Features and Services Fundamentals — Book 3 of 6 (D to H)
Avaya Communication Server 1000

7.5
NN43001-106, 05.09
April 2011

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Contents

Chapter 1: New in this release

The following sections detail what is new in *Avaya Feature and Services Fundamentals - Book 3 of 6* (NN43001-106) for Avaya Communication Server 1000 (Avaya CS 1000) Release 7.5.

- Features on page 25
- Other changes on page 25

Features

See the following sections for information about feature changes.

Extended Local Calls

To provide Aura with information regarding local calls on a CS 1000 Access Element, the Extended Local Calls (ELC) feature routes all local calls to a preconfigured ELC route.

To enable the ELC feature on a telephone, set ELC class of service. From an end user point of view, ELC calls do not differ from, and are originated in the same manner as, local calls.

When you enable ELC class of service, all local calls for a telephone are routed through ELC SIP trunks if possible. Calls are processed locally only when both parties disable ELC class of service.

See Extended Local Calls on page 295.

Other changes

This release contains no other changes.

Revision History

April 2012	Standard 05.09. This document is up-issued to include the chapter E.164 Enhancement.
December 2011	Standard 05.08. This document is up-issued to support the removal of End of Life (EoL) and Manufactured Discontinued (MD) hardware content and associated diagrams.

December 2011	Standard 05.07. This document is up-issued for release 7.5 to indicate that CLID Manipulation Data Block (CMDB) is not supported for Local Termination (LTER).
May 2011	Standard 05.06. This document is up-issued to support Communication Server 1000 Release 7.5.
March 2011	Standard 05.05. This document is up-issued to support Communication Server 1000 Release 7.5.
March 2011	Standard 05.04. This document is up-issued to support Communication Server 1000 Release 7.5.
February 2011	Standard 05.03. This document is up-issued to remove legacy feature and hardware content that is no longer applicable to or supported by Communication Server 1000 systems.
November 2010	Standard 05.01 and 05.02. These documents were issued to support Communication Server 1000 Release 7.5.
June 2010	Standard 04.01. This document is up-issued to support Communication Server 1000 Release 7.0.
December 2009	Standard 03.03. This document is up-issued to support Communication Server 1000 Release 6.0.
June 2009	Standard 03.02. This document is up-issued to support Communication Server 1000 Release 6.0.
May 2009	Standard 03.01. This document is up-issued to support Communication Server 1000 Release 6.0.
July 2008	Standard 02.06. This document has been up-issued to support Communication Server Release 5.5.
January 2008	Standard 02.05. This document has been up-issued to support Communication Server Release 5.5.
December 2007	Standard 02.04. This document has been up-issued to support Communication Server Release 5.5.
June 2007	Standard 01.04. This document is up-issued to revise the 500 Telephone Features and Bandwidth Management Support for Network Wide Virtual Office chapters in Book 1 and the Conference Warning Tone Enhancement chapter in Book 2).
June 2007	Standard 01.03. This document is up-issued to revise the Software Licenses chapter in Book 6.
June 2007	Standard 01.02. This document is up-issued to revise the Network Music feature implementation in Book 5.
May 2007	Standard 01.01. This document is up-issued to support Communication Server 1000 Release 5.0.
July 2006	Standard 17.00.This document is up-issued to reflect changes in technical content for M3900 Full Icon Support feature, M3900

	Set-to-Set Messaging feature and M3900 series digital telephone feature reference.
April 2007	Standard 16.00. This document is up-issued to add keycode commands for CP PIV, IPMG on CS1000E to the following : operating parameters; and LD 97, Call redirection by Day ; the CRDAY prompt ; and Call Direction by Time of Day, Flexible Feature Codes and to correct Message Intercept for Set Status Lockout and SECA001 alarm message.
January 2006	Standard 15.00. This document is up-issued to add Converged Office feature ; changes to interactions with Call Forward All Calls (Book 1) and (Book 2) and IP Phones to supported sets referenced in Selectable Conferee Display and Disconnect to (Book 3).
August 2005	Standard 14.00. This document is up-issued to support Communication Server 1000 Release 4.5.
September 2004	Standard 13.00. This document is up-issued for Communication Server 1000 Release 4.0.
October 2003	Standard 12.00. This document is issued for Succession 3.0.
November 2002	Standard 11.00. This document is up-issued to support Meridian 1 Release 25.40 and Succession Communication Server for Enterprise (CSE) 1000, Release 2.0. This is book 2 of a 3 book set.
January 2002	Standard 10.00. This document is up-issued to include content for Meridian 1 Release 25.40 and Succession Communication Server for Enterprise 1000, Release 1.1.
April 2000	Standard 9.00. This document is up-issued for Release 25.0x. to include removal of: redundant content; references to equipment types except Options 11C, 51C, 61C, and 81C; and references to previous software releases.
June 1999	Standard 8.00. This document is up-issued for Generic Release 24.2x.
October 1997	Standard 7.00. This document is up-issued to remove certain application-specific features and to place them in their appropriate Nortel Networks technical publications (technical documents).
August 1996	Standard 6.00. This document is up-issued to incorporate the features Automatic Number Identification, Automatic Trunk Maintenance, Multi Tenant Service, Radio Paging and X08/11 Gateway.
December 1995	Standard 5.00. This document is up-issued for Release 21.1x.
July 1995	Standard 4.00.This document is up-issued for Release 21.
October 1994	Standard 2.0. This document is up-issued for Release 20.1x.

July 1994	Standard 1.0. This document is up-issued for Release 20.0x.

Chapter 2: Customer service

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Chapter 3: Features and Software options

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Package Name	Number	Mnemonic	Release
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Package Name	Number	Mnemonic	Release
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Package Name	Number	Mnemonic	Release
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Package Name	Number	Mnemonic	Release
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Package Name	Number	Mnemonic	Release
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Package Name	Number	Mnemonic	Release
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Package Name	Number	Mnemonic	Release
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• Call Pickup, Directed			
• Directory Number Delayed Ringing			
Directory Number Expansion (7 Digit)	150	DNXP	13
• Directory Number Expansion			
• Directory Number			
- Flexible Attendant Directory Number			
- Listed Directory Numbers			
- Single Appearance Directory Number			

Package Name	Number	Mnemonic	Release
- Multiple Appearance Directory Number			
- Prime Directory Number			
• Diskette Overflow Warning			
• Display of Calling Party Denied			
Distinctive Ringing	74	DRNG	4/9
• Distinctive/New Distinctive Ringing			
Do Not Disturb, Group	16	DNDG	1
• Do Not Disturb Group			
Do Not Disturb, Individual	9	DNDI	1
• Do Not Disturb			
• Network Individual Do Not Disturb (see also packages 127, and 159)			
• Electronic Brand lining			
Emergency Services Access Calling Number Mapping	331	ESA_CLMP	23
• Emergency Services Access (See also packages 329 and 330)			
Emergency Services Access Supplementary	330	ESA_SUPP	23
• Emergency Services Access (See also packages 329 and 331)			
Emergency Services Access	329	ESA	23
• Emergency Services Access (See also packages 330 and 331)			
• End of Selection			
• End of Selection Busy			
• End-of-Dialing on Direct Inward/Outward Dialing Incoming Call Indicator Enhancement			
End-To-End Signaling	10	EES	1
• Attendant End-to-End Signaling			
• End-to-End Signaling			
Enhanced ACD Routing	214	EAR	17
• Enhanced Automatic Call Distribution Routing			
• MFC Interworking with AML Based Applications (see also packages 128, and 215)			
Enhanced Call Trace	215	ECT	18

Package Name	Number	Mnemonic	Release
• Customer Controlled Routing			
• MFC Interworking with AML Based Applications (see also packages 128, and 214)			
Enhanced Controlled Class of Service	173	ECCS	15
Enhanced DPNSS Services	288	DPNSS_ES	21
• DPNSS1 Executive Intrusion			
Enhanced DPNSS1 Gateway	284	DPNSS189I	20
• Enhanced DPNSS1 Gateway			
Enhanced Hot Line	70	HOT	4/10
• Hot Line			
• Network Intercom			
• Enhanced input/output buffering			
• Enhanced Maintenance (Patching)			
Enhanced Music	119	EMUS	12
• Music, Enhanced			
Enhanced Night Service	133	ENS	20
• Enhanced Night Service			
• Enhanced package printout			
• Equal Access Compliance			
Euro ISDN Trunk - Network Side	309	MASTER	22
• EuroISDN Trunk - Network Side			
Euro ISDN	261	EURO	20
• ISDN – Advice of Charge for EuroISDN			
• ISDN BRI and PRI Trunk Access for Europe (EuroISDN)			
• EURO ISDN Continuation			
Euro Supplementary Service	323	ETSI_SS	22
• EuroISDN Call Completion Supplementary Service			
Executive Distinctive Ringing	185	EDRG	16
• Executive Distinctive Ringing			
FCC Compliance for DID Answer Supervision	223	FCC68	17
• Federal Communications Commission Compliance for DID Answer Supervision			

Package Name	Number	Mnemonic	Release
Feature Group D	158	FGD	17
• Feature Group D (Inbound to Meridian 1)			
• Federal Communications Commission Compliance for Equal Access			
• First-Second Degree Busy Indication			
• First-Second Degree Busy Indication, ISDN			
• Flexible Attendant Call Waiting Thresholds			
• Flexible Busy Tone Timer			
Fiber Network	365	FIBN	25
Flexible Call Back Queuing	61	FCBQ	1
• Flexible Call Back Queuing			
Flexible Direct Inward Dialing	362	FDID	24
• Flexible Direct Inward Dialing			
Flexible Feature Codes	139	FFC	15
• Call Forward/Hunt Override Via Flexible Feature Code			
• China Number 1 Signaling – Flexible Feature Codes			
• Dial Access to Group Calls (see also package 48).			
• Direct Inward Dialing Call Forward No Answer Timer			
• Electronic Lock Network Wide/Electronic Lock on Private Lines			
• Flexible Feature Codes			
• Automatic Wake FFC Delimiter			
• Call Forward Destination Deactivation			
Flexible Numbering Plan	160	FNP	14
• Alternative Routing for DID/DOD			
• Flexible Numbering Plan			
• Special Dial Tones after Dialed Numbers			
• Flexible Numbering Plan Enhancement			
• Flexible Orbiting Prevention Timer			
Flexible Tones and Cadences	125	FTC	16
• Flexible Tone and Digit Switch Control			
• Reverse Dial on Routes and Telephones			
• Tones and Cadences			

Package Name	Number	Mnemonic	Release
Forced Charge Account	52	FCA	1
• Charge Account, Forced			
French Type Approval	197	FRTA	15
• Camp-on to a Set in Ringback or Dialing			
• Forward No Answer Call Waiting Direct Inward Dialing			
• Group Hunt Queuing (see also package 120)			
• Group Hunt Queuing Limitation Enhancement (see also package 120)			
• Loopback on Central Office Trunks			
Geographic Redundancy Primary system	404	GRPRIM	4.0
Geographic Redundancy Secondary system	405	GRSEC	4.0
Group Call	48	GRP	1
• Dial Access to Group Calls (see also package 139).			
• Group Call			
• Group Hunt Queuing Limitation (see also package 120)			
Group Hunt/DN Access to SCL	120	PLDN	15
• Group Hunt Queuing (see also package 197)			
• Group Hunt Queuing Limitation (see also package 131)			
• Group Hunt Queuing Limitation Enhancement (see also package 197)			
• Group Hunt			
• Speed Call Directory Number Access			
• Handset Volume Reset			
• Handsfree Download (Meridian Digital Telephones			
• Held Call Clearing			
H323 Virtual Trunk	399	H323_VTRK	3.0
• IP Peer Networking Phase 2			
• Branch Office			
HiMail Fax Server	195	FAXS	18
History File	55	HIST	1
• History File			
Hold in Queue for IVR	218	IVR	18

Package Name	Number	Mnemonic	Release
Hospitality Management	166	HOSP	16
Hospitality Screen Enhancement	208	HSE	17
• Hospitality Enhancements: Display Enhancements			
• Hunting By Call Type			
• Hunting			
- Circular Hunting			
- Linear Hunting			
- Secretarial Hunting			
- Short Hunting			
- Data Port Hunting			
- Trunk Hunting			
• Incoming Call Indicator Enhancement			
Incoming DID Digit Conversion	113	IDC	12
• China Number 1 Signaling Trunk Enhancements (see also packages 49, 128, and 131)			
• DNIS Name Display (see also packages 95, and 98)			
• Incoming DID Digit Conversion			
• Incoming Trunk Programmable Calling Line Identification			
• Incremental Software Management			
• Input/Output Access and System Limits			
Integrated Digital Access	122	IDA	16
• Analog Private Network Signaling System (APNSS) (see also packages 190, 123, and 124)			
• DASS2/DPNSS1 – Integrated Digital Access (see also packages 123 and 124)			
• DPNSS1 Satellite			
• DASS2/DPNSS INIT Call Cutoff			
Integrated Message System UST and UMG are part of IMS Package	35	IMS	2
• Integrated Messaging System Link			
Integrated Services Digital Network Application Module Link for Third Party Vendors	153	IAP3P	13
• Application Module Link			
• Network Application Protocol Link Enhancement			

Package Name	Number	Mnemonic	Release
Integrated Services Digital Network BRI Trunk Access	233	BRIT	18
• Integrated Services Digital Network Basic Rate Interface (see also packages 216, and 235)			
Integrated Services Digital Network Supplementary Features	161	ISDN INTLSUP	14
• Call Connection Restriction (see also packages 146 and 147)			
• Direct Inward Dialing to Network Calling			
• Incoming Digit Conversion Enhancement			
• Network Time Synchronization			
• X08 to X11 Gateway			
Integrated Services Digital Network Signaling Link	147	ISL	13
• Call Connection Restriction (see also packages 146 and 161)			
Integrated Services Digital Network	145	ISDN	13
• Backup D-Channel to DMS-100/250 and AT&T 4ESS			
• Call Pickup Network Wide			
• D-Channel Error Reporting and Monitoring			
• Integrated Services Digital Network (ISDN) Primary Rate Interface			
• Network Name Display (Meridian 1 to DMS-100/250)			
• Total Redirection Count			
• T309 Time			
• Integrated Voice and Data			
Intercept Computer Interface	143	ICP	10
• Intercept Computer Dial from Directory			
• Intercept Computer Enhancements			
• Intercept Computer Flexible DN Length			
• Intercept Computer Interface			
• Intercept Computer Network Screen Activation and Flexible DN interactions			
• Intercept Treatment Enhancements			
Intercept Treatment	11	INTR	1
• Intercept Treatment			

Package Name	Number	Mnemonic	Release
Inter-Exchange Carrier	149	IEC	13
• Inter Exchange Carrier			
Internal CDR	108	ICDR	10
• Internal Call Detail Recording			
International 1.5/2.0 Mbit/s Gateway	167	GPRI	18
• Radio Paging			
• International Meridian 1			
International nB+D	255	INBD	20
• ISDN PRI Do Trunk Access for Japan (nB+D)			
International Primary Rate Access (CO)	146	PRA	13
• Call Connection Restriction (see also packages 147 and 161)			
• Integrated Services Digital Network Primary Rate Access			
• Integrated Services Digital Network Primary Rate Access Central Office Connectivity to Japan D70			
International Primary Rate Access	202	IPRA	15
• Integrated Services Access/Call by Call Service Selection Enhancements			
• Integrated Services Digital Network Primary Rate Access to 1TR6 Connectivity			
• Integrated Services Digital Network Primary Rate Access to NUMERIS Connectivity			
• Integrated Services Digital Network Primary Rate Access to SwissNet 2 Connectivity			
• Integrated Services Digital Network Primary Rate Access to SYS-12 Connectivity			
International Supplementary Features	131	SUPP	9
• IODU/C			
IP Expansion	295	IPEX	25.40
• IP Expansion			
IP Media Gateway	403	IPMG	4.0
ISDN Semi-Permanent Connection	313	ISPC	22
• ISDN Semi-Permanent Connections for Australia			
• Italian Central Office Special Services (see also packages 129, and 157)			

Package Name	Number	Mnemonic	Release
Japan Central Office Trunks	97	JPN	9
• Japan Central Office Trunk			
Japan Digital Multiplex Interface	136	JDMI	14
• Japan Digital Multiplex Interface			
Japan Telecommunication Technology Committee	335	JTTC	23
• Japan TTC Common Channel Signaling			
Japan Tone and Digit Switch	171	JTDS	14
• Japan Tone and Digit Switch			
Last Number Redial	90	LNR	8
• Last Number Redial			
Limited Access to Overlays	164	LAPW	16
• B34 Dynamic Loss Switching (see also packages 131 and 203)			
• Faster I/O			
• Limited Access to Overlays			
• Limited Access to Overlays Password Enhancement			
• Teletype Terminal Access Control in Multi-Customer Environment (see also package 131)			
Line Load Control	105	LLC	10
• Line Load Control			
• Line Lockout			
Local Steering Code Modifications	137	LSCM	10
• Local Steering Code Modifications			
• Lockout, DID Second Degree Busy and MFE Signaling Treatments			
• Loop Start Answer Supervision XUT			
• Loop Start Supervisory Trunks			
• Loop Start Supervisory Trunks (Incoming Calls)			
Location Code Expansion	400	LOCX	4.0
M2000 Digital Sets	88	DSET	7
• Distinctive Ringing for Digital Telephones			
• M2317 Telephones			
• Flexible Voice/Data Terminal Number			

Package Name	Number	Mnemonic	Release
M2250 Attendant Console • Digital Attendant Console	140	DCON	15
M2317 Digital Sets • M2317 Digital Sets	91	DLT2	9
M3000 Digital Sets • M3000 Telephones	89	TSET	7
M3900 Full Icon Support • M3900 Full Icon Support	397	ICON_ PACKAGE	3.0
M3900 Phase III Virtual Office Enhancement • Virtual Office Enhancement	387	VIR_OFF_ ENH	25.40
M3900 Ring Again	396	M3900_RG A_PROG	3.0
M911 Enhancement Display • 10/20 Digit ANI on 911 Calls	249	M911 ENH	25
Maid Identification • Maid Identification • Make Set Busy and Voice Call Override	210	MAID	17
Make Set Busy • Make Set Busy • Make Set Busy Improvement • Malicious Call Trace on Direct Inward Dialing	17	MSB	1
Malicious Call Trace • Enhanced Malicious Call Trace • Malicious Call Trace • Malicious Call Trace DN/TN Print • Malicious Call Trace Idle • Manual Line Service • Manual Service Recall to Attendant • Manual Signaling (Buzz) • Manual Trunk Service	107	MCT	10

Package Name	Number	Mnemonic	Release
MAT 5.0	296	MAT	22
• Meridian 1 Attendant Console Enhancements (see also package 76)			
Meridian 1 Companion Option	240	MCMO	19
• Avaya Integrated DECT			
MCDN End to End Transparency	348	MEET	24
Meridian 1 Enhanced Conference, TDS and MFS	204	XCT0	15
• Meridian 1 Enhanced Conference, TDS and MFS			
Meridian 1 Fault Management	243	ALRM_FILTER	19
• Alarm Management			
• Meridian 1 Initialization Prevention and Recovery			
Meridian 1 Packet Handler	248	MPH	19
• Meridian 1 Packet Handler			
Meridian 1 Superloop Administration (LD 97)	205	XCT1	15
• Extended DID/DOD Software Support – Europe			
• Extended Flexible Central Office Trunk Software Support			
• Extended Tone Detector and Global Parameters Download (see also package 203)			
• Generic XFCOT Software Support			
Meridian 1 XPE	203	XPE	15
• B34 Codec Static Loss Plan Downloading			
• B34 Dynamic Loss Switching (see also packages 131, and 164)			
• Extended Multifrequency Compelled Sender/Receiver			
• Extended Tone Detector and Global Parameters Download (see also package 205)			
• Intelligent Peripheral Equipment Software Support Enhancements			
Meridian 911	224	M911	19
• Meridian 911 Enhancements – Call Abandon			
• Meridian 911 Enhancements – MADN Display Coordination			
Meridian Hospitality Voice Service	179	HVS	16
• Meridian Hospitality Voice Services			

Package Name	Number	Mnemonic	Release
Meridian Link Modular Server	209	MLM	16
• Meridian Link Enhancements			
Meridian SL-1 ST Package	96	SLST	9
• Meridian SL-1 ST Package			
Message Intercept	163	MINT	15
• Message Intercept			
Message Waiting Center	46	MWC	1
• Message Waiting Lamp Maintenance			
• Message Waiting Unconditional			
Message Waiting Indication Interworking with DMS	219	MWI	19
• Message Waiting Indication (MWI) Interworking			
Mobile Extensions	412	MOBX	5.50
Modular Telephone Relocation			
Multifrequency Compelled Signaling	128	MFC	9
• China Number 1 Signaling Trunk Enhancements (see also packages 49, 113, and 131)			
• China Number 1 Signaling – Active Feature Dial Tone (see also package 126)			
• China Number 1 Signaling – Audible Alarm (see also package 126)			
• China Number 1 Signaling – Vacant Number Announcement (see also package 126)			
• India Phase 2			
• R2 Multifrequency Compelled Signaling (MFC) DID/DTMF DOD			
• R2 Multifrequency Compelled Signaling (MFC) Selective Route To Attendant			
• MFC Interworking with AML Based Applications (see also packages 214 and 215)			
• R2 Multifrequency Compelled Signaling Timer Control			
• Semi-Compelled MFC and Calling Name Identification Charges			
Multifrequency Signaling for Socotel	135	MFE	10
• Multifrequency Signaling for Socotel			

Package Name	Number	Mnemonic	Release
Multi-Language I/O Package	211	MLIO	16
• Multi-language TTY Input/Output			
Multi-Language Wake Up	206	MLWU	16
• Multi-language Wake Up			
• Multi-Party Operation Enhancements			
Multi-Party Operations	141	MPO	20
• Attendant Clearing during Night Service			
• Multi-Party Operations			
• Multiple Appearance DN Redirection Prime			
• Multiple Console Operation			
Multiple Queue Assignment	297	MQA	21
• Multiple Queue Assignment			
Multiple-Customer Operation	2	CUST	1
• Multiple Customer Operation			
Multiple-Tenant Service	86	TENS	7
• Multi-Tenant Service			
Multi-purpose Serial Data Link Serial Data Interface	227	MSDL SDI	19
• Multi-purpose Serial Data Link Serial Data Interface			
Multi-purpose Serial Data Link Single Terminal Access	228	MSDL STA	19
• Single Terminal Access			
Multi-purpose Serial Data Link	222	MSDL	18
• Multi-purpose Serial Data Link			
Multi-Site Mobility Networking	370	MSMN	25
Multi-User Login	242	MULTI_USER	19
• Multi-User Login			
Music Broadcast	328	MUSBRD	23
• Music Broadcast			
Music	44	MUS	1
• Music			
Network Alternate Route Selection	58	NARS	1

Package Name	Number	Mnemonic	Release
• Equi-distribution Network Attendant Service Routing (see also package 159)			
• Network Alternate Route Selection/Basic Alternate Route Selection Enhancement – Local Termination (see also package 57)			
• Network Anti-tromboning			
• Virtual Network Services/Virtual Directory Number Expansion (see also package 183)			
Network Attendant Service	159	NAS	20
• Equi-distribution Network Attendant Service Routing (see also package 58)			
• Network Individual Do Not Disturb (See also packages 9 and 127).			
Network Authorization Code	63	NAUT	1
• Network Authorization Code			
Network Automatic Call Distribution	207	NACD	15
• Network Automatic Call Distribution			
Network Call Back Queuing	38	MCBQ	2
• Network Call Back Queuing			
Network Call Transfer	67	NXFR	3
Network Class Of Service	32	NCOS	1
• Network Class of Service			
Network Message Services	175	NMS	16
Network Priority Queuing	60	PQUE	1
• Network Priority Queuing			
Network Signaling	37	NSIG	2
• Network Signaling			
Network Speed Call	39	NSC	2
• Network Speed Call			
Network Traffic Measurements	29	NTRF	1
• Network Traffic Measurement			
New Flexible Code Restriction	49	NFCR	2

Package Name	Number	Mnemonic	Release
• China Number 1 Signaling Trunk Enhancements (see also packages 113, 128, and 131)			
• New Flexible Code Restriction			
New Format CDR	234	FCDR	18
• Call Detail Recording Time to Answer			
• CDR on Busy Tone			
Next Generation Connectivity	324	NGEN	22
NI-2 Call By Call Service Selection	334	NI-2 CBC	23
• Night Restriction Classes of Service			
• Night Service			
• Night Service Enhancements – All Calls Remain Queued for Night Service			
• Night Service Enhancements – Recall to Night DN			
• Night Service Enhancements – Requeuing of Attendant Present Calls			
• Night Service Enhancements – Requeuing of Attendant Present Calls			
NI-2 Name Display Service	385	NDS	25.40
• NI-2 Name Display Supplementary Service			
Avaya Symposium Call Center	311	NGCC	22
North America National ISDN Class II Equipment	291	NI2	21
• North American Numbering Plan			
• Off-Hook Alarm Security			
Observe Agent Security	394	OAS	3.0
• Observe Agent Security			
Off-Hook Queuing	62	OHQ	1
• Network Drop Back Busy and Off-hook Queuing (see also package 192)			
Office Data Administration System	20	ODAS	1
• Office Data Administration System			
• Off-Premise Extension			
On Hold On Loudspeaker	196	OHOL	20
• On-Hook Dialing			

Package Name	Number	Mnemonic	Release
Open Alarms	315	OPEN ALARM	22
Operator Call Back (China #1)	126	OPCB	14
• Busy Verify on Calling Party Control Calls			
• China Number 1 Signaling – Active Feature Dial Tone (see also package 128)			
• China Number 1 Signaling – Audible Alarm (see also package 128)			
• China Number 1 Signaling – Called Party Control			
• China Number 1 Signaling – Calling Number Identification on Outgoing Multifrequency Compelled Signaling			
• China Number 1 Signaling – Calling Party Control			
• China Number 1 Signaling – Flexible Timers			
• China Number 1 Signaling – KE Multifrequency Compelled Tandem Signaling			
• China Number 1 Signaling – Malicious Call Trace Enhancement			
• China Number 1 Signaling – Off-hook Tone			
• China Number 1 Signaling – Toll Call Identification			
• China Number 1 Signaling – Toll Operator Call Back			
• China Number 1 Signaling – Toll Operator Call Back Enhancement			
• China Number 1 Signaling – Vacant Number Announcement (see also Package 128)			
Optional Features	1	OPTF	1
• Autodial			
• Call Forward All Calls			
• Ring Again			
• Speed Call			
• Speed Call on Private Lines (see also package 0)			
• Speed Call/Autodial with Authorization Codes (see also package 34)			
• Speed Call Delimiter (see also package 34)			
Optional Outpulsing Delay	79	OOD	5
• Optional Outpulsing Delay			

Package Name	Number	Mnemonic	Release
Originator Routing Control	192	ORC_RVQ	18
• Network Drop Back Busy and Off-hook Queuing (see also package 62)			
• Remote Virtual Queuing			
• Out-of-Service Unit			
Outpulsing, asterisk (*) and octothorpe (#)	104	OPAO	
• Outpulsing of Asterisk "*" and Octothorpe "#"			
Overlap Signaling (M1 to M1 and M1 to 1TR6 CO)	184	OVLP	15
• Overlap Signaling			
• Overlay 45 Limited Repeats			
• Overlay Cache Memory			
• Override			
• Paging			
• Partial Dial Timing			
• PBX (500/2500) Telephones			
• Periodic Camp-on Tone			
• Periodic Clearing			
• Periodic Clearing Enhancement			
• Periodic Clearing on RAN, ACD, and Music			
Personal Call Assistant	398	PCA	3.0
• Personal Call Assistant			
Phantom TN	254	PHTN	20
• Phantom TNs			
• Position Busy with Call on Hold			
PPM/Message Registration	101	MR	10
• Advice of Charge Real-time Supplementary Services for NUMERIS and SWISSNET (see also package 131)			
• Advice of Charge – Charging Information and End of Call for NUMERIS Connectivity (see also package 131)			
• Message Registration			
• Periodic Pulse Metering			
• Predictive Dialing			
Pretranslation	92	PXLT	8

Package Name	Number	Mnemonic	Release
• Pretranslation			
• Preventing Reciprocal Call Forward			
Priority Network Override	389	PONW	25.40
• Network Break-in and Force Disconnect			
Priority Override/Forced Camp-On	186	POVR	20
• Forced Camp-on and Priority Override			
• Privacy			
• Privacy Override			
• Privacy Release			
• Private Line Service			
Proactive Voice Quality Management	401	PVQM	4.0
Property Management System Interface	103	PMSI	10
• Property Management System Interface			
• Public Switched Data Service			
Pulsed E&M (Indonesia, French Colisée)	232	PEMD	18
• Pulsed E&M DTI2 Signaling			
Q Reference Signaling Point Interface	263	QSIG	20
• Integrated Services Digital Network QSIG Basic Call			
QSIG Generic Functional protocol	305	QSIG GF	22
• ISDN QSIG Generic Functional Transport			
QSIG Supplementary Service	316	QSIG-SS	22
• ISDN QSIG Call Completion			
• ISDN QSIG Call Diversion Notification			
• ISDN QSIG Path Replacement			
Radio Paging	187	RPA	15
• Radio Paging			
• Radio Paging Product Improvements			
• Recall to Same Attendant			
• Recall with Priority during Night Service			
• Recall With Priority during Night Service			
• Recall With Priority during Night Service Network Wide			

Package Name	Number	Mnemonic	Release
Recorded Announcement Broadcast	327	RANBRD	23
• Recorded Announcement Broadcast			
Recorded Announcement	7	RAN	1
• Recorded Announcement			
Recorded Overflow Announcement	36	ROA	2
• Recorded Overflow Announcement			
• Recorded Telephone Dictation			
• Recovery of Misoperation on the Attendant Console			
• Recovery on Misoperation of Attendant Console			
• Reference Clock Switching			
• Reference Clock Switching (see also packages 75, 129, and 154)			
Remote IPE	286	REMOTE_IPE	
• Remote Intelligent Peripheral Equipment			
Remote Virtual Queuing	192	RVQ	18
• Network Drop Back Busy and Off-hook Queuing (see also package 62)			
• Remote Virtual Queuing			
Resident Debug	82	RSDB	9
• Restricted Call Transfer			
• Ring and Hold Lamp Status			
• Ringback Tone from Meridian 1 Enhancement			
Ringing Change Key	193	RCK	15
• Ringing Change Key			
Room Status	100	RMS	10
• Room Status			
Scheduled Access Restrictions	162	SAR	20
• Scheduled Access Restrictions			
• Secrecy Enhancement			
• Secretarial Filtering			
• Seizure Acknowledgment			

Package Name	Number	Mnemonic	Release
• Selectable Conferee Display and Disconnect			
• Selectable Directory Number Size			
Semi-Automatic Camp-On	181	SACP	15
• Attendant Blocking of Directory Number			
• Attendant Idle Extension Notification			
• Semi-Automatic Camp-On			
• Serial Port Expansion			
Series Call	191	SECL	15
• Series Call			
Set Relocation	53	SR	1
• Automatic Set Relocation			
• Short Buzz for Digital Telephones			
• Short Memory Test			
• Single Digit Access to Hotel Services			
Set-to-Set Messaging	380	STS	25
• Set-to-Set Messaging			
Single Term Access	228	STA	19
• Single Term Access			
• Slow Answer Recall Enhancement			
• Slow Answer Recall for Transferred External Trunks			
• Source Included when Attendant Dials			
SIP Gateway and Converged Desktop	406	SIP	4.0
Soft Switch	402	SOFTSWITCH	4.0
Spanish KD3 DID/DOD interface	252	KD3	20
• KD3 Direct Inward Dialing/Direct Outward Dialing for Spain			
• Special Signaling Protocols			
• Special Trunk Support			
• Speed Call Directory Number Access			
• Speed Call on Private Lines (see also package 1)			
• Speed-Up Data Dump			

Package Name	Number	Mnemonic	Release
Station Activity Records	251	SCDR	20
• Station Activity Records			
Station Camp-On	121	SCMP	20
• Station Camp-On			
Station Category Indication	80	SCI	7
• Station Category Indication			
Station Loop Preemption	106	SLP	10
Station Specific Authorization Codes	229	SSAU	19
• Station Specific Authorization Code			
• Station-to-Station Calling			
Stored Number Redial	64	SNR	3
• Stored Number Redial			
Supervisory Attendant Console	93	SUPV	8
• Supervisory Attendant Console			
Supervisory Console Tones	189	SVCT	20
• System Capacity Enhancements			
System Errors and Events Lookup	245	SYS_MSG_ LKUP	19
• System Message Lookup			
System Speed Call	34	SSC	2
• Speed Call/Autodial with Authorization Codes (see also package 1)			
• Speed Call, System			
• Speed Call Delimiter (see also package 34)			
• Telephones (PBX)			
• Teletype Terminal Access Control in Multi-Customer Environment (see also package 164)			
• Telset Call Timer Enhancement			
Time and Date	8	TAD	1
• Time and Date			
Tone Detector Special Common Carrier	66	SCC	7
Traffic Monitoring	168	TMON	

Package Name	Number	Mnemonic	Release
Trunk Anti-Tromboning	293	TAT	21
• Trunk Anti-Tromboning			
Trunk Barring	132	TBAR	20
• Trunk Barring			
Trunk Failure Monitor	182	TFM	15
• Trunk Failure Monitor			
• Trunk Failure Monitor Enhancement			
Trunk Hook Flash (Centrex)	157	THF	14
• Centrex Switchhook Flash			
• Italian Central Office Special Services (see also packages 129, and 131)			
• Trunk to Trunk Connections			
• Trunk Traffic Reporting Enhancement			
Trunk Verification from Station	110	TVS	9.32
• Trunk Verification from a Station			
• Uninterrupted Line Connection			
United Kingdom	190	UK	16
• Analog Private Network Signaling System (APNSS) (see also packages 122, 123, and 124)			
• UK Analogue Hardware Support			
Universal ISDN Gateways	283	UIGW	20
• Universal ISDN Gateway			
• Variable Flash Timing and Ground Button			
• Variable Guard Timing			
VIP Auto Wake Up	212	VAWU	17
• Hospitality Enhancements: V.I.P. Auto Wake Up			
Virtual Network Services	183	VNS	16
• Virtual Network Services			
• Virtual Network Services/Virtual Directory Number Expansion (see also package 58)			
• Voice Call			
Virtual Office	382	VIRTUAL_ OFFICE	25

Package Name	Number	Mnemonic	Release
• Branch Office			
• Emergency Services For Virtual Office			
• Internet Telephone Virtual Office			
• Virtual Office			
Virtual Office Enhancement	387	VOE	3.0
• Branch Office			
• Emergency Services For Virtual Office			
• Internet Telephone Virtual Office			
X08 to X11 Gateway	188	L1MF	15
• X08 to X11 Gateway			
Zone Call Admission Control	407	ZCAC	4.5
• Adaptive Network Bandwidth Management			

Chapter 4: Departmental Listed Directory Number

Contents

This section contains information on the following topics:

Feature description

The Departmental Listed Directory Number (DLDN) feature allows specified telephones sharing the same numbering plan to belong to one subgroup out of a possible six subgroups within a system customer group. Each Departmental Listed Directory Number (DLDN) subgroup is identified by one of the customer's Listed Directory Numbers (LDNs). Calls to specific Listed Directory Numbers (LDN), or dial-0 calls from subgroup telephones, are directed to the attendant console or consoles assigned to that LDN.

When the Departmental Listed Directory Number (DLDN) feature is implemented, a departmental attendant console is presented with calls from the following sources:

- Incoming external trunk calls routed to the LDN from:
    - an auto-terminate trunk (CO, FX, or WATS) whose Auto-Terminate Number (ATDN) is the LDN

- a Direct Inward Dialing (DID) trunk whose DID number is the same as the LDN

• Calls that originate from internal telephones or TIE trunks when:

- a telephone user dials the LDN
- a telephone user associated with a departmental attendant console dials 0, or
- a TIE-line user dials the LDN.

The DLDN feature associates attendant consoles with an LDN. Up to 63 attendant consoles can be associated with one LDN.

For call distribution purposes, all attendant consoles within a subgroup are made members of a circular list. When a call is received, it is presented to the next listed console after the one that was last offered a call, thus ensuring that LDN calls are distributed in an equitable way. LDN calls, dial-0 calls, and associated timed recalls are serviced according to a circular list for the particular LDN.

On receiving an LDN type call, the system searches for an idle attendant console and tests whether or not that console is configured to answer a call for the dialed Directory Number (DN). If the attendant console is not configured to answer calls for that LDN, the next idle attendant console is tested. If an attendant console that can answer the call is found, the call is presented with the appropriate Loop and Incoming Call Indicator (ICI) lamps lit. If no idle attendant console for the LDN is found, the call is placed in the incoming call queue for all attendant consoles within the customer group.

The Call Waiting indication is provided to all attendant consoles within the customer group. If an Incoming Call Indicator (ICI) key has been provisioned for the LDN, a lamp indication (with no buzz) is provided to all idle attendant consoles within the customer group and may be answered by pressing the appropriate key.

When an attendant presses the Release key, the system checks to see if there are any calls waiting in the queue. If there are calls waiting, it tests whether or not the attendant console, if it is next in the circular list, can answer the first call in the queue. If the call can be answered, it is presented to the attendant console. Otherwise it is put back into the queue and another call is sought. If no calls for the LDN are found, the attendant console is idled and the Release lamp is lit.

Operating parameters

An optional assignment of ICI keys is allowed to provide a visual indication of the LDN (LD 15).

If the DN Expansion package is equipped, all LDNs can have up to seven digits.

Feature interactions

Attendant Position Busy

If all attendant consoles in a LDN group are in position busy, calls to that LDN are not automatically presented to any attendant console in the customer group and will enter the attendant queue for that customer group. Other attendants outside the LDN queue may only answer LDN calls in the attendant queue by pressing the relevant LDN ICI key, if configured. No buzz is provided as the call is in the attendant queue and not the loop key.

Attendant Supervisory Console

The supervisory capabilities extend to all attendant consoles defined within the customer group. The attendant console serving as supervisor should be a member of every DLDN group so that it can serve all groups when operating in the Normal mode.

Call Forward Busy Call Forward No Answer Call Forward

Call Forward No Answer to the attendant and Call Forward Busy operate like Call Forward to 0, and are routed to any idle attendant console in the customer group.

Centralized Attendant Service

LDN calls are not screened for Centralized Attendant Service (CAS). When a CAS key is pressed at a CAS remote attendant console, LDN calls will be handled at the CAS main as if the DLDN feature did not exist.

Console Operations

DLDN is a way of directing attendant calls. The feature has some similarities to MTS, but it overrides Multi-tenant Service (MTS) and is therefore not affected by Console Presentation.

Directory Number

With the Network-Wide LDN feature, telephones using DLDN have access to two additional LDNs, even though DLDN is not supported over a network.

Interdepartmental Attendant Transfers

Interdepartmental Attendant Transfers operate normally, except that if there is a recall, it will be to the appropriate department rather than to the last attendant that extended the call.

Listed Directory Numbers, Network Wide

Departmental LDN is not supported over the network; however, this feature does provide two more LDNs for the DLDN feature.

Network-Wide Listed Directory Number

DLDN is not supported over a network; however, Network-Wide LDN provides two additional LDNs for DLDN.

Multiple Console Operation

Departmental Listed Directory Number (DLDN) supports the assignment of 63 consoles per DLDN.

Night Service

DLDN does not affect Night Service (including TAFAS). Calls presented to the LDN from an external source will queue for the night bell. All other attendant calls receive busy treatment if the night Directory Number (DN) is busy.

Feature packaging

Departmental Listed Directory (DLDN) package 76 has no other package dependencies.

Feature implementation

Task summary list

The following is a summary of the tasks in this section:

1. Table 1: LD 15 - Enable the Departmental Listed Directory Number feature for a customer. on page 71

   Enable the Departmental Listed Directory Number feature for a customer.
2. Table 2: LD 10 - Configure Departmental Listed Directory Number for analog (500/2500 type) telephones. on page 72

   Configure Departmental Listed Directory Number for analog (500/2500 type) telephones.
3. Table 3: LD 11 - Configure Departmental Listed Directory Number for Meridian 1 proprietary telephones. on page 73

   Configure Departmental Listed Directory Number for Meridian 1 proprietary telephones.

Table 1: LD 15 - Enable the Departmental Listed Directory Number feature for a customer.

Prompt	Response	Description
REQ:	CHG	Change.
TYPE:	LDN	Departmental Listed Directory Numbers
CUST		Customer number
	0-99	Range for Large System and Avaya Communication Server 1000E (Avaya CS 1000E) system.
- OPT	NLDN	Network wide LDN allowed. XLDN = Network wide LDN denied (default).
- DLDN	(NO) YES	(Disable) enable DLDN.
- LDN0	xxxx	Listed Directory Number Zero.
- LDA0	1 - 63 ALL	Attendant consoles associated with LDN 0.
- LDN1	xxxx	Listed Directory Number One.

Prompt	Response	Description
- LDA1	1 - 63 ALL	Attendant console number associated with LDN 1.
- LDN2	xxxx	Listed Directory Number Two.
- LDA2	1 - 63 ALL	Attendant console number associated with LDN 2.
- LDN3	xxxx	Listed Directory Number Three.
- LDA3	1 - 63 ALL	Attendant console number associated with LDN 3.
- LDN4	xxxx	Listed Directory Number Four.
- LDA4	1 - 63 ALL	Attendant console number associated with LDN 4.
- LDN5	xxxx	Listed Directory Number Five.
- LDA5	1 - 63 ALL	Attendant console number associated with LDN 5.
- ICI	xx LD0 xx LD1 xx LD2 xx LD3 xx LD4 xx LD5	Incoming Call Indication for Listed Directory Numbers Zero to Five (xx = key number 00-19).
To remove an LDN, enter an X before the Directory Number. An LDN cannot be removed if any attendant consoles are associated with it. To remove an associated attendant console, enter an X at the LDA prompt before the attendant number.		

Table 2: LD 10 - Configure Departmental Listed Directory Number for analog (500/2500 type) telephones.

Prompt	Response	Description
REQ:	CHG	Change.
TYPE:	500	Telephone type.
TN		Terminal number
	l s c u	Format for Large System, Avaya MG 1000B, and CS 1000E system where l = loop, s = shelf, c = card, u = unit.
LDN	(NO) 0-3	Telephone associated with LDN (0-3 or none). Choose NO to remove this telephone from the group.

Table 3: LD 11 - Configure Departmental Listed Directory Number for Meridian 1 proprietary telephones.

Prompt	Response	Description
REQ:	CHG	Change.
TYPE:	a...a	Telephone type. Type **?** for a list of possible responses.
TN		Terminal number
	l s c u	Format for Large System, MG 1000B, and CS 1000E system, where l = loop, s = shelf, c = card, u = unit.
LDN	(NO) 0-3	Telephone associated with LDN (0-3 or none). Choose NO to remove this telephone from the group.

Feature operation

No specific operating procedures are required to use this feature.

Chapter 5: D-channel Expansion

The D-channel Expansion feature increases the total number of possible D-channels in a multiple group system. The D-channel Expansion feature increases the number of physical I/O addresses permitted for D-channel application to 16 for each network group. For each MSDL physical I/O address, up to four ports are available for D-channel use. With the D-Channel Expansion feature, the software supports up to 255 D-channels.

For more information on the D-Channel Expansion feature, see *Avaya ISDN Primary Rate Interface Fundamentals, NN43001-569*.

Chapter 6: Dial Access to Group Calls

Contents

This section contains information on the following topics:

Feature description

This feature allows attendants and users of analog (500/2500 type) telephones, and Meridian 1 proprietary telephones to make a Group Call by dialing a Flexible Feature Code (FFC). telephone users may continue to use a Group Call key. The customer can define whether or not the originator of the Group Call has control of the active call. In the Group Call List, if GRPC = YES, the originator has control: when the originator goes on hook, the call is terminated. If GRPC = NO and the originator goes on hook, the Group Call acts like a conference call: the call remains active until all members go on hook.

For more information on group calls, see Group Call on page 423 in this book.

Operating parameters

All group stations must have Warning Tone Allowed (WTA) Class of Service.

Because analog (500/2500 type) telephones have no lamp state, there is no indication to the call originator that all group members have answered.

If a Group Call is originated using an FFC from a DN key of a Meridian 1 proprietary telephone, or a loop key on an attendant console, the DN lamp state does not display the status of the Group Call.

A Group Call member that has disconnected from the call cannot be reconnected to the call.

Feature interactions

The following features cannot be applied to a Group Call:

- Call Forward No Answer
- Call Forward Busy
- Call Join
- Call Park
- Conference
- Hunting
- Privacy Release, and
- Ring Again.

AC15 Recall: Transfer from Norstar

If Norstar sends a recall signal in order to initiate a consultation, the consultation will not be authorized because it is not possible to put a group call on hold. It is however possible to transfer a party to a group call using an AC15 trunk.

On Hold on Loudspeaker

If a group call is initiated from a set with Dealer Allowed Class of Service, the conference is built up on the assigned loop of the loudspeaker or speech monitor system channel since this is a potential On Hold on Loudspeaker call.

Feature packaging

Dial Access to Group Call requires the following packages:

- Group Call (GRP) package 48
- Flexible Feature Codes (FFC) package 139

Feature implementation

Task summary list

The following is a summary of the tasks in this section:

1. Table 4: LD 18 - Configure the Group Call List table for Group Call control. on page 79

   Configure the Group Call List table for Group Call control.
2. Table 5: LD 57 - Configure Flexible Feature Codes for Group Calls. on page 79

   Configure Flexible Feature Codes for Group Calls.

Table 4: LD 18 - Configure the Group Call List table for Group Call control.

Prompt	Response	Description
REQ	CHG	Change.
TYPE	GRP	Group Call data block.
CUST	xx	Customer number, as defined in LD 15
GRNO	0-63	Number of the Group Call list.
STOR	xx yyy...y	Group member number (xx) and associated DN (yyy...y).

Table 5: LD 57 - Configure Flexible Feature Codes for Group Calls.

Prompt	Response	Description
REQ	CHG	Change.
TYPE	FFC	Flexible Feature Codes.

Prompt	Response	Description
CUST	xx	Customer number, as defined in LD 15
GRPF	xxxx	Group Call code.
- GRCL	0-63	Group Call List number.

Feature operation

To make a Group Call,

- Press the Group Call key. All group members are automatically called. The LCD indicator beside the Group Call key flashes until all members have answered. Then it lights steadily.

or

- Dial the Group Call FFC. All group members are automatically called. When an originating station makes a Group Call using an FFC, all idle stations in the group are rung. Busy stations are given Call Waiting or Camp-On, if equipped, along with a special warning tone.

Chapter 7: Dial Intercom

Contents

This section contains information on the following topics:

Feature description

Dial Intercom (DI) allows a customer to arrange stations within the system into separate Dial Intercom Groups (DIGs). A total of 100 stations can belong to each Dial Intercom Group (DIG). One-digit dialing is required for a Dial Intercom Group (DIG) of up to 10 stations, and two-digit dialing is required for a DIG of up to 100 stations.

Meridian 1 proprietary telephones can be equipped with a separate DIG key/lamp pair for each DIG of which it is a member. Single-line telephone users can belong to only one DIG and may not have any non-DIG Directory Numbers (DNs).

Voice or ring may be specified on a DIG basis for Meridian 1 proprietary telephones. If voice is specified, an idle station rings once for two seconds. The calling party is then connected and may make a voice announcement. If ring is implemented, normal ringing is received until the called party answers. This feature provides the option of an announcement or a two-way speech path.

The ring option must be used if a 500 telephone is a member of the group.

Distinctive ringing for Dial Intercom

This feature allows a user to differentiate between an incoming call and a Dial Intercom (DI) call. The Dial Intercom (DI) ringing has a different cadence than the regular Directory Number (DN) ringing and Distinctive Ringing.

Distinctive Ringing for DI is assigned on a per-customer basis. The cadence is 0.5 sec. on and 0.5 sec. off, repeatedly.

Dial Intercom Handsfree Voice Call

Dial Intercom Handsfree Voice Call can be used with the M2317 and M2616 telephones.

Handsfree Voice Call provides the option of configuring VCC/DIG (with voice option) to be answered in either handsfree mode or loudspeaker only mode. Calls answered in handsfree (HVA) mode establish a two-way voice path, while those answered in loudspeaker only (HVD) mode establish only a one-way voice path from the calling telephone to the destination telephone.

Dial Intercom Handsfree Voice Call applies only to voice option DIG calls.

Operating parameters

A maximum of 2046 DIGs can be established per customer.

Calls are restricted to stations within the DIG only. Trunks cannot be accessed using the DIG key, and cannot be added to a DIG call using the Conference feature.

A DIG member number must be a single appearance Directory Number (DN) within a specified DIG.

DI analog (500/2500 type) telephones cannot dial the attendant or be dialed by the attendant.

A DI telephone cannot be assigned a member number that conflicts with the Special Prefix (SPRE) code. In the case of double-digit DIG values, the first digit cannot be the same as the SPRE code. For example, if the SPRE code is 7, the member number cannot be 7 or any number from 70 through 79. A two-digit SPRE code, such as 77, allows 99 DIG member numbers (00, 01-76, and 78-99). With no SPRE code defined, 100 DIG members are possible.

Call Transfer and Conference cannot take place to telephones outside the DIG.

Handsfree Voice Call allowed/denied is set at the system level and can only be used with digital telephones that have handsfree capabilities (such as M2317 and M2616), and requires Class of Service Handsfree Allowed (HFA) on the destination telephone, which is set at the telephone level.

Basic Rate Interface (BRI) telephones do not support the Handsfree feature.

Feature interactions

Auto Answer Back (AAB)

This feature is not affected by the Handsfree Voice Call feature.

Autodial Speed Call

The Dial Intercom code may be dialed using Autodial or Speed Call.

Automatic Line Selection

A Dial Intercom DN is selected by Incoming Ringing Line Selection and Outgoing Line Selection.

Call Forward Call Waiting

The Call Forward and Call Waiting features do not apply to a Dial Intercom appearance.

Call Party Name Display

The display on telephones connected by Dial Intercom shows the group member's DIG number plus Call Party Name Display information.

Call Pickup

Call Pickup may be used by Meridian 1 proprietary telephones if the telephones are all in the same DIG and Call Pickup Group and the ring option is specified for the DIG.

Call Pickup Network Wide

The Dial Intercom feature is not supported network wide. Any pickup attempt from a distant node to a local intercom call will be rejected, because the far-end user is considered as not being in the same intercom group.

Digit Display

The digit display will be cleared when the DIG key is pressed. When the user dials the DI code, the digits of the code are displayed. When the call is answered, the DI code of the calling party appears on the display of the called party.

If either party presses the Release key or goes on hook during a DIG call, the displays of both parties are cleared. If either party presses the Hold key, the display of the holding station is cleared but the display of the other station remains unchanged. When the held call is reestablished, the holding station redisplays the DIG number of the other party.

Conference Call Transfer

When using Conference or Transfer, the voice option is not provided if the call is terminated before the conference or transfer is completed. If an analog (500/2500 type) telephone is part of a Dial Intercom Group (DIG), the user of the telephone can conference only with another user whose telephone is within the same Dial Intercom Group (DIG).

Display of Calling Party Denied

Display information on sets that are involved in a Dial Intercom Group (DIG) call is based on the individual Class of Service of each set. If a DN is denied for a set involved in a DIG call, the DIG number for that set is replaced by one dash (–) in the case of 10 DIG stations. For 100 DIG stations, the DIG number is replaced by two dashes (– –).

Hot Line

The analog (500/2500 type) Hot Line telephones cannot be members of Dial Intercom Groups (DIGs).

Station features

DI can be used in combination with the following features:

Feature	Meridian 1 proprietary telephones	Analog (500/2500 type) telephones
Autodial Speed Call Digit	• • • • • • • • • • •	
Display Make Set Busy		
Override Release Hold Call		
Pickup Conference Call		
Transfer Ring Again		• • • •

Tones, Flexible Incoming

For Dial Intercom Group (DIG) calls with the voice (V) option, if the telephone receiving the call is busy, the user hears one buzz followed by a flashing indicator. This is how DIG works with or without FIT.

Feature packaging

Dial Intercom (DI) package 21 has no other package dependencies.

Feature implementation

Task summary list

The following is a summary of the tasks in this section:

1. Table 6: LD 15 - Enable Dial Intercom for a customer. on page 86

Enable Dial Intercom for a customer.

2. Table 7: LD 10 - Configure Dial Intercom for analog (500/2500 type) telephones. on page 86

   Configure Dial Intercom for analog (500/2500 type) telephones.

3. Table 8: LD 11 - Configure Dial Intercom for Meridian 1 proprietary telephones. on page 87

   Configure Dial Intercom for Meridian 1 proprietary telephones.

4. Table 9: LD 15 - Configure Handsfree Voice Call for the system. on page 87

   Configure Handsfree Voice Call for the system.

Table 6: LD 15 - Enable Dial Intercom for a customer.

Prompt	Response	Description
REQ:	CHG	Change.
TYPE:	FTR	Features and options
CUST		Customer number
	0-99	Range for Large System and Avaya Communication Server 1000E (Avaya CS 1000E) system.
- DGRP	0-2046	Maximum number of DIGs that can be defined for the customer. The maximum number of DIGs allowed is 2046.

Table 7: LD 10 - Configure Dial Intercom for analog (500/2500 type) telephones.

Prompt	Response	Description
REQ:	NEW CHG	Add, or change. Single line telephones cannot have both a Dial Intercom Group number and a standard DN. To add this feature, you must remove the telephone from the database and build it again, as a Dial Intercom Group member.
TYPE:	500	Telephone type.
TN		Terminal number
	l s c u	Format for Large System, Avaya MG 1000B, and CS 1000E system, where l = loop, s = shelf, c = card, u = unit.
DES	a...x	ODAS set designator. a...x = one-to-six character alphanumeric designator.
CUST	xx	Customer number, as defined in LD 15
DIG	xxxx yy	xxxx = Dial Intercom group number (0-2046). yy = member number (0-99) within the group.

Prompt	Response	Description
		The maximum number of DIGs allowed is to 2046.

Table 8: LD 11 - Configure Dial Intercom for Meridian 1 proprietary telephones.

Prompt	Response	Description
REQ:	CHG	Change.
TYPE:	a...a	Telephone type. Type **?** for a list of possible responses.
TN		Terminal number
	l s c u	Format for Large System, MG 1000B, and CS 1000E system, where l = loop, s = shelf, c = card, u = unit.
KEY	xx DIG aaa bb c	Add a Dial Intercom key, where: xx = key number aaa = group number (0-2046) bb = member number (0-99), and c = r (ring) or v (voice). The maximum number of DIGs allowed is to 2046.

Table 9: LD 15 - Configure Handsfree Voice Call for the system.

Prompt	Response	Description
REQ:	CHG	Change.
TYPE:	FTR	Features and options
CUST		Customer number
	0-99	Range for Large System and CS 1000E system.
- OPT	(HVD) HVA	Handsfree Voice Call (Denied) Allowed.

Feature operation

An example of a Dial Intercom call is listed below.

Dial Intercom Call

To make a Dial Intercom call:

1. Lift the handset and dial the **Intercom** key.
2. Dial the one- or two-digit code for the DIG member.

If your telephone and the telephone you are calling are configured for the voice option, you can deliver a voice message after two seconds of ringing.

To answer a Dial Intercom call when you are on a line other than your DIG line:

1. Release the current call or place it on hold.
2. Press **Intercom**.

Dial Intercom Handsfree Voice Call

Examples of both Handsfree Voice Call options are listed below.

HVA option

The originating telephone (telephone A) places a DIG call to the destination telephone (telephone B).

- Telephone B rings once.
- After one ring, telephone B automatically answers the call in Handsfree mode.

  The DN and handsfree LCDs are lit and a two-way voice path is established.

HVD option

Telephone A places a call to telephone B.

- Telephone B rings once.
- After one ring, telephone B automatically answers the call in loudspeaker only mode.

  The DN LCD is lit and the handsfree LCD remains dark, establishing a one-way voice path from telephone A to telephone B. At this point, telephone A is unable to hear the person at telephone B.

  To establish a two-way voice path, telephone B must either go off hook, or press the Handsfree button.

Chapter 8: Dial Pulse/Dual-tone Multifrequency Conversion

Contents

This section contains information on the following topics:

Feature description

With the Dial Pulse/Dual Tone Multifrequency Conversion feature, Dial Pulse (DP) signals from analog (500/2500 type) telephones, Dial Pulse (DP) TIE lines, Meridian 1 proprietary telephones, or attendant consoles are automatically converted to Dual-tone Multifrequency (DTMF) signals for transmission over trunks equipped for Dual-tone Multifrequency (DTMF) service. Dual-tone Multifrequency (DTMF) signals from single-line 2500 telephones are automatically converted for transmission over rotary-dial-only trunks, such as TIE lines. This eliminates the need for duplicate dials.

DTMF calling allows the use of 2500 telephones, equipped with push-button dials, to transmit digits through audible tones to the system equipment. This feature provides the ability to use any combination of telephones. However, 2500 telephones cannot use DTMF to control dictation equipment when the dictation trunk is specified as Dial Pulse (DP).

Operating parameters

There are no operating parameters associated with this feature.

Feature interactions

There are no feature interactions associated with this feature.

Feature packaging

This feature is included in base system software.

Feature implementation

There are no specific implementation procedures for this feature.

Feature operation

No specific operating procedures are required to use this feature.

Chapter 9: Dial Tone Detection

Contents

This section contains information on the following topics:

Feature description

The Dial Tone Detection (DTD) feature is needed because the first digit cannot be sent until the dial tone is detected on calls to a Public Switched Telephone Network (PSTN). This avoids the outpulsing of digits before the PSTN is ready to accept them, thus avoiding either the loss of digits or the misrouting of calls. The possibility of circumventing code dialing limitations is also minimized by the feature.

The feature is configurable on a route basis for all types of routes.

The time-out for the route is statistically averaged over the last eight times that Dial Tone Detection was involved. Either the running-average time or the pre-overlay programmed minimum time is used as the trunk time out, whichever is greater. Dial Tone Detection can be invoked every time an outgoing trunk route is selected, regardless of the selected feature.

Operating parameters

There are no operating parameters associated with this feature.

Feature interactions

Digital Trunk Interface (DTI) - Commonwealth of Independent States (CIS)

Dial tone detection is supported in the CIS, but with the limitation of low reliability of the tone provided by the Public Exchange.

ISDN Semi Permanent Connections for Australia

To convey D-channel signaling over an ISPC link, the route associated to the link at the system configured as MASTER must detect a dialtone.

Three Wire Analog Trunk - Commonwealth of Independent States (CIS)

Dial Tone detectors are supported with the limitations of the reliability of the tone provided by the Public Exchange.

Feature packaging

Dial Tone Detector (DTD) package 138.

Feature implementation

Gather data for each customer's number to be configured for the DTD feature.

Task summary list

The following is a summary of the tasks in this section:

1. Table 10: LD 13 - Create or modify data blocks for Digitone Receivers. on page 93

   Create or modify data blocks for Digitone Receivers.
2. Table 11: LD 16 - Create or modify data for trunk routes. on page 93

   Create or modify data for trunk routes.
3. Table 12: LD 17 - Modify the system hardware and software parameters. on page 93

   Modify the system hardware and software parameters.

Table 10: LD 13 - Create or modify data blocks for Digitone Receivers.

Prompt	Response	Description
...		
TYPE	DTD	Dial Tone Detection.

Table 11: LD 16 - Create or modify data for trunk routes.

Prompt	Response	Description
...		
DTD	(NO) YES	Dial Tone Detection is (is not) to be performed on this route.

Table 12: LD 17 - Modify the system hardware and software parameters.

Prompt	Response	Description
...		
DTDT	NO	No Dial Tone Detection tests are required.

Feature operation

No specific operating procedures are required to use this feature.

Chapter 10: Dialed Number Identification Service

The Automatic Call Distribution (ACD) Dialed Number Identification Service (DNIS) shows the last three or four digits of the dialed DNs received from auto-terminated Direct Inward Dialing (DID) and TIE trunks on the display for ACD agents. The maximum number of characters allowed is 27, including spaces.

In telemarketing environments, DNIS can reduce the time needed to serve a call. For example, the dialing plan can be configured so the DNIS digits represent product lines or services. The ACD agent can then answer incoming calls with the correct response

For more information about Dialed Number Identification Service, see *Avaya Automatic Call Distribution Fundamentals, NN43001-551*.

Chapter 11: Digit Display

Contents

This section contains information on the following topics:

Feature description

There are two types of Digit Displays: Attendant Console Digit Display and Meridian Modular Telephone Digit Display.

Attendant Console Digit Display

The M2250 attendant console is a Digital console with a 4-line, 40-character wide alphanumeric Liquid Crystal Display. The console's LCD displays the information presented in Table 13: LCD alphanumeric display information on page 97.

Table 13: LCD alphanumeric display information

Line	Display information
1	Displays the time and date.
2	Displays call source information.
3	Displays call destination information.
4	Displays console status information.

Meridian Modular Telephone Digit Display

This feature allows the automatic display of information relevant to normal call processing and feature activation on any Meridian Modular Telephone equipped with a 16-digit display. A key/ lamp pair is also provided to enable the station user to obtain information manually, independent of call processing activity.

Time and Date are displayed with an additional Time and Date (TAD) key.

⚠ **Caution:**

This option should not be used when a Prime DN appears on another telephone as a Prime DN. Severe real-time penalties will occur (ERR040 message).

The following display options are available:

- No Digit Display (NDD)

  This is the default option.

- Automatic Digit Display (ADD)

  This option allows the display of digit information during call processing. ADD allows the automatic display of a calling party number on an incoming call to the Prime DN on a telephone.

- Standard Delayed Display (DDS)

  Provides calling party information, displayed after answer only.

- Tandem Digit Display (TDD)

  With this option, when an incoming call is presented to a busy Meridian Modular Telephone with display, the Calling Line Identification and Call Party Name Display information is automatically displayed on the busy telephone.

Automatic displays will show the following:

- Number dialed
- Number of calling party
- Call Pickup
- Call Waiting party, and
- Time and date.

Press the Display (DSP) key, then the feature key to display information associated with these features:

- ACD in-calls

  If the Display Key is used to view information defined on the ACD DN key of an agent serving multiple queues, then the ACD DN displayed will be the current queue being served if the agent is active on a call. The last queue is served if the agent is not serving an ACD call or the Primary ACD DN if the agent is logged off.

- Autodial number

  When the telephone is inactive and the DSP key is pressed, followed by the autodial key, the number stored against the key will be displayed.

- Autoline

  To display the DN programmed for the Autoline key, the attendant presses the Autoline key when the console is idle or in Position Busy. On an analog console, to display a DN that is longer than eight digits, the attendant presses the display key after pressing the Autoline key.

- Buzz DN

  When the telephone is inactive and the DSP key is pressed, followed by the Buzz DN key, the number stored against the key will be displayed.

- Call Forward party

  When the telephone is inactive and the DSP key is pressed, followed by the Call Forward key, the number stored against the key will be displayed.

- Call Park

  The Park DN of the most recently parked call can be re-displayed on proprietary telephones equipped with displays, a Park key, and a Display key. This is done by pressing the Display key, then the Park key. The attendant can display the last call parked by pressing the Park key when no loop key is active.

- Call Pickup

  To display Call Pickup, press the Display key, followed by the Call Pickup key.

- Call Waiting party

  Pressing the Call Waiting key to answer a waiting call makes that call active. The call can be placed on hold by pressing the Call Waiting key again, or by pressing any idle DN key on the set. If the Display key is pressed before the Call Waiting key, the call waiting party information is displayed.

- Conference

  While in a conference call, the Display (DSP) key can be used to obtain information on other keys. However, the Display key is blocked when the CSD key is active.

- DN key (Meridian Modular Telephones)

  While the key is active (established, outgoing ringing) will show the source of the destination. While the key is active but not answered (that is, ringing) will show the source of the originator. While inactive will show the number stored that will be used for the 'last number redial' function (if configured).

- Hot Line

Hot Type I calls are supported by the Display key feature; pressing the Display key and then the Hot Type I key will show the target DN on the originating station's display.

- Message Waiting

  When the telephone is inactive and the DSP key is pressed, followed by the Message Waiting key, the number stored against the key will be displayed.
- Ring Again party

  When the telephone is inactive and the DSP key is pressed, followed by the Ring Again key, the number stored against the key will be displayed.
- Speed Call number

  To display a stored entry the user presses the Display key and the Speed Call key and dials the list number. The list number cannot be abbreviated.
- Voice Call party

  When the telephone is inactive and the DSP key is pressed, followed by the Voice Call key, the number stored against the key will be displayed.

Operating parameters

Digit Display must be enabled for all console types in LD 15, using the prompt OPT.

Only telephones equipped with a Digit Display module can use this feature.

The Display Time and Display Date key cannot be assigned to key 0.

