off Repeating

(c) Austria

Name	EMEA		Basic Tone Setting for EMEA (This feature)	
	Signal Frequency (Hz)/ Sensitivity Level (dBm)	Duration of Connection	Signal Frequency (Hz)/ Sensitivity Level (dBm)	Duration of Connection
Internal Dial Tone	425 Hz/-3 dBm	150ms-on, 150ms-off, 150ms-on, 150ms-off, 150ms-on, 700ms-off, Repeating	425 Hz/- 3 dBm	150ms-on, 150ms-off, 150ms-on, 150ms-off, 150ms-on, 750ms-off, Repeating
Enquiry Dial Tone	425 Hz/ -3 dBm	150ms-on, 150ms-off, 150ms-on, 150ms-off, 150ms-on, 700ms-off, Repeating	425 Hz/ -3 dBm	150ms-on, 150ms-off, 150ms-on, 150ms-off, 150ms-on, 750ms-off, Repeating
Congestion Tone	425 Hz/ -3 dBm	200ms-on, 200ms-off Repeating	425 Hz/ -3 dBm	400ms-on, 400ms-off Repeating
Diversion Active Dial Tone	425 Hz/ -3 dBm	350ms-on, 10ms-off Repeating	425 Hz/ -3 dBm	375ms-on, 25ms-off Repeating
Call Waiting Tone Burst	425 Hz/ -23 dBm	200ms-on, 200ms-off, 200ms-on, off	425 Hz/ -3 dBm	192ms-on, 192ms-off, 192ms-on, off
Park Tone	425 Hz/ -3dBm	200ms-on, 200ms-off Repeating	425 Hz/ -3dBm	50ms-on, 50ms-off Repeating
Routing Tone	425 Hz/ -12dBm 950 Hz/ -13dBm 1400 Hz/ -13dBm 1800 Hz/ -13dBm 1400 Hz/ -13dBm 950 Hz/ -13dBm	1s-on, 100ms-on, 100ms-on, 100ms-on, 100ms-on, 100ms-on - Repeating	425 Hz/ -6 dBm	7 times 75ms on/75ms off, followed by one time 75ms on, 875ms off - Repeating
Break-in Ticker	425 Hz/ -3dBm	500ms-on, 500ms-off Repeating	425 Hz/ -3dBm	4 times 32 ms-on/ 960ms-off - Repeating
Add-on Ticker	425 Hz/ -3dBm	500ms-on, 500ms-off Repeating	425 Hz/ -3dBm	4 times 32 ms-on/ 960ms-off - Repeating
Howler Tone	1800 Hz/ -13 dBm	250ms-on, 250ms-off Repeating	2000 Hz/ +3 dBm	32ms-on, 32ms-off Repeating

(d) Italy

Name	EMEA		Basic Tone Setting for EMEA (This feature)	
	Signal Frequency (Hz)/ Sensitivity Level (dBm)	Duration of Connection	Signal Frequency (Hz)/ Sensitivity Level (dBm)	Duration of Connection
First External Dial Tone	425 Hz/-6 dBm	600ms-on, 1s-off, 200ms-on, 200ms-off, 600ms-on, 1s-off, Repeating	425 Hz/- 6 dBm	600ms-on, 1.075s-off, 200ms-on, 200ms-off, 600ms-on, 1.075s-off, Repeating
Congestion Tone	425 Hz/ -6 dBm	200ms-on, 200ms-off Repeating	425 Hz/ -6 dBm	500ms-on, 500ms-off Repeating
Diversion Active Dial Tone	350 Hz + 440 Hz/ -9 dBm + -9dBm	320ms-on, 10ms-off Repeating	350 Hz + 440 Hz/ -9 dBm + -9dBm	350ms-on, 25ms-off Repeating
Confirmation Tone	425 Hz/ -6 dBm	100ms-on, 100ms-off, 100ms-on, 100ms-off, 100ms-on, 1.5s-off, Repeating	425 Hz/ -6dBm	96ms-on, 96ms-off, 96ms-on, 96ms-off, 96ms-on, 1504ms-off - Repeating 3 times, then silence
Call Waiting Tone Burst	425 Hz/ -11 dBm	100ms-on, 4.9s-off Repeating	425 Hz/ -6 dBm	96ms-on, 4896ms-off, Repeating
Camp-on Busy Tone to Caller	425 Hz/ -6 dBm	200ms-on, 200ms-off 200ms-on, 3.4s-off Repeating	425 Hz/ -6dBm	200ms-on, 200ms-off 200ms-on, 3.15s-off Repeating
Hold Tone	425 Hz/ -6 dBm	200ms-on, 200ms-off 200ms-on, 3.4s-off Repeating	425 Hz/ -6dBm	200ms-on, 200ms-off 200ms-on, 3.15s-off
Park Tone	425 Hz/ -6 dBm	500ms-on, 500ms-off Repeating	425 Hz/ -6 dBm	100ms-on, 100ms-off Repeating
Break-in Ticker	425 Hz/ -3dBm	10ms-on, 3s-off Repeating	425 Hz/ -3dBm	10 times 32 ms-on/ 480ms-off - Repeating every 32s
Add-on Ticker	425 Hz/ -3dBm	4s-on, 30s-off Repeating	425 Hz/ -3dBm	10 times 32 ms-on/ 480ms-off - Repeating every 32s
Howler Tone	1800 Hz/ -13 dBm	250ms-on, 250ms-off Repeating	2000 Hz/ +3 dBm	32ms-on, 32ms-off Repeating

(e) Belgium

Name	EMEA		Basic Tone Setting for EMEA (This feature)	
	Signal Frequency (Hz)/ Sensitivity Level (dBm)	Duration of Connection	Signal Frequency (Hz)/ Sensitivity Level (dBm)	Duration of Connection
Number Unobtainable Tone	450 Hz/-4 dBm	180ms-on, 180ms-off Repeating	450 Hz/-4 dBm	200ms-on, 200ms-off, Repeating
Congestion Tone	450 Hz/-4 dBm	180ms-on, 180ms-off Repeating	450 Hz/ -4 dBm	200ms-on, 200ms-off Repeating
Confirmation Tone	450 Hz/ -4 dBm	40ms-on, 40ms-off Repeating	450 Hz/ -4dBm	480ms-on, 480ms-off Repeating
Call Waiting Tone Burst	1400 Hz/ -13 dBm	250ms-on, 5s-off Repeating	450 Hz/ -4 dBm	256ms-on - Repeating every 4s
Park Tone	450 Hz/ -4 dBm	500ms-on, 500ms-off Repeating	450 Hz/ -4 dBm	200ms-on, 200ms-off Repeating
Break-in Ticker	425 Hz/ -3dBm	Continuous	425 Hz/ -3dBm	5 times 32 ms-on/ 480ms-off - Repeating
Add-on Ticker	425 Hz/ -3dBm	4s-on, 30ms-off Repeating	425 Hz/ -3dBm	5 times 32 ms-on/ 480ms-off - Repeating every 32s
Howler Tone	1800 Hz/ -13 dBm	250ms-on, 250ms-off Repeating	2000 Hz/ +3 dBm	32ms-on, 32ms-off Repeating

(f) Denmark

Name	EMEA		Basic Tone Setting for EMEA (This feature)	
	Signal Frequency (Hz)/ Sensitivity Level (dBm)	Duration of Connection	Signal Frequency (Hz)/ Sensitivity Level (dBm)	Duration of Connection
Internal Ring Tone	425 Hz/-3 dBm	630ms-on, 100ms-off 270ms-on, 4s-off	425 Hz/-3 dBm	1s-on, 4s-off, Repeating
Number Unobtainable Tone	950 Hz/-13 dBm 1800 Hz/-13 dBm 1400 Hz/-13 dBm	330ms-on, 330ms-on 300ms-on, 2s-off Repeating	425 Hz/ -6 dBm	50ms-on, 50ms-off Repeating
Diversion Active Dial Tone	400 Hz/ -9 dBm	330ms-on, 20ms-off Repeating	400 Hz/ -9dBm	350ms-on, 25ms-off Repeating
Confirmation Tone	425 Hz/-3 dBm	1s-on, 250ms-off Repeating	425 Hz/-3 dBm	992ms-on, 256ms-off Repeating
Call Waiting Tone Burst	425 Hz/ -23 dBm	50ms-on, 100ms-off 50ms-on, 12s-off Repeating	425 Hz/ -3 dBm	64ms-on, 96ms-off, 64ms-on -repeating every 4s
Hold Tone	33.33 Hz x 400 Hz -17 dBm (suppressed carrier)	200ms-on, 200ms-off 200ms-on, 200ms-off 200ms-on, 10s-off Repeating	33.33 Hz x 400 Hz -17 dBm (suppressed carrier)	200ms-on, 200ms-off 200ms-on, 200ms-off 200ms-on, 9s-off Repeating
Park Tone	425 Hz/ -3 dBm	250ms-on, 250ms-off Repeating	425 Hz/ -6 dBm	50ms-on, 50ms-off Repeating
Break-in Ticker	425 Hz/ -3dBm	Continuous	425 Hz/ -3dBm	5 times 32 ms-on/ 480ms-off - Repeating every 4s
Add-on Ticker	425 Hz/ -3dBm	2s-on, 15s-off Repeating	425 Hz/ -3dBm	2 times 32 ms-on/ 480ms-off - Repeating every 32s
Howler Tone	1800 Hz/ -13 dBm	250ms-on, 250ms-off Repeating	2000 Hz/ +3 dBm	32ms-on, 32ms-off Repeating

(g) Sweden

Name	EMEA		Basic Tone Setting for EMEA (This feature)	
	Signal Frequency (Hz)/ Sensitivity Level (dBm)	Duration of Connection	Signal Frequency (Hz)/ Sensitivity Level (dBm)	Duration of Connection
Busy Tone	425 Hz/-9 dBm	250ms-on, 250ms-off Repeating	425 Hz/-9 dBm	250ms-on, 250ms-off, Repeating
Number Unobtainable Tone	950 Hz/-13 dBm 1400 Hz/-13 dBm 1800 Hz/-13 dBm	330ms-on, 30ms-off 330ms-on, 30ms-off 330ms-on, 1s-off Repeating	425 Hz/ -6 dBm	50ms-on, 50ms-off Repeating
Diversion Active Dial Tone	425 Hz/-9 dBm	320ms-on, 20ms-off Repeating	425 Hz/ -9dBm	350ms-on, 25ms-off Repeating
Confirmation Tone	425 Hz/-9 dBm	320ms-on, 20ms-off Repeating	425 Hz/ -9 dBm	1504ms-on, 512ms-off Repeating
Call Waiting Tone Burst	425 Hz/ -23 dBm	200ms-on, 500ms-off 200ms-on, off (Not repeating)	425 Hz/ -9 dBm	192ms-on, 512ms-off, 192ms-on, off
Park Tone	425 Hz/ -9 dBm	250ms-on, 750ms-off Repeating	425 Hz/ -6 dBm	50ms-on, 50ms-off Repeating
Break-in Ticker	1400 Hz/ -27dBm	100ms-on, 1.5s-off Repeating	1400 Hz/ -27dBm	2 times 96 ms-on/ 1504ms-off - Repeating every 4s
Add-on Ticker	1400 Hz/ -27dBm	360ms-on, 15s-off Repeating	1400 Hz/ -27dBm	352ms-on, off Repeating every 32s
Howler Tone	1800 Hz/ -13 dBm	250ms-on, 250ms-off Repeating	2000 Hz/ +3 dBm	32ms-on, 32ms-off Repeating

(h) Swiss

Name	EMEA		Basic Tone Setting for EMEA (This feature)	
	Signal Frequency (Hz)/ Sensitivity Level (dBm)	Duration of Connection	Signal Frequency (Hz)/ Sensitivity Level (dBm)	Duration of Connection
Diversion Active Dial Tone	340 Hz/ -9 dBm + 425 Hz/-9 dBm, 425 Hz/-9 dBm	1.1s-on, 1.1s-on Repeating	425 Hz (-9dBm) + 340 Hz (-14 dBm)	975ms-on, 25ms-off, Repeating
Confirmation Tone	425 Hz/-9 dBm, 950 Hz/-9 dBm, 425 Hz/-9 dBm, 950 Hz/-9 dBm,	200ms-on, 200ms-on, 200ms-on, 200ms-on, 1s-off Repeating	425 Hz/ -19 dBm	800ms-on/992ms-off Repeating
Call Waiting Tone Burst	425 Hz/ -23 dBm	200ms-on, 200ms-off 200ms-on, 4s-off Repeating	425 Hz/ -19 dBm	192ms-on, 192ms-off, 192ms-on - Repeating every 4s
Camp-on Busy Tone to Caller	425 Hz/ -9 dBm	1s-on, 3.6s-off, 500ms-on, 500ms-off, 500ms-on, 3.6s-off Repeating	425 Hz/ -9 dBm	1s-on, 3.75s-off, 500ms-on, 500ms-off, 500ms-on, 3.75s-off Repeating
Hold Tone	425 Hz/ -9 dBm	1s-on, 3.6s-off, 500ms-on, 500ms-off, 500ms-on, 3.6s-off Repeating	425 Hz/ -9 dBm	1s-on, 3.75s-off, 500ms-on, 500ms-off, 500ms-on, 3.75s-off Repeating
Park Tone	425 Hz/ -9 dBm	500ms-on, 500ms-off Repeating	425 Hz/ -9 dBm	200ms-on, 200ms-off Repeating
Routing Tone	950 Hz/ -9 dBm, 425 Hz/ -9 dBm	500ms-on, 500ms-on 2s-off - Repeating	425 Hz/ -6 dBm	7 times 75ms on/off, followed by one time 75 ms on, 875ms off Repeating
Break-in Ticker	425 Hz/-23 dBm	200ms-on, 2s-off Repeating	425 Hz/ -23 dBm	2 times 192-ms on/ 1792ms-off - Repeating 4s
Add-on Ticker	425 Hz/-23 dBm	450ms-on, 15s-off Repeating	425 Hz/ -23 dBm	448ms-on -repeating every 32s
Howler Tone	1800 Hz/-13 dBm	250ms-on, 250ms-off Repeating	2000 Hz/ +3 dBm	32ms-on, 32ms-off Repeating

(i) UK

Name	EMEA		Basic Tone Setting for EMEA (This feature)	
	Signal Frequency (Hz)/ Sensitivity Level (dBm)	Duration of Connection	Signal Frequency (Hz)/ Sensitivity Level (dBm)	Duration of Connection
External Ring Tone	33.33 Hz x 400 Hz -17 dBm (suppressed carrier)	400ms-on, 200ms-off, 400ms-on, 2s-off Repeating	33.33 Hz x 400 Hz -17 dBm (suppressed carrier)	1s-on, 3s-off Repeating
Busy Tone	400 Hz -12 dBm	380ms-on, 380ms-off Repeating	400 Hz/ -12 dBm	375ms-on/375ms-off Repeating
Diversion Active Dial Tone	33.33 Hz x 400 Hz -17 dBm (suppressed carrier)	320ms-on, 10ms-off Repeating	33.33 Hz x 400 Hz -17 dBm (suppressed carrier)	350ms-on/25ms-off Repeating
Call Waiting Tone Burst	33.33 Hz x 400 Hz -17 dBm (suppressed carrier)	500ms-on, off	33.33 Hz x 400 Hz -17 dBm (suppressed carrier)	512ms-on, off
Camp-on Busy Tone to Caller	33.33 Hz x 400 Hz -17 dBm (suppressed carrier)	1s-on, 4s-off, Repeating	33.33 Hz x 400 Hz -17 dBm (suppressed carrier)	1s-on, 3s-off Repeating
Hold Tone	33.33 Hz x 400 Hz -17 dBm (suppressed carrier)	1s-on, 4s-off, Repeating	33.33 Hz x 400 Hz -17 dBm (suppressed carrier)	1s-on, 3s-off Repeating
Routing Tone	400 Hz/ -12 dBm	250ms-on, 2.5s-off Repeating	400 Hz/ -12 dBm	250ms-on, 2.75s-off Repeating
Break-in Ticker	425 Hz/-3 dBm	1s-on, 4s-off, Repeating	425 Hz/ -3 dBm	2 times 32ms-on/ 480ms-off Repeating every 4s
Add-on Ticker	425 Hz/-3 dBm	200ms-on, 200s-off, 200ms-on, 5s-off Repeating	425 Hz/ -3 dBm	2 times 32ms-on/ 352ms-off Repeating every 4s
Howler Tone	1800 Hz/-13 dBm	250ms-on, 250ms-off Repeating	2000 Hz/ +3 dBm	32ms-on, 32ms-off Repeating