Feature interactions

Attendant Break-In

During Attendant Break-In, the Attendant Console Digit Display shows the DN of the incoming call and the destination DN until the Attendant extends the incoming call to the destination DN and releases the connection.

Autodial Tandem Transfer

Digit Display allows the automatic display of information relevant to normal call processing if the sets have display capability and the Class of Service is ADD or DDS. When the THF key is pressed the display gets cleared, and pressing the ADL key causes the ADL digits to be displayed. However, no ADL digits will be displayed if no Tone and Digit Switch (TDS)/XCT is available to generate the Dual-tone Multifrequency (DTMF) tones for the ADL digits.

Automatic Redial

Dialed numbers are displayed when the Automatic Redial (ARDL) feature is activated. The calling party can dial digits even though a busy tone indication is given.

Digits dialed while on hold are not displayed.When the calling party accepts a redialed call, the dialed numbers are displayed. If the Display (DSP) key and appropriate RGA key are pressed while a call is on hold, the number redialed is displayed.

China - Flexible Feature Codes - Outgoing Call Barring Enhanced Flexible Feature Codes - Outgoing Call Barring

Meridian Modular Telephones with displays do not display the OCB level and the Station Control Password (SCPW) when OCB FFCs are dialed. This protects the security of the SCPW.

Centralized Multiple Line Emulation

The digit display of the station picking up a parked call recall shows the parked call's access code followed by the parked call's access-identification code. If the picked-up call is a group member call, the display shows the group number of the picked-up station.

Dial Intercom

The digit display will be cleared when the Dial Intercom Group (DIG) key is pressed. When the user dials the DI code, the digits of the code are displayed. When the call is answered, the DI code of the calling party appears on the display of the called party.

If either party presses the Release key or goes on hook during a DIG call, the displays of both parties are cleared. If either party presses the Hold key, the display of the holding station is cleared but the display of the other station remains unchanged. When the held call is reestablished, the holding station redisplays the DIG number of the other party.

Digital Private Network Signaling System (DPNSS1)/Digital Access Signaling System (DASS2) Uniform Dialing Plan (UDP) Interworking

The digit display rules for DPNSS1 UDP are based on what is currently done in an MCDN.

Group Hunt

Until a call is answered, the calling party will see the dialed DN. When the call is answered, the caller will see the dialed DN appended with the DN and name, if Calling Party Name Display (CPND) is equipped, of the called party. The terminating set will always see the originating DN appended by a Pilot DN.

Hot Line

A Display key on a telephone with a Hot Line appearance will display the Hot Line target DN data stored for that key.

INIT ACD Queue Call Restore

Call information associated with Digit Display is lost after system initialization and call restoration.

LOGIVOX Telephone

During manual dialing or last number redial, the display shows the dialed digits, even if the set has display denied Class of Service. If the set has LOGIVOX denied Class of Service, each digit is shown twice.

Override Override, Enhanced Override, Priority

The Digit Display of the telephone being overridden changes to the Directory Number (DN) of the overriding telephone once Priority Override is accomplished.

Pretranslation

The Pretranslation digit is displayed as it was dialed, but if the call is put on hold, the digits of the pretranslated DN are displayed

Feature packaging

Digit Display (DDSP) package 19 has no other feature package dependencies.

Feature implementation

Task summary list

The following is a summary of the tasks in this section:

1. Table 14: LD 15 - Configure Digit Display for attendant consoles for each customer. on page 103

   Configure Digit Display for attendant consoles for each customer.
2. Table 15: LD 11 - Configure Digit Display for Meridian Modular Telephones. on page 104

   Configure Digit Display for Meridian Modular Telephones.
3. Table 16: LD 12 - Configure Digit Display for each attendant console. on page 104

   Configure Digit Display for each attendant console.

Table 14: LD 15 - Configure Digit Display for attendant consoles for each customer.

Prompt	Response	Description
REQ:	CHG	Change.
TYPE:	ATT_DATA	Attendant console options
CUST		Customer number
	0-99	Range for Large System and Avaya Communication Server 1000E (Avaya CS 1000E) system.

Prompt	Response	Description
- OPT	(XDP) IDP	(Exclude) include Digit Display capability for attendant consoles of this customer.

Table 15: LD 11 - Configure Digit Display for Meridian Modular Telephones.

Prompt	Response	Description
REQ:	CHG	Change.
TYPE:	a...a	Telephone type. Type **?** for a list of possible responses.
TN		Terminal number
	l s c u	Format for Large System, Avaya MG 1000B, and CS 1000E system, where l = loop, s = shelf, c = card, u = unit.
CLS	(NDD) DDS ADD	Telephone is not equipped with a Digit Display. Calling Party information is displayed after call is answered (delayed display source). Calling Party information is displayed during call processing (Automatic Digit Display).
KEY	xx DSP xx TAD	Add a Digit Display key (must be key/lamp pair). Add a Time and Date key (must be key/lamp pair). xx = key number.

Table 16: LD 12 - Configure Digit Display for each attendant console.

Prompt	Response	Description
REQ	CHG	Change.
TYPE	2250	Attendant console type.
TN		Terminal number
	l s c u	Format for Large System, MG 1000B, and CS 1000E system, where l = loop, s = shelf, c = card, u = unit.
KEY	xx DCW xx DDT xx DPD xx DPS xx DTM xx MDT xx MTM	Configure console key for: • Display Call Waiting key • Display Date key • Display Destination key • Display Source key • Display Time key • Display/change Date key • Display/change Time key

Feature operation

No specific operating procedures are required to use this feature.

Chapter 12: Digital Private Network Signaling System

British Telecom's Digital Private Signaling System No. 1 (DPNSS1) is the open signaling protocol standard for intelligent private network digital connections. DPNSS1 provides the signaling capability to establish simple telephony and data calls, as well as supplementary features.

The following DPNSS1 features have been introduced:

- DASS2/DPNSS1 – Integrated Digital Access
- DPNSS1 Attendant Call Offer
- DPNSS1 Attendant Timed Reminder Recall and Attendant Three-party Service
- DPNSS1 Call Back when Free and Next Used
- DPNSS1 D-channel Handler Interface Expansion
- DPNSS1/DASS2 to ISDN PRI Gateway
- DPNSS1 Extension Three-party Service
- DPNSS1 Loop Avoidance
- DPNSS1 Redirection
- D-Channel Interface Expansion for DASS2/DPNSS1
- DPNSS1 Route Optimization
- DPNSS1 Step Back on Congestion
- DPNSS1 Executive Intrusion, and
- DPNSS1/DASS2 Uniform Dialing Plan Interworking.

For more information on DPNSS1, see *Avaya DPNSS1 Fundamentals (NN43001-572)*.

Chapter 13: Digital Trunk Interface - Commonwealth of Independent States

Contents

This section contains information on the following topics:

Feature description

The information presented in this section does not pertain to all market regions. Contact your system supplier or your Avaya representative to verify support of this product in your area.

The Commonwealth of Independent States (CIS) Digital Trunk Interface (DTI) feature allows the system to connect to Direct Inward Dialing (DID)/Central Office Trunk (COT) trunks to a CIS Public Exchange/Central Office and to a CIS toll exchange.

To satisfy the unique requirements of CIS DTI signaling, two new trunk cards have been introduced: a dual 2 Mbps Enhanced Network (ENET) styled digital trunk card (CDTI2). The CDTI2 card provides 60 voice or data 64 kbps channels. The card occupies one card slot on the common equipment shelf (CDTI2).

In addition to most of the features provided by digital trunks, the CDTI2 card provides the following features intended for the CIS market:

- CIS digital trunk signaling (outgoing, incoming toll, and incoming local calls)
- Automatic Number Identification (ANI) transmission for outgoing calls on request from the Public Exchange
- Special disconnect procedure (two-way release) on incoming local answered calls initiated by the Public Exchange to provide Malicious Call Trace
- Unanswered free special service calls – outgoing calls that remain unanswered are recognized in a special manner to allow the called party (special service operator) to disconnect the calls
- CIS transmission plan
- Downloading the required firmware mode per loop, and
- Dial tone provided internally to the calling party by the system after seizure of an outgoing CIS trunk. However, for outgoing call terminating to a busy, vacant, invalid, or restricted DN, the system does not provide busy/overflow tone. The Public Exchange will send the tone on the speech path.

The CIS DTI trunk provides significant improvement on real-time impact for dial pulse outpulsing and digit collection by transferring these processes from the software to the firmware. The trunk state change validation timing is performed by the firmware. A Firmware Unproductive Timer is used to prevent a call on a CIS trunk from remaining unanswered for too long.

Operating parameters

The CDTI2 card does not support Periodic Pulse Metering, continuous pulse detection, or echo suppression.

The only line signaling supported for CIS is a two-bit ABCD protocol.

The data in ANI always refers to the originator of the outgoing call. If the call is transferred, the ANI information is not changed and therefore may be different than that of the set currently involved in the call.

On outgoing toll calls, there is no delay. On outgoing local calls, there is a 700 millisecond delay in the Answer signal recognition before the call is established.

Incoming and outgoing trunks cannot be mixed within the same route.

Toll Operator Break-In/Trunk Offer is not supported.

Toll Operator Manual Ringing is not supported.

MF Shuttle Register Signaling is not supported.

Only ANI transmission is supported.

Feature interactions

Authorization Code

An extension may refer to an Authorization Code to seize an outgoing CIS DTI trunk. The Authorization Code category is used to build the Automatic Number Identification (ANI) message. Thus, a set having a CIS restricting call category can complete a call to the Public Network using the Authorization Code.

Called Party Disconnect Control

This feature may not be used in the CIS market because of its signaling requirements.

Computer to circuit-switched network interface

Computer to circuit switched network Interface is not supported on CDTI2 because the protocol conversion is not supported.

Call Detail Recording

If ANI is requested to be output in the Call Detail Recording (CDR) record, it will not refer to the CIS DTI2 ANI.

Data Transmission

All features connected with Data Transmission must be used with caution, because the ANI interaction can happen at any time during an outgoing call, thus destroying the transmitted data and disrupting the call.

Dial Tone Detection

Dial tone detection is supported, but with the limitation of low reliability of the tone provided by the Public Exchange.

Incoming Digit Conversion Enhancement Incoming DID Digit Conversion

The construction of an ANI message does not care if Incoming Digit Conversion is used. The DN sent as ANI is the actual DN of the set, not necessarily the Direct Inward Dialing (DID) number to dial to reach the set. Therefore, if an external party uses a DN, delivered in an ANI message, for making a call to the corresponding extension, the call may fail.

Japan DTI2

All features related to Japan DTI2 may not be used, because the proper Scan and Signaling Distributor (SSD) messages are not supported in the CDTI2 firmware.

Multiple Appearance Directory Number

Since the ANI category is defined on a per set basis, two stations with the same Multiple Appearance Directory Number (MADN) can be assigned different ANI categories.

Periodic Pulse Metering

Periodic Pulse Metering is not supported.

Pulsed E and M DTI2 Signaling

Pulsed E&M is not supported.

R2MFC Calling Number Identification

The category (CAC) used to build the R2MFC Calling Number Identification (CNI) for the analog, digital, and Basic Rate Interface (BRI) sets is also used to build the CIS Automatic

Number Identification (ANI). The meaning of CAC is different between the R2MFC CNI signalization and the CIS signalization (analog BRI and digital). R2MFC CAC prompt values are in the range of 0-10, with the default value of 0. CIS CAC prompt values are in the range of 0-9, with the default value of 3.

If the MFC package is equipped, but not the CIST package, the CAC prompt uses the R2MFC range and default. If the CIST package is equipped, whether or not the MFC package is equipped, the CAC prompt uses the CIS range and default.

Special Dial Tones after Dialed Numbers

Special Dial Tones can be used to provide dial tone after the system user has dialed the digit "9" (Local Exchange access code).

Tandem Switching

If an ISDN TIE incoming trunk (MCDN, QSIG, DPNSS1) with Calling Line Identification (CLID) and Originating Line Identification (OLI) available seizes the CIS DTI2 outgoing trunk, the ANI DN to be used for sending to the CIS Public Exchange is extracted from this CLID/OLI.

In any other case, the ANI sent to the CIS Public Exchange is based on the local system node (that is, tandem node) definitions.

Virtual Network Services

Virtual Network Services via CIS DTI2 is not supported.

Feature packaging

This feature is packaged as Commonwealth of Independent States Trunk Interface (CIST) package 221.

The following packages are required:

- Flexible Tones and Cadences (FTC) package 125
- 2 Mbps Digital Trunk Interface (DTI2) package 129
- International Supplementary Features (SUPP) package 131
- Flexible Numbering Plan (FNP) package 160
- Meridian 1 Intelligent Peripheral Equipment (XPE) package 203

- Meridian 1 Extended Conference, TDS and MFS (XCT0) package 204, and
- Meridian 1 Superloop Administration (XCT1) package 205.

Feature implementation

Task summary list

The following is a summary of the tasks in this section:

1. Table 17: LD 17 - Change system configuration data on page 115

   Change system configuration data.
2. Table 18: LD 73 - Define DTI2 data on page 115

   Define DTI2 data.
3. Table 19: LD 73 - Define the SICA table for CDTI2 on page 116

   Define the SICA table for CDTI2/CSDTI2.
4. Table 20: LD 97 - Define dial pulse make-break ratio on page 118

   Define dial pulse make-break ratio.
5. Table 21: LD 15 - Define busy tone/overflow tone time out. on page 119

   Define busy tone/overflow tone time out.
6. Table 22: LD 16 - Add or change route data for CIS DTI2 trunks. on page 119

   Add or change route data for CIS DTI2 trunks.
7. Table 23: LD 16 - Add or change route data for an incoming, non CIS DTI2, trunk on page 121

   Add or change route data for an incoming, non CIS DTI2, trunk.
8. Table 24: LD 14 - Add or change trunk data for CIS DTI2 incoming and outgoing trunk on page 121

   Add or change trunk data for CIS DTI2 incoming and outgoing trunk.
9. Table 25: LD 10 - Specify ANI category for CIS DTI2 calls on page 122

   Specify ANI category for CIS DTI2 calls.
10. Table 26: LD 11 - Specify ANI category for CIS DTI2 calls. on page 122

    Specify ANI category for CIS DTI2 calls.

11. Table 27: LD 12 - Specify ANI category for CIS DTI2 calls on page 123

    Specify ANI category for CIS DTI2 calls.
12. Table 28: LD 27 - Add or change Digital Subscriber Loop (BRI set) for CIS on page 123

    Add or change Digital Subscriber Loop (BRI set) for CIS.
13. Table 29: LD 88 - Add or change the Authcode data block on page 123

    Add or change the Authcode data block.
14. Table 30: LD 56 - Configure the dial tone on page 124

    Configure the dial tone.
15. Table 31: LD 56 - Configure Tone to Last Party on page 124

    Configure Tone to Last Party.
16. Table 32: LD 18 - Add or change Speed Call lists on page 125

    Add or change Speed Call lists, System Speed Call lists, Group Call lists, Enhanced Hot Line lists, Pretranslation lists, and Special Service lists. Special Service lists can now handle the Special Service Unanswered Call (SSUC) call type.

Table 17: LD 17 - Change system configuration data

Prompt	Response	Description
REQ	CHG	Change.
TYPE	PARM	System Parameters
...		
- PCML	(MU) A	System Companding Law.
TYPE	CEQU	Common Equipment
...		
- DTI2	<loop> <loop>...	Define CDTI2 loops exactly like existing DTI2/ SDC2.

Table 18: LD 73 - Define DTI2 data

Prompt	Response	Description
REQ	CHG	Change.
TYPE	DTI2	DTI2 Data Block.
FEAT	LPTI	Loop timers and some other per DTI2 loop defined parameters.
LOOP	<loop>	DTI2 loop number.

Prompt	Response	Description
CDTI2	YES	CDTI2 card.
P DIGIT (S)	PXXX	DP outpulsing will be sent on signaling bit A.
P METR (R)	NO	Pulse Metering.
SASU	1024	Seize Acknowledgment Supervision timer is defined in milliseconds (rounded to the closest multiple of 128 milliseconds).
MFAO	YES	Multi-frame alignment option used.
SZNI	NO	Seize Not Idle option not used.
LCLB	NO	Lockout Clear Back option (send CLR-BK signal to DID in lockout) not used.
UCFS	1101	Unequipped Channel Fault Signal – ABCD bits to be sent on unequipped channel. The default value of 1101 is acceptable.
MFF	(AFF) CRC	Alternate Frame Format or CRC4 may be chosen.
...		
FRFW	NO	Prompted only if French Type Approval (FRTA) package 197 is equipped.
CISFW	YES	Prompted only if Commonwealth of Independent States – Digital Trunk Interface (CIST) package 221 is equipped and CDTI2 = YES.

Table 19: LD 73 - Define the SICA table for CDTI2

Prompt	Response	Description
REQ	CHG	Change.
TYPE	DTI2	DTI2 Data Block.
FEAT	ABCD	Signaling category.
SICA	2-16	SICA table number.
...		
Incoming/ Outgoing Calls		
IDLE(S)	0101, 1101	Idle on backward sent, where: 0101 = incoming trunks (local and toll), and 1101 = outgoing trunks.
IDLE(R)	0101, 1101	Idle on backward sent, where: 0101 = incoming trunks (local and toll), and 1101 = outgoing trunks.

Prompt	Response	Description
FALT(S)	1101	Fault (referred to as blocked in CCITT terminology).
FALT(R)	1101	Fault (referred to as blocked in CCITT terminology).
Incoming Calls		
E_SEZ(R)	1001	Seize.
SEZD(R)	NO	Seize for voice calls.
P CALL (R)	NO	Signal sent during seize by an incoming CO trunk.
SEZA (S)	1101	Seize Acknowledge.
- TIME	150	Time in milliseconds.
P DIGT(R)	Pxxx	DP Digits received decadic pulses.
NRCV(S)	NO	Number received.
P EOSF(S)	NO	Pulsed End of Selection Free is not used.
EOSF(S)	NO 1001	Steady End of Selection Free, where: NO = local trunk, and 1001 = incoming toll trunk.
P EOSB(S)	NO	Pulsed End of Selection Busy is not used.
EOSB (S)	0001	Steady End of Selection Busy.
P OPCA(R)	NO	Operator calling.
E_CONN(S)	1001 1101	Connect Send (Answer), where: 1001 = local trunk, and 1101 = incoming toll trunk.
CONN(R)	1001 0001	Connect received, where: 1001 = local trunk, and 0001 = incoming toll trunk.
P RRC(S)	NO	Register recall.
P BURS(S)	NO	Bring up receiver for L1 networking.
P BURS(R)	NO	Bring up receiver for L1 networking.
CLRB(S)	0001 1001	Clear Back (B Ring Off), where: 0001 = local trunk, and 1001 = incoming toll trunk.
CLRF(R)	0001 NO	Clear Forward (A Ring Off), where: 0001 = local trunk (used only to start two-way release), and NO = incoming toll trunk.
P OPRS(R)	NO	Operator manual recall.
P NXFR(S)	NO	Network transfer.
P ESNW(S)	NO	ESN wink.
P CAS(S)	NO	Centralized attendant.

Prompt	Response	Description
Outgoing Calls		
E_SEZ(S)	1001	Seize.
SEZD(S)	NO	Seize for data calls.
SEZA(R)	1101	Seize Acknowledge.
P WNKS(R)	NO	Wink start.
P EOS(R)	NO	End of selection busy.
E_CONN(S)	NO	Connect.
E_CONN(R)	1001	Connect Receive (Answer).
P OPRC(R)	NO	Operator recall for special services.
P BURS(S)	NO	Bring up receiver for L1 networking.
P BURS(R)	NO	Bring up receiver for L1 networking.
CLRB(R)	0001	Clear Back (B Ring Off).
CLRF(S)	NO	Clear Forward (the same as the IDLE(S) signal).
P NXFR(R)	NO	Network transfer.
P ESNW(R)	NO	ESN wink.
P CAS(R)	NO	Centralized Attendant Service.

Table 20: LD 97 - Define dial pulse make-break ratio

Prompt	Response	Description
REQ	CHG	Change.
TYPE	SYSP	System parameters.
INTN	YES	A-law should be used as system companding law.
...		
P10R	(50)-70	Make-break ratio for primary 10 pulses per second dial pulse dialing.
P12R	(50)-70	Make-break ratio for secondary 10 pulses per second dial pulse dialing.
P20R	(50)-70	Make-break ratio for 20 pulses per second dialing.

Table 21: LD 15 - Define busy tone/overflow tone time out.

Prompt	Response	Description
REQ:	NEW CHG	New or change.
TYPE:	TIM	Gate opener.
...		
- BOTO	30	Busy tone/overflow tone time out (in seconds).

Table 22: LD 16 - Add or change route data for CIS DTI2 trunks.

Prompt	Response	Description
REQ	NEW CHG	New or change.
TYPE	RDB	Route Data Block.
...		
TKTP	DID COT	Trunk type, where: DID = incoming trunks route, and COT = outgoing trunks route.
...		
DTRK	YES	Digital trunk.
DGTP	DTI2	Digital trunk type for route.
...		
ICOG	ICT OGT	Incoming trunk. Outgoing trunk.
...		
CNTL	YES	Changes to controls or timers.
TIMR	DDL 0	Delay Dial Timer not needed.
TIMR	DSI 49992	Disconnect supervision timer (five-second value, rounded to the nearest 128 ms.).
TIMR	EOD 13952	End of dial timer (default value).
TIMR	SFB 25	Seize Fail Busy timer. The recommended value for trunks with seizure supervision is 25 seconds.
TIMR	GTI 0	Incoming Guard timer must be defined equal to zero. Incoming CIS DTI2 trunks only. For CIS DTI2 trunks no guard timing is necessary on the incoming side. Immediately after sending the "IDLE" signal, the incoming trunk may be reseized by the CO.
TIMR	ATO 128-(4992)-65408	ANI time out timer in milliseconds. For CIS outgoing trunk routes this defines the time delay

Prompt	Response	Description
		performed after the outpulsing of the toll access code. During this delay further outpulsing is temporarily halted until the special message from the card firmware confirms that a successful ANI request/response interaction has been performed.
...		
NEDC	ORG ETH	Near end disconnect control, where: ORG = originating end disconnect control for incoming calls, and ETH = either end control for outgoing calls.
FEDC	ORG ETH	Far end disconnect control, where: ORG = originating end disconnect control for incoming calls, and ETH = either end control for outgoing calls.
CDPC	NO	The system is not the only controlling party on incoming calls.
...		
OPCB	NO	External operator features not allowed on this route.
...		
CGPC	NO	Calling party control of calls not enabled.
CDCT	NO	Called party control of call is not enabled.
DDO	NO	Do not delay digit outpulsing for DOD trunks.
...		
DTD	NO	Dial tone detection is not to be performed on this route.
...		
CDR	YES	CDR to output for calls on trunks in this route.
...		
OAL	YES	CDR on all outgoing calls.
...		
OAN	NO	CDR on answered outgoing calls. It is not used because of free special service calls, which are not answered.
NATL	NO	North American toll scheme is not used.

Prompt	Response	Description
TDG	8	Toll digit (list of digits after the trunk access code which indicate toll calls). This can also be defined in LD 18.
...		
PRDL	YES	Partial dial timing is equipped using EOD.
DNSZ	(0)-7	Number of digits expected on DID routes. 0 (the default) indicates no fixed value. This value must be defined according to the numbering plan.
...		
BTT	30	Duration of busy/overflow tone to be returned on DID route in seconds.
...		
LEC	0-9999999	Local Exchange Code.
ADDG	0-9	Additional digit.
CAC	0-(3)-9	Route ANI category.
ANDN	0-9999999	Route ANI DN.

Table 23: LD 16 - Add or change route data for an incoming, non CIS DTI2, trunk

Prompt	Response	Description
REQ	NEW CHG	New or change.
TYPE	RDB	Route Data Block.
...		
ICOG	ICT IAO	Incoming trunk. Incoming and outgoing trunk.
...		
CAC	0-(3)-9	Route ANI category.
ANDN	0-9999999	Route ANI DN.
RDNL	0-(4)-7	Remote DN Length.

This trunk may be any kind of trunk. If this trunk, used as an incoming trunk, originates an outgoing call to a CIS DTI2 trunk, its CAC and ANDN are used in the ANI information sent out.

Table 24: LD 14 - Add or change trunk data for CIS DTI2 incoming and outgoing trunk

Prompt	Response	Description
REQ	NEW CHG	New or change.

Prompt	Response	Description
TYPE	DID COT	Direct Inward Dialing (for incoming trunks), or Central Office Trunk (for outgoing trunks).
...		
SICA	2-16	Signaling category table number. Standard default SICA table (number 1) may not be used for CIS DTI2 trunks. CIS DTI2 trunks for incoming local and incoming toll calls must have different SICA tables.
PDCA	(1)-16	PAD table number.
PCML	A	Only A-law companding may be used on the CIS DTI2 trunk line.
...		
CIST	(NO) YES	This prompt appears for incoming trunks only (ICOG = ICT in LD 16), where: YES = toll trunk, and NO = local trunk.
...		
CLS	(DIPF DIP	Dial pulse execution, where: DIP = outpulsing by firmware, digit collection – traditional, by software, and DIPF = outpulsing and digit collection are performed by firmware.
	(P10) P12	Make-break ratio for dial pulse dialing.

Table 25: LD 10 - Specify ANI category for CIS DTI2 calls

Prompt	Response	Description
REQ:	NEW CHG	Add or change.
TYPE:	500	500/2500 telephone data block.
...		
CAC	0-(3)-9	Specify ANI category for CIS DTI2 calls.
CLS	(DNAA) DNAD	DN of set (allowed) not allowed for use in ANI messages.

Table 26: LD 11 - Specify ANI category for CIS DTI2 calls.

Prompt	Response	Description
REQ:	NEW CHG	Add or change.
TYPE:	a...a	Telephone type. Type **?** for a list of possible responses.

Prompt	Response	Description
...		
CAC	0-(3)-9	Specify ANI category for CIS DTI2 calls.
CLS	(DNAA) DNAD	DN of set (allowed) not allowed for use in ANI messages.

Table 27: LD 12 - Specify ANI category for CIS DTI2 calls

Prompt	Response	Description
REQ	NEW CHG	Add or change.
TYPE	2250	Attendant console type.
...		
DNAN	(DNAA) DNAD	For CIS ANI purposes, the ANI DN will be LDN0 (defined in LD 15).

Table 28: LD 27 - Add or change Digital Subscriber Loop (BRI set) for CIS

Prompt	Response	Description
REQ	NEW CHG	Add or change.
TYPE	DSL	Digital Subscriber Loop.
...		
CAC	0-(3)-9	Specify ANI category for CIS DTI2 calls.
CLS	(DNAA) DNAD	DN of set (allowed) not allowed for use in ANI messages.

Table 29: LD 88 - Add or change the Authcode data block

Prompt	Response	Description
REQ	NEW CHG	New or change.
TYPE	AUB	Authcode data block.
...		
CLAS	(0)-115	Classcode value assigned to Authcode (NAUT).
...		
NCOS	(0)-99	Network Class of Service group number.
CAC	0-(3)-9	ANI category for CIS DTI2 calls.

Table 30: LD 56 - Configure the dial tone

Prompt	Response	Description
REQ	NEW CHG	New or change.
TYPE	DTAD	Special dial tone after dialed number.
DDGT	9	Use "9" as the outgoing local access code.
TONE	SRC1	Dial tone to be provided after the dialed digit 9 (Source Tone 1).
...		
REQ	NEW CHG	New or change.
TYPE	FTC	Flexible Tones and Cadences data block.
TABL	0-31	FTC table number.
DFLT	0-31	Default FTC table.
...		
SRC	YES	Change Source Tones (SRC1-SCR8).
SRC1		Source Tone 1.
TDSH	0 0 0 3	Tone number 3 provides 400 Hz, -23 db.
XTON	159	NT8D17 TDS tone code: 420 Hz, -25 db, A-law.
XCAD	0	NT8D17 cadence code for FCAD (steady tone).

Table 31: LD 56 - Configure Tone to Last Party

Prompt	Response	Description
REQ	NEW CHG	New or change.
TYPE	FTC	Flexible Tones and Cadences data block.
TABL	0-31	FTC table number.
DFLT	0-31	Default FTC table.
RING	<CR>	
HCCT	YES	Change the TDS card controlled cadence tones.
...		
TLP		Tone to Last Party.
TDSH	0 0 31 3	Cadence 31 in MCAD table will provide repeating 256 ms burst and 256 ms silence. Tone number 3 provides 440 Hz, -23 db.
XTON	159	NT8D17 TDS tone code: 420 Hz, -25 db, A-law.

Prompt	Response	Description
XCAD	31	NT8D17 cadence code for FCAD.
TLTP	30	Tone to Last Party timer in seconds.
...		
REQ	NEW CHG	New or change.
TYPE	MCAD	Master Cadence table.
WCAD	31	Cadence number.
CDNC	0051 0051	Repeating 256 ms burst and 256 ms silence.
...		
REQ	NEW CHG	New or change.
TYPE	FCAD	Firmware Cadence table.
WCAD	31	Cadence number.
CDNC	0060 0060	Repeating 300 ms burst and 300 ms silence.

Table 32: LD 18 - Add or change Speed Call lists

Prompt	Response	Description
REQ	NEW CHG	Add, or change.
TYPE	SSL	Special Service List.
SSL	1-15	SSL number.
SSDG	0-999	Special Service Digits combination.
CDPC	(NO) YES	Called Party Control mark.
TOLL	(NO) YES	Toll access code mark.
ALRM	(NO) YES	Alarm digits mark.
TNDM	(NO) YES	Tandem mark. Send MFC "H" tandem signal.
SSUC	(NO) YES	Special Service Unanswered Call mark. If the outgoing call is recognized as SSUC (first 1-4 digits outpulsed to the trunk are equal to the SSDG with SSUC = YES), then such a call requires some specific disconnect treatment.

Feature operation

No specific operating procedures are required to use this feature.

Chapter 14: Digitone Receiver Enhancements

Contents

This section contains information on the following topics:

Feature description

The Digitone Receiver Enhancements feature consists of the Digitone Receiver Time out Enhancement and the Quadruple Density Digitone Receiver Card.

An enhancement to Digitone receiver (DTR) time out prevents the situation in which the far-end of an outgoing call from a Dual-tone Multifrequency (DTMF) telephone or trunk is answered before speechpath can be established.

This problem can occur when trunks without answer supervision are used, and the called party answers quickly. Without answer supervision, the speech path is established upon time out of the end-of-dialing timer. It is possible for the far-end station to answer before this time out.

The timer enhancement will prevent this situation from occurring by holding back outpulsing of the last digit until a half-second before end-of-dialing time out. This leaves only a half-second interval in which the far-end station could answer before speechpath is established.

This DTR timer enhancement applies to DTRs of all densities, and for all trunk calls made from DTMF telephones or trunks, except for:

- MFC or MFE calls
- terminating trunks that have answer supervision
- Electronic Switched Network (ESN) calls

Operating parameters

This feature is not supported on the 1.5 Mbit Digital Trunk Interface (DTI).

Feature interactions

There are no feature interactions associated with this feature.

Feature packaging

This feature is included in base system software.

Feature implementation

Table 33: LD 13 - Create or modify data blocks for Digitone Receivers.

Prompt	Response	Description
...		
TN		Terminal number
	l s c u	Format for Large System, Avaya CS 1000 Media Gateway 1000B (Avaya MG 1000B), and Avaya Communication Server 1000E (Avaya CS 1000E) system, where l = loop, s = shelf, c = card, u = unit.
CDEN	4D	Enter 4D if the unit is on a quadruple density circuit pack (not allowed if the network loop is not configured for quadruple density).

Feature operation

No specific operating procedures are required to use this feature.

Chapter 15: Direct Inward Dialing Call Forward No Answer Timer

Contents

This section contains information on the following topics:

Feature description

This feature introduces the Direct Inward Dialing Call Forward No Answer (DFNR) timer that, when expired, allows an unanswered Direct Inward Dialing (DID) call to be routed to the attendant after the last stage of Call Forward No Answer (CFNA) or hunt treatment has been completed (the maximum number of CFNA or hunt steps is two). The DFNR timer is customer-defined as a number of ring cycles in LD 15.

The operation of the DFNR option can be overridden or disabled, depending on the definition of the FNAD option in LD 15. If FNAD has been defined as attendant (ATT), the DFNR timer is overridden, since unanswered DID calls are automatically routed to the attendant. If FNAD has been defined as NO, DFNR is disabled. If FNAD has been defined as Hunt (HNT) or Forward DN (FDN), the DFNR timer is applied after the last stage of CFNA or hunt treatment has been completed.

Operating parameters

DFNR does not apply to Automatic Call Distribution (ACD) calls, nor does it apply to non-DID calls.

The DFNR overrides the Forward Number Allowed (FNA) or Forward Number Denied (FND) Class of Service of the called party.

Feature interactions

Attendant Recall

The Direct Inward Dialing Call Forward No Answer Timer does not apply to an answered DID call that is extended to an unanswered station by the attendant – the call is recalled to the attendant using the Attendant Recall feature.

Call Forward No Answer Hunting

Call Forward No Answer and Hunting take precedence over the Message Center feature.

Call Waiting Redirection

The Direct Inward Dialing Call Forward No Answer Timer is applied after the last stage of Call Forward No Answer or SFNA treatment resulting from the Call Waiting Redirection feature for DID Call Waiting calls.

Feature packaging

The Direct Inward Dialing Call Forward No Answer (DFNR) feature requires the Flexible Feature Codes (FFC) package 139.

Feature implementation

Table 34: LD 15 - Define the Number of Ring Cycles.

Prompt	Response	Description
REQ:	NEW CHG	Add. Change.
TYPE:	RDR	Call Redirection
...		
- DFNR	(0)-15	DID Forward No Answer Ring cycles, prompted if the FNAD prompt is not set to ATT or NO. Defines the number of ringing cycles before a DID call is Slow Answer recalled to the attendant console after the last stage of CFNA or Hunt treatment has been completed (the maximum number of CFNA or hunt steps is two). If DFNR = 0 then DID CFNA is disabled

Feature operation

No specific operating procedures are required to use this feature.

Chapter 16: Direct Inward Dialing Recall Features on DTI2 for Italy

Contents

This section contains information on the following topics:

Feature description

Direct Inward Dialing (DID) Recall Features on DTI2 for Italy consists of DID Offering and DID Recall.

DID Offering

When a DID call placed on a DTI2 trunk terminates on a busy set, the system replies by sending an End of Selection Busy (EOSB) signal on the calling channel to inform the Public Exchange/Central Office that no further call modification will be performed. Busy tone is returned while waiting for the release signal from the Central Office (IDLE). The new DID Offering feature enables the external Central Office operator to reroute the call to the attendant by sending the Operator Recall Signal (OPRS) instead of the IDLE signal. Upon receipt of the OPRS signal, the call is presented to the attendant console on the Recall (RLL) Incoming Call Indicator (ICI) key.

DID Recall

When an established DID call placed on a DTI2 trunk is released by called party (internal set), the system sends a Clear Backward (CLRB) signal on the calling channel to inform the Central Office that the call has been disconnected. Upon receipt of this signal, the Central Office should reply with the IDLE signal to confirm the disconnection of the call. At this point, the new DID recall feature allows the external Central Office operator to reroute the call to the attendant console by sending the OPRS signal instead of IDLE. The system will detect the OPRS as a valid signal and the call will be presented to the attendant console on the RLL ICI key.

Operating parameters

Both DID Offering and DID Recall currently only support Type Approval in Italy and are not commercially available.

The QPC536 Digital Trunk Interface and NTAK10 (XDTI) cards are required.

This feature only works on DTI2 trunks.

Feature interactions

Basic Rate Interface (BRI) Special Call Forward Busy

This feature takes precedence over the DID offering; when the conditions for the BRI Special Call Forward Busy are met, the call is diverted to the attendant console without waiting for the OPRS signal. When the BRI Special Call Forward Busy feature fails or is not enabled, busy tone is returned to the Central Office and the DID offering can be activated.

Forward Busy

The DID offering is available only after the End of Selection Busy signal has been sent by the Central Office. This signal is provided to the Central Office trunk only if the busy set is configured with Forward Busy Denied (FBD) Class of Service.

Network Attendant Services (NAS)

Incoming DID calls which are Offered or Recalled to the attendant may receive NAS treatment. This feature requires no modification.

Feature packaging

Direct Inward Dialing (DID) Recall Features on DTI2 for Italy are included in the existing 2 Mbit Digital Trunk Interface (DTI2) package 129, which requires International Supplementary Features (SUPP) package 131.

Feature implementation

DID Offering

Task summary list

The following is a summary of the tasks in this section:

- Table 35: LDs 10/11 -Set the Class of Service to FBD. on page 138

  Set the Class of Service to FBD.
- Table 36: LD 16 - Set DID Recall for this Rate. on page 138

  Set DID Recall for this Rate.
- Table 37: LD 73 - Configure the SICA table for the DID Offering feature. on page 138

  Configure the SICA table for the DID Offering feature.
- Table 38: LD 16 - Set DID Recall to Attendant for this rate. on page 139

  Set DID Recall to Attendant for this rate.
- Table 39: LD 73 - Configure the SICA table for the DID Recall feature. on page 139

  Configure the SICA table for the DID Recall feature.

Table 35: LDs 10/11 -Set the Class of Service to FBD.

Prompt	Response	Description
REQ:	NEW CHG	New or change.
TYPE:	a...a	Telephone type. Type **?** for a list of possible responses.
TN		Terminal number
	l s c u	Format for Large System, Avaya CS 1000 Media Gateway 1000B (Avaya MG 1000B), and Avaya Communication Server 1000E (Avaya CS 1000E) system, where l = loop, s = shelf, c = card, u = unit.
CLS	FBD	Forward Busy Denied.

Table 36: LD 16 - Set DID Recall for this Rate.

Prompt	Response	Description
REQ	NEW CHG	New, or change.
TYPE	RDB	Route data block.
...		
RCAL	DRA	Set DID Recall to ATTN for this route.

Table 37: LD 73 - Configure the SICA table for the DID Offering feature.

Prompt	Response	Description
REQ	CHG	Change.
TYPE	DTI2	2 Mbit DTI.
FEAT	ABCD	Digital signaling category table.
SICA	nn	Signaling Category table number.
INCOMING/OUTGOING CALLS		
IDLE(R)	ABCD	(Receive) IDLE signal bits.
INCOMING CALLS		
P EOSB(S)	ABCD	End of Selection Busy (receive) signal.
- TIME	(100)-150	Duration of the EOSB(S) signal in milliseconds.
...		
P OPRS(R)	ABCD	Operator (receive) recall signal.
- TIME	xxxx yyyy	Time for OPRS(R) in milliseconds, where: xxxx = 8-(48)-2040, and yyyy = xxxx-(128)-2040.

DID Recall

Table 38: LD 16 - Set DID Recall to Attendant for this rate.

Prompt	Response	Description
REQ	NEW CHG	New, or change.
TYPE	RDB	Route Data Block.
...		
RCAL	DRA	Set DID Recall to attendant for this route.

Table 39: LD 73 - Configure the SICA table for the DID Recall feature.

Prompt	Response	Description
REQ	CHG	Change.
TYPE	DTI2	2 Mbit DTI.
FEAT	ABCD	Digital signaling category table.
SICA	nn	Sica table number.
...		
INCOMING CALLS		
CLRB(S)	ABCD	Clear Backward (send) signal.
...		
P OPRS(R)	ABCD	Operator (receive) recall signal.
- TIME	xxxx yyyy	Time for OPRS(R) in milliseconds, where: xxxx = 8-(48)-2040, and yyyy = xxxx-(128)-2040.

Feature operation

No specific operating procedures are required to use this feature.

Chapter 17: Direct Inward Dialing to TIE Connection

Contents

This section contains information on the following topics:

Feature description

This feature allows DID-to-TIE connections, subject to all trunk barring, Trunk Group Access Limitations (TGAR), Trunk Access Restriction Groups (TARG), and other Class of Service limitations. When the end-of-dialing timer detects that end-of-dialing is reached for an outgoing TIE trunk the Call Forward No Answer (CFNA) timer is started.

If the CFNA timer expires prior to detecting an answer signal the call is intercepted to the attendant. If a routed call receives a busy signal from an extension, the busy signal is returned to the DID. If the DID does not go on-hook before the CFNA recall timer expires, the call is routed to the attendant.

Operating parameters

The Central Office must be equipped to handle the special signaling requirements associated with the DID-to-TIE Connection feature described above.

The DID-to-TIE Connection feature is not available on 1.5 Mbps digital, Japanese DMI, PRI2 or DPNSS trunks.

Feature interactions

There are no feature interactions associated with this feature.

Feature packaging

International Supplementary Features (SUPP) package 131.

Feature implementation

Task summary list

The following is a summary of the tasks in this section:

1. Table 40: LD 15 - Allow DID-to-TIE connections. on page 142

   Allow DID-to-TIE connections.
2. Table 41: LD 16 - Define the Number of digits expected on DID rate. on page 143

   Define the Number of digits expected on DID rate.

Table 40: LD 15 - Allow DID-to-TIE connections.

Prompt	Response	Description
REQ:	NEW CHG	Add, or change.
TYPE:	NET	ISDN and ESN networking options.
...		
- DITI	(NO) YES	DID-to-TIE connections (are not) are allowed.

Table 41: LD 16 - Define the Number of digits expected on DID rate.

Prompt	Response	Description
...		
DNSZ	(0)-7	Number of digits expected on DID route. 0 indicates no fixed number.

Feature operation

No specific operating procedures are required to use this feature.

Chapter 18: Direct Inward System Access

Contents

This section contains information on the following topics:

Feature description

Direct Inward System Access (DISA) allows selected users to access the system from the public or private network by dialing a special Directory Number (DN) assigned by the customer. The number can be dialed from any Digitone telephone outside the network. Once the Direct Inward System Access (DISA) call has been answered, the user can access any of the following features and capabilities offered through Direct Inward System Access:

- Calls to any station within the customer group
- Trunk calls (such as calls to a Public Exchange/Central Office, a TIE trunk, or paging and dictation trunks)
- Basic/Network Authorization Code (BAUT/NAUT)
- Call Detail Recording (CDR) and Call Detail Recording Charge Account, and
- Basic/Network Alternate Route Selection (BARS/NARS) and Automatic Number Identification (ANI) route selection.

Each special Directory Number (DN) dialed by a DISA user is associated with a particular DISA Directory Number. Any number of DISA DNs can be assigned, provided that they are consistent with the numbering plan of the customer. Access rights are determined by the Class of Service and Trunk Group Access Limitations (TGAR) associated with the DISA number. Calls to DISA can be placed on dedicated, auto-terminate incoming trunks (Central Office [CO], Foreign

Exchange [FX], or Wide Area Telephone Service [WATS]) and TIE or Direct Inward Dialing (DID) trunks, all of which must have proper supervision.

As a safeguard against unauthorized use, an authorization code or special security code of one to eight digits can be assigned for each DISA DN. The security code must be entered before any system resources can be used. Additionally, a secure data password can be provided to enable the customer to create, modify, or remove information concerning DISA.

Operating parameters

The features not available to DISA users are those that require a switchhook flash (such as Call Transfer, Conference, Hold, or Ring Again). Also unavailable are features requiring that predefined data be assigned for the DN (for example, Speed Call), and other features that are not applicable to DISA calls (such as Call Pickup and Call Forward).

Any CO, FX, or WATS trunk route can be designated as an auto-terminate route, allowing incoming calls in the route to terminate on one particular DN rather than going to the attendant. Several trunks can specify the same DISA DN, or each trunk can specify a different DISA DN.

Only trunks that give disconnect supervision can be used to provide access to DISA. Therefore, trunks dedicated to DISA (CO, FX, or WATS) must have a ground start signaling arrangement. Incoming DISA calls on trunks without disconnect supervision will not be allowed. For these calls, overflow tone is given to TIE, DID, and Common Controlled Switching Arrangement (CCSA) trunk calls, and calls on CO, FX, and WATS trunks are intercepted to the attendant.

Trunks dedicated to DISA can also be used as normal outgoing trunks.

For Avaya Communication Server 1000E (Avaya CS 1000E) systems, analog trunks used for DISA must be provisioned on an Avaya Media Gateway 1000E (Avaya MG 1000E). . DISA must also be packaged on the CS 1000E for analog trunks and digital trunks.

DISA is not supported for High Scalability Survivable SIP Media Gateways.

Feature interactions

Access Limitations

Access limitations are assigned to the DISA DN as they are to any station within the system. Separate access limitations are also assigned to authorization codes used by DISA callers.

Attendant Busy Verify Busy Verify

Attendant Busy Verify applies only to DNs within the system. If an attendant tries to use the feature to enter a DISA DN, overflow tone is returned.

Basic/Network Alternate Route Selection (BARS/NARS)

The BARS/NARS features function on a DISA call as if it had been originated from inside the system.

Basic/Network Authorization Code (BAUT/NAUT)

This feature can be used in conjunction with DISA to allow a user access to more resources than are normally available. The Authorization Code must be entered, in addition to the security code (if required), using the applicable Special Prefix (SPRE) code followed by the authorization access code 6, or by an applicable Flexible Feature Code. If authorization codes are required, a valid Authorization Code must be entered after the DISA security code (no SPRE code is needed).

Call Detail Recording

If the customer and trunk route on which the incoming DISA call is being made have the applicable Call Detail Recording (CDR) options in effect, particulars of the call are recorded when it is established. There is no special indication on the CDR record that this was a DISA call. If the incoming trunk route is not specified for CDR options, recording depends on what has been specified by the customer for any outgoing trunks seized by the DISA caller.

Call Forward/Hunt Override Via Flexible Feature Code

DISA is not supported. Any attempt to dial the Call Forward/Hunt Override Via Flexible Feature Code will be ignored and access denied treatment will be returned.

Call Page Network Wide (PAGENET)

Paging trunk routes must have the response PGNU programmed for the NACC prompt in LD 16 (RDB) in order to allow DISA access from (Central Office [CO], Foreign Exchange [FX], Wide Area Telephone Service [WATS]) or Direct Inward Dialing (DID) trunks.

China Number 1 Signaling - Called Party Control

If an external station is allowed access to the trunk on which a Special Service resides using Direct Inward System Access (DISA), the station may also access that Special Service. However, Called Party Control is not supported.

Digital Private Network Signaling System (DPNSS1)/Digital Access Signaling System (DASS2) Uniform Dialing Plan (UDP) Interworking

DISA is not supported in a DPNSS1 UDP network.

Electronic Lock Network Wide/Electronic Lock on Private Lines

The Electronic Lock feature cannot be activated or deactivated when accessing the node through DISA.

Generic XFCOT Software Support

This feature allows selected external users to access the system switch by dialing a special directory number, and to use some features of the system as an internal station.

A Direct Inward System Access (DISA) call is allowed on a disconnect supervised or unsupervised loopstart trunk. If a caller on an unsupervised loopstart trunk disconnects during a DISA operation, it is detected by a dial time out or when the call is answered.

Caller disconnection during a DISA operation is detected by a disconnect-supervised loopstart trunk on an XFCOT card and the operation can then be ended.

ISDN QSIG/EuroISDN Call Completion

Call Completion on Busy Subscriber (CCBS) and Call Completion No Response (CCNR) are not supported on Direct Inward System Access (DISA) calls when the call destination is busy.

Line Lockout Flexible Line Lockout

The defined Flexible Line Lockout treatment is provided to DISA calls.

New Flexible Code Restriction

If the Direct Inward System Access (DISA) DN has a TLD, CUN, or CTD Class of Service, calls made through DISA are eligible for NFCR treatment.

Night Service Enhancements

It is not possible to assign a Night Service Group Number to any trunk that is a member of a route that is set to auto-terminate on a DISA DN.

Pretranslation

Direct Inward System Access calls are automatically assigned XLST 0.

Scheduled Access Limitations

Direct Inward System Access (DISA) numbers are not assigned to Scheduled Access Limitations (SAR) groups and therefore are not affected by SAR schedules.

DISA can be used to manually modify the SAR schedule, provided that the correct FFC and Authorization Code are dialed.

Feature packaging

Direct Inward System Access (DISA) is package 22 and has no other feature package dependencies.

Feature implementation

Task summary list

The following is a summary of the tasks in this section:

1. Table 42: LD 24 - Configure the Direct Inward System Access feature for a customer. on page 150

   Configure the Direct Inward System Access feature for a customer.
2. Table 43: LD 16 - Define an auto-terminate trunk route for Direct Inward System Access. on page 150

   Define an auto-terminate trunk route for Direct Inward System Access.
3. Table 44: LD 14 - Define Direct Inward System Access DNs for trunks in an auto-terminate trunk route. on page 151

   Define Direct Inward System Access DNs for trunks in an auto-terminate trunk route.

Table 42: LD 24 - Configure the Direct Inward System Access feature for a customer.

Prompt	Response	Description
REQ	NEW CHG	Add, or change.
TYPE	DIS	DISA data.
CUST	xx	Customer number, as defined in LD 15
SPWD	xxxx	System secure data password (0001-9999) allows modifications to the DISA data block. 0000 = disable the password (see LD 15).
DN	xxx...x	DN for DISA access.
SCOD	X xx...xx	DISA security code (1-8 digits). X = remove security code.
AUTR	(NO) YES	Authorization Code is not or is required.
TGAR	xx	Trunk Group Access Restriction to be applied to calls made using DISA (0-15). TGAR can be from 0 to 31.
NCOS	xx	Network Class of Service to be applied to DISA calls.
COS		Class of Service to be applied to DISA calls.
	UNR CUN SRE TLD CTD FRE FR1 FR2	Unrestricted. Conditionally unrestricted. Semi-restricted. Toll restricted. Conditionally toll restricted. Fully restricted. Fully restricted 1. Fully restricted 2

Table 43: LD 16 - Define an auto-terminate trunk route for Direct Inward System Access.

Prompt	Response	Description
REQ	NEW CHG	Add, or change.
TYPE	RDB	Route data block.

Prompt	Response	Description
CUST	xx	Customer number, as defined in LD 15
ROUT		Route number
	0-511	Range for Large System and CS 1000E system.
TKTP	aaa	Trunk type.
AUTO	(NO) YES	Route is not or is arranged to auto-terminate incoming calls on the DISA DN.
ICOG	IAO ICT OGT	Incoming and outgoing trunk.
ACOD	xxxx	Trunk route access code.

Table 44: LD 14 - Define Direct Inward System Access DNs for trunks in an auto-terminate trunk route.

Prompt	Response	Description
REQ	NEW CHG	Add, or change.
TYPE	COT FEX WAT	Trunk type.
TN		Terminal number
	l s c u	Format for Large System, Media Gateway 1000B, and CS 1000E system, where l = loop, s = shelf, c = card, u = unit.
XTRK	XUT	Universal trunk card (prompted for Superloops).
CUST	xx	Customer number, as defined in LD 15
RTMB		Route number and Member Number
	0-511 1-4000	Range for Large System and CS 1000E system.
ATDN	xxx...x	DISA DN on which incoming calls are to auto-terminate.
SIGL	GRD	Ground Start signaling.

Feature operation

To dial into the system from the public network:

1. Dial the DISA number. You hear a dial tone.
2. Dial the security code, if required.
3. Dial the Authorization Code, if required.

Chapter 19: Direct Inward System Access on Unsupervised Trunks

Contents

This section contains information on the following topics:

Feature description

With this enhancement, Direct Inward System Access (DISA) is allowed on Public Exchange/ Central Office (CO), FEX, and WATS trunks without disconnect supervision. Without the enhancement, DISA calls on these trunks are intercepted to the attendant. The Timed Forced Disconnect Timer is used to prevent the permanent seizure of the Central Office trunk in cases where the far-end goes on-hook first.

Operating parameters

There are no operating parameters associated with this feature.

Feature interactions

There are no feature interactions associated with this feature.

Feature packaging

Direct Inward System Access (DISA) package 22.

Feature implementation

Table 45: LD 16 - Configure the Timed Forced Disconnect Timer.

Prompt	Response	Description
...		
MFC	(NO) YES	Respond with YES to enable Multifrequency Compelled Signaling.
- TIMR	TFD (0)-3600	Timed Force Disconnect in 30-second increments.

In addition, make sure the DISA feature is configured as described in the DISA feature description contained within this document.

Feature operation

To dial into the system from the public network:

1. Dial the DISA number. You hear a dial tone.
2. Dial the security code, if required.
3. Dial the Authorization Code, if required.

Chapter 20: Direct Private Network Access

Contents

This section contains information on the following topics:

Feature description

The Direct Private Network Access feature provides enhancements to the processing of Direct Inward System Access (DISA) and Authcode Last Request calls. This feature complements system capabilities to provide an arrangement suitable for long distance resellers. Typically, subscribers to these resellers' services dial in through a DISA port and require some automated digit manipulation, recorded announcements and Authcodes for billing purposes. This feature offers the following capabilities:

DISA Digit Insertion

Once a DISA Directory Number (DN) is accessed, the system automatically inserts from 1 to 31 digits to save the caller from having to manually enter these digits. Dial tone is provided if the system expects to receive more digits from the caller in order to complete the call. If no additional digits are required, the call terminates automatically.

DISA Recorded Announcement (RAN)

A caller may be greeted with a Recorded Announcement once a DISA DN is accessed. The caller can begin dialing anytime during the greeting, in which case the greeting is stopped and the call is processed. If the Recorded Announcement finishes, dial tone is provided if more digits are expected from the caller to complete the call. As with the case of DISA Digit Insertion, the call terminates automatically if no additional digits are required.

Authcode Last Retry Request

For an Authcode Last Request call, if a caller enters an authorization code (Authcode) that is invalid, the caller is prompted to enter an Authcode again. The reprompt for the Authcode takes the form of either an Authcode Last Retry Request dial tone or a RAN before the Authcode Last Retry Request dial tone.

If configured, the RAN indicates to the caller that a wrong Authcode has been entered. While RAN is being given, all dialed digits are ignored.

If a caller realizes they have misdialed, an octothorpe (#) can be pressed which allows the user to immediately re-enter the Authcode. If an invalid Authcode is entered for a second time, the existing invalid Authcode treatment results.

Operating parameters

DISA Digit Insertion, DISA RAN, and Authcode Last Retry can be activated individually or can be combined to work in conjunction with one another.

DISA Digit Insertion and DISA RAN can be optionally assigned on a per DISA basis in LD 24, and are only applicable to DISA calls.

Authcode Last Retry can be optionally assigned on a per customer basis in LD 88, and is applicable to all call types supporting Authcode Last.

All existing DISA limitations apply to the DISA Digit Insertion and DISA RAN functionalities.

All existing RAN limitations apply to the DISA RAN and Authcode Last Retry functionalities.

All existing Authcode Last limitations apply to the Authcode Last Retry functionality.

To support DISA RAN and the Authcode Last Retry RAN function, the system must be equipped with all the necessary RAN hardware.

Feature interactions

Attendant Console Operation

Authcode Last Retry Not Configured

If an invalid Authcode is entered by an attendant, overflow tone is given as soon as a sufficient number of Authcode digits has been entered. If the attendant enters some digits for an Authcode that is less than the number of digits defined in LD 88, silence is heard.

Authcode Last Retry Configured

If the caller is an attendant and the Authcode entered is invalid, once a sufficient number of digits has been entered, the Authcode Last Request dial tone is immediately given to reprompt for the Authcode. If the attendant enters some digits for an Authcode that is less than the number of digits defined in LD 88, silence is heard. Since there is no interdigit time out for an attendant console, no Authcode Last Request dial tone will be given for retry.

Authcode Last Request tone will be heard immediately prompting for Authcode Retry if the attendant enters an octothorpe "#" followed by some digits.

Authorization Code Security Enhancement

Only when an Authcode retry fails will a Security Administration (SECA) message be printed to the configured MTC, FIL console and/or the configured History File.

Autodial

If Autodial is programmed with a valid Authcode for Authcode Last followed by an octothorpe "#", the existing Authcode Last operation will reject the Authcode as an invalid Authcode. If Authcode Last Retry is defined, the caller will be reprompted for the Authcode.

Call Detail Recording

Digits inserted by DISA Digit Insertion are reflected in the Call Detail Recording (CDR)record.

When a caller is reprompted for an Authcode due to Authcode Last Retry, and a new Authcode is entered, the second Authcode will overwrite the first entry. Therefore, the CDR record only reflects the last Authcode entered.

Pretranslation

Digits automatically inserted by DISA Digit Insertion are pretranslated during call processing in the same manner as if the caller had manually dialed the digits.

Speed Call

If a Speed Call entry is programmed with a valid Authcode for Authcode Last followed by an octothorpe "#", the existing Authcode Last operation will reject the Authcode as an invalid Authcode. If Authcode Last Retry is defined, the caller will be reprompted for the Authcode.

Feature packaging

This feature is packaged under Direct Private Network Access (DPNA) package 250.

DISA Digit Insertion requires the following additional package:

- Direct Inward System Access (DISA) package 22

DISA RAN requires the following additional packages:

- Direct Inward System Access (DISA) package 22
- Recorded Announcement (RAN) package 7

Authcode Last Retry requires the following additional packages:

- Basic Authorization Code (BAUT) package 25
- Network Authorization Code (NAUT) package 63
- Recorded Announcement (RAN) package 7 when an Authcode Last Retry RAN is required

Feature implementation

Task summary list

The following is a summary of the tasks in this section:

1. Table 46: LD 24 - Modify the Direct Inward System Access data block. on page 159

   Modify the Direct Inward System Access data block.
2. Table 47: LD 88 - Modify the authorization code data block. on page 160

   Modify the authorization code data block.

DISA DN Data

Configure RAN routes (LD 16) and RAN trunks (LD 14) as per existing procedures.

Table 46: LD 24 - Modify the Direct Inward System Access data block.

Prompt	Response	Description
REQ	NEW CHG	New, or change.
TYPE	DIS	DISA data.
CUST	xx	Customer number, as defined in LD 15
...		
RANR		RAN route number for "Authcode Last" prompt (NAUT)
	0-511	Range for Large System and Avaya Communication Server 1000E (Avaya CS 1000E) system.
- RTMR	(0)-10-300	The maximum amount of time (in seconds) that a caller can wait for an available RAN trunk before being removed from the RAN queue and proceeding as if DISA RAN has been completed.
	(0)	Removes and deactivates the timer.
DGTS	x...x	Digits for DISA Digit Insertion. Up to 31 digits can be defined.
	(X)	Removes and deactivates DISA Digit Insertion.

Prompt	Response	Description
- DLTN	(YES) NO	Dial tone needed after digit insertion. Dial tone not needed after digit insertion.

Authcode Data

Configure RAN routes (LD 16) and RAN trunks (LD 14) as per existing procedures.

Table 47: LD 88 - Modify the authorization code data block.

Prompt	Response	Description
REQ	NEW CHG	New, or change.
TYPE	AUB	Authcode data.
CUST	xx	Customer number, as defined in LD 15
...		
RANR	0-511	RAN Route number For Large Systems
RTRY	(NO) YES	Disable Authcode Last Retry. Enable Authcode Last Retry.
- RAN2		Route number for Authcode - last Retry RAN
	0-511	Range for Large System and CS 1000E system.
CLAS	xx	Class code value assigned to authcode.

Feature operation

Operational Sequence of a DISA Call

Step	User Action	Result
1.	Dials DISA DN.	If DISA Security Access Code is required, special dial tone is given, and the caller continues to Step 2. Otherwise the caller skips to Step 3.
2.	Enters the Security Access Code.	The dial tone is removed as soon as the first digit is dialed. If the security access code entered is valid, the caller continues to Step 3. Otherwise, the existing

Step	User Action	Result
		treatment for invalid Security Access code is given when the interdigit timer expires.
3.	<no user action>	If Authcode is required, normal dial tone is given, and the caller continues to Step 4. Otherwise, the caller skips to Step 5.
4.	Enters an Authcode.	The dial tone is removed as soon as the first digit is dialed. If the Authcode entered is valid, the caller continues to Step 5. Otherwise, the existing invalid Authcode treatment is given when the interdigit timers times out.
5.	<no user action>	If DISA Digit Insertion is not configured, the caller immediately continues to Step 6. Otherwise, the digits defined for DISA Digit Insertion are automatically inserted into the call register before the caller continues to Step 6.
6.	<no user action>	If DISA RAN is configured, a RAN greeting is provided, and the caller continues to Step 7. Otherwise, the caller skips to Step 8.
7.	a) The caller listens to the RAN greeting; or	a) If DISA Digit Insertion is not defined, or DISA Digit Insertion specifies to give dial tone to prompt the caller to enter more digits, the caller continues to Step 8. Otherwise, the inserted digits are immediately processed for call completion.
	b) begins dialing before the RAN is finished.	b) The RAN greeting is stopped as soon as the first digit is dialed. The dialed digits are appended into the call register (that is, if DISA Digit Insertion is defined, the dialed digits are stored after the inserted digits), and the call is processed for call completion.
8.	<no user action>	Dial tone is given and the caller continues to Step 9.
9.	Dials digits to originate the call.	Dial tone is removed as soon as the first digit is dialed. The dialed digits are appended into the call register (that is, if DISA Digit Insertion is defined, the dialed digits are stored after the inserted digits), and the call is processed for call completion.

Operational Sequence of Authcode Last

Step	User Action	Result
1.	Makes an outgoing call that requires Authcode Last.	Authcode Last Request dial tone is given. If Authcode Last RAN is defined, RAN precedes the dial tone. The caller continues to Step 2.

Step	User Action	Result
2.	Dials one of the following:	The Authcode Last Request dial tone is removed as soon as the first digit is dialed. Then depending on the digit input, one of the following occurs:
	a) A valid Authcode.	a) The call is processed for call termination.
	b) An invalid Authcode followed by "#".	b) If Authcode Last Retry is defined, Authcode Last Request dial tone is immediately given (if Authcode Last Retry RAN is defined RAN precedes the dial tone), and the caller continues to Step 3. If Authcode Last Retry is not defined, when the interdigit timer expires the existing invalid Authcode treatment is given.
	c) An invalid Authcode.	c) If Authcode Last Retry is defined: • If the caller is an attendant, Authcode Last Request dial tone is immediately given (if Authcode Last Retry RAN is defined RAN precedes the dial tone), and the caller continues to Step 3. • If the caller is not an attendant, when the interdigit timer expires Authcode Last Request dial tone is again given (if Authcode Last Retry RAN is defined RAN precedes the dial tone), and the caller continues to Step 3. If Authcode Last Retry is not defined, when the interdigit timer times out the existing invalid Authcode treatment is given.
3.	Dials one of the following:	The Authcode Last Request dial tone is removed as soon as the first digit is dialed. Then depending on the digit input, one of the following occurs:
	a) A valid Authcode	a) The call is processed for call termination.
	b) An invalid Authcode followed by "#".	b) When the interdigit timer times out, the existing invalid Authcode treatment is given.
	c) An invalid Authcode.	c) When the interdigit timer times out, the existing invalid Authcode treatment is given.

Example of a DPNA Call Using All Three Functions

In this example, User A calls from home to a DISA DN and subsequently to an ESN number as defined in the DISA Digit Insertion. When prompted for an Authcode, User A initially enters an invalid one, before being reprompted for the authcode (See Figure 1: DPNA call using all three functions on page 163).

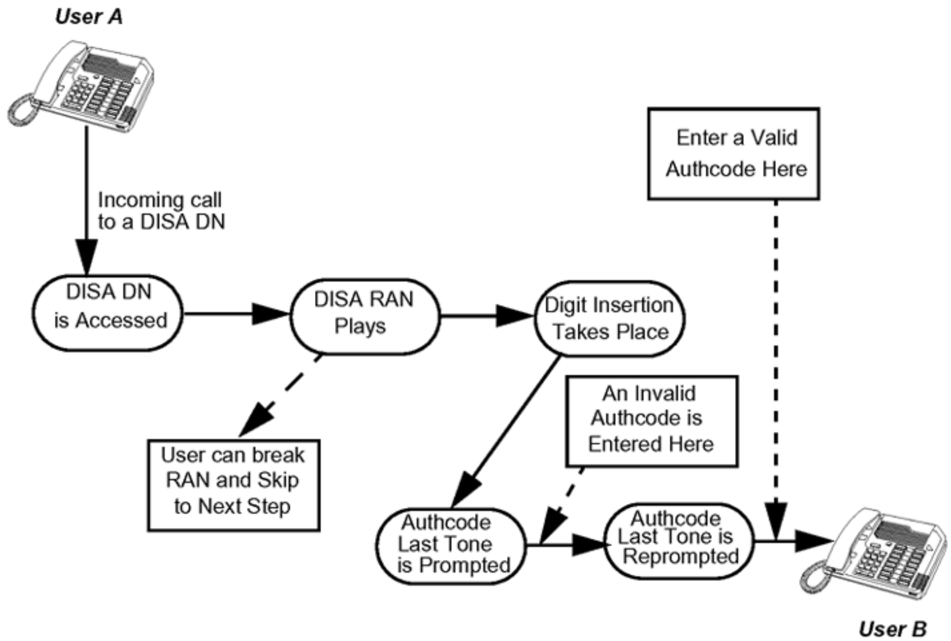

Figure 1: DPNA call using all three functions

Chapter 21: Directory Number

Refer to the following feature modules in this book for information on Directory Number:

- Directory Number Delayed Ringing on page 167
- Directory Number Expansion on page 173
- Flexible Attendant Directory Number on page 177

- Listed Directory Numbers
- Multiple Appearance Directory Number
- Prime Directory Number in *Avaya Features and Services Fundamentals, NN43001-106*, Book 5
- Single Appearance Directory Number in *Avaya Features and Services Fundamentals, NN43001-106*, Book 6

For Network-Wide Listed Directory Number, see *Avaya ISDN Primary Rate Interface Fundamentals, NN43001-569*.

Chapter 22: Directory Number Delayed Ringing

Contents

This section contains information on the following topics:

Feature description

There are two types of Directory Number keys: ringing and non-ringing. The Directory Number Delayed Ringing (DNDR) feature offers the ability to provide an audible notification (for example, ringing or buzzing) after a specified delay to non-ringing keys for a particular Terminal Number (TN). These keys can be either Single Call Non-Ringing (SCN) or Multiple Call Non-Ringing (MCN).

When an incoming call is presented to an SCN/MCN key, the associated lamp starts flashing. If Directory Number Delayed Ringing is defined for the set, an audible notification is given after a defined number of seconds (from 1 to 120 seconds). The DNDR value is defined in LD 11, and the feature is disabled if zero is selected as the delay value. When the feature is disabled, all Single Call Non-Ringing (SCN) or Multiple Call Non-Ringing (MCN) keys for this particular TN will not receive audible notifications.

Operating parameters

Only Meridian 1 proprietary telephones with DN key type SCN or MCN may use this feature; analog (500/2500 type) telephones are not supported.

When enabling the Directory Number Delayed Ringing feature and zero is entered as delay value, the desired Single Call Ringing or Multiple Call Ringing key must to changed to Single Call Non-Ringing (SCN) or Multiple Call Non-Ringing (MCN).

The DNDR feature is enabled on a TN basis. Thus, all or none of the SCN/MCN keys for the TN will receive the audible notification.

For a single call, two appearances of a Multiple Appearance Directory Number (MADN) may ring simultaneously if their DNDR values differ by two seconds or less.

The DNDR value can be different for multiple TNs with the same DN appearance; therefore, the audible notification may begin at different times for a single call.

Feature interactions

Any feature that works with non-ringing keys works with the DNDR feature whether or not the key is ringing.

Attendant Administration Automatic Wake-Up

Attendant Administration and the Automatic Wake-Up features are not supported.

Attendant Blocking of Directory Number

The Attendant Blocking of DN feature will override the Directory Number Delayed Ringing feature and ring the blocked DN immediately when the SACP key is pressed to ring the blocked DN.

Attendant Recall Automatic Timed Reminder Recalls

If a dialed set has DNDR defined, and an attendant re-extends a call without releasing it, the DNDR timing is not reset. If the value of the recall timer is less than that of the DNDR timer, the call is recalled to the attendant before audible notification begins.

Attendant Recall Enhancement

With this feature, when a call to a set is recalled to the attendant, the ringing is stopped on that set. If the attendant re-extends the call and ringing is applied again, the DNDR delay is also applied.

Buzzing

If a set is defined with DNDR delay and there is an incoming call to another SCN/MCN DN key on the same set, buzzing (or short buzzing) is applied after the DNDR delay timer expires.

Call Forward No Answer Call Forward No Answer, Second Level

The DNDR feature allows the SCN/MCN (non ringing keys) to actually ring after a definable period of time (DNDR prompt in LD 11). If the time before CFNA takes effect is less than the DNDR time for a particular set, CFNA will forward this call before any SCN/MCN keys can ring on this set. Note that CFNA is defined in the number of rings and DNDR is defined in seconds.

If the Forward DN set is busy or invalid when the call is forwarded, the call will return to the originally called set. However, the DNDR delay timer will be reapplied to the called set if DNDR is defined.

If a call is forwarded, as per existing operation, this call will be treated as a new incoming call to the forward DN. For example, if the forward DN has a DNDR value defined, a new timer will begin timing according to the forward DN's DNDR delay.

Call Waiting

Call Waiting tones apply to SCN/MCN keys as per existing operation. The DNDR delay does not apply, and the user is informed of the incoming call immediately.

Data Calls Private Line Ringing (PVN) Private Line Non-Ringing Set-Based Administration Enhancements

These features are not supported by the Directory Number Delayed Ringing feature.

Distinctive/New Distinctive Ringing

The DNDR feature applies to the Distinctive Ringing feature; what applies to normal ringing with DNDR also applies to distinctive ringing.

Flexible Incoming Tones

If DNDR is enabled, the Flexible Incoming Tones buzz is delayed as with any type of audible notification.

Group Call

When a group call is made to an SCN/MCN key with Directory Number Delayed Ringing (DNDR) defined, audible notification will be given after the DNDR delay has expired.

Ringing Change Key

If an SCR/MCR key is toggled from "ringing" to "non-ringing", the DNDR feature will apply to the key. If an SCR/MCR key is toggled again from "non-ringing" to "ringing", the key will be rung immediately and DNDR will no longer apply.

If an SCN/MCN key is toggled from "non-ringing" to "ringing", the DNDR key will ring immediately and DNDR will no longer apply. If an SCN/MCN is toggled again from "ringing" to "non-ringing", the key will not ring immediately and the DNDR feature will apply to the key.

Short Buzz for Digital Telephones

If a set is defined with DNDR delay and there is an incoming call to another SCN/MCN DN key on the same set, buzzing (or short buzzing) is applied after the DNDR delay timer expires.

Spanish KD3 Forced Disconnect

Spanish KD3 Digital Trunk Signaling Direct Inward Dialing (DID) disconnects an incoming call if the destination does not answer in 60 seconds. If the DNDR delay is set to a value of more than 60 seconds, the KD3 DID will terminate the call and the destination never receives the audible notification.

User Selectable Call Redirection

With User Selectable Call Redirection (USCR) a user can change the number of CFNA/DFNA ringing cycles. If the user changes the CFNA/DFNA value so that CFNA takes place before the DNDR timer runs out, none of the SCN/MCN keys will receive an audible notification. See the interaction with Call Forward No Answer.

Feature packaging

This feature is included in base system software.

Feature implementation

Table 48: LD 11 - Configure the delay value (in seconds).

Prompt	Response	Description
REQ	NEW CHG	Add, or change.
TYPE	a...a	Telephone type. Type **?** for a list of possible responses.
TN		Terminal number
	l s c u	Format for Large System, Avaya CS 1000 Media Gateway 1000B (Avaya MG 1000B), and Avaya Communication Server 1000E (Avaya CS 1000E) system, where l = loop, s = shelf, c = card, u = unit.
...		
DNDR	(0)-120	Delay value in seconds. A DNDR value of zero disables the feature. If the DNDR value is an odd number, it is rounded up to the next even number. If REQ = NEW, the delay value is 0 (the default); otherwise the existing value appears.
...		
KEY	xx SCN yyyy xx MCN yyyy	Key number, Single Call Non-Ringing, DN. The key must be SCN or MCN.

Feature operation

No specific operating procedures are required to use this feature.

Chapter 23: Directory Number Expansion

Contents

This section contains information on the following topics:

Feature description

This feature increases the number of digits allowed for internal Directory Numbers (DNs), from a maximum of four digits per DN to seven digits per DN. The following internal DNs have been expanded:

- Single-line telephone DNs
- Multi-line telephone DNs
- Trunk Group Access codes
- Attendant DNs (including local attendant in Centralized Attendant Service)
- Listed Directory Numbers (LDNs)
- Coordinated Dialing Plan (CDP) steering codes
- Automatic Call Distribution (ACD) DNs
- ACD position IDs
- Direct Inward System Access (DISA) DNs
- Centralized Attendant Service (CAS) hold DNs

- Release Link Trunk (RLT) DNs in Centralized Attendant Service)
- System Park DNs
- Test line DNs, and
- Data service DNs.

The following DN types are not expanded:

- Special Prefix (SPRE)
- Basic/Network Alternate Route Selection (BARS/NARS) access codes
- Route Selection Automatic Number Identification (RSANI) access code, and
- Automatic Modem Pooling (AMP) all-digital-connection prefix.

Along with Directory Number Expansion (DNXP), Call Detail Recording Expansion (CDRE) package 151 is available to allow Call Detail Recording (CDR) records to accommodate the increased digit field lengths. Call Detail Recording (CDR) package 4 and Directory Number Expansion (DNXP) package 150 are required for CDRE.

Operating parameters

The number of DNs that can be configured is limited by the available protected data store in the system.

DNXP does not enhance existing feature capability other than allowing an internal DN with up to seven digits.

If DNXP is equipped, the system communicates with any attached Auxiliary Processor (AUX), except ACD-D, in a new message format containing expanded DN fields. Therefore, the respective Auxiliary Processor (AUX) software must be upgraded to handle longer DNs in new messages.

If a message is sent to an Auxiliary Processor (AUX) that is not capable of handling expanded DNs, only the last four digits are included in the message.

Incoming Digit Conversion (IDC) translates a maximum of four digits only.

The Automatic Number Identification (ANI) calling number is always seven digits long. It is obtained by combining the Automatic Number Identification Listed Directory Number (ANI LDN) with one of the following:

- DN of the analog (500/2500 type) telephone
- Prime DN of the digital telephone

- Automatic Number Identification (ANI) attendant number, specified on a per customer basis, and
- Automatic Number Identification (ANI) trunk number, specified on a per trunk group basis.

With the DNXP package equipped, if an Automatic Number Identification Listed Directory Number (ANI LDN) is not defined, then the full seven digits of an internal DN can be used as the ANI calling number. If an ANI LDN is defined and internal DNs are longer than four digits, only the leading digits of the DNs are retained in the ANI calling number.

CDRE must be equipped to allow the printing of seven-digit DNs in the CDR records.

An Automatic Identification of Outward Dialing (AIOD) station identification number remains four digits long. If a DN is longer, only the leading digits are retained as the Automatic Identification of Outward Dialing (AIOD) station identifier.

Service-change and print overlays with DN-related prompts and commands have been modified to accommodate seven-digit DNs if the DNXP package is equipped.

Feature interactions

ACD-C Reports

When the DNXP package is equipped, each DN-related field is expanded to seven digits.

ACD Load Management

ACD Load Management commands have been modified to allow longer DN-related fields (ACD DN, position ID, route access code).

Automatic Identification of Outward Dialing

The Automatic Identification of Outward Dialing (AIOD) station identifier and trunk identifier remains four digits long. If the total number of digits in the AIOD prefix and internal DN exceeds four, only the leading digits of the station DN are retained as the AIOD identifier.

Automatic Number Identification

If the DN Expansion package is equipped, the Automatic Number Identification billing number (ANAT) can have up to seven digits. The total number of digits for ANAT and Automatic Number Identification listed DN (ANLD) cannot exceed seven.

Auxiliary processors

Any AUX or application processor that shares or exchanges system internal DN-related information with the system must be modified to handle the longer DN format. Otherwise, only the four trailing digits will be included in the message.

The presence of DNXP has an impact on the following types of AUX:

- Auxiliary Processor Link (APL)
- Application Module Link (AML)
- Standard Serial Data Interface (SDI) with application interface to the system, and
- Standard SDI without application interface to the system.

Background Terminal Interface

When the DNXP package is equipped, any background terminal command, response, or display containing a DN is allowed to have a DN of up to seven digits.

Coordinated Dialing Plan

Coordinated Dialing Plan (CDP) steering codes are expanded to a maximum of seven digits. The maximum number of digits for a complete CDP DN has increased from seven to ten (a three-digit steering code followed by a seven-digit internal DN).

With DNXP, the maximum number of leading digits to be deleted from a Local Steering Code (LSC) is expanded to seven digits, due to longer CDP numbers.

Digit and Name Display

If longer DNs are defined, the left most digits may be scrolled out on a digit display, depending on the size of the display window.

Direct Inward Dialing

Depending on the number of Direct Inward Dialing (DID) digits outpulsed by the Public Exchange/Central Office (CO), the system can insert a unique string of prefix digits to the incoming Direct Inward Dialing (DID) digits on a per DID trunk group basis to form a final internal DN. The number of digits that can be inserted for a DID (or TIE) trunk group has been expanded from six to eight digits.

Do Not Disturb

If the Directory Number Expansion (DNXP) package is equipped, DNs can have up to seven digits.

Electronic Switched Network

With DNXP, a seven-digit Location Code (LOC) call to an Electronic Switched Network (ESN) switch can be terminated to an internal DN of up to seven digits. A Digit Manipulation Index associated with a Home Location Code is used to properly terminate the calls.

Flexible Attendant Directory Number

The attendant DN can have up to seven digits if the Directory Number Expansion (DNXP) package is equipped.

Integrated Services Digital Network

See *Avaya ISDN Primary Rate Interface Fundamentals, NN43001-569*.

Night Service

If the Directory Number Expansion (DNPX) package is equipped, the Night DNs can be up to seven digits; otherwise, the DN can be a maximum of four digits.

Single Appearance Directory Number

The DN can have up to seven digits if the Directory Number Expansion package is equipped.

Feature packaging

Directory Number Expansion (DNXP) package 150 has no other feature package dependencies.

Feature implementation

There are no specific implementation procedures for this feature.

Feature operation

No specific operating procedures are required to use this feature.

Chapter 24: Distinctive/New Distinctive Ringing

Contents

This section contains information on the following topics:

Feature description

In commercial applications, the ability to have telephones with a distinctive ring is useful for distinguishing various call types. The Distinctive Ringing capability is enabled for specific trunk groups.

The Tone and Digit Switch (TDS) card provides Meridian 1 proprietary telephones with distinctive ringing cadence. This card provides a distinctive ringback tone of 440 Hz + 480 Hz on incoming calls on the designated trunks, timed for 1.64 on and 0.36 off. On single-line telephones, the normal ringing pattern is 2 on and 4 off. Distinctive Ringing for single-line telephones is 1.54 on and 0.38 off.

New Distinctive Ringing

This feature provides a new ringing cadence of 0.512 on and 0.512 off, followed by 1.024 on and 4.096 off, for all telephone types.

Distinctive Ringing for Dial Intercom

This feature allows a user to differentiate between an incoming call and a Dial Intercom call. The Dial Intercom ringing has a different cadence than regular Directory Number (DN) ringing or Distinctive Ringing.

Distinctive Ringing for Dial Intercom is assignable on a per-customer basis. The cadence is 0.5 on and 0.5 off, repeatedly.

Operating parameters

Distinctive Ringing requires 2.5 times as much "on" ringing time as routine ringing. The number of simultaneously ringing lines per ringing generator is reduced according to the proportion of incoming calls that receive Distinctive Ringing. For example, if 50 percent of all calls receive Distinctive Ringing, the number of simultaneous ringing lines is reduced from 20 to 14 per ringing generator.

Feature interactions

Attendant calls

When an incoming trunk call is extended by an attendant, the terminating extension rings distinctively.

Call Forward Busy

Calls modified by Call Forward Busy are not given Distinctive Ringing as they terminate on the attendant console.

Call Forward No Answer, Second Level

The ringing cadence for all telephones in a chain of call redirections remains the same as for the original DN called.

Call Waiting Redirection

The existing Distinctive Ringing Call Forward No Answer feature is applied to calls from a Distinctive Ringing enabled trunk. If such an incoming call is receiving Call Waiting treatment on sets with Distinctive Ringing, Call Forward No Answer (CFNA), and the Call Waiting Redirection feature enabled, the DFNA timer is applied to the call instead of the CFNA timer. The Call Waiting warning tone, if enabled, is not changed by Distinctive Ringing. If that call is not answered before the expiration of the DFNA timer, CFNA treatment is given using the Call Waiting Redirection feature.

Directory Number Delayed Ringing

The Directory Number Delayed Ringing (DNDR) feature applies to the Distinctive Ringing feature; what applies to normal ringing with DNDR also applies to distinctive ringing.

Flexible Tones and Cadences

With the Flexible Tones and Cadences package equipped, the Call Park Recall Ring Cadence (RBCS) specified in LD 56 has precedence over the Distinctive or New Distinctive Ringing given for Call Park recall.

ISDN Semi Permanent Connections for Australia

For ISDN Semi Permanent Connections for Australia (ISPC) calls, Distinctive/New Distinctive Ringing is provided according to the configuration of the route associated to the phantom trunk TN. This configuration is independent of the route associated to the real TN.

Night Service

Incoming calls terminating on a night Directory Number (DN) ring distinctively.

Telephones

The Meridian digital telephone Distinctive Ringing (defined by the Class of Service in LD 11) specifies the frequency and the warble-tone rate, and does not pertain to the Distinctive Ringing feature as referred to in this feature description.

For example, suppose New Distinctive Ringing is enabled and a call comes in from a Distinctive Ringing enabled trunk. If the call terminates on a Meridian digital telephone with DR2 Class of Service, it rings with DR2 (frequency and warble tone), but with a cadence of 0.512 on and 0.512 off, followed by 1.024 on and 4.096 off.

Telephone features

Calls modified by the following features receive Distinctive or New Distinctive Ringing:

- Call Forward All Calls
- Call Forward No Answer
- Flexible Call Forward No Answer
- Call Park
- Call Transfer
- Conference
- Hunting

User Selectable Call Redirection

The single parameter previously used to define distinctive ringing cycles (DFNA) is expanded to three (DFN0-2), with the Ringing Cycle Options (RCO) parameter used to select the specific DFNA entry for each telephone.

Virtual Network Services

An incoming call using VNS on a Bearer trunk defined with the prompt DRNG = YES will ignore this value and will perform the treatment as if the value of this prompt was DRNG = NO.

Feature packaging

Distinctive/New Distinctive Ringing (DNRG) package 74 has no feature package dependencies.

Distinctive Ringing for Dial Intercom is included in Dial Intercom (DI) package 21.

Distinctive Ringing for digital telephones is included in Digital Telephones (DSET) package 88.

Feature implementation

Task summary list

The following is a summary of the tasks in this section:

1. Table 49: LD 15 - Enable or disable Distinctive Ringing for Dial Intercom calls and specify Call Forward No Answer timing for trunks with Distinctive Ringing. on page 183

   Enable or disable Distinctive Ringing for Dial Intercom calls and specify Call Forward No Answer timing for trunks with Distinctive Ringing.
2. Table 50: LD 17 - Specify Distinctive or New Distinctive Ringing. on page 184

   Specify Distinctive or New Distinctive Ringing.
3. Table 51: LD 16 - Enable or disable Distinctive Ringing for each incoming or incoming/outgoing trunk route. on page 184

   Enable or disable Distinctive Ringing for each incoming or incoming/outgoing trunk route.
4. Table 52: LD 11 - Specify Distinctive/New Distinctive Ringing Class of Service for Meridian 1 proprietary telephones. on page 184

   Specify Distinctive/New Distinctive Ringing Class of Service for Meridian 1 proprietary telephones.

Table 49: LD 15 - Enable or disable Distinctive Ringing for Dial Intercom calls and specify Call Forward No Answer timing for trunks with Distinctive Ringing.

Prompt	Response	Description
REQ:	CHG	Change.
TYPE:	FTR	Features and options.
CUST		Customer number
	0-99	Range for Large System and Avaya Communication Server 1000E (Avaya CS 1000E) system.
- IRNG	(NO) YES	(Disable) enable Distinctive Ringing for Dial Intercom calls.
DFNA	1-(4)-15	The number of distinctive ringing cycles before Call Forward No Answer is activated for calls with Distinctive Ringing (the default is 4).

Table 50: LD 17 - Specify Distinctive or New Distinctive Ringing.

Prompt	Response	Description
REQ	CHG	Change.
TYPE	CFN PARM	Configuration Record. System parameters.
PARM	(NO) YES	Change system parameters.
- NDRG	(NO) YES	(Disable) enable New Distinctive Ringing (DRNG). Prompted only if DRNG is equipped.

Table 51: LD 16 - Enable or disable Distinctive Ringing for each incoming or incoming/ outgoing trunk route.

Prompt	Response	Description
REQ	CHG	Change.
TYPE	RDB	Route Data Block.
CUST	xx	Customer number, as defined in LD 15
ROUT		Route number
	0-511	Range for Large System and CS 1000E system.
DRNG	(NO) YES	(Disable) enable Distinctive Ringing for incoming calls.

Table 52: LD 11 - Specify Distinctive/New Distinctive Ringing Class of Service for Meridian 1 proprietary telephones.

Prompt	Response	Description
REQ:	CHG	Change.
TYPE:	a...a	Telephone type. Type **?** for a list of possible responses.
TN		Terminal number
	l s c u	Format for Large System, MG 1000B, and CS 1000E system, where l = loop, s = shelf, c = card, u = unit.
CLS	DRGX	Distinctive ring type (DRG1), DRG2, DRG3, DRG4, where: DRG1 = high fast tone, frequency 667/500 Hz. DRG2 = high slow tone, frequency 667/500 Hz. DRG3 = low fast tone, frequency 250/333 Hz. DRG4 = low slow tone, frequency 250/333 Hz. The DRG3/4 distinctive ringing for M2006 and M2008 telephones are different: DRG3 = low fast tone, frequency 1600/2000 Hz. DRG4 = low slow tone, frequency 1600/2000 Hz.

Feature operation

No specific operating procedures are required to use this feature.

Chapter 25: Distinctive Ringing by DN

Contents

This section contains information on the following topics:

Feature description

Distinctive Ringing by DN (DRDN) allows a distinctive ringing cadence to be configured for each DN key. The ability to have sets with a distinctive ring is useful for distinguishing calls with different DNs and is only available on Meridian Modular sets.

Distinctive ringing is an enhancement to the existing Executive Distinctive Ringing (EDRG) feature. This existing feature supports a distinctive ringing cadence when a call is made from an executive set. The Distinctive Ringing by DN feature enhances the EDRG feature by introducing two new functionalities.

The EDRG feature is determined by Class of Service as executive and it will ring distinctively. The existing functionality of EDRG is modified to allow the ringing cadence to be defined on a DN key basis rather than a TN basis.

A sub prompt for every DN key configures distinctive ringing index for incoming and outgoing calls. There are two available features for incoming and outgoing calls:

- Distinctive Ringing by call source, per DN-key: The distinctive ringing given to the called set is determined by the call source (calling set). This functionality is the same as the EDRG feature, except it is DN-key based rather than set based
- Distinctive Ringing by call destination, per DN-key: The distinctive ringing given to the called set is determined by the call destination (called set) and is also based on the DN-key of the called set.

With these enhancements, a DN-key can be configured to give a distinctive ring to the terminating set, and receive a distinctive ring for incoming calls.

Operating parameters

The precedence order for the different distinctive ringing cadences to ring the terminating set in a call is:

- Distinctive Ringing for an Incoming trunk call
- Distinctive Ringing by DN by call source
- Executive Ringing by DN call destination
- Distinctive Ringing by DN by call destination

The Private Line Ringing (PVR)/ Non-Ringing (PVN) keys are not supported by the DRDN features.

No DRDN functionality is supported on the Voice Call (VCC) keys since no DN is assigned to a VCC key.

A total of five distinctive ringing cadences used by DRDN are supported. Therefore a set with more than five DNs will have at least two DN-keys with the same distinctive ringing cadences.

The functionality of DRDN is limited to the following DN-keys; otherwise, normal ringing is given.

- Single Call Ringing (SCR)
- Single Call Non-ringing (SCN)
- One-way HOTLine (HOT)
- Two-way HOTLine
- Conference Hotline (CH)

The following Meridian sets can support DRDN:

- M2006
- M2008
- M2008HF
- M2616
- M2016
- M2216
- M2317

Feature interactions

Attendant Extended Call

A call from a set with DRDN extended from the attendant to the called set rings distinctively with the DNRO ringing cadence as configured on the originating set. It the attendant set is not configured for DRDN and the called set is equipped with DRDN then the called set rings with the DNRI ringing cadence as configured on the called set. If DRDN is not configured, normal ringing is given.

Call Forward All Calls

The forwarded call rings distinctively the called set if the originating set is configured with DRDN. If DRDN is not configured on the originating set then the called set rings distinctively, otherwise normal ringing is given.

Call Forward No Answer, Second Level

The ringing cadence for all telephones in a chain of call redirections remains the same as for the original DN called. When CFNA is activated for a set, distinctive ringing is given to the called set if the originator set is configured with DRDN, otherwise normal ringing is given.

Call Transfer

The ringing of the redirected call is determined by the set that has originated the call and not by the set transferring the call. The transferred call distinctively rings the called set if the originating set is configured with DRDN. If the originating set is not configured with DRDN then the ringing of the transferred call is determined by the called set.

Conference

The conference call is either scanned for a call marked as distinctive or a set designed as an executive set. The conferee with the highest index determines the ringing for the new call. The index of the conferees across the network checks if the network supports NAS supplementary messaging.

Dial Intercom Call

A Dial Intercom call is distinguished from a normal call since it has a different cadence configured in the FTC table. Dial Intercom takes precedence over the existing EDRG feature.

Distinctive Ringing

Existing Distinctive Ringing by DN (defined by the Class of Service in LD 11) specifies the frequency and the tone rate where the DRDN features supports the cadences.

Distinctive Ringing by an Incoming Trunk Call

All calling sets marked as distinctive rings the called set with a distinctive ring. The distinctive ring is determined by the index configured for the calling set. This takes precedence over DRDN.

Group Call

Distinctive ringing takes priority over the ringing cadence selected by the DRDN feature.

Hunting

Hunting occurs when the called set is busy. If the originating set is configured with DRDN the called set rings distinctively. A called set on a network call will ring distinctively with the cadence determined by the ringing index received across the network.

Enhanced Hotline

Enhanced Hotline DN-keys are required to support the functionality of the DRDN feature. A call made from Hotline DN-keys rings the called set with the index as configured for DNRO of the key. An incoming call to the HOT key rings the set with the index configured for DNRI.

Flexible Tones and Cadences

With the Flexible Tones and Cadences package 125 equipped, the Call Park Recall Ring Cadence (RBCS) specified in LD 56 has precedence over the Distinctive feature and Distinctive Ringing by DN given for Call Park recall.

Multiple Appearance DN

Distinctive Ringing by DN does not support Multiple Appearance DNs. Therefore, each appearance of a DN configured on a different set cannot be configured to allow different ringing cadences.

Night Service

Incoming calls terminating on a night Directory Number (DN) that has been set up with DRDN ring distinctively. If DRDN is not configured on the calling set, the night DN rings distinctively, otherwise normal ringing is given.

Feature packaging

The following packages are required for Distinctive Ringing by DN:

- Distinctive Ringing (DRNG) package 74
- Flexible Tones and Cadences (FTC) package 125
- Executive Distinctive Ringing (EDRG) package 185

Network Distinctive Ringing (NDRG) for feature functionality over the ISDN requires:

- Distinctive Ringing (DRNG) package 74
- Integrated Service Digital Network (ISDN) package 145
- Integrated Service Digital Network International (ISDN_INTL_SUP) package 161

Feature implementation

Task summary list

The following is a summary of the tasks in this section:

1. Table 53: LD 56 - Define the ringing cadence for analog (500/2500 type) sets, and the network and distinctive ringing tone for proprietary sets. on page 192

   Define the ringing cadence for analog (500/2500 type) sets, and the network and distinctive ringing tone for proprietary sets.

2. Table 54: LD 11 - Define the distinctive ringing cadence/tone to be used for Meridian 1 proprietary telephones and define Class of Service. on page 193

   Define the distinctive ringing cadence/tone to be used for Meridian 1 proprietary telephones and define Class of Service.

Table 53: LD 56 - Define the ringing cadence for analog (500/2500 type) sets, and the network and distinctive ringing tone for proprietary sets.

Prompt	Response	Description
REQ	CHG	Change existing data.
TYPE	FTC	Flexible Tones and Cadence table.
TABL	0-31	FTC table number.
RING	YES	Tones and cadences for ringing.
...	...	
NDR1 PBX	0-(2)-15	Network Distinctive Ring 1 cadence for analog (500/2500 type) sets.
NDR1 BCS		Network Distinctive Ring 1 cadence for proprietary sets.
- XTON	0-(2)-15	NT8D17 TDS Tone code.
- XCAD	0-(2)-15	NT8D17 TDS Cadence code.
NDR2 PBX	0-(2)-15	Network Distinctive Ring 2 cadence for analog (500/2500 type) sets.
NDR2 BCS		Network Distinctive Ring 2 cadence for proprietary sets.
- XTON	0-(2)-15	NT8D17 TDS Tone code.
- XCAD	0-(2)-15	NT8D17 TDS Cadence code.

Prompt	Response	Description
NDR3 PBX	0-(2)-15	Network Distinctive Ring 3 cadence for analog (500/2500 type) sets.
NDR3 BCS		Network Distinctive Ring 3 cadence for proprietary sets.
- XTON	0-(2)-15	NT8D17 TDS Tone code.
- XCAD	0-(2)-15	NT8D17 TDS Cadence code.
NDR4 PBX	0-(2)-15	Network Distinctive Ring 4 cadence for analog (500/2500 type) sets.
NDR4 BCS		Network Distinctive Ring 4 cadence for proprietary sets.
- XTON	0-(2)-15	NT8D17 TDS Tone code.
- CAD	7	NT8D17 TDS Cadence code.

Table 54: LD 11 - Define the distinctive ringing cadence/tone to be used for Meridian 1 proprietary telephones and define Class of Service.

Prompt	Response	Description
REQ:	NEW CHG	Add new data. Change existing data.
TYPE:	a...a	Telephone type. Type **?** for a list of possible responses.
TN		Terminal number
	l s c u	Format for Large System, Avaya CS 1000 Media Gateway 1000B (Avaya MG 1000B), and Avaya Communication Server 1000E (Avaya CS 1000E) system, where l = loop, s = shelf, c = card, u = unit.
DES	d...d	Office Data Administration System (ODAS) Station Designator of 1-6 alphanumeric characters.
CUST	xx	Customer number, as defined in LD 15
...		
CLS	DRDA	Distinctive Ringing by DN enabled. (DRDD) is the default.
...	...	
KEY	xx aaa yyyy	Telephone function key assignments for this feature, where: • xx = key number. • aaa = key type for this feature. These key types include: HOT D (one way and two way hotline), MCR, MCN, SCR, SCN and CH D. • yyyy = Directory Number. The maximum number of distinctive ringing cadences is five. Therefore, a set configured with more than five DNs,

Prompt	Response	Description
		say six, can provide distinctive ringing for five of the six DNs. Any call originating from other than the above mentioned keys gives the default ring to the terminating sets.
- MARP	NO	Multiple Appearance DN Redirection Prime.
- DNRO	(0)-4	Distinctive Number Ringing index for outgoing calls.
- DNRI	(0)-4	Distinctive Number Ringing index for incoming calls.

Feature operation

No specific operating procedures are required to use this feature.

Chapter 26: Do Not Disturb

Contents

This section contains information on the following topics:

Feature description

Individual Do Not Disturb (DNDI) allows the attendant to place a particular Directory Number (DN) in Do Not Disturb (DND) mode. A DN in this mode is free to originate calls, but appears busy to incoming calls. An attendant dialing a Directory Number in Do Not Disturb mode receives a visual indication and can override it temporarily by using Busy Verify (BVR) and signal source. To activate Individual Do Not Disturb (DNDI), a separate Individual Do Not Disturb (DNDI) key/lamp pair must be assigned to each applicable attendant console.

Analog (500/2500 type) telephones can be equipped with a Do Not Disturb lamp. Common Control Switching Arrangement (CCSA) and LPA Class of Service must be allowed.

Calls will receive the customer-specified intercept treatment (for example, busy tone, Recorded Announcement (RAN), or attendant). An enhancement to DND provides the ability to route calls to the Hunt DN instead of to the intercept treatment. Table 55: Do Not Disturb intercept treatments on page 196 lists possible intercept treatments based on responses to the prompts Do Not Disturb Intercept Treatment (DNDT) and Do Not Disturb Hunt (DNDH) in LD 15.

Table 55: Do Not Disturb intercept treatments

Call type	Hunt	DNDT = BST		DNDT = RAN		DNDT = ATT	
		DNDH No	DNDH Yes	DNDH No	DNDH Yes	DNDH No	DNDH Yes
DID							
Analog (500/ 2500 type) telephone	Allow	H	H	R	H	H	H
	Deny	A	A	R	R	A	A
Meridian 1 proprietary telephone	Allow	A	H	R	H	A	H
	Deny	A	A	R	R	A	A
Attendant							
Analog (500/ 2500 type) telephone	Allow	H	H	B	H	H	H
	Deny	B	B	B	B	B	B
Meridian 1 proprietary telephone	Allow	B	H	B	H	B	H
	Deny	B	B	B	B	B	B
Internal							
Analog (500/ 2500 type) telephone	Allow	H	H	R	H	H	H
	Deny	B	B	R	R	A	A
Meridian 1 proprietary telephone	Allow	B	H	R	H	A	H
	Deny	B	B	R	R	A	A
H = Follow Hunt Directory Number (DN) A = Intercept to attendant B = Busy tone R = RAN treatment							

Group Do Not Disturb (DNDG) allows an attendant to place predefined groups of DNs in DND mode. A DN can belong to many DND groups.

If a DN belongs to more than one DND group, the DND status of the DN might not be consistent with the DND status of each group. For example, if one of the DN's groups is removed from

DND mode, the DN is also removed from DND mode even if another group to which the DN belongs is still in DND mode.

To enable Group Do Not Disturb (DNDG), the DNDI package must be equipped. DNDI allows the user to activate, cancel, and verify the presence of the feature. A separate Group Do Not Disturb (DNDG) key is assigned to each attendant console for activating the DNDG feature.

Operating parameters

A maximum of 100 groups (0-99) can be defined per customer. Each group can contain up to 127 DNs.

A maximum of 20 DNDG keys can be equipped on an M2250 attendant console. Alternatively, the DNDI key plus dial-access can be used to activate DND for up to 100 groups.

To activate DNDG using a DNDG key, a group of telephones must be defined for that key (see LD 26).

For Individual Do Not Disturb (DNDI), a Direct Inward Dial (DID) call to a DN with DND active goes to the attendant if DNDT in LD 15 is set to BST or ATT. If the attendant is in Night Service, DID calls go to the night DN, if one is specified.

For Group Do Not Disturb (DNDG), if a DN is busy or has DND active, a DID caller gets a busy tone. If DNDT in LD 15 is set to CDB or RAN, and a DN is busy or has DND active, the DID caller gets RAN and then goes to the attendant.

Feature interactions

Attendant Alternative Answering

A DN in the DND mode is free to originate calls but appears busy to incoming calls. Call Forward All Calls takes precedence over DND indication on Attendant Alternative Answering (AAA) DNs.

Attendant Blocking of Directory Number

The Attendant Blocking of DN feature will override the Do Not Disturb feature. If the dialed DN of the set that has the Do Not Disturb feature active is idle, the DN will be blocked and if the DN is busy, busy tone will be heard.

Attendant Break-In

For a telephone with Do Not Disturb in effect, Break-In is temporarily denied to the attendant. The Break-In lamp uses slow flash to indicate this situation. Using the Break-In key prior to dialing the destination DN circumvents this situation. After the Break-In, the telephone returns to its prior status.

Attendant Break-In to Inquiry Calls

The operation of Do Not Disturb is overridden on a analog (500/2500 type) telephone that has inadvertently been placed on-hook during a Break-In conference to allow it to be re-rung by the attendant.

If the controlling party goes on hook in a Break-In conference, and is being re-rung by the attendant, the ringing takes precedence over Do Not Disturb that may be applied to the set.

Automatic Wake Up

When a telephone is configured for Do Not Disturb, a wake up call can still be presented.

Call Forward All Calls Hunting

If activated, Call Forward All Calls, Call Forward, Internal Calls and Hunting take precedence over DND busy indication.

Call Forward/Hunt Override Via Flexible Feature Code

Do Not Disturb is not overridden by the Call Forward/Hunt Override Via FFC feature.

Call Park

Calls can be parked on and by DNs in DND mode. When a telephone in DND mode parks a call, the call will not return to the DND telephone. It recalls to the attendant.

Camp-On, Forced

Telephones with Do Not Disturb enabled cannot be camped on to with Forced Camp-On. Overflow tone is returned to telephones attempting Forced Camp-On.

China - Attendant Monitor

If an attendant attempts to monitor a DN which has Do Not Disturb activated and is idle, idle DN treatment is given.

Digital Private Signaling System 1 (DPNSS1) Executive Intrusion

Executive Intrusion is not allowed if either of these features is active at the requested party.

Directory Number Expansion

If the Directory Number Expansion (DNXP) package is equipped, DNs can have up to seven digits.

Group Hunt

Do Not Disturb (DND) has priority over Group Hunting. Group Hunting will skip over sets with DND active.

Hunting

If activated, Hunting takes precedence over Do Not Disturb busy indication.

Idle Extension Notification

It is not possible to request for Idle Extension Notification towards an extension that has the Do Not Disturb feature activated.

The Idle Extension Notification feature is not supported on DPNSS networks.

It is not possible to request Idle Extension Notification towards an extension that is Second Degree Busy. Idle Extension Notification is only possible on an extension that is First Degree Busy.

It is not possible to set Idle Extension Notification towards a pilot DN.

Intercept Computer Dial from Directory

This feature can be activated for an extension DN as follows:

- Press an idle Loop key, and press the Do Not Disturb Individual (DND IND) key on the attendant console.
- Dial a DN from the ICT.
- Press the DND IND key once more, and terminate the procedure by pressing the Release key on the attendant console.

The same approach applies when cancelling Do Not Disturb for a set.

To override Do Not Disturb for an extension DN:

- Press an idle Loop key on the attendant console.
- Dial a DN from the Intercept Computer (ICT).

Press the DND IND key on the attendant console.

ISDN QSIG/EuroISDN Call Completion

An incoming notification overrides a set with Do Not Disturb (DND) activated. Call Completion requests can be applied to sets with the DND feature activated. However, this request does not advance until the DND feature is deactivated.

Last Number Redial

A Hot Line key cannot be redialed using the Last Number Redial feature.

Make Set Busy and Voice Call Override

Voice calls are not allowed on a set with attendant-activated Do Not Disturb.

Meridian Hospitality Voice Services

Individual Do Not Disturb (DND) allows the attendant to place a Directory Number into DND mode. A DN in this mode is free to originate calls, but appears busy to incoming calls. With MHVS equipped, a new prompt (DNDH) allows callers to be redirected to Meridian Mail for voice mail services. A called telephone must have Hunting Allowed (HTA) class of service, and Hunt to Meridian Mail and DNDH in LD 15 must both be set to YES.

Network Individual Do Not Disturb

An attendant may receive a visual indication of the state of a set belonging to Group Do Not Disturb mode, whether this set is located on the local node or any other network node.

Network Intercom

Hot Type I calls ignore the Do Not Disturb feature. Hot Line calls are presented to the defined target, even when DND is activated.

Night Station

A Night Station DN can be placed in DND mode.

Override

Priority Override

Telephones with DND enabled cannot be overridden. Overflow (fast busy) tone is returned to telephones attempting Priority Override.

Private Line Service

DND cannot be used on Private Lines.

Feature packaging

Do Not Disturb, Individual (DNDI) package 9 has no feature package dependencies.

Do Not Disturb, Group (DNDG) package 16 requires DNDI package 9.

Do Not Disturb Hunt requires Meridian Hospitality Voice Services (MHVS) package 179.

Feature implementation

Task summary list

The following is a summary of the tasks in this section:

1. Table 56: LD 15 - Specify the treatment received by calls to a number in Do Not Disturb mode. on page 203

   Specify the treatment received by calls to a number in Do Not Disturb mode.
2. Table 57: LD 26 - Add or change a Group Do Not Disturb. on page 203

   Add or change a Group Do Not Disturb.
3. Table 58: LD 26 - Merge one or more defined Do Not Disturb groups into another DND group, retaining their status as groups. on page 203

   Merge one or more defined Do Not Disturb groups into another DND group, retaining their status as groups.
4. Table 59: LD 26 - Print Do Not Disturb group data. on page 204

   Print Do Not Disturb group data.
5. Table 60: LD 12 - Add or change Individual or Group Do Not Disturb keys on an attendant console. on page 204

   Add or change Individual or Group Do Not Disturb keys on an attendant console.
6. Table 61: LD 10 - Enable or disable lamp for analog (500/2500 type) telephones. on page 205

   Enable or disable lamp for analog (500/2500 type) telephones.

Table 56: LD 15 - Specify the treatment received by calls to a number in Do Not Disturb mode.

Prompt	Response	Description
REQ	CHG	Change.
TYPE	FTR	Features and options
CUST		Customer number
	0-99	Range for Large System and Avaya Communication Server 1000E (Avaya CS 1000E) system.
- DNDL	(NO) YES	Do Not Disturb lamp for analog (500/2500 type) telephones.
TYPE	INT	Intercept Treatment Option.
- DNDT	(BST) ATT RAN	Busy tone treatment for Do Not Disturb (DND) numbers. Attendant treatment for DND numbers. Recorded announcement for DND numbers.
- - RRT	xxx	Route number for the recorded announcement for calls to a DND number (prompted if DNDT = RAN).
TYPE	RDR	Call Redirection.
- DNDH	(NO) YES	(Disallow) Allow Do Not Disturb Hunt.

Table 57: LD 26 - Add or change a Group Do Not Disturb.

Prompt	Response	Description
REQ	CHG REM	Change, or remove DN in DND group.
TYPE	DND	Do Not Disturb Group data block.
CUST	xx	Customer number, as defined in LD 15
GPNO	0-99	DND group to be added or changed.
STOR	xxx...x	DN to be added or changed in the DND group; repeat to add other DNs.
RMOV	xxx...x	DN to be removed from a DND group. Prompted if REQ = REM.

Table 58: LD 26 - Merge one or more defined Do Not Disturb groups into another DND group, retaining their status as groups.

Prompt	Response	Description
REQ	MRG CHG REM OUT	Merge DND groups. Add a DND group from a list of merged DND groups. Remove DND group from a merged group. Remove a DND group that consists of a list of merged DND groups.

Prompt	Response	Description
TYPE	DND	Do Not Disturb Group data block.
CUST	xx	Customer number, as defined in LD 15
GPNO	0-99	Number of the DND group to be created through merging of other DND groups.
GRP1	G0-G99	Number of the first DND group to be merged (total number of members in all merged DND groups cannot exceed 127). Prompted if REQ = MRG.
GRP2	G0-G99	Number of the second DND group to be merged (total number of members in all merged DND groups cannot exceed 127). Prompted if REQ = MRG.
GRP	G0-G99	Number of the DND group to be merged (total number of members in all merged DND groups cannot exceed 127). Prompted if REQ = MRG.
STOR	G0-G99	Specify the number of the DND group to be added to a list of merged DND groups. Prompted if REQ = CHG.
RMOV	G0-G99	Specify the number of the DND group to be removed from a list of merged DND groups. Prompted if REQ = REM.

Table 59: LD 26 - Print Do Not Disturb group data.

Prompt	Response	Description
REQ	PRT	Print.
TYPE	DND	Do Not Disturb Group data block.
CUST	xx	Customer number, as defined in LD 15
GPNO	0-99 <CR>	DND group to be printed. Print all DND group data.

Table 60: LD 12 - Add or change Individual or Group Do Not Disturb keys on an attendant console.

Prompt	Response	Description
REQ	CHG	Change.
TYPE	2250	Attendant console type.
TN		Terminal number
	l s c u	Format for Large System, MG 1000B, and CS 1000E system, where l = loop, s = shelf, c = card, u = unit.

Prompt	Response	Description
KEY	xx DDL	Add an Individual Do Not Disturb key, where xx = 0-19 for M2250 consoles.
KEY	xx GND 0-99	Add a DND group key, where xx = 0-19 for M2250 consoles.

Table 61: LD 10 - Enable or disable lamp for analog (500/2500 type) telephones.

Prompt	Response	Description
REQ:	CHG	Change.
TYPE:	500	Telephone type.
TN		Terminal number
	l s c u	Format for Large System, MG 1000B, and CS 1000E system, where l = loop, s = shelf, c = card, u = unit.
CLS	(LPD) LPA (CCSD) CCSA	(Disable) enable lamp. Controlled Class of Service (denied) allowed.

Feature operation

Individual Do Not Disturb

To activate DNDI using the DNDI key (Attendant Console):

1. Select an idle loop key.
2. Press DNDI.
3. Dial the DN of the telephone to place into DND mode.
4. Press DNDI again. (Ignore status of indicator.)
5. Press Rls.

To deactivate DNDI, follow the same steps.

Group Do Not Disturb

There are two ways to activate DNDG: with the DNDG key or with the DNDI key.

To activate DNDG using the DNDG key (Attendant Console):

1. Press DNDG. This key already has a defined group assigned to it. The associated indicator remains steadily lit to indicate that all telephones in that DND group are in DND mode.
2. Press Rls.

To deactivate DNDG:

- Press DNDG.

To activate DNDG using the DNDI key (Attendant Console):

1. Select an idle loop key.
2. Press DNDI.
3. Press the octothorpe (#) key.
4. Dial the group number.
5. Press # again.
6. Press DNDI again.
7. Press Rls.

Chapter 27: DTMF Handling using RFC2833

Contents

This section contains information on the following topics:

Feature description

Traditionally, on VoIP trunks (e.g. H.323), a key press on a set would result in either; 1) a tone being sent across the VoIP stream which could be distorted due to the compression characteristics of the codec or 2) a message would be sent to the Call Server which would provide an out of band signal to the far end indicating a DTMF tone. With RFC2833, a key press on a set is translated into a packet(or packets) that flow with the VoIP stream to the far end. This special packets are RFC2833 packets which contain the DTMF key that was pressed. The same idea applies to TDM devices that are involved in a VoIP call. The VGW TN that converts the TDM stream to VoIP also detects a tone on the TDM side and translates it to RFC2833 packets on the VoIP side. As well, the VGW TN can receive an RFC2833 packet on the VoIP side and generates a tone on the TDM side.

Operating parameters

The system is always on.

Feature interactions

Media Security Phase 1

There is the interaction between RFC2833 and Media Security Phase 1 features in VGW TN selection agorithm. RFC2833 feature allocates RFC2833 capable VGW TNs if available, and the other ones if necessary. Media Security Phase 1 feature allocates SRTP capable VGW TNs if available, and the other ones if necessary. The search is optimized to be as fast as possible; that is, in one pass, both (RFC2833 capable and SRTP capable) are targeted first.

Active Call Failover (ACF)

There is the interaction between RFC2833 and Active Call Failover (ACF) features. In Geographic Redundant systems, if IP Phase 2 set is in a call and primary Call Server and Signalling Server are out of service, the IP Phone will reregister to the Secondary Signalling Server. Since ACF feature provides that only local calls survives Primary Call Server failure, RFC2833 feature is not affected. In the Main Office/Branch Office setup with ACF, the Branch Office IP Phone (normal mode) to Branch Office TDM set call will survive the Main Office failure. This is now local Branch Office call and RFC2833 will not be supported.

Feature packaging

The following table describes the total required X21 packaging for this feature to be operable.

Table 62: Total required X21 packaging

Package Mnemonic	Package Number	Package Description	Package Type (New or Existing or Dependency)	Applicable Market
BASE			EXISTING	Global
SIP	406	SIP trunks can be configured	EXISTING	Global

Feature implementation

There will be no new provisioning for the RFC2833 feature.

DTMF detection can be enabled/disabled, on a per node basis, in Element Manager (Communication Server 1000 Release 4.5). It is applicable to VGW TNs only, not IP Phones. It is not RFC2833 configuration, but it does not affect the RFC2833.

If the RFC2833 is negotiated successfully (both sides advertised their RFC2833 Rx capability), then the RFC2833 will be used in both Tx and Rx directions no matter how the DTMF detection is provisioned.

If the RFC2833 is not negotiated successfully, then the RFC2833 will not be used for DTMF transport. DTMF digits will be transported either out-of-band or in-band (tone as audio) depending on the DTMF detection configuration. The behavior is the same as in CS 1000 4.5.

Feature operation

The feature operates without user input.

Chapter 28: Dual Signaling on Analog Trunks

Contents

This section contains information on the following topics:

Feature description

A telephone user can select any interexchange carrier for any given call by using a Carrier Access Code (CAC). A CAC comprises an Equal Access identifier and a Carrier Identification Code (CIC). Avaya refers to a call preceded by a CAC as an Equal Access call.

The Dual Signaling on Analog Trunks feature allows Dial Pulse signaling and Digitone signaling to be applied separately to incoming and outgoing calls on one trunk. It reduces the number of Digitone Receiver (DTR) units required on the system since these units are no longer necessary for incoming calls on trunks programmed with the new DPDT Class of Service.

Dual Signaling on Analog Trunks introduces the following trunk Classes of Service in LD 14:

- DPDT = digit information is received as Dial Pulse and sent as Digitone
- DTDP = digit information is received as Digitone and sent as Dial Pulse

Prior to the introduction of Dual Signaling on Analog Trunks, a similar functionality was available when trunks were programmed for DTMF signaling. Dial Pulse calls, if received, were analyzed and handled by the Tone and Digit Switch or Extended Conference and Tone Service card. A DTR was reserved, needlessly, for the duration of the signaling.

The following diagram shows one application of the feature.

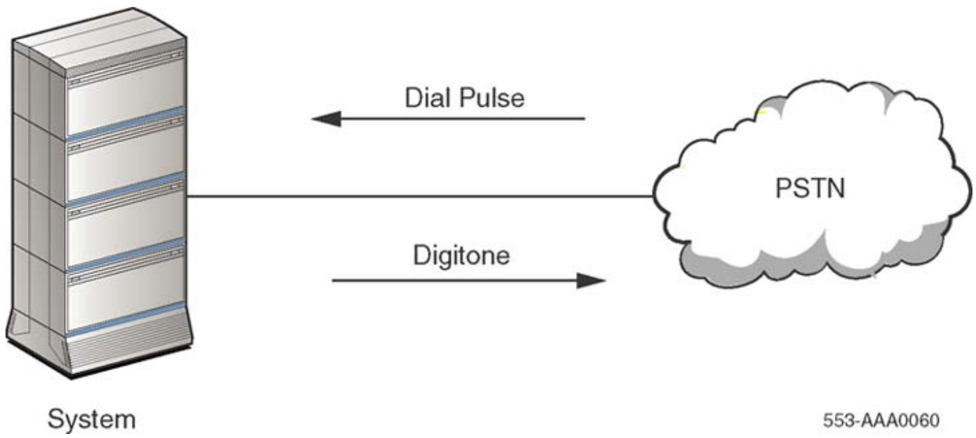

Figure 2: System connected to the CO through analog trunks interface

This feature enables a trunk to be configured in one of the following ways:

- incoming Dial Pulse - outgoing Dial Pulse
- incoming DTMF - outgoing DTMF
- incoming Dial Pulse - outgoing DTMF
- incoming DTMF - outgoing Dial Pulse

Operating parameters

The new Classes of Service (DPDT and DTDP) are mutually exclusive with DIP, DTN, MFC, MFE, MFK, MFR and MFX.

If Dual Signaling on Analog Trunks is used on a trunk with DPDT programmed, a DTR is not involved with incoming trunk traffic.

This feature is available on analog DID and TIE trunks only.

CLS DPDT/DTDP can only be configured on routes with the ICOG prompt set to IAO (incoming and outgoing).

Feature interactions

There are no feature interactions associated with this feature.

Feature packaging

This feature is included in base system software.

Feature implementation

Table 63: LD 14 - Configure the trunk with the Dual Signaling on Analog Trunks Class of Service.

Prompt	Response	Description
REQ	CHG	Change existing data.
TYPE	aaa	Trunk type. xxx = DID, TIE.
TN		Terminal number
	l s c u	Format for Large System, Avaya CS 1000 Media Gateway 1000B (Avaya MG 1000B), and Avaya Communication Server 1000E (Avaya CS 1000E) system, where l = loop, s = shelf, c = card, u = unit.
CLS	(DIP) DPDT DTDP	Dial Pulse. Incoming Dial Pulse - outgoing Digitone. Incoming Digitone - outgoing Dial Pulse.

Feature operation

No specific operating procedures are required to use this feature.

Chapter 29: E.164 Enhancement

Contents

This section contains information on the following topics:

Feature description

The E.164 enhancement feature allows manipulation of the calling number digits, numbering plan indicator (NPI), and type of number (TON) for incoming DID calls if the TON/NPI of the calling number is E.164/UNKNOWN or UNKNOWN/UNKNOWN and the National Access Code (NATC) / International Access Code (INTC) is configured in the Customer Data Block (CDB). If the leading digits in the calling number match the INTC code, the TON/NPI is set to E.164/INTERNATIONAL and the matching digits are removed. If the leading digits in the calling number match the NATC code, the TON/NPI is set to E.164/NATIONAL and the matching digits are removed. The only configuration requirement for this feature is configuring NATC/INTC in the Networking block in the CDB.

Operating parameters

There are no operating parameters associated with this feature.

Feature interactions

Flexible CLID Manipulation Table

The E.164 Enhancement feature interacts with the Flexible CLID Manipulation Table feature. When a call comes in, CLID Manipulation Data Block (CMDB) rules are applied first and E.164 rules are applied second. Administrators can avoid unwanted interactions (E.164 rules overwriting CMDB rules) by configuring the Flexible CLID Manipulation Table as follows:

- Configure the CMDB so that after CMDB rules apply, TON/NPI is not configured to UNKNOWN/UNKNOWN. In this configuration the E.164 enhancement feature does not make any additional changes to the calling number.
- If you want to configure TON/NPI to UNKNOWN/UNKNOWN, you can configure the CMDB to remove the leading digits of the calling number. In this case, there is not a match for the INTC or CNTC codes, and the CLID does not satisfy the matching rules for applying E.164 enhancement features.

Note:

This feature is supported only for incoming ISDN DID routes.

For information about the Flexible CLID Manipulation Table feature, see Flexible CLID Manipulation Table on page 341 in this book.

For information about E.164 CLID operation on Office Communication Server (OCS), see *Avaya Converged Office Fundamentals — Microsoft Office Communications Server 2007, NN43001–121*.

Feature packaging

E.164 Enhancement has no feature package dependencies.

Feature implementation

Configure the National Access Code (NATC) and International Access Code (INTC) in the Networking Block in the CDB in LD 15. For more information, see *Software Input Output Reference - Administration, NN43001-611*.

Table 64: LD 15 — Configure the Networking Data Block for NATC and INTC

Prompt	Response	Description
...		
NATC	x	National Access Code
INTC	xxx	International Access Code

Feature operation

No specific operating procedures are required to use this feature.

Chapter 30: Early Media for Universal Extensions

Contents

This section contains information on the following topics:

Feature description

The Early Media feature provides the ability for SIP trunk calls to determine a speech path before receiving the information from the call destination. Early media consists of audio and video media that is exchanged before a call session is accepted by the call destination.

Early media generated by a caller typically consists of voice commands or dual-tone multifrequency (DTMF) tones that drive interactive voice response (IVR) systems. Typical examples of early media generated by a call destination are ring tone and announcements.

Early media occurs from the moment the initial INVITE is sent until the User Agent Server (UAS) generates a final response. It can be unidirectional or bidirectional and generated by the call originator, the call recipient, or both.

Operating parameters

Note:

The Early Media class of service (CLS) is only available for Universal Extensions (UEXT).

The Early Media class of service can be enabled (ELMA) or denied (ELMD). The default value for the class of service depends on the type of UEXT.

The following UEXTs have a default value of ELMA (Early Media Allowed):

- TLSV

The following UEXTs have a default value of ELMD (Early Media Denied):

- MOBX
- SIPN
- SIP3
- FMCL
- SIPL

> **Note:**
> The SIPL class of service cannot be changed to ELMA.

UEXTs with a CLS of ELMA provide early media to incoming calls until the call is established. Examples of early media include in-band ringback tones, customized tones from mobile phones, and various greetings and announcements. UEXTs with a CLS of ELMD only provide local ringback to incoming calls until they are established. No early media is provided.

For each MADN (Multiple Appearance Directory Numbers) group, only one UEXT can render early media to a caller. For example, if a MADN group contains several UEXTs and if during a call to the DN several trunks begin to provide early media, only the early media of the trunk associated with the UEXT ELMA is played. However, the caller connects to the UEXT endpoint that establishes the call, regardless of whether or not the endpoint belongs to a trunk that provides early media.

Feature interactions

There are no feature interactions associated with this feature.

Feature packaging

This feature is included in base system software.

Feature implementation

Table 65: LD 11 - Early Media Class of Service configuration

Prompt	Response	Description
REQ	aaa	NEW or CHG
TYPE	bbb	Terminal type. This can be any Universal Extension (UEXT).
CUST	xx	Customer number.
CLS	(ELMA) / ELMD	Class of Service option, where: • ELMA = Early media allowed (default) • ELMD = Early media denied
...		

Element Manager

You can select and configure the Early Media class of service using the Phones configuration page in Element Manager. For information about Phones configuration using Element Manager, see *Avaya Element Manager System Reference—Administration, NN43001-632*.

Feature operation

No specific operating procedures are required to use this feature.

Chapter 31: Electronic Brandlining

Contents

This section contains information on the following topics:

Feature description

The Electronic Brandlining (EBLN) feature enhances the display functionality of Meridian Modular sets. This feature allows the second line on the idle display screen of a Meridian Modular set to show a custom display.

Previous to the Electronic Brandlining feature, when a Meridian Modular set is in the idle state, only the time and date is shown on the first line of the display screen and the second line is blank.

The display screen of a Meridian Modular set contains two lines with 24 character spaces on each line. Previously, the second line on the display screen of an idle Meridian Modular set was blank. With the Electronic Brandlining feature, however, a custom display is shown left justified on the second line of the idle display screen.

Incremental Software Management

A License parameter is introduced with the Electronic Brandlining feature. This License parameter is used to transfer custom display information from the Order Management System to system software. The Electronic Brandlining License value is copied from the appropriate tape/keycode/file into system software during sysload. The system software then sends the custom display to the display screen of a Meridian Modular set.

The Electronic Brandlining License value contains one of the following:

- a Terminal Text Broadcast customized text string value
- a default value

The value of the Electronic Brandlining parameter determines the content of the Electronic Brandlining custom display.

LD 22 is modified to print the Electronic Brandlining License parameters. When REQ = SLT (Print System Limits: Incremental Software Management) in LD 22, the License parameters, system limits, and keywords are printed. The printing of the Electronic Brandlining custom display output is added after the License parameters.

Customers can deliver License parameters through keycodes. A keycode is a machine-generated digitally signed list of customer capabilities and authorized software release. A security keycode scheme protects License parameters.

To expand License limits, customers must order and install a new keycode. This installation is performed using the Keycode Management feature. All Keycode Management commands are executed in LD 143. To make the expansion effective, the customer must sysload. For more information about keycode installation, see *Avaya CS 1000M and Meridian 1 Large System Upgrades Overview, NN43021-458*.

Custom Displays

The Electronic Brandlining feature provides the following two custom displays:

- Terminal Text Broadcast Customized Text
- Default "AVAYA" or blank display

Terminal Text Broadcast Customized Text

When the Electronic Brandlining License parameter is equal to the Terminal Text Broadcast value, the customized brandline to be displayed is initially defaulted to AVAYA. This brandline can then be configured to display a different customized brandline.

The customized brandline can have a maximum of 24 characters, each of which must be supported by the North American Meridian Modular set display firmware. Version 18 firmware supports 7-bit ASCII Roman characters and 8-bit non-ASCII Roman characters (See Tables Table 66: Valid 7-bit ASCII Roman Characters on page 226 and Table 67: Valid 8-bit non-ASCII Roman Characters on page 227). Alphanumeric and punctuation characters are supported.The customized brandline is configured on a system basis (LD 17).

Figure 3: An idle Meridian Modular display screen with a customized brandline displayed on page 225 shows an example of a customized brandline displayed on the idle screen of a Meridian Modular set.

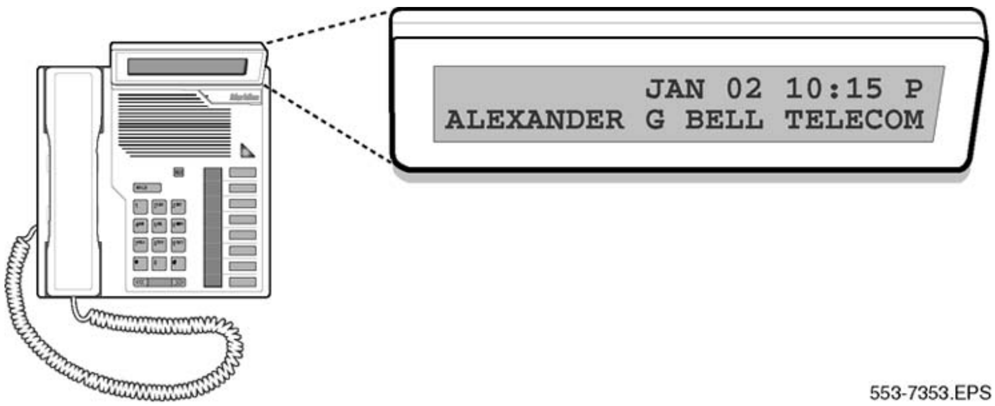

Figure 3: An idle Meridian Modular display screen with a customized brandline displayed

In addition to displaying a customized brandline, the Terminal Text Broadcast functionality can also be used to broadcast a customized text string on the idle display screen of a Meridian Modular set. The text string can have a maximum of 24 supported characters (See Tables Table 66: Valid 7-bit ASCII Roman Characters on page 226 and Table 67: Valid 8-bit non-ASCII Roman Characters on page 227). The customized text string is configured on a system basis (LD 17).

The customized text string can be composed of a single blank space. In this case, the second line of the idle display screen is blank, as per existing functionality.

To enter the customized brandline or text string in LD 17, use one of the following methods:

- Enter a line of supported characters followed by a Carriage Return (<CR>) at the IDLE_DISP_STRING prompt in LD 17.
- Enter a valid character one at a time using either a supported character or its two digit hexadecimal representation at the IDLE_DISP_CHAR prompt in LD 17. The end of input is indicated when only a <CR> is entered or when the 24th character is entered.

Figure 4: An idle Meridian Modular display screen with a customized text string displayed on page 226 shows an example of a customized text string displayed on the idle screen of a Meridian Modular set.

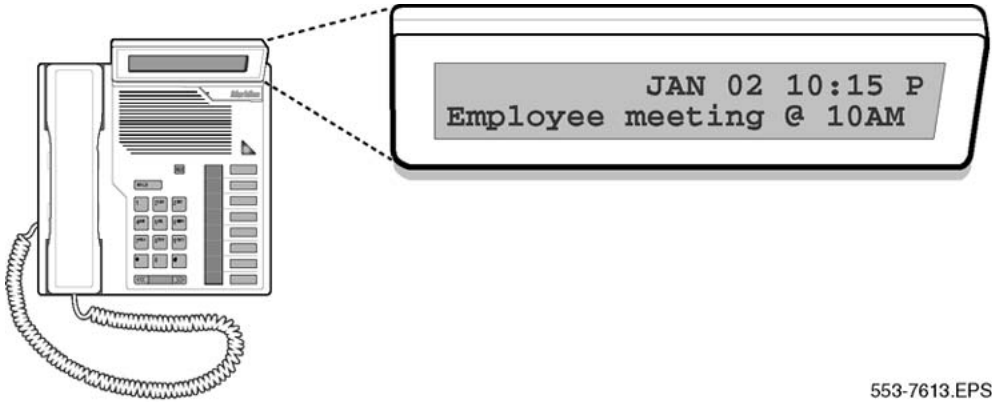

Figure 4: An idle Meridian Modular display screen with a customized text string displayed

Supported characters

Table Table 66: Valid 7-bit ASCII Roman Characters on page 226 lists the 7-bit ASCII Roman characters and the corresponding hexadecimal representations that are supported by the Electronic Brandlining feature.

Table 66: Valid 7-bit ASCII Roman Characters

20 <space>	21 !	22 "	23 #	24 $	25 %
26 &	27 ’	28 (	29)	2A *	2B +
2C ,	2D -	2E .	2F /	30 0	31 1
32 2	33 3	34 4	35 5	36 6	37 7
38 8	39 9	3A :	3B ;	3C <	3D =
3E >	3F ?	40 @	41 A	42 B	43 C
44 D	45 E	46 F	47 G	48 H	49 I
4A J	4B K	4C L	4D M	4E N	4F O
50 P	51 Q	52 R	53 S	54 T	55 U
56 V	57 W	58 X	59 Y	5A Z	5B [
5C \	5D]	5E ^	5F _	60 `	61 a
62 b	63 c	64 d	65 e	66 f	67 g
68 h	69 i	6A j	6B k	6C l	6D m
6E n	6F o	70 p	71 q	72 r	73 s
74 t	75 u	76 v	77 w	78 x	79 y
7A z	7B {	7C |	7D }	7E ~	7F ¿

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Table 67: Valid 8-bit non-ASCII Roman Characters on page 227 lists the 8-bit non-ASCII Roman characters and the corresponding hexadecimal representations that are supported by the Electronic Brandlining feature.