(j) Spain

Name	EMEA		Basic Tone Setting for EMEA (This feature)	
	Signal Frequency (Hz)/ Sensitivity Level (dBm)	Duration of Connection	Signal Frequency (Hz)/ Sensitivity Level (dBm)	Duration of Connection
External Ring Tone	425 Hz -12 dBm	1s-on, 4s-off Repeating	425 Hz/-12 dBm	300ms-on, 400ms-off, 300ms-on, 4s-off Repeating
Busy Tone	425 Hz -12 dBm	500ms-on, 500ms-off Repeating	425 Hz/ -12 dBm	500ms-on, 500ms-off Repeating
Number Unobtainable Tone	425 Hz -12 dBm	60ms-on, 60ms-off Repeating	425 Hz/ -12 dBm	50ms-on, 50ms-off Repeating
Diversion Active Dial Tone	425 Hz -9 dBm	320ms-on, 10ms-off Repeating	425 Hz -9 dBm	350ms-on, 25ms-off Repeating
Call Waiting Tone Burst	425 Hz -9 dBm	250ms-on, 250ms-off Repeating	425 Hz -9 dBm	2 times 256ms-on/ 256ms-off - Repeating every 4s
Park Tone	425 Hz -9 dBm	500ms-on, 500ms-off Repeating	425 Hz -12 dBm	50ms-on, 50ms-off Repeating
Routing Tone	Not specified (EMEA default tone)	Not specified (EMEA default tone)	425 Hz -6 dBm	7 times 75ms-on/off, followed by one time 75ms on, 875ms off Repeating
Break-in Ticker	425 Hz/-3 dBm	Continuous	425 Hz/ -3 dBm	5 times 32ms-on/ 480ms-off - Repeating
Add-on Ticker	425 Hz/-3 dBm	4s-on, 30s-off Repeating	425 Hz/ -3 dBm	5 times 32ms-on/ 480ms-off - Repeating every 32s
Howler Tone	1800 Hz/-13 dBm	250ms-on, 250ms-off Repeating	2000 Hz/ +3 dBm	32ms-on, 32ms-off Repeating

(k) Greece

Name	EMEA		Basic Tone Setting for EMEA (This feature)	
	Signal Frequency (Hz)/ Sensitivity Level (dBm)	Duration of Connection	Signal Frequency (Hz)/ Sensitivity Level (dBm)	Duration of Connection
Congestion Tone	425 Hz -3 dBm	80ms-on, 80ms-off Repeating	425 Hz/-3 dBm	200ms-on, 200ms-off Repeating
Confirmation Tone	425 Hz -3 dBm	3.5s-on, off	425 Hz/-3 dBm	3488ms-on, off
Call Waiting Tone Burst	425 Hz -9 dBm	1s-on, off	425 Hz/-3 dBm	960ms-on, off
Camp-on Busy Tone to Caller	25 Hz x 425 Hz -10 dBm (suppressed carrier)	200ms-on, 200ms-off, 200ms-on, 2s-off, Repeating	25 Hz x 425 Hz -10 dBm (suppressed carrier)	200ms-on, 200ms-off, 200ms-on, 1.9s-off, Repeating
Hold Tone	25 Hz x 425 Hz -10 dBm (suppressed carrier)	200ms-on, 200ms-off, 200ms-on, 2s-off, Repeating	25 Hz x 425 Hz -10 dBm (suppressed carrier)	200ms-on, 200ms-off, 200ms-on, 1.9s-off, Repeating
Park Tone	425 Hz/-3 dBm	160ms-on, 480ms-off Repeating	425 Hz/ -3 dBm	200ms-on, 200ms-off Repeating
Routing Tone	Not specified (EMEA default tone)	Not specified (EMEA default tone)	425 Hz/ -6 dBm	7 times 75ms-on/off, followed by one time 75ms on, 875ms off Repeating
Break-in Ticker	950 Hz/-13 dBm	1s-on, 4s-off Repeating	950 Hz/-13 dBm	480ms-on, off Repeating every 4s
Add-on Ticker	950 Hz/-13 dBm	4s-on, 6s-off Repeating	950 Hz/ -13 dBm	480ms-on, off Repeating every 32s
Howler Tone	1800 Hz/-13 dBm	250ms-on, 250ms-off Repeating	2000 Hz/ +3 dBm	32ms-on, 32ms-off Repeating

(l) South Africa

Name	EMEA		Basic Tone Setting for EMEA (This feature)	
	Signal Frequency (Hz)/ Sensitivity Level (dBm)	Duration of Connection	Signal Frequency (Hz)/ Sensitivity Level (dBm)	Duration of Connection
External Ring Tone	33.33 Hz x 400 Hz -3 dBm	400ms-on, 200ms-off, 400ms-on, 2s-off	33.33 Hz x 400 Hz -3 dBm	1s-on, 2s-off Repeating
Confirmation Tone	450 Hz / -8 dBm 1800 Hz / -13dBm	200ms 200ms	400 Hz/ -9dBm	608ms-on,192ms-off, Repeating (Max.16s)
Call Waiting Tone Burst	400 Hz -9 dBm	250ms-on, off	400 Hz/ -9dBm	256ms-on, off repeating every 4s
Break-in Ticker	425 Hz -3 dBm	2s-on, off	425 Hz -3 dBm	5 times 32ms-on/ 480ms-off, repeating every 4s
Add-on Ticker	425 Hz -3 dBm	500ms-on, 10s-off Repeating	425 Hz -3 dBm	4 times 32ms-on/ 480ms-off
Howler Tone	1800Hz -13 dBm	250ms-on, 250ms-off Repeating	<PCM Howler> 2000Hz	<PCM Howler> 32ms-on, 32ms-off

APPENDIX C Reference Information for Customizing/Editing Your Language Files through Optional Language Pack Loading

This appendix provides supplementary information about editing on-screen text through Optional Language Pack Loading [O-42].

The DT800/DT700/DT400/DT300 terminals have an LCD providing three lines of text information; The bottom line shows Soft keys, the middle line shows date and time, and the top line shows operation-related information. In the order of area from bottom to top of the screen, the following topics are explained:

- Soft Key Labels — Section 1
- Abbreviation of Name of Month and Day Name of the Week (Kind Number 44 and 45) — Section 2
- Non-Call Operations — Section 3
- Call Operations — Section 4
- Service-Specific Information Message — Section 5
- Help Messages — Section 6
- Trunk Class Numbers and Abbreviations — Section 7
- Annex — Section 8

The procedures explained in Sections 1 and 2 are relatively simple and straightforward because they are directly related to one name or one function key. However, procedures provided in Sections 3 and 4 are more complicated because text strings of information messages can be used in multiple features.

Note: Kind Number and Data Number referred to in this appendix are the “Kind” column and the “Data Number” column in the Excel sheet EXLANG_0001.xls, respectively.

1. Soft Key Labels

The Soft key labels classified as Kind Number 16 and 17 (see Tables 1 and 2) can be modified through both the maintenance command, AFDD, and Optional Language files. On the other hand, the Soft key labels classified as other Kind Numbers, shown in Table 3, can be modified only through Optional Language files.

The advantage of using Optional Language files instead of the AFDD command is that once you have completed modifying the language file, you can distribute copies of the file to other sites.

Kind Number = 16

FKY	Index	Service Name	Default	Deutsche	Kind No.	Data No.
1	C-2	Call Forwarding - Busy Line	FDB	RW-FB	16	48
2	C-5	Call Forwarding - All Calls	FDA	RW-A	16	49
3	E-1	Executive Right-of-Way	E-OVR	AFROW	16	50
4	C-12	Call Waiting	CW	ANR_W	16	51
5	C-1	Call Back	CB	RR	16	52
6	M-19	Message Reminder	MSG-R	NRERI	16	53
7	S-24D	Save and Repeat	S&R	SP_WW	16	54
8	L-6D	Last Number Called	LAST#	WW	16	55
9	B-8	Boss-Secretary-Message Waiting Lamp Control Set	MWSET	NRLSE	16	56
10	C-6	Call Hold	C_HOLD	RUFHA	16	57
11	B-8	Boss-Secretary-Message Waiting Lamp Control Cancel	MWOFF	NRLLÖ	16	58
12	C-7	Call Pickup - Group	PICK	PICKG	16	59
13	T-2	Three-way Calling	CONF	3KONF	16	60
14	V-2	Voice Call	VOICE	SPRUF	16	61
15	S-3	Speed Calling - System	S-SPD	KWSYS	16	62
17	T-2	Three-Way Calling - CAS shift-flash	CAS	CAS	16	64
19	U-1	UCD Busy Out	UCDBO	UCDBE	16	66
21	F-3D	Flash Button	FLASH	ERDTA	16	68
22	C-3	Call Forwarding - Don't Answer	FDN	RW-KA	16	69
25		ISDN Malicious Call (Australia and U.A.E only)	C_RPT	FANGN	16	72
26	S-21	Speed Calling - Station	G_SPD	KWNST	16	73
48	P-18	Privacy Release	P-RLS	PRIVE	16	95
50	D-11D	Do Not Disturb	DND	RABW	16	112
51		Microphone	MIC	MIC	16	113

Kind Number = 16

FKY	Index	Service Name	Default	Deutsche	Kind No.	Data No.
55	S-26D	Speed Calling - One Touch - Page Changeover	PAGE	KWSEL	16	117
60	M-47	Manual Signaling Key	SIG	SIGNT	16	122

Kind Number = 17

FKY	Index	Service Name	Default	Deutsche	Kind No.	Data No.
83	A-121	Add-On Conference - Retain Conference	CKEEP	KONFM	17	18
84	S-127D	Serial Call -D	SCALL	SER_R	17	19
85	I-21	Internal Zone Paging	IZP	IZPSE	17	20
87	D-105D	Hotel Functions - Wake Up Set	WUS	WECKE	17	22
88	D-105D	Hotel Functions - Wake Up Reset	WUR	WECKA	17	23
89	D-105D	Hotel Functions - Do Not Disturb Set	DDS	RABWE	17	24
90	D-105D	Hotel Functions - Do Not Disturb Reset	DDR	RABWA	17	25
91	D-105D	Hotel Functions - Room Cut Off Set	RCS	ZS_AN	17	26
92	D-105D	Hotel Functions - Room Cut Off Reset	RCR	ZS_AB	17	27
93	D-105D	Hotel Functions - Message Waiting Set	MWS	MW_E	17	28
94	D-105D	Hotel Functions - Message Waiting Reset	MWR	MW_A	17	29
95	D-105D	Hotel Functions - Check In	C/I	CKEIN	17	30
96	D-105D	Hotel Functions - Check Out	C/O	CKAUS	17	31
97	D-105D	Hotel Functions - Status	STS	STATU	17	32
98	D-105D	Hotel Functions - Audit	AUD	PRÜF	17	33
99	D-105D	Hotel Functions - Enter	ENT	AUSFÜ	17	34
100	D-105D	Hotel Functions - Clear Enter	CE	LÖSCH	17	35
101	D-105D	Hotel Functions - END	END	ENDE	17	36
113	C-157	Call ID Change Display	IDCHG	ID_WS	17	48
115	C-7	Call Pickup Expand	GPICK	GRANN	17	50
117	R-49	Ringer Mute	RMUTE	RUTOA	17	52
120		Next Page of Soft Keys (Scroll Key)	>>>	>>>	17	55
121		Previous Page of Soft Keys (Back Scroll Key)	<<<	<<<	17	56
122		Soft Key Assignment	S-KEY	TAPRO	17	57

Kind Number = 17

FKY	Index	Service Name	Default	Deutsche	Kind No.	Data No.
123		Off Hook Ringing	OHR	FEHLR	17	58
124		Mute	MUTE	STUMM	17	59
125		Headset/Handset	HSET	HEADS	17	60
126	N-45	PS Key	PS	PS_TA	17	61
127	D-156	Dual Station Call - Boss Move	B_MV	CHUML	17	62
128	C-150	Call Block	IDRST	GRRUF	17	63
129	C-160	Call Trace	RPT	MITSC	17	64
131	S-157	Disconnect	DISC	CLI	17	66
136	S-137	Selectable Station Call Pickup	S PCK	EGRAN	17	71
137	S-137	Selectable Station Call Pickup Group	GSPCK	EGRAG	17	72
142	D-159	LOGOUT for IP Enabled Dterm	L-OUT	L-OUT	17	77
148	L-53	Call Forwarding - Logout	FDL	RWLÖ	17	83
149	M-107	Message Audible Alert	MSGA	SPRAL	17	84
151	L-54	MA-ID Setting	MAID	MA-ID	17	86
156	I-48	Visitor Key	VISIT	GAST	17	91
163	D-165	Privacy Key	PVCY	PV_NR	17	98
167	M-118	Mobility Access	MACC	MACC	17	102
177	M-117	Language Change	LANG	SPRACH	17	112
178	C-195	Call History	iCall	ANRUFE	17	113
180	P-83	Phonebook	Pbook	Telb	17	115
190	D-170	Display Reversing	RVERS	INVRS	17	125

The following Soft key labels can be modified through Optional Language files only.

Service-specific Soft Keys (Kind Number = 23,24, 29, 32, 47, 49, 51, and 54)

Index	Service Name	Default	Deutsche	Kind No.	Data No.
P-83	Phonebook - Switch to Local PhoneBook	Local	Lokal	23	1
	Phonebook - Switch to System PhoneBook	Sys	Sys	23	2
	Phonebook - Extension	Extn	Extn	23	3
	Phonebook - Cancel	Cncel	Abbr	23	4
	Phonebook - Search	Srch	Suche	23	5
	Phonebook - Add	Add	Einf	23	6
	Phonebook -Next	->	->	23	7
	Phonebook - Clear	Clear	CLS	23	8
	Phonebook -Lower Case	Lcase	Eu-kl	23	9
	Phonebook - Upper Case	Ucase	Eu-gr	23	10
	Phonebook - Next	>>>	>>>	23	13
	Phonebook - Previous	<<<	<<<	23	14
	Phonebook - OK	OK	OK	23	15
	Phonebook -Previous	Prev	oben	23	16
	Phonebook - Next	Next	unten	23	17
	Phonebook - More	More	mehr	23	18
	Phonebook - Edit	Edit	Aend	23	19
	Phonebook - Delete	Del	Loesh	23	20
	Phonebook - Prefix	Prfix	Pref	23	21
C-195	Call History - Up Cursor	(up)	(up)	24	0
	Call History - Down Cursor	(down)	(down)	24	1
	Call History - Night Cursor	>>>	>>>	24	3
	Call History - Left Cursor	<<<	<<<	24	4
	Call History - Prefix	Prfix	Pref	24	5
	Call History - Delete	DELETE	LOESCH	24	6
	Call History - End	END	ENDE	24	7
S-162	Preset Dial - Clear	Clr	Zrück	29	1
	Preset Dial - Cancel	Cnl	Ende	29	2
	Preset Dial - More	More	Mehr	29	3

Service-specific Soft Keys (Kind Number = 23,24, 29, 32, 47, 49, 51, and 54)

Index	Service Name	Default	Deutsche	Kind No.	Data No.
M-117	Multilingual Display - Select	SELECT	SELECT	32	1
	Multilingual Display - Default	DFLT	DFLT	32	2
	Multilingual Display - Cancel	CANCEL	CANCEL	32	3
	Multilingual Display - Menu	MENU	MENU	32	4
	Multilingual Display - Next	NEXT	NEXT	32	5
	Multilingual Display - Set	SET	SET	32	6
D-169	DESI-Less Terminal - Upper case	UCase	Eu-gr	47	0
	DESI-Less Terminal - Lower case	LCase	Eu-kl	47	1
	DESI-Less Terminal - Arrows, Back and Next	<- -> BK >>>	<- -> ZURUCK >>>	49	0
	DESI-Less Terminal - Insert, Delete and Next	INS DEL >>>	EINF LOESCH >>>	49	1
	DESI-Less Terminal - Set, Clear, Cancel and Next	SET CLEAR CANCEL >>>	SETZEN ENTF ABRUCH >>>	49	2
C-195	Call History - Left Cursor	<<<	<<<	51	0
	Call History - Right Cursor	>>>	>>>	51	1
	Call History - Delete	DELETE	LOESCH	51	2
	Call History - End	END	ENDE	51	3
R-60	Redial - Up Arrow	(up)	(up)	54	1
	Redial - Down Arrow	(down)	(down)	54	2
	Redial - Delete	Delete	Loesch	54	3
	Redial - End	End	Ende	54	4
	Redial - One	One	Eine	54	5
	Redial - All	All	Alles	54	6
	Redial - Cancel	Cancel	Abbr	54	7

2. Abbreviation of Name of Month and Day Name of the Week (Kind Number 44 and 45)

You can modify the abbreviation of the name of month and the day name of the week. The abbreviation cannot exceed three characters in length.