Table 67: Valid 8-bit non-ASCII Roman Characters

A0 <NASP>	A1 Ľ	A2 ¢	A3 £	A4 Ń	A5 ¥
A6 Ś	A7 Ď	A8 ¨	A9 ©	AA Ň	AB Ť
AC Ź	AD Ÿ	AE ®	AF Ż	B0 °	B1 ±
B2 Ą	B3 Ł	B4 Ř	B5 L´	B6 Ć	B7 Č
B8 Ě	B9 Š	BA º	BB Ę	BC Ŕ	BD U̇
BE Ž	BF ¿	C0 À	C1 Á	C2 Â	C3 Ã
C4 Ä	C5 Å	C6 Æ	C7 Ç	C8 È	C9 É
CA Ê	CB Ë	CC Ì	CD Í	CE Î	CF Ï
D0 Đ	D1 Ñ	D2 Ò	D3 Ó	D4 Ô	D5 Õ
D6 Ö	D7 ×	D8 Ø	D9 Ù	DA Ú	DB Û
DC Ü	DD Ý	DE Þ	DF ß	E0 à	E1 á
E2 â	E3 ã	E4 ä	E5 å	E6 æ	E7 ç
E8 è	E9 é	EA ê	EB ë	EC ì	ED í
EE î	EF ï	F0 ð	F1 ñ	F2 ò	F3 ó
F4 ô	F5 õ	F6 ö	F7 ÷	F8 ø	F9 ù
FA ú	FB û	FC ü	FD ý	FE þ	FF ÿ

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Characters that are listed in Table 66: Valid 7-bit ASCII Roman Characters on page 226 and Table 67: Valid 8-bit non-ASCII Roman Characters on page 227 are available with North American Version 18 firmware. Individual TTYs may not match the characters and hexadecimal representations in the same way as shown in Tables Table 66: Valid 7-bit ASCII Roman Characters on page 226 and Table 67: Valid 8-bit non-ASCII Roman Characters on page 227.

Default Electronic Brandlining Display

If the Terminal Text Broadcast custom display is not chosen, then the Electronic Brandlining parameter value indexes into a default brandline. The default brandline is "AVAYA", the Meridian Modular set manufacturer (Avaya). This default brandline is displayed left justified on the second line of the idle display screen of a Meridian Modular set.

The default brandline is enabled/disabled on a system basis (LD 17). When the AVAYA_BRAND prompt is set to NO, the second line of the idle Meridian Modular display screen is blank, as per existing functionality.

Figure 5: An idle Meridian Modular display screen with the Default Electronic Brandlining, AVAYA displayed on page 228 shows the default brandline (AVAYA) displayed on an idle Meridian Modular set.

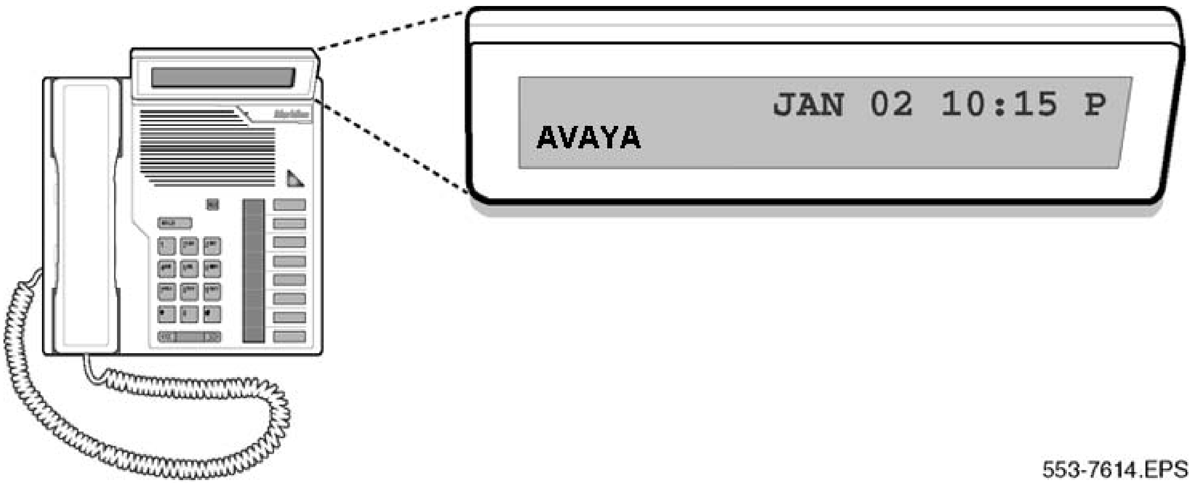


Figure 5: An idle Meridian Modular display screen with the Default Electronic Brandlining, AVAYA displayed

Operating parameters

The Electronic Brandlining feature applies to Meridian Modular sets that are equipped with a display screen and the appropriate Meridian Modular display firmware. Meridian Modular sets include: M2008, M2016, M2616, M2216ACD1, and M2216ACD2.

The Meridian Modular display firmware, North American Version 18 (Three Language Display) or later, is required for Meridian Modular sets to use the Electronic Brandlining feature. North American Version 18 firmware supports English, French, and Spanish.

The North American Version 18 firmware stores and displays the custom display. If the custom display is sent to a Meridian Modular set without the new firmware, the extra Scan and Signal Distributor (SSD) messages are ignored.

There is an incremental impact of sending SSD messages for a customized brandline. Therefore, it is recommended that no brandlining be done for heavily loaded systems experiencing delays on the High Speed Link (HSL). Instead, the default EBLN brandline can be chosen. Only one SSD message is sent whether the AVAYA_BRAND prompt is set to YES or NO. To minimize the number of SSD messages with the Terminal Text Broadcast custom display, a blank display can be configured. In this case, the customized text string is composed of a single blank space, and only one SSD message is sent for the same real time impact as the default EBLN custom display.

The custom display can have a maximum of 24 characters. Each character must be supported by North American Version 18 firmware.

Version 18 firmware supports 7-bit ASCII Roman characters and 8-bit non-ASCII Roman characters, regardless of whether or not the Multi-language TTY Input/Output (MLIO) package 211 is equipped. Alphanumeric and punctuation characters are supported.

When the MLIO package is restricted, if the "Valid 8-bit non-ASCII Roman Characters" that are supported are used in a custom display, then a 7-bit TTY may not be able to print the characters. If not, then each character is replaced with an underscore character.

If the MLIO package is not restricted and a 7-bit TTY is used, the 8-bit supported characters cannot be printed correctly. Instead, the service change administration interfaces may print garbage characters and/or the interfaces may lock.

When the MLIO package is not restricted, the system sends the valid 8-bit characters to the TTY, rather than the underscore characters. With the MLIO package equipped, it is assumed that the TTY is capable of handling 8-bit characters. If the TTY is capable of entering the "¿" 7-bit character and all other supported 8-bit characters directly, then these characters are accepted by the system, without using the hexadecimal values for the Terminal Text Broadcast customized text. The hexadecimal values can, however, still be used for entries.

The "!" character cannot be entered directly from the TTY keyboard. It can be entered, however, through character-by-character input (IDLE_DISP_CHAR nn prompt in LD 17), using its hexadecimal value.

When the system does not recognize a temporary power outage on a Meridian Modular set, the screen may remain blank until the custom display information, along with the time and date information, is downloaded again.

If the new Electronic Brandlining parameter has an invalid value, the default display is shown. In this case, conversion should have defaulted the AVAYA_BRAND to YES, and as long as this prompt has not been changed, "AVAYA" is displayed.

If the Electronic Brandlining parameter is set to the Terminal Text Broadcast value and the customized text string is configured as "AVAYA" or blank, the AVAYA_BRAND option does not apply. The AVAYA_BRAND option only applies to toggles between "AVAYA" and a blank second line if the Electronic Brandlining parameter is set to the default value.

For new systems, the AVAYA_BRAND prompt is automatically set to YES (default), and the "AVAYA" default brandline is displayed. For the Terminal Text Broadcast option, the AVAYA_BRAND field is automatically set to YES (default); although, the AVAYA_BRAND field is not applicable nor is it output in LDs 17 and 22. The Terminal Text Broadcast customized brandline is initially set to the default "AVAYA" brandline.

No changes are made to the features which currently output information on the second line of the idle display screen of a Meridian Modular set. These features and their output have precedence over the Electronic Brandlining feature. The following idle screens take precedence over the Electronic Brandlining feature: Automatic Answerback, Call Forward, Logged Out, Make Set Busy, Not Ready, and Overflow Busy.

Feature interactions

Automatic Answerback

When Automatic Answerback (AAB) is activated on a Meridian Modular set, the second line of the idle display screen shows "AUTO ANSWER ACTIVATED".

The Electronic Brandlining custom display is not shown when AAB is activated.

Call Forward All Calls Internal Call Forward

When Call Forward All Calls or Internal Call Forward is activated on a Meridian Modular set, the second line of the display screen shows "CFWD" on the idle screen. The Electronic Brandlining custom display is not shown when Call Forward All Calls or Internal Call Forward is activated.

When Call Forward All Calls or Internal Call Forward is de-activated on a Meridian Modular set, the second line of the display screen shows "CALL FORWARD CANCELLED" on the idle screen for a few seconds. The Electronic Brandlining custom display is not shown while "CALL FORWARD CANCELLED" is displayed. When the "CALL FORWARD CANCELLED" display times out, the Electronic Brandlining custom display is shown.

Digital Set Display Download

With the Electronic Brandlining feature, the existing time and date messages are modified to include the Electronic Brandlining custom display as part of its data (if applicable).

Display key

When the Display (DSP) key is first pressed, the display screen is blank. When any other key is pressed after the DSP key is pressed, all relevant information is displayed.

The Electronic Brandlining custom display is not displayed during the DSP key process until Lamp Audit updates the display screen with the time and date (when applicable).

Do Not Disturb

When a set is in the Do Not Disturb (DND) mode, the second line of the idle display screen is blank. Therefore, the second line displays the Electronic Brandlining custom display when the Electronic Brandlining feature is enabled (if applicable).

Limited Access to Overlays

The existing functionality of the Limited Access to Overlays (LAPW) feature is not changed as a result of the Electronic Brandlining feature.

The Terminal Text Broadcast configuration of a customized text string in LD 17 is password protected by level 2 system administration (PWD2). The added implementation of PWD2 in LD 17 is required to allow configuration of the Terminal Text Broadcast customized text string.

As per existing functionality, when LAPW is disabled on a system, the PWD2 password is restricted to a 4-digit password composed of the hexadecimal digits 0-9 and/or A-F.

As per existing functionality, when LAPW is enabled, PWD2 can be configured as a 16-digit alphanumeric password. LAPW then applies to the PWD2 prompt.

Make Set Busy

When Make Set Busy (MSB) is activated on a Meridian Modular set, the second line of the idle display screen shows "SET BUSY ACTIVATED".

The Electronic Brandlining custom display is not shown when Make Set Busy is activated.

Set Based Administration

When a service change is made by Set Based Administration (SBA), the downloading of the time, date, and the Electronic Brandlining custom display (if applicable) is induced.

Set Relocation

Automatic Set Relocation (ASR) and Modular Telephone Relocation (MTR) include the "plugging in" of a Meridian Modular set for its feature operation. When a Meridian Modular set is "plugged in", the power-on-reset induces the downloading of the time, date, and Electronic Brandlining custom display (if applicable).

System Access Enhancements

The existing functionality of the System Access Enhancements (SAE) feature is not changed as a result of the Electronic Brandlining feature.

The SAE feature applies to the added implementation of the PWD2 prompt in LD 17 for the Terminal Text Broadcast configuration of a customized text string.

Feature packaging

This feature is included in base system software.

Feature implementation

Task summary list

The following is a summary of the tasks in this section:

1. Table 68: LD 17 - Configure the AVAYA Electronic Brandline. on page 232

   Configure the AVAYA Electronic Brandline.
2. Table 69: LD 17 - Enter a customized text string. on page 233

   Enter a customized text string.
3. Table 70: LD 11 - Enable the display on a Meridian Modular set. on page 234

   Enable the display on a Meridian Modular set.

Table 68: LD 17 - Configure the AVAYA Electronic Brandline.

Prompt	Response	Description
REQ	CHG	Change existing data.
TYPE	PARM	System Parameters.
...		
AVAYA_BRAND	(YES)	"AVAYA" Electronic Brandline is displayed (default).

Prompt	Response	Description
	NO	"AVAYA" Electronic Brandline is not displayed. AVAYA_BRAND is only prompted when parameter is set to the default value.

Table 69: LD 17 - Enter a customized text string.

Prompt	Response	Description
REQ	CHG	Change existing data.
TYPE	PARM	System Parameters.
...		
IDLE_SET_DISPLAY aaaa		The current customized text string "aaaa" is shown. This information is displayed for confirmation only.
- - MODIFY		Gateway to new EBLN Terminal Text Broadcast configurations.
	(NO)	Enter NO to keep existing configuration (default).
	YES	Enter YES to prompt for further EBLN Terminal Text Broadcast configuration.
- - PWD2	x...x	Password 2. The second level administration password is needed to allow configuration of the Terminal Text Broadcast customized text string.
- - SUPPORTED_TEXT_ONLY		Change customized text string by text string input.
	(YES)	Enter YES to input by text string, and the IDLE_DISP_STRING prompt is prompted.
	NO	Enter NO to input character by character, and the IDLE_DISP_CHAR nn prompt is prompted.
- - - IDLE_DISP_STRING	bbbb	Enter the customized text string. IDLE_DISP_STRING is prompted only if SUPPPORTED_TEXT_ONLY = YES. A maximum of 24 supported characters are accepted and validated. For a blank display, enter <CR> only.
IDLE_SET_DISPLAY bbbb		The customized text (bbbb) entered at the IDLE_DISP_STRING prompt is shown. This information is displayed for confirmation only. It is confirmed at the following OK prompt.

Prompt	Response	Description
- - OK		Confirm the validated Terminal Text Broadcast customized text string (bbbb) entered at the IDLE_DISP_STRING prompt.
	(YES)	Enter YES to keep the new text string as "bbbb".
	NO	Enter NO to input a new Terminal Text Broadcast customized text string, and the Supported_TEXT_ONLY prompt is re-prompted.
...		
- - - IDLE_DISP_CHAR nn		Enter the customized text string character by character.
	c	c = one supported character
	hh	hh = 2 hexadecimal digits (0-9, A-F, a-f), representing a supported character. nn (01-24) is the position of the character in the customized text string. The IDLE_DISP_CHAR prompt is only prompted if SUPPORTED_TEXT_ONLY = NO. It is reprompted until <CR> only is entered or until nn is the 24th character that has been entered.
IDLE_SET_DISPLAY cccc		The customized text string (cccc) entered at the IDLE_DISP_CHAR prompt is shown. This information is displayed for confirmation only. It is confirmed at the following OK prompt.
- - OK		Confirm the validated Terminal Text Broadcast customized text string (cccc) entered at the IDLE_DISP_CHAR nn prompts.
	(YES)	Enter YES to keep the new text string as "cccc".
	NO	Enter NO to input a new Terminal Text Broadcast customized text string, and the SUPPORTED_TEXT_ONLY prompt is re-prompted.

Table 70: LD 11 - Enable the display on a Meridian Modular set.

Prompt	Response	Description
REQ:	NEW CHG	Add new data. Change existing data.
TYPE:	a...a	Telephone type. Type **?** for a list of possible responses.

Prompt	Response	Description
TN		Terminal number
	l s c u	Format for Large System, Avaya CS 1000 Media Gateway 1000B (Avaya MG 1000B), and Avaya Communication Server 1000E (Avaya CS 1000E) system, where l = loop, s = shelf, c = card, u = unit.
...		
CLS	(ADD) DDS	Digit Display options Automatic Digit Display (default). Delay Display. When CLS = DDS, the display is activated after the call is answered.
...		

Feature operation

No specific operating procedures are required to use this feature.

Chapter 32: Electronic Switched Network

Contents

This section contains information on the following topics:

Feature description

The Electronic Switched Network (ESN) group of features is designed to support voice and circuit-switched voiceband data telecommunications needs for multiple-location customer applications.

Basic Authorization Code

The Basic Authorization Code (BAUT) feature provides up to 5000 authorization codes of 1 to 14 digits that allow selected users to temporarily override system access limitations by dialing a Special Service Prefix (SPRE) code, the digit 6, and the Basic Authorization Code (BAUT). The Basic Authorization Code (BAUT) is used for general applications and is described in *Avaya Basic Network Feature Fundamentals*.

Basic Alternate Route Selection

Basic Alternate Route Selection (BARS) enables calls placed to another location to be routed automatically over the least expensive route. After the Basic Alternate Route Selection (BARS) access code and the desired number have been dialed, Basic Alternate Route Selection (BARS) automatically tries alternate routes to the destination and completes the call over the least expensive route available at the time of dialing. BARS is described in detail in *Avaya Basic Network Feature Fundamentals, NN43001-579*.

Call Back Queuing

Call Back Queuing (CBQ) is an optional feature available to systems equipped with the Basic/Network Alternate Route Selection (BARS/NARS) or Coordinated Dialing Plan (CDP) features. If all facilities are busy when an individual places a BARS, NARS, or CDP call, Call Back Queuing (CBQ) enables the individual to invoke the Ring Again (RGA) feature and receive a callback from the system when a facility becomes available. Call Back Queuing (CBQ) is described in detail in Network Queuing description or *Avaya Basic Network Feature Fundamentals, NN43001-579*.

Call Back Queuing to Conventional Mains

Call Back Queuing to Conventional Mains (CBQCM) enables call originators at a Conventional Main (any type of switch, including switches that are part of an Electronic TIE Network [ETN]) to access the CBQ feature at the serving ESN Node. When offered CBQ by the Node, users at the Conventional Main dial their extension number to accept the CBQ offer. When facilities become available at the Node, it initiates a CBQ callback to the call originator at the Conventional Main. See *Avaya Basic Network Feature Fundamentals, NN43001-579* for a detailed description of Call Back Queuing to Conventional Mains (CBQCM).

Coordinated Call Back Queuing

Coordinated Call Back Queuing (CCBQ) enables telephones eligible for Ring Again (RGA) at the Main to be offered CBQ when network calls are blocked at the serving Node. When facilities become available at the Node, the call originator at the Main is alerted by a callback (identical to an RGA callback) from the Node. Coordinated Call Back Queuing (CCBQ) requires that the Main and associated Node be equipped with Network Signaling. See *Avaya Basic Network Feature Fundamentals, NN43001-579* for a detailed description of Coordinated Call Back Queuing (CCBQ).

Coordinated Call Back Queuing Against Main

Coordinated Call Back Queuing Against Main (CCBQAM) is an enhancement to the CCBQ feature that allows a station at the Node to be offered CBQ if a call is blocked at the Main. When facilities become available at the Main, the call originator at the Node is alerted by a callback from the Main. The Network Signaling feature must be equipped at both the Main and the Node for Coordinated Call Back Queuing Against Main (CCBQAM) implementation.

Coordinated Dialing Plan

Coordinated Dialing Plan (CDP) enables a customer with a number of switches to coordinate the dialing plan of stations at these switches. The Coordinated Dialing Plan (CDP) feature allows the telephone user to call any other telephone within a CDP group by dialing a three- to seven-digit number assigned to the station. CDP can be arranged to provide a centralized public exchange network capability that channels access to the public network through a single system switch within the CDP group.

CDP routes Direct Inward Dialed (DID) calls over Central Office (CO) and Wide Area Telephone Service (WATS) trunks using a Distant Steering Code (DSC). The feature is controlled by the Customer Data Block (LD 15). This applies to CO, WATS, Data Terminal Interface (DTI), and Integrated Services Digital Network (ISDN) trunks.

CDP is described in detail in the *Avaya Dialing Plans Reference, NN43001-283*.

Flexible ESN 0 Routing

Flexible ESN 0 Routing allows the routing of calls on different routes based on a few predefined non-leftwise unique dialing sequences. "Leftwise unique" means that each entry cannot match the left most portion of any other entry in the table. For example, if "123" is an entry in the table, then no other entry may begin with "123."

The ESN translation table will allow any or all of the following non-leftwise unique numbers (along with their associated route lists) to be entered into the ESN translation table:

- 0
- 00
- 01
- 011

Flexible ESN "0" Routing is part of the existing BARS (57) and Network Alternate Route Selection (NARS) (58) packages and has no interaction with other features besides these. Since NARS has two translation tables, two Flexible ESN "0" Routing data blocks will be included in NARS. This means that a call could be configured to route in two different ways.

This feature applies to all route types and network types supported by ESN. For information about the appropriate prompts and responses in Service Change (LD 90), see *Avaya Software Input Output Reference — Administration, NN43001-611*.

Network Alternate Route Selection

Network Alternate Route Selection (NARS) is an integral part of Avaya's ESN. Network Alternate Route Selection (NARS) is designed for large business customers with numerous distributed operating locations, enabling the customer to tie together the switches at the various operating locations to create a private telecommunications network. NARS is described in detail in the *Avaya Basic Network Feature Fundamentals, NN43001-579*.

BARS/NARS Incoming Trunk Group Exclusion

Incoming Trunk Group Exclusion (ITGE) is an enhancement to the BARS/NARS feature. Standard call blocking is applied on outgoing calls to a specific Numbering Plan Area (NPA), NXX, Special Number (SPN), or Location Code (LOC) at the ESN node if the call is from a specific incoming trunk group.

This prevents loopback routing through the caller's home switch (home NPA, NXX). Calls that should have been made off-net from the caller's home switch are blocked outgoing at the Node. Main users are prevented from using the ESN to make calls to certain NPA, NXX, SPN, or Location Codes (LOC) that they are restricted from making at the home switch.

Incoming Trunk Group Exclusion (ITGE) provides full ten-digit restriction for NPA and SPN codes, seven-digit restriction for NXX codes, and three-digit restriction for Location Code (LOC) codes.

Detailed information on this enhancement is provided in *Avaya Basic Network Feature Fundamentals, NN43001-579*.

NARS Multiple DID Office Code Screening

Multiple DID Office Code Screening is an enhancement to the On-Net to Off-Net Overflow capability of the NARS feature. This enhancement permits network calls that undergo on-net to off-net conversion to terminate at any Directory Number (DN) that has been defined in the LOC data block of memory. This data block allows the definition of multiple office codes (NXX) and/or multiple Directory Number (DN) ranges of the following types:

- single office code/single Directory Number (DN) range
- single office code/multiple DN ranges
- multiple office codes/single DN range
- multiple office codes/multiple DN ranges

NARS Multiple DID Office Code Screening operates within the following parameters:

- Only one Numbering Plan Area (NPA) per LOC is allowed.
- Ranges defined within a LOC must be unique. Overlapping or duplication of ranges is not permitted.
- The number of digits must be the same in each Direct Inward Dialing (DID) range.
- A maximum of 20 Direct Inward Dialing (DID) ranges may be defined per location code.

BARS/NARS Off-Net Number Recognition

Off-Net Number Recognition is an enhancement to the Basic/Network Alternate Route Selection (BARS/NARS) feature for ESN, and for the BARS feature for standalone applications.

Off-Net numbers that terminate at an ESN Node or Main, or at a Conventional Main, can be routed through the private network by means of TIE trunks. BARS/NARS Off-Net Number Recognition prevents unnecessary TO and FROM terminations through CO trunks, at the terminating end, when a caller dials a DID or Direct Distance Dialing (DDD) call to a location in the private network. Calls are handled on the basis of customer-defined parameters stored in Network Translation Tables and Supplementary Digit Recognition/Restriction Blocks.

Detailed information is provided in *Avaya Basic Network Feature Fundamentals, NN43001-579*.

BARS/NARS 11-Digit Translation

This feature expands the ESN BARS/NARS translation capabilities from a maximum of four digits to a maximum of 11 digits for route selection.

Possible conflicts between translatable codes (NPA, NXX, LOC, SPN) are eliminated by 11-Digit Translation. By allowing translation of more than four leading digits, unique nonconflicting routing to a destination is possible. More than one route list can exist for each specific code of a type. For example, the NXX 727 could only translate into one route list previously.

With 11-Digit Translation, up to 128 route lists for BARS and up to 256 for NARS may be defined, extending translation deeper into the dialed code. The codes must be leftwise unique. If an NXX of 7271 is defined, any other 727 entries must be extended to four digits.

BARS/NARS 11-Digit Translation is discussed in greater detail in *Avaya Basic Network Feature Fundamentals, NN43001-579*.

Network Authorization Code

The Network Authorization Code (NAUT) feature provides up to 50,000 authorization codes. Network Authorization Code (NAUT) incorporates all the features of the BAUT feature, adds a conditionally last option for entering an Authorization Code after dialing an ESN call, and enables the attendant to enter an Authorization Code. Network Authorization Code (NAUT) is described in detail in *Avaya Basic Network Feature Fundamentals, NN43001-579*.

Network Call Transfer

Network Call Transfer (NXFER) enhances the operation of Call Transfer (XFER) between two switches when a call is transferred back to the originating switch. The regular Call Transfer feature requires two TIE trunks to complete the call. With Network Call Transfer (NXFER), if the call is transferred back to the originating switch as the same TIE trunk group, the originating switch completes the transfer within itself and the TIE trunks are dropped. For a detailed description of Network Call Transfer (NXFER), see *Avaya Basic Network Feature Fundamentals, NN43001-579*. The benefits derived from the NXFER feature include:

- minimal use of access TIE lines
- improved transmission performance, since TIE lines are not used for the completed connection
- operation identical to that of Call Transfer (XFER)

NXFER operates within the following parameters:

- Meridian 1 proprietary telephones must be equipped with a Call Transfer key.
- Network Signaling (NSIG) must be provided on both switches.

Network Signaling

Network Signaling (NSIG) provides a proprietary signaling protocol for transmission of network call information between switches that operate in a private network environment with Basic/ Network Alternate Route Selection (BARS/NARS) or CDP. Network Signaling (NSIG) can be equipped at the Node and Main switches. For a detailed description of Network Signaling, see *Avaya Basic Network Feature Fundamentals, NN43001-579*.

NSIG supports transmission or reception of information between the following switch types:

- System Node to System Node
- System Node to System Main
- System Node to an Electronic TIE Network (ETN) switch
- System Main to System Node
- ETN switch to System Node

Information transmitted and received from one switch to another can include the following:

- call type
- called number
- Network Class of Service (NCOS)
- Traveling Class of Service (TCOS)
- Traveling Class Mark (TCM)
- queue identification number (for CCBQ)

NSIG operates within the following parameters:

- A Main can connect to only one Node, and both switches must be equipped with the NSIG feature.
- TIE trunks between Nodes and Mains must be arranged for Dual-tone Multifrequency (DTMF) sending/receiving and wink-start operation.
- System Node compatibility with Electronic TIE Network (ETN) switches is limited to seven-digit on-network and ten-digit off-network calls.

Network Traffic

The Network Traffic (NTRF) feature enables traffic data related to BARS, NARS, and CDP to be retrieved and output at a traffic TTY. The network traffic measurements (in addition to the switch traffic measurements) are described in detail in *Avaya Traffic Measurement: Formats and Outputs Reference, NN43001-750*.

Network Speed Call

Network Speed Call (NSC) enables a user who is normally restricted from making network calls to make such a call through BARS/NARS, provided that the destination is a number defined in a System Speed Call (SSC) list. When such a call is placed, the CLS and TGAR limitations are lifted and a Network Class of Service (NCOS), associated with the SSC list, is assigned for the duration of the call. NSC is described in detail in *Avaya Basic Network Feature Fundamentals, NN43001-579*.

Off Hook Queuing

Off Hook Queuing (OHQ) is an optional feature available at any switch equipped with BARS, NARS, or CDP. If all facilities are busy when an individual places a BARS, NARS, or CDP call, the OHQ feature enables the individual to wait off hook for a programmed length of time until a facility becomes available. OHQ is described in *Avaya Basic Network Feature Fundamentals, NN43001-579*.

Operating parameters

See the appropriate publication for each ESN feature.

Feature interactions

See *Avaya Electronic Switched Network: Signaling and Transmission, NN43001-280* for ESN feature interactions.

Feature packaging

Basic Authorization Code (BAUT) package 25 requires:

- Charge Account/Authorization Code (CAB) package 24.

Basic Alternate Route Selection (BARS) package 57 requires:

- Basic Routing (BRTE) package 14
- Network Class of Service (NCOS) package 32

Coordinated Dialing Plan (CDP) package 59 requires:

- Basic Routing (BRTE) package 14
- Network Class of Service (NCOS) package 32
- Flexible Call Back Queuing (FCBQ) package 61

Network Alternate Route Selection (NARS) package 58 requires:

- Basic Routing (BRTE) package 14
- Network Class of Service (NCOS) package 32

Network Authorization Code (NAUT) package 63 requires:

- Charge Account/Authorization Code (CAB) package 24
- Basic Authorization Code (BAUT) package 25

and at least one of the following:

- Basic Alternate Route Selection (BARS) package 57
- Network Alternate Route Selection (NARS) package 58 or
- Coordinated Dialing Plan (CDP) package 59

Network Call Transfer (NXFR) package 67 requires:

- Network Class of Service (NCOS) package 32
- Network Signaling (NSIG) package 37

Network Signaling (NSIG) package 37 requires:

- Network Class of Service (NCOS) package 32

Network Traffic (NTRF) package 29 requires at least one of the following:

- Basic Alternate Route Selection (BARS) package 57
- Network Alternate Route Selection (NARS) package 58
- Coordinated Dialing Plan (CDP) package 59
- Priority Queuing (PQUE) package 60
- Flexible Call Back Queuing (FCBQ) package 61, or
- Off Hook Queuing (OHQ) package 62

Network Speed Call (NSC) package 39 requires:

- System Speed Call package (SSC) package 34

and at least one of the following:

- Basic Alternate Route Selection (BARS) package 57, or
- Network Alternate Route Selection (NARS) package 58

Off Hook Queuing (OHQ) package 62 requires

- Basic Queuing (BQUE) package 28

and at least one of the following:

- Basic Alternate Route Selection (BARS) package 57, or
- Network Alternate Route Selection (NARS) package 58

Feature implementation

See *Avaya Electronic Switched Network: Signaling and Transmission, NN43001-280* for ESN implementation.

Feature operation

See *Avaya Electronic Switched Network: Signaling and Transmission, NN43001-280* for ESN operation.

Chapter 33: Emergency Services Access

Emergency Services Access (ESA) is a feature that places a customer in compliance with new federal legislation that requires the Private 911 type of functionality provided by ESA. Please note, however, that the ESA feature is also generally useful for users who are not subject to legislation, and is broad enough to be used in different countries. For example, it will be appreciated by any customer who wants to route emergency calls in a special manner, or who wants to be notified when a telephone user makes an emergency call. It would also appeal to a customer who wishes to have ESA calls answered on-site, on the business premises, rather than being forwarded to the Public Services Answering Point (PSAP).

See *Avaya Emergency Services Access Fundamentals, NN43001-613* for complete information.

Chapter 34: End of Selection

Contents

This section contains information on the following topics:

Feature description

This feature allows an End of Selection (EOS) signal to be sent back on a Direct Inward Dialing (DID) trunk to inform the Public Exchange/Central Office that the dialing phase of the call has been completed. The signal will be sent back when one of the following occurs:

- the DID call terminates on an idle station or attendant, an Automatic Call Distribution (ACD) queue, or a busy station
- the call has been intercepted (the DN is busy, not in service, or prohibited), and
- the interdigit timer has expired or an incomplete DN has been dialed.

Operating parameters

The Central Office must be equipped to handle the special signaling requirements associated with the End of Selection feature described above.

The End of Selection feature is available with either the QPC357 or NTD9447 pack for analog trunks, or the QPC536 pack for 2 Mbit digital trunks. It is not available on 1.5 Mbit digital trunks or Japanese DMI trunks.

If the DN size is specified, the End of Selection feature allows a trunk to be locked out if the correct number of digits are not received, or if termination has not been completed when the correct number of digits have been received.

The End of Selection signal is not supported by R2 Multifrequency.

Feature interactions

There are no feature interactions associated with this feature.

Feature packaging

International Supplementary Features (SUPP) package 131.

Feature implementation

Table 71: LD 16 - Create or modify data for trunk routes:

Prompt	Response	Description
...		
EOS	(NO) YES	End of Selection (EOS) signal is enabled; no EOS signal. EOS and BSY signals are enabled.

Feature operation

No specific operating procedures are required to use this feature.

Chapter 35: End of Selection Busy

Contents

This section contains information on the following topics:

Feature description

This feature can be used where there is a requirement for the system to send a busy signal to the Public Exchange/Central Office when the call terminates in a busy connection. The signal will be sent 500 to 900 milliseconds after the end of selection signal is sent and informs the Central Office to release the connection and return busy tone to the originating source.

Operating parameters

The Central Office must be equipped to handle the special signaling requirements associated with the End of Selection Busy feature described above.

The End of Selection Busy feature is only available on the NTD9447 or the QPC536 pack, and is not supported by R2 Multifrequency Compelled Signaling.

Feature interactions

There are no feature interactions associated with this feature.

Feature packaging

International Supplementary Features (SUPP) package 131.

Feature implementation

Table 72: LD 16 - Create or modify data for trunk routes.

Prompt	Response	Description
...		
EOS	BSY	End of Selection (EOS) and BSY signals are enabled.

Feature operation

No specific operating procedures are required to use this feature.

Chapter 36: End-of-Dialing on Direct Inward/Outward Dialing

Contents

This section contains information on the following topics:

Feature description

This feature monitors an outgoing Direct Inward Dialing (DID) or Direct Outward Dialing (DOD) call to determine whether additional digits are dialed after the route access code seizes the trunk. If no digits are dialed in 15 seconds, the trunk is disconnected.

Operating parameters

The Public Exchange/Central Office must be equipped to handle the special signaling requirements associated with the End-of-dialing on DID/DOD feature described above.

The End-of-Dialing on DID/DOD feature is not available on 1.5 Mbit digital trunks or Japanese Digital Multiplex Interface (DMI) trunks.

Feature interactions

There are no feature interactions associated with this feature.

Feature packaging

International Supplementary Features (SUPP) package 131.

Feature implementation

No change to existing configuration is required for the End-of-dialing on Direct Inward/Outward Dialing feature.

Feature operation

No specific operating procedures are required to use this feature.

Chapter 37: End-to-End Signaling

Contents

This section contains information on the following topics:

Feature description

The End-to-End Signaling (EES) feature enables a station to send Digitone end-to-end signaling through an established outgoing connection. EES provides fast reliable service and an optional feedback tone to the originator, as specified in LD 56. In addition, EES eliminates the use of a conference loop for sending EES tones to the connected parties.

To use EES, the following prompt or prompts need to be set in LD 15: EEST = NO (no feedback tone, default value) or EEST = YES, DTMF = NO (single optional feedback tone, as specified in LD 56).

An outgoing connection from a digital telephone is considered established after the end of dialing time is elapsed. Alternatively, an outgoing call can be established after the end of dialing time is elapsed, or can be established immediately by pressing an octothorpe (#) after the last digit is dialed.

Attendant End-to-End Signaling

The attendant can send DTMF tones to either the source or destination party using the AEES key on the attendant console. If there are two receiving parties on the current active loop key, the attendant can press the EXCL SRC or EXCL DEST key to exclude one of the connected parties before pressing the AEES key (defined in LD 12). Only one party on the active loop key (source or destination) can receive the DTMF signal. After pressing the AEES key, the attendant can press digits to send DTMF tones out to the source or destination party. To terminate the EES operation, the attendant should press the AEES key again. The states for the EXCL SRC, EXCL DEST, SRC loop, and DEST loop keys remain the same as before the EES key is pressed.

Operating parameters

The EES capability extends to internal analog (500/2500 type) telephone calls and incoming trunk calls.

A call must be established before using the EES feature. An outgoing call is considered established 14 seconds (DP trunk) or four seconds (2500-type telephone or Digitone trunk) after the last digit has been outpulsed. The length of this delay can be changed through service change. EES can be performed after end of dial time out, or when an answer supervision has been received from the far end, by pressing the octothorpe key (#) after the last digit.

EES is allowed only on CO, FEX, WATS, TIE, CCSA, DID, and CAMA trunk types.

EES is not available on analog (500/2500 type) telephones.

EES eliminates the use of the conference loop.

The AEES key, like other flexible programmable keys, cannot be configured on key 0 or key 1 of the attendant console.

There is a 5.4 dB difference between when EEST is set to YES (provide end-to-end signaling feedback tone) and when it is set to NO (provide no tone). An attenuation of 5.4 dB using the conference pads is applied to the EES tone if user feedback is to be given.

Feature interactions

End-to-End Signaling feature interactions

Agent/All Observe

In the Agent/All Observe mode, a supervisor, agent, and customer are all in a conference call. This feature uses Conference EES.

Attendant End-to-End Signaling

An attendant console in Attendant End-to-End Signaling mode can communicate with the source or destination party through in-band DTMF tones on an established speech path. The attendant console is treated like any other telephone.

Autodial Tandem Transfer

EES is used to send the Automatic Dialing (ADL) digits to the Public Exchange/Central Office (CO). With Autodial Tandem Transfer (ATX), the 911 agent can use the ADL key or manually dial the digits, or use a combination of both methods, to dial the third party's number. The ADL key can be pre-programmed with a prefix and the remaining digits can be dialed manually to distinguish between different numbers.

To get uniform feedback tone when using the ADL key along with manual dialing, set the DTMF prompt to NO in LD 15.

Call Modification

If EES is in progress, Call Modification is blocked. If Call Modification were not blocked, it might not be performed correctly during EES.

Call Detail Recording Record

An option in the Customer Data Block (LD 15) defines whether EES digits should be captured in the Call Detail Recording (CDR) record or not. This can prevent EES digits that contain sensitive information, like account numbers and passwords, from appearing in the CDR record.

Call Party Name Display

When entered after a call is answered, EES digits are displayed immediately following the CPND name of the connected party. Leading DN digits and name characters may be shifted out of the display window.

Conference End-to-End Signaling

Improved EES does not apply when the parties are in a conference call. In conference EES, a Tone and Digit Switch (TDS) loop is attached to the conference loop when a digit is pressed by one of the conferenced parties, and TDS is released when the digit is released. The setting of the EEST prompt determines whether the DTMF feedback tone is provided or not. The DTMF prompt is ignored for Conference EES.

EuroISDN Continuation

End-to-End Signaling is supported on all outgoing EuroISDN routes as soon as the CALL PROCEEDING message with a Progress Indicator is received.

EuroISDN Trunk - Network Side

EuroISDN Master Mode

End-to-End Signaling, which allows in-band dialing to be performed on ISDN trunks before and after the call has been answered, is supported on the EuroISDN Trunk - Network Side connectivity.

In the case of tandem with ISDN trunks, the necessary information to allow the End-to-End Signaling feature is tandemed to the ISDN trunk. At this point, it becomes the responsibility of the end user switch to provide the End-to-End Signaling service.

Multi-Party Operations - Three-Party Service

The party receiving the patience tone or the Misoperation ringback is not able to use EES.

Silent Observe

EES supports the Silent Observe feature of Automatic Call Distribution (ACD), like any other feature that involves EES between two telephones. A supervisor can use this ACD feature to silently observe an agent.

Stored Number Redial

End-to-End Signaling (EES) activates after a call to a trunk is established by expiration of the end-of-dial timer. Further digits dialed are not stored by the SNR feature once it is in EES mode.

Attendant Administration

While in the Attendant Administration mode, pressing the AEES key is ignored.

Attendant End-to-End Signaling feature interactions

Attendant Barge-In Attendant Busy Verify

While in the Barge-In/Busy Verify mode, the console cannot enter AEES mode.

Attendant Features

Activating Automatic Wake Up, Call Park, Charge Account, Calling Party Number, Hold, Release, or another loop key will terminate AEES operation.

Attendant Position Busy Centralized Attendant Service Night Service

These features work together with Attendant End-to-End Signaling (AEES). However, do not press one of these feature keys while using AEES, or the Dual-tone Multifrequency (DTMF) code signals may be blocked.

Attendant Supervisory Console

The supervisor can operate AEES if there is a call on the active loop key. An attendant in AEES mode can be monitored by the supervisor.

Conference

While in AEES mode, the receiving party cannot initiate a conference call.

End-to-End Signaling (station level)

The attendant console and the telephone receiving AEES cannot both activate EES simultaneously.

Interposition call

When an attendant is actively connected to another console using Interposition Attendant Call, AEES is blocked. During an Interposition Call Transfer, however, the console that is actively connected to a telephone can perform AEES, provided the party connected to the other attendant console is excluded.

Meridian Hospitality Voice Services - Digit Key

Attendant End-to-End Signaling and Digit Key are mutually exclusive. Being in AEES mode overrides the use of the Digit Key.

Trunk connection

On incoming ground start CO or Direct Inward Dialing (DID) trunks without Answer Supervision, you must press the Release (RLS) key on the console to exit AEES mode and drop the connection.

Feature packaging

End-to-End Signaling and Attendant End-to-End Signaling are both part of package 10 and have no feature package dependencies.

Feature implementation

Task summary list

The following is a summary of the tasks in this section:

1. Table 73: LD 15 - Enable End-to-End Signaling tone feedback on page 260

   Enable End-to-End Signaling tone feedback.
2. Table 74: LD 12 - Add End-to-End Signaling key to attendant console on page 261

   Add End-to-End Signaling key to attendant console.
3. Table 75: LD 56 - Specify the cadence for the EES feedback tone on page 261

   Specify the cadence for the EES feedback tone.

Table 73: LD 15 - Enable End-to-End Signaling tone feedback

Prompt	Response	Description
REQ:	CHG	Change.

Prompt	Response	Description
TYPE:	FTR	Features and Options.
CUST		Customer number
	0-99	Range for Large System and Avaya Communication Server 1000E (CS 1000E) system.
- EEST	(NO) YES	NO = No EES feedback tone is given to the telephone. YES = EES feedback tone is given; the type is defined by the DTMF prompt.
- DTMF	(NO) YES	NO = Use EES for single feedback tone. YES = Use EES for DTMF feedback tone.
...		
TYPE	CDR	CDR and charge account options.
- ECDR	(NO) YES	NO = Do not capture EES digits in the CDR record. YES = Capture EES digits in the CDR record.

Table 74: LD 12 - Add End-to-End Signaling key to attendant console

Prompt	Response	Description
REQ	CHG	Change.
TYPE	2250	Attendant console type.
TN		Terminal number
	l s c u	Format for Large System, Avaya MG 1000B , and CS 1000E system, where l = loop, s = shelf, c = card, u = unit.
KEY	xx EES	Add EES key (xx = key number) (cannot be key 0 or 1).

Table 75: LD 56 - Specify the cadence for the EES feedback tone

Prompt	Response	Description
REQ	CHG NEW	Change, or add.
TYPE	FTC	Flexible Tones and Cadences.
TABL	x	FTC table number.
HCCT	YES	Hardware Controlled Cadence.
EEST		No response expected; this is an informational prompt.
- TDSH	i bb cc tt	TDS external, burst, cadence, and tone.
- XTON	0-255	NT8D17 TDS tone code.

Prompt	Response	Description
- XCAD	0-255	NT8D17 cadence code for FCAD.

Feature operation

No specific operating procedures are required to use this feature.

Chapter 38: End-to-End Signaling Display Enhancement

Contents

This section contains information on the following topics:

Feature description

The End-to-End Signaling Display Enhancement (EESDSP) feature enhances the existing End-to-End Signaling (EES) feature. EES digits can communicate private information such as account numbers, authorization codes, and passwords. In some environments, showing this information can be a security issue. EESDSP feature provides the option to show or block the EES digits from appearing on a set's display screen. The customer can enable or disable this option at the EES Digit Display (EESD) prompt in the Customer Data Block.

With the EESDSP feature enabled, the user's display shows all the EES digits as dialed. EES digits display when you enter them after a call is answered. The digits appear following the Call Party Name Display (CPND) name of the connected party. Initial digits and name characters may move out of the display window if necessary. With the EESDSP feature disabled, the user's display does not change, keeping the established call information.

Operating parameters

The EES feature must be enabled for the EESDSP feature to function.

The EESDSP feature applies only to the EES digit display functionality of the existing EES features. The EES digits are not displayed on the sets of the other parties in an established call.

The EESDSP feature does not apply to a networking environment.

The EESDSP feature applies to proprietary sets, Basic Rate Interface (BRI) sets, and attendant consoles with a display screen enabled to show entered EES digits and EES capabilities.

Attendant consoles require Attendant EES (AEES), which is enabled by configuring and using the programmable AEES key.

Feature interactions

The EESDSP feature does not change the production of tones for EES digits, or the processing or sending of EES digits. This feature only gives the customer the option to show or block all EES digits on the display.

Attendant End-to-End Signaling

For Attendant End-to-End Signaling (AEES), place the attendant console in EES mode by pressing the AEES key. When in EES mode, you can dial EES digits. The attendant console can send the EES Dual-tone Multifrequency (DTMF) tones to either the source or destination party.

When the End-to-End Signaling Display Enhancement option is enabled, the attendant console display shows the EES digits entered while in the EES mode. The digits appear on the second line of the attendant console's display. If disabled, the attendant console display does not change.

Call Party Name Display (CPND)

With the EESDSP option enabled, EES digits appear after the Call Party Name Display (CPND) name of the connected party. Initial digits and name characters may move out of the display window if necessary.

With the EESDSP option disabled, the set display does not change from the established CPND display.

Conference End-to-End Signaling

The EESDSP option changes the display of the EES digits as dialed for all the EES features, including Conference EES.

End-to-End Signaling

The EESDSP option has no effect on the digits dialed before the system is in EES mode. In EES mode, digits dialed from a set with a digital display appear on the display when the EESDSP option is enabled. When you disable the EESDSP option, the display does not show the dialed EES digits.

Improved End-to-End Signaling

The EESDSP feature changes the display of EES digits the same for both Improved End-to-End Signaling (IEES) and EES.

Feature packaging

The End-to-End Signaling Display Enhancement (EESDSP) feature requires End-to-End Signaling (EES) package 10.

Feature implementation

Table 76: LD 15 - Enable the End-to-End Signaling Display Enhancement feature.

Prompt	Response	Description
REQ:	CHG	Change existing data.
TYPE:	FTR	Customer Features and options.
CUST		Customer number
	0-99	Range for Large System and Avaya Communication Server 1000E (Avaya CS 1000E) system.

Prompt	Response	Description
....	....	
EEST	(NO) YES	EES Tone to originating party. Do not send feedback to the originator. Send feedback tone to the originator. Enhanced EES signaling is provided when EEST=YES and DTMF=NO.
- DTMF	(YES) NO	EES feedback tone. EES for DTMF feedback tone. EES for single tone feedback (only prompted if EEST=YES).
EESD	(NO) YES	EES digit display. Do not display the EES digits. Display all EES digits.
TTBL	(0)–31	Tone Table number.
....	....	

Feature operation

No specific operating procedures are required to use this feature.

Chapter 39: Equal Access Compliance

Contents

This section contains information on the following topics:

Feature description

A telephone user can select any interexchange carrier for any given call by using a Carrier Access Code (CAC). A CAC comprises an Equal Access identifier and a Carrier Identification Code (CIC). Avaya refers to a call preceded by a CAC as an Equal Access call.

Federal Communications Commission (FCC) requirements

FCC Part 68 regulations require that any equipment or software manufactured or imported on or after April 17, 1992, and installed by any aggregator, must allow all users to use Equal Access codes to selectively access the long distance carrier of their choice. As defined in FCC docket 90-313, an aggregator is any business that, in the ordinary course of operations, makes telephones available to the public, or to transient users of the premises, for interstate telephone calls using a provider of operator services. Aggregators include hotels or motels, hospitals, universities, airports, gas stations, or pay telephone owners.

Aggregators, although they must allow callers access to any long distance caller, are permitted to block calls selectively. Selective equal access lets aggregators choose to block direct-dialed calls that result in charges to the originating telephone. Aggregators cannot block operator-assisted calls.

Avaya complies with the FCC Equal Access rules in dockets 90-313, 91-35, and their appendixes.

Equal Access dialing plans

The system software supports Equal Access dialing plans as follows:

- It allows operator-assisted North American and international dialing.
  - CAC + 0
  - CAC + 0 + (NPA) + NXX + XXXX, and
  - CAC + 01 + CC + NN.
- It allows or denies direct North American and international dialing.
  - CAC + 1 + (NPA) + NXX + XXXX, and
  - CAC + 011 + CC + NN.

Legend:

CAC = Carrier Access Code (101XXXX)

NPA = Numbering Plan Area (area code in the North American Numbering Plan)

NXX = end-office code (N = any digit except 0 or 1; X = any digit (0–9))

XXXX = any four digits

CC = Country Code

NN = National Number

Route types

Equal Access Compliance supports COT, FEX, WAT, DID, and TIE routes.

A TIE route is supported only if standard signaling is specified in LD 16 (SIGO = STD). To enable Equal Access call limitations to function properly, Digital Trunk Interface (DTI) TIE routes must be voice only. (DTI TIE routes configured as voice/data are not supported for connection to a Public Exchange/Central Office.) TIE routes must be either outgoing or incoming/outgoing (ICOG = IAO or OGT).

Call restriction

Call restriction relies on fixed pattern recognition to determine which calls can be denied. Switch administrators can restrict two kinds of direct-dialed Equal Access calls: North American calls with the 101XXXX+1+NPA+ NXX+XXXX format and international calls with the 101XXXX +011+CC +NN format. If either restriction option is chosen, the administration must verify that the Original Carrier Access Code (OCAC) flag is correctly set.

Call limitations do not affect attendant calls.

Calls blocked by Equal Access are not directed to alternate routes.

BARS/NARS routing

Equal Access determines limitations without looking at a call's originating type (ESN or Direct Access). Routing has no effect on Equal Access call restriction: calls receive the same restriction treatment whether they originate from a trunk access code or from BARS/NARS. Equal Access is not a BARS/NARS feature and does not require BARS/NARS dialing.

To configure BARS/NARS to route Equal Access calls, simply use a special number (SPN) of 10 (the Equal Access code) to identify the calls as Equal Access calls and route them accordingly.

Example

Configure BARS/NARS for Equal Access call routing, assuming that calls originate from Customer 0 and go out over Route 10. To route Equal Access calls originating from Customer 0 over Route 10, using route list index 100 and access code 1 (AC1).

Operating parameters

There are no operating parameters associated with this feature.

Feature interactions

There are no feature interactions associated with this feature.

Feature packaging

Equal Access compliance is included in base system software. Network Class of Service (NCOS) package 32 is required to configure Equal Access.

Feature implementation

The configuration in this example routes all Equal Access calls placed through BARS/NARS with access code 1 (AC1) over route 10. Set the SPN to "101".

In this example set Equal Access toll calls for NCOS = 4. Note that Equal Access toll calls placed through direct trunk access to route 10 also will be blocked.

Task summary list

The following is a summary of the tasks in this section:

1. Table 77: LD 17 - Set OCAC as appropriate. on page 271

   Set OCAC as appropriate.
2. Table 78: LD 86 - Set the route list index to Route 10. on page 271

   Set the route list index to Route 10.
3. Table 79: LD 90 - Establish an SPN for the Equal Access code. on page 272

   Establish an SPN for the Equal Access code.
4. Table 80: LD 87 - Configure a NCOS for Equal Access. on page 272

   Configure a NCOS for Equal Access.
5. Table 81: LD 10 - Assign a NCOS to an Analog Telephone. on page 272

   Assign a NCOS to an Analog Telephone.
6. Table 82: LD 11 - Assign a NCOS to a Digital Telephone. on page 272

   Assign a NCOS to a Digital Telephone.
7. Table 83: LD 16 - Enable Equal Access for this route. on page 273

   Enable Equal Access for this route.

Carrier Identification Code Expansion supports and extends the General Carrier Restriction method of blocking calls. Given the expansion in the number of Carrier Identification Codes (CIC), it is no longer practical to support Selective Carrier Restriction functionality. Carrier Identification Code Expansion continues to provide the selective blocking function required by the FCC; Avaya and the FCC interpret the term "selective" differently. For these reasons, prompts pertaining to General Carrier Restriction and Selective Carrier Restriction in LD 16 no longer appear.

Customers who chose the ITOL prompt in LD 16 to block international calls should also have international calls blocked at the Public Exchange/Central Office to reduce the likelihood of unauthorized international calls. The carrier or Central Office operator intercept interdigit timer typically expires in four to six seconds. The system end of dial timers, End-of Dial Timer for non-Digitone Trunks (EOD) and End-of Dial-Timer for Digitone Trunks (ODT), are defaulted to 14 and four seconds respectively. ODT can be raised to seven seconds to prevent Digitone stations from bypassing Equal Access limitations of Digital Distance Dialing international calls.

The interdigit timeout for non-leftwise-unique prefixes 0 and 01 is fixed for a given carrier network. Therefore, Equal Access connects the call to the Central Office trunk if the user dials Carrier Access Code + 0 and allows the end-of-dialing timer to expire. Equal Access blocks the same call if the caller presses the octothorpe (#) key and cancels the EOD or ODT. The caller cannot bypass the EQAR prompt in LD 16 provided that the EOD and ODT are set long enough to exceed the inter-digit timeout on the carrier networks.

Before and during the permissive period, when both the three-character and the four-character CIC are allowed, current Equal Access users must set the Original Carrier Access Code (OCAC) flag to YES in LD 17. OCAC should be set to NO (default).

New Equal Access customers do not need to change the OCAC flag until the feature is configured.

Table 77: LD 17 - Set OCAC as appropriate.

Prompt	Response	Description
REQ	CHG	Change existing route data.
TYPE	CFN PARM	Configuration Record. System parameters.
PARM	YES	Change system parameters.
- NDRG	(NO) YES	(Disable) enable new distinctive ringing.
- OCAC	(NO) YES	Support original CAC format (must be set to YES during interim period, NO following interim period).

Table 78: LD 86 - Set the route list index to Route 10.

Prompt	Response	Description
REQ	NEW CHG	Create, or change database.
CUST	xx	Customer number, as defined in LD 15
FEAT	RLB	Route List Block.
RLI	100	Use route list index 100 to route Equal Access calls.
ENTR	0	Route entry number for this route list index (0 if this is the first entry).
ROUT	10	Send Equal Access calls over Route 10.

Table 79: LD 90 - Establish an SPN for the Equal Access code.

Prompt	Response	Description
REQ	NEW	New ESN translation table entry.
CUST	xx	Customer number, as defined in LD 15
FEAT	NET	Network translation table entry.
TRAN	AC1	Access code 1 is used to originate the Equal Access calls.
TYPE	SPN	SPN translation entry.
SPN	101	SPN (Equal Access code).
RLI	100	Use route list index 100 to route Equal Access calls.

Table 80: LD 87 - Configure a NCOS for Equal Access.

Prompt	Response	Description
REQ	CHG	Change NCTL data.
CUST	xx	Customer number, as defined in LD 15
FEAT	NCTL	Change NCTL block.
NCOS	4	Network Class of Service group number.
EQA	YES	This NCOS permits Equal Access call restriction capabilities.

Table 81: LD 10 - Assign a NCOS to an Analog Telephone.

Prompt	Response	Description
REQ:	CHG	Change existing set data.
TYPE:	aaa	Specify set type.
TN		Terminal number
	l s c u	Format for Large System, Avaya CS 1000 Media Gateway 1000B (Avaya MG 1000B), and Avaya Communication Server 1000E (Avaya CS 1000E) system, where l = loop, s = shelf, c = card, u = unit.
NCOS	4	Network Class of Service group number.

Table 82: LD 11 - Assign a NCOS to a Digital Telephone.

Prompt	Response	Description
REQ:	CHG	Change existing set data.

Prompt	Response	Description
TYPE:	a...a	Telephone type. Type ? for a list of possible responses.
TN		Terminal number
	l s c u	Format for Large System, Avaya MG 1000B, and CS 1000E system, where l = loop, s = shelf, c = card, u = unit.
NCOS	4	Network Class of Service group number.

Table 83: LD 16 - Enable Equal Access for this route.

Prompt	Response	Description
REQ	CHG	Change existing route data.
TYPE	RDB	Change Route Data Block.
CUST	xx	Customer number, as defined in LD 15
ROUT	10	
EQAR	(NO) YES	Enter YES to enable Equal Access and selective blocking for this route. A YES response triggers the next two prompts.
- NTOL	(DENY) ALOW	Specify that Equal Access North American calls billed to originating telephone are to be denied.
- ITOL	(DENY) ALOW	Specify that Equal Access international calls billed to originating telephone are to be denied.

Feature operation

No specific operating procedures are required to use this feature.

Chapter 40: Extended DID/DOD Software Support - Europe

Contents

This section contains information on the following topics:

Feature description

This feature provides software support for the European Extended Direct Inward Dialing (XDID)/Direct Outward Dialing (DOD) cards. These cards are the NT5K36AA (German XDID pack), NT5K84AA (Swiss XDID pack) and NTAG04AA (Dutch XCOT/DID). The new packs enable the system to have the following Intelligent Peripheral Equipment (IPE) DID/DOD functionalities.

Seizure acknowledgment on outgoing traffic

In order to provide this functionality, LD 14 has to be modified in order to allow Ear and Mouth (E&M) signaling to be configured for DID trunk on an XDID card. A new "Trunk Type and Signaling" in the type 2 Channel Download message defined for DID-E&M has to be downloaded onto the firmware. LD 14 must have a configuration of ACWK = YES, and LD 16 must have a configuration of "trunk type" (TYPE) = DID, "signaling" (SIGL) = EAM, and "start arrangement on outgoing" (STRO) = IMM.

End of dialing on DOD

No software changes are required to provide this functionality.

Interdigit timer on DID

To provide this functionality, the Partial Dial (PRDL) prompt has to be configured as BSY or YES.

End-of-selection signal on DID

To provide this functionality, a new outgoing SSD message, "End of Signaling", has been defined. The End-of-selection (EOS) prompt in LD 16 has to be configured to BSY or YES.

End-of-selection busy signal on DID

To provide this functionality, a new Outgoing SSD message "End of Signaling Busy", has been defined. The End of Selection (EOS) prompt in LD 16 has to be configured to BSY.

Provision of busy tone, ringback tone, and overflow tone for DID callers

No software changes are required to provide this functionality.

Restricted/unrestricted DID Class of Service for DID calls

No software changes are required to provide this functionality.

DID to TIE connection, subject to configured trunk barring and Class of Service limitations

To provide this functionality, the DITI prompt in LD 15 has to be configured to YES.

Line Break Alarm

To provide this functionality, an incoming SSD message, BAR, has been defined to trigger the Trunk Failure Monitor feature whenever a problem situation arises on the line. A new SSD message, UNBAR, has been defined to clear the problem indications provided by the Trunk Monitor feature. LD 14 has to be configured with a Class of Service of trunk barring allowed (BARA) or denied (BARD). This Class of Service is downloaded onto the XDID/DOD cards.

Static loss pad

One of two loss pads (either long or short) can be selected on a per trunk basis. To provide this functionality, LD 14 has to be configured with a Class of Service of either SHL (short line) or LOL (long line). The configured pad type is downloaded onto the XDID/DOD cards.

Disconnect supervision

To support this functionality, the software has been changed so that an XDID card can provide disconnect supervision for a DID trunk with Ear and Mouth (E&M) signaling. The software has also been changed to refrain from sending an End of Selection (EOS) signal when an incoming trunk call is being disconnected.

DID digit collection type

To support this functionality, the type of incoming DID digit collection is configured against a Class of Service and downloaded to the XDID card.

Unsupported Class of Service

If an attempt is made to download an unsupported configuration during regular enabling of the pack or during audit, the pack responds with a problem report type 3 message. The error message ERR5327 is printed out on the TTY and the trunk is disabled.

Incoming Digit collection

This functionality only applies to Dual-tone Multifrequency (DTMF) DID trunks. The software must be ready to accept incoming digits regardless of whether or not an "Enable Digit Collection" message is sent. To support this functionality, the trunk must be configured with an incoming start arrangement (prompt STRI = IMM in LD 14). Message H0019 is sent when a Digitone Receiver (DTR) signal is found.

Proceed to Send message to the firmware

A "Proceed to Send" message must be sent to the firmware in cases of non dial pulse trunks, as soon as the software is able to receive digits. To support this functionality, LD 14 must be configured with DTCR = YES. A new H0019 message is sent when a Digitone Receiver (DTR) signal is found for Dual-tone Multifrequency (DTMF) signaling, or when a Multifrequency Compelled (MFC) sender/receiver is found for Multifrequency Compelled (MFC) signaling. If a DTR signal is not found, the call is released.

PPM and Buffered PPM downloadable on a per country basis

To support this functionality, Periodic Pulse Metering and Buffered Periodic Pulse Metering (PPM) are enabled on a per trunk basis, rather than on per card basis. Configuration of PPM and Buffered PPM is still done on a per route basis.

Audit conflict reporting and PPM event reporting

To support this functionality, a channel and card parameter download audit is performed during initialization and when LD 30 is run as a midnight routine. This is to ensure that the software configuration is the same as the configuration stored in the hardware. If a discrepancy is detected, the software information is stored in the hardware and an error message is printed on the TTY. Also, for PPM recording, two new type 5 messages have been defined to report hardware problems. These are the TRK Event: Partial Metering Detection Failure message and the TRK Event: Fatal Metering Detection Failure message.

On partial PPM failure, a TRK516 error message is printed on the TTY. If PPM is configured, CDR records for any calls in progress may be incorrect. If Busy Tone Supervision is configured, busy tone may not have been detected for calls in progress. On fatal PPM failure, a TRK517 error message is printed on the TTY. If PPM is configured, further PPM reporting is disabled until the pack is either disabled and then reenabled, or removed and then reinserted. The CDR record for any call in progress is incorrect. If Busy Tone Supervision is configured, tone supervision can no longer be performed until the pack is either disabled and then reenabled, or replaced.

Network DID and Enhanced Night Service groups on DID

No software changes are required to support these functionalities on the XDID/DOD cards.

Held call clearing

No software changes are required to support this functionality on the XDID/DOD cards.

Unequipped channel notification

To support this functionality, a channel download message is sent to the XDID pack whenever a trunk on the pack is removed.

Call blocking

Before disabling a trunk, the software requires confirmation that the trunk is in the idle state. To support this functionality, the software disable sequence has been modified. The software waits for an idle state message from the XDID pack before sending a disable message to the trunk. If the idle message is not received before the disconnect supervision (DSI) timer expires, the software prints the TRK136 (Release Failure on the Unit) error message. The trunk is placed in lockout state. If the disable sequence was started from an overlay, a TRK520 (No Far End Release) error message is printed. The trunk remains in lockout until a Far End Release message is received on the pack.

Number Reception message

This is a Dutch Central Office (CO) requirement. When sufficient digits are received at the Dutch CO, the battery is reversed. When the Dutch COT/DID pack (NTAG04AA) detects this reversal, it sends a Number Reception message. This functionality is a software enhancement.

When an ENLC command is performed on an XDID/E&M card, the card is first reset and then messages are downloaded to the firmware to reflect the software trunk state. This prevents the software database from being in conflict with the firmware database. If an XDID/E&M card unit is in busy state, the SSD message H.A004 is printed. If the unit is in barred state, the SSD message H.A003 is printed.

Operating parameters

There are no operating parameters associated with this feature.

Feature interactions

DID/DOD

This feature provides the same feature interactions as the following DID/DOD features:

- End of Selection, End of Selection Busy
- Provision of Tones
- Selectable DN Size
- Partial Dial Timing
- Seizure Acknowledgment
- DID Restricted Class of Service
- DID to TIE Connection, and
- Enhanced Night Service.

Japanese DID trunk

For Japanese DID trunk support, DID to TIE (DTOT) package 176 must be removed due to tariff limitations.

Federal Communications Commission (FCC) Compliance for DID Answer Supervision

If FCC Compliance for DID Answer Supervision (FC68) package 223 is configured on XDOD units, it may lead to incorrect call status. Therefore, equipping this package is not recommended.

Trunk Failure Monitor

When a BAR message indicating a problem situation is received, a TRK501 message is printed on the TTY, the uppermost key lamps light up on the attendant console, and the trunk is placed in the BUSY state to prevent the trunk from being seized for new outgoing calls. The reception of an UNBAR message indicates that the problem situation has been cleared. A TRK502 message is printed on the TTY, the lamps on the attendant console are darkened, and the trunk is idled.

BARA CLS must be configured on the XDID trunk for the described process to occur.

XDID/DOD and XFCOT

Software support for European XDID/DOD cards and software support for European XFCOT cards provide similar functionality in the following areas:

- Trunk Failure Monitor processing
- Downloading of PPM information
- Configuration and downloading of static pad setting for short line and long line, and
- Configuration download processing. Fields that are not filled due to configuration limitations are left blank and are not validated or interpreted by the firmware. The fields are treated as unused fields.

The DTCR (Digit Collection Ready) prompt has replaced the DTRA (Digitone Receiver Attached) prompt in LD 14.

Feature packaging

The Extended DID/DOD Software Support feature requires the following packages:

- Meridian 1 Superloop Administration (XCT1) package 205, which has the following requirements:
    - Intelligent Peripheral Equipment (XPE) package 203
    - International Supplementary Features (SUPP) package 131
    - ISDN Supplementary Features (ISDNS) package 161
    - PPM/Message Registration (MR) package 101

- Trunk Failure Monitor (TFM) package 182.

Feature implementation

Task summary list

The following is a summary of the tasks in this section:

1. Table 84: LD 15 - In the Customer Data Block, allow DID to TIE connections. on page 282

   In the Customer Data Block, allow DID to TIE connections.
2. Table 85: LD 16 - Define a DID/DOD trunk route for Germany and Switzerland. on page 282

   Define a DID/DOD trunk route for Germany and Switzerland.
3. Table 86: LD 14 - Define an XDID card unit. on page 283

   Define an XDID card unit.

Table 84: LD 15 - In the Customer Data Block, allow DID to TIE connections.

Prompt	Response	Description
REQ:	NEW CHG	Add or change.
TYPE:	NET	ISDN and ESN Networking options
...		
- DITI	YES	DID to TIE connections are allowed.

Table 85: LD 16 - Define a DID/DOD trunk route for Germany and Switzerland.

Prompt	Response	Description
REQ	NEW CHG	Add new data, or change existing data.
TYPE	RDB	Route Data Block
CUST	xx	Customer number, as defined in LD 15
ROUT		Route number
	0-511	Range for Large System and Avaya Communication Server 1000E (Avaya CS 1000E) system.
TKTP	DID	Trunk Type Direct Inward Dial

Prompt	Response	Description
ICOG	IAO	Incoming Outgoing trunk Incoming And Outgoing
ACOD	xxxxxxx	Access Code for the trunk route
CNTL	YES	Change Controls or timers
TIMR		Trunk Timers
	EOD 10112	End-of-Dial timer
	GTI 128	Incoming Guard timer
	GTO 128	Outgoing Guard timer
	ICF 0	Incoming Flash timer
	OGF 0	Outgoing Flash timer
	DSI 360000	Disconnect Supervision timer
NEDC	ETH	Near End Disconnect Control Either end control
FEDC	ETH	Far End Disconnect Control Either end control
...		
MR	PPM XLD	Message Registration Buffered PPM signals to be counted on this route M&MM Lead non-buffered is used.
PRDL	BSY	Partial Dial timing Busy signal is sent on timeout
EOS	BSY	End Of Selection signal EOS and busy signals are enabled
ACKW	(NO) YES	Acknowledgement seizure signal
BTT	100	Busy Tone Time

Table 86: LD 14 - Define an XDID card unit.

Prompt	Response	Description
REQ	NEW CHG	Create a New Data Block. Change an existing Data Block.
TYPE	DID	Direct Inward Dial trunk data block.
...		
XTRK	XDID	Extended (Intelligent Peripheral Equipment [IPE]) Direct Inward Dialing trunk.
...		
SIGL	EAM	Ear And Mouth (E&M) signaling (note that this prompt uses the letter "A", instead of the "&" which is more commonly used in the abbreviation of Ear and Mouth).

Prompt	Response	Description
STRI	IMM	Immediate Start arrangement Incoming.
STRO	IMM	Immediate Start arrangement Outgoing.
...		
CLS		Class of Service. The Class of Service parameters to be downloaded onto the XDID card unit.
	(LOL) SHL	Enter (LOL for long line) or SHL for short line static loss pad selection.
	(BARD) BARA	Barring (Denied) Allowed.
...		
DTCR	(NO) YES	Digit Collection Ready. Incoming digit collection ready; (do not) send acknowledgment when digit collection resources (DTR, MFC sender/receiver) are ready and attached.

Feature operation

No specific operating procedures are required to use this feature.

Chapter 41: Extended Flexible Central Office Trunk Software Support

Contents

This section contains information on the following topics:

Feature description

This feature provides software support for the following Extended Flexible Central Office Trunk (XFCOT) cards to meet the requirements of the following countries:

- NT5K70AA (German 8D)
- NT5K71AA (German 4D)
- NT5K82AA (Swiss)
- NT5K90AA (Danish PPM)
- NT5K90BA (Danish non-PPM)
- NT5K93AA (Norwegian PPM)
- NT5K93BA (Norwegian non-PPM)
- NTAG03AA (Dutch COT)
- NTAG04AA (Dutch DID/COT)
- NT5K18BA (New Zealand)

- NT5K99AA (Spanish PPM), and
- NT5K99BA (Spanish non-PPM).

The NT5K18AA (UK XFCOT) is not affected by the software changes introduced to support the XFCOT packs.

The following supervision, based on loop start signaling, is supported:

Battery Supervision Central Office Trunk (COT)

A battery supervised COT uses polarity detectors to provide seize, answer and disconnect supervision on all outgoing calls, and disconnect supervision on incoming calls. The supervision is performed by reversing the polarity from the Public Switched Telephone Network (PSTN) line. The battery supervised COT is configured in LD 14 with BAT.

ARF Supervision Central Office Trunk

ARF is an Ericson type series Public Exchange which provides disconnect supervision on both incoming and outgoing loop start Central Office trunk calls; on outgoing calls, seize supervision is also provided. Supervision is based on battery reversal detection. The signaling used to provide this supervision is called ARF signaling. The ARF supervised COT is configured in LD 14 with ARF.

Tone Supervised Central Office Trunk with downloadable Busy Tone parameters

A tone supervised COT has a busy tone detector on each unit. Busy tone is provided by the PSTN when the far end releases from outgoing and incoming trunks. The tone supervised COT is configured in LD 14 with BTS. This tone supervision depends on the busy tone frequency and cadence characteristics, as configured on a card basis using the Busy Tone ID (BTID) prompt in LD 14.

Loop Break Supervised Central Office Trunk

This type of signaling provides disconnect supervision by detecting a calibrated battery removal from the PSTN. The loop break COT supervision is configured in LD 14 with LBS.

Unsupervised Central Office Trunk

An unsupervised COT has neither polarity, battery, nor busy tone detector. Thus, no answer or disconnect supervision is provided for incoming or outgoing calls. A trunk is configured as unsupervised in LD 14 using other than BAT, LBS, ARF, or BTS.

Autoguard

Autoguard provides seize supervision on outgoing trunk calls. Autoguard is configured in LD 14 with SEIZ = YES.

Extended Flexible Central Office Trunk Software Support also provides the following capabilities:

- Trunk Barring. The XFCOT card can detect signaling from the PSTN that a trunk is barred, and that any call on the trunk must be dropped. The trunk unit is then marked software busy (busy barred) so that no outgoing calls may be made. A TRK514 message is printed on the TTY. A STAT (status) command in LD 32 or 36 yields a "Busy Barred" status. When the PSTN signals that the trunk unit may be unbarred, the software idles the trunk unit and a TRK515 message is printed on the TTY. Barring is configured on a per unit basis in LD 14 against a CLS of BARA. The BARA CLS is downloaded onto the XFCOT card.
- Static Loss Pad Selection. Trunk pad selection controls transmission loss. A pad may be inserted within or outside an XFCOT trunk card to allow a call to terminate on a station or another trunk. Two pad types are available to support long line or short line. The pad types are configured in LD 14 on a per unit basis, against a Class of Service of SHL for short line or a Class of Service of LOL for long line. The SHL or LOL is downloaded onto the XFCOT card.
- Enabling and disabling of Periodic Pulse Metering (PPM). The user configures PPM on a per route basis; the software configures the trunk on a per unit basis.
- Enabling and disabling of Buffered Periodic Pulse Metering (PPM), on a per trunk basis.
- A PPM ID that designates PPM parameters. This is configured in LD 14 against the PPID prompt. This value is downloaded onto the XFCOT firmware so that the appropriate PPM parameter may be selected.
- A four-unit card. The NT5K71AA four-unit quad density card has been introduced to meet German requirements.
- Mixed Central Office Trunk and Direct Inward Dialing on the same XFCOT card. In LD 14, the XFCOT card may be configured as being either COT or DID.
- ALS signaling. ALS, available only on the NTAG04AA (Dutch DID/COT) unit, is combined COT/DID signaling. Additions to the ground start signaling have been added for XFCOT support. On the near end, a partial release message is sent instead of a full release message. On an outgoing call, the number reception is accepted and interpreted by the

software. Number reception is a battery reversal signaling from the CO, indicating that it has received sufficient dialing information. The ALS signaling type is configured in LD 14 against the SIGL prompt.

- Balance impedance adjustment. It is possible to download the balance and termination impedance configured by a craftsperson for a NT5K90AA (Danish PPM) or NT5K90BA (Danish non-PPM) unit. The termination impedance is defaulted to value of 600 ohms. The balance impedance may be configured in LD 14 using the BIMP prompt, as 600 ohms or 3COM (three-component).
- Flash hook signaling, for features requiring a flash hook operation. The flash hook signal instructs a pack to send a flash hook signal to the PSTN. The features that require a flash hook are Malicious Call Trace and Centrex Switchhook Flash.

Error reporting and auditing is also provided. New problem reports are defined so that the XFCOT card can notify the software when the dialing speeds or companding laws are not supported by the hardware. If these new error reports are received from the XFCOT card, an error message is printed on the TTY. A channel and card parameter download audit is performed during initialization and when LD 30 is run as a midnight routine to ensure that the software configuration is the same as the configuration stored in the hardware. If a discrepancy is detected, the software information is stored in the hardware, and an error message is printed on the TTY.

For PPM recording, two type 5 messages have been defined to report hardware problems. These are the TRK Event: Partial Metering Detection Failure message, and the TRK Event: Fatal Metering Detection Failure message. Also, a type 12 channel configuration message and a type 13 channel audit configuration message have been introduced. The type 12 message provides the hardware with certain card configuration information, so that the card may be able to inform the software when certain configurations are not supported on the pack, and perform message filtering based on the software configuration. The type 13 message provides configuration download messages during the midnight routine.

The following table summarizes the downloaded software configurations that each XFCOT card supports.

Table 87: Downloaded configurations for XFCOT cards

XFCOT card	Hardware I.D. supported	Signaling supported	Downloaded SUPN supported	Periodic Pulse Metering (PPM)
NT5K18AA	01, 13, 14	COT (GRD, LGR, LDC)	SUPN	per pack
NT5K16BA	00, 01	COT (LOP, GRD)	BTS	per pack & unit
NT5K70AA	00	COT (LOP)	BTS	per unit
NT5K71AA	00	COT (LOP)	BTS	per unit
NT5K82AA	00	COT (LOP)	BTS, LBS, BAR	per pack & unit
NT5K90AA	00	COT (LOP)	BTS, ARF	per unit

XFCOT card	Hardware I.D. supported	Signaling supported	Downloaded SUPN supported	Periodic Pulse Metering (PPM)
NT5K90BA	00	COT (LOP)		none
NT5K93AA	00	COT (LOP)	BTS	per pack & unit
NT5K93BA	00	COT (LOP)	BTS	none
NT5K99AA	00	COT (LOP)	BTS	per unit
NTAG03AA	00	COT (LOP)	BTS	per pack & unit
NTAG04AA	26, 27	COT (ALS) DID (EAM)		per pack & unit

Operating parameters

The flash hook implementation for the Centrex Switchhook Flash feature does not provide flexible timing, as is provided by non-XFCOT packs. The timing is hard-coded onto the pack at 90 milliseconds.

The new XFCOT trunks cannot support the PPM frequency characteristics, configured as the PPM ID, for each trunk. The PPM ID is configured for the first trunk configured for the pack, and cannot be changed unless all trunks are removed from the pack and then reconfigured. The same limitations apply to the busy tone indication ID.

Only static pad selection is supported on the new XFCOT cards. Pad selection on a per call or per event basis is not supported.

Loop Start Supervisory Trunks and Japanese Central Office Trunks are not supported on the new XFCOT cards.

The B34 Codec support is not provided by this feature. The B34 Codec configured on a card allows the software to download an actual loop value for pads, rather than long line or shot line notations.

Periodic Clearing is not supported on the new XFCOT cards.

Feature interactions

Dial Tone Detector

A Dial Tone Detector notifies the software that a dial tone has been received for an outgoing call. With the XFCOT cards, dial tone detection is not attempted until a SEIZE ACKNOWLEDGE signal is received for those supervisions that require such a signal.

European XDID/DOD

Software support for European XDID/DOD cards and software support for European XFCOT cards provide similar functionality in the following areas:

- Trunk Failure Monitor or barring
- Downloading of PPM information
- Configuration and downloading of static pad setting for short line and long line, and
- Configuration download processing. Fields that are not filled due to configuration limitations are left blank and are not validated or interpreted by the firmware. The fields are treated as unused fields.

Federal Communications Commission (FCC) Compliance for DID Answer Supervision

If FCC Compliance for DID Answer Supervision (FC68) package 223 is configured on XFCOT units, it may lead to incorrect call status. Therefore, equipping the FCC package is not recommended.

Trunk Failure Monitor

When a BAR message indicating a problem situation is received, a trunk message is printed on the TTY, the uppermost key lamps light up on the attendant console, and the trunk is placed into BUSY state to prevent the trunk from being seized for new outgoing calls. The reception of an UNBAR message indicates that the problem situation has been cleared. A message is printed on the TTY, the lamps on the attendant console are darkened, and the seized trunk is idled. Note that BARA Class of Service must be configured on the trunk for the described processing to occur.

UK XFCOT (NT5K18AA)

For the UK XFCOT card, the NT5K18AA, there are no changes in configuration and operation except in the following areas:

- For static pad setting, the configuration for short line and long line has been changed from TRC to SHL for short line, and NTC to LOL for long line.
- The PPM configuration is done on a per route basis.
- Only one value is now downloaded for the PPM ID, on all UK cards.
- Only COT trunks are supported on the NT5K18AA. The NTAG04AA card supports COT and DID trunks.
- The balance impedance may now be configured on the NT5K90AA (Danish PPM) or NT5K90BA (Danish non-PPM) card.

Feature packaging

Meridian 1 Superloop Administration (XCT1) package 205.

Dependencies: Intelligent Peripheral Equipment (XPE) package 203; PPM/Message Registration (MR) package 101; Trunk Failure Monitor (TFM) package 182; and Trunk Hook Flash (Centrex) (THF) package 157.

Feature implementation

Table 88: LD 14 - Configure the trunk parameters for the new XFCOT cards.

Prompt	Response	Description
...		
CDEN	4D 8D	Card Density, where: 4D = Quad density, and 8D = Octal density.
...		
SIGL	ALS	ALS signaling on COT trunk with ground start (applies to the NTAG04AA unit only).

Prompt	Response	Description
BIMP	(3COM) 600	Three-component complex impedance. 600 ohms.
...		
SEIZ	(NO) YES	Automatic Guard Detection for outgoing trunk.
PPID	(0)-15	PPM country ID. Must be configured if PPM is enabled on the route. One PPID type per card. Trunks must be removed from a card to change PPID. Choose from one of the following PPM IDs, according to country: (0) – UK (50 Hz). 1 – France (50 Hz). 2 – France (12 Hz). 3 – Germany (16 kHz). 4 – Switzerland (12 kHz). 5 – Denmark (12 kHz). 6 – Norway (16 kHz). 7 – Belgium (16 kHz). 8 – Spain (12 kHz). 9 – Portugal (12 kHz). 10 – Holland (50 Hz). 11-15 – Reserved for future use.
BTID	(0)-15	Busy Tone Country ID. Must be configured for BTS supervised XCOT trunk. One BTID type per card. Trunks must be removed from card to change BTID. Choose from one of the following Busy Tone IDs, according to country: (0) – CCITT. 1-2 – Reserved fro future use. 3 – Germany. 4 – Switzerland. 5 – Denmark. 6 – Norway. 7-9 – Reserved for future use. 10 – Holland. 11-15 – Reserved for future use.
CLS		Class of Service options.
	(SHL) LOL	Enter SHL for short line (LOL for long line) static loss pad selection.
	(XBAT) BAT	Enter BAT for battery supervised COT; (XBAT) for no battery supervision.
	(XARF) ARF	Enter ARF for ARF supervised COT; (XARF) for no ARF supervision.
	(XLBS) LBS	Enter LBS for loop break supervised COT; (XLBS) for no loop break supervision.
	(XBTS) BTS	Enter BTS for tone supervised COT; (XBTS) for no tone supervision.
	(BARD) BARA	Enter BARA to allow barring; (BARD) to deny barring.
SUPN	(NO) YES	Enter SUPN = NO or the appropriate supervision type.
STVP	BAT ARF	Entering any of the following prompts will now override the previously configured type.

Prompt	Response	Description
	LBS BTS	Enter BAT for battery supervised COT. Enter ARF for ARF supervised COT. Enter LBS for loop break supervised COT. Enter BTS for tone supervised COT. The XBAT, XARF, XLBS, and XBTS prompts are no longer applicable.
SHL/LOL, BARA/BARD remain appropriate responses for the CLS prompt.		

Feature operation

No specific operating procedures are required to use this feature.

Chapter 42: Extended Local Calls

Contents

This section contains information on the following topics:

Feature description

To provide Aura with information regarding local calls on a CS 1000 Access Element, the Extended Local Calls (ELC) feature routes all local calls to a preconfigured ELC route.

To enable the ELC feature on a telephone, set ELC class of service. From an end user point of view, ELC calls do not differ from, and are originated in the same manner as, local calls.

When you enable ELC class of service, all local calls for a telephone are routed through ELC SIP trunks if possible. Calls are processed locally only when both parties disable ELC class of service.

Telephones supporting the Extended Local Calls feature

The following telephones support the Extended Local Calls feature:

- Analog telephones:
  - PBX telephones (500, 2500);
- Digital telephones:
  - Aries telephones (2006, 2008, 2016, 2616, 2216);

- Avaya 3900 Series Digital Deskphones (3901, 3902, 3903, 3904, 3905);
- Touchphone, Compact and Delta telephones (3000, 2112, 2018, 2009, 2317, 2003);
- DCS for Wireless phones (DCS);

• IP Phones:

- Avaya 1100 Series IP Deskphones (1110, 1120, 1140, 1150, 1165);
- Avaya 1200 Series IP Deskphones (1210, 1220, 1230);
- Avaya 2000 Series IP Deskphones (2001P2, 2002P1, 2002P2, 2004P1, 2004P2, 2007);
- Avaya 2033 IP Conference Phone;
- WLAN Handsets 2210 Series (2210, 2211, 2212);
- Avaya 2050 IP Softphone;
- SIP Line Phones: UEXT with UXTY = SIPL, SIPN or SIP3.

Operating parameters

There are no operating parameters associated with this feature.

Feature interactions

- Access Restrictions. If you define FR2 class of service for a telephone, then the access to or from TIE trunks is restricted for that telephone. ELC calls are local, and the Access Restriction feature does not affect incoming or outgoing ELC calls.
- Attendant Console. All calls between an Attendant Console and telephones with ELC enabled (ELCA) class of service (CLS) are terminated locally. All calls from and to an Attendant Console are terminated without ELC changes.

```
Example: Break-in feature.
Attendant Console is configured with Break-in key.
Telephones A and B have ELCA CLS.
```

a. Telephone A calls telephone B.

b. The call between telephones A and B is established as an Extended Local Call.

c. The Attendant presses the Break-in key and dials telephone A.

d. The Attendant hears the busy tone and presses the Break-in key again.

e. The call between the Attendant and telephone A is local and the conference call is established.

```
Example: Basic Call from an Attendant
Telephone A has ELCA CLS.
```

a. An Attendant calls telephone A.

b. The call between telephone A and the Attendant is established locally.

- Automatic Call Distribution (ACD). The behavior of an ACD call depends only on the class of service of the originator. If the originator has ELCA class of service, an ACD call is established as an Extended Local Call. If the originator has disabled ELC (ELCD) CLS, an ACD call is established as a local call.
- Bandwidth Management.
  - Turn on VPNI in the Customer Data Block, to ensure correct bandwidth counting.
  - For some codecs the calculated bandwidth of Extended Local Calls is 2 kbits less than the actual bandwidth usage. For example, for codec type G.711 u-law 20ms NOVAD (codec number 4) the actual average bandwidth usage is 190 kbit. The calculated bandwidth usage is 188 kbit.
  - The ELC feature does not interoperate with the adaptive bandwidth management feature.
- Call Forward No Answer. When a telephone with ELCD CLS calls a telephone with ELCA CLS and the ELCA CLS telephone forwards the call no answer to an ELCD CLS telephone, the call will be established as an Extended Local Call, and not as a local call.
- Call Park. If an ELCA telephone parks a call, the call will be established as an Extended Local Call, regardless of the calls originating or terminating class of service.

```
Example:
Telephones A and C have ELCD CLS.
Telephone B has ELCA CLS.
```

a. Telephone A calls telephone B.

b. Telephone B parks the call.

c. Telephone C dials the parked DN.

d. The call between telephones A and C is an Extended Local Call.

- Call Pickup
  - If a telephone with ELCA CLS picks up a local call, the call will be established locally; not as an Extended Local Call.

```
Example:
Telephones A and B have ELCD CLS.
Telephone C has ELCA CLS.
```

i. Telephone A calls telephone B.

ii. Telephone C picks up the call from telephone B.

iii. The call between telephones C and A is not an Extended Local Call.

- If a telephone with ELCD CLS picks up an Extended Local Call, the call will be established as an Extended Local Call, regardless of the originating or terminating CLS. (Both, the originating and terminating CLS may be ELCD).

```
Example:
Telephones A and C have ELCD CLS.
Telephone B has ELCA CLS.
```

i. Telephone A calls telephone B.

ii. Telephone C picks up the call from telephone B.

iii. The call between telephones C and A is an Extended Local Call.

- Call Trace . The Call Trace functionality is enhanced for Extended Local Calls. For example, the Call Trace functionality can monitor both calling parties simultaneously in an Extended Local Call.
- Conference Call. If a conference call consists of more than three parties and all conference legs are Extended Local, then the call will be dropped for all parties after the originator of the conference call releases the call.

  Ad hoc Conference Tearing Down. If the originator of a conference call between three parties goes onhook, the call establishes between the other two parties in accord with their ELC class of service configuration.
- ELC class of service. ELC enabled class of service cannot be set if ELC is disabled for the customer. If there are ELC enabled telephones, then ELC can not be disabled in LD15.
- Group Call. ELC CLS does not interoperate with the Group Call feature. A Group call from an ELCA CLS telephone to ELCA CLS telephones is established as a local call.

```
Example #1:
Telephones A, B, and C have ELCA CLS.
Telephone D has ELCD CLS.
```

a. Telephone A calls the group of telephones B, C and D.

b. Telephone B answers the call.

c. The call between telephones A and B is established locally; not as an Extended Local Call.

d. Telephones C and D answer the call. The conference call is established.

```
Example #2:
Telephones A, B, C and D have ELCA CLS.
```

a. Telephone A calls the group of telephones B, C and D.

b. Telephone B answers the call.

c. The call between telephones A and B is established locally; not as an Extended Local Call.

d. Telephones C and D answer the call. The conference call is established.

```
Example #3:
Telephones A, B, C and D have ELCD CLS.
```

a. Telephone A calls the group of telephones B, C and D.

b. Telephone B answers the call.

c. The call between telephones A and B is established locally; not as an Extended Local Call.

d. Telephones C and D answer the call. The conference call is established.

- Music on Hold. Music on Hold for ELC calls is configured in the same manner as trunk calls; set MUS to YES and assign appropriate music route to the ELC Route Data Block.
- Override. If an ELC enabled telephone overrides an established call, the override is performed locally.
- Routes and Trunks. You can only use SIP ELC routes, SIP ELC trunks and ELC route lists for the ELC feature.

  You can only define ELC RLB as ELC_RLI, in the Customer Data Block (CDB).

  You can not do the following with routes and trunks:

  - Remove a SIP ELC route, if trunks for the route exist.
  - Change a non-SIP ELC route to a SIP ELC route, and you can not change a SIP ELC route to a non-SIP ELC route, if trunks for the route exist.
  - Move trunks from a SIP ELC route to a non-SIP ELC route, nor from a non-SIP ELC route to a SIP ELC route.
  - Remove a SIP ELC route, if it is shown in any ELC Route List Block (RLB).
  - Change a SIP ELC route to a non-SIP ELC route (changing Protocol ID (PCID) in Route Data Block (RDB)), if it is shown in any ELC RLB.
  - Change a non-SIP ELC route to a SIP ELC route (changing PCID for the route in RDB), if it is shown in any RLB.
  - Configure non-ELC routes as ELC RLB entries, and you can not configure ELC RLB entries as non-ELC routes.
  - Define ELC RLB for other features.
  - Change the ELC prompt value for an existing RLB.
  - Use ELC RLB in steering codes.

- Transferring calls.
  - When you transfer an ELC call to an external party, the ELC trunks are not released after the transfer is completed. The transferred call occupies two ELC trunks and one trunk for the external call.
  - When you transfer an external call through ELC, the ELC trunks are not released after the transfer is completed. The transferred call occupies two ELC trunks and one trunk for the external call.
- Trunk optimization (TRO). The ELC feature does not interoperate with TRO. Turn off TRO on the route data block intended for Extended Local Calls.
- Voice Call. ELC class of service does not interoperate with the Voice Call feature. Voice Calls only can be established locally, regardless of the originating and terminating class of services.

Feature packaging

The Extended Local Calls feature requires the following packages:

- CDR (package 4)
- NARS (package 58)
- FNP (package 160)

  Note:
  You must enable FNP for the customer.
- TATO (package 312)
- SIP (package 406)

Feature implementation

Task summary

1. Add new ELC route.
2. Configure ELC trunks in the ELC route.
3. Configure ELC Route List Block.
4. Define the configured ELC Route List Block in Customer Data Block.
5. Configure ELC class of service enabled phones.

Configuring ELC routes in Element Manager

1. In the Element Manager navigator, click **Routes and Trunks > Routes and Trunks**.
2. Click **Add Route**.

   The New Route Configuration window displays.

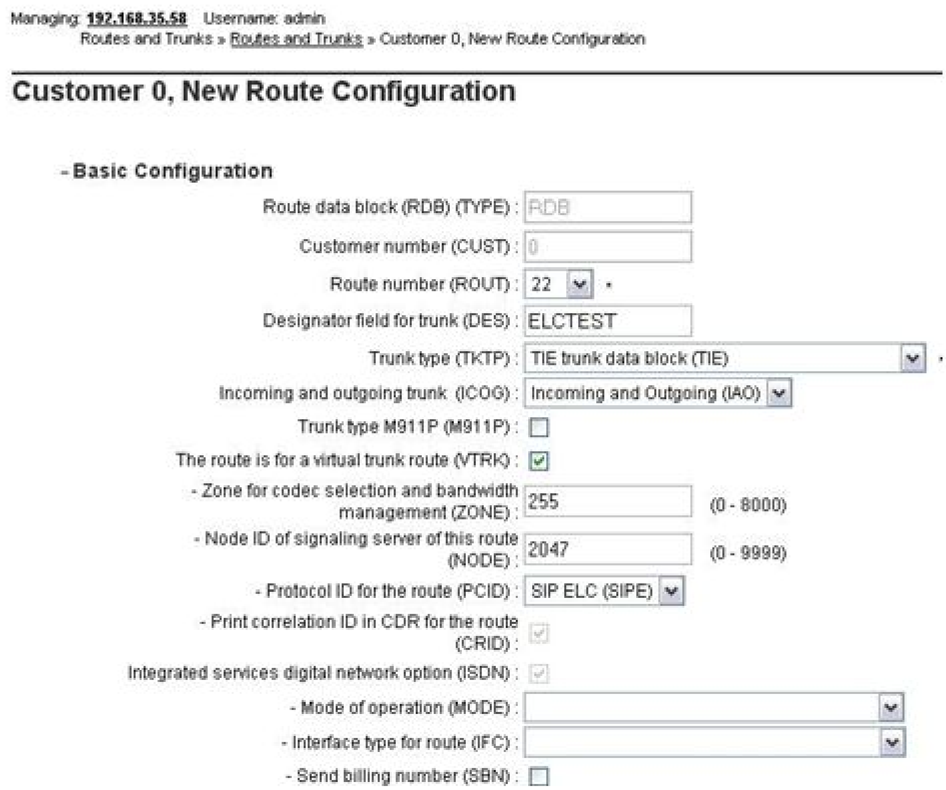


Figure 6: New Route Configuration

3. From the Trunk Type (TKTP) list, select **TIE trunk data block (TIE)**.
4. From the Incoming and outgoing trunk (ICOG) list, select **Incoming and Outgoing (IAO)**.
5. Select **This route is for a virtual trunk route (VTRK)** check box.
6. From the Protocol ID for the route (PCID) list, click **SIP ELC (SIPE)**.

   The **Print correlation ID in CDR route (CRID)** check box is selected automatically.

   The **Access code for the trunk route (ACOD)** field is not displayed when SIP ELC (SIPE) is selected.
7. Enter valid values in all required fields on the New Route Configuration window.
8. Click **Save**.

Configuring ELC trunks in the ELC route in Element Manager

1. In the Element Manager navigator, click **Routes and Trunks > Routes and Trunks**.
2. Click **Add Trunk** for the configured ELC route.

   The TIE Trunk Data Block window displays.

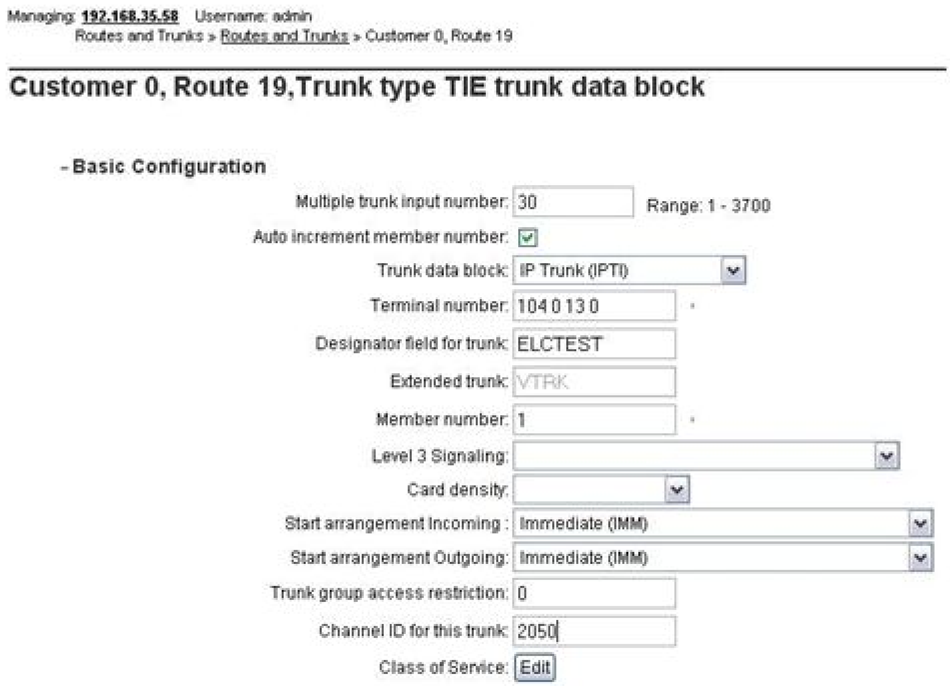


Figure 7: TIE Trunk Data Block

3. Enter a value in the **Multiple trunk input number** field.
4. From the Trunk data block list, select **IP Trunk (IPTI)**.
5. Enter valid values in all required fields on the TIE Trunk Data Block window.
6. Click **Save**.

Configuring ELC Route List Block in Element Manager

1. In the Element Manager navigator, click **Dialing and Numbering Plans > Electronic Switched Networks**.

   The Electronic Switched Network (ESN) window displays.
2. From **Network Control & Services**, select **Route List Block (RLB)** and fill in the new Route List Block number.
3. Click **Add**.

   The Route List Block window displays.

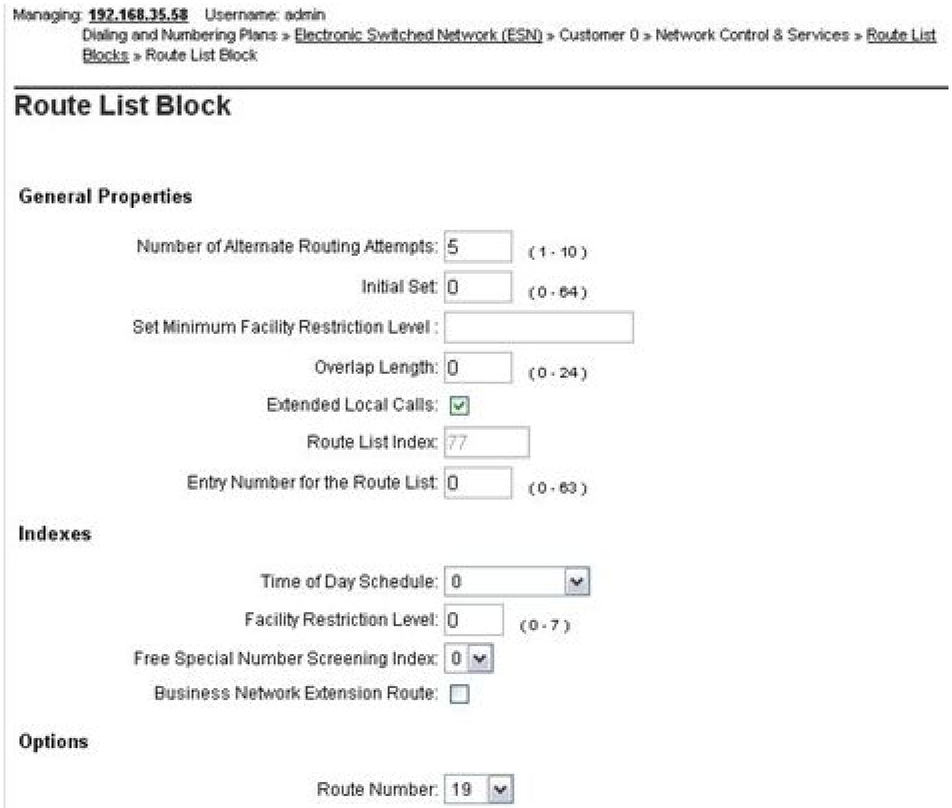


Figure 8: Route List Block

4. Select the **Extended Local Calls** check box.
5. From the **Route Number** list, select the preconfigured ELC route.
6. Enter valid values in all required fields on the Route List Block window.
7. Click **Save**.

Defining the configured Route List Block in the Customer Data Block in Element Manager

1. In the Element Manager navigator, select **Customers**.
2. Click the appropriate **Customer number** and then click the **Feature packages** link.

   The Feature Packages window displays.

Figure 9: Feature Packages

3. Click **Flexible Numbering Plan Package: 160** to see more options.
4. Select the **Enable Flexible Numbering Plan for customer** check box.
5. Select the **Extended Local Calls** check box.
6. In the **Extended Local Calls Route list index** field, enter the preconfigured Route List Block index.
7. Click **Save**.

Enabling ELC class of service in Element Manager

1. In the Element Manager navigator, select **Phones**.
2. Click **Add**.

   The New Phones window displays.

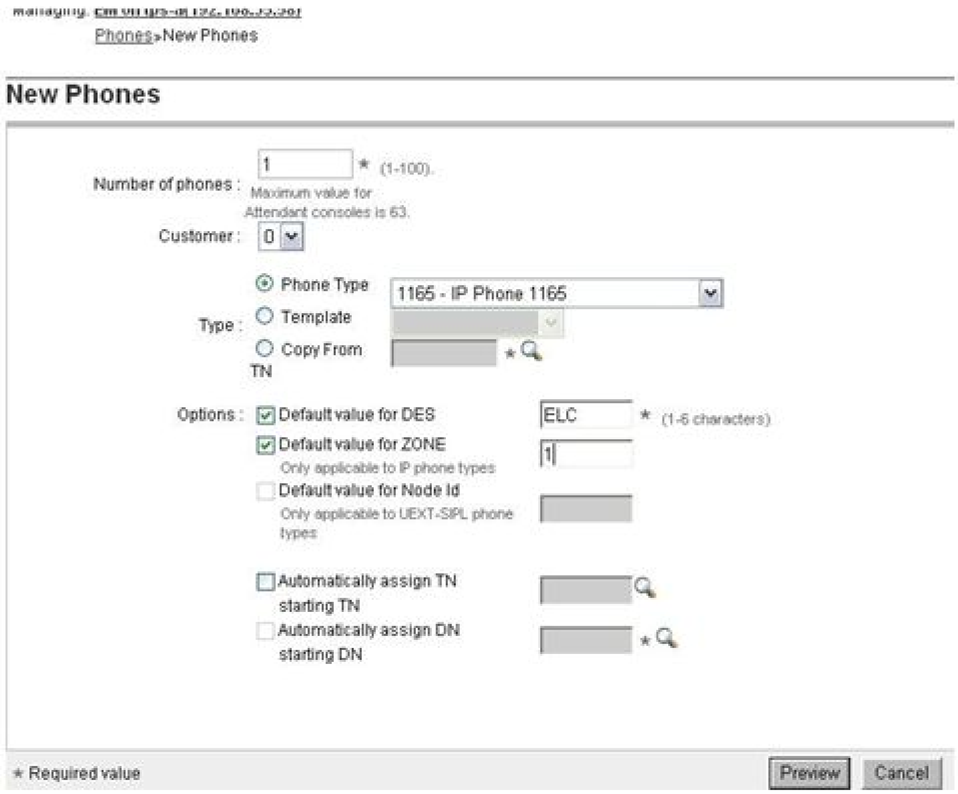


Figure 10: New Phones

3. Enter valid values in all required fields on the New Phones window.
4. Click **Preview**.

   The Phone Details window displays.

Figure 11: Phone Details

5. From the **ELC Extended Local Calls** list in the Features pane, select **Allowed** .
6. Enter valid values in all required fields on the Phone Details window.
7. Click **Save**.

Feature operation

You do not need any specific operating procedures to use this feature.

Chapter 43: Extended Multifrequency Compelled Sender/Receiver

Contents

This section contains information on the following topics:

Feature description

This feature provides combined Multifrequency Compelled Signaling (MFC) and MFE signaling for SOCOTEL, using the Extended Multifrequency Compelled Sender/Receiver NT5K21AA card. This card is based on the XDTR card NT8D16AB.

Although the NT5K21AA card provides both MFC and MFE signaling, it may only be configured as one or the other: it is not possible to configure certain units as MFC and other units as MFE, on the same card. If there is a requirement for both MFC and MFE signaling, then two NT5K21AA cards may be configured – one for MFC and one for MFE.

In support of the NT5K21AA card, the system software has been modified as follows:

- Four DS-30X channels are provided for simultaneous generation and detection (forward and backwards) of MFC digits
- Four DS-30X channels are provided for alternate generation and detection of MFE digits (software selectable)
- A DS-30X channel of A10 formatted signaling is provided for communication between the system CPU and the NT5K21AA pack

- A-law and μ-law PCM encoding schemes are both supported
- Any one of 16 tone output levels may be specified for each channel
- Any one of four levels may be specified as a minimum receiver acceptance level
- Special MFC functions, such as pulse or automatic mode, are provided
- Card-ID information, configured during the manufacture of the NT5K21AA pack, is stored in the EEPROM message
- Hardware self-test and troubleshooting capabilities, including loop-back of PCM channels at the NT5K21AA, are provided by maintenance software, and
- The standard faceplate Enable/Disable Status Indicator LED is provided.

Most of the existing command structure for signaling has been maintained, with the following exceptions:

- During RESET, the NT5K21AA card is configured as either MFC or MFE, and as either A-law or μ-law
- More comprehensive self-test results are provided
- The minimum receiver acceptance level (MFL) is downloaded, and
- An extended range is provided for the MFC digit level (MFL).

Operating parameters

Both A-law and μ-law, which are software selectable, are supported. But when a Companding Law is selected in LD 97, it is supported on a system basis.

System parameters have to be downloaded on the NT5K21AA card in the following cases:

- When the NT5K21AA card is enabled in LD 32 and 54
- During service changes and initialization
- When a new NT5K21AA unit is defined in LD 13, and
- When an NT5K21AA card is moved to another card, in LD 13.

The default system parameters for downloading are NT5K21AA card type MFC, Mμ-law companding law, and a Minimum Receiver Acceptance level of -36 dB.

The following Card-LAN interface capabilities are supported by the NT5K21AA card:

- Periodic Intelligent Peripheral Equipment (IPE) polling of the status of the NT5K21AA card
- Requesting of card-ID, card type, and firmware version for auto-configuration, and
- Requesting of configuration data, including the DS-30X signaling type, during power up and RESET.

Feature interactions

There are no feature interactions associated with this feature.

Feature packaging

- Intelligent Peripheral Equipment (XPE) package 203.
- Multifrequency Compelled Signaling (MFC) package 128.
- Multifrequency Signaling for SOCOTEL (MFE) package 135.

Feature implementation

Task summary list

The following is a summary of the tasks in this section:

1. Table 89: LD 97 - Download the system parameters, and define the Multifrequency Minimum Receiver Level (MFRL). on page 309

   Download the system parameters, and define the Multifrequency Minimum Receiver Level (MFRL).

2. Table 90: LD 14 Create the trunk data block and define the range of Multifrequency Digit Level on page 310

   Create the trunk data block and define the range of Multifrequency Digit Level

Table 89: LD 97 - Download the system parameters, and define the Multifrequency Minimum Receiver Level (MFRL).

Prompt	Response	Description
...		
MFRL	0-(2)-3	Multifrequency minimum Receiver Level for XMFC/ XMFE (NT5K21) for Superloop only. 0 = -28 dBM. 1 = -32 dBM. 2 = -36 dBM (the default). 3 = -40 dBM.

Table 90: LD 14 Create the trunk data block and define the range of Multifrequency Digit Level

Prompt	Response	Description
...		
MFL	(0)-15	Multifrequency digit level. Expanded from 0-7 to 0-15 for Superloop only. Enter the MFC digit level required for signals to the Public Switched Telephone Network (PSTN). Superloop codes and values: 0 = -8 dBM. 1 = -11 dBM. 2 = -12 dBM. 3 = -13 dBM. 4 = -14 dBM. 5 = -15 dBM. 6 = -16 dBM. 7 = -31 dBM. 8 = -4 dBM. 9 = -5 dBM. 10 = -6 dBM. 11 = -7 dBM. 12 = -9 dBM. 13 = -10 dBM. 14 = spare. 15 = spare. Levels 0-7 are already defined.

Feature operation

No specific operating procedures are required to use this feature.

Chapter 44: Extended Tone Detector Global Parameters Download

Contents

This section contains information on the following topics:

Feature description

An Extended Tone Detector (XTD) card is capable of performing both Dual-tone Multifrequency (DTMF) and Dialtone (DT) detection. It is possible to download parameters onto the card so that it may be customized for a particular environment. These parameters are the A-law/Mμ-law for the Extended DTMF (XDTR) portion of the card, and the first stage dialtone detection (DT).

This feature allows several new parameters and a new message to be downloaded onto the new global XTD pack, the NT5K48AA. The new parameters, grouped under the categories of first stage dialtone detection, second stage dialtone detection, and XDTR minimum accept level, are:

- flexible first stage dialtone detection
    - frequency band (expanded operation)
    - minimum detect level
    - minimum validation time

- break duration
- cadence type

• flexible second stage dialtone detection

- second stage configuration, and

• flexible XDTR minimum accept level

The new message is the Detect Second Stage Dialtone. It allows the NT5K48AA to distinguish between using the first stage dialtone detection parameters and the second stage dialtone detect parameters for detecting dialtone.

To configure the first and second stage dialtone detection parameters, a new type, DTD, and associated prompts have been introduced in LD 97. This prompt allows a craftsperson to create up to eight different XTD tables containing the parameters. In LD 13, a table is associated with each XTD card. These parameters are downloaded onto each XTD card.

To configure the flexible XDTR minimum accept level parameter, a new type, DTR, and associated prompt MINL (that defines the minimum accept level, on a per-system basis) have been introduced in LD 97. This parameter is downloaded onto each XTD card and DTR card.

Operating parameters

Since there is only one parameter for the second stage dialtone detector (the craftsperson, in LD 97, enters a value between 0-15 to indicate which of the 16 options to use), the parameters for second stage dialtone detection hardware operation are hardcoded with limited flexibility. The NT5K48AA has to be modified to provide second stage configuration, if it is to be introduced to a country that has an undefined configuration.

The default values for all parameters are for the Swiss standards. However, if the UK Program (UK) package 190 is equipped, the UK recommended default values are used.

Feature interactions

There are no feature interactions associated with this feature.

Feature packaging

Intelligent Peripheral Equipment (XPE) package 203; and Meridian 1 Superloop Administration (XCT1) package 205.

Feature implementation

Task summary list

The following is a summary of the tasks in this section:

1. Table 91: LD 97 - Configure all the first- and second- stage dial tone detection parameters (TYPE = DTD). on page 313

   Configure all the first- and second- stage dial tone detection parameters (TYPE = DTD).
2. Table 92: LD 97 - Configure the flexible XDTR minimum accept level parameter (TYPE = DTR). on page 314

   Configure the flexible XDTR minimum accept level parameter (TYPE = DTR).
3. Table 94: LD 13 - Define the protected data block of the XTD card. on page 315

   Define the protected data block of the XTD card.
4. Table 95: LD 16 - Define the route protected data block of the XTD card. on page 315

   Define the route protected data block of the XTD card.

Table 91: LD 97 - Configure all the first- and second- stage dial tone detection parameters (TYPE = DTD).

Prompt	Response	Description
...		
TYPE	DTD	First- and second-stage dialtone detection parameters.
XTDT	(0)-7	Extended Tone Detection Table. XDTD table number in which the parameters are stored. Table 0 can be changed but must not be removed. Table 0 always exists and is initialized to default values.

Prompt	Response	Description
DFQ	0-(4)-15	Dial Tone Frequency band for 1st dial tone, which is the number of the dial tone frequency band chosen in the hardcoded frequency table. With United Kingdom (UK) package 190 the default value for DFQ = 0.
MDL	10-(20)-40	Minimum Detect Level for 1st dial tone in dBm, which is the absolute value of the minimum detect level. Odd input is rounded down. With United Kingdom (UK) package 190 the default value for MDL = 30 (-30 dBM).
MVT	100-(400)-1600	Minimum Validation Time for dial tone in milliseconds. Input that is not a multiple of 100 is rounded down to the next multiple of 100. With United Kingdom (KUK) package 190 the default value for MVT = 300.
BRK	(0)-240	Break Duration (maximum) for 1st dial tone in milliseconds. Input that is not a multiple of 16 is rounded down to the next multiple of 16.
CAD	(0)-15	Cadence type for 1st dial tone, which is the number of the cadence pattern in the hardcoded table.
SSC	(0)-15	Second Stage Configuration, which is the configuration number for the second stage dial tone detection to be set in the firmware.

Table 92: LD 97 - Configure the flexible XDTR minimum accept level parameter (TYPE = DTR).

Prompt	Response	Description
...		
TYPE	DTR	First- and second-stage dial tone detection parameters.
MINL	3-(42)-48	Minimum accept level for Digitone Receivers in dBM, which is the absolute value of the minimum accept level. Input that is not a multiple of 3 is rounded down to a valid multiple of 3. With United Kingdom (UK) package 190 the default value for MINL = 45 (-45 dBm).

Refer to Table 93: Recommended parameters according to country on page 314 for recommended configuration values for each country. The default values given in parenthesis are for non-UK countries.

Table 93: Recommended parameters according to country

Country	DFQ	MDL	MVT	BRK	CAD	SSC	MINL
Germany	1	-16 dBm	1000 ms.	0 ms.	0	—	-45 dBm
France	0	-24 dBm	1000 ms.	30 ms.	0	0	-30 dBm

Country	DFQ	MDL	MVT	BRK	CAD	SSC	MINL
Sweden	1	-28 dBm	1000 ms.	60 ms.	0	—	-28 dBm
Norway	1	-32 dBm	1400 ms.	0 ms.	0	—	-45 dBm
Switzerland	4	-28 dBm	1000 ms.	0 ms.	0	—	-30 dBm
Spain	2	-32 dBm	1000 ms.	0 ms.	0	0	-30 dBm
UK (330/440)	0	-30 dBm	500 ms.	0 ms.	0	—	-45 dBm
UK (33/50)	3	-30 dBm	900 ms.	0 ms.	0	—	-45 dBm
Denmark	1	TBD	TBD	0 ms.	0	—	-45 dBm
Holland	0	TBD	TBD	TBD	0	—	-30 dBm
New Zealand	1	TBD	TBD	TBD	0	—	-45 dBm

Table 94: LD 13 - Define the protected data block of the XTD card.

Prompt	Response	Description
...		
XTDT	(0)-7	Extended Tone Detector Table Number, prompted when TYPE = XTD. If a table other than 0 is entered, it must have already been configured in LD 97.

Table 95: LD 16 - Define the route protected data block of the XTD card.

Prompt	Response	Description
...		
XTDT	(0)-7	Extended Tone Detector Table Number, prompted with Meridian 1 Superloop Administration (XCT1) package 205. Must be the same value as defined in LD 13. If a table other than 0 is entered, it must have already been configured in LD 97.

Feature operation

No specific operating procedures are required to use this feature.

Chapter 45: FCC Compliance for DID Answer Supervision

Contents

This section contains information on the following topics:

Feature description

This feature is designed to meet the requirements in the United States, Section 68.314(h) of Part 68, and the DOC requirements in Canada, Section 3.22 of CSO3 Part 1, for answer supervision of redirected telephone calls to ensure proper billing.

This feature is designed specifically for telephone calls coming in through Direct Inward Dialing (DID) trunks. Answer supervision for all other types of telephone calls is not affected. This feature works in conjunction with the following types of calls:

- Direct Inward Dialing (DID) calls terminating at the system and forwarded to a Recorded Announcement (RAN).
- Direct Inward Dialing (DID) calls forwarded by the system through the public switched network (PSN) to another number in the Public Exchange/Central Office (CO), or to another system.

  On North American COT, FEX, and WATS trunks, Central Offices do not always return answer supervision. When no answer supervision is returned, the system software uses the end-of-dial timer for non-Digitone trunks (EOD timer), or the end-of-dial timer for Digitone trunks (ODT timer) to verify call connection. For Federal Communications

Commission (FCC) compliance, the EOD and ODT timers will still be used for incoming DID calls, except that EOD is capped at 20 seconds even if configured for more.

This feature handles incoming DID calls over Data Terminal Interface (DTI), Integrated Services Digital Network (ISDN), and analog trunks. Outgoing calls over Central Office (CO) and TIE trunks are also handled. System components involved include trunks, the system, and the CO.

The following explains how the system components handle answer supervision.

- Analog, DTI, and ISDN incoming trunks: These are covered as long as they are DID incoming trunks. For incoming analog and DTI trunks, answer supervision or pseudo-answer supervision is returned by the system to the CO, if necessary. For incoming ISDN trunks, the connect message is returned instead.
- Analog, DTI, and ISDN outgoing trunks: For incoming DID calls, the answer and disconnect supervisor (SUPN) of the outgoing trunk is forced to NO. The EOD or ODT timer simulates the return of answer supervision.
- For DID calls terminating at the system, the system returns answer supervision based on the terminating condition. For DID calls forwarded to Public Switched Networks (PSN) or private networks, the system returns answer supervision based on the condition of the outgoing trunk (whether answered or timed out).
- CO: The system provides the pseudo-answer for DID calls because the system cannot return answer supervision.

DID calls terminating at the system

The requirements for a DID call terminating at the system to return answer supervision to the incoming DID trunk are shown in the following table. The ASUP prompt in LD 16 is kept for other types of calls, but the system software enforces the correct settings to return answer supervision if a Recorded Announcement (RAN) is used for DID calls, regardless of the value originally specified in the service change.

Table 96: Returning Answer Supervision for DID calls terminating at the system

DID call terminating status	Answer supervision returned with FCC Compliance
Answered by the called DID station	Yes
Answered by an attendant	Yes
Routed to dialing prompt	Yes
Routed to Meridian Mail	Yes
Routed to Recorded Announcement, including invalid number, not in service, and not assigned announcements	Yes

DID call terminating status	Answer supervision returned with FCC Compliance
Routed to Recorded Announcement by Automatic Call Distribution (ACD), including invalid number, not in service, and not assigned announcements	Yes
Not answered	No
Busy signal	No
Recorder signal	No

Calls forwarded to Public Switched Network

Because it is uncertain whether or not the far end will return answer supervision, the system uses the EOD and ODT timers. If the system has not detected the return of answer supervision upon timeout of the outgoing CO trunk, the system sends pseudo-answer supervision to the incoming DID trunk. This timer is set in LD 16 on a per-route basis. When a CO trunk is configured, system software forces the value of SUPN to NO. Consequently, system software does not expect the return of answer supervision, and returns answer supervision in the following cases:

- The system receives answer supervision from the outgoing CO trunk before the EOD or ODT timer of the outgoing route expires.
- The system does not receive answer supervision from the outgoing trunk and the EOD or ODT timer of the outgoing route expires; pseudo-answer supervision is generated.

There are still some cases in which the SUPN value for CO trunks is assigned to YES if the CO supports a reverse battery mechanism.

With FCC Compliance, a more stringent mechanism is introduced to apply SUPN = NO in LD 14 to all CO trunks, even those configured as polarity sensitive. Service-changeable EOD or ODT timers are always used for incoming DID calls to enforce the return of answer supervision. In this case:

EOD = 128-19,968 milliseconds (ms) (default time is 13,952 ms), and

ODT = 256-16,128 ms (default time is 4,096 ms).

The EOD timer expires at 20 (20,000 ms) for FCC Compliance. For outgoing DID calls, the EOD upper limit is 32,640 ms.

DID calls forwarded to private networks

Answer supervision is not always returned on TIE trunks because some TIE trunks leased from public carriers are connected to COs that do not support answer supervision.

Currently, the system provides the SUPN prompt (LD 14) to specify the availability of answer supervision on certain types of trunks, including TIE, CAM, Common Control Switching Arrangement (CCSA), and CAA (CCSA Automatic Number Identification [ANI]). If SUPN is YES, and it is an outgoing trunk, the system does not return answer supervision to the incoming DID trunk unless answer supervision is received from that outgoing trunk. If the user specifies NO, the system returns pseudo-answer supervision upon EOD or ODT timeout. Such implementation causes short billing and overcharge problems.

To solve this problem, a treatment similar to the one implemented on CO trunks is used on the trunks in this category. The system enforces SUPN = NO without changing the SUPN value.

For incoming DID calls routed to private networks, SUPN is enforced to NO to ensure the return of answer supervision on the outgoing TIE, CO, FEX, WATS, CAM, CAA, and CCSA trunks. If answer supervision is not returned when the end of dial timeout occurs, the system disregards the original value of SUPN set by the user and forces the return of answer supervision.

When the call comes from a DID trunk, the following outgoing trunks are affected: TIE, CO, FEX, WATS, CAM, CAA, and CCSA.

Operating parameters

Allowing system equipment to be operated in such a manner as to not provide proper answer supervision signaling is in violation of Part 68 rules.

This equipment returns answer supervision signals to the public switched telephone network (PSTN) when:

- answered by the called station
- answered by the attendant
- routed to a Recorded Announcement that can be administered by the Customer Premises Equipment (CPE) user, and
- routed to a dial prompt.

This equipment returns answer supervision on all DID calls forwarded back to the PSTN. Permissible exceptions are when:

- a call is unanswered
- a busy tone is received, and
- a reorder tone is received.

Feature interactions

Extended DID and DOD Software Support - Europe

If FCC Compliance for DID Answer Supervision (FC68) package 223 is configured on XDOD units, it may lead to incorrect call status. Therefore, equipping this package is not recommended.

Extended Flexible Central Office Trunk Software Support

If FCC Compliance for DID Answer Supervision (FC68) package 223 is configured on XFCOT units, it may lead to incorrect call status. Therefore, equipping the FCC package is not recommended.

Feature Group D and Japan DID trunks

Feature Group D trunks and Japan (JPN) DID trunks are not affected by this feature.

ISDN trunks

Both incoming and outgoing Integrated Services Digital Network (ISDN) trunks are affected by this feature.

- For ISDN incoming DID trunks, the connect message is returned when answer supervision is returned or when the end of dial timer expires.
- For ISDN outgoing trunks, the end of dial timer is added to the protocol to simulate the EOD timer when a connect message is not returned from the far end; the system generates a pseudo-answer supervision to send to the incoming trunk.

Intercept Recorded Announcement

With this feature, incoming DID calls that are intercepted to a Recorded Announcement (RAN) are provided with answer supervision.

Feature packaging

This feature requires Federal Communications Commission Compliance for DID Answer Supervision (FC68) package 223.

Feature implementation

There are no specific implementation procedures for this feature.

Feature operation

No specific operating procedures are required to use this feature.

Chapter 46: FCC Compliance for Equal Access

Contents

This section contains information on the following topics:

Feature description

This feature brings the system into compliance with the Equal Access portion of the Federal Communication Commission (FCC) 68 ruling. This calls for the optional restriction of two types of direct-dialed Equal Access toll calls, while allowing all other Equal Access dialing sequences (with the exception of operator cut-through) and call processing operations.

The two types of Equal Access calls that may be restricted are:

- North American toll calls (1+NPA+NXX+XXXX),where NPA = Number Plan Area, NXX = any three digits with N being any digit except 0 or 1, and XXXX = any four digits, and
- International toll calls (011+CC+NN),where CC = Country Code and NN = National Number. FCC compliant dialing plans.

Table 97: FCC compliant dialing plans.

Dialing Format	Destination
Allow: 10XXX+0 10XXX+0+(NPA)+NXX+XXXX	Operator of carrier specified by XXX. Operator function of carrier specified by

Dialing Format	Destination
10XXX+0+SAC+NXX+XXX 10XXX+01+CC+NN	XXX. Subscribed carrier specified by XXX. Operator function of carrier specified by XXX.
Allow/Deny: 10XXX+1+(NPA)+NXX+XXXX 10XXX+011+CC+NN	Carrier specified by XXX. Carrier specified by XXX.
where: XXX = any three digits, XXXX = any four digits, NPA = Number Plan Area, NXX = any three digits with N being any digit except 0 or 1, CC = Country Code, NN = National Number.	

This feature provides two methods of restricting Equal Access toll calls, General Carrier Restriction (GCR), and Selective Carrier Restriction (SCR). These limitations, configured in LD 16, require that the originating set have a Network Class of Service of Equal Access. The Equal Access restriction for an NCOS group is configured in LD 87.

GCR permits a configuration of allowing or denying all North American Equal Access toll calls and all international Equal Access toll calls. This GCR restriction is based on call type only, and does not take into account the dialed Carrier Identification Code. SCR uses the New Flexible Code Restriction (NFCR) feature to place a more selective restriction on Equal Access toll calls, based on the dialed Carrier Identification Code (CIC). So, for example, Equal Access toll calls for a carrier with a CIC of 434 could be denied, while Equal Access toll calls for a carrier with a CIC of 225 could be allowed.

GCR is the simplest method to implement and requires no additional memory. It is therefore recommended that GCR be used if there is no need to restrict Equal Access toll calls based on carrier usage. SCR is more difficult to set up and requires additional memory. Use this method only if there is a strong need to restrict Equal Access toll calls based on carrier usage.

Since both methods can be active at the same time, the optimum solution in some cases would be to implement a combination of GCR and SCR. If, for example, a requirement exists to restrict all North American Equal Access toll calls and only certain international Equal Access toll calls, based on carrier usage, then GCR could be configured to handle the North American Equal Access toll calls while SRC could be configured to handle the international Equal Access toll calls.

Operating parameters

The same requirements for normal calls using the New Flexible Code Restriction (NFCR) feature apply to calls made under the Selective Carrier Restriction method, except that Equal Access operator calls (10XXX0) are allowed to be completed while Equal Access international toll calls (10XXX011) are denied.

This feature could require extra memory when operating under the Selective Carrier Restriction method (as much as 15.5K words of protected data storage when fully configured). Insufficient memory may limit the number of CIC codes which may be restricted.

This feature only supports COT, FEX, WAT, DID, and TIE routes with Standard Signaling.

This feature does not support network signaling, since the intention is to restrict Equal Access calls directly terminating at the Central Office and not at another network node.

This feature does not restrict calls made by an attendant.

The # sign is not outpulsed by the system, as recommended in the FCC Bellcore North American Dialing Plan.

The operator cut-through dialing sequence of 10XXX#, which is recommended in the FCC Bellcore North American Dialing Plan, is not supported on the system.

Feature interactions

New Flexible Code Restriction

The New Flexible Code Restriction (NFCR) feature has been modified to allow for the restriction of Equal Access international toll calls (10XXX+011+CC+NN) while not restricting Equal Access operator calls (10XXX+0).

Feature packaging

This feature is not packaged, however the following packages are required to make it operational: Network Class of Service (NCOS) package 32 is required for both the General Carrier Restriction and Selective Carrier Restriction methods; and New Flexible Code Restriction (NFCR) package 49 is required for the Selective Carrier Restriction method.

Feature implementation

Task summary list

The following is a summary of the tasks in this section:

1. Table 98: LD 16 - Apply Equal Access call restriction to this route. on page 326

   Apply Equal Access call restriction to this route.
2. Table 99: LD 87 - Specify whether Equal Access with a NCOS group is to be associated or not. on page 326

   Specify whether Equal Access with a NCOS group is to be associated or not.

Table 98: LD 16 - Apply Equal Access call restriction to this route.

Prompt	Response	Description
...		
EQAR	(NO) YES	Enable Equal Access limitations. Prompted when TKTP = CO, FEX, WAT, or ISA, and ICOG = OGT, or IAO.
- GCR	(NO) YES	General Carrier Restriction to restrict Equal Access calls.
- - NTOL	(DENY) ALOW	North American toll calls (that is, 1+ calls).
- - ITOL	(DENY) ALOW	International toll calls (that is, 011+ calls).
- SCR	(NO) YES	Selective Carrier Restriction to restrict Equal Access calls. Prompted when EQAR = YES, and New Flexible Code Restriction is enabled. NTOL and ITOL must both be ALOW.

Table 99: LD 87 - Specify whether Equal Access with a NCOS group is to be associated or not.

Prompt	Response	Description
...		
- EQA	(NO) YES	Equal Access (is not) is associated with this NCOS group.

Feature operation

The dialing sequence for Equal Access calls on the system is:

- Access Code (either trunk or NARS/BARS)
- Carrier Access Code (CAC). The CAC is comprised of the Equal Access code (10) and the Carrier Identification Code (CIC) (any three digits). The CIC specifies the carrier that will handle the call,
- Telephone number.

The dialing sequence can contain two special characters, the asterisk (*) and the number sign (#). The * sign within a dialing invokes a three-second pause in the call processing procedure, and has no bearing on call restriction routines. The # sign within a dialing sequence signifies the end of the dialing sequence, and that it can be examined by call restriction routines. The only exception occurs when all international Equal Access toll calls have been restricted on a switch. In this case, direct-dialed Equal Access operator calls may not terminate with the # sign (in order to avoid possible fraud when calls are placed from trunks with Digitone Class of Service).

The asterisk (*) used to introduce a pause while outpulsing digits is supported on analog and DTI trunks, but not supported on ISDN trunks. On ISDN trunks, if the OPAO feature is enabled, the asterisk (*) is outpulsed as a called party digit.

Chapter 47: First-second Degree Busy Indication

Contents

This section contains information on the following topics:

Feature description

This feature provides an attendant with an indication of whether a party is first degree or second degree busy. If party A is established on a call to party B, and the attendant tries to connect to party A, party A is considered to be first degree busy. If party C is camped-on or call waiting to party A, party A is then considered to be second degree busy.

Operating parameters

There are no operating parameters associated with this feature.

Feature interactions

There are no feature interactions associated with this feature.

Feature packaging

This feature is packaged in International Supplementary Features (SUPP) package 131.

Feature implementation

Table 100: LD 15 - At the OPT prompt, deny or allow Attendant Busy Display.

Prompt	Response	Description
REQ:	NEW CHG	Add, or change.
TYPE:	ATT_DATA	Attendant console options
...		
- OPT	(ABDD) ABDA	Attendant Busy Display (denied) allowed.

Feature operation

The first degree busy indication is as normal. For second degree busy indication, normal busy tone is given to the attendant, and the display -O (meaning Occupied Second Degree) is given on the last four right-hand spaces of the console display.

Chapter 48: Flexible Attendant Call Waiting Thresholds

Contents

This section contains information on the following topics:

Feature description

When there are no calls waiting in the attendant queue, the Call Waiting Lamp on all attendant consoles is dark. The lamp is lit as soon as the first call arrives that can not be presented to a console.

When the number of calls waiting in the attendant queue exceeds the upper threshold, defined by the CWCL prompt in LD 15, the Call Waiting Lamp (CWL) state on all attendant consoles is changed from lit to flash (60 impulses per minute).