The text will appear in the middle line of the DT terminal LCD (e.g. “11:23 AM TUE 7 AUG 2012”).

Month and Day Name

Text Type	Meaning	Default	Deutsche	Kind No.	Data No.
Month	January	JAN	JAN	44	0
	February	FEB	FEB	44	1
	March	MAR	MRZ	44	2
	April	APR	APR	44	3
	May	MAY	MAI	44	4
	June	JUN	JUN	44	5
	July	JUL	JUL	44	6
	August	AUG	AUG	44	7
	September	SEP	SEP	44	8
	October	OCT	OKT	44	9
	November	NOV	NOV	44	10
	December	DEC	DEZ	44	11
Day Name	Sunday	SUN	SO	45	0
	Monday	MON	MO	45	1
	Tuesday	TUE	DI	45	2
	Wednesday	WED	MI	45	3
	Thursday	THU	DO	45	4
	Friday	FRI	FR	45	5
	Saturday	SAT	SA	45	6

3. Non-Call Operations

Non-Call operations are categorized into two groups; First group is controlled by the Telephony Server and the second group is controlled by a DT terminal itself.

This section includes the following lists of information message text used in non-call operations:

- Non-Call Operations Controlled by the Telephony Server (Table 5 Kind Number 18)
- Cursor Up/Down Operation (Table 6 Kind Number 19)
- Non-Call Operations Controlled by Terminal (Table 7 Kind Number 26)
- Non-Call Operations Controlled by Terminal (Table 7 Kind Number 33-43)

Non-Call Operations Controlled by the Telephony Server

Operation	Index	Service Name	Default	Deutsche	Kind No.	Data No.
	M-19	Message Reminder	MSG	NRER	18	0
Redial	L-6D	Last Number Called	LCR [*]/SPD[.]	WW [*]/ZW[.]	18	3
Feature+2		Handset Receiving Volume Control - Small	RCV VOL SMALL	HÖRLAUTST LEISE	18	6
Feature+2		Handset Receiving Volume Control - Large	RCV VOL LARGE	HÖRLAUTST LAUT	18	7
Feature+3		Ringer Tone Select	RINGER TONE	KLINGELTON	18	9
Feature+4		Transmit/Receive Volume Control - Down	VOLUME DOWN	LAUTSTÄRKE LEISE	18	12
Feature+4		Transmit/Receive Volume Control - Up	VOLUME UP	LAUTSTÄRKE LAUT	18	13
Feature+5		Hands-free - On	HANDS FREE ON	FREISPRECHEN AN	18	15
Feature+6		Hands-free - Off	HANDS FREE OFF	FREISPRECHEN AUS	18	18
Feature+7		Call indicator Lamp - On	INDICATOR ON	INDIKATOR AN	18	21
Feature+7		Call indicator Lamp - Off	INDICATOR OFF	INDIKATOR AUS	18	22
Feature+8	S-162	Dynamic Dial Pad - OFF	DirectDialPadOFF	DIREKTWAHL AUS	18	24
Feature+8	S-162	Dynamic Dial Pad - ON	DirectDialPad ON	DIREKTWAHL AN	18	25
Feature+8	S-162	Selective Dialing - Normal	Dial Mode Normal	Wahlmodus Normal	18	33
Feature+8	S-162	Selective Dialing - Direct	Dial Mode Direct	Wahlmodus Direkt	18	34

Non-Call Operations Controlled by the Telephony Server

Operation	Index	Service Name	Default	Deutsche	Kind No.	Data No.
Feature+8	S-162	Selective Dialing - Preset	Dial Mode Preset	Wahlmodus Block-wahl	18	35

The table below lists information displayed at the left-bottom corner when the UP or DOWN key is pressed in several times.

Cursor Up/Down Operation

Press Up/Down When	What to Control	Default	Deutsche	Kind No.	Data No.
Talking on handset	Handset Volume	HANDSET	HÖRER	19	0
Talking on speakerphone	Speaker Volume	SPEAKER	LAUTSPR	19	1
Ringing	Ringer Volume	RING	KLINGEL	19	2
Ringing while off-hook	Off-Hook Ringer Volume	OFF-H R	OFF-HOOK	19	3
Idle	LCD Contrast	LCD	LCD	19	4
unknown	Back Ground Music	BGM	BGM	19	5

You can modify the text of a menu item of DT300 terminal. The menu tree is shown below.

Non-Call Operations Controlled by Terminal

Text of Menu Item (Customizable)				Meaning	Kind/ Data No.
[Menu]				Menu root display	26/1
|					
+-----	Call History			Menu item for call history	26/2
|	|				
|	[Call History]			Call history menu display	26/5
|	|				
|	+-----	Incoming Calls		Menu item for incoming calls	26/6
|	|	|			
|	|	[Incoming calls]		Incoming call menu display	26/8
|	|	|			
|	|	+-----	Unknown	Unknown number list	26/9
|	|	|			
|	|	+-----	No Data	No incoming call records	26/10
|	|	|			
|	|	+-----	Delete One	Deleting one entry	26/32
|	|	|			
|	|	+-----	Delete All	Deleting all entries	26/33
|	|	|			
|	|	+-----	Dialing Error	Deleting all entries	26/34
|	|				
|	+-----	Outgoing Calls		Menu item for outgoing calls	26/7
|		|			
|		[Outgoing Calls]		Outgoing call menu display	26/11
|		|			
|		+-----	No Data	No outgoing call records	26/12
V		|			

Continues to next page

Non-Call Operations Controlled by Terminal

Text of Menu Item (Customizable)				Meaning	Kind/ Data No.
Continued from previous page					
|		|			
|		+-----	Delete One	Deleting one entry	26/32
|		|			
|		+-----	Delete All	Deleting all entries	26/33
|		|			
|		+-----	Dialing Error	Deleting all entries	26/34
|					
+-----	Directory			Menu item for phonebook	26/3
|					
+-----	Settings			Menu item for settings	26/4
	|				
	[Settings]			Settings menu display	26/13
	|				
	+-----	Ring Volume		Menu item for ringer volume	26/14
	|	|			
	|	+-----	[Ring Volume]	Ringer volume display	26/19
	|	|			
	|	+-----	[Headset Ring Volume]	Headset ringer volume display	26/20
	|				
	+-----	Off Hook Ring		Menu item for off-hook ring	26/15
	|	|			
	|	[Off Hook Ring]		Off-hook ring menu display	26/21
	|	|			
	|	+-----	Disable	Disabling off-hook ringing	26/22
	|	|			
	|	+-----	Enable	Enabling off-hook ringing	26/23
	V				
Continues to next page					

Non-Call Operations Controlled by Terminal

Text of Menu Item (Customizable)			Meaning	Kind/ Data No.
Continued from previous page				
|				
+-----	Back Light		Menu item for backlight	26/16
|	|			
|	[Back Light]		Backlight menu display	26/24
|	|			
|	+-----	Disable	Disabling backlight	26/25
|	|			
|	+-----	Enable	Enabling backlight	26/26
|				
+-----	Headset		Menu item for headset	26/17
|	|			
|	[Headset]		Headset menu display	26/27
|	|			
|	+-----	Disable	Disabling headset	26/28
|	|			
|	+-----	Enable	Enabling headset	26/29
|				
+-----	Peripherals		Menu item for peripherals	26/18
	|			
	[Peripherals]		Peripherals menu display	26/30

Non-Call Operations Controlled by Terminal (Deutsche)

Text of Menu Item (Customizable)				Meaning	Kind/ Data No.
[Menü]				Menu root display	26/1
+-----	Anruferliste			Menu item for call history	26/2
	[Anruferliste]			Call history menu display	26/5
	+-----	Ankommende Rufe		Menu item for incoming calls	26/6
		[Ankommende Rufe]		Incoming call menu display	26/8
		+-----	Unbekannt	Unknown number list	26/9
		+-----	Keine Daten	No incoming call records	26/10
		+-----	Eine Stelle löschen	Deleting one entry	26/32
		+-----	Alles löschen	Deleting all entries	26/33
		+-----	Wahlprozedurfehler	Deleting all entries	26/34
	+-----	Abgehende Rufe		Menu item for outgoing calls	26/7
		[Abgehende Rufe]		Outgoing call menu display	26/11
		+-----	Keine Daten	No outgoing call records	26/12

Continues to next page

Non-Call Operations Controlled by Terminal (Deutsche)

Text of Menu Item (Customizable)				Meaning	Kind/ Data No.
Continued from previous page					
|		|			
|		+-----	Eine Stelle löschen	Deleting one entry	26/32
|		|			
|		+-----	Alles löschen	Deleting all entries	26/33
|		|			
|		+-----	Wahlprozedurfehler	Deleting all entries	26/33
|					
+-----	Verzeichnis			Menu item for phonebook	26/3
|					
+-----	Einstellungen			Menu item for settings	26/4
	|				
	[Einstellungen]			Settings menu display	26/13
	|				
	+-----	Ruftonlautstärke		Menu item for ringer volume	26/14
	|	|			
	|	+-----	[Ruftonlautstärke]	Ringer volume display	26/19
	|	|			
	|	+-----	[Hör-Spr-G.Ruflautst.]	Headset ringer volume display	26/20
	|				
	+-----	Rufsign w Gespräch		Menu item for off-hook ring	26/15
	|	|			
	|	[Rufsign w Gespräch]		Off-hook ring menu display	26/21
	|	|			
	|	+-----	Deaktivieren	Disabling off-hook ringing	26/22
	|	|			
	|	+-----	Aktivieren	Enabling off-hook ringing	26/23
	|				
Continues to next page					

Non-Call Operations Controlled by Terminal (Deutsche)

Text of Menu Item (Customizable)			Meaning	Kind/ Data No.
Continued from previous page				
|				
+-----	Hintergrundlicht		Menu item for backlight	26/16
|	|			
|	[Hintergrundlicht]		Backlight menu display	26/24
|	|			
|	+-----	Deaktivieren	Disabling backlight	26/25
|	|			
|	+-----	Aktivieren	Enabling backlight	26/26
|				
+-----	Hör Sprechgarnitur		Menu item for headset	26/17
|	|			
|	[Hör Sprechgarnitur]		Headset menu display	26/27
|	|			
|	+-----	Deaktivieren	Disabling headset	26/28
|	|			
|	+-----	Aktivieren	Enabling headset	26/29
|				
+-----	Peripheriegeräte		Menu item for peripherals	26/18
	|			
	[Peripheriegeräte]		Peripherals menu display	26/30

4. Call Operations

Some text strings of information messages are used in multiple features. For example, the text string “FORWARD” is used when call forwarding destination is set or canceled.

There are two groups for the call-operation-related text strings; One group is the ACTION group and the other group is the RESULT group.

(a) Where ACTION and RESULT information is displayed:

← 8-character length →	← 8-character length →	← 8-character length →	
ACTION	RESULT		
11:41 AM THU 9 AUG 2012			
Soft key 1	Soft key 2	Soft key 3	Soft key 4

As shown above, the text string for ACTION information is displayed at the left-top corner with a maximum length of eight characters. The text string for RESULT information is displayed at the middle position of the top line, following the ACTION information text string. With the combination of these two types of texts, most of procedures for regularly-used phone operations are guided.

When you make a simple phone call, these areas are left blank. Status information such as “CONNECTING” or “CALLING” will not be displayed. However, when another action is taken, these areas will be used.

(b) Holding a call / Being held

When you put someone on hold by pressing the Hold key, or you are on hold by someone else, the ACTION area displays the text HOLD, which is saved with Kind Number=13 and Data Number=04 in the Optional Language file.

	ACTION	RESULT
Default text message	HOLD	
Kind No., Data No.	13, 04	none

	ACTION	RESULT
Deutsche text	HALTEN	
Kind No., Data No.	13, 04	none

When you press the Hold key twice to put the call on “exclusive hold,” in which no one else can retrieve the call, the following text string is displayed on the LCD.

	ACTION	RESULT
Default text message	E-HOLD	
Kind No., Data No.	14, 00	none

	ACTION	RESULT
Deutsche text	EXKLHALT	
Kind No., Data No.	14, 00	none

(c) Transferring a call

When you press the Transfer key during a call, you will hear the second dial tone, and the ACTION display will show “TRANSFER” (13,06). At this moment, the other party will see “HOLD” (13,04) in his/her ACTION area.

	ACTION	RESULT
Default text message	TRANSFER	
Kind No., Data No.	13, 06	none

	ACTION	RESULT
Deutsche text	VERBINDE	
Kind No., Data No.	13, 06	none

When you have completed dialing the transfer destination number, you hear the Ring Back Tone. The ACTION display on the other party, who is being transferred, will change to “TRANSFER” (13,06). This “TRANSFER” (13,06) indication remains until you replace the handset to complete the transferring operation.

(d) Encountering busy

When your destination party is on another call and you are not automatically forwarded then you will see “BUSY” (13,05) in the ACTION area.

	ACTION	RESULT
Default text message	BUSY	
Kind No., Data No.	13, 05	none

	ACTION	RESULT
Deutsche text	BESETZT	
Kind No., Data No.	13, 05	none

(e) Dialing invalid number

In case of dialing an unregistered number or forbidden number, you will see "REST" (13,08) in the ACTION area.

Note: "REST" means restriction.

	ACTION	RESULT
Default text message	REST	
Kind No., Data No.	13, 08	none

	ACTION	RESULT
Deutsche text	ABGELEHN	
Kind No., Data No.	13, 08	none

A recommended text is "Invalid."

(f) Setting and canceling Call Forwarding and being forwarded

When you press a Call Forwarding key, you will see "FORWARD" (13,02) in the ACTION display area. This message prompts you to enter a forwarding destination number when you press a forwarding key and go off-hook, or when you press a forwarding key with the handset off hook and hear Dial Tone.

	ACTION	RESULT
Default text message	FORWARD	
Kind No., Data No.	13, 02	none

	ACTION	RESULT
Deutsche text	WEITERL	
Kind No., Data No.	13, 02	none

If you successfully enter the destination number to the “FORWARD” (13,02) prompt, you will see “SET” (04,01) message on the RESULT area, which is in the middle of the top line on the LCD.

	ACTION	RESULT
Default text message	FORWARD	SET
Kind No., Data No.	13, 02	(04, 01)

	ACTION	RESULT
Deutsche text	WEITERL	EIN
Kind No., Data No.	13, 02	(04, 01)

The text string “SET” (04,01) can be used in other operations, such as “Do Not Disturb.”