When the number of calls waiting in the attendant queue drops below the lower threshold, defined by the CWCL prompt in LD 15, the CWL state on all attendant consoles is changed from flash to lit.

When there are no more calls waiting in the attendant queue, the CWL is turned off.

The Flexible Attendant Call Waiting Thresholds (FACWT) feature allows the thresholds to be defined as a percentage of the active consoles, consoles which are not in Position Busy or Night Service, or as a fixed number. The feature is activated on a customer basis by responding with FACA (Flexible Attendant Call Waiting Thresholds Allowed) to the OPT (Option) prompt in LD 15.

Operating parameters

The upper threshold must be greater than or equal to the lower threshold.

The maximum number of attendants multiplied by the threshold maximum percentage must equal less than 65 535 (due to storage requirements).

Feature interactions

Attendant Overflow Position

The Attendant Overflow Position is not counted as an active attendant.

Recall to Same Attendant

The Recall to Same Attendant (RTSA) feature has precedence over the Flexible Attendant Call Waiting Thresholds feature. If either RSAA or RSXA options are selected, RTSA has precedence over FACWT in determining the Call Waiting Lamp state. If one or more RTSA calls are waiting in the attendant queue, RTSA will set the Call Waiting Lamp state to wink (30 impulses per minute).

RTSA calls are not included when the FACWT feature determines the number of calls waiting.

Feature packaging

The Flexible Attendant Call Waiting Thresholds is packaged under International Supplementary Features (SUPP) package 131.

Feature implementation

Task summary list

The following is a summary of the tasks in this section:

1. Table 101: LD 15 - The Customer Data Block on page 333

   The Customer Data Block service change accepts the options FACD and FACA to be defined as a customer option. The range and usage of the CWCL thresholds is defined by the FAC option selected. To allow the calls waiting thresholds to be defined as percentages respond to the OPT prompt with FACA. To allow the calls waiting thresholds to be defined as number of calls respond to the OPT prompt with FACD.

2. Table 102: LD 21 - Print Routine 2 is modified to include OPT FACD or FACA setting and the new CWCL range settings in the Customer Data Block printout. on page 334

   Print Routine 2 is modified to include OPT FACD or FACA setting and the new CWCL range settings in the Customer Data Block printout.

3. Table 103: LD 93 - As for the Customer Data Block, the CWCL threshold usage is changed with the selection of a FAC option in the Customer Data Block. on page 334

   As for the Customer Data Block, the CWCL threshold usage is changed with the selection of a FAC option in the Customer Data Block.

Table 101: LD 15 - The Customer Data Block

Prompt	Response	Description
REQ:	CHG NEW	Modify or create data block.
TYPE:	ATT_DATA	Attendant console options
CUST		Customer number
	0-99	Range for Large System and Avaya Communication Server 1000E (Avaya CS 1000E) system.
...		
- OPT	(FACD) FACA	Options for customer: (Flexible Attendant Call Waiting Thresholds Denied), Flexible Attendant Call Waiting Thresholds Allowed. (Denies), Allows the attendant Call

Prompt	Response	Description
		Waiting thresholds to be defined as a percentage of active attendants.
...		
- CWCL	xxxyyy	Call Waiting Lamp thresholds Where xxx defines the lower threshold and yyy defines the upper threshold.
	(0)-255(0)-255	Valid ranges for number of calls when FACD is entered in response to OPT.
	(0)-1000(0)-1000	Valid ranges for percentages when FACA is entered set in response to OPT.

The Customer Data Block service change accepts the options FACD and FACA to be defined as a customer option. The range and usage of the CWCL thresholds is defined by the FAC option selected. To allow the calls waiting thresholds to be defined as percentages respond to the OPT prompt with FACA. To allow the calls waiting thresholds to be defined as number of calls respond to the OPT prompt with FACD.

Table 102: LD 21 - Print Routine 2 is modified to include OPT FACD or FACA setting and the new CWCL range settings in the Customer Data Block printout.

Prompt	Response	Description
REQ	PRT	Request: Print data block.
TYPE	CDB	Customer Data Block
CUST	xx	Customer number, as defined in LD 15

Table 103: LD 93 - As for the Customer Data Block, the CWCL threshold usage is changed with the selection of a FAC option in the Customer Data Block.

Prompt	Response	Description
REQ	CHG NEW PRT	Request: Modify, create or print data block.
TYPE	CPGP	Console Presentation Group Parameters.
CUST	xx	Customer number, as defined in LD 15
CPG	1-63	Console Presentation Group: ACG (Attendant Console Group) number.
...		
AODN	...	
CWCL	xxxyyy	Call Waiting Lamp thresholds Where xxx defines the lower threshold and yyy defines the upper threshold.
	(0)-255(0)-255	Valid ranges for number of calls when FACD is set in response to OPT in LD 15.

Prompt	Response	Description
	(0)-1000(0)-1000	Valid ranges for percentages when FACA is set in response to OPT in LD 15.

Feature operation

If the customer has the FACA option selected in the Customer Data Block (LD 15) the thresholds are defined as a percentage of the number of active attendants. The thresholds are specified on a customer and tenant Console Presentation Group (CPG) level basis. If the Flexible Attendant Call Waiting Thresholds Denied (FACD) option is selected, the thresholds are defined as fixed numbers and the operations remain the same as when this feature is not used.

When the FACA option is used, the CWL state is updated each time the number of calls waiting or the number of active attendants changes. Any integer between 0-1000 can be set for the Call Waiting thresholds percentage. The following tables illustrate the operation when FACA is selected and the lower limit is defined as 100 percent of active attendants and the upper limit is defined as 200 percent of active attendants (CWCL 100 200).

Table 104: Upper and lower limits of calls waiting versus number of active attendants

Number of active attendants	Number of calls waiting in queue to achieve 100% lower limit	Number of calls waiting in queue to achieve 200% upper limit
1	1	2
2	2	4
3	3	6

Table 105: CWL state versus number of active attendants

Number of active attendants	Number of calls in queue												
	0	1	2	4	6	8	6	4	3	2	1	0	
1	D	L	L	F	F	F	F	F	F	F	L	D	CWL state
2	D	L	L	L	F	F	F	F	F	L	L	D	
3	D	L	L	L	L	F	F	F	L	L	L	L	

Number of active attendants	Number of calls in queue												
	0	1	2	4	6	8	6	4	3	2	1	0	
Legend: D = Dark, L = Lit, F = Flash.													

Chapter 49: Flexible Attendant Directory Number

Contents

This section contains information on the following topics:

Feature description

The Flexible Attendant Directory Number (FADN) specifies the Directory Number (DN) that provides access to the attendant, replacing the usual 0. The DN may be any DN in the numbering plan, but it must be used only for the attendant and in all situations in which 0 is normally used.

Operating parameters

The attendant DN may be used only for the attendant. One attendant DN is allowed per customer and all attendants must have the same DN.

Feature interactions

Directory Number Expansion

The attendant DN can have up to seven digits if the Directory Number Expansion (DNXP) package is equipped.

Feature packaging

This feature is included in base system software.

Feature implementation

Table 106: LD 15 - Define or change the attendant Directory Number.

Prompt	Response	Description
REQ:	CHG	Change.
TYPE:	ATT_DATA	Attendant console options
...		
- ATDN	xxx...x	Number dialed to reach the attendant (the default is 0).

Feature operation

No specific operating procedures are required to use this feature.

Chapter 50: Flexible Busy Tone Timer

Contents

This section contains information on the following topics:

Feature description

The feature provides a flexible length of time that a caller on a Direct Inward Dialing (DID) route hears busy or overflow tone, when it is normally encountered. The time that the tone is presented is overlay programmable from 2 to 254 seconds.

Operating parameters

There are no operating parameters associated with this feature.

Feature interactions

There are no feature interactions associated with this feature.

Feature packaging

This feature is packaged in the International Supplementary Features (SUPP) package 131.

Feature implementation

Table 107: LD 16 - Set data for Flexible busy or overflow time to implement the flexible Busy Tone Timer feature:

Prompt	Response	Description
...		
BTT	2-(30)-254	Enter busy/overflow time to be returned on DID routes in seconds.

Feature operation

No specific operating procedures are required to use this feature.

Chapter 51: Flexible CLID Manipulation Table

Contents

This section contains information on the following topics:

Feature description

Use the Flexible CLID manipulation table to configure matching rules and replacement rules linked to an incoming route and outgoing RLI for the Calling Line Identification (CLID).

If all conditions in the matching rule of the first tuple in the Flexible CLID manipulation table are satisfied, the CLID is modified according to the corresponding replacement rule. If all conditions in the matching rule are not satisfied, the next tuple is examined in an attempt to match its matching rule. If there are no configured tuples for which all conditions in the matching rule are satisfied, the CLID is left unchanged.

Note:
The CLID Manipulation Data Block (CMDB) feature is not supported for Local Termination (LTER).

Operating parameters

There are no operating parameters associated with this feature.

Feature interactions

This feature is supported only for Calling numbers.

This feature is supported only for incoming route or outgoing RLI of UIPE, non-UIPE and DASS/DPNSS calling numbers.

Configure up to 256 Flexible CLID manipulation tables. Include a maximum of 16 tuples in each Flexible CLID manipulation table.

The E.164 Enhancement feature interacts with the Flexible CLID Manipulation Table feature. For more information, see E.164 Enhancement on page 215 in this book.

Feature packaging

This feature is included in the base system software.

Feature implementation

Configure the CLID Manipulation Data Block (CMDB) in LD 86. For more information, see *Software Input Output Reference - Administration, NN43001-611*.

Each CMDB has a set of tuples, matching rules and replacement rules. An example tuple is : <RULENUM> <MNUM> <MID> <MNPI> <MTON> <MDR><MNOD> <DEL> <INST> <RNPI> <RTON>.

Table 108: Matching rules

Prompt	Response	Description

MNUM	`CLNG (Calling number)`
	`DC (Don't Change)`
	`CONN (Connected number)`
		Matching Number Type

MID	`X-XXXXXXX up to 8 digits; each digit can be between 0 and 9.` `DC (Don't change)`	Matching Initial Digits
MNPI	`E164, PRIV, E163, TELE, X121, NATL` `DC (Don't change)`	Matching Numbering Plan Indicator
MTON	`UKWN, INTL, NATL, SPN, LOCL, ELOC, CDP, NCHG, CCS7`	Matching Type of Number

MDR	`GT (greater than),`
	`EQ (equal to),`
	`LT (lesser than),`
	`DC (Don't change)`
		Matching Digit Relation

MNOD	`0-32,` `DC (Don't change)`	Matching Number of Digits

Table 109: Replacement rules

Prompt	Response	Description
DEL	`1 - 32 digits,` `NCGH (No change)`	Delete initial digits
INST	`X-XXXXXXX up to 8 digits; each digit can be between 0 and 9.` `NCGH (No change)`	Insert initial digits
RNPI	`E164, PRIV, E163, TELE, X121, NATL` `NCGH (No change)`	Replacement Numbering Plan Indicator
RTON	`UKWN, INTL, NATL, SPN, LOCL, ELOC, CDP, NCHG, CCS7`	Replacement Type of Number

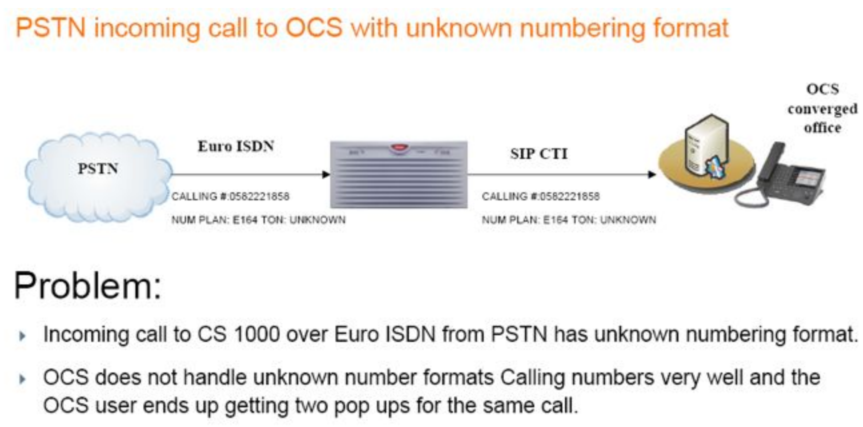
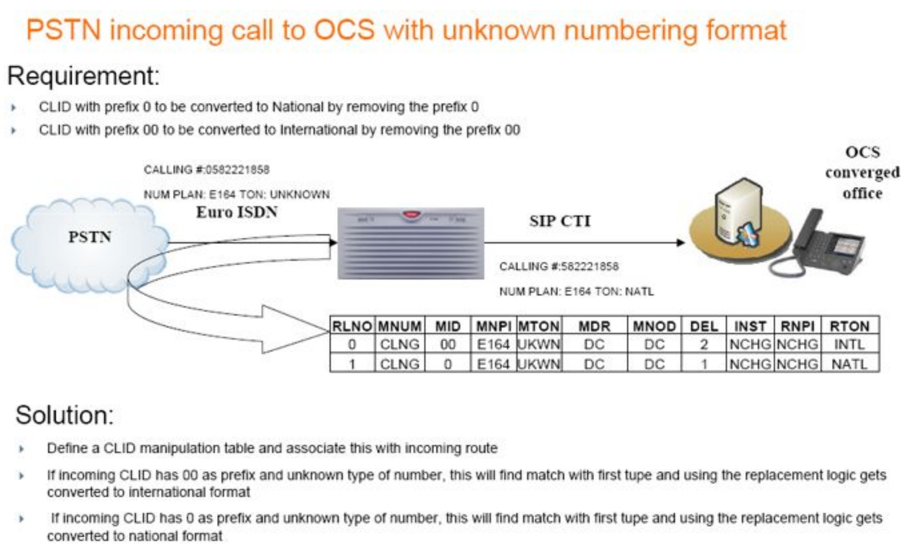


Figure 12: Incoming call to OCS with unknown numbering format



RLNO	MNUM	MID	MNPI	MTON	MDR	MNOD	DEL	INST	RNPI	RTON
0	CLNG	00	E164	UKWN	DC	DC	2	NCHG	NCHG	INTL
1	CLNG	0	E164	UKWN	DC	DC	1	NCHG	NCHG	NATL

Figure 13: Solution to incoming call to OCS with unknown numbering format

Next example depicts a regulatory requirement that the CLID displayed from an outbound contact center must be a dialable number.

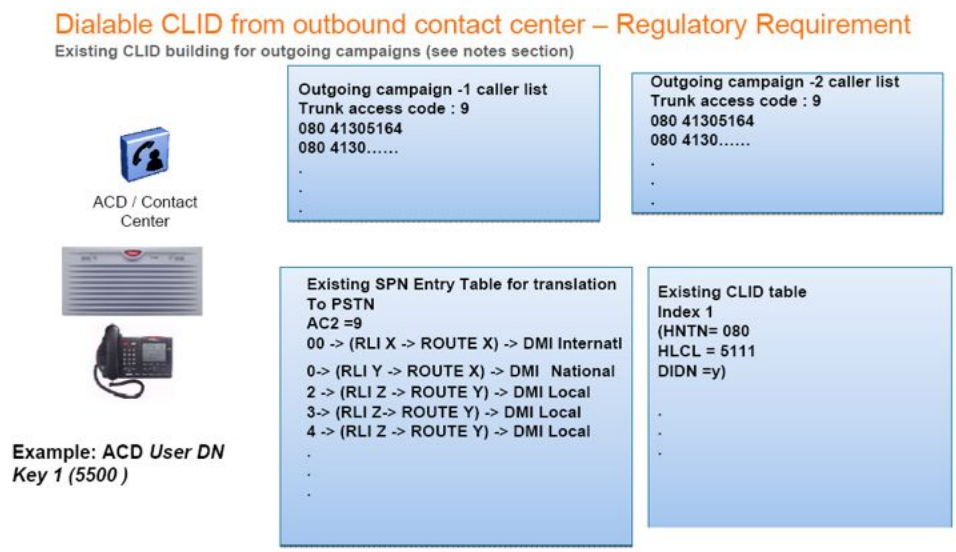


Figure 14: Existing CLID for outgoing campaigns

example

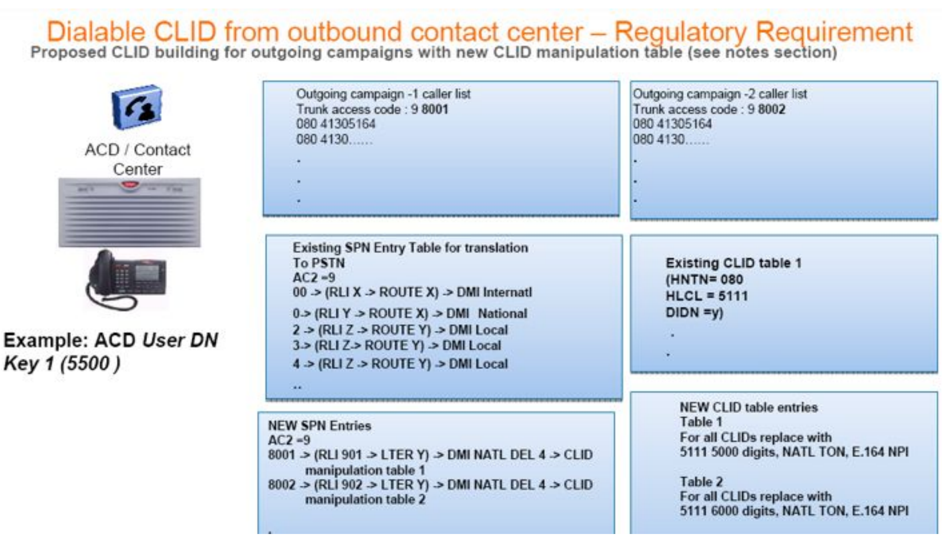


Figure 15: Proposed CLID for outgoing campaigns

Associating a CMDB table to a route or RLI entry

The following table shows an example configuration:

```
**** CMDB **** TBNO 1

-------------------------------------------------------------------------------
RLNO    MNUM    MID      MNPI    MTON    MDR     MNOD    DEL     INST    RNPI    RTON
-------------------------------------------------------------------------------
1       CLNG    10       PRIV    CDP     EQ      4       2       5674     PRIV    CDP
-------------------------------------------------------------------------------

To associate the CMDB table with the RLI,

In overlay 86, rlb
REQ new/chg
CUST 1
FEAT rlb
RLI 1
ENTR 0
LTER NO
ROUT xx
.
.
.
CTBL 1


To associate the table on the incoming route

In overlay 16, rdb
REQ new/chg
TYPE rdb
CUST 1
ROUT xx
.
.
.
ARDN
CTBL 1
```

Matching Digit Relation (MDR)

In the following configuration, the MDR value is EQ (equal to) and the MNOD (matching number of digits is 4). CMDB conversion occurs only if the calling number has 4 digits.

```
**** CMDB **** TBNO 1

-------------------------------------------------------------------------------
RLNO    MNUM    MID      MNPI    MTON    MDR     MNOD    DEL     INST    RNPI    RTON
-------------------------------------------------------------------------------
1       CLNG    10       PRIV    CDP     EQ      4       2       5674     PRIV    CDP
-------------------------------------------------------------------------------
```

In the following configuration, the MDR value is GT (greater than) and the MNOD (matching number of digits is 2). CMDB conversion only occurs if the calling number contains more than 2 digits.

```
**** CMDB **** TBNO 1

-------------------------------------------------------------------------------
RLNO    MNUM    MID      MNPI    MTON    MDR     MNOD    DEL     INST    RNPI    RTON
-------------------------------------------------------------------------------
1       CLNG    10       PRIV    CDP     GT      2       2       5674     PRIV    CDP
-------------------------------------------------------------------------------
```

Feature operation

You do not need any specific operating procedures to use this feature.

Chapter 52: Flexible Dial Tone Detection

Contents

This section contains information on the following topics:

Feature description

The Flexible Dial Tone Detection (FDTD) feature permits the system to wait for and detect a Second Dial Tone (SCDT) before automatic or manual dialing of outgoing toll calls. The wait-for-tone position in the digit outpulsing is user configurable, thus providing flexible digit validation. This feature is an enhancement to the Dial Tone Detection (DTD) feature.

The break-in outpulsing can occur after a defined digit sequence, or after a defined number of digits have been outpulsed. Digit outpulsing is halted and the Dial Tone Detector is reconnected. With the FDTD feature, it is no longer necessary to use the * to create pauses in outpulsing.

This feature has the following three options:

Dial Tone Position (DTP)

With the DTP option an Outgoing Access Code (OAC) is selected. Then FDTD verifies the dialed digits against the OAC (for example, country code) of up to four digits. When DTP is set, only the OAC digits are outpulsed before the DTD is reconnected. The DTP is the position immediately after the OAC. Up to four OACs can be specified.

Count Detection (CNT)

With the CNT option, the system will send a pre-defined number of digits (up to fifteen) before digit outpulsing is halted and the DTD is reconnected. Digit counting is done either one digit at a time, or as a string if fast Tone and Digit Switch (TDS) outpulsing is set up.

Digit Sequence (DGTS)

With the DGTS option, a table of up to 245 entries could be created of unique one-to-four digit sequences where the DTD should be reconnected after.

Operating parameters

There are no operating parameters associated with this feature.

Feature interactions

There are no feature interactions associated with this feature.

Feature packaging

This feature is packaged under Dial Tone Detector (DTD) package 138.

Feature implementation

Task summary list

The following is a summary of the tasks in this section:

1. Table 110: LD 16 - Set data for Flexible busy or overflow time. on page 351

Set data for Flexible busy/overflow time.

2. Table 111: LD 56 - Create tone and ringing parameters for one or more customers. on page 351

   Create tone and ringing parameters for one or more customers.

Table 110: LD 16 - Set data for Flexible busy or overflow time.

Prompt	Response	Description
...		
DTD	(NO) YES	Dial Tone Detection (is not) is to be performed on this route.
SCDT	(NO) YES	Secondary dial tone (will not) will be used on route.

Table 111: LD 56 - Create tone and ringing parameters for one or more customers.

Prompt	Response	Description
...		
TYPE	FDTD	Flexible Dial Tone Detection data.

Feature operation

No specific operating procedures are required to use this feature.

Chapter 53: Flexible Direct Inward Dialing

Prior to the introduction of the Flexible Direct Inward Dialing (FDID) feature, hotels were required to purchase a large number of DID numbers that matched the number of hotel guest rooms. These DID DNs must be coordinated with the local exchange and become permanent in the system.

The FDID feature allows hotels to assign a temporary DID number to a guest room using a Property Management System (PMS) or Background Terminal (BGD).

For more information, see *Avaya Hospitality Features: Description and Operation, NN43001-553*.

Chapter 54: Flexible Feature Codes

Contents

This section contains information on the following topics:

Feature description

Flexible Feature Codes (FFCs) are user-defined numbers of up to four digits that can be used in place of existing Special Prefix (SPRE) codes. With DN Expansion (DNXP) package 150, Flexible Feature Codes (FFCs) can be up to seven digits long. The Flexible Feature Code (FFC) feature allows customers to define different dialing codes for different features. There is no limit to the number of FFCs per prompt as long as each one is unique.

This feature allows the use of digits 0 through 9, and the asterisk (*) and octothorpe (#) to activate features. Special Prefix (SPRE) dialing feature is still supported, with or without the FFC feature enabled. However, the Special Prefix (SPRE) must be assigned in LD 15 in order for FFCs to operate for those features that also use SPRE codes.

The FFC package allows analog (500/2500 type) telephones to activate these features:

- Automatic Wake Up (AWU)
- Electronic Lock (ELK)
- Override, and
- Remote Call Forward (RCFW).

Customers define one or more codes at their discretion in LD 57 (FFC). For Service Change updates, see *Avaya Software Input Output Reference — Administration, NN43001-611*.

The basic FFC operation allows a telephone to access features normally available by dialing SPRE codes. FFCs are not supported, however, on a Meridian 1 proprietary telephone that is attempting a call pickup on a Dial Intercom ringing call.

A telephone can access a feature using FFC only if that telephone can currently access the same feature using SPRE dialing.

Any telephone, that does not currently have SPRE access, receives intercept treatment when dialing FFCs. Telephone operation remains the same (only the codes are different) so that the FFC code is dialed instead of the SPRE code. Therefore, each feature enabled must have an FFC individually defined.

When FFCT is YES in LD 57, the system returns a confirmation tone to the user after completing some feature operations.

The confirmation tone is the same as the special dial tone.

FFC allows analog (500/2500 type) telephones to Override established calls, based on the telephone's programmed Class of Service. Analog (500/2500 type) telephones can also activate and deactivate Call Forward by dialing a single FFC.

The confirmation Tone for FFC allows analog (500/2500 type) telephones and Meridian 1 proprietary telephones to receive a special tone when certain functions are complete. Confirmation Tone is returned following these events:

- Automatic Wake Up (any function)
- Call Forward (deactivate)
- Electronic Lock (any function)
- Ring Again (activate or deactivate)
- Room Status (any function)
- Speed Call Controller (add to Speed Call list), and
- Store Number (erase).

Confirmation Tone for FFC is returned when a predefined string is used as the end-of-dialing indicator for the following activities:

- Call Forward (activate)
- Permanent Hold (any function)
- Speed Call (store)
- Store Number (store), and
- Flexible Feature Code (any verification).

Confirmation Tone is provided for Speed Call store after the End-of-Dial string, such as the octothorpe (#), is entered.

Operating parameters

The SPRE feature must exist in order for FFC to operate.

The FFCs selected must be unique numbers up to seven digits long. They cannot conflict with any Directory Number (DN) already in the dialing plan.

LD 57 can allow no more than 100 FFCs to be modified in a single pass through Service Change.

Customers using the octothorpe (#) as part of their dialing plan can use a predefined string of digits for end-of-dialing indicators.

Changes to the Station Control Passwords (SCPWs) do not take affect until after a data dump and SYSLOAD. Configuring the system or enabling the feature changes SCPL = 0 in LD 15 to any length. This change takes effect immediately. Any other change to SCPL in LD 15 requires a data dump and SYSLOAD before taking effect. When the Station Control Password Length (SCPL) is changed, all associated passwords change accordingly at the next data dump and SYSLOAD. Changing SCPL from three to five automatically inserts leading zeros before all existing three-character passwords. Conversely, changing SCPL from five to three automatically truncates the leading characters of all existing five-character passwords.

Feature interactions

Attendant Blocking of Directory Number

If a Flexible Feature Code is dialed after pressing the Semi-automatic Camp-on (SACP) key to initiate an Attendant Blocking of DN attempt, overflow tone will be provided and the attempt cancelled.

Automatic Wake Up

Telephones can activate Automatic Wake Up (AWU) features for their own station with Common Controlled Switching Arrangement Class of Service.

The Automatic Wake Up feature may be active at the same time as Multiple Wake Up.

The attendant query function is not supported for Multiple Wake Up.

Multiple Wake Up from Attendant Consoles is not supported.

The Background Terminal (BGT) is not supported for Multiple Wake Up.

If one Automatic Wake Up time has been set using the Automatic Wake Up Activate (AWUA) FFC, only three additional Multiple Wake Up calls may be entered using the Multiple Wake Up Activate (MWUA) FFC.

Call Forward All Calls

When FFC is configured for a customer, #1 automatically becomes the FFC DN for both Call Forward Activate (CFWA) and Call Forward Deactivate (CFWD). When the same DN is used for both CFWA and CFWD, FFC toggles the Call Forward activated/deactivated state of the telephone. When Call Forward is activated for a telephone, entering #1 automatically deactivates Call Forward, no matter what follows #1. When Call Forward is deactivated for a telephone, the result of entering #1 depends on what follows:

- If the telephone goes on hook immediately, Call Forward is activated for the telephone to its previous call forward number.
- If a valid DN is entered after #1, call forward is activated for the telephone to that valid DN.
- If an invalid DN is entered after #1, call forward remains deactivated for the telephone.

Call Forward Attendant and Network-Wide Remote

If the Outpulsing of Asterisk and Octothorpe (OPAO) package is equipped, the octothorpe (#) is treated as a dialed digit and does not signal the end of dialing. From one to three end-of-entry characters are defined in LD 15.

Call Pickup Call Pickup, Directed

FFC codes are not supported on a Meridian 1 proprietary telephone during an attempt to pick up a Dial Intercom ringing call.

China - Flexible Feature Codes - Outgoing Call Barring

Flexible Feature Codes containing a "*" or an "#" will always be allowed by Outgoing Call Barring (OCB). Therefore, FFCs which can be used to make a call should be entirely numeric if barring of them is required.

Some FFCs are equivalent to Special Prefix functions and these will be subject to barring based on the equivalent Special Prefix codes, even if the FFC is entirely numeric.

Controlled Class of Service

If Electronic Lock (ELK) is activated, the CCRS Class of Service is used whether Controlled Class of Service (CCOS) is active or not. ELK takes precedence over CCOS. If ELK is deactivated, the set is treated as per existing operation.

When FFC ELKA and a password is entered, this set will use the CCRS Class of Service configured in LD 15. The CCRS Class of Service will always be used whether or not CCOS is currently controlling the set's Class of Service. When FFC ELKD and a password is entered, the set will use the appropriate Class of Service associated with this set. If CCOS is enabled for the set, the associated customer Class of Service is used (that is, CCRS, ECC1, or ECC2). If CCOS is not enabled for this set, the set's own Class of Service is used.

When FFC ELK is deactivated, the set reverts back to the Class of Service as it should be without FFC ELK, instead of always reverting back to the set's Class of Service (that is, if CCOS is enabled, it will use the customer's Class of Service; if CCOS is not enabled, it will use the set's Class of Service).

Intercept Treatment

If Intercept Treatment has been specified for a call to a vacant number (CTVN), the Digit Display (DDs) on the attendant console is affected by Flexible Feature Codes (FFCs). If no FFC has been defined, the dialed digits are displayed up to and including the first digit that fails to match any Directory Number (DN). If one or more FFCs have been defined, the dialed digits are displayed, up to and including the first digit that fails to match any FFC.

ISDN QSIG and EuroISDN Call Completion

Analog (500/2500 type) set can use Flexible Feature Codes (FFCs) to activate Call Completion to Busy Subscriber requests.

Pretranslation

Flexible Feature Codes must be accessible through a Pretranslation Table entry in order for users to activate features in this manner.

The Flexible Feature Code (FFC) feature will not be affected if the FFCs begin with "*" or "#", since before translation begins if the first digit is an "*" or "#" pretranslation will not be done. If any digits follow the FFC code, the first of the digits that follows will be pretranslated.

Special Prefix

Users are still able to use Special Prefix (SPRE) dialing (if the feature is enabled) with or without FFC defined.

Speed Call, System

With Flexible Feature Code (FFC), a confirmation tone is provided for Speed Call store after the end-of-dial (EOD) string is entered.

Feature packaging

Flexible Feature Codes (FFC) package 139 requires Controlled Class of Service (CCOS) package 81 only if Electronic Lock (ELK) is desired.

In addition, the SPRE dialing feature must be enabled for FFC functions.

2500 Telephone Features (SS25) package 18, and 500 Set Dial Access to Features (SS5) package 73 are required to support the following features:

- Call Forward
- Speed Call Controller
- Speed Call User
- Permanent Hold
- Call Park, and
- System Speed Call.

Feature implementation

Task summary list

The following is a summary of the tasks in this section:

1. Table 112: LD 15 - Set parameters for Flexible Feature Code. on page 361

Set parameters for Flexible Feature Code.

2. Table 113: LD 10 - Set Station Control Password Length for analog (500 and 2500 type) telephones. on page 361

   Set Station Control Password Length for analog (500/2500 type) telephones.

3. Table 114: LD 11 - Set Station Control Password Length for Meridian 1 proprietary telephones. on page 362

   Set Station Control Password Length for Meridian 1 proprietary telephones.

4. Table 115: LD 57 - Define numbers for Flexible Feature Code. on page 362

   Define numbers for Flexible Feature Code.

Table 112: LD 15 - Set parameters for Flexible Feature Code.

Prompt	Response	Description
REQ:	CHG	Change.
TYPE:	CCS	Controlled Class of Service options.
CUST		Customer number
	0-99	Range for Large System and Avaya Communication Server 1000E (Avaya CS 1000E) system.
- CCRS	aaa	Controlled Class of Service (CCOS) (assigned when Electronic Lock (ELK) is activated), where: aaa = UNR (Unrestricted), TLD (Toll Denied), CTD (Conditionally Toll Denied), CUN (Conditionally Unrestricted), SRE (Semi-restricted), FRE (Fully Restricted), FR1 (Fully Restricted Level 1), FR2 (Fully Restricted Level 2).
TYPE	FFC	Flexible Feature Code Options.
- SCPL	x	Station Control Password Length (SCPL), 0-8. Entering 0 disables ELK and Remote Call Forward (RCFW) features at next data dump and SYSLOAD.
- FFCS	(NO) YES	(Do not) change FFC end-of-dialing indicator.
- - STRL	x	String length 1-3 (prompted only if FFCS = YES).
- - STRG	aaa	Character string to be used (up to string length; prompted only if FFCS = YES).

Table 113: LD 10 - Set Station Control Password Length for analog (500 and 2500 type) telephones.

Prompt	Response	Description
REQ:	CHG	Change.
TYPE:	500	Telephone type.

Prompt	Response	Description
TN		Terminal number
	l s c u	Format for Large System, Avaya MG 1000B, and CS 1000E system, where l = loop, s = shelf, c = card, u = unit.
SCPW	xx...xx X	Station Control Password (must be same length as SCPL in LD 15; enter X to delete password).
CLS	CCSA	Enable CCOS for Electronic Lock (ELK) and Remote Call Forward (RCFW).

Table 114: LD 11 - Set Station Control Password Length for Meridian 1 proprietary telephones.

Prompt	Response	Description
REQ:	CHG	Change.
TYPE:	a...a	Telephone type. Type **?** for a list of possible responses.
TN		Terminal number
	l s c u	Format for Large System, MG 1000B, and CS 1000E system, where l = loop, s = shelf, c = card, u = unit.
SCPW	xx...xx X	Station Control Password. Must be the same length as SCPL in LD 15. Enter X to delete the password. Delete the password only if SCPL = 0; otherwise receive an error code for no password to fit the SCPL.
CLS	CCSA	Enable CCOS for ELK and Remote Call Forward (RCFW).

Table 115: LD 57 - Define numbers for Flexible Feature Code.

Prompt	Response	Description
REQ	NEW CHG OUT	Build new FFC data block, change FFC data block, remove FFC code.
TYPE	FFC	Flexible Feature Codes.
CUST	xx	Customer number, as defined in LD 15
FFCT	(NO) YES	FFC Confirmation Tone.
CEPT	(NO) YES	Conférence Européen des Postes Tel defaults are (not) allowed, to be defined (prompted only if REQ = NEW).
REP*	n <CR>	Single-character replacement for * and # in CEPT defaults. Create defaults only.

Prompt	Response	Description
ALL	(NO) YES	(Do not) remove all FFCs (prompted only if REQ = OUT).
CODE	aaaa ALL <CR>	FFC type. All prompts. No prompts.
- ADMN	xxxx	Set-based Administration sequence code.
- AFTO	xxxx	DSN Flash Precedence code.
- AREM	xxxx	Automatic Set Removal code.
- ASRC	xxxx	Automatic Set Relocation code.
- ATDA	xxxx	Autodial Activate code.
- ATDD	xxxx	Autodial Deactivate code.
- ATVF	xxxx	DSN Immediate Precedence code.
- ATVM	xxxx	Priority Precedence code.
- ATVP	xxxx	DSN Flash Override Precedence code.
- AUTH	xxxx	Authorization code.
- AVNR	xxxx	DSN Routine Call code.
- AUTH	xxxx	Authorization Code.
- AWUA	xxxx	Automatic Wake Up Activate code.
- AWUD	xxxx	Automatic Wake Up Deactivate code.
- AWUV	xxxx	Automatic Wake Up Verify code.
- BNRA	xxxx	Busy Number Redial code.
- BNRD	xxxx	Busy Number Redial Deactivate code.
- CCFA	xxxx .	Customer Call Forward code.
- CCFD	xxxx	Customer Call Forward Deactivate code.
- CDRC	xxxx	Call Detail Recording Charge Account code.
- CFDD	xxxx	Call Forward Destination Deactivation.
- CFHO	xxxx	Call Forward/Hunt Override code.
- CFWA	xxxx	Call Forward All Calls Activate code.
- CFWD	xxxx	Call Forward All Calls Deactivate code.
- CFWV	xxxx	Call Forward All Calls Verify code.
- COND	xxxx	Conference Diagnostics code.
- CPAC	xxxx	Park Access Call code.
- CPP	xxxx	Calling Party Privacy code.

Prompt	Response	Description
- CPPO	xxxx	Calling Party Privacy Override code.
- CPRK	xxxx	Park Call code.
- CSHF	xxxx	Centrex Switchhook Flash code.
- C6DS	xxxx	Six-Party Conference code.
- CWGA	xxxx	Call Waiting Activate code.
- CWGD	xxxx	Call Waiting Deactivate code.
- DEAF	xxxx	Deactivate Ring Again and FWD codes.
- DPVS	xxxx	Data Port Verification code.
- ELKA	xxxx	Electronic Lock Activate code.
- ELKD	xxxx	Electronic Lock Deactivate code.
- EOVR	xxxx	Enhanced Override code.
- FDIS	xxxx	Forced Disconnect code.
- GHTA	xxxx	Group Hunt Termination Allowed code.
- GHTD	xxxx	Group Hunt Termination Disallowed code.
- GRHP xxx	xxxx	Group Hunt Pilot DN
-- CFBA	(NO) YES	Call Forward Busy Activate
-- USE	(GPHT)	Initiate Group Hunting.
-- LSN	nnn	List number.
-- HTYP	aaa	Hunting type.
- GRPF	xxxx	Group Call code.
- GRCL	xxxx	Group Call List number.
- HIDN	xxxx	Hospitality Identification (Hospitality Management) code.
- HOLD	xxxx	Permanent Hold code.
- ICFA	xxxx	Internal Call Forward Activate code.
- ICFD	xxxx	Internal Call Forward Deactivate code.
- ICFV	xxxx	Internal Call Forward Verify code.
- INST	xxxx	Set-based Administration Installer code.
- HREL	xxxx	Hospitality Relocation (Hospitality Management) code.
- ICPA	xxxx	Intercept Computer Interface Activate code.
- ICPD	xxxx	Intercept Computer Interface Deactivate code.
- ICPO	xxxx	Intercept Computer Interface Override code.

Prompt	Response	Description
- ICPP	xxxx	Intercept Computer Interface Print code.
- IMS	xxxx	Integrated Message System access code.
- ITXX	xxxx	For â€œ1xxâ€ Special Services (up to four digits) CO route.
-- RTXX	xxxx	CO route number for the â€œ1xxâ€ service.
- LILO	xxxx	Log In-Log Out for 500/2500 ACD sets code.
- MLIO	xxxx	Multi-language I O code.
- MNT	xxxx	Enter set-based Maintenance sequence code.
- IMS	xxxx	Integrated Message System Access code.
- LILO	xxxx	Login, Logoff code for analog (500/2500 type) ACD telephones.
- MNTC	xxxx	Maintenance Access code.
- MSBA	xxxx	Make Set Busy Activate code.
- MSBD	xxxx	Make Set Busy Deactivate code.
- MTRC	xxxx	Malicious Call Trace code.
- MWRA	xxxx	Multiple Wake Up Repeat Activate code.
- MWUA	xxxx	Multiple Wake Up Activate code.
- MWUD	xxxx	Multiple Wake Up Deactivate code.
- NRDY	xxxx	Not Ready Activate or Deactivate code for analog (500/2500 type) ACD telephones.
- OVRD	xxxx	Override/Priority Override code.
- OCBA	xxxx	Outgoing Call Barring feature code.
- OCBD	xxxx	Outgoing Call Barring Deactivate code.
- OCBV	xxxx	Verify the Outgoing Call Barring feature code.
- PCAA	xxxx	Personal Call Assistant Activate code.
- PCAD	xxxx	Personal Call Assistant Deactivate code.
- PCAV	xxxx	Personal Call Assistant Verify code.
- PGAP	xxxx	Answer Parallel Paging code.
- PGIP	xxxx	Initiate Parallel Paging code.
- PGSP	xxxx	Initiate Serial Paging code.
- PLDN	xxxx	Pilot DN code.
-- USE	aaaa	Use (aaaa = GPHT, SCLU, or SCLC).
-- LSNO	nnnn	List Number.

Prompt	Response	Description
-- HTYP	aaa	Hunting Type (aaa = (LIN) or RRB)
-- CFWI	(NO) YES	Call Forward All Calls have priority over Group Hunting.
-- MQUE	aaaa	Limit to calls Queued against pilot DN (aaa = (ALL), 0, 1, or ACTM).
- PONW	xxxx	Priority Override Network Wide.
- PUDN	xxxx	Pick Up Directory Number code.
- PUGR	xxxx	Pick Up Group code.
- PUGR	xxxx	Pick Up Group code.
- PURN	xxxx	Pick Up Ringing Number code.
- RCFA	xxxx	Remote Call Forward Activate code.
- RCFD	xxxx	Remote Call Forward Deactivate code.
- RCFV	xxxx	Remote Call Forward Verify code.
- RDLN	xxxx	Redial Last Number code.
- RDNE	xxxx	Redial Number Erase code.
- RDSN	xxxx	Redial Saved Number code.
- RDST	xxxx	Redial Store code.
- RGAA	xxxx	Ring Again Activate code.
- RGAD	xxxx	Ring Again Deactivate code.
- RGAV	xxxx	Ring Again Verify code.
- RMST	xxxx	Room Status code.
- RPAN	xxxx	Radio Paging Answer Call code.
- RPAX	xxxx	Radio Paging Access code.
- SADS	xxxx	Scheduled Access Restriction Disable code.
- SAEN	xxxx	Scheduled Access Restriction Enable code.
- SALK	xxxx	Scheduled Access Restriction Lock code.
- SAUN	xxxx	Scheduled Access Restriction Unlock code.
- SCPC	xxxx	Station Control Password Change code.
- SFAC	xxxx	Secretarial Filtering Access code.
- SPCC	xxxx	Speed Call Controller code.
- SPCE	xxxx	Speed Call Erase code.
- SPCU	xxxx	Speed Call User code.
- SSPU	xxxx	System Speed Call User code.

Prompt	Response	Description
- TFAS	xxxx	Trunk Answer from Any Station code.
- TNDN	xxxx	Enter the DN-to-TN conversion utility code.
- TRMD	xxxx	Terminal Diagnostics code.
- TRVS	xxxx	Trunk Verification code.
- USCR	xxxx	User Selectable Call Redirection.
- USER	xxxx	Set-based Administration User code.
- USTA	xxxx	User Status code.
- VTLF	xxxx	Virtual Office Terminal Logoff. - VTLN xxxx Virtual Terminal Login
- VTLN	xxxx	Virtual Terminal Login.

Feature operation

For some features, the user can dial a different FFC to activate or deactivate a feature or to verify some feature operations. The tone for each event (activate, deactivate, verify) is the same as the default Confirmation Tone (special dial tone).

The Electronic Lock and Remote Call Forward FFCs are described here because Electronic Lock is packaged with Flexible Feature Codes and affects Remote Call Forward.

For information about using FFCs for other features, see the individual feature descriptions.

Electronic Lock

Electronic Lock (ELK), packaged with FFC, provides an SCPW for changing the status from the telephone. The SCPW also protects against changes to the Remote Call Forward (RCFW) feature. Entering a password length of 0 in LD 15 (SCPL) disables password control for both ELK and RCFW. Operating ELK requires enabling CCOS package 81.

To change the Class of Service from a telephone:

1. Dial the Electronic Lock Activate (ELKA) code.
2. Dial the SCPW. The telephone's Class of Service is changed to the CCRS value defined in LD 15.

To return the telephone to the originally defined Class of Service:

1. Dial the Electronic Lock Deactivate (ELKD) code.
2. Dial the SCPW. The telephone's Class of Service is changed to the values defined in LD 10 and LD 11.

Because the Class of Service defined for CCRS in LD 15 is usually lower than the Class of Service defined in LD 10 or LD 11, the Class of Service for a telephone is lowered by dialing the Electronic Lock Activate (ELKA) FFC and the password associated with that telephone. The user can activate from a remote telephone by dialing the ELKA FFC, the SCPW and the Directory Number to be changed. The same operation can deactivate the feature, using the Electronic Lock Deactivate (ELKD) code programmed in LD 57.

ELK operation has the following requirements:

- CCOS allowed, with CCSA Class of Service in LD 10 and LD 11, and CCRS defined in LD 15
- Set the password length in LD 15, at the SCPL prompt
- Add passwords in LD 10 and LD 11, at the SCPW prompt, and
- FFCT = YES in LD 57.

To change the SCPW for ELK:

1. Select a free extension.
2. Dial the SCPC code.
3. Dial the SCPW for your telephone.
4. Dial the new password.
5. To confirm, dial the new password again.
6. Hang up or press Rls.

Chapter 55: Flexible Feature Code Boss Secretarial Filtering

Contents

This section contains information on the following topics:

Feature description

The Flexible Feature Code Boss Secretarial Filtering (FFCSF) feature allows a set, designated as a "secretary" set, to filter calls coming in to a "boss" set. A boss or secretary set can be any Meridian 1 proprietary set or 16-button Dual-tone Multifrequency (DTMF) set. Filtering is a form of call screening, in which the calls coming into the boss set are presented to the secretary set to be answered and possibly transferred back to the boss set.

A boss set can have only one secretary set, while a secretary set can have an unlimited number of boss sets.

Operating parameters

A set cannot simultaneously be configured as a boss set and a secretary set.

The FFCSF Flexible Feature Code must be unique and not conflict with the customer dialing plan.

Secretary DNs which are programmed on a boss set cannot already be part of the customer's DN plan, nor conflict with it.

The FFCSF feature cannot be applied to sets having Multiple Appearance DNs.

In a networking environment, a boss set and secretary set must be on the same node.

Easy Change (ECHG) requests cannot be made against the Secretarial Filtering (SFLT) and Secretarial Forwarding DN (SFDN) prompts in LDs 10 and 11.

Feature interactions

Attendant Blocking of Directory Number

The FFC Boss Secretary Filtering feature will be overridden. If an Attendant Blocking of DN attempt is made for a set that has the Boss Secretary Filtering feature active, the dialed DN will be blocked if idle. If it is busy, busy tone will be heard.

Attendant-Extended Calls

Attendant-extended third-party calls to a boss set will be subject to filtering if filtering on the boss set is active for all calls. If filtering is allowed for external calls only, the attendant will be filtered only if the third party is external.

Call Forward All Calls

Although Call Forward All Calls and FFCSF can be equipped on the same set, they cannot both be active at the same time. There is no precedence of one over the other; it is not possible to activate one if the other is active on the set.

Call Forward Busy Call Waiting

A Call Forward Busy or Call Waiting to a boss set with filtering active is routed to the secretary set.

Call Forward and Busy Status

If the secretary set is a Meridian 1 proprietary telephone, or a compact digital set, it can be equipped with a Call Forward and Busy Status (BFS) key/lamp pair, to perform the following:

- monitor the status of the Call Forward feature on a boss set
- activate/deactivate the Call Forward feature on a boss set
- monitor whether or not a boss set is busy on a call, and
- override the Call Forward All Calls feature on a boss set, in order to place a call to the boss set.

The above functions, however, can only be performed by the secretary set while it is in an unattended state, since BFS and FFCSF cannot be active simultaneously.

Camp-On

When an attendant is attempting to Camp-on a call to a boss set with filtering active, the call is routed to the secretary set, if the filtering is active for all calls. If filtering is active for external calls only, the call is routed to the secretary set if the call is an external call.

Hot Line Private Line

FFCSF takes precedence over Private Line and Hot Line.

Hunting

A boss set with filtering activated is passed over by Hunting; the next hunt sequence is to the secretary set.

Lockout, DID Second Degree Busy, and MFE Signaling Treatment

Flexible Feature Code Boss Secretarial Filtering takes precedence over lockout and second degree busy.

Network Intercom

Hot Type I calls override this feature (for instance, Hot Type I calls are not filtered by FFC Boss Secretarial filtering). The call terminates on the Boss' set and is not forwarded to the secretary.

FFC Boss Secretarial Filtering takes precedence over enhanced Hot Type D calls. In this case, if FFC Boss Secretarial Filtering is active, calls terminate on the secretary's set.

Voice Call

A call to a Voice Call key on a boss set with filtering active is not filtered to the secretary set.

Feature packaging

This feature is packaged under Boss Secretarial Filtering (FFCSF), package 198.

Feature implementation

Task summary list

The following is a summary of the tasks in this section:

1. Table 116: LD 10 or LD 11 - Respond to the Secretarial Filtering (SFLT) prompt. on page 372

   Respond to the Secretarial Filtering (SFLT) prompt.
2. Table 117: LD 57 - Define the Secretarial Filtering Access Code. on page 373

   Define the Secretarial Filtering Access Code.

Table 116: LD 10 or LD 11 - Respond to the Secretarial Filtering (SFLT) prompt.

Prompt	Response	Description
...		

Prompt	Response	Description
SFLT	(NO) BOSS SEC	Secretarial Filtering, prompted with Boss Secretarial Filtering (FFCSF) package 198. Designate a telephone by entering either BOSS for boss set, SEC for secretary set, or NO for no designation. SEC, (NO), and <CR> take you to the next prompt.
- SFDN	xxxx	Secretarial Forwarding DN of secretary set to which filtered calls should be forwarded, prompted if response to SFLT = BOSS.

Table 117: LD 57 - Define the Secretarial Filtering Access Code.

Prompt	Response	Description
...		
SFAC	xxxx	Secretarial Filtering Access code.

Feature operation

The FFCSF feature may be accessed from the boss set and secretary set using the same Flexible Feature Code (FFC) followed by a control digit.

On a boss set, the following control digits can be dialed:

- 7, to activate filtering for all external calls
- 8, to activate filtering for all external and internal calls, and
- 9, to cancel filtering.

Confirmation tone is given to the boss set after filtering has been successfully activated or deactivated, or if filtering was already activated. Afterwards, a special dial tone (the same as the one used to indicate that Call Forward is active on a set) is provided to the boss set whenever it goes off-hook, as an audible reminder that the feature is active.

If filtering could not be activated by the boss set due to one of the following conditions, overflow tone is returned:

- the secretary set assigned to the boss set is not attended, or
- Call Forward All Calls is active on the boss set.

On a secretary set, the following control digits may be dialed:

- 8: to place the secretary set in attended state, allowing calls to be filtered to it from a boss set.
- 9: to place the secretary set in unattended state and to disable the boss set filtering.

In either case, confirmation tone is returned to the secretary set.

Chapter 56: Flexible Orbiting Prevention Timer

Contents

This section contains information on the following topics:

Feature description

The Orbit Prevention feature prevents an infinite loop from being created in a network-wide Call Forward configuration resulting from set A being call forwarded (all calls) to set B at another node, which in turn has been call forwarded back to set A. A check is provided through the Flexible Orbiting Prevention Timer (FOPT) that prohibits any set from call forwarding more than one call off-node for a period of 6 seconds.

The Orbit Prevention feature allows the Flexible Orbiting Prevention Timer (FOPT), to be service changeable from 0 to 30 seconds (even numbers only). If a value of 0 is defined, then Orbit Prevention is disabled and call forwarding is not inhibited in any way.

Operating parameters

There are no operating parameters associated with this feature.

Feature interactions

There are no feature interactions associated with this feature.

Feature packaging

This feature is included in base system software.

Feature implementation

Table 118: LD 15 - Enter an even value between 0-30 seconds, at the FOPT prompt to define the Orbit Prevention Timer.

Prompt	Response	Description
REQ:	NEW CHG	Add, or change.
TYPE:	NET	ISDN and ESN Networking Options.
...		
- FOPT	0-(6)-30	Flexible Orbiting Prevention Timer. The number of seconds in two-second intervals that CFW should be suspended on a set that has just forwarded a call off node. If an odd number is entered, the number is rounded up to the next even number, and the message "FOPT ROUNDED TO xx" is printed.

Feature operation

No specific operating procedures are required to use this feature.

Chapter 57: Flexible Tone and Digit Switch Control

Contents

This section contains information on the following topics:

Feature description

This feature allows the system to generate the many tones and cadences required for call processing in various countries. The system must be equipped with Flexible Tone and Digit Switch (TDS) circuit packs. One TDS pack is inserted in each network shelf in place of a network circuit pack.

The TDS packs are pre-overlay programmed with certain basic tone characteristics (frequencies, levels and cadences) which are then combined in various ways to produce the following tones:

- ACD ring-again ringback tone
- busy tone
- call forward dial tone
- call forward message-waiting dial tone
- camp-on confirm tone
- control dial tone

- dial tone
- dial-0 recall tone
- hold confirmation tone
- listed DN tone
- message waiting dial tone
- overflow tone
- preemption tone
- ringback tone
- test tone

These tones are also service changeable in LD 56. When call processing requires a particular tone, software sends the code defining that tone to the TDS pack. The TDS pack then generates the tone.

A number of other tones and associated cadences are available from the TDS but are assigned by software in LD 56. These are:

- agent observe tone
- call waiting tone
- intrusion tone
- override tone

The following tones are likely to be defined as bursts, but are still software controlled:

- ATV completion busy tone
- observe blocking tone
- off-hook queuing tone
- set relocate tone
- telset messaging alert tone
- telset messaging OK tone
- telset status update tone

Three exceptions to the categories of tones described so far are special dial tone, expensive route warning tone, and precedence call waiting tone. These tones are flexible only in their sound and not in their cadence.

Also included are distinctive or precedence ringing for analog (500/2500-type) and digital telephones. Refer to LD 56 in *Avaya Software Input Output Reference — Administration, NN43001-611* for the identification of these tones and cadences.

The tone and ringing requirements of the customer determine which TDS is required.

This feature also provides the following:

- an additional make/break ratio is available for ten pulses per second dialpulsing
- variable inter-digit pause time is flexible and can be assigned in LD 56 for digitone and dialpulse digits
- two additional DTMF outpulsing rates are available and assigned in LD 17

Operating parameters

There are no operating parameters associated with this feature.

Feature interactions

There are no feature interactions associated with this feature.

Feature packaging

This feature is packaged under Flexible Tones and Cadences (FTC) package 125.

Feature implementation

Task summary list

The following is a summary of the tasks in this section:

1. Table 119: LD 17 - Modify the system hardware and software parameters. on page 380

   Modify the system hardware and software parameters.
2. Table 120: LD 56 - Modify or change customer tone and ringing parameters. on page 380

   Modify or change customer tone and ringing parameters.

Table 119: LD 17 - Modify the system hardware and software parameters.

Prompt	Response	Description
...		
PARM	(NO) YES	Change system parameters.
ABCD	(NO) YES <CR>	16-Button DTMF operation is (is not) enabled. Original value is left unchanged.
DTRB	100	100 millisecond bursts of DTMF with 100 millisecond interdigit pause.
	50	50 millisecond bursts of DTMF with 50 millisecond interdigit pause.
	60	60 millisecond bursts of DTMF with 90 millisecond interdigit pause.
	70	70 millisecond bursts of DTMF with 70 millisecond interdigit pause.
NASA	YES	

Table 120: LD 56 - Modify or change customer tone and ringing parameters.

Prompt	Response	Description
...		
TYPE	FTC	Flexible tone and ringing.

Feature operation

No specific operating procedures are required to use this feature.

Chapter 58: Flexible Trunk to Trunk Connections

Contents

This section contains information on the following topics:

Feature description

The Flexible Trunk to Trunk Connections (FTT) feature controls trunk to trunk connections for Transfer, Supervised Conference, and unsupervised Conference, based upon the Station's Class of Service. This feature is used with or without the Trunk Barring (TBAR) feature. The Flexible Trunk to Trunk Connections feature provides the following options at a set level:

- allows trunk to trunk connections for Transfer and Conference
- denies trunk to trunk connections for Transfer and Conference
- allows trunk to trunk connections for Supervised Conference only, and denies trunk to trunk connections for Transfer and unsupervised Conference

The Conference feature allows additional parties to join an established call. One internal Directory Number must always be involved in the Conference call for a Supervised Conference. A system user can conference two or more trunks and then drop out of the conference, leaving the other trunks connected. This is an unsupervised Conference.

When Flexible Trunk to Trunk Connections is used in conjunction with the Trunk Barring feature, one of the following options may be selected:

- Additional set level limitations can be added to the existing Customer level Trunk Barring.
- The limitations placed by Trunk Barring, based upon the set's Flexible Trunk to Trunk Connections Class of Service, can be lifted.
- All set based trunk to trunk connections can be controlled for Conference and Transfer, depending upon the set's Flexible Trunk to Trunk Connections Class of Service, whether or not the route is barred by TBAR.

The functionality of the Flexible Trunk to Trunk Connections feature is activated by Flexible Trunk to Trunk Connections Options (FTOP prompt) in the Customer Data Block and controlled by the Station's Class of Service. The options that are available at a Customer level are dependent upon whether or not Trunk Barring (package 132) is configured.

Functionality of Flexible Trunk to Trunk Connections without Trunk Barring configured

When Flexible Trunk to Trunk Connections is used without Trunk Barring configured, the following Classes of Service are provided at a set level:

- When CLS = FTTU, Flexible Trunk to Trunk Connections Unrestricted, trunk to trunk connections are allowed for both Conference and Transfer. Flexible Trunk to Trunk Connections Unrestricted (FTTU) is the default value.
- When CLS = FTTR, Flexible Trunk to Trunk Connections Restricted (FTTR), trunk to trunk connections are denied for both Conference and Transfer.
- When CLS = FTTC, Flexible Trunk to Trunk Connections Conditional, trunk to trunk connections are allowed for Supervised Conference. Trunk to trunk connections are denied for Transfer and unsupervised Conference. Flexible Trunk to Trunk Connections Conditional (FTTC) is the default for new sets.

When Flexible Trunk to Trunk Connections is used without Trunk Barring configured, the following Flexible Trunk to Trunk Connections Options are available at a Customer level:

- When FTOP = FRES, Flexible Trunk to Trunk Connections Restricted, the Flexible Trunk to Trunk Connections feature does not function. The customer can still configure the set's Class of Service; however, the Class of Service does not take effect. Flexible Trunk to Trunk Connections Restricted (FRES) is the default value.
- When FTOP = FTLY, Flexible Trunk to Trunk Connections Only, trunk to trunk connections are controlled exclusively by the Flexible Trunk to Trunk Connections feature, based upon the set's Class of Service.

Figure 16: Functionality of Flexible Trunk to Trunk Connections without Trunk Barring configured on page 383 illustrates the functionality of Flexible Trunk to Trunk Connections without Trunk Barring configured.

In Figure 16: Functionality of Flexible Trunk to Trunk Connections without Trunk Barring configured on page 383, Set B is established with Trunk Route A and initiates a transfer or a conference with Trunk Route C.

Flexible Trunk to Trunk Connections Options (FTOP) = Flexible Trunk to Trunk Connections Only (FTLY)

Referring to Figure 16: Functionality of Flexible Trunk to Trunk Connections without Trunk Barring configured on page 383, when Flexible Trunk to Trunk Connections Options is set to Flexible Trunk to Trunk Connections Only (FTLY) and the Class of Service of Set B is set to Flexible Trunk to Trunk Connections Unrestricted (FTTU), the following is true:

- Telephone B can complete the Call Transfer between Trunk Routes A and C, as long as no other limitations apply.
- Telephone B can conference Trunk Routes A and C and then disconnect. In this case, Trunk Routes A and C remain connected, as long as no other limitations apply.

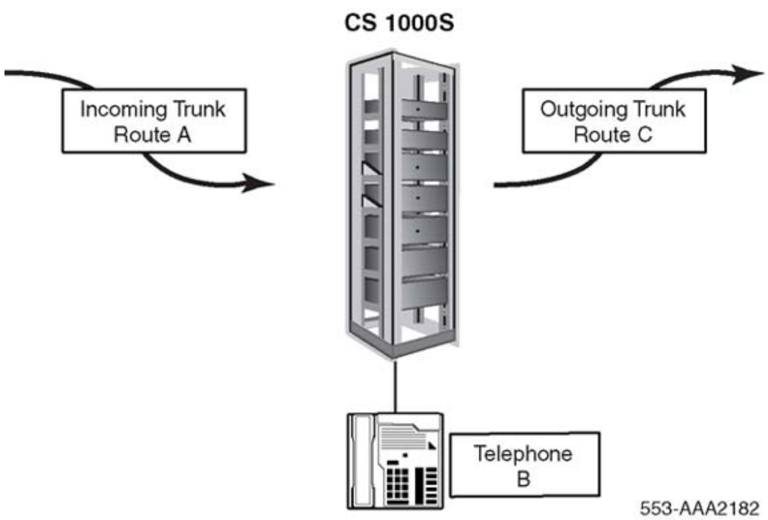


Figure 16: Functionality of Flexible Trunk to Trunk Connections without Trunk Barring configured

When Flexible Trunk to Trunk Connections Options is set to Flexible Trunk to Trunk Connections Only (FTLY) and the Class of Service of Set B is set to Flexible Trunk to Trunk Connections Restricted (FTTR), the following is true:

- Telephone B cannot transfer Incoming Trunk Route A to Outgoing Trunk Route C.
- Telephone B cannot complete the conference involving Trunk Routes A and C.

With Class of Service set to Flexible Trunk to Trunk Connections Restricted (FTTR), a consultation connection initiated by telephone B to Trunk Route C is not affected by Flexible Trunk to Trunk Connections.

Referring to Figure 16: Functionality of Flexible Trunk to Trunk Connections without Trunk Barring configured on page 383, when Flexible Trunk to Trunk Connections Options is set to

Flexible Trunk to Trunk Connections Only (FTLY) and the Class of Service of Set B is set to Flexible Trunk to Trunk Connections Conditional (FTTC), the following is true:

- Telephone B cannot complete the Call Transfer from Trunk Route A to Trunk Route C.
- Telephone B can complete the Supervised Conference with Trunk Routes A and C, as long as no other limitations apply. If Set B drops out of this conference, Trunk Routes A and C are disconnected.

Table 121: CLS and FTOP Matrix for Flexible Trunk to Trunk Connections without Trunk Barring configured on page 384 is a matrix that summarizes the possible selections for Station Class of Service and Flexible Trunk to Trunk Connections Options available for Flexible Trunk to Trunk Connections without Trunk Barring configured.

Table 121: CLS and FTOP Matrix for Flexible Trunk to Trunk Connections without Trunk Barring configured

Station Classes of Service (LDs 10 and 11)	Customer Level Options (LD 15)	
	FTOP = FRES (Default)	FTOP = FTLY
CLS = FTTU (Default for existing sets)	No effect on Class of Service. Existing limitations still apply.	Allows trunk to trunk connections for both Transfer and Conference.
CLS = FTTR	No effect on Class of Service. Existing limitations still apply.	Blocks all trunk to trunk connections for Transfer and Conference.
CLS = FTTC (Default for new sets)	No effect on Class of Service. Existing limitations still apply.	Allows trunk to trunk connections for Supervised Conference only. Denies trunk to trunk connections for Transfer and unsupervised Conference.

Functionality of Flexible Trunk to Trunk Connections for Supervised Conference

For Supervised Conference, at least one internal set must be involved in the conference. With the Flexible Trunk to Trunk Connections feature configured, if the last set that drops out of the conference has Class of Service set to Flexible Trunk to Trunk Connections Restricted (FTTR) or Flexible Trunk to Trunk Connections Conditional (FTTC), the call is disconnected. If the last set that drops out of the conference has Class of Service (CLS) set to Flexible Trunk to Trunk Connections Unrestricted (FTTU), the call is not disconnected.

Figure 17: Flexible Trunk to Trunk Connections for Supervised Conference on page 385 illustrates the functionality of Flexible Trunk to Trunk Connections for Supervised Conference.

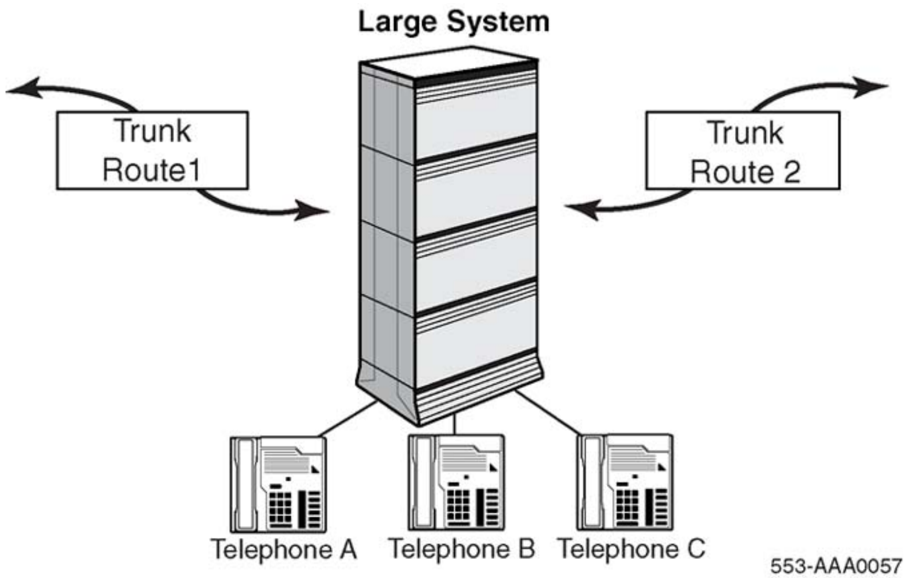


Figure 17: Flexible Trunk to Trunk Connections for Supervised Conference

Referring to Figure 17: Flexible Trunk to Trunk Connections for Supervised Conference on page 385, Telephones A, B, and C have Class of Service set to Flexible Trunk to Trunk Connections Unrestricted (FTTU), Flexible Trunk to Trunk Connections Restricted (FTTR), and Flexible Trunk to Trunk Connections Conditional (FTTC) respectively. Telephones A, B, and C and Trunk Routes 1 and 2 are involved in a conference.

- If A is the last internal telephone to drop out of the conference, the call is not disconnected by the Flexible Trunk to Trunk Connections feature, as Class of Service is set to Flexible Trunk to Trunk Connections Unrestricted (FTTU) for A. Other limitations, however, may cause the call to disconnect. This is an unsupervised conference. The present functionality is maintained.
- If B is the last internal telephone to drop out of the conference, the call is disconnected, as Class of Service is set to Flexible Trunk to Trunk Connections Restricted (FTTR) for B.
- If C is the last internal telephone to drop out of the conference, the call is disconnected, as Class of Service is set to Flexible Trunk to Trunk Connections Conditional (FTTC) for C.

Functionality of Flexible Trunk to Trunk Connections with Trunk Barring configured

Trunk Barring provides the option of denying a direct or modified connection between Customer defined routes. Trunk Barring works with Route Access Restriction Tables (ARTs), as defined in LD 56.

When the Flexible Trunk to Trunk Connections feature is used with Trunk Barring (TBAR) configured, additional flexibility in controlling the trunk to trunk connections for Transfer and Conference is provided.

If Flexible Trunk to Trunk Connections is implemented with the Trunk Barring feature, the following four options are available at a customer level:

- When FTOP = FRES, Flexible Trunk to Trunk Connections Restricted, the Flexible Trunk to Trunk Connections feature does not function. The customer can still configure the set's Class of Service; however, the Class of Service does not take effect. Flexible Trunk to Trunk Connections Restricted (FRES) is the default value.
- When FTOP = TBFT, Trunk Barring Flexible Trunk to Trunk Connections, additional limitations are applied, depending upon the set's Class of Service. Trunk to trunk connections barred by TBAR always remain restricted. Connections not barred by TBAR utilize the set's Class of Service.
- When FTOP = FTTB, Flexible Trunk to Trunk Connections Trunk Barring, Flexible Trunk to Trunk Connections lifts TBAR limitations for routes barred by TBAR, based upon the set's Class of Service. Flexible Trunk to Trunk Connections does not apply any new limitations for non-barred routes.
- When FTOP = FTLY, Flexible Trunk to Trunk Connections Only, trunk to trunk connections for Transfer or Conference that are on barred and non-barred routes are controlled exclusively by the Flexible Trunk to Trunk Connections feature.

The Flexible Trunk to Trunk Connections feature provides the same Class of Service options at a set level with or without Trunk Barring configured. (CLS = FTTU, FTTR, FTTC).

Flexible Trunk to Trunk Connections Options (FTOP) = Trunk Barring Flexible Trunk to Trunk Connections (TBFT)

Figure 18: Functionality of Flexible Trunk to Trunk Connections with Trunk Barring configured and FTOP = TBFT on page 387 illustrates Flexible Trunk to Trunk Connections functionality with Trunk Barring configured and FTOP set to TBFT in LD 15.

In Figure 18: Functionality of Flexible Trunk to Trunk Connections with Trunk Barring configured and FTOP = TBFT on page 387, B is established on a call with Trunk Route 1. Trunk Routes 1 and 2 are not barred by TBAR, but Trunk Routes 3 and 4 are barred connection to any other route. Trunk Routes 1, 2, 3, and 4 are both incoming and outgoing.

Referring to Figure Figure 18: Functionality of Flexible Trunk to Trunk Connections with Trunk Barring configured and FTOP = TBFT on page 387, when Flexible Trunk to Trunk Connections Options is set to Trunk Barring Flexible Trunk to Trunk Connections (TBFT) and the Class of

Service of B is set to Flexible Trunk to Trunk Connections Unrestricted (FTTU), the following is true:

- B can complete the Call Transfer between Trunk Routes 1 and 2.
- B can conference Trunk Routes 1 and 2 and then disconnect. In this case, Trunk Routes 1 and 2 remain connected, as TBAR does not bar the connection between the two trunks.
- B cannot complete Transfer or Conference from Trunk Routes 1 or 2 to Trunk Routes 3 or 4, as these trunk routes are barred by TBAR.

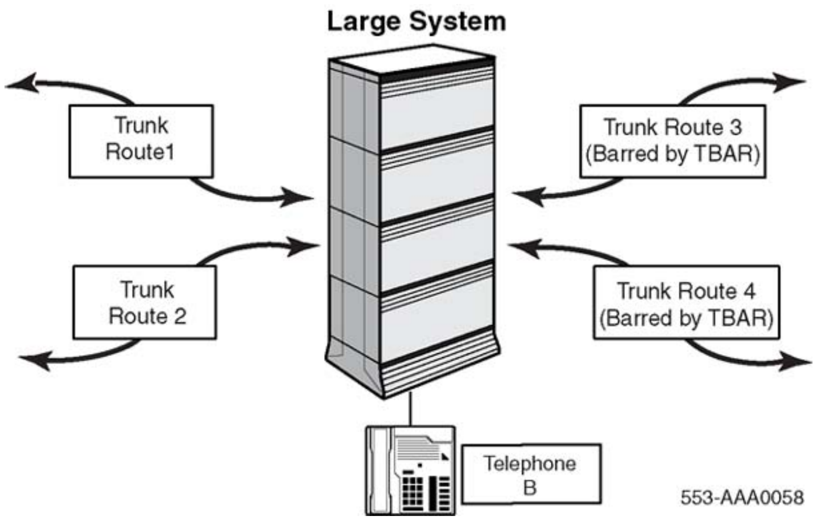


Figure 18: Functionality of Flexible Trunk to Trunk Connections with Trunk Barring configured and FTOP = TBFT

Referring to Figure 18: Functionality of Flexible Trunk to Trunk Connections with Trunk Barring configured and FTOP = TBFT on page 387, when Flexible Trunk to Trunk Connections Options is set to Trunk Barring Flexible Trunk to Trunk Connections (TBFT) and the Class of Service of B is set to Flexible Trunk to Trunk Connections Restricted (FTTR), the following is true:

- B cannot complete Transfer or Conference with Trunk Routes 1 and 2, even though the connectivity between the trunks is allowed by TBAR. This Class of Service functions as though the two trunks are blocked by the Trunk Barring feature.
- B cannot complete Transfer or Conference from Trunk Routes 1 and 2 to Trunk Routes 3 or 4, as these trunks are barred by TBAR.

Referring to Figure 18: Functionality of Flexible Trunk to Trunk Connections with Trunk Barring configured and FTOP = TBFT on page 387, when Flexible Trunk to Trunk Connections Options is set to Trunk Barring Flexible Trunk to Trunk Connections (TBFT) and the Class of Service of Set B is set to Flexible Trunk to Trunk Connections Conditional (FTTC), the following is true:

- B cannot complete the Call Transfer from Trunk Route 1 to Trunk Route 2
- B can complete the Supervised Conference with Trunk Routes 1 and 2. If Set B drops out of this conference, Trunk Routes 1 and 2 are disconnected.
- B cannot complete both Transfer and Conference from Trunk Routes 1 or 2 to Trunk Routes 3 or 4, as these trunks are barred by TBAR.

Flexible Trunk to Trunk Connections Options (FTOP) = Flexible Trunk to Trunk Connections Trunk Barring (FTTB)

Figure 19: Flexible Trunk to Trunk Connections with Trunk Barring configured and FTOP = FTTB on page 388 illustrates Flexible Trunk to Trunk Connections functionality with Trunk Barring configured and FTOP set to FTTB.

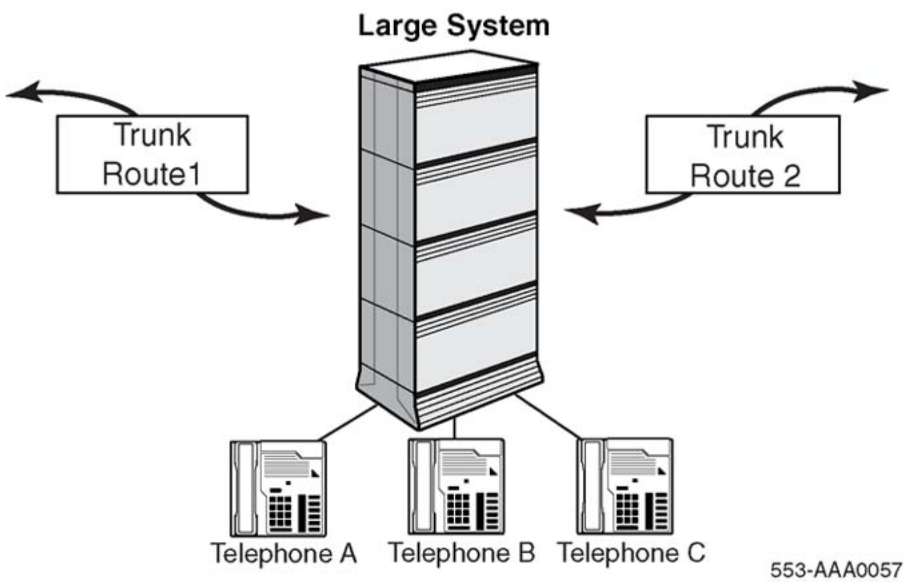


Figure 19: Flexible Trunk to Trunk Connections with Trunk Barring configured and FTOP = FTTB

In Figure 19: Flexible Trunk to Trunk Connections with Trunk Barring configured and FTOP = FTTB on page 388, Routes 1, 2, 3, and 4 are both incoming and outgoing. Access to different trunks is given as follows:

- From Trunk Route 1, connection is allowed to Trunk Routes 2, 3, and 4.
- From Trunk Route 2, connection is allowed to Trunk Routes 1, 3, and 4.
- From Trunk Route 3, connection is denied to Trunk Routes 1, 2, and 4.
- From Trunk Route 4, connection is denied to Trunk Routes 1, 2, and 3.

In short, any call from/to Trunk Route 1 or Trunk Route 2 is allowed. Any call from Trunk Route 3 and Trunk Route 4 is denied to all other trunk routes.

Referring to Figure 19: Flexible Trunk to Trunk Connections with Trunk Barring configured and FTOP = FTTB on page 388, when the Flexible Trunk to Trunk Connections Options is set to Flexible Trunk to Trunk Connections Trunk Barring (FTTB) and the Class of Service of B is set to Flexible Trunk to Trunk Connections Unrestricted (FTTU), all TBAR limitations for Transfer and Conference are lifted.

On a TBAR unrestricted trunk, B receives a call on incoming Trunk Route 1. The call is established. B initiates a call with any of the Trunk Routes 2, 3, or 4.

When TBAR does not restrict connection from Trunk Route 1 to any other trunk route:

- B can transfer the call on Trunk Route 1 to any of the Trunk Routes 2, 3, or 4.
- B can conference the call on Trunk Route 1 with any of the Trunk Routes 2, 3, or 4.

On a TBAR restricted trunk, B receives a call on incoming Trunk Route 3. The call is established. B initiates a call with any of the Trunk Routes 1, 2, or 4.

In this case, TBAR restricts connection from Trunk Route 3 to any other trunk route. However, as B has CLS set to FTTU, the TBAR restriction is lifted for B. Therefore:

- B can transfer the call on Trunk Route 3 to any of the Trunk Routes 1, 2, or 4
- B can conference the call on Trunk Route 3 with any of the Trunk Routes 1, 2, or 4.

Referring to Figure 19: Flexible Trunk to Trunk Connections with Trunk Barring configured and FTOP = FTTB on page 388, when the Flexible Trunk to Trunk Connections Options (FTOP) is set to Flexible Trunk to Trunk Connections Trunk Barring (FTTB) and the Class of Service of B is set to Flexible Trunk to Trunk Connections Restricted (FTTR), the existing TBAR functionality is retained.

On a TBAR unrestricted trunk, B receives a call on incoming Trunk Route 1, which is a TBAR unrestricted trunk. The call is established. B initiates a call with any of the Trunk Routes 2, 3, or 4.

When TBAR does not restrict connection from Trunk Route 1 to any other trunk route:

- B can transfer the call on Trunk Route 1 to any of the Trunk Routes 2, 3, or 4.
- B can conference the call on Trunk Route 1 with any of the Trunk Routes 2, 3, or 4.

On a TBAR restricted trunk, B receives a call on incoming Trunk Route 3. The call is established. B initiates a call with any of the Trunk Routes 1, 2, or 4.

In this case, TBAR restricts connection from Trunk Route 3 to any other trunk route. As B has Class of Service set to FTTR, the TBAR restriction is not lifted for this set.

- B cannot transfer the call on Trunk Route 3 to any of the Trunk Routes 1, 2, or 4.
- B cannot conference the call on Trunk Route 3 with any of the Trunk Routes 1, 2, or 4.

Referring to Figure 19: Flexible Trunk to Trunk Connections with Trunk Barring configured and FTOP = FTTB on page 388, when the Flexible Trunk to Trunk Connections Options is set to

Flexible Trunk to Trunk Connections Trunk Barring (FTTB) and the Class of Service of B is set to Flexible Trunk to Trunk Connections Conditional (FTTC), TBAR limitations for Supervised Conference are lifted. TBAR limitations for Transfer and unsupervised Conference are maintained.

On a TBAR unrestricted trunk, B receives a call on incoming Trunk Route 1. The call is established. B initiates a call with any of the Trunk Routes 2, 3, or 4.

When TBAR does not restrict connection from Trunk Route 1 to any other trunk route:

- B can transfer the call on Trunk Route 1 to any of the Trunk Routes 2, 3, or, 4.
- B can conference the call on Trunk Route 1 with any of the Trunk Routes 2, 3, or 4.

On a TBAR restricted trunk, B receives a call on incoming Trunk Route 3. The call is established. B initiates a call with any of the Trunk Routes 1, 2, or 4.

When TBAR restricts connection from Trunk Route 3 to any other trunk routes:

- B cannot transfer the call on Trunk Route 3 to any of the Trunk Routes 1, 2, or 4.

However, as B has Class of Service set to FTTC, the TBAR restriction is lifted for Supervised Conference. Therefore:

- B can conference the call on Trunk Route 3 with any of the Trunk Routes 1, 2, or 4.
- Once B drops out of the conference, the two remaining TBAR trunks are disconnected.

Since all telephones that are already present in the system default to a Class of Service of Flexible Trunk to Trunk Connections Unrestricted (FTTU), when the Customer Option is changed to Flexible Trunk to Trunk Connections Trunk Barring (FTTB), TBAR limitations for all telephones are lifted for Conference and Transfer. Therefore, the Class of Service must be changed to Flexible Trunk to Trunk Connections Restricted (FTTR), in order to maintain the existing TBAR functionality. The telephone sets that are new to the system default to a Class of Service of FTTC.

Flexible Trunk to Trunk Connections Options (FTOP) = Flexible Trunk to Trunk Connections Only (FTLY)

When Flexible Trunk to Trunk Connections Options is set to Flexible Trunk to Trunk Connections Only (FTLY) and the Class of Service is set to Flexible Trunk to Trunk Connections Unrestricted (FTTU), trunk to trunk connections are allowed for both Conference and Transfer, irrespective of whether or not TBAR is activated.

When Flexible Trunk to Trunk Connections Options is set to Flexible Trunk to Trunk Connections Only (FTLY) and the Class of Service is set to Flexible Trunk to Trunk Connections Restricted (FTTR), trunk to trunk connections are denied for both Conference and Transfer, irrespective of whether or not TBAR is activated.

When Flexible Trunk to Trunk Connections Options is set to Flexible Trunk to Trunk Connections Only (FTLY) and the Class of Service is set to Flexible Trunk to Trunk Connections Conditional (FTTC), trunk to trunk connections are allowed for Supervised Conference only.

Trunk to trunk connections for Transfer and unsupervised Conference are denied, irrespective of whether or not TBAR is activated.

Table 122: CLS and FTOP Matrix for Flexible Trunk to Trunk Connections with TBAR configured on page 391 is a matrix that summarizes the possible selections for Station Class of Service and Flexible Trunk to Trunk Connections Options available for Flexible Trunk to Trunk Connections when Trunk Barring is configured.

Table 122: CLS and FTOP Matrix for Flexible Trunk to Trunk Connections with TBAR configured

Station Classes of Service (LDs 10 and 11)	**Customer Level Options (LD 15)**			
	FTOP = FRES (Default)	FTOP = TBFT	FTOP = FTTB	FTOP = FTLY
CLS = FTTU (Default for existing sets)	Existing TBAR functionality.	Existing TBAR functionality.	Lifts all TBAR limitations for Transfer and Conference.	Allows Transfer and Conference, irrespective of whether or not TBAR is activated, unless other limitations exist.
CLS = FTTR	Existing TBAR functionality.	Blocks all trunk to trunk connections for both Transfer and Conference.	Existing TBAR functionality.	Blocks all trunk to trunk connections for Transfer and Conference.
CLS = FTTC (Default for new sets)	Existing TBAR functionality.	Allows trunk to trunk connections for Supervised Conference on non-TBAR routes. Denies Transfer and unsupervised Conference on all trunk to trunk connections not blocked by TBAR.	Lifts TBAR limitations for Supervised Conference only. Maintains TBAR limitations for unsupervised Conference and Transfer.	Blocks trunk to trunk connections for Transfer and unsupervised Conference. Allows trunk to trunk connections for Supervised Conference.

Operating parameters

Flexible Trunk to Trunk Connections is configured at a set level, by defining the Class of Service (CLS) prompt in LDs 10 or 11.

All existing telephone sets default to a Class of Service of Flexible Trunk to Trunk Connections Unrestricted (FTTU) upon initial software conversion. When new telephone sets are added and configured, they default to a Class of Service of Flexible Trunk to Trunk Connections Conditional (FTTC).

In the Customer Data Block, Flexible Trunk to Trunk Connections Options can be set to Trunk Barring Flexible Trunk to Trunk Connections (TBFT) and Flexible Trunk to Trunk Connections Trunk Barring (FTTB) only when the Trunk Barring is configured. Flexible Trunk to Trunk Connections Options (FTOP) is set to the default, Flexible Trunk to Trunk Connections Restricted (FRES), to maintain the existing functionality.

A telephone with a Class of Service of Flexible Trunk to Trunk Connections Restricted (FTTR) cannot initiate a Conference call to an outgoing trunk, although it can be included in a conference. If this type of telephone is the last set to disconnect from the conference, the call is ended. The established trunks are released.

If a conference is on hold and an additional telephone attempts to join the conference over a barred trunk route and through a telephone that has Class of Service set to Flexible Trunk to Trunk Connections Restricted (FTTR), then Flexible Trunk to Trunk Connections does not permit a consultation connection. This is as per the existing operation.

If more than two trunks are involved in a call and all internal calls drop from the conference, Flexible Trunk to Trunk Connections does not affect the Conference disconnection.

Multiple Appearance, Single Call Arrangement DNs allow a single call to be active on the DN, regardless of its number of appearances. If the Single Call Ringing DN is established in a call, another appearance of the DN can enter into the call, if the Privacy feature is not in effect, by going off hook or by pressing the Multiple Appearance Single Call DN key. Flexible Trunk to Trunk Connections limitations for Conference are applicable in such a case.

As per the existing operation, answer and disconnect supervision is a requirement for Transfer and Conference.

Flexible Trunk to Trunk Connections does not support Basic Rate Interface (BRI) telephone sets or attendant console operations.

Flexible Trunk to Trunk Connections supports Analog and ISDN trunks. R2MFC and AC15 signaling is also supported. Flexible Trunk to Trunk Connections does not support Service trunks, such as Recorded Announcement (RAN), Paging (PAG), Dictation (DIC), Music (MUS), and Automatic Wake Up Recorded Announcement (AWR).

Call Redirection features are not supported with Flexible Trunk to Trunk Connections.

With Flexible Trunk to Trunk Connections, unless the Trunk to Trunk Connection feature is configured, two outgoing trunk connections are blocked for Transfer and unsupervised Conference.

Customer Controlled Routing (CCR), Meridian Link, and Application Module Link (AML) applications are not affected by the Flexible Trunk to Trunk Connections feature.

When adding a new telephone, the default Class of Service is Flexible Trunk to Trunk Connections Conditional (FTTC). This could impact an application's ability to conference or transfer a call to an outgoing trunk. In the case where this functionality is required, the Class of Service must be changed on the set to Flexible Trunk to Trunk Connections Unrestricted (FTTU).

Flexible Trunk to Trunk Connections blocks the initiation of Conference. Applications, such as Break In, Barge In, Bridging, and Overriding, are not supported.

Feature interactions

Access Limitations

Access Limitations limits terminal access to the exchange network, private network, and certain features and services. During the call origination process, access checks are made by the system on the following:

- Class of Service of an individual terminal
- Trunk Group Access Limitations (TGAR) code of a terminal, if a direct trunk access code is dialed or as an optional feature when a Basic Alternate Route Selection (BARS) or Network Alternate Route Selection (NARS) access code is dialed
- area code and exchange code, if dialed by terminals with toll denied or conditional toll denied Class of Service, using direct trunk access codes and Code Restriction Tables
- Network Class of Service (NCOS) of a terminal, if Basic Alternate Route Selection (BARS)/Network Alternate Route Selection (NARS) or Coordinated Dialing Plan (CDP) access codes are dialed, or if direct trunk access codes are dialed and New Flexible Code Restriction (NCFR) tables are programmed

Previously restricted connections by any feature other than Trunk Barring cannot be lifted or avoided by the Flexible Trunk to Trunk Connections feature. Basically, all existing limitations apply with the exception of Trunk Barring limitations.

Call Transfer

If Flexible Trunk to Trunk Connections allows a telephone to transfer to an outgoing trunk, Access Limitations can still block the transfer. If a telephone is denied transfer by the Flexible Trunk to Trunk Connections feature, then the transfer is blocked regardless of Access Limitations.

For a transfer to be completed, both Access Limitations and Flexible Trunk to Trunk Connections must allow the transfer.

Conference

If the Flexible Trunk to Trunk Connections feature allows a telephone to conference to an outgoing trunk, then Conference is allowed unless it is blocked by other existing limitations. If a telephone disconnects from a conference, Flexible Trunk to Trunk Connections limitations verify whether the telephone is allowed to transfer the call between the two trunks. If allowed, this unsupervised conference is completed, unless and until barred by another feature.

Attendant Console Operations

Flexible Trunk to Trunk Connections does not support attendant console operations. If an attendant attempts to extend an originating trunk connection on a route barred by the Trunk Barring feature, overflow tone is provided. The Flexible Trunk to Trunk Connections feature does not lift this restriction.

Although attendant consoles have a Conference key, Flexible Trunk to Trunk Connections does not apply any limitations.

Basic Alternate Route Selection Network Alternate Route Selection Coordinated Dialing Plan Flexible Numbering Plan

Regardless of the method of dialing used to originate the call with the outgoing trunk, Flexible Trunk to Trunk Connections limitations apply for Transfer and Conference.

Call Redirection

Call Forward features

When a telephone performs Call Forward to an external trunk and receives an incoming trunk call, it may result in a trunk to trunk connection. The Flexible Trunk to Trunk Connections Station Class of Service is not applied when forwarding incoming trunk calls to a barred route.

Call Pickup

The new Station's Classes of Service, introduced by the Flexible Trunk to Trunk Connections feature, do not impose any limitations on Call Pickup.

Meridian Mail Trunk Access Restriction

Flexible Trunk to Trunk Connections limitations do not apply to Meridian Mail Trunk Access Restriction (MTAR). Irrespective of the Station's Class of Service, external calls are prevented from being transferred/conferenced to Meridian Mail.

Multi-Party Operations - Call Join

The functionality of Flexible Trunk to Trunk Connections applies to conferences made by the Call Join operation.

No Hold Conference

When a Meridian 1 proprietary telephone is established with a trunk call and a No Hold Conference is initiated, Trunk Barring limitations do not apply, and the conference is completed. However, if the last internal telephone involved in the No Hold Conference has a Class of Service of Flexible Trunk to Trunk Connections Conditional (FTTC) or Flexible Trunk to Trunk Connections Restricted (FTTR), then the call is disconnected if that telephone drops out of the conference.

Scheduled Access Limitations

With the Flexible Trunk to Trunk Connections feature configured, existing limitations are not avoided. Additional limitations imposed by Flexible Trunk to Trunk Connections Classes of Service are introduced when Scheduled Access Limitations is configured.

Toll Operator Break In

The Flexible Trunk to Trunk Connections Classes of Service have no impact on Toll Operator Break In.

Trunk Access From Any Station

There is no limitation with the new Flexible Trunk to Trunk Connections Station Classes of Service that can restrict the station from picking up the call by Trunk Access From Any Station (TAFAS).

Trunk to Trunk Connection

Flexible Trunk to Trunk Connections takes precedence over the Trunk to Trunk Connection feature.

Virtual Network Services

Flexible Trunk to Trunk Connections does not apply any limitations to existing Virtual Network Services (VNS) functionality.

Feature packaging

This feature is included in base system software.

Feature implementation

Task summary list

The following is a summary of the tasks in this section:

1. Table 123: LD 15 - Configure Flexible Trunk to Trunk Connections options. on page 397

   Configure Flexible Trunk to Trunk Connections options.

2. Table 124: LD 10 - Configure Flexible Trunk to Trunk Connections for analog (500 and 2500 type) sets. on page 398

   Configure Flexible Trunk to Trunk Connections for analog (500/2500 type) sets.

3. Table 125: LD 11 - Configure Flexible Trunk to Trunk Connections for Meridian 1 proprietary sets. on page 398

   Configure Flexible Trunk to Trunk Connections for Meridian 1 proprietary sets.

Table 123: LD 15 - Configure Flexible Trunk to Trunk Connections options.

Prompt	Response	Description
REQ:	CHG	Change existing data.
TYPE:	NET	Trunk and network options.
CUST		Customer number
	0-99	Range for Large System and Avaya Communication Server 1000E (Avaya CS 1000E) system.
...		
FTOP		Flexible Trunk to Trunk Connections Options.
	(FRES) TBFT	FTT feature is inactive. FTT adds new limitations on connections not barred by TBAR.
	FTTB	FTT lifts TBAR limitations for routes barred by TBAR. FTT cannot add any new limitations for non-barred routes.
	FTLY	All set based trunk to trunk connections for Transfer and Conference are controlled by FTT only.
...		

Table 124: LD 10 - Configure Flexible Trunk to Trunk Connections for analog (500 and 2500 type) sets.

Prompt	Response	Description
REQ:	CHG	Change existing data.
TYPE:	500	500/2500 type telephone data block.
TN		Terminal number
	l s c u	Format for Large System, Avaya MG 1000B, and CS 1000E system, where l = loop, s = shelf, c = card, u = unit.
...		
CLS	(FTTC) FTTU FTTR	Flexible Trunk to Trunk Connections Conditional (default for new sets). Flexible Trunk to Trunk Connections Unrestricted (default). Flexible Trunk to Trunk Connections Restricted.
...		

Table 125: LD 11 - Configure Flexible Trunk to Trunk Connections for Meridian 1 proprietary sets.

Prompt	Response	Description
REQ:	CHG	Change existing data.
TYPE:	a...a	Telephone type. Type **?** for a list of possible responses.
TN		Terminal number
	l s c u	Format for Large System, MG 1000B, and CS 1000E system, where l = loop, s = shelf, c = card, u = unit.
...		
CLS	(FTTC)	Flexible Trunk to Trunk Connections Conditional (default for new sets).
	FTTU	Flexible Trunk to Trunk Connections Unrestricted (default for existing sets).
	FTTR	Flexible Trunk to Trunk Connections Restricted.
...		

Feature operation

No specific operating procedures are required to use this feature.

Chapter 59: Flexible Voice and Data Terminal Number

Contents

This section contains information on the following topics:

Feature description

The Flexible Voice/Data Terminal Number feature allows both bearer (B) channels on the M2000 series Meridian 1 proprietary sets to be available for either voice or data calls on a dynamic (per-call) or static basis. This feature has been developed exclusively for VISIT equipment functionality.

With the dynamic capabilities of this feature, a user has access to two simultaneous voice or two simultaneous data connections on the Time Compression Multiplexing (TCM) loop. This capability is practical for users with various desktop multimedia applications, such as VISIT video, that require various combinations of voice and data connections on a per-call basis.

Dynamic voice/data Terminal Numbers (TNs) have two Directory Numbers (DNs) to place and receive calls. The primary voice DN is assigned to key 00 on the telephone. Another key is assigned to the data DN. This key is designated as the data mode key. All data calls are placed and received using this key. Any other secondary DN keys assigned to a dynamic voice/data TN can place and receive voice calls only.

With the static capabilities of this feature, each B-channel on the set is configured as either voice or data. This provides the opportunity for two voice or two data B-channels on the same

TCM loop. This configuration doubles the density of the digital line card (XDLC). Since the TN has either a voice or data Class of Service, calls placed from any DN key on the set are either voice or data.

Operating parameters

There are no limitations against placing data calls on hold.

When a Terminal Number (TN) is in the voice mode, the short hunting feature is terminated when a Data Mode (DTM) key is encountered.

Data calls to a dynamic voice/data TN are not redirected. All TN redirection features such as Call Forward All Calls and Hunting are applicable to voice calls only. If a data call is not presented to the DTM key the call is given reorder tone.

A Data Mode (DTM) key can be assigned to M2000 series proprietary sets with the exception of the M2006 set.

Data Directory Numbers (DNs) for dynamic voice/data TNs cannot have Multiple Appearance DNs (MADNs).

A dynamic voice/data TN can only have one data DN.

No audible progress tones, such as dial tone or ringback, are provided for data calls to or from dynamic TNs. Only Time Compression Multiplexing (TCM) progress messages are sent for data calls. Audible progress tones are provided for voice calls.

If set relocation takes place, upper and lower TNs of a Time Compression Multiplexing (TCM) loop are relocated together. This occurs even if upper and lower TNs were assigned as dynamic or static. A relocated lower TN (0-15) must be in voice mode. Following relocation, both TNs maintain their prior voice or data settings.

When a service change is performed on a dynamic TN in data mode, it is automatically changed to voice mode.

To prevent improper setup, the new Class of Service Flexible Terminal Number Allowed (FLXA) must be specified to assign Data Class of Service to a lower TN or Voice Class of Service to an upper TN.

Feature interactions

Call Forward All Calls Call Forward, Internal Calls

Voice calls directed to a dynamic voice/data Terminal Number are forwarded, if either of these features are enabled. Data calls, to a dynamic voice/data TN, are not forwarded.

Call Redirection

If a call is not presented to the Data Mode (DTMK) key, the call is given reorder tone.

Call Waiting Camp-On

These features are not supported on data calls to a dynamic voice/ data TN.

Call Waiting and Camp On are supported for voice calls to dynamic voice/ data TN. However, no tone is inserted during a Camp On attempt if the Terminal Number is in a busy data mode.

Message Waiting Forward Busy Call Forward Busy

Voice calls directed to a call processing busy dynamic voice/data TN are redirected using Message Waiting Forward Busy or Call Forward Busy provided these features are configured for the TN. Data calls to dynamic voice/data TNs are not redirected.

Voice Call

If a dynamic TN has a single appearance DN key that terminates on a Voice Call (VCC) key, the called party hears a single beep if occupied on another DN. However, if the called party is a dynamic TN in data mode, the DN key lamp flashes. A beep is not provided.

Feature packaging

The Flexible Voice/Data Terminal Number feature is contained in M2000 Digital Sets (DSET) package 88.

Feature implementation

Task summary list

The following is a summary of the tasks in this section:

1. Table 126: LD 11 - Assign the Static Voice Terminal Number. on page 404

   Assign the Static Voice Terminal Number.
2. Table 127: LD 11 - Assign the Static Data Terminal Number. on page 405

   Assign the Static Data Terminal Number
3. Table 128: LD 11 - Assign the Dynamic Terminal Number. on page 405

   Assign the Dynamic Terminal Number.

Table 126: LD 11 - Assign the Static Voice Terminal Number.

Prompt	Response	Description
REQ:	NEW CHG	New, or Change.
TYPE:	a...a	Telephone type. Type **?** for a list of possible responses.
TN		Terminal number
	l s c u	Format for Large System, Avaya CS 1000 Media Gateway 1000B (Avaya MG 1000B), and Avaya Communication Server 1000E (Avaya CS 1000E) system, where l = loop, s = shelf, c = card, u = unit.
CLS	FLXA VCE	Flexible voice/data allowed.FLXA is only required if voice TN unit is less than 15. (FLXD) = Flexible voice/ data denied (default). This Class of Service can only be assigned to 2006, 2008, 2016, 2216 or 2616 sets. When configured to CLS = FLXA, Voice Class of

Prompt	Response	Description
		Service (VCE) can be assigned to the upper TN unit (16 - 31).

Table 127: LD 11 - Assign the Static Data Terminal Number.

Prompt	Response	Description
REQ:	NEW CHG	New, or Change.
TYPE:	a...a	Telephone type. Type **?** for a list of possible responses.
TN		Terminal number
	l s c u	Format for Large System, Avaya MG 1000B, and CS 1000E system, where l = loop, s = shelf, c = card, u = unit.
CLS	FLXA DTA	Flexible voice/data allowed. FLXA is only required if data TN unit is greater than 15. (FLXD) = Flexible voice/data denied (default). This Class of Service can only be assigned to 2006, 2008, 2016, 2216 or 2616 sets. When configured to CLS = FLXA, Data Class of Service (DTA) can be assigned to the lower TN unit (0 -15).

Table 128: LD 11 - Assign the Dynamic Terminal Number.

Prompt	Response	Description
REQ:	NEW CHG	New, or Change.
TYPE:	a...a	Telephone type. Type **?** for a list of possible responses.
TN		Terminal number
	l s c u	Format for Large System, MG 1000B, and CS 1000E system, where l = loop, s = shelf, c = card, u = unit.
CLS	FLXA VCE	Flexible voice/data allowed. (FLXD) = Flexible voice/ data denied. This Class of Service can only be assigned to 2006, 2008, 2016, 2216 or 2616 sets. When configured to CLS = FLXA Voice Class of Service (VCE) can be assigned to the upper TN unit (16 - 31) and Data Class of Service (DTA) can be assigned to the lower TN unit (0 -15). A Single Call Ringing (SCR) key can be designated a Data Mode (DTMK) key.
DTMK	xx	Key assignment for Data Mode Key. This key must be a single appearance SCR/SCN key and cannot be assigned key 00.

Prompt	Response	Description
- KEY	00 aaa xxxx	Prime Directory Number Key, where aaa = SCR, SCN, MCR or MCN and xxxx = Voice Directory number
- KEY	xx SCR yyyy xx SCN yyyy	Single Call Ringing Single Call Non Ringing Data Mode Key, where xx = key number and yyyy = Data Directory Number.

When call processing switches between voice and data mode on the dynamic Terminal Number, some Class of Service option data is automatically modified. In data mode, the dynamic TN has options Warning Tone Denied (WTD) and Maintenance Telephone Denied (MTD). When switched back to voice mode, the original settings for these options is automatically restored, and the Class of Service is not printed.

Feature operation

No specific operating procedures are required to use this feature.

Chapter 60: Forced Camp-On and Priority Override

Contents

This section contains information on the following topics:

Feature description

Forced Camp-On is similar to the regular station-to-station Camp-On, except that it can be done without an internal or external call on hold. When used with Priority Override, the capability is called Enhanced Override.

Priority Override allows an established call to be broken into and another call presented to the desired party. Before Break-In occurs, a warning tone is given to all parties involved in the established call. The set performing the override must have a priority level equal to or higher than the set being overridden.

Operating parameters

Priority Override and Forced Camp-On can operate independently of each other.

All stations involved in an established call being broken into must have Warning Tone Allowed (WTA) Class of Service.

Priority Override and Forced Camp-On cannot be applied to telephones involved in any of the following:

- non-established call
- conference call
- attendant call
- Release Link attendant call
- attendant call through Centralized Attendant Service or a Primary Rate Access/Integrated Services Digital Network trunk
- Automatic Call Distribution (ACD) call
- data call
- parked call
- call waiting call
- held call
- operator Call Back or toll operator Break-In call
- Make Set Busy active, or
- Do Not Disturb active.

External trunks cannot perform priority override. They can be overridden only if they are the undesired party of an established call being broken into.

Feature interactions

Multi-party Operations

With Multi-Party Operations (MPO), when a consultation call is made on a set equipped with Priority Override, a control digit has to be dialed from the set to perform a recall and return the call on hold.

Override

When Priority Override is activated, it replaces normal override. Once Priority Override has been performed on a set, its Digit Display shows the DN of the overriding set.

Feature packaging

Priority Override/Forced Camp-On (POVR) is packaged under package 186.

Dependencies:

- Flexible Feature Codes (FFC) package 139
- Multi-party Operations (MPO) package 141

Feature implementation

Task summary list

The following is a summary of the tasks in this section:

1. Table 129: LD 10, LD 11 - Respond to CLS prompt on page 410

   Respond to CLS prompt with CPFA to allow Camp-On to another set, or CPFD to deny such Camp-On. Respond to PLEV prompt with a value between 1 and 7, to set the priority level for this set.
2. Table 130: LD 11 - Respond to the KEY prompt with a key number on page 410

   Respond to the KEY prompt with a key number, followed by EOVR, to define an Enhanced Override key for each Meridian 1 proprietary telephone.
3. Table 131: LD 14 - Trunks to be involved on page 410

   Trunks to be involved in such Camp-On override calls must have Warning Tone Allowed (WTA) Class of Service allowed
4. Table 132: LD 15 - AFCO prompt on page 410

   To select either automatic or manual forced Camp-On for a customer, respond to the AFCO prompt with either YES (automatic) or NO (manual).
5. Table 133: LD 16 - At the PLEV prompt, specify priority levels for trunk routes. on page 410

   At the PLEV prompt, specify priority levels for trunk routes.
6. Table 134: LD 57 - The Enhanced Override flexible feature code on page 411

The Enhanced Override flexible feature code must be defined by responding to the EOVR prompt with an appropriate FFC digit sequence to be assigned that function.

The following are additional to the definitions required for telephone configuration without this feature.

Table 129: LD 10, LD 11 - Respond to CLS prompt

Prompt	Response	Description
...		
CLS	(CPFA) CPTD	Forced Camp-On from another set (allowed) denied.
PLEV	0-(2)-7	Priority Level, prompted with Priority Override/Forced Camp-On (POVR) package 186. 2 = set can override sets of level 1 and 2, and can be overridden by sets of level 2-7.

Table 130: LD 11 - Respond to the KEY prompt with a key number

Prompt	Response	Description
...		
KEY	xx EVOR	Key number; Enhanced Override.

Table 131: LD 14 - Trunks to be involved

Prompt	Response	Description
...		
CLS	(WTA) WTD	Warning Tone (allowed) denied.

Table 132: LD 15 - AFCO prompt

Prompt	Response	Description
...		
AFCO	(NO) YES	(Manual) Automatic Forced Camp-On, prompted with Priority Override/Forced Camp-On (POVR) package 186.

Table 133: LD 16 - At the PLEV prompt, specify priority levels for trunk routes.

Prompt	Response	Description
...		
PLEV	0-(2)-7	Priority Level, prompted with Priority Override/Forced Camp-On (POVR) package 186. 2 = set can override sets of level 1 and 2, and can be overridden by sets of level 2-7

Table 134: LD 57 - The Enhanced Override flexible feature code

Prompt	Response	Description
...		
- EOVR	xxxx	Enhanced Override (manual Forced Camp-On followed by Priority Override).

Feature operation

Forced Camp-On is activated automatically (if Automatic Forced Camp-On is defined), or it can be activated manually using the Enhanced Override (EOVR) key on Meridian 1 proprietary telephones or the Enhanced Override Flexible Feature Code on analog (500/2500 type) telephones. If the EOVR key is pressed again or the Enhanced Override Flexible Feature Code dialed again, Priority Override is activated.

If Forced Camp-On is not equipped, the first depression of the EOVR key, or the first dialing of the Enhanced Override Flexible Feature Code activates Priority Override.

To activate Priority Override, the user of an analog (500/2500 type) telephone dials the Override Flexible Feature Code, while the user of a Meridian 1 proprietary telephone presses the Override key (OVR). Priority Override can also be activated using the Enhanced Override Flexible Feature Code or the Enhanced Override key (EOVR), as described previously.

Chapter 61: Forward No Answer Call Waiting Direct Inward Dialing

Contents

This section contains information on the following topics:

Feature description

The Forward No Answer Call Waiting Direct Inward Dialing (FCWD) feature allows a Direct Inward Dialing (DID) call that encounters a busy set with Call Waiting Allowed to be routed to an attendant (or recalled to the night DN during Night Service), if it is not answered within a customer-defined period (between 2-126 seconds). If Return to Same Attendant is equipped, the call is routed to the first available attendant.

Operating parameters

There are no operating parameters associated with this feature.

Feature interactions

Call Waiting Redirection

With the Call Waiting Redirection feature also enabled, the Call Waiting Redirection feature takes precedence over the FCWD feature. The existing CFNA also takes precedence over the existing Attendant Recall of Call Waiting calls. Since the Call Waiting Redirection feature applies CFNA treatment to a Call Waiting call while the FCWD feature applies an attendant recall timer, the Call Waiting Redirection feature also has precedence over the FCWD timer.

Feature packaging

This feature is packaged under French Type Approval (FRTA), package 197.

Feature implementation

Table 135: LD 15 - Respond to FCWD prompt with an even-numbered value between 0 and 126 seconds.

Prompt	Response	Description
REQ:	NEW CHG	Add, or change.
TYPE:	RDR	Call Redirection.
...		
- FCWD	(0)-126	Number of seconds a DID call should wait on a set before being forwarded to the attendant, prompted with French Type Approval (FRTA) package 197. If (0) is chosen, the call is not forwarded to an attendant. Valid entries are even numbers between 1 and 126; odd numbers are rounded down.

Feature operation

No specific operating procedures are required to use this feature.

Chapter 62: Generic XFCOT Software Support

Contents

This section contains information on the following topics:

Feature description

The Generic XFCOT is a circuit card developed to meet the North American Transmission Plan, with the following functionalities:

- tone supervision
- battery supervision
- Periodic Pulse Metering (PPM)
- loopstart signaling

This feature provides the choice of Dynamic Pad Switching in the North American Environment for Central Office trunks (LD 97), enhances the trunk-to-trunk connection, and improves the use of disconnect supervision in features like ACD, Meridian Mail, DISA, Call Park, and Camp-On.

With this feature, a disconnect-supervised loopstart Central Office trunk follows normal XFCOT rules for trunk-to-trunk connection and disconnection.

Functionality is provided on the following IPE circuit cards:

- NTCK16AD for PPM/BAT/BTS
- NTCK16BD for BAT/BTS

Operating parameters

There are no operating parameters associated with this feature.

Feature interactions

Automatic Call Distribution

This feature is used when a large number of incoming calls are to be answered by a designated group of telephone sets. Calls that cannot be answered immediately are put in an Automatic Call Distribution (ACD) queue.

ACD is allowed on disconnect supervised or unsupervised loopstart trunks. If a caller on an unsupervised loopstart trunk disconnects while the call is in an ACD queue, it is detected when the call is answered by an ACD agent.

With this development, caller disconnection is detected by disconnect-supervised loopstart trunk on an XFCOT card and disconnected callers are then dropped from the ACD queue.

Other ACD operations that require a disconnect-supervised COT such as INTERFLOW, NCFW and NITE RAN are now allowed on a disconnect-supervised loopstart trunk on an XFCOT card.

Call Park

Call Park feature allows an attendant or telephone user to place a call in parked state (connected to a parked DN) where it can be retrieved by any attendant console or station set. If the call is not retrieved after a customer-defined time, the call is recalled to the telephone user who parked it.

Call Park is allowed on disconnect-supervised or unsupervised IPE loopstart Central Office trunks. If a caller on an unsupervised loopstart trunk disconnects while the call is in parked state is detected when the parked call is recalled or answered.

Caller disconnection during park state is detected by a disconnect supervised loopstart trunk on an XFCOT card. The disconnected caller is then dropped from the parked DN.

Camp-On

The Camp-On feature allows an attendant to route one additional call to a busy DN so it can be rung when it becomes free. If the busy DN is not free after a customer-defined time, the call is recalled to the attendant.

A call from a loopstart disconnect supervised or unsupervised loopstart trunk can be camped on. If a caller on an unsupervised loopstart trunk disconnects while the call is camped on, it is detected when the call is recalled or answered.

Caller disconnection during Camp-On operation is detected by a disconnect-supervised loopstart trunk on an XFCOT card and the camped on call is dropped.

Digital Trunk Interface (DTI) Pad Switching

The DTI pad process intervenes when a DTI port is involved in a connection. It is independent from the normal pad and it handles the DTI port side and the other port side.

This process is changed to handle XFCOT units when the North American Transmission Plan is selected as XUT units.

Direct Inward System Access

This feature allows selected external users to access the system switch by dialing a special directory number, and to use some features of the system as an internal station.

A Direct Inward System Access (DISA) call is allowed on a disconnect supervised or unsupervised loopstart trunk. If a caller on an unsupervised loopstart trunk disconnects during a DISA operation, it is detected by a dial time out or when the call is answered.

Caller disconnection during a DISA operation is detected by a disconnect-supervised loopstart trunk on an XFCOT card and the operation can then be ended.

European XFCOT Software Support

This feature supports international IPE trunks with new functionalities such as supervision on loopstart trunk, PPM, and static pad switching.

The Generic XFCOT Software Support is a product improvement of this feature regarding the pad switching, the trunk-to-trunk disconnection, and the use of disconnect supervision for loopstart trunk in some features.

Meridian Mail

The Meridian Mail feature allows a caller to leave a voice mail message for a person unable to be reached. Once the caller is connected to the voice mail- there is a maximum duration allowed for the message after which the call is disconnected.

Meridian Mail is allowed on disconnect supervised or unsupervised loopstart trunks. If a caller on an unsupervised loopstart trunk disconnects while accessing Meridian Mail, the call is disconnected when the connection-time to the mail box exceeds the maximum duration.

Caller disconnection is detected by the disconnect-supervised loopstart trunk on an XFCOT card and the caller is then dropped from the queue for messaging service or from the mail box.

Periodic Clearing

Periodic Clearing is the sending of a periodic signal from the system to a Central Office when an incoming call has been answered but is not in an established state (for instance, ringing, held, parked). The connection is disconnected if the originator goes on-hook.

The Periodic Clearing condition is timed by the disconnect timer (DCTI) to prevent this situation from lasting for an extended time. When the DCTI timer expires the trunk is disconnected.

The Disconnect Timer can be used without having the feature Periodic Clearing configured particularly when the Central Office trunk has no disconnect supervision. It can be disabled by setting the DCTI to 0 in LD 16.

A loopstart trunk can be marked as disconnect supervised. When it has a class of service providing disconnect supervision, in Periodic Clearing condition the trunk is disconnected when the calling station releases the call.

Feature packaging

This feature is packaged under the following packages:

- Intelligent Peripheral Equipment (XPE) package 203
- International Supplementary Features (SUPP) package 131

- Meridian 1 Enhanced Conference, TDS and MFS (XCT0) package 204
- Meridian 1 Superloop Administration (XCT1) package 205 (unrestricted when the XPE package is equipped)

Feature implementation

Task summary list

The following is a summary of the tasks in this section:

1. Table 136: LD 97 - Choose the North American Transmission Plan by answering YES to the NATP prompt. on page 421

   Choose the North American Transmission Plan by answering YES to the NATP prompt.
2. Table 137: LD 14 - Define the Periodic Pulse Metering parameters on a per country basis. This prompt is used to define a disconnect supervised loopstart trunk on an XFCOT. on page 421

   Define the Periodic Pulse Metering parameters on a per country basis. This prompt is used to define a disconnect supervised loopstart trunk on an XFCOT.

Table 136: LD 97 - Choose the North American Transmission Plan by answering YES to the NATP prompt.

Prompt	Response	Description
REQ	aaa	Request (CHG, END, PRT)
TYPE	LOSP	TYPE = LOSP (Loss Plan Tables)
NATP	YES	North American Transmission Plan for Generic XFCOT

Table 137: LD 14 - Define the Periodic Pulse Metering parameters on a per country basis. This prompt is used to define a disconnect supervised loopstart trunk on an XFCOT.

Prompt	Response	Description
REQ	NEW	New.
TYPE	COT	Central Office Trunk.
TN		Terminal number
	l s c u	Format for Large System, Avaya CS 1000 Media Gateway 1000B (Avaya MG 1000B), and Avaya Communication

Prompt	Response	Description
		Server 1000E (Avaya CS 1000E) system, where l = loop, s = shelf, c = card, u = unit.
XTRK	XCOT	Type is IPE COT.
CDEN	(8D)	Card density is 8D.
SIGL	LOP	Loop start signaling.
PPID	xx	Where xx is one of the following: 0 – United Kingdom (50 Hz) 1 – France (12 Khz) 2 – France (50 Hz) 3 – Germany, Egypt, Turkey, Venezuela, Indonesia, Finland (16 Khz) 4 – Switzerland, Ireland, Portugal, Italy, Spain, Lebanon, Turkey (12 Khz) 5 – Denmark (12 Khz) 6 – Norway, Belgium (16 Khz) 7 – Holland (50 Hz) 8 – Australia (two different packs) (12Khz/50 Hz) 9-15 – Reserved for future use.
BTID	xx	Enter the country busy tone ID as follows: 0-2 – Reserved for future use 3 – Germany, Ireland 4 – Switzerland 5 – Denmark 6 – Norway, Kuwait, Chile, Venezuela, Indonesia, Thailand, Korea 7 – Holland 8 – Australia, Mexico 9 – Ireland 10 – Taiwan, Brazil, Tortola, Mexico 11 – Singapore 12 – Argentina, Italy 13 – Lebanon, Italy 14 – Turkey 15 – Reserved for future use.
SUPN	YES (NO)	Trunk Supervision required (not required)
STYP	BTS BAT	Busy tone supervision enabled Loop break supervision enabled
CLS	(LOL) SHL (DIP) DTN (P10) P20 P12	Attenuation pads in (out). Digitone signaling (digipulse). Make-break ration for pulse dialing speed.

Feature operation

No specific operating procedures are required to use this feature.

Chapter 63: Group Call

Contents

This section contains information on the following topics:

Feature description

Group Call allows a user of a Meridian 1 proprietary telephone to place a call to up to ten Directory Numbers (DNs) simultaneously by activating a Group Call key. The called DNs must have been previously defined as members of a group.

Each customer within the system can have up to 64 groups assigned. Each group has up to 20 group members. Any DN in the system can be assigned as a member of a group, and a DN can be a member of more than one group.

Each Multiple Appearance, MCR/MCN DN reduces the number of telephone sets that can be added to a Group Call. For example, if two telephones have the same MCR appearance of a DN, the number of telephones in the Group Call becomes 19. That is, each appearance of a DN counts as one member, up to a maximum of 20, of the Group Call.

Multiple Appearance, SCR/SCN DNs count as one member of a Group Call, irrespective of its number of DN appearances.

Groups are defined through Service Change in LD 18. When a group is defined, each member of the group is assigned a member number. If network or conference blocking is encountered, members are assigned priorities for connection to the Group Call in the order of their group member numbers (member 0 has the highest priority). It is recommended that group members be assigned from different network loops to minimize the possibility of network blocking.

The Group Call key is used to originate a Group Call to all members of the group to which the Group Call key is assigned. The Group Call key for a given group can appear on more than one telephone. More than one Group Call key can be assigned to a group, but only one Group Call key can be active for a given group at any time. A telephone with a Group Call key need not be equipped with a Directory Number (DN) that is defined as a group member.

Activation of a Group Call key originates a call to all assigned members of the group. When the first member of the group answers, ringback tone is removed and a speech path is set up between the member and the originator of the call. As subsequent members answer, they are added to the call. The lamp associated with the Group Call key at the originator's telephone flashes until all members of the group have answered the call.

If a Directory Number (DN) is actively engaged in a call and a Group Call is originated for that DN, either the Group Call is camped on or Call Waiting is activated for the DN and a special warning tone is provided. The special warning tone consists of three rapid bursts of tone followed by 10 seconds of silence, then an additional three rapid bursts of tone.

An active Group Call is under complete control of the originator of the call. If the originator goes on hook, the call is completely broken down. Members who are taking part in a Group Call can disconnect from the call at any time, but once disconnected, they cannot be reconnected.

Operating parameters

A Group Call can be originated only from a Meridian 1 proprietary telephone with a Group Call key.

Group Call does not support data calls.

With the Extended Conference TDS (XCT) card (NT8D17), audio interference can occur if many of the call participants are on "older-style" analog (500/2500-type) sets. This problem does not occur when participants are on Meridian Digital Telephones or "newer-style" electronic analog (500/2500-type) sets. Analog (500/2500-type) sets produce more audio noise and degrade the sound quality of the conference as the number of participants exceeds 12-15 members. As the number of participants drops below this threshold, the sound quality of the Group Call returns to normal.

The maximum number of members per group is 20.

Note:

For CS 1000E with MGC media gateways, the maximum members per group is 20.

The maximum number of groups per customer is 64.

Each group member DN must have a Warning Tone Allowed Class of Service.

Off-premise Extension (OPX) lines cannot be members of a group.

Calls to a DN that is active in a Conference call, or Group Call, are blocked.

Feature interactions

Automatic Line Selection

This feature is not selected for automatic Outgoing Line Selection or Non-Ringing Line Selection. It is selected for Incoming Ringing Line Selection.

Call Forward All Calls

A Group Call to a telephone with Call Forward active is forwarded one step only. The Call Forward number must be a valid DN.

Call Forward and Hunt Override Via Flexible Feature Code

It is not possible to use Call Forward/Hunt Override FFC as a Group Call DN.

Call Pickup

This feature can be used to answer a Group Call if it is activated by a valid telephone in the same Call Pickup group, or by using Directory Number (DN) Pickup or Group Pickup.

Call Pickup Network Wide

The Group Call feature does not allow a remote party in a Group Call list. Therefore, a Group Call cannot be picked up by a remote station. If during the network scanning a Group Call is found, it will be ignored and the network scanning will continue.

Call Transfer Conference

Neither Call Transfer nor Conference can be initiated during a Group Call. If an analog (500/2500 type) telephone user flashes the switchhook during an established Group Call, the user is dropped from the call.

Directory Number Delayed Ringing

When a group call is made to an SCN/MCN key with Directory Number Delayed Ringing (DNDR) defined, audible notification will be given after the DNDR delay has expired.

Display of Calling Party Denied

The calling party's display shows the DN of the last set to connect into the Group Call regardless of the Class of Service. The called set displays the Group Number only.

Hold

Only the originator of a Group Call can put the Group Call on hold.

Hot Line

Hot Lines can be members of a Group Call. They cannot, however, have a Group Call key.

ISDN QSIG and EuroISDN Call Completion

Call Completion cannot be applied to a Group Call.

Make Set Busy Individual Do Not Disturb

A Group Call to a telephone in Make Set Busy or Individual Do Not Disturb mode cannot be completed. The telephone will not be rung and is not counted as part of the Group Call (for instance, if all other members in the group have answered, the lamp next to the Group Call key on the originator's telephone lights steadily).

Network Intercom

When Directory Number Delayed Ringing (DNDR) is defined and an incoming call to set configured with Hot Type I or D Key and DNDR occurs, the set winks until the DNDR timer expires. After this timer expires, the set rings as normal.

Short Buzz for Digital Telephones

The special three-second buzz for Group Call is not affected by this feature.

Telephone features

The following features cannot be applied on a Group Call:

- Call Forward No Answer
- Call Forward Busy
- Call Forward/Hunt Override Via Flexible Feature Code
- Call Join
- Call Park
- Call Transfer
- Conference
- Hunting
- Privacy Release
- Ring Again

Feature packaging

This feature is packaged under Group Call (GRP), package 48 and has no feature package dependencies.

Feature implementation

Task summary list

The following is a summary of the tasks in this section:

1. Table 138: LD 18 - Add or change a Group Call list. on page 428

   Add or change a Group Call list.
2. Table 139: LD 11 - Add or change Group Call for Meridian 1 proprietary telephones. on page 428

   Add or change Group Call for Meridian 1 proprietary telephones.
3. Table 140: LD 20 - Print Group Call data. on page 429

   Print Group Call data.

Table 138: LD 18 - Add or change a Group Call list.

Prompt	Response	Description
REQ	CHG	Change.
TYPE	GRP	Group Call data block.
CUST	xx	Customer number, as defined in LD 15
GRNO	0-63	Number of the Group Call list.
STOR	xx yyy...y <CR>	Group member number (xx) and associated DN (yyy...y). End input of stored Group Call entries

Table 139: LD 11 - Add or change Group Call for Meridian 1 proprietary telephones.

Prompt	Response	Description
REQ:	CHG	Change.
TYPE:	a...a	Telephone type. Type ? for a list of possible responses.
TN		Terminal number
	l s c u	Format for Large System, Media Gateway 1000B , and CS 1000E system, where l = loop, s = shelf, c = card, u = unit.
KEY	xx GRC yy	Add a Group Call key, where:

Prompt	Response	Description
		xx = key number, and yy = Group Call list number (0-63).

Table 140: LD 20 - Print Group Call data.

Prompt	Response	Description
REQ	PRT	Print.
TYPE	GRP	Group Call data.
CUST	xx	Customer number, as defined in LD 15
GRNO	0-63 <CR>	Number of the Group Call group. Print data for all Group Call groups.

Feature operation

To make a Group Call:

- Press Group Call. All group members are automatically called. The LCD indicator beside the Group Call key flashes until all members have answered. Then it lights steadily.

To make a Group Call using a Flexible Feature Code, see the feature module Dial Access to Group Calls on page 77 of this document.

Chapter 64: Group Hunt

Contents

This section contains information on the following topics:

Feature description

Group Hunting is similar to the Hunting feature. If a call encounters a busy DN and a Group Hunting Pilot DN is specified, the call is routed to the next idle DN in a prearranged group. Unlike the existing Hunting feature, Group Hunting allows a customer to:

- Configure all members of a hunt group in one block instead of many different station data blocks.
- Prevent Group Hunt termination on any idle member through a Group Hunt Deactivate Flexible Feature Code (FFC) or through a GHD (Group Hunt Deactivate) key.
- Limit the hunting steps to the total number of DNs in the list.
- Initiate hunting by dialing or accessing a Group Hunt Pilot DN directly.
- Configure a DN to be a member of more than one hunt group.

Pilot DN

Pilot DNs are defined as PLDN Flexible Feature Codes (FFC) in LD 57. Pilot DNs are used in two ways:

1. If the USE prompt is set to GPHT, then the Pilot DN is defined to activate Group Hunting.
2. If the USE prompt is set to Speed Call List Controller (SCLC) or Speed Call List User (SCLU), then the Pilot DN is defined to access the Speed Call or System Speed Call lists that are associated with the Pilot DN.

Termination conditions

When a Group Hunt Pilot DN is dialed, Group Hunting searches the list associated with the Pilot DN, according to the hunt type specified, until one of the following conditions is met:

1. idle DN is encountered
2. Automatic Call Distribution (ACD) DN, Integrated Voice Messaging Service (VMS) DN, Message Center (MC) DN, Listed Directory Number (LDN), or attendant DN is encountered
3. route access code is encountered
4. ESN access code is encountered
5. Group Hunt Pilot DN is encountered
6. all DNs in the group are hunted to, or the maximum number of hunting groups is reached

If condition 1 or 2 is met, then incoming calls are completed to that DN. All DNs listed in condition 2 are associated with a queue. Remember the following when configuring these Group Hunt lists:

- These DNs always appear idle to a hunt cycle, regardless of their actual status. The hunt always redirects to the indicated destination, and never comes back into the Group Hunt list, therefore these calls are never queued against the Pilot DN.
- It is recommended that if these DNs must be used in a Group Hunt list, only one such DN be used. This DN must always be the last entry in the list.
- Also, linear hunting must be used. In this configuration, any redirected call is subject to the call processing treatment of the destination.
- Listed DNs can be configured as a last entry in a hunt group list, if linear hunting is used. The redirected call is presented to the associated LDN Incoming Call Indicator (ICI) key on the attendant console. The call can be transferred back to the Hunt Group Pilot DN; once transferred, it cannot be recalled to the attendant.
- Attendant DNs can be configured as a last entry in a hunt group list, if linear hunting is used. The call can be transferred back to the Hunt Group Pilot DN; once transferred, it cannot be recalled to the attendant.
- Automatic Call Distribution (ACD) DNs can be configured as a last entry in a hunt group list, if linear hunting is used. The call can be transferred back to the Hunt Group Pilot DN.

If the ACD queue has the Hunt Group Pilot DN defined as the night DN, the call is transferred back into the hunt group list.

If termination condition 3 or 4 is met, that call termination depends on either the access code or the number that followed. Therefore, remember the following when configuring Group Hunt lists:

- Use only one access code for each Group Hunt list. The access code must always be the last entry in the list.
- Use linear hunting. In this configuration, any redirected call is subject to the call processing treatment of the destination.
- If an access code is used as a Group Hunt member, it must be entered as "access code and complete destination number" to ensure proper routing to the destination, not just the access code alone.
- Trunk optimization does not apply.

If termination condition 5 is met, the search ends for the current list and begins for the list associated with the new Pilot DN. A Pilot DN cannot be a member of its own Hunting Group.

If termination condition 6 is met, then incoming calls are placed in a queue in the order of arrival. They are then presented to the next DNs in the group as the members become available.

Direct Inward Dialing (DID) calls are placed in a Group Hunting queue only if the group is still in service. If the group is not in service (if all of its members have deactivated Group Hunting), DID calls are routed directly to the attendant.

Calls are removed from a Group Hunting queue when they are abandoned, when they are presented to an available member, or when they are attendant-extended calls and the slow answer recall timer has expired.

Ringback tone is heard by callers who wait in Group Hunting queues for service.

If the attempted DN for termination by Group Hunting is not a valid member or number, an error message (ERR 8985) prints, hunting terminates, and calls route to overflow tone as specified by the intercept treatment.

Hunt types

Two types of Group Hunt are provided: linear and round robin. Only one hunt type is allowed per Group Hunt List.

- Linear: Hunting starts at the first DN in the list and ends when one of the conditions mentioned in Termination conditions on page 432 is met.
- Round Robin: Hunting starts at the DN next in the list to the last DN that was hunted to. Hunting ends when one of the conditions mentioned in Termination conditions on page 432 is met.

Group Hunt Lists

Group Hunt lists are defined and modified in LD 18. The Pilot DN entered for each list must have been previously defined as a Group Hunt FFC in LD 57. When a Group Hunt list is defined, the members are assigned a member number as in configuring a Speed Call List. The maximum DN size of each member is 31 digits. The list members can have one of the following DN types:

1. Single or Multiple Appearance DN
2. Listed Directory Number
3. Attendant DN
4. Automatic Call Distribution (ACD-DN), VMS-DN, MC-DN
5. Route access code (route access code + number)
6. Electronic Switch Network (ESN) number (for example, access code + number)
7. Group Hunt Pilot DN
8. Radio Paging access code followed by a complete DN

A Group Hunt list can also be modified through a Speed Call or System Speed Call Controller key, through an analog (500/2500 type) telephone feature Speed Call Controller, or through Group Hunting Speed Call or System Speed Call Controller Flexible Feature Codes (FFC).

Composition of Group Hunt lists

Authorized Group Hunt list members belong to one of the following categories:

- Set-associated DNs (DN type 1) — These DNs are associated with sets and/or keys in a stand-alone system. They are any of the following:
    - Single-appearance DNs
    - Multiple-appearance, single-call arrangement DNs
    - Multiple-appearance, multiple-call arrangement DNs
    - If MADNs with multiple-call arrangement are to be used in Group Hunt lists, they must have only one Prime DN appearance.
    - MADNs on analog (500/2500-type) sets with the MCRA class of service (Multiple-call arrangement) are not supported in Group Hunt lists. If one appearance of this MADN is busy, all other appearances are also considered busy by the Group Hunt Cycle.

- A set-associated DN can only be defined 96 times as a Group Hunt list member in the system.

- System-associated DNs (DN types 2, 3, 4, 7, 8) — These DNs are associated with other destinations than extensions in a stand-alone system.
- Routing-associated DNs (DN types 5, 6) — These DNs are associated with destinations outside the stand-alone system.