If you cancel the Call Forwarding feature by pressing the Call Forwarding key again while off-hook, you will see “CANCEL” (04,02) in the RESULT area.

	ACTION	RESULT
Default text message	FORWARD	CANCEL
Kind No., Data No.	13, 02	(04, 02)

	ACTION	RESULT
Deutsche text	WEITERL	ABBRUCH
Kind No., Data No.	13, 02	(04, 02)

When a call is forwarded, both the destination party and originating party will see the same ACTION information message. Three types of Call Forwarding messages are generated according to Call Forwarding events.

[Call Forward - All Calls]

	ACTION	RESULT
Default text message	CF ALL	
Kind No., Data No.	14, 01	none

	ACTION	RESULT
Deutsche text	RWALLE	ABBRUCH
Kind No., Data No.	14, 01	none

[Call Forward - Busy]

	ACTION	RESULT
Default text message	CF BUSY	
Kind No., Data No.	14, 02	none

	ACTION	RESULT
Deutsche text	RWBESETZT	ABBRUCH
Kind No., Data No.	14, 02	none

[Call Forward - No Answer]

	ACTION	RESULT
Default text message	CF NANS	
Kind No., Data No.	14, 03	none

	ACTION	RESULT
Deutsche text	RWKEINAW	ABBRUCH
Kind No., Data No.	14, 03	none

(g) Setting and canceling Do Not Disturb

As described in the Call Forwarding section, the text messages “SET” (04,01) and “CANCEL” (04,02) are also used to indicate Do Not Disturb status.

[Do Not Disturb Activated]

	ACTION	RESULT
Default text message	DND	SET
Kind No., Data No.	13, 24	04, 01

	ACTION	RESULT
Deutsche text	RUFABWEI	EIN
Kind No., Data No.	13, 24	04, 01

[Do Not Disturb Canceled]

	ACTION	RESULT
Default text message	DND	CANCEL
Kind No., Data No.	13, 24	04, 02

	ACTION	RESULT
Deutsche text	RUFABWEI	ABBRUCH
Kind No., Data No.	13, 24	04, 02

(h) Complete Table of ACTION Information Messages

ACTION - 1 (Kind Number = 13)

* = Not used

Index	Service - Operation	Default	Deutsche	Kind No.	Data No.
S-7,8,9	Station Hunting	HUNTING	FANGEN	13	1
C-2,3,5	Call Forwarding	FORWARD	WEITERL	13	2
U-1	Uniform Call Distribution	UCD	UCD	13	3
C-6	Call Hold	HOLD	HALTEN	13	4
	Busy	BUSY	BESETZT	13	5
	Transfer	TRANSFER	VERBINDE	13	6
T-2	Three-Way Calling	CONF	KONFEREN	13	7
	Restricted to Proceed	REST	ABGELEHN	13	8
	Error Indication	ERROR	FEHLER	13	9
A-1	Camp-On	CAMP ON	AUFSCHAL	13	10
C-17	Recall after Consultation Hold	RECALL	RÜCKNAHM	13	11
B-3	Operator Busy Verification	OPE BV	ABFRPRÜF	13	12
C-1	Call Back	CALLBACK	RÜCKRUF	13	13
C-7	Call Pickup	PICK UP	HERANHOL	13	14
S-3	Speed Calling - System	SPEED	KURZWAHL	13	15
S-61	Speed Calling Override - System	OVERRIDE	ÜBERSCHR	13	16
S-15	Recall from Serial Call	RECALL	FOLGERF	13	17
O-2	Outgoing Trunk Queuing	OG-Q	WARTSCHL	13	18
P-1	Paging Access	PAGING	PERSRUFE	13	19
R-16	Radio Paging	P BELL	SIGNALRF	13	20
S-24D	Save and Repeat -D	S&R	SPEI&WW	13	21
M-19	Message Reminder	MESSAGE	NACHRERI	13	22
V-2	Voice Call	VOICE	SPRACHRF	13	23
D-11D	Do Not Disturb - D	DND	RUFABWEI	13	24
P-18	Privacy Release	PRIVACY	PRIVAT	13	25
A-30	Automatic/Manual Intercom	ICM	INTERKOM	13	26
C-12	Call Waiting - Terminating	C WAIT	RUFWARTE	13	27
*	Not Answer	NOT ANS	KEINEANT	13	28

ACTION - 1 (Kind Number = 13)

* = Not used

Index	Service - Operation	Default	Deutsche	Kind No.	Data No.
*	Data Release	RELEASE	VERSION	13	29
*	Data Ready	READY	DATA OK	13	30
*	Data Call	D	DATEN	13	31

ACTION - 2 (Kind Number = 14)

* = Not used

Index	Service - Operation	Default	Deutsche	Kind No.	Data No.
E-4D	Exclusive Hold -D	E-HOLD	EXKLHALT	14	0
C-5	Call Forwarding - All Calls	CF ALL	RWALLE	14	1
C-2	Call Forwarding - Busy Line	CF BUSY	RWBESZT	14	2
C-3	Call Forwarding - Don't Answer	CF NANS	RWKEINAW	14	3
*	Data Call - Set	SET	SETZEN	14	4
*	Data Call - Reset	RESET	RÜCKSETZ	14	5
*	Data Call - Data	DATA	DATEN	14	6
O-6	Off-Hook Alarm	OFF ALM	FEHLBEDI	14	7
P-9	Priority Call 1	PRI 1	PRIOR1	14	8
	Priority Call 2	PRI 2	PRIOR2	14	9
	Priority Call 3	PRI3	PRIOR3	14	10
*	Data Call - Wait for Modem to Ans	WAIT	WAMODANT	14	11
*	Data Call - DTE Status	D SUP	DTE ÜB?	14	12
*	Data Call - DTE Status	DTE SUP	DTE ÜB	14	13
A-19	Attendant Night Transfer	TRANSFER	NACHTSCH	14	14
C-29	Call Park	CALLPARK	RF PARK	14	15
U-1	Uniform Call Distribution - Busy Out	BUSY OUT	BGS EXT	14	16
ACD	ACD - Assistance	ASSIST	ACDUNTST	14	17
	ACD - Emergency	HELP	ACDHILFE	14	18
	ACD - Monitor	MONITOR	ACDMONIT	14	19
	ACD - Overflow	OVERFLOW	ACDÜBERL	14	20
	ACD - Stroke Count	STROKE	ACDBELZÄ	14	21

ACTION - 2 (Kind Number = 14)

* = Not used

Index	Service - Operation	Default	Deutsche	Kind No.	Data No.
M-85D	Message Waiting Indication -D	MW	NRWARTET	14	22
M-48	Multi Channel Recording - Record	REC	AUFNAHME	14	24
*	Collect Call	COLLECT	VERBINDG	14	25
M-48	Multi Channel Recording - Replay	REPLAY	WIEDERGA	14	26
A-3	Attendant Console - Operator Transfer	XFR OPE	VERMITTL	14	27
*	DK Operation - Originating Call	ORG	RFERZEU	14	28
*	DK Operation - Answering Call	ANSWER	ANTWORT	14	29
F-31	Follow Phone	SWAP	TAUSCH	14	31

ACTION - 3 (Kind Number = 15)

Index	Service - Operation	Default	Deutsche	Kind No.	Data No.
R-49	Ringer Mute	RINGER	RFTONAUS	15	1
C-28	Call Forwarding - All Calls - Outside	CF-OUT	RW_EXT	15	2
F-17	Function Button User Setting	FUNCTION	FT_BELEG	15	3
D-159	Password Registration -New	NEW	PW NEU	15	5
	Password Registration - Old	OLD	PW ALT	15	6
	Password Registration - Repeat	REPEAT	PW WDH	15	7
I-46	Individual Call Block	C-BLOCK	BLRF ERW	15	8
A-142	Anonymous Call Block	ANON	BLRF KMT	15	9
D-162	Distinctive Ringing List	DR-LIST	BLRF LST	15	10
L-53	Logged Out IP Station - Call Destination	CF LOUT	RWL-AUS	15	11
M-107	Message Audible Alert	MSG ALRT	NR ALARM	15	12
	Message Audible Alert	MSG	NACHRCHT	15	13
L-54	Least Cost Routing for MG-PRI	MAID	MA-IDSET	15	14
D-164D	Day-Night Mode Status -D - Night	NIGHT	NACHT	15	15
	Day-Night Mode Status -D - Day	DAY	TAG	15	16
I-49	Internal Zone Paging - Multi-Zone	M-IZP	MEHRZOSC	15	17
S-168	SP Controlled SIP Terminal	CF OUT	SIPRW EX	15	19

ACTION - 3 (Kind Number = 15)

Index	Service - Operation	Default	Deutsche	Kind No.	Data No.
M-120	Multi-Boss/Multi-Secretary - Boss Absent	ABSENT	ABWESEND	15	23
	Multi-Boss/Multi-Secretary - Boss Present	PRESENT	ANWESEND	15	24
M-118	Mobility Access - Mobility Access Station	MS	MS	15	26

(i) Complete Table of RESULT Information Messages

RESULT (Kind Number = 4)

Frequently Used	Meaning	Default	Deutsche	Kind No.	Data No.
Yes	Function Activated	SET	EIN	4	1
Yes	Function Canceled	CANCEL	ABBRUCH	4	2
Yes	Busy Line	BUSY	BESETZT	4	3
Yes	Restricted/Invalid	REST	RESTRIKT	4	4
No	Wait	WAIT	WARTEN	4	5
No	OK	OK	OK	4	6
No	On	ON	AN	4	7
No	Off	OFF	AUS	4	8
No	Clear	CLEAR	LÖSCHEN	4	9
No	Error	ERROR	FEHLER	4	10
No	Mode	MODE	MODUS	4	13
Not used	Data Call - ER Signal On	PR?	BEREIT?	4	28
	Data Call - Set	SET	EIN	4	29
	Data Call - Not Answered	NOT ANS	KEINE AW	4	30
	Data Call - Established	D	D	4	31

5. Service-Specific Information Message

The following text information messages are dedicated to specific services and they are not shared by other services.

Service-Specific Status Indication 1 (Kind Number = 6, 7, 21, and 22)

Index	Service - Operation	Default	Deutsche	Kind No.	Data No.
C-2,3,5,25	Call Forwarding - Set by Operator	FWD SET OPR	RUFWL VERMITLNG	6	1
F-6	Faulty Trunk Report	BAD CALL NOTIFY	ANRUF FEHLER	6	2
T-62	Timed Reminder	WAKE UP	WECKEN	6	3
	Off-Hook Alert	HANG UP!	AUFLEGEN!	6	4
E-11	Emergency Call	EMERGENCY CALL	NOTRUF	6	5
	Nailed Down Connection (Not Used)	NAILED DOWN CONN	FESTGELEGT VERB	6	6
	Data Connection (Not Used)	EQP NOT READY	EQP NICHT BEREIT	6	7
C-160	Call Trace	CALL REPORT	ANRUF BERICHT	7	0
D-159	DtermIP - Log Out	LOGOUT?	LOGOUT?	7	1
D-165	Dialed Number Privacy Mode - Normal	NORMAL MODE	NORMAL MODUS	7	4
	Dialed Number Privacy Mode - Mode 1	PRIVACY MODE 1	VERDECKT MODUS1	7	5
	Dialed Number Privacy Mode - Mode 2	PRIVACY MODE 2	VERDECKT MODUS2	7	6
L-54	LCR for MG-PRI - MA-ID Set	MAID SET	MASE EIN	21	1
	LCR for MG-PRI - MA-ID Display	MAID	MASE	21	2
	LCR for MG-PRI - MA-ID Error	MAID ERROR	MASE FHLR	21	3
	LCR for MG-PRI - MA-ID Restricted	MAID RESTRICTION	MASE RST	21	4
	LCR for MG-PRI - MA-ID Error	MAID ERROR	MASE FHLR	21	5

Service-Specific Status Indication 1 (Kind Number = 6, 7, 21, and 22)

Index	Service - Operation	Default	Deutsche	Kind No.	Data No.
P-83	Phonebook - Display	Phone book	Telefonbuch	22	1
	Phonebook - Local Phonebook	Local Phone book	Lokales Telefonbuch	22	2
	Phonebook - Add Name	Add name	Name einfuegen	22	3
	Phonebook - Name	Name	Name	22	4
	Phonebook - Number	Number	Nummer	22	5
	Phonebook - Remote Space at the End	Remove space at the end	Leerzeichen loeschen!	22	6
	Phonebook - Entry is Added	Entry is added	Eintrag gesichert	22	7
	Phonebook - Local Phonebook is Full	Local phone book is full	Lokales Telbuch ist voll	22	8
	Phonebook - Entry is Not Added	Entry is not added	Eintrag nicht gesicnert!	22	9
	Phonebook - Entry is Not Found	Entry is not found	Eintrag nicht gefunden!	22	10
	Phonebook - Delete this Entry?	Delete this entry?	Eintrag loeschen?	22	11
	Phonebook - Entry is Deleted	Entry is deleted	Eintrag geloescht!	22	12
	Phonebook - Prefix Number Input?	Prefix Number Input?	Kennziffer eingabe	22	13
	Phonebook - Entry is Edited	Entry is edited	Eintrag geaendert!	22	14
	Phonebook - Entry is Not Edited	Entry is not edited	Eintrag nicht geaendert!	22	15
	Phonebook - Entry Cannot be Deleted	Entry cannot be deleted	Eintrag nicht geloescht!	22	16
	Phonebook - Local Phonebook Empty	Local phone book empty	Lokales Telb ist leer!	22	17
	Phonebook - Edit Name	Edit name	Name aendern	22	18
	Phonebook - Entry is Not Added	Entry is not added	Eintrag nicht gesichert!	22	19
	Phonebook - System Phonebook Empty	Sys phone book empty	Sys Telb ist leer!	22	20
	Phonebook - Search	Search	Suche	22	21

Service-Specific Status Indication 2 (Kind Number = 30, 31, 48, 50, 52, and 53)

Index	Service - Operation	Default	Deutsche	Kind No.	Data No.
x	OAI Service - Read Error	READ ERROR	LESE FEHLER	30	1
x	E_ACD Service - SUP	SUP	SUP	30	2
C-150	Call Block - Entry	BLOCK ENTRY	BLOCK EING	30	3
	Call Block - Clear	BLOCK CLEAR	BLOCK LÖSCH	30	4

Service-Specific Status Indication 2 (Kind Number = 30, 31, 48, 50, 52, and 53)

Index	Service - Operation	Default	Deutsche	Kind No.	Data No.
S-127D	Serial Call -D - Set	SERIAL SET	SERIELL SETZEN	30	5
	Serial Call -D - Canceled	SERIAL CANCEL	SERIELL RSETZEN	30	6
	Serial Call -D - Restricted	RST	EINSCHRÄNKUNG	30	7
C-146	Call Hold - Conference	CALL HOLD CNF	KONF RUF HALTEN	30	8
P-18	Privacy Release - Conference Restricted	CNF RST	KONF EINSCHR	30	9
	Privacy Release - Set	P-RLS SET	PRIV AUS SETZEN	30	10
	Privacy Release - Reset	P-RLS RESET	PRIV AUS RSETZEN	30	11
	Privacy Release - Set	PRIVACY SET	PRIV SETZEN	30	12
	Privacy Release - Canceled	PRIVACY CNCL	PRIV PSETZEN	30	13
	Privacy Release - Wait to Join?	WAIT P-RLS?	WARTE PRIV AUS?	30	14
	Privacy Release - Permit to Join?	P-RLS?	PRIV AUS?	30	15
A-121	Add-On 8 Party Conference - Retain Conf	RETAIN CONF	HALTEN KONF	30	16
	Add-On 8 Party Conference - Disconnect	DISCONNECT CONF	TRENNEN KON	30	17
A-30	Intercom - Intercom Call	ICM	INTERKOM	30	18
	Intercom - Conference Restricted	CNF RST	KONF EINSCHR	30	19
	Intercom - Intercom Call	ICM:	IKM:	30	20
M-48	Multi Channel Recording - Start	RECORDING START	AUFNAHME START	30	21
	Multi Channel Recording - Playing Start	PLAYING START	ABSPIELEN START	30	22
	Multi Channel Recording - End	RECORDING END	AUFNAHME ENDE	30	23
Not Released in EMEA	Voice Mail - REC Denied	REC DENIED	AUFN ABGELEHNT	30	24
	Voice Mail - Message	VM BUSY	NACHRICHT	30	25
	Voice Mail - REC Denied	REC CANCEL	AUFN ABGELEHNT	30	26
	Voice Mail - REC Denied	POST COMPLETE	AUFN ABGELEHNT	30	27
	Voice Mail - VM Busy	OVERRIDE	VM BESETZT	30	28
	Voice Mail - REC Cancel	RECALL	AUFN ABBRUCH	30	29
	Voice Mail - POST Complete	OG-Q	POST KOMPLETT	30	30
M-117	Multilingual Display	ENGLISH	GERMAN	31	0
D-169	DESI-Less Terminal - Name Assign	Push Speed Calling key	KURZWAHLTASTE DRUECKEN	48	0

Service-Specific Status Indication 2 (Kind Number = 30, 31, 48, 50, 52, and 53)

Index	Service - Operation	Default	Deutsche	Kind No.	Data No.
C-195	Call History - No Record in Incoming Call list	No List	KEINE LISTE	50	0
	Call History - Answered Call	(no indication)	(no indication)	52	0
	Call History - Missed Call	!!	!!	52	1
R-60	Redial - Redial	Redial	Wahlwiederholung	53	1
	Redial - No Record in Outgoing Call List	No list	Keine Liste	53	2
	Redial - Delete Outgoing Call Record	Delete	Löschen	53	3

6. Help Messages

Help Messages (Kind Number = 8)

Key Name	Help Text	Kind No.	Data No.
PRESS ANY SOFT KEY	FOR HELP PRESS ANY SOFT KEY FOR HELP	8	0
CALL FORWARD - BUSY	FORWARDS CALLS TO A NEW DESTINATION WHEN LINE IS BUSY -- TO USE: GO OFFHOOK PRESS KEY AND ENTER NEW DESTINATION	8	288
CALL FORWARD - ALL	FORWARDS ALL CALLS TO A NEW DESTINATION TO USE: GO OFF-HOOK PRESS KEY AND ENTER NEW DESTINATION	8	294
EXECUTIVE OVERRIDE	BARGE?IN TO BUSY STATION WARNING TONE IS SENT PRESS KEY WHEN INTERNAL BUSY IS ENCOUNTERED??NOT ENABLED FOR ALL USERS	8	300
CALL WAITING--- CAMPED ON	CALLS MAY BE ANSWERD BY DEPRESSING KEY WHEN A CALL WAITING TONE IS HEARD -- PRESS KEY TO ALTERNATE CALLERS	8	306
CALL BACK -- PRESS KY TO	HAVE A BUSY STATION RING YOUR PHONE WHEN IT BECOMES IDLE CALL BACK IS CANCELLED IF YOUR STATION IS NOT IDLE	8	312
SAVE AND REPEAT - PRESS	AFTER DIALING TO SAVE DIALED NUMBER TEMPORARILY-AFTER SAVING DEPRESS TO DIAL - AFTER USE NUMBER IS ERASED	8	324
MESSAGE WAITING SET	PRESS TO ACTIVATE MW LIGHT AT CALLED STATION USE THE MW-OFF KEY TO CANCEL	8	336
CALL HOLD	DEPRESS KEY WHILE ON A CALL TO PUT CALLER ON HOLD WHEN RELEASING A CURRENT CALL""RECALLED FROM SUSPENDED CALL	8	342
MESSAGE WAITING CAN-CEL	PRESS KEY TO EXTINGUISH A PREVIOUSLY SET MESSAGE WAITING	8	348
CALL PICKUP - GROUP	THIS KEY WILL ALLOW YOU TO ANSWER ANY RINGING STATION WITHIN YOUR DEFINED CALL PICKUP GROUP	8	354
THREE PARTY CONFER-ENCE	WHILE ON A CALL PRESS XFER KEY DIAL 3RD PARTY WHEN READY TO START CONFERENCE DEPRESS THIS KEY	8	360

Help Messages (Kind Number = 8)

Key Name	Help Text	Kind No.	Data No.
VOICE CALL	PRESS KEY AFTER DIALING TO SET A VOICE PATH FROM YOUR STATION TO THE TARGET STATION'S SPEAKERPHONE	8	366
SYSTEM SPEED DIALING	PROVIDES ACCESS TO SYSTEM SPEED DIAL PRESS KEY AND DIAL THE THREE OR FOUR DIGIT SPEED DIAL CODE	8	372
SWITCH HOOK FLASH - CAS	DEPRESSING KEY WHILE ON AN OUTSIDE TRUNK WILL SEND A SWITCH HOOK FLASH THIS KEY IS NORMALLY USED FOR CENTREX XFER	8	384
UCD BUSY OUT	MEMBERS OF A UCD GROUP CAN PRESS THIS KEY TO LOG IN TO OR OUT OF THE UCD CALL HUNT - NORMAL OPERATION CONTINUES	8	396
FLASH	THIS KEY PREFORMS THE SAME FUNCTION AS THE TRANSFER KEY	8	408
CALL FORWARD - NO ANSWER	FORWARDS CALLS TO A NEW DISTINATION WHEN NOT ANSWER -- TO USE: GO OFFHOOK PRESS KEY AND ENTER NEW DESTINATION	8	414
MALICIOUS CALL TRACE	PRESS KEY TO INITIATE CALL TRACE FOR USE ONLY ON ISDN CALLS (AUSTRALIAN)	8	432
GROUP SPEED DIALING	PROVIDES ACCESS TO GROUP SPEED DIAL PRESS KEY AND DIAL THE TWO DIGIT SPEED DIAL CODE	8	438
PRIVACY RELEASE	PRESS KEY TO JOIN A CALL IN PROGRESS OR TO LET ANOTHER PARTY JOIN YOUR CALL BOTH PARTIES MUST BE IN SAME LINE GROUP	8	570
DO NOT DISTURB	PRESSING KEY WHILE IDLE RESTRICTS INCOMING CALLS USING WHILE OFF HOOK SETS PRIVACY PRESS AGAIN TO CANCEL	8	672
MICROPHONE ON/OFF	TURNS THE MICROPHONE ON AND OFF FOR USE WITH SPEAKER-PHONE - MIC LED ILLUMINATES WHEN ON	8	678
BUTTON PAGE	TWO FUNCTIONS MAY BE ASSIGNED TO EACH FEATURE OR SPEED KEY USE THIS KEY TO TOGGLE BETWEEN PAGES	8	702
HEADSET ON OFF	PRESS BUTTON TO GO OFF-HOOK USING THE HEADSET PRESS WHEN CALL IS COMPLATED TO HANG-UP	8	708
RELEASE KEY	PRESS KEY WHEN USING A HEADSET TO RELEASE FROM THE CALL WITHOUT WAITING FOR THE PARTY TO HANG-UP (FOR ACD USE)	8	726
MANUAL SIGNALING	THIS KEY WILL CAUSE A ONE SECOND RING TO OCCUR AT THE PRE-DETERMINED SIGNALING STATION	8	732

Help Messages Kind Number = 9 Part 1 - Data Number 108 - 378

Key Name	Help Text	Kind No.	Data No.
KEEP CONFERENCE	PRESS KEY TO ALLOW ALL OUTSIDE PARTIES TO STAY ON CONFERENCE WHEN ALL INTERNAL PARTIES HANG-UP	9	108
SERIAL CALL	USE PRIOR TO XFER-CAUSES CALLER TO RECALL TO AN AUTOMATED ATTENDANT UPON COMPLETION OF THE FIRST CALL	9	114
INTERNAL ZONE PAGE	PRESS KEY TO VOICE CALL ALL MEMBERS OF YOURS PAGE GROUP - THE PARTY MAY ANSWER BY DEPRESSING THE PRIME LINE KEY	9	120
WAKE UP SET -- SETS AUTO	WAKE UP FOR A GUEST ROOM OR ADMIN STATION. TO SET: PRESS KEY"" ENTER TIME AND STATION"" PRESS ENTER"" THEN END.	9	132
WAKE UP RESET -- CANCELS	AUTO WAKE UP FOR A GUEST ROOM OR ADMIN STATION TO RESET: PRESS KEY"" ENTER STATION"" PRESS ENTER"" THEN END.	9	138
DO NOT DISTURB SET --	SETS DND FOR GUEST ROOM OR ADMIN STATION. TO SET: PRESS KEY"" ENTER STATION"" PRESS ENTER"" THEN END.	9	144
DO NOT DISTURB RESET	SETS DND FOR GUEST ROOM OR ADMIN STATION. TO RESET: PRESS KEY"" ENTER STATION"" PRESS ENTER"" THEN END.	9	150
ROOM CUT OFF SET	TEMPORARILY DISABLES A GUEST ROOM PHONE. TO SET: PRESS KEY"" ENTER STATION"" PRESS ENTER"" THEN END.	9	156
ROOM CUT OFF RESET	TEMPORARILY DISABLES A GUEST ROOM PHONE. TO RESET: PRESS KEY"" ENTER STATION"" PRESS ENTER"" THEN END.	9	162
MESSAGE WAITING SET	SETS MW INDICATION ON A GUEST/ADMIN STATION. TO SET: PRESS KEY"" ENTER STATION"" PRESS ENTER"" THEN END.	9	168
MESSAGE WAITING RESET	CANCELS MW INDICATION ON GUEST/ADMIN STATION. TO RESET: PRESS KEY"" ENTER STATION"" PRESS ENTER"" THEN END.	9	174
CHECK IN	SETS A GUEST ROOM TO CHECK IN STATUS. TO SET: PRESS KEY"" ENTER STATION"" PRESS ENTER"" THEN END.	9	180
CHECK OUT	SETS A GUEST ROOM TO CHECK OUT STATUS. TO SET: PRESS KEY"" ENTER STATION"" PRESS ENTER"" THEN END.	9	186
PLEASE REFER TO	A MANUAL PLEASE REFER TO A MANUAL	9	192
PLEASE REFER TO	A MANUAL PLEASE REFER TO A MANUAL	9	198
ENTER DATA	ALLOWS THE USER TO ENTER DATA TO SET VARIOUS SERVICE FEATURES FOR A GUEST OR ADMINISTRATION STATION.	9	204
CLEAR ENTRY	ALLOWS THE USER TO CANCEL OR CLEAR DATA IF AN ERROR HAS OCCURRED.	9	210
END FUNCTION	ALLOWS THE USER TO FINISH OR COMPLETE AN OPERATION.	9	216
CALLER ID DISPLAY CHANGE	USE KEY TO TOGGLE TO BETWEEN STANDARD DISPLAY AND ALTERNATE DISPLAY	9	288
CALL PICKUP - GROUP	THIS KEY WILL ALLOW YOU TO ANSWER ANY RINGING STATION WITHIN YOUR EXPANDED CALL PICKUP GROUP	9	300

Help Messages Kind Number = 9 Part 1 - Data Number 108 - 378

Key Name	Help Text	Kind No.	Data No.
CALL HOLD CONFER-ENCE	WHEN CALL IS IN PROGRESS AND A CALL IS ON HOLD ON OTHER LINE PRESS XFER THIS KEY-LINE NUMBER TO CREATE THREE WAY CNF	9	306
RINGER MUTE	PRESS THIS KEY TO ELIMINATE THE AUDIBLE RINGING TO RESTORE THE RINGING PRESS KEY AGAIN	9	312
NEXT PAGE OF DISPLAY	PRESS THIS KEY TO SCROLL TO THE NEXT DISPLAY SCREEN USE THE KEY TO RETURN TO PREVIOUS DISPLAY SCREEN	9	330
PREVIOUS PAGE OF DIS-PLAY	PRESS THIS KEY TO SCROLL TO THE PREVIOUS DISPLAY - USE THE TO SCROOL TO THE NEXT PAGE	9	336
PLEASE REFER TO	A MANUAL PLEASE REFER TO A MANUAL	9	348
PLEASE REFER TO	A MANUAL PLEASE REFER TO A MANUAL	9	354
HEADSET ON OFF	PRESS BUTTON TO GO OFF-HOOK USING THE HEADSET PRESS WHEN CALL IS COMPLETED TO HANG-UP	9	360
PS KEY	FORWARDS ALL CALLS TO PS PS KEY FORWARDS ALL CALLS TO PS	9	366
BOSS MOVE	IN BOSS-SEC SERVICE WHEN THE SEC PUSHES BOSS LINE THE STA-TION ASSIGNED IN BOSS STATION BEFOREHAND IS CALLED	9	372
CALL BLOCK	INCOMING CALL RESTORICT WHEN A REGISTRATION NUMBER	9	378

Help Messages Kind Number = 9 Part 2 - Data Number 384 - 762

Key Name	Help Text	Kind No.	Data No.
CALL TRACE	THIS FUNCTION REPORT INFORMATION OF LAST CALLING NUM-BER TO ADMINISTRATOR	9	384
DISC KEY	PRESS THIS KEY TO RELEASE THE LAST CALLED PARTY AND RECONNECT TO THE HELD PARTY	9	396
CALL REDIRECT	THIS FUNCTION SENDS A CALL IMMEDIATELY FROM A RINGING STATION TO THE CALL FORWARD NO ANSWER DESTINATION	9	402
REMOTE HOLD	THIS FUNCTION PLACES A TRANSFERED CALL ON HOLD AT THE DESTINATION STATION	9	408
DESIGNATE GROUP PICUP	IT CAN RESPOND TO RINGI -NG OTHER PICUP GROUP -S. IT IS POSSI-BLE BY YOU PUSH THIS KEY AND DIALING A GROUP NO.	9	414
PLEASE REFER TO	A MANUAL PLEASE REFER TO A MANUAL	9	426
PLEASE REFER TO	A MANUAL PLEASE REFER TO A MANUAL	9	432
CALL FORWARD - STA	FORWARDS STA CALLS TO A NEW DESTINATION TO USE: GO OFF-HOOK PRESS KEY AND ENTER NEW DESTINATION	9	438

Help Messages Kind Number = 9 Part 2 - Data Number 384 - 762

Key Name	Help Text	Kind No.	Data No.
CALL FORWARD - COT	FORWARDS OUT TRUNK TO A NEW DESTINATION TO USE: GO OFF-HOOK PRESS KEY AND ENTER NEW DESTINATION	9	444
LOGOUT KEY	WHEN THIS KEY IS PRESSED 2 TIMES IN 5 SEC"" THIS STATION WILL BE LOGOUT.	9	462
EXPANDED SPEED DIAL-ING	PROVIDES ACCESS TO EXPANDED SPEED DIAL PRESS KEY AND DIAL THE TWO DIGIT SPEED DIAL CODE	9	486
CALL FORWARD - LOGOUT	FORWARDS CALLS TO A NEW DESTINATION WHEN STA LOGOUT -- TO USE: GO OFFHOOK PRESS KEY AND ENTER NEW DESTINATION	9	498
VOICE MAIL MESSAGE ALERT	PRESS THIS KEY TO ENABLE AUDIBLE MESSAGE ALERT TO CAN-CEL PRESS THIS KEY AGAIN	9	504
SET OF MA-ID	SELECT OPTIMAL ROUTE FROM MA?ID WHEN OUTGOING CALLS: GO OFF-HOOK PRESS KEY AND ENTER NEW MA-ID	9	516
BY PRESSING THIS KEY""	WHEN THERE IS A CALLING TO BOSS"" BOSS S STATI ON WILL BE RINGING AND SECRETARY S STATION WILL BE NOT RINGING	9	522
MULTI-ZONE IZP	PRESSING THIS KEY IZP S GROUPS 1-4 TO USE GO OFF?HOOK AND PRESS THE MIZP KEY	9	534
CALL PARK DELUXE-1	THIS KEY WILL ALLOW YOU TO HOLD CURRENT CALL TO YOUR DEFINED CALLPARK DELUXE GROUP1 AND ANSWER IT`S HOLDING CALL	9	558
CALL PARK DELUXE-2	THIS KEY WILL ALLOW YOU TO HOLD CURRENT CALL TO YOUR DEFINED CALLPARK DELUXE GROUP2 AND ANSWER IT`S HOLDING CALL	9	564
CALL PARK DELUXE-3	THIS KEY WILL ALLOW YOU TO HOLD CURRENT CALL TO YOUR DEFINED CALLPARK DELUXE GROUP3 AND ANSWER IT`S HOLDING CALL	9	570
CALL PARK DELUXE-4	THIS KEY WILL ALLOW YOU TO HOLD CURRENT CALL TO YOUR DEFINED CALLPARK DELUXE GROUP4 AND ANSWER IT`S HOLDING CALL	9	576
CALL PARK DELUXE-5	THIS KEY WILL ALLOW YOU TO HOLD CURRENT CALL TO YOUR DEFINED CALLPARK DELUXE GROUP5 AND ANSWER IT`S HOLDING CALL	9	582
Privacy Mode	By pressing this button the number dialed will be shown or hidden and stored or not stored in LNR Memory *=Not stored	9	588
SPEED DIAL NAME ASSIGN	Used to program name to SPD key to program: press name button then SPD. Enter Info via key pad then press SET.	9	600
MOBILE SERVICE	DISPLAY FORWARDED NUMBER IF MA KEY IS PUSHED MOBILE SERVICE CANCELLED IF MA KEY IS PUSHED IN STATE OF OFF-HOOK	9	612
LANGUAGE	LANGUAGE DISPLAYED CAN BE CHANGED LANGUAGE LANGUAGE DISPLAYED CAN BE CHANGED	9	672

Help Messages Kind Number = 9 Part 2 - Data Number 384 - 762

Key Name	Help Text	Kind No.	Data No.
CALLERS LIST	Incoming Calls History will be displayed. CALLERS LIST Incoming Calls History will be displayed.	9	678
Phone book	There is a Local and a System phone book Phone book There is a Local and a System phone book	9	690
B&W Reverse	By depressing this key"" B&W of display will reverse.	9	750
MOBILE SERVICE	SET DATA WILL BE CLEARED IF A BUTTON IS PUSH AFTER AN OFF-HOOK WHEN MOBILE SERVICE DATA SET	9	762

Help Messages (Deutsche) (Kind Number = 8)

Key Name	Help Text	Kind No.	Data No.
FÜR HILFE BELIEBIGE	FUNKTIONSTASTE DRÜCKEN FÜR HILFE BELIEBIGE FUNKTIONSTASTE DRÜCKEN	8	0
RUFWEITERLEITUNG BESETZT	RUFWEITERLEITUNG BEI BESETZT AN NEUE ZIELSTELLE - EIN: HÖRER ABHEBEN TASTE DRÜCKEN NEUE ZIELSTELLE EINGEBEN	8	288
RUFWEITERLEITUNG - ALLE	WEITERLEITUNG ALLER RUFE AN NEUE ZIELSTELLE - EIN: HÖRER ABHEBEN TASTE DRÜCKEN NEUE ZIELSTELLE EINGEBEN	8	294
AUFSCHALTEN	AUFSCHALTEN MIT WARNTON AUF BESETZTEN APPARAT TASTE DRÜCKEN BEI INTERN BESETZT - NICHT FÜR ALLE TEILNEHMER VERFÜGBAR	8	300
ANRUF WARTET	WEITEREN RUF (WARTETON) MIT DRÜCKEN DER TASTE HERANHOLEN - ZWISCHEN TEILNEHMERN MIT DRÜCKEN DER TASTE WECHSELN	8	306
RÜCKRUF	TASTE DRÜCKEN: RÜCKRUF WENN FREI VON GEGENSTELLE - RÜCKRUF WIRD GELÖSCHT FALLS IHR ANSCHLUSS BESETZT IST	8	312
SPEICHERN/WIEDERHOLEN	GEWÄHLTE NUMMER SPEICERN UND SPÄTER WIEDER WÄHLEN - FÜR ANWÄHLEN DER RUFNUMMER TASTE ERNEUT DRÜCKEN	8	324
MITTEILUNG WARTET	DRÜCKEN DER TASTE AKTIVIERT MW-LEUCHTE AM ENDGERÄT DES GERUFENEN TEILNEHMERS - LÖSCHEN ÜBER MW-AUS-TASTE	8	336
GESPRÄCH PARKEN	GESPRÄCH MIT DRÜCKEN DER TASTE PARKEN ZURÜCK ZU GEPARKTEM RUF: HÖRER ABHEBEN LINE-TASTE (BLINKT) DRÜCKEN	8	342
MITTEILUNG WARTET AUS	MIT DRÜCKEN DER TASTE WIRD EINE ZUVOR AKTIVIERTE ANZEIGE MITTEILUNG WARTET GELÖSCHT	8	348
GRUPPENRUF HERANHOLEN	HERANHOLEN EINES EINTREFFENDEN RUFES AN EIN MITGLIED IHRER GRUPPE: TASTE DRÜCKEN	8	354
DREIERKONFERENZ	TRANSFER-TASTE WÄHREND GESPRÄCH DRÜCKEN DRITTEN TEILNEHMER ANWÄHLEN UND DIESE TASTE BETÄTIGEN	8	360

Help Messages (Deutsche) (Kind Number = 8)

Key Name	Help Text	Kind No.	Data No.
SPRACHANRUF	DRÜCKEN DER TASTE NACH DEM WÄHLEN BAUT EINEN SPRECH-WEG VON IHREM ENDGERÄT ZUM LAUTSPRECHER DER ZIEL-STELLE AUF	8	366
KURZWAHL- SYSTEM	ZUGRIFF AUF DIE SYSTEMKURZWAHL TASTE DRÜCKEN UND DREIODER VIERSTELLIGEN KURZWAHL-CODE EINGEBEN	8	372
SWITCH HOOK FLASH - CAS	MIT DRÜCKEN DER TASTE WÄHREND EINER EXTERNVERBINDUNG WIRD EIN SWITCHHOOK-FLASH GESENDET (FÜR CENTREX)	8	384
UCD BESETZT - AUS	AN-/ABMELDUNG FANGEN FÜR UCD-MITGLIEDER MIT DRÜCKEN DER TASTE - OHNE EINFLUSS AUF DEN NORMALBETRIEB	8	396
FLASH-TASTE	DIE FUNKTION DIESER TASTE ENTSPRICHT DER FUNKTION DER TRANSFER-TASTE	8	408
RUFWEITERLEITUNG-FREI	WEITERLEITUNG ALLER RUFE AN NEUE ZIELSTELLE EIN: HÖRER ABHEBEN - TASTE DRÜCKEN - NEUE ZIELSTELLE EINGEBEN	8	414
FANGEN	FANGEN BÖSWILLIGER ANRUFER STARTEN: TASTE DRÜCKEN (NUR FÜR ISDN-RUFE)	8	432
KURZWAHL GRUPPE	ZUGRIFF AUF DIE GRUPPENKURZWAHL TASTE DRÜCKEN UND ZWEISTELLIGEN KURZWAHLCODE EINGEBEN	8	438
PRIVAT-FREIGABE	PRIVAT-FREIGABE FÜR MITHÖREN LAUFENDER GESPRÄCHE - ALLE TEILNEHMER MÜSSEN IN GLEICHER GRUPPE SEIN	8	570
RUFABWEISUNG	ANKOMMENDE ANRUFE ABWEISEN PRIVACY-FUNKTION: HÖRER ABHEBEN - TASTE DRÜCKEN AUS: TASTE DRÜCKEN	8	672
MIKROFON EIN/AUS	EIN-/AUSSCHALTEN DES MIKROFONS FÜR LAUTSPRECHERBETRIEB - EINGESCHALTET: ROTE LED RECHTS UNTEN LEUCHTET	8	678
FUNKTIONSEBENE WECHSELN	FEATURE- UND KURZWAHLTASTEN SIND JE ZWEI FUNKTIONEN ZUWEISBAR FÜR WECHSEL FUNKTIONSEBENE: TASTE DRÜCKEN	8	702
HEADSET EIN - AUS	TASTE DRÜCKEN: HÖRER ABNEHMEN ÜBER HEADSET GESPRÄCH BEENDEN: TASTE ERNEUT DRÜCKEN	8	708
AUSLÖSE-TASTE	BEI NUTZUNG EINES HEADSETS TASTE DRÜCKEN UM VERBIND-UNG VOR AUFLEGEN DER GEGENSTELLE AUSZULÖSEN (FÜR ACD)	8	726
1-SEKUDEN RUF	MIT DRÜCKEN DIESER TASTE ERFOLGT EIN 1-SEKUNDEN-RUFTON AN EINE VORAB DEFINIERTE ZIELSTELLE	8	732

Help Messages (Deutsche) Kind Number = 9 Part 1 - Data Number 108 - 378

Key Name	Help Text	Kind No.	Data No.
KONFERENZ HALTEN	EXTERNE TEILNEHMER KÖNNEN KONFERENZ AUCH WEITER-FÜHREN WENN ALLE INTERNEN TEILNEHMER AUFGELEGT HABEN	9	108
KETTENGESPRÄCH	NUTZUNG VOR RUFWEITERLEITUNG - WIEDERHERBEIRUF AN AUTO-ATTENDANT NACH BEENDIGUNG DES ERSTEN GESPRÄCHS	9	114
PAGING-GRUPPE INTERN	TASTE DRÜCKEN FÜR SPRACHRUF AN ALLE MITGLIEDER DER EIGENEN PAGING-GRUPPE - ANTWORT ÜBER PRIME-LINE-TASTE	9	120
WECKRUF EIN: AKTIV-IERT	WECKRUF FÜR NST IN GÄSTE ZIMMER / VERWALTUNG EIN: TASTE DRÜCKEN"" UHR ZEIT UND NST EINGEBEN ENTER DRÜCKEN BEEN-DEN	9	132
WECKRUF AUS: LÖSCHT	WECKRUF FÜR NST IN GÄSTE ZIMMER / VERWALTUNG LÖSCHEN: TASTE DRÜCKEN"" NST EINGEBEN"" ENTER DRÜCKEN UND BEEN-DEN.	9	138
RUFABWEISUNG - EIN	AKTIVIERT RABW FÜR GÄSTE ZIMMER / VERWALTUNG EIN: TASTE DRÜCKEN"" NST EINGEBEN"" ENTER-TASTE DRÜCKEN UND BEEN-DEN.	9	144
RUFABWEISUNG - AUS	DEAKTIVIERT RABW FÜR GÄS TEZIMMER / VERWALTUNG AUS: TASTE DRÜCKEN"" NST EINGEBEN"" ENTER-TASTE DRÜCKEN UND BEENDEN.	9	150
ZIMMER SPERREN	DEAKTIVIERT TELEFON IM GÄSTEZIMMER TEMPORÄR EIN: TASTE DRÜCKEN"" NST EINGEBEN"" ENTER-TASTE DRÜCKEN UND BEEN-DEN.	9	156
ZIMMER FREIGEBEN	REAKTIVIERT TELEFON IM GÄSTEZIMMER LÖSCHEN: TASTE DRÜCKEN"" NST EINGEBEN"" ENTERTASTE DRÜCKEN""BEENDE	9	162
NACHRICHT WARTET - EIN	AKTIVIERT MW-ANZEIGE AN NST GAST/VERWALTUNG. EIN: TASTE DRÜCKEN"" NST EINGEBEN"" ENTER-TASTE DRÜCKEN UND BEEN-DEN.	9	168
NACHRICHT WARTET - AUS	LÖSCHT MW-ANZEIGE AN NST GAST/VERWALTUNG. LÖSCHEN: TASTE DRÜCKEN"" NST EINGEBEN"" ENTER-TAS TASTE DRÜCKEN""BEENDE	9	174
CHECK IN	SETZT GÄSTEZIMMER AUF STATUS CHECK-IN. TASTE DRÜCKEN"" NST EINGEBEN"" ENTER-TASTE DRÜCKEN"" BEENDEN	9	180
CHECK-OUT	SETZT GÄSTEZIMMER AUF STATUS CHECK-OUT. TASTE DRÜCKEN"" NST EINGEBEN"" ENTER-TASTE DRÜCKEN"" BEENDEN	9	186
BITTE SCHLAGEN SIE	IM HANDBUCH NACH BITTE SCHLAGEN SIE IM HANDBUCH NACH	9	192
BITTE SCHLAGEN SIE	IM HANDBUCH NACH BITTE SCHLAGEN SIE IM HANDBUCH NACH	9	198
DATEN EINGEBEN	ERMÖGLICHT DEM TEILNEHMER VERSCHIEDENE LEISTUNGS-MERKMALE FÜR GÄSTEODER VERWALTUNGSNST ZU AKTIVIEREN	9	204
EINTRAG LÖSCHEN	ERMÖGLICHT DEM TEILNEHMER BEI AUFTRETEN EINES FEHLERS EINGEGEBENE DATEN ZU LÖSCHEN	9	210

Help Messages (Deutsche) Kind Number = 9 Part 1 - Data Number 108 - 378

Key Name	Help Text	Kind No.	Data No.
FUNKTION ENDE	ERMÖGLICHT DEM TEILNEHMER EINE FUNKTION ZU BEENDEN ODER ZU VERVOLLSTÄNDIGEN.	9	216
ANZEIGE ANRUFER ÄNDERN	TASTE DRÜCKEN UM ZWISCHEN STANDARDANZEIGE DER ANRUFERKENNUNG UND ALTERNATIVER ANZEIGE UMZUS-CHALTEN	9	288
GRUPPENRUF HERANHO-LEN	HERANHOLEN VON RUFEN AN ENDGERÄTE INNERHALB DER EIGENEN ERWEITERTEN RUFGRUPPE ÜBER DRÜCKEN DER TASTE	9	300
RUF PARKEN - KON-FERENZ	EINRICHTEN EINER DREIERKONFERENZ MIT BESTEHENDER GESPRÄCHSVERBINDUNG UND GEPARKTEM RUF (ÜBER TASTE LINE-NR)	9	306
TONRUF STUMMSCHALTEN	AKUSTISCHER RUFTON AUS: TASTE DRÜCKEN AKUSTISCHER RUF-TON EIN: TASTE ERNEUT DRÜCKEN	9	312
ANZEIGE NÄCHSTE SEITE	SCROLLEN AUF NÄCHSTE SEITE: TASTE DRÜCKEN ZURÜCK AUF VORHERIGE SEITE: - TASTE DRÜCKEN	9	330
ANZEIGE VORHERIGE SEITE	SCROLLEN AUF VORHERIGE SEITE: TASTE DRÜCKEN ZUR NACH-STEN SEITE: -TASTE DRÜCKEN	9	336
BITTE SCHLAGEN SIE	IM HANDBUCH NACH BITTE SCHLAGEN SIE IM HANDBUCH NACH	9	348
BITTE SCHLAGEN SIE	IM HANDBUCH NACH BITTE SCHLAGEN SIE IM HANDBUCH NACH	9	354
HEADSET EIN/AUS	GESPRÄCH ANNEHMEN: TASTE DRÜCKEN GESPRÄCH BEENDEN: TASTE ERNEUT DRÜCKEN	9	360
PS-TASTE	LEITET ALLE RUFE AN PS WEITER PS-TASTE LEITET ALLE RUFE AN PS WEITER	9	366
CHEF WAHL	RUF DER VOREINGESTELLTEN CHEF-NST WENN BEI AKTIVER CHEF-SEK SCHALTUNG SEK DIE CHEF-TASTE DRÜCKT	9	372
RUF SPERREN	EINTREFFENDEN RUF SPERREN WENN ER VON EINGETRAGENER RUFNUMMER AUS ERFOLGT	9	378

Help Messages (Deutsche) Kind Number = 9 Part 2 - Data Number 384 - 762

Key Name	Help Text	Kind No.	Data No.
FANGEN	DIESE FUNKTION MELDET DIE DATEN DER LETZTEN ANRUFENDEN RUFNUMMER AN DEN ADMINISTRATOR	9	384
VERBINDUNG AUFHEBEN	LÖST VERBINDUNG ZUM ZULETZT GESPROCHENEN TEILNEHMER AUS - STELLT VERBINDUNG MIT GEPARKTEN TEILNEHMER HER	9	396
RUF WEITERSCHALTEN	DIESE FUNKTION SCHALTET EINEN RUF VON EINER ANGERUFENEN NST AN DIE ZIELSTELLE RUFWEITERSCH. KEINE ANTWORT WEITER	9	402
REMOTE HALTEN	STELLT EINEN WEITERGELEITETEN RUF AUF DER ZIEL-NST IM STATUS "" HALTEN"" ZU.	9	408
GRUPPE FÜR PICK WÄHLEN	HERANHOLEN VON RUFEN FÜR ANDERE GRUPPEN DURCH DRÜCKEN DER TASTE UND WAHL DER GRUPPEN-NR.	9	414
BITTE SCHLAGEN SIE	IM HANDBUCH NACH BITTE SCHLAGEN SIE IM HANDBUCH NACH	9	426
BITTE SCHLAGEN SIE	IM HANDBUCH NACH BITTE SCHLAGEN SIE IM HANDBUCH NACH	9	432
RUF WEITERLEITEN - NST	LEITET NST-RUFE AN EINE NEUE ZIELSTELLE EIN: HÖRER ABNEHMEN - TASTE DRÜCKEN - NEUE ZIELSTELLE EINGEBEN.	9	438
RUF WEITERLEITEN - AMT	LEITET AMTSLEITUNGSRUF AN NEUE ZIELSTELLE EIN: HÖRER ABNEHMEN - TASTE DRÜCKEN - NEUE ZIELSTELLE EINGEBEN.	9	444
LOGOUT-TASTE	WIRD DIESE TASTE ZWEIMAL INNERHALB VON 5 SEKUNDEN GEDRÜCKT"" SO ERFOLGT LOGOUT FÜR DIESE NEBENSTELLE.	9	462
ERWEITERTE KURZ-WAHL	BIETET ZUGRIFF AUF ERWEITERTE KURZWAHL TASTE DRÜCKEN UND ZWEISTELLIGEN KURZWAHLCODE WÄHLEN	9	486
RUF WEITERLEITEN-LOGOUT	LEITET RUFE AN EINE NEUE ZIELSTELLE WENN NST LOGOUT - EIN: HÖRER ABNEHMEN - TASTE DRÜCKEN ZIELSTELLE EINGEBEN.	9	498
VM-NACHRICHT - ALARM	TASTE DRÜCKEN"" UM EINEN AKUSTISCHEN ALARM ZU AKTIVIEREN - TASTE ERNEUT DRÜCKEN"" UM AKTI VIERUNG ZU LÖSCHEN.	9	504
EINSTELLUNG ZS-ID	OPTIMALE ROUTE VON ZS-ID WÄHLEN BEI ABGEHENDEN RUFEN: HÖRER ABNEHMEN - TASTE DRÜCKEN UND NEUE ZS-ID EINGEBEN.	9	516
SEKRETARIAT ABWESEND	DRÜCKEN DER TASTE FÜHRT BEI RUFEN AN CHEF"" DASS DIE CHEF-NST UND NICHT MEHR DIE SEK-NST GERUFEN WIRD	9	522
MULTI-ZONE IZP	MIZP-TASTE DRÜCKEN.	9	534
RUF PARKEN DELUXE-1	TASTE DRÜCKEN: PARKEN DES MOMENTANEN RUFS AUF DEFINIERTER CALLPARK DELUXE GRUPPPE1 UND GEPARKTEN RUF ANNEHMEN.	9	558
RUF PARKEN DELUXE-2	TASTE DRÜCKEN: PARKEN DES MOMENTANEN RUFS AUF DEFINIERTER CALLPARK DELUXE GRUPPPE2 UND GEPARKTEN RUF ANNEHMEN.	9	564
RUF PARKEN DELUXE-3	TASTE DRÜCKEN: PARKEN DES MOMENTANEN RUFS AUF DEFINIERTER CALLPARK DELUXE GRUPPPE3 UND GEPARKTEN RUF ANNEHMEN.	9	570

Help Messages (Deutsche) Kind Number = 9 Part 2 - Data Number 384 - 762

Key Name	Help Text	Kind No.	Data No.
RUF PARKEN DELUXE-4	TASTE DRÜCKEN: PARKEN DES MOMENTANEN RUFS AUF DEFINIERTER CALLPARK DELUXE GRUPPPE4 UND GEPARKTEN RUF ANNEHMEN.	9	576
RUF PARKEN DELUXE-5	TASTE DRÜCKEN: PARKEN DES MOMENTANEN RUFS AUF DEFINIERTER CALLPARK DELUXE GRUPPPE5 UND GEPARKTEN RUF ANNEHMEN.	9	582
PRIVAT-MODUS	TASTE DRÜCKEN: DIE GEWÄHLTE RUFNUMMER WIRD ANGEZEIGT/ VERBORGEN UND IM LNR-SPEICHER GESPEICHERT/NICHT GESPE-ICHERT.	9	588
NAME AUF KURZWAHL-TASTE	NAME EINER KURZWAHLTASTE ZUWEISEN: TASTE NAME DRÜCKEN DANN KURZWAHLTASTE - DATEN EINGEBEN - SET DRÜCKEN.	9	600
MOBILER DIENST	ANZEIGE UMGELEITETE NR. WENN MA-TASTE GEDR. MOBILER DIENST AUS WENN MATASTE GEDR. UND HOERER ABGEHOBEN	9	612
SPRACHE	ANGEZEIGTE SPRACHE IST EINSTELLBAR SPRACHE ANGEZEIGTE SPRACHE IST EINSTELLBAR	9	672
ANRUFER LISTE	EINGEGANGENE ANRUFE WERDEN ANGEZEIGT ANRUFER LISTE EINGEGANGENE ANRUFE WERDEN ANGEZEIGT	9	678
Telefonbuch	Lokales und System Telefonbuch Telefonbuch Lokales und System Telefon-buch	9	690
Invertierte Anzeige	Durch Dr Taste wird die DisplayDarstellung invertiert	9	750
MOBILE SERVICE	SET DATA WILL BE CLEARED IF A BUTTON IS PUSH AFTER AN OFF-HOOK WHEN MOBILE SERVICE DATA SET	9	762

7. Trunk Class Numbers and Abbreviations

The trunk class indicates the type of trunk line for public or tie-lines. The trunk type is programmed as a numerical parameter called Trunk Class (TCL) in the "Route Class" data with the maintenance command ARTD. The following table indicates TCL number and the abbreviation of the trunk line.

Note that the abbreviation cannot exceed four characters in length.

Trunk Class Numbers and Abbreviations

Trunk Class	Meaning	Default	Deutsche	Kind No.	Data No.
1	Direct Distance Dialed (DDD) line	DDD	DDD	2	1
2	Foreign eXchange line	FX	FX	2	2
3	Wide Area Telephone Service line	WATS	WATS	2	3
4	Tie line	TIE	TIE	2	4
5	Common Channel Switching Arrangement	CCSA	CCSA	2	5
6	Toll terminal	TOLL	TOLL	2	6
7	Centralized Attendant Service (CAS) line	SATE	SATE	2	7
8	Paging	PG	PG	2	8
9	Code call trunk			2	9
10	Dictation trunk			2	10
11	General paging	PG	PG	2	11

8. Annex

a. List of Kind Number (Part 1)

Kind No.	Meaning	Note
1	Language information (number, name, version)	Only version number can be changed.
2	Trunk class numbers and abbreviations	Section 7.
3	On-call operation 4-character result information message (old)	Not used in Europe for Dterm series III or older
4	On-call operation 8-character result information message	Section 4. - (i)
5	12-character information message	OPR only
6	Service-specific 16-character information message - A	Section 5.
7	Service-specific 16-character information message - B	Section 5.
8	HELP messages for function keys - basic	Section 6.
9	HELP messages for function keys - extended	Section 6.
10	Service-specific 4-character information message - A (old)	Not used in Europe for Dterm series III or older
11	Service-specific 4-character information message - A (old)	Not used in Europe for Dterm series III or older
12	Service-specific 4-character information message - A (old)	Not used in Europe for Dterm series III or older
13	On-call (Call) operation 8-character information message - A	Section 4.
14	On-call (Call) operation 8-character information message - B	Section 4.
15	On-call (Call) operation 8-character information message - C	Section 4.
16	Soft key label for function keys - basic	Section 1.
17	Soft key label for function keys - extended	Section 1.
18	Off-call (Non-call) operation information message - server	Section 3.
19	Cursor operation - up/down	Section 3.
20	Service-specific 16-character information message - Message Area ID (old)	Not used in Europe for Dterm series III or older
21	Service-specific 24-character information message - Message Area ID	Section 5.
22	Service-specific information message - Phonebook	Section 5.
23	Service-specific Soft key label - Phonebook	Section 1.
24	Service-specific Soft key label - Call History, Prefix and Phonebook	Section 1.
25	Service-specific information message - Call History and Prefix	Section 5.
26	DT300 terminal menu text	Section 3.
27	DT300 short cut menu text	Section 3.

b. List of Kind Number (Part 2)

Kind No.	Meaning	Note
28	Service-specific information message - Preset Dial	Section 5.
29	Service-specific Soft key label - Preset Dial	Section 1.
30	Service-specific status indication	Section 5.
31	Service-specific information message - Language Select by Terminal	Section 5.
32	Service-specific Soft key label - Language Select by Terminal	Section 1.
33	SIP multi-line terminal login error - Override?	Section 3.
34	SIP multi-line terminal login error - Illegal LENS	Section 3.
35	SIP multi-line terminal login error - Not Acceptable Here	Section 3.
36	SIP multi-line terminal login error - Illegal Equipment Type	Section 3.
37	SIP multi-line terminal login error - PH Partially Available	Section 3.
38	SIP multi-line terminal login error - Exceeded Capability	Section 3.
39	SIP multi-line terminal login error - Resource Unavailable	Section 3.
40	SIP multi-line terminal login error - Parse Error	Section 3.
41	SIP multi-line terminal login error - Error Code	Section 3.
42	SIP multi-line terminal login error - Soft key	Section 3.
43	SIP multi-line terminal login error - Display	Section 3.
44	Month name	Section 2.
45	Day of week	Section 2.
46	Service-specific information message - Intercom	Section 5.
47	Service-specific Soft key label - SPD name character select	Section 1.
48	Service-specific information message - SPD name	Section 5.
49	Service-specific Soft key label - SPD name - input	Section 1.
50	Service-specific information message - Call History	Section 5.
51	Service-specific Soft key label - Call History (Default)	Section 1.
52	Service-specific information message - Call History - modified	Section 5.
53	Service-specific information message - Redial	Section 5.
54	Service-specific Soft key label - Redial	Section 1.

UNIVERGE SV9500
Data Programming Manual - Business

NWD-165800-001

Revision Sheet

V5: DATE OCTOBER, 2017

FEATURE LIST

2,3,7,22

CHAPTER 2

42,85,86,87,104

CHAPTER 3

274,275,276,277,278,279,280,
281,282,283,284,285,286,287,
288,289,290,291,292,314,457,
458,459,594,596,617,618,619,
620,621,622,623,624,625,641,
642,643,644,645,646,647,648,
649,715,716,717,737,738,739,
740,741,795,798,800,830,842,
859,860,861,863,865,904,982,
983,1085,1090,1091,1152,1158,
1178,1179,1183,1185,1206,
1384,1385,1387,1404,1405,
1419,1421,1494,1496,1498,
1500,1605,1717,1720,1896,
1908,1911,1915,1980,1983,
1996,2012,2017,2018,2019,
2020,2021,2024,2030,2033,
2034,2036,2037,2040,2165,
2166,2167,2168,2169,2170,
2172,2176,2177,2178,2179,
2181,2195,2207,2211,2284,
2431,2432,2437,2440,2444,
2519,2558,2600,2602,2603,
2611,2613,2614,2615,2624,
2628,2633,2634,2639,2640,
2641,2643,2644,2657,2661

V4: DATE JULY, 2017

FEATURE LIST

6,20,22

CHAPTER 2

5,6,10,11,12,13,15,19,24,29,31,
76,169,173

CHAPTER 3

209,210,211,227,228,244,247,
248,270,368,428,438,466,467,
473,474,475,511,552,565,566,
567,568,569,570,571,572,573,
574,575,576,577,578,579,580,
581,582,583,601,607,608,610,
611,651,652,653,684,698,699,
700,706,707,708,760,819,911,
938,951,953,954,955,1187,1211,
1212,1291,1305,1426,1427,
1428,1429,1430,1431,1432,
1459,1462,1466,1467,1468,
1471,1472,1473,1475,1477,
1478,1506,1549,1690,1691,
1698,1731,1732,1733,1734,
1739,1741,1742,1743,1745,
1853,1880,1896,1897,1919,
1986,1987,1988,1989,1990,
1991,1992,1993,1994,1995,
1996,1997,1998,1999,2000,
2001,2002,2003,2004,2005,
2006,2007,2008,2009,2010,
2011,2109,2117,2130,2131,2132,
2133,2134,2135,2136,2137,
2138,2139,2140,2141,2142,
2143,2144,2145,2146,2147,
2148,2156,2159,2173,2174,
2176,2179,2180,2181,2182,
2195,2199,2201,2202,2203,
2205,2206,2207,2208,2212,
2215,2251,2252,2273,2274,
2276,2310,2311,2502,2531,
2574,2575,2620,2621,2622,
2623,2624,2626,2627,2628

APPENDIX C

33

V4: DATE MARCH, 2017

FEATURE LIST

2, 6, 7, 8, 11, 12, 13, 15, 17, 18,
19, 22, 23, 24, 25

CHAPTER 2

5, 13, 14, 15, 16, 40, 41, 42, 43,
44, 46, 60, 62, 85, 93, 100, 101,
120, 121, 122, 123, 124, 125,
127, 128, 129, 130, 131, 132,
133, 134, 171, 172, 173, 174,
175, 179, 180, 181, 182, 183,
184, 185

CHAPTER 3

241, 259, 260, 261, 285, 286,
296, 342, 344, 346, 356, 357,
362, 363, 367, 370, 371, 427,
435, 436, 437, 439, 440, 459,
460, 466, 467, 468, 570, 601,
602, 603, 604, 605, 606, 607,
608, 609, 610, 611, 612, 613,
614, 615, 680, 681, 682, 683,
684, 685, 686, 687, 688, 689,
690, 691, 692, 693, 694, 695,
696, 697, 698, 699, 700, 701,
702, 703, 704, 705, 706, 708,
709, 710, 711, 712, 713, 716,
717, 718, 719, 720, 721, 722,
723, 755, 760, 929, 930, 950,
951, 1033, 1132, 1133, 1134,
1137, 1156, 1158, 1163, 1165,
1166, 1167, 1168, 1169, 1170,
1171, 1172, 1179, 1183, 1184,
1185, 1189, 1190, 1200, 1212,
1295, 1298, 1299, 1300, 1350,
1352, 1366, 1377, 1378, 1382,
1383, 1398, 1543, 1569, 1571,
1577, 1578, 1581, 1663, 1665,
1667, 1750, 1751, 1752, 1753,
1782, 1784, 1817, 1818, 1819,
1872, 1881, 1891, 1892, 1893,
1894, 1895, 1896, 1913, 1940,
2038, 2095, 2101, 2102, 2134
2135, 2136, 2141, 2142, 2143,
2146, 2154, 2156, 2158, 2318,
2356, 2378, 2386, 2387, 2430,
2482, 2484, 2524, 2562, 2570,
2607, 2609, 2610, 2612, 2613,
2622

V4: DATE OCTOBER, 2016

FEATURE LIST

5, 11, 15, 22, 24

CHAPTER 2

14, 15, 39, 40, 41, 42, 43, 45, 84, 99, 100, 119, 120, 121, 122, 123, 124, 126, 127, 128, 129, 130, 131, 132, 133, 172

CHAPTER 3

191, 239, 265, 307, 317, 401, 434, 467, 557, 558, 659, 781, 912, 913, 981, 982, 993, 994, 995, 996, 997, 998, 999, 1000, 1001, 1098, 1104, 1126, 1136, 1145, 1321, 1322, 1323, 1324, 1325, 1326, 1327, 1328, 1329, 1330, 1331, 1332, 1333, 1334, 1336, 1337, 1338, 1339, 1340, 1342, 1343, 1346, 1347, 1349, 1350, 1352, 1431, 1432, 1433, 1434, 1435, 1436, 1437, 1438, 1439, 1440, 1441, 1442, 1443, 1445, 1447, 1449, 1451, 1453, 1454, 1455, 1456, 1457, 1458, 1459, 1461, 1462, 1463, 1464, 1465, 1469, 1470, 1471, 1472, 1473, 1474, 1480, 1481, 1483, 1484, 1485, 1486, 1487, 1488, 1491, 1492, 1493, 1494, 1495, 1496, 1497, 1498, 1505, 1506, 1507, 1508, 1509, 1517, 1545, 1546, 1548, 1549, 1553, 1555, 1557, 1611, 1612, 1613, 1614, 1615, 1616, 1617, 1618, 1619 1620, 1621, 1730, 1750, 1751, 1772, 1886, 1887, 1888, 1889, 1890, 1891, 1892, 1893, 1894, 1895, 1896, 1897, 1898, 1899, 1900, 1901, 1969, 1985, 2023, 2052, 2053, 2059, 2087, 2188 2261, 2264, 2315, 2316, 2317, 2318, 2319, 2323, 2324, 2327, 2328, 2329, 2337, 2338, 2339, 2340, 2341, 2344, 2345, 2359, 2360, 2361, 2362, 2363, 2364, 2365, 2366, 2367, 2368, 2384, 2387, 2419, 2496, 2497, 2500, 2502, 2503, 2504, 2505, 2506, 2507, 2508, 2509, 2510, 2517, 2523, 2525, 2526, 2527, 2528, 2529, 2530, 2531, 2536, 2538, 2539, 2541, 2542, 2549, 2565, 2566, 2567, 2568, 2569, 2570, 2571

APPENDIX C

69

V3: DATE APRIL, 2016

FEATURE LIST

1, 2, 7, 8, 12, 13, 15, 16, 17, 18, 21, 22

CHAPTER 1

2, 3

CHAPTER 2

24, 35, 48, 79, 83, 91, 92, 93, 94, 128, 169, 171, 172

CHAPTER 3

177, 178, 179, 194, 199, 200, 219, 220, 225, 228, 229, 248, 251, 252, 270, 271, 281, 291, 297, 314, 391, 393, 403, 405, 413, 423, 439, 444, 450, 456, 539, 555, 558, 569, 575, 582, 586, 591, 595, 601, 602, 619, 649, 650, 651, 657, 658, 677, 689, 700, 710, 715, 719, 745, 782, 791, 811, 831, 906, 907, 932, 934, 935, 947, 948, 1020, 1031, 1035 1036, 1114, 1127, 1128, 1143, 1144, 1145, 1155, 1156, 1308, 1309, 1325, 1335, 1342, 1343, 1345, 1406, 1409, 1478, 1479, 1480, 1481, 1514, 1521, 1529, 1530, 1531, 1532, 1533, 1534, 1535, 1536, 1537, 1538, 1540, 1541, 1543, 1544, 1545, 1546, 1548, 1549, 1550, 1552, 1553, 1554, 1556, 1557, 1558, 1559, 1560, 1561, 1562, 1563, 1564, 1565, 1566, 1567, 1568, 1571, 1572, 1574, 1575, 1576, 1577, 1578, 1580, 1582, 1584, 1585, 1587, 1588, 1589 1590, 1591, 1592, 1593, 1594, 1595, 1596, 1679, 1680, 1682, 1683, 1684, 1687, 1695, 1697, 1699, 1700, 1701, 1703, 1704, 1710, 1711, 1712 1713, 1715, 1716, 1718, 1744, 1745, 1746, 1747, 1749, 1796, 1797, 1799, 1800, 1821, 1822, 1823, 1829, 1834, 1835, 1845, 1846, 1890, 1921, 1922, 1930, 1938, 1939, 1940, 1941, 1942, 1943, 1944, 1945, 1946, 1947, 1948, 1952, 1957, 1960, 1961, 1989, 1991, 1993, 1994, 2000, 2002, 2004, 2020, 2051, 2052, 2078, 2103, 2127, 2236, 2238, 2268, 2281, 2282, 2299, 2308, 2330, 2331, 2333, 2335, 2336, 2337, 2340, 2342, 2343, 2344, 2346, 2347, 2348, 2349, 2350, 2351, 2352, 2353, 2354, 2355, 2367, 2379, 2382, 2383, 2384, 2411, 2412, 2420, 2423, 2424, 2438, 2470, 2474, 2475, 2476, 2477, 2478, 2479

APPENDIX C

26

V2: DATE NOVEMBER, 2015

CHAPTER 3

1121, 2022

V2: DATE OCTOBER, 2015

FEATURE LIST

2, 3, 5, 6, 7, 8, 9, 10, 11, 12, 13, 14, 15, 16, 17, 18, 19, 20, 21, 22, 23

CHAPTER 2

8, 9, 10, 11, 12, 14, 15, 16, 17, 18, 19, 20, 22, 23, 24, 25, 26, 27, 28, 29, 30, 31, 32, 33, 34, 35, 36, 37, 38, 39, 40, 42, 44, 45, 46, 47, 49, 50, 51, 52, 55, 56, 60, 61, 62, 63, 64, 65, 66, 69, 70, 71, 73, 75, 76, 79, 80, 81, 82, 83, 84, 87, 89, 91, 92, 94, 95, 96, 97, 98, 115, 116, 118, 121, 127, 135, 137, 138, 139, 142, 146, 147, 148, 149, 151, 152, 154, 155, 156, 157, 161, 162, 165, 166, 167, 169

CHAPTER 3

174, 175, 176, 357, 385, 386, 387, 405, 420, 440, 451, 453, 558, 560, 561, 595, 596, 597, 601, 602, 603, 604, 605, 606, 619, 620, 621, 622, 643, 644, 904, 905, 934, 936, 937, 948, 983, 1098, 1100, 1101, 1102, 1103, 1104, 1105, 1106, 1107, 1108, 1109, 1110, 1113, 1114, 1116, 1120, 1122, 1123, 1124, 1125, 1127, 1128, 1129, 1130, 1242, 1243, 1249, 1250, 1353 1356, 1357, 1359, 1360, 1555, 1556, 1565, 1603, 1604, 1605, 1606, 1607, 1608, 1609, 1610, 1611, 1612, 1636, 1637, 1638, 1639, 1640, 1641, 1642, 1645, 1716, 1717, 1718, 1719, 1720, 1721, 1723, 1724, 1725, 1726, 1727, 1728, 1729, 1730, 1731, 1759, 1776, 1814, 1816, 1848, 1849, 1853, 1854, 1861, 1869, 1870, 1871, 1873, 1940, 1941, 2056, 2059, 2074, 2075, 2076, 2077, 2082, 2083, 2095, 2098, 2155, 2156, 2157, 2159, 2230, 2232, 2235, 2236, 2242, 2302, 2369, 2370, 2372, 2374, 2376, 2377, 2378, 2379, 2383, 2384, 2385, 2387, 2388, 2390, 2391, 2392, 2393, 2394, 2395, 2396, 2397, 2398, 2399, 2400, 2402, 2403, 2404, 2407, 2408, 2411

. **V2**: DATE MARCH, 2015

FEATURE LIST

2, 3, 5, 6, 7, 8, 9, 10, 11, 12, 13, 14, 15, 16, 17, 18, 19, 20, 21, 22, 23

CHAPTER 1

2, 3

CHAPTER 2

17, 18, 19, 23, 24, 27, 28, 30, 32, 33, 35, 36, 38, 39, 40, 41, 42, 43, 45, 46, 47, 48, 49, 58, 64, 65, 69, 70, 71, 72, 73, 74, 75, 82, 84, 85, 88, 89, 90, 91, 92, 93, 96, 98, 100, 101, 124, 125, 127, 136, 144, 146, 147, 148, 151, 155, 156, 157, 158, 160, 161, 163, 164, 165, 166, 170, 171, 174, 175

CHAPTER 3

183, 184, 185, 186, 187, 188, 189, 190, 191, 192, 193, 194, 195, 196, 200, 204, 205, 208, 209, 214, 218, 225, 230, 236, 240, 241, 243, 250, 251, 252, 258, 266, 267, 268, 269, 270, 271, 272, 273, 274, 277, 278, 279, 287, 288, 289, 291, 293 295, 297, 298, 299, 302, 303, 318, 319, 321, 332, 333, 375 376, 390, 394, 410, 412, 414, 416, 420, 436, 437, 440, 447 448, 452, 454, 455, 456, 459 466, 480, 481, 483, 484, 485, 501, 503, 505, 506, 508, 514, 515, 519, 521, 525, 526, 535, 539, 541, 542, 545, 548, 549, 555, 556, 557, 558, 559, 560, 561, 562, 563, 565, 566, 569 570, 577, 578, 579, 581, 583, 587, 588, 589, 591, 592, 597 601, 603, 604, 606, 607, 609, 610, 611, 612, 613, 614, 620, 625, 627, 628, 629, 639, 640, 642, 644, 651, 653, 654, 655, 656, 657, 658, 659, 660, 661, 662, 663, 664, 665, 669, 670, 671, 672, 673, 675, 676, 678, 679, 684, 687, 688, 689, 693, 695, 698, 717, 725, 748, 757, 759, 760, 761, 768, 769, 784, 786, 787, 788, 789, 790, 791, 792, 793, 794, 795, 796, 797, 798, 800, 801, 802, 803, 805, 816, 820, 822, 823, 836, 858, 859, 861, 868, 872, 873, 876, 897, 898, 899, 900, 901, 902, 903, 904, 905, 906, 907, 908, 909, 910, 911, 912, 915, 922, 924, 925, 932, 934, 940, 942, 943, 944, 952, 960, 961, 962, 963, 964, 965, 966, 967, 968, 969, 970, 971, 972, 973, 974, 975, 976, 1002, 1004, 1006, 1008, 1009, 1010, 1011, 1016, 1025, 1031, 1033, 1046, 1047, 1050, 1051, 1065, 1066, 1074, 1075, 1076, 1080, 1083, 1103, 1107, 1109, 1110, 1111, 1112, 1113, 1114, 1116, 1117, 1121, 1122, 1124, 1127, 1128, 1130, 1131, 1133, 1135, 1136, 1137, 1138, 1145, 1146, 1147, 1148, 1151, 1152, 1156, 1157, 1158, 1159, 1164, 1188, 1189, 1208, 1210, 1213, 1214, 1215, 1225, 1230, 1237, 1238, 1239, 1249, 1252, 1253, 1259, 1260, 1298, 1299, 1300, 1301, 1302, 1307, 1308, 1309, 1314, 1315, 1316, 1317, 1318, 1319, 1320, 1321, 1322, 1323, 1324, 1328, 1329, 1330, 1331, 1338, 1339, 1340, 1341, 1342, 1343, 1344, 1345, 1348, 1349, 1351, 1352, 1356, 1357, 1367, 1370, 1379, 1393, 1394, 1395, 1416, 1417, 1420, 1421, 1451, 1452, 1453, 1459, 1461, 1475, 1476, 1478, 1480, 1482, 1483, 1484, 1485, 1486, 1487, 1488, 1495, 1496, 1500, 1502, 1504, 1505, 1509, 1512, 1513, 1520, 1521, 1523, 1524, 1525, 1526, 1527, 1528, 1529, 1530, 1531, 1532, 1533, 1536, 1543, 1545, 1549, 1552, 1553, 1554, 1555, 1556, 1557, 1558, 1559, 1560, 1571, 1572, 1575, 1576, 1581, 1582, 1585, 1591, 1592, 1594, 1596, 1608, 1609, 1614, 1616, 1617, 1618, 1619, 1620, 1621, 1622, 1623, 1624, 1625, 1641, 1648, 1649, 1650, 1651, 1652, 1653, 1654, 1655, 1675, 1676, 1677, 1683,

1684, 1685, 1686, 1687, 1692,
1693, 1694, 1695, 1696, 1712,
1726, 1729, 1730, 1731, 1732,
1733, 1734, 1736, 1737, 1738,
1739, 1740, 1741, 1742, 1743,
1748, 1750, 1751, 1752, 1753,
1754, 1759, 1762, 1771, 1782,
1792, 1793, 1798, 1801, 1802,
1804, 1806, 1807, 1808, 1809,
1810, 1811, 1818, 1819, 1820,
1821, 1828, 1831, 1832, 1833,
1834, 1835, 1840, 1841, 1842,
1843, 1844, 1849, 1850, 1853,
1854, 1855, 1856, 1861, 1880,
1881, 1893, 1905, 1910, 1914,
1916, 1918, 1919, 1921, 1926,
1931, 1933, 1934, 1953, 1971,
1972, 1976, 1977, 1978, 1979,
1980, 1982, 1983, 1984, 1985,
1987, 1989, 1995, 1996, 1997,
1998, 1999, 2016, 2019, 2029,
2030, 2031, 2032, 2033, 2034
2036, 2037, 2038, 2039, 2040,
2041, 2043, 2044, 2045, 2054,
2055, 2059, 2061, 2065, 2068,
2083, 2085, 2086, 2091, 2092,
2104, 2107, 2153, 2156, 2159,
2163, 2164, 2165, 2166, 2168,
2195, 2196, 2197, 2198, 2199,
2203, 2206, 2207, 2208, 2212,
2213, 2214, 2215, 2216, 2219,
2220, 2223, 2224, 2226, 2227,
2228, 2231, 2232, 2233, 2234,
2235, 2236, 2237, 2238, 2239,
2240, 2242, 2245, 2246, 2252,
2254, 2261, 2264, 2293, 2300,
2301, 2302, 2304, 2307, 2308,
2309, 2327, 2342, 2343, 2344,
2345, 2358, 2359, 2360, 2362,
2363, 2364, 2365, 2370, 2382,
2383, 2385, 2387, 2388, 2393,
2395, 2397, 2401, 2403, 2404,
2405, 2406, 2407, 2408, 2409,
2410, 2411, 2412, 2413, 2414,
2417, 2418, 2421, 2425, 2426,
2427, 2428, 2429, 2434, 2435,
2436

V1: DATE MAY, 2014

Chapter 1

3

Chapter 2

43

Chapter 3

172, 173, 237, 281, 282, 283,
303, 447, 495, 496, 558, 598,
607, 714, 721, 773, 776, 777,
886, 887, 890, 900, 917, 918,
1311, 1312, 1313, 1316, 1317,
1319, 1320, 1324, 1325, 1512,
1518, 1528, 1550, 1551, 1565,
1583, 1610, 1842, 2013, 2014,
2018, 2020, 2042, 2044, 2045,
2050, 2051, 2063, 2162, 2174,
2177, 2180, 2181, 2183, 2184,
2188, 2190, 2191, 2192, 2311