System- or routing-associated DNs always appear available during a Group Hunt cycle. Set-associated DNs do not appear available. Therefore, the Group Hunt cycle always redirects to system-associated and routing-associated DNs when they are met in the list, and the call is never queued against the Pilot DN.

If Linear Hunt is configured for a Group Hunt list that has system- or routing-associated DNs, make these DNs the last entries in the list. List members entered with a higher index are never reached because the system- or routing- associated DNs always route the call.

There are four supported Group Hunt lists:

- Type I Group Hunt lists set-associated DNs — These use Linear Hunt type. When the user dials the PLDN, the Group Hunt Cycle offers the call to the first list member. If this member is not eligible, the next member is tried. This process continues until one is found, or until the complete list is searched. Figure 20: Group Hunt cycle for Type 1 Group Hunt lists on page 436 describes the Group Hunt cycle for Type I Group Hunt lists.

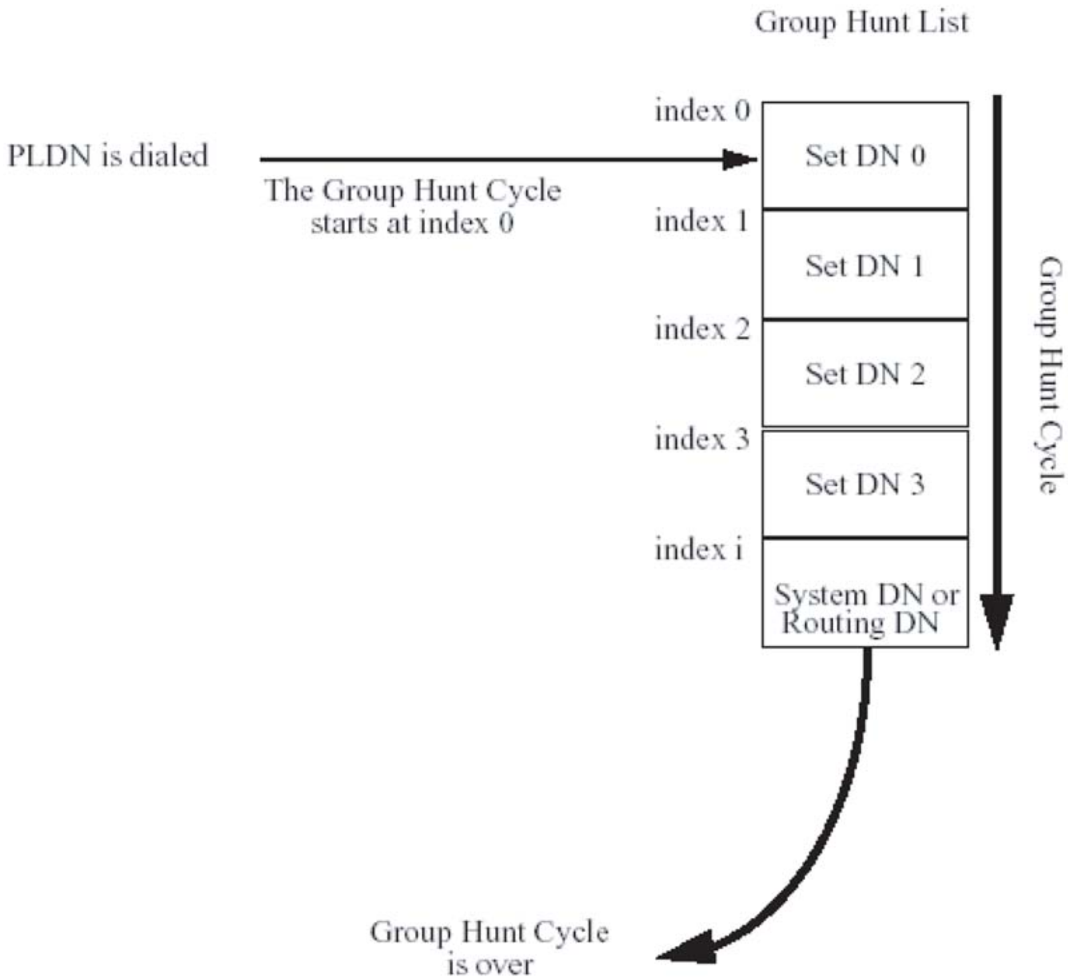


Figure 20: Group Hunt cycle for Type 1 Group Hunt lists

- Type II Group Hunt lists set-associated DNs — These use Round Robin Hunt type. When the user dials the PLDN, the Group Hunt Cycle offers the call to the index next to the last index that was hunted to in the previous cycle. If this member is not eligible, the next DN member is tried. This process continues until one is found, or until the whole list is searched. Figure 21: Group Hunt cycle for Type II Group Hunt lists on page 437 describes the Group Hunt cycle for Type II Group Hunt lists.

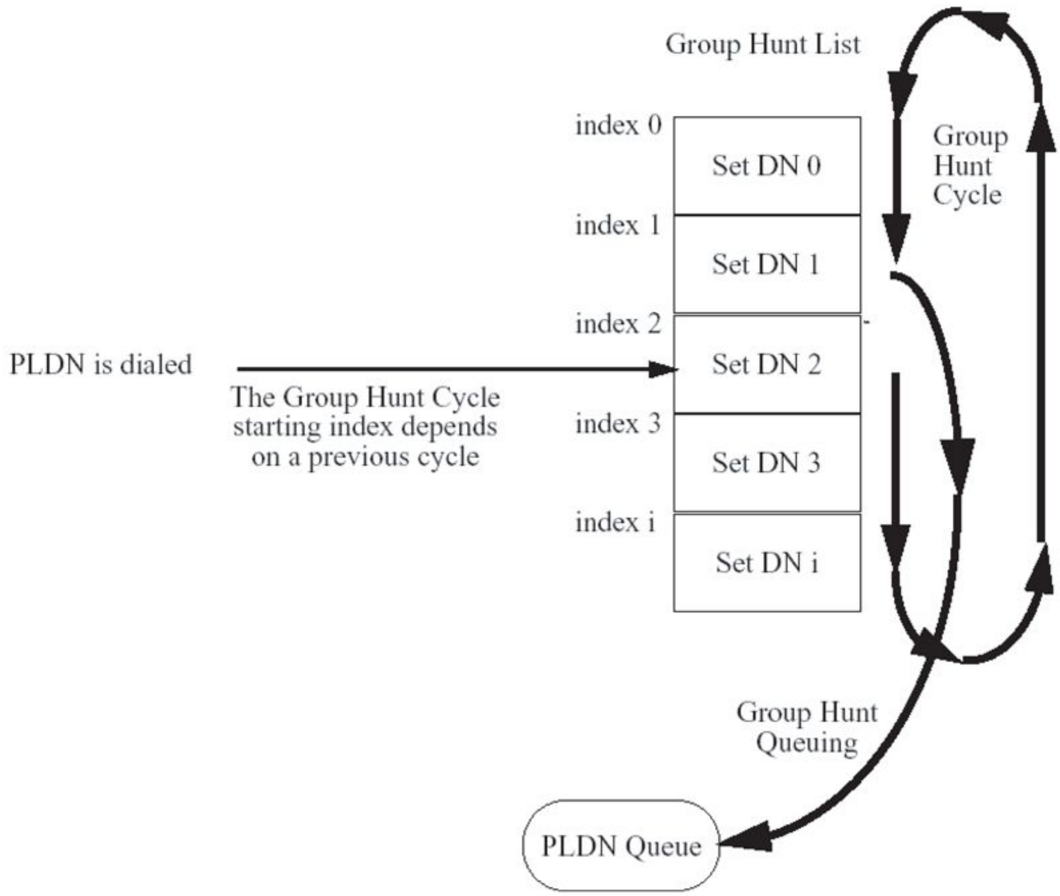


Figure 21: Group Hunt cycle for Type II Group Hunt lists

- Type III Group Hunt lists set-associated DNs except for the highest index position, which is filled with a system- or routing-associated DN — These use Linear hunt type. When the user dials the PLDN, the Group Hunt Cycle offers the call to the first list member. If this member is not eligible, the next member is tried. This process continues until one is found. If no member is found, the call is offered to the highest index position (System- or Routing-associated DN). There is no queuing. Figure 22: Group Hunt cycle for Type III Group Hunt lists on page 438 describes the Group Hunt cycle for Type III Group Hunt lists.

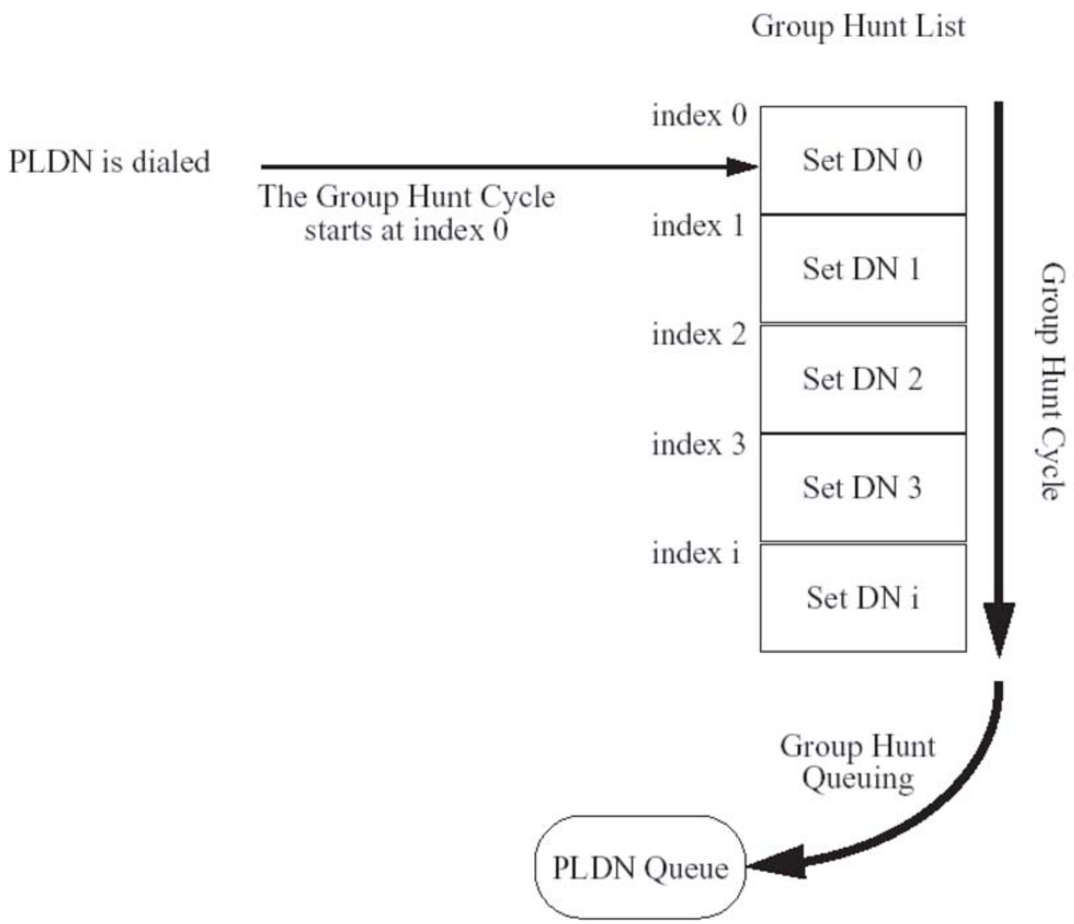


Figure 22: Group Hunt cycle for Type III Group Hunt lists

- Type IV Group Hunt lists system- or routing-associated DNs of the same DN type — These use Round Robin hunt type. Once the PLDN has been dialed, the Group Hunt Cycle offers the call to the index next to the last index that was hunted to during the previous cycle. There is no further search in the list nor is there any queuing. Figure 23: Group Hunt cycle for Type IV Group Hunt lists on page 439 describes the Group Hunt cycle for Type IV Group Hunt lists.

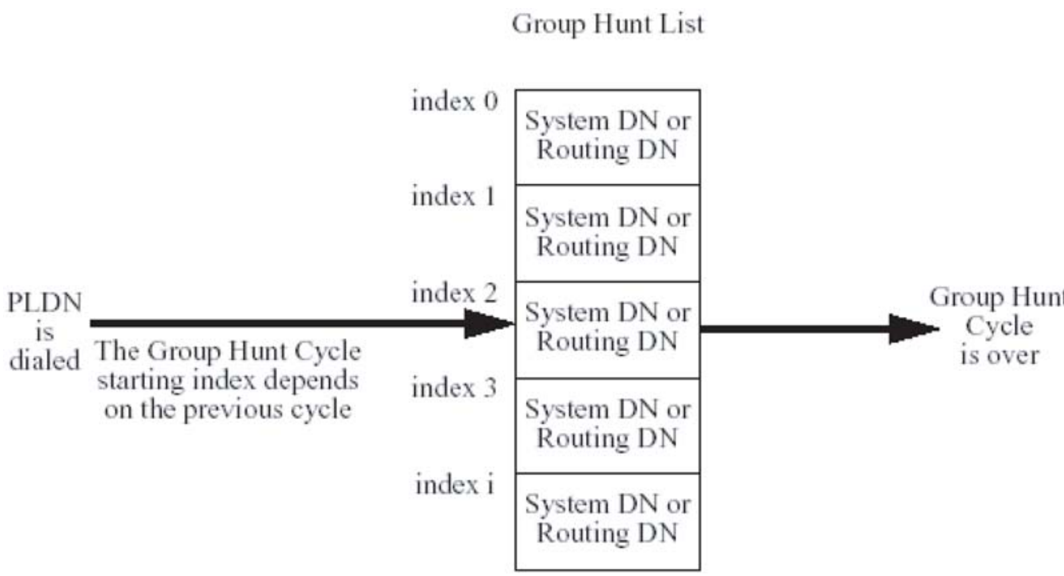


Figure 23: Group Hunt cycle for Type IV Group Hunt lists

Important:
The user must define the Group Hunt list to match one of the four list types.

Queuing

If all members of a Group Hunt list are busy, calls are queued against the Pilot DN of that Group Hunt list. Ring-back tone is provided. There are several options available to control the number of calls allowed to be queued against any given Pilot DN. These options are:

- Group Hunt Queuing Limitation allows the system administrator to select, through service change, the number of calls allowed to queue against the Pilot DN. The selection is made by responding to the Maximum Queue (MQUE) prompt in LD 57. The valid responses to this prompt are:
    - 0: No calls allowed to queue.
    - 1: One call allowed to queue.
    - ALL: No limit to the number of calls allowed to queue.
    - ACTM: The number of calls allowed to queue must be less than or equal to the number of active Group Hunt list members.

Group Hunt Deactivate and Activate

Group Hunt Deactivate allows an idle Set-associated DN to be made non-eligible to Group Hunt calls. A Group Hunt Deactivate FFC code (GHTD) is available to both analog (500/2500-

type) and digital sets. The Group Hunt Deactivate key (GHD) can be configured on digital sets.

1. Digital sets
    a. If the station user activates the DN key, dials the GHTD code and then dials PLDN, the DN is deactivated from the Group Hunt list associated with PLDN. An overflow tone is given if the operation is not successful.
    b. If the station user activates the DN key and dials the GHTD code, the DN is deactivated from all Group Hunt lists to which this DN belongs. An overflow tone is given if the operation is not successful.
    c. If the station user activates the GHD key (key lamp is dark), the Prime DN (defined on key 0) is deactivated from all Group Hunt lists to which this Prime DN belongs. The GHD key lamp is lit if the operation is successful. The GHTD code must be used to deactivate non-Prime DNs.
2. Analog (500/2500-type) sets
    a. If the station user goes offhook, dials the GHTD code, and then dials PLDN, the DN is deactivated from the Group Hunt list associated with PLDN. An overflow tone is given if the operation is not successful.
    b. If the station user goes offhook and dials the GHTD code, the DN is deactivated from all Group Hunt lists to which this DN belongs. An overflow tone is given if the operation is not successful.

If the DN is a Multiple-Appearance DN, all appearances of the DN are deactivated when a DN member executes one of the deactivation processes.

Group Hunt Activate allows an idle Set-associated DN to return to the active state from the Group Hunt Deactivate state. A Group Hunt Activate FFC code (GHTA) is available to both analog (500/2500-type) and digital sets. A Group Hunt Deactivate key (GHD) can be configured on digital sets.

1. Digital sets
    a. If the station user activates the DN key, dials the GHTA code and then dials PLDN, the DN is activated again in the Group Hunt list associated with PLDN. Overflow tone is given if the operation is not successful.
    b. If the station user activates the DN key and dials the GHTA code, the DN is activated again for all Group Hunt lists to which this DN belongs. Overflow tone is given if the operation is not successful.
    c. If the station user activates the GHD key (key lamp is lit), the Prime DN (defined on key 0) is activated again for all Group Hunt lists to which this Prime DN belongs. The GHD key lamp is darkened if the operation is successful.
2. Analog (500/2500-type) sets

a. If the user goes offhook and dials the GHTA code, and then dials PLDN, the DN is activated again for the Group Hunt list associated with PLDN. An overflow tone is given if the operation is not successful.

b. If the station user goes offhook and dials the GHTA code, the DN is activated again for all Group Hunt lists to which this DN belongs. An overflow tone is given if the operation is not successful. The GHTA code must be used to activate non-Prime DNs.

If the DN is a Multiple-Appearance DN, all appearances of the DN are activated again when a DN member executes an activation processes.

Access to Group Hunt lists

A Group Hunt list can be accessed by dialing the associated Pilot DN, through:

- manual dialing
- automatic dialing (such as Autodial, Hotline, Speed Call)
- redirection (such as Call Transfer, Call Forward, Hunt)
- ACD Night Service
- ACD interflow/overflow
- trunk access

A Pilot DN can be accessed like any other DN in the network. Any network user can access all Group Hunt lists defined for a network from anywhere in the network. This allows a centralized Group Hunt list to be set up for all network users.

However, Group Hunting is not possible across the network because calls encountering access code entries are always directed to the destination and never return to the hunt queue.

Operating parameters

The Group Hunting feature does not support data calls.

Hunting is limited to the following:

- the total number of DNs in the group
- a maximum of 30 hunting groups for each hunting sequence (for multi-group systems)
- a maximum of 18 hunting groups for each hunting sequence (for all other systems)

Hunting can be limited to the total number of DNs in the group, to 30 hunting groups per hunting sequence for multi-group systems, or to 18 hunting groups per hunting sequence for all other systems.

A maximum of 31 digits can be entered in each list entry.

A maximum of 96 entries can be placed in each list.

A specific station can be defined within a group, among different groups, or a combination thereof a maximum of 96 times.

A maximum of 8000 Group Hunt lists can be defined on a system (programmable through the existing MSCL prompt in LD 17 and reduced by the number of defined Speed Call and System Speed Call lists.)

For larger applications, the ACD package must be equipped to optimize call control and call distribution.

It is recommended that the Group Hunt feature be primarily used with set-associated DNs.

A Group Hunt pilot DN cannot be a member of its own list.

Round Robin hunting should only be used if all entries in the Group Hunt list are the same type (for example, all set-associated DNs or system-associated DNs).

A Pilot DN can be accessed from a network TIE trunk. Also, members of the Group Hunt list can be located at remote nodes.

Feature interactions

Attendant Timed Recalls

Attendant-extended calls to a PLDN will recall to the attendant when the recall timer expires.

- If the call extends to an eligible member in the Group Hunt list, the Slow Answer Recall Timer of the customer applies.
- If the attendant-extended call is queued to the PLDN, the Call Waiting Timer of the customer applies.
- If the attendant-extended call is queued to the PLDN, the Call Waiting Recall Timer is started as explained above; when a list member becomes eligible and is offered the call, the timer re-starts with its Slow Answer Recall value.

If the call is extended to an eligible member that has Call Forward No Answer configured, the Flexible Call Forward No Answer Timer applies instead of the Slow Answer Recall Timer. Ringing in the Group Hunt list continues as long as allowed by CFNA.

Access Limitations

If a routing-associated DN is programmed in a Group Hunt list, access limitations apply, based on the Class of Service of the calling station or route, the TGAR of the calling station or route, or both.

Attendant Alternative Answering

A Pilot DN can be defined as an alternative DN. Calls forwarded to a Pilot DN as an alternative DN are directed to the next DN in the group.

Attendant Blocking of Directory Number

It is not possible to activate the Attendant Blocking of DN feature for a Pilot DN. If an attempt is made to block a PLDN, the attempt will be canceled and overflow tone will be returned. If a DN that is a member in a Group Hunt (or Hunt) list is blocked by the Attendant Blocking of DN feature, the DN is considered to be busy.

Attendant Break-in and Toll Operator Break-in

Attendant Break-in and Toll Operator Break-in will not be supported when dialing a Pilot DN directly.

Attendant Busy Verify

An attendant is not allowed to busy-verify when dialing a Pilot DN directly.

Attendant Overflow Position

A PLDN cannot be configured as an Attendant Overflow DN (AODN).

Call Forward All Calls

When Group Hunt attempts to terminate on a DN, which has Call Forward All Calls active, it will continue with the next DN in the group if the attempted DN is busy, or if the DN is idle and the response to the Call Forward Ignore (CFWI) prompt in LD 57 is NO. If the attempted DN is idle and the response to the CFWI prompt in LD 57 is YES, then the system terminates Group Hunt and ring stations associated with the DN.

Call Forward Busy

Group Hunting has priority over the Call Forward Busy feature.

If the DN to be terminate has FBA (Forward Busy Allowed) Class of Service is busy, then Group Hunting continues with the next DN in the group.

Call Forward by Call Type

A Pilot DN can be configured as the redirection DN (HUNT, FDN, EHT, EFD) for the CFCT feature. The interaction is the same as for the Call Forward No Answer feature.

Call Forward External Deny

A Pilot DN cannot be configured as the Call Forward All Calls redirection DN if the set has the CFXD capability allowed.

Call Forward and Hunt Override through Flexible Feature Code

Primary Line Directory Numbers (PLDNs) are not overridden by the Call Forward/Hunt Override Via FFC feature. Any attempt will be ignored and access denied treatment will result.

Call Forward No Answer

Call Forward No Answer (CFNA) can optionally be configured to use a Pilot DN. This option is available when the HUNT DN or the FDN is defined as a Pilot DN.

If an idle station attempted for termination has CFNA defined, then the station will be rung. If the station does not answer within the customer specified number of ring cycles, then Group Hunting will continue with the next DN in the group. The calling party will continue to hear ringback tone until one of the conditions mentioned in Termination conditions on page 432 (the last condition is not applicable in this case) is met, or until they release the call.

French Type Approval (FRTA) package 197 restricted:

If a user, for example X (internal or external) dials the PLDN, the telephone sets rings as described in the following sections.

Consider Set A as the first valid member of FNA class of service found by the Group Hunt Cycle and it rings CFNx times. CFNx is the number of ring cycles before the Call Forward Number Answer redirection takes place. It is configured in Overlay 15.

Consider Set B as another valid member searched in the Group Hunt list. If set B has FND class of service, it rings until X disconnects. If set B has FNA class of service, it rings CFNx times; another valid member for example, set C is searched in the list, and the process repeats until one of the following conditions are met:

- set-associated DN with FND class of service
- system-associated DN
- routing-associated DN

After set A rung CFNx times, if the next eligible member in the list is set A, it rings again.

French Type Approval (FRTA) package 197 equipped:

For internal calls, the telephone ringing works as mentioned in the previous section. For external calls (CO, DID), the calls are routed to the attendant after set A rings CFNx times.

Call Forward No Answer by Call Type

CFNA by Call Type can be configured to use a Pilot DN. This option is available when the EFD or EHT DN is defined as a Pilot DN.

When Group Hunting terminates on an idle station with Call Forward No Answer by Call Type active, treatment will be the same as in the case of CFNA.

Call Detail Recording on Redirected Incoming Calls

For the Call Detail Recording on Redirected Incoming Calls feature, in the case of Group Hunt, the Pilot DN is the one before the last set in the redirection chain.

Call Transfer

Any call can be transferred to a Group Hunt Pilot DN. If there are no idle sets available for the call transfer, the call is queued to the Pilot DN and the caller receives ringback tone. If the call cannot be queued because the queue threshold has been reached, the caller receives busy tone.

Call Waiting

Call Waiting to a Pilot DN is not be supported.

Camp-on

Camping an incoming call on to a Pilot DN is not be supported.

Digit Display and Name Display

Until a call is answered, the calling party sees the dialed DN. When the call is answered, the caller sees the dialed DN appended with the DN and name of the calling party, if Calling Party Name Display (CPND) is equipped. The terminating set will always see the originating DN appended with a Pilot DN.

Digital Private Network Signaling System (DPNSS1) and Digital Access Signaling System (DASS2) Uniform Dialing Plan (UDP) Interworking

Only basic DPNSS1 UDP calls are supported with Group Hunting. Interactions between DPNSS1 Supplementary Services and Group Hunting are not supported.

DPNSS1 Diversion

Only simple DPNSS1 calls support Group Hunting. All DPNSS1 supplementary services do not support Group Hunting.

Do Not Disturb

Do Not Disturb (DND) has priority over Group Hunting. Group Hunting will skip over sets with DND active.

Enhanced Night Service

If a Pilot DN is defined as one of the NITE DNs from the list associated with the Trunk Night Group, then incoming calls directed to the Pilot DN will be presented to the next idle DN in the hunt group.

Electronic Switched Network

Group Hunting can be applied to Network calls. An Electronic Switched Network (ESN) access code (trunk steering code), if encountered during Group Hunting, will terminate the hunting sequence.

Hunting

Group Hunting has priority over Hunting. If the DN attempted for termination by Group Hunting has HTA COS, and if it is busy, Group Hunting continues with the next DN in the group instead of following the DN's hunting configuration.

ISDN QSIG and EuroISDN Call Completion

Call Completion to Busy Subscriber cannot be applied to Pilot DN when no idle set is located during a Group Hunt call.

Last Number Redial Stored Number Redial

A Pilot DN will be stored as a Last Number Redial (LNR) and Stored Number Redial (SNR) number when it is dialed directly.

Make Set Busy

Make Set Busy (MSB) has priority over Group Hunting. Group Hunting will skip over sets with MSB active.

Multiple Appearance Directory Number

While Multiple Appearance DNs (MADN) single call arrangements are treated the same as Single Appearance DNs (SADN), MADN multiple call arrangements must be avoided in a Group Hunt list.

With MADN multiple call arrangement, the idle or busy status of the MADN is determined by the Terminal Number (TN) data block of the prime appearance of the called DN. If there is more than one prime appearance of the called DN, the idle or busy status is then selected from the last TN in the DN block for the MADN (DNB prompt in LD 22). This means that there can be idle appearances of the MADN, while the hunt cycle regards them as busy and attempts to terminate on the next idle member of the Group Hunt list.

If an MADN multiple call arrangement must be used, a supervisor set must be assigned to the hunt group. This supervisor set must be given the one and only prime appearance of the MADN. Any other appearance must have the MADN programmed as a secondary DN (any DN key other than 0). In this way, the supervisor set controls the status of the MADN and thus the Group Hunt treatment. If the supervisor set is busy, the hunt does not terminate on the MADN.

Multi-Party Operations

As per the existing Multi-Party Operations (MPO) feature, recovery of misoperation of Call Transfer will not be applied to incoming calls which are transferred on ringing to a Pilot DN by transferring parties who are waiting in GPHT queues for service.

Night Answer by Time of Day

If a Pilot DN is defined as one of the NITE DNs in LD 15, then incoming calls directed to the Pilot DN will be presented to the next idle DN in the group. At the instant of changeover (change from one night DN to another), Group Hunting, if still active, will keep on hunting for the next idle DN in the group.

Night Service

If a Pilot DN is defined as a NITE DN or trunk NITE DN, then incoming calls directed to the NITE DN or trunk NITE DN will be presented to the next idle station in the hunt group.

On Hold on Loudspeaker

Group Hunt to a loudspeaker DN can be programmed, but will be ignored if configured as Make Set Busy (MSB) by call processing.

Override Ring Again

Override and Ring Again will not be supported.

Recall to Same Attendant

Calls redirected from a Group Hunt list through the listed DN or flexible attendant DN, and transferred back to the Pilot DN, are recalled if the Slow Answer Recall Timer expires. However, in practical configurations, the hunt terminates on the entry with the listed DN or attendant DN before the Slow Answer Recall Timer expires; consequently, the call is not redirected to that DN and presented on the applicable ICI key on the console. Therefore, the call is never presented as a recall, so that Recall to the Same Attendant does not apply.

Recorded Announcement

Calls which are queued against the Group Hunt Pilot DN cannot receive Recorded Announcement.

Remote Call Forward

If Call Forward All Calls is activated remotely, the interaction with Group Hunting is the same as Call Forward All Calls.

Ring Again on No Answer

Ring Again on No Answer cannot be applied if the DN dialed was a Pilot DN.

Slow Answer Recall

Calls extended by the attendant to the Group Hunt Pilot DN are recalled to the same attendant, after the Slow Answer Recall timer expires. This only applies to a standalone configuration; Network Attendant Service (NAS) is not supported.

Tenant Service

If a Pilot DN is defined as a Tenant NITE DN, then incoming calls directed to the Pilot DN will be presented to the next idle DN in the hunt group.

Total Redirection Count

Group Hunt takes precedence over the Total Redirection Count feature, in that the TRCNT limit is not applied to a Group Hunt call.

Warning Tone

Warning Tone is not applied to queued calls, if the French Type Approval package (197) is not equipped. If the French Type Approval package (197) is equipped, a warning tone of Camp-on can be provided to the first active member of a Group Hunt list that has Warning Tone Allowed (WTA) Class of Service (COS). Any new call in the queue is announced to the next set in the hunt chain that has WTA COS.

16-Button Digitone and Multifrequency Operation

Group Hunt Pilot DN (GRHP) function will not be supported. Group Hunting and Speed Call DN Access can be accessed through the Autodial function.

Feature packaging

Group Hunt requires the following packages:

For markets other than France:

- Group Hunt/DN Access to SCL (PLDN) package 120, which has the following dependencies:
    - System Speed Call (SSC) package 34
    - International Supplementary Features (SUPP) package 131 where applicable
    - Flexible Feature Codes (FFC) package 139

For the French market only:

- French Type Approval (FRTA) package 197 and Group Hunt/DN Access to SCL (PLDN) package 120, which has the following dependencies:
    - System Speed Call (SSC) package 34
    - International Supplementary Features (SUPP) package 131
    - Flexible Feature Codes (FFC) package 139

Feature implementation

Task summary list

The following is a summary of the tasks in this section:

1. Table 141: LD 22 - Verify that software package 120 is equipped. on page 452

   Verify that software package 120 is equipped.
2. Table 142: LD 17 - Enter the number of Group Hunt lists allowed in the system. on page 452

   Enter the number of Group Hunt lists allowed in the system.
3. Table 143: LD 15 - Enter a Group Hunt PLDN. on page 453

   Enter a Group Hunt PLDN.
4. Table 144: LD 10 - Enter a Group Hunt Pilot DN (PLDN). on page 453

   Enter a Group Hunt Pilot DN (PLDN).
5. Table 145: LD 11 - Enter a Group Hunting Denied (GHD) key and enter a Group Hunt PLDN. on page 454

   Enter a Group Hunting Denied (GHD) key and enter a Group Hunt PLDN.

   LD 11 is modified to disallow the removal of the last appearance of a Single Call Non-ringing (SCN), Single Call Ringing (SCR), Multiple Call Non-ringing (MCN), or

Multiple Call Ringing (MCR) DN which is part of a Group Hunt list. This ensures the DN is removed from all Group Hunt lists prior to being removed from a set.

6. Table 146: LD 12 - Enter a Group Hunt PLDN. on page 455

   Enter a Group Hunt PLDN.

7. Table 147: LD 14 - Enter a Group Hunt PLDN. on page 455

   Enter a Group Hunt PLDN.

8. Table 148: LD 18 - Create or modify Group Hunt lists. on page 455

   Create or modify Group Hunt lists.

   Responses are required to the following prompts when a Group Hunt list is modified, created, or removed. This overlay disallows the removal of a Group Hunt list if it is still associated with a PLDN that exists in LD 57. This ensures the PLDN is removed prior to removing the Group Hunt list.

9. Table 149: LD 20 - Print the Group Hunt list. on page 456

   Print the Group Hunt list. This includes PLDN entries.

10. Table 150: LD 57 - Define, change, or print data associated with FFC. on page 456

    Define, change, or print data associated with FFC.

11. Table 151: LD 57 - Configure Flexible Feature Codes data block for Group Hunt Termination. on page 457

    Configure Flexible Feature Codes data block for Group Hunt Termination.

12. Table 152: LD 81 - Count or list all stations equipped with the GHD key. on page 457

    Count or list all stations equipped with the GHD key.

13. Table 153: LD 83 - Display the GHD key data. on page 458

    Display the GHD key data.

Table 141: LD 22 - Verify that software package 120 is equipped.

Prompt	Response	Description
REQ	PRT	Print information.
TYPE	PKG	Type of information to print: equipped packages.

Table 142: LD 17 - Enter the number of Group Hunt lists allowed in the system.

Prompt	Response	Description
REQ:	CHG	Change the Configuration Record.

Prompt	Response	Description
TYPE:	PARM	System Parameters
...	...	
MSCL	0-8191	Number of Group Hunt lists allowed in the system.

Table 143: LD 15 - Enter a Group Hunt PLDN.

Prompt	Response	Description
REQ:	NEW	Add new data.
TYPE:	NIT	Night Service Options
...		
- NIT1	x...x	First Night service by time of day DN can be defined as a PLDN.
- TIM1	...	Hour and Minute for First Night Service DN.
- NIT2	x...x	Second Night service by time of day DN can be defined as a PLDN.
- TIM2	...	Time for Second Night Service DN.
- NIT3	x...x	Third Night service by time of day DN can be defined as a PLDN.
- TIM3	...	Time for Third Night Service DN.
- NIT4	x...x	Fourth Night service by time of day DN can be defined as a PLDN.
- TIM4	...	Hour and Minute for Fourth Night Service DN.

Table 144: LD 10 - Enter a Group Hunt Pilot DN (PLDN).

Prompt	Response	Description
REQ:	NEW CHG	Add new data, or change existing data.
...		
IAPG	0-15	Meridian Link Unsolicited Status Message (USM) group
HUNT	x...x	Hunt DN of the next station in the Hunt chain. Hunt DN can be defined as a PLDN.
...		
AACD	(NO) YES	Associate set (AST) telephone
FTR	FTR	Enter the feature name and related data.

Prompt	Response	Description
	EFD x...x	External Flexible call forward DN (a Group Hunt pilot can be entered.) External Call Forward No Answer DN can be defined as a PLDN.
	EHT x...x	External Hunt DN External Hunt DN can be defined as a PLDN.
	FDN x...x	Flexible Call Forward No Answer Call Forward No Answer DN can be defined as a PLDN.

Table 145: LD 11 - Enter a Group Hunting Denied (GHD) key and enter a Group Hunt PLDN.

Prompt	Response	Description
REQ:	NEW CHG	Add new data, or change existing data.
...		
AOM	0-2	Number of Add-on Modules. AOM appears if TYPE = M2216 and M2616
FDN	x...x	Flexible CFNA DN Call Forward No Answer DN can be defined as a PLDN.
...		
ICT	0-<NIPN>	Intercept Computer Terminal or printer number Number of Intercept Positions (NIPN) is defined in LD 15.
EFD	x...x	Flexible CFNA DN for External calls External Call Forward No Answer DN can be defined as a PLDN.
HUNT	x...x	Hunt DN of next station in hunt chain Hunt DN can be defined as a PLDN.
EHT	x...x	External Hunt DN External Hunt DN can be defined as a PLDN.
...		
LANG	(0)-5 X	Language choice for Automatic Wakeup (AWU) calls. Prompted with Multi-language Wakeup (MLWU) package 206.
KEY	xx aaa yyyy	Telephone key assignments.
	xx CFW yy z...z	Call Forward key Key number (xx), Call Forward function (CFW), length (yy), Call Forward target DN (z...z) can be defined as a PLDN.
	xx GHD	Key number (xx), Group Hunting Denied function (GHD).

Prompt	Response	Description
		The GHD key is added to allow a station user to toggle the Primary (key 0) Directory Number (PDN) in and out of all groups of which that PDN is a member.

Table 146: LD 12 - Enter a Group Hunt PLDN.

Prompt	Response	Description
REQ	NEW CHG	Add new data, or change existing data.
...		
ICP	(NO) YES	Intercept Computer available.
AADN	x...x	Attendant Alternate Answering DN. Alternate Answering DN can be defined as a PLDN.

Table 147: LD 14 - Enter a Group Hunt PLDN.

Prompt	Response	Description
REQ	NEW CHG	Add new data, or change existing data
...		
NGRP	(0)-9	Night Service Group number
NITE	x...x	Night Service directory number Night service DN can be defined as a PLDN.
ATDN	x...x	Auto-terminate DN Auto-terminate DN can be defined as a PLDN.
MNDN	x...x	Manual Directory Number Manual DN can be defined as a PLDN.

Table 148: LD 18 - Create or modify Group Hunt lists.

Prompt	Response	Description
REQ	CHG MOV NEW OUT	Change, move, create, or remove a data block.
TYPE	GHT	Group Hunt data block
LSNO	1-8190 <CR>	List Number Group Hunt lists Use only when REQ = CHG and TYPE = GHT
CUST	0-99 0-31	Customer number, as defined in LD 15.
PLDN	x...x	Pilot DN: Prompted when REQ = NEW or CHG and LSNO = <CR>.
DNSZ	4-(16)-31	Directory Number Size Maximum length of DN allowed for Group Hunt list. Prompted when REQ = NEW or

Prompt	Response	Description
		CHG and LSNO = <CR>. After DNSZ is defined, it should not be changed. Print the list in LD 20, remove it with REQ = OUT, and rebuild the list with the new DNSZ.
SIZE	1-96 1-1000	Size of list Maximum DNs in Group Hunt list Range is 1 to 96 entries if response to TYPE is GHT. Range is 1 to 1000 if response to TYPE is SCL or SSC.
STOR	...	Store: Enter entry (member) number (x...x) and Group Hunt target DN (y...y). For TYPE = GHT the input format is Group Hunt entry and digits stored against it: Where:
	x...x y...y <CR>	x...x = GHT entry number from 0 to 95 y...y = digits stored Stop STOR prompt In Group Hunting the member number must conform with the SIZE; the number of digits must conform with DNSZ.
	xx<space><CR>	Remove entry
WRT	(YES) NO	Write Write information to data store.

Table 149: LD 20 - Print the Group Hunt list.

Prompt	Response	Description
REQ	PRT	Print data block.
TYPE	GHT	Type of data block: Group Hunt list.
LSNO	1-8190 <CR>	Group Hunt lists. Print all lists
SIZE 1-96		The list size is printed if a Group Hunt list number is entered against the LSNO prompt. SIZE is not printed if <CR> is entered against the LSNO prompt.

Table 150: LD 57 - Define, change, or print data associated with FFC.

Prompt	Response	Description
REQ	CHG NEW	Add new data, or change existing data.
TYPE	FFC	Flexible Feature Codes data block.
CUST	0-99 0-31	Customer number, as defined in LD 15
FFCT	<CR>	Flexible Feature Confirmation Tone.
CODE	PLDN	Code to be modified or created: Pilot DN.
- PLDN	xxxx <CR>	Pilot DN: enter Pilot DN to be modified or created. Enter a carriage return to proceed to next prompt.

Prompt	Response	Description
-- USE	GPHT	USE: enter USE for Pilot DN Group Hunting.
-- LSNO	xxxx	List Number: enter Group Hunt list number. Group Hunt list must exist in LD 18.
-- HTYP	(LIN) RRB	Hunting Type: enter either (Linear) or Round Robin as the type of hunting to be used for the Group Hunt list.
-- CFWI	(NO) YES	Call Forward All Calls Idle: Where: enter NO if Group Hunting is to skip idle stations with Call Forward All Calls active, or enter YES if Group Hunting is to terminate on idle stations with Call Forward All Calls active.
MQUE	0 1 (ALL) ACTM	Maximum Queue (maximum number of calls allowed to queue against the Pilot DN. Where: Enter 0 to deny all calls from queuing Enter 1 to allow only one no call to queue Enter ALL, the default, to allow all calls to queue or Enter ACTM to limit the number of calls allowed to queue to be less than or equal to the number of active members of the Group Hunt list.

Table 151: LD 57 - Configure Flexible Feature Codes data block for Group Hunt Termination.

Prompt	Response	Description
REQ	CHG NEW	Add new data, or change existing data.
TYPE	FFC	Flexible Feature Codes data block.
CUST	0-99	Customer number, as defined in LD 15.
FFCT	<CR>	Flexible Feature Confirmation Tone.
CODE	GHTA	Code to be modified or created: Group Hunt Termination Allowed.
- GHTA	x...x	Enter code to be dialed to allow Group Hunt termination on a set.
CODE	GHTD	Code to be modified or created: Group Hunt Termination Denied.
- GHTD	x...x	Enter code to be dialed to deny Group Hunt termination on a set.

Table 152: LD 81 - Count or list all stations equipped with the GHD key.

Prompt	Response	Description
REQ	NEW CHG	Add new data, or change existing data.

Prompt	Response	Description
	CNT	Print a count of telephones equipped with the features specified in response to the FEAT prompt.
	LST	List telephones equipped with the features specified in response to the FEAT prompt.
...	...	
FEAT	GHD	Group Hunt Deactivation key
FEAT	<CR>	<CR> enters the default

Table 153: LD 83 - Display the GHD key data.

Prompt	Response	Description
REQ	TNB	Print the TN blocks in Designation order.
...	...	
... KEY xx GHD ...		GHD key data is printed each time it is configured on a set.

Feature operation

No specific operating procedures are required to use this feature.

Chapter 65: Group Hunting Queuing Limitation

Contents

This section contains information on the following topics:

Feature description

The Group Hunting Queuing Limitation feature restricts the maximum number of calls that can be queued against a Pilot Directory Number (DN).

The Group Hunting Queuing Limitation feature adds a prompt (MQUE - Maximum Queue) in LD 57 which allows a user to define a limit on the number of calls queued against a Pilot DN. The valid inputs are:

- 0 No calls can be queued.
- 1 One call can be queued.
- ALL All calls may be queued (default).
- <CR> Setting is left as is.

When the maximum is exceeded, the next call that attempts to queue will be given busy treatment.

The following are examples of the treatment calls receive with MQUE set to the various settings:

MQUE set to 0

1. Pilot DN Z can hunt two sets, A and B. Both of these sets are busy.
2. Set (or DID trunk) C dials Pilot DN Z.
3. If C is a set it receives busy tone and cannot be queued, but if it is a Direct Inward Dialing (DID) trunk it receives whatever busy treatment has been requested for that DID route.

MQUE set to 1

1. Pilot DN Z can hunt two sets, A and B. Both of these sets are busy.
2. Set C dials the Pilot DN Z. The call is queued.
3. Set D dials the Pilot DN Z. This call receives busy tone.
4. Set A goes on-hook first. The first call is presented to set A.

MQUE set to ALL

This option disables the Group Hunt Queuing Limitation enhancement. With ALL selected there is no limit as to the number of calls which can be queued against the Pilot DN.

Operating parameters

There are no operating parameters associated with this feature.

Feature interactions

Camp-on

No Camp-on tone is provided for Group Hunting.

Music

No music is provided for Group Hunting Queuing Limitation.

Feature packaging

This feature is packaged under the International Supplementary Features (SUPP) package 131; Group Hunt/DN Access to SCL (PLDN) package 120; and all PLDN package (120) dependencies.

Feature implementation

Table 154: LD 57 - Modify, create, or print Flexible Feature Codes

Prompt	Response	Description
REQ	CHG NEW PRT	Request: Modify, create, or print data block.
TYPE	FFC	Flexible Feature Codes.
CUST	xx	Customer number, as defined in LD 15
...	xxxx	Automatic Call Distribution Directory Number.
CODE	PLDN	Code to modify, create: Pilot Directory Number.
- PLDN	XXXX	Enter PLDN to be modified or created.
- - USE	GPHT	Use of this PLDN, Group Hunt Pilot DN.
...		
- - CFWI	...	
- - MQUE		Maximum Queue – Maximum number of calls that may be queued against a Group Hunt Pilot DN.
	(ALL) 0 1 <cr>	All calls may be queued (default). No calls can be queued. One call may be queued. Use default setting if this is a new Pilot DN, leave existing setting as is if the Pilot DN is being modified.
...		

Printing the FFC data block will include the MQUE prompt and its response.

Feature operation

No specific operating procedures are required to use this feature.

Chapter 66: Group Hunting Queuing Limitation Enhancement

Contents

This section contains information on the following topics:

Feature description

This feature introduces a Group Hunt Threshold (GHTH) which limits the number of calls that can be linked in the Pilot DN (PLDN) waiting queue. The threshold is calculated dynamically and is set equal to the number of active members in the group hunt list. This limits the number of calls in the PLDN queue to one per active member of the group hunt list. The feature is activated using the MQUE prompt in LD 57. The MQUE prompt now accepts a response of ACTM (Active Members) to invoke the GHTH.

Along with the Group Hunt Threshold this feature introduces the use of Camp-on tone to indicate that there are queued calls.

Operating parameters

Although Automatic Call Distribution (ACD) DNs, Integrated Voice Messaging Service (VMS) DNs, Listed Directory Numbers (LDNs), Route access codes, Electronic Switched Network

(ESN) access codes, and other Pilot DNs can be defined as a group hunt list member, it is recommended that they are not used due to the fact that these targets are considered as active when computing the threshold, regardless of their actual state.

Feature interactions

Call Forward by Call Type Call Forward No Answer

An external call is made to the PLDN. An idle group hunt list member station is rung but does not answer. If the member station has Call Forward No Answer (FNA) or Call Forward by Call Type Allowed (CFTA) Class of Service, then the call is transferred to the attendant after the number of ring cycles defined for Call Forward No Answer has been reached. If the call is an internal call, then the system searches for another idle group hunt list member.

Call Transfer

If a call is transferred to the PLDN, and all Group Hunt list members are busy, the call is queued to the PLDN, if the number of queued calls is less than the Group Hunt Threshold limit. If the number of queued calls has reached the Group Hunt Threshold limit, the call is not queued and busy tone is returned to the transferring party.

Feature packaging

This feature is packaged under French Type Approval (FRTA) package 197 and Group Hunt/ DN Access to SCL (PLDN) package 120.

Feature implementation

Table 155: LD 57 - For the Group Hunt Queuing Limitation Enhancement, responses to the following prompts are required.

Prompt	Response	Description
REQ	CHG NEW	Modify or create data block.

Prompt	Response	Description
TYPE	FFC	Flexible Feature Codes data block.
CUST	xx	Customer number, as defined in LD 15
FFCT	<CR>	Flexible Feature Confirmation Tone.
CODE	PLDN	Pilot DN.
PLDN	xxxx <CR>	Enter Pilot DN to be modified or created. Enter a carriage return to proceed to next prompt
USE	GPHT	Enter use for Pilot DN. Group Hunting.
LSNO	xxxx	Enter group hunt list number. Group hunt list must exist in LD 18.
HTYP	(LIN) RRB	Enter either (Linear) or Round Robin as the type of hunting to be used for the group hunt list.
CFWI	(NO) YES	Call Forward All Calls Idle: enter NO if Group Hunting is to skip idle stations with Call Forward All Calls active, or enter YES if Group Hunting is to terminate on idle stations with Call Forward All Calls active.
MQUE	ACTM	Maximum Queue (maximum number of calls allowed to queue against the Pilot DN.): enter ACTM (Active Members) to limit the number of calls allowed to queue to be less than or equal to the number of active members of the group hunt list.

Feature operation

A group hunt list member is active if any call to the PLDN can terminate on the member set when it is idled. Conversely, a group hunt list member is not active if Group Hunt Termination Denied (GHTD) Flexible Feature Code (FFC) is dialed, and, or, Call Forward All Calls is active for the member and Call Forward Ignore (CFWI) in LD 57 is NO for the PLDN.

When the response to the MQUE prompt is ACTM and a call is routed to or dials a PLDN and it cannot terminate on an active member station, the call is linked to the PLDN queue (if the number of calls waiting in the PLDN queue is lower than the threshold limit). If the number of calls waiting in the PLDN queue reaches the threshold limit, calls are no longer linked to the PLDN queue. If the call is an internal call or attendant-extended call, busy tone is given to the originating party. If the originating call is a Direct Inward Dialing (DID) or Central Office (CO) trunk, it is routed to the attendant as a Call Forward Busy call. The attendant console display shows the PLDN (the attendant cannot Break-in or Busy Verify to a PDLN).

When a call is queued against a PLDN, Camp-on tone is given to the first member of the group hunt list having Warning Tone Allowed (WTA) Class of Service. If none of the members has WTA Class of Service, the Camp-on tone is not provided.

Chapter 67: Handset Volume Reset

Contents

This section contains information on the following topics:

Feature description

This feature is supported by the A44 chip in Meridian digital sets and causes a telephone's handset volume to be reset to a specified volume every time that the telephone user hangs up or uses handsfree. If the user wishes to adjust the volume, the user must manually do so for each call.

Operating parameters

There are no operating parameters associated with this feature.

Feature interactions

There are no feature interactions associated with this feature.

Feature packaging

This feature is included in base system software.

Feature implementation

Table 156: LD 17 - Define the Handset Volume Reset setting.

Prompt	Response	Description
REQ	CHG	Change.
TYPE	CFN ATRN	Configuration Record. Aries Transmission.
...		
ATRN	YES	Aries (Meridian Modular) transmission parameters. Only prompted if response to TYPE is CFN.
...		
- VOLR	(NO) YES	Volume Reset.

Feature operation

When a transmission download occurs, following a SYSLOAD or when the set line cord is plugged in, the option setting defined in LD 17 is included in the message. The message is interpreted by the set firmware and the appropriate setting is applied. A system initialization will not download this message.

Chapter 68: Handsfree Transmission Parameter Download

Contents

This section contains information on the following topics:

Feature description

This feature provides parameters to support the handsfree transmission parameter download on Meridian 1 proprietary telephones. These parameters are downloaded to each telephone upon system reload or set power-up, after the handset parameters.

Two prompts are defined in LD 17 allowing control of handsfree transmit and receive loudness ratings.

Operating parameters

There are no operating parameters associated with this feature.

Feature interactions

There are no feature interactions associated with this feature.

Feature packaging

This feature is included in base system software.

Feature implementation

Task summary list

The following is a summary of the tasks in this section:

1. Table 157: LD 11 - Modify the system hardware and software parameters. on page 470

   Modify the system hardware and software parameters.
2. Table 158: LD 17 - Create or modify the digital telephone data blocks. on page 471

   Create or modify the digital telephone data blocks.
3. Table 159: LD 22 - Print Handsfree transmission parameter download settings. on page 471

   Print Handsfree transmission parameter download settings.

Table 157: LD 11 - Modify the system hardware and software parameters.

Prompt	Response	Description
...		
CLS	(HFD) HFA	Digital Telephone Handsfree (denied) allowed. Not allowed on M2006, M2008, M2016S, or M2216 sets. M2016 must be defined as a 2616 with HFD Class of Service allowed for M2616 sets.

Prompt	Response	Description
		HFA is the default for M2317 sets.

Table 158: LD 17 - Create or modify the digital telephone data blocks.

Prompt	Response	Description
REQ	CHG	Change.
TYPE	CFN ATRN	Configuration Record. Aries Transmission.
ATRN	YES	Meridian Modular telephone transmission parameters. Only prompted if response to TYPE is CFN.
- HRLR	(0)-8 32-40	Handsfree receive objective loudness rating.
- HTLR	(0)-11 32-54	Handsfree transmit objective loudness rating.

Table 159: LD 22 - Print Handsfree transmission parameter download settings.

Prompt	Response	Description
REQ	PRT	Print.
TYPE	CFN ATRN	Configuration. Meridian Modular telephone transmission parameters. When the Handsfree transmission download parameters are printed they are output as their dB value (that is, an input of 0 in response to the HRLR prompt is printed as +42.00, while a response of 0 to the HTLR prompt is printed as -44.00).

Input value	HRLR (dB)	HTLR (dB)		Input value	HRLR (dB)	HTLR (dB)
0	+42.00	-44.00		21	N.A.	N.A.
1	+42.85	-43.50		22	N.A.	N.A.
2	+43.70	-43.50		23	N.A.	N.A.
3	+44.55	-43.00		24	N.A.	N.A.
4	+45.40	-42.50		25	N.A.	N.A.
5	+46.25	-42.00		26	N.A.	N.A.
6	+47.10	-42.00		27	N.A.	N.A.
7	+47.95	-41.50		28	N.A.	N.A.
8	+48.80	-41.00		29	N.A.	N.A.
9	N.A.	-40.50		30	N.A.	N.A.
10	N.A.	-40.50		31	N.A.	N.A.

Input value	HRLR (dB)	HTLR (dB)		Input value	HRLR (dB)	HTLR (dB)
11	N.A.	-40.00		32	+42.00	-44.00
12	N.A.	N.A.		33	+41.15	-44.50
13	N.A.	N.A.		34	+40.30	-45.00
14	N.A.	N.A.		35	+39.45	-45.00
15	N.A.	N.A.		36	+38.60	-45.50
16	N.A.	N.A.		37	+37.75	-46.00
17	N.A.	N.A.		38	+36.90	-46.50
18	N.A.	N.A.		39	+36.05	-46.50
19	N.A.	N.A.		40	+35.20	-47.00
20	N.A.	N.A.		41	N.A.	-47.50
42	N.A.	-48.00		53	N.A.	-52.00
43	N.A.	-48.00		54	N.A.	-52.50
44	N.A.	-48.50		55	N.A.	N.A.
45	N.A.	-49.00		56	N.A.	N.A.
46	N.A.	-49.50		57	N.A.	N.A.
47	N.A.	-49.50		58	N.A.	N.A.
48	N.A.	-50.00		59	N.A.	N.A.
49	N.A.	-50.50		60	N.A.	N.A.
50	N.A.	-51.00		61	N.A.	N.A.
51	N.A.	-51.00		62	N.A.	N.A.
52	N.A.	-51.50		63	N.A.	N.A.

All values are Objective Loudness Ratings (OLR) measured without inserted loss or gain for trunk card interfaces and computed per IEEE methods. Receive ratings are at maximum volume. Transmit ratings are measured in an anechoic environment with less than 25 dBa room noise.

Feature operation

Whenever a download occurs, following SYSLOAD or when the telephone line cord is plugged in, the Relative Loudness Rating settings defined in LD 17 are included in the message. The message is interpreted by the telephone firmware and the appropriate settings are applied.

Chapter 69: Held Call Clearing

Contents

This section contains information on the following topics:

Feature description

The Held Call Clearing feature allows both the active call and the held call to be released when the user of a Meridian 1 proprietary telephone replaces the handset. Pressing the Release key only releases the active call.

For Single Appearance DNs, an on-hook action from a station clears the active call and all held calls belonging to that station. Pressing the Release key clears only the active call on the station. Activated feature keys, not involving an active or held call on the set, are not affected by the on-hook or Release key action. If an on-hook action occurs while a feature key is being activated, the system follows the Release key functionality. In most cases, this causes the feature key to be idled.

Where several DNs appear on the same set, an on-hook or Release key action does not affect any unanswered incoming calls which are unanswered call waiting calls or are in a ringing state, whether or not the ringing tone is audible. Answered call waiting calls (those which are active or being held) are cleared by an on-hook action. A Release key action clears only active call-waiting calls.

For Multiple Appearance DNs, an on-hook action from a station having one appearance of a Multiple Appearance DN clears only the current active call and the held calls belonging to that station. Pressing the Release key clears only the active call on the station. Calls active or held on another appearance of the same DN, on a different set, are not affected.

For Data DNs, an on-hook or Release key action clears active data calls on a Data DN. A data call is considered active on a set when the "Data Shift" LED is lit. A call on a Data DN which is not the set's active call is not affected by an on-hook or Release key action. For data terminals, only active data calls are released by an on-hook or Release key action.

Operating parameters

The Held Call Clearing feature cannot be used on analog (500/2500 type) telephones.

Feature interactions

Call Park

A call put on hold during a Call Park is not cleared by an on-hook action on that set.

Call Transfer

Active Call Transfer calls are cleared by either an on-hook or Release key action. Held Call Transfer calls are cleared only by an on-hook action, and not by a Release key action.

Called Party Control on Internal Calls

With Called Party Control on Internal Call enabled, a call on hold is not cleared when the calling party releases. This occurs whether or not the Held Call Clearing feature has been activated.

Conference

Active Conference calls are cleared by an on-hook or Release key action. Conference calls being held are cleared by an on-hook action only, and not by a Release key action. In either case, all other parties on the conference remain connected.

Handsfree

For a set equipped with a Handsfree add-on unit, the on-hook action is suppressed if the Handsfree key is pressed simultaneous to the on-hook. In this case, all active and held calls on the set are not affected by the on-hook action. For a Meridian M1000 or digital telephone, an on-hook action does not affect an active call on the set. In all cases, a Release key action clears an active call, whether in handsfree mode or not.

Misoperation on Call Transfer

An on-hook action clears a call that is put on hold during Call Transfer. This action may lead to a misoperation if the user of the set from which the call is being transferred goes on-hook before a valid DN is dialed. In this case, the misoperation is handled in the same manner as for a 500-type set.

On Hold on Loudspeaker

Going on-hook when Held Call Clearing is activated will clear the loudspeaker as for a normal held call. Therefore, it is recommended not to use this feature with the On Hold on Loudspeaker feature.

Feature packaging

This feature is packaged under International Supplementary Features (SUPP), package 131.

Feature implementation

Table 160: LD 15 - Activate Held Call Clearing in response to the HCC prompt to implement this feature.

Prompt	Response	Description
REQ:	NEW CHG	Add, or change.
TYPE:	FTR	Features and options

Prompt	Response	Description
...		
- HCC	(NO) YES XFER	Held Call Clearing is to be activated, (deactivated) or set to transfer the held call.

Feature operation

Place the handset of your Meridian 1 proprietary telephone on-hook to release both the active and held call.

Pressing the Rls key only releases the active call.

Chapter 70: History File

The History File provides the capability to allocate an area of protected data to store system messages until a printout is requested by a technician. The size of the History File is defined on a system basis and can be up to 65 534 characters. Since one word of protected data stores two History File characters, the size of the History File is up to 32 767 words of protected data.

For a complete description of the History File, see *Avaya System Management Reference, NN43001-600*.

Chapter 71: Hong Kong Digital Trunk Interface

Contents

This section contains information on the following topics:

Feature description

This feature modifies the 1.5 Mbps Digital Trunk Interface (DTI) in order to allow the system to interface with the Hong Kong Telephone Company (HKT). The design modification alters the Dual-tone Multifrequency (DTMF) signaling protocol to conform with the HKT requirements. This DTMF design modification involves altering the AB bit protocol used in the DID/TIE convention, which is the convention used for the system to HKT connectivity. The AB bit values for the normal DID/TIE convention are reversed for the HKT interface. For example, if the normal convention for a DID/TIE going off-hook requires that AB bit values 0 and 0 be sent to the far end, the convention for HKT is that AB bit values 1 and 1 be sent.

This feature also meets the requirement of requiring the system, after a trunk seizure, to wait 600 milliseconds before accepting the dialed digits from the far end. This 600 milliseconds dialing delay is provided by the Dial Delay Timer, whose maximum configurable delay has been extended to 1,023 milliseconds. The timer is set on a per-route basis.

Operating parameters

Hong Kong Digital Trunk Interface modification applies only to 1.5 Mbits DTI trunks.

Feature interactions

There are no feature interactions associated with this feature.

Feature packaging

This feature is packaged under 1.5 Mbps Digital Trunk Interface (DTI) package 75.

Feature implementation

Task summary list

The following is a summary of the tasks in this section:

1. Table 161: LD 14 - Respond to the CLS prompt by entering HKA to allow the Hong Kong feature modification. on page 480

   Respond to the CLS prompt by entering HKA to allow the Hong Kong feature modification.
2. Table 162: LD 16 - Enter a dialing pause for the Dial Delay Timer at the Time prompt: on page 481

   Enter a dialing pause for the Dial Delay Timer at the Time prompt:

Table 161: LD 14 - Respond to the CLS prompt by entering HKA to allow the Hong Kong feature modification.

Prompt	Response	Description
...		

Prompt	Response	Description
CLS	(HKD) HKA	Hong Kong DTI (denied) allowed. May only be used with DTI TNs with DTN CLS on DID or TIE routes.

Table 162: LD 16 - Enter a dialing pause for the Dial Delay Timer at the Time prompt:

Prompt	Response	Description
...		
- TIMR	DDL 0-(70)-511	Dial Delay Timer. A value of 0 disables the timer.

Feature operation

No specific operating procedures are required to use this feature.

Chapter 72: Hot Line, Enhanced

Contents

This section contains information on the following topics:

Feature description

Enhanced Hot Line (EHOT) provides Hot Line services to telephones with programmable keys. This feature is designed for, and is compatible with, analog (500/2500 type) telephones and Meridian 1 proprietary telephones. All capabilities from Flexible Hot Line (HOT) are provided to any key/lamp pair for one- and two-way Hot Lines on a per station basis. When the handset is lifted, or when a preprogrammed key is activated, the system speed calls a preprogrammed DN. Hot Lines access a set of terminal numbers programmed by direct entry using LD 11, or by list entry such as by System Speed Call (SSC) using LD 18. There is no difference in operation for the Hot Line user.

Once a Hot Line call enters the ringing state, it is the same as any normal call.

Enhanced Hot Line (EHOT) allows a distinction between analog (500/2500 type) telephone Hot Lines and manual Hot Lines without dial capabilities. For example, telephones with EHOT enabled and dial facilities support Dial Access features such as Call Transfer or Conference calling.

A Hot Line key can be defined with a Directory Number (DN) of its own, allowing other calls to terminate on that HOT key. The DN must be defined before it can be specified as the DN for a HOT key. For Meridian 1 proprietary telephones, the HOT key must be assigned to a DN during Service Change to create a two-way Hot Line. Analog (500/2500 type) telephones are always two-way Hot Lines, as they always have a DN assigned.

Operating parameters

Incoming calls to Hot Line telephones or keys can be restricted to calls originating from other Hot Line telephones or keys, Voice Call keys, and Group Call keys. This restriction is turned on or off on a per customer basis.

Telephones without a keypad or rotary dial cannot be assigned the Enhanced Hot Line Allowed (EHTA) Class of Service.

A maximum of 31 digits can be stored against a Hot Line telephone or key.

Only one Hot Line list is allowed per customer.

HOT cannot access a list created by the list-entry method for Enhanced Hot Line (EHOT).

A specific Hot Line key on a Meridian 1 proprietary telephone can have access to only one entry in the Hot Line list, but more than one telephone can have access to the same entry.

Analog (500/2500 type) telephones with Manual Line (MNL) Class of Service cannot be defined as Enhanced Hot Line Allowed (EHTA); Enhanced Hot Line Denied (EHTD) is the default. Users of these telephones must continue to use the HOT feature.

If a key is assigned as an EHOT Directory Number (DN), all appearances of that DN must also be EHOT keys.

Feature interactions

Attendant Administration

Use of an attendant console to change the database for EHOT is not supported.

Autodial

Flexible Hot Line and/or Enhanced Hot Line are mutually exclusive with the Autodial feature.

Automatic Answerback

The Automatic Answerback feature is fully compatible with a two-way Hot Line key assigned as the Prime DN.

Automatic Call Distribution

A Hot Line DN key can be assigned to an Automatic Call Distribution telephone.

Automatic Line Selection

Since the Hot Line key acts as a Single Call Ring (SCR) key, incoming ringing line preference can be applied. Outgoing line preference automatically selects a line other than the current Hot Line, so that a Hot Line call is not accidentally activated.

Automatic Redial

An Automatic Redial (ARDL) call can be activated from an Enhanced Hot Line key. However, the call is only redialed when the calling party's HOT key is free.

Call Forward Busy Call Forward No Answer Hunting

Any Hot Line telephone can be assigned Call Forward Busy, Call Forward No Answer and Hunting (excluding Short Hunt) Class of Service, but it applies only to the two-way Hot Line capability.

Call Park

Analog (500/2500 type) Hot Line telephones with EHTA and XFA Class of Service are allowed to park calls using the established Call Park procedures. Once a call is parked on an analog (500/2500 type) Hot Line telephone and the telephone is placed on hook, it cannot be unparked. Parked calls will recall to the parking telephone after the Call Park timeout. Two-way Meridian 1 proprietary telephone Hot Line stations that are equipped with a Call Park key/lamp pair are allowed to park calls in the normal fashion. As with analog (500/2500 type) telephones, a call parked from a Hot Line key cannot be picked up using the same key.

Call Pickup

Telephones with two-way Hot Line keys, and analog (500/2500 type) Hot Line telephones, can be assigned to pickup groups. Incoming Hot Line calls may be picked up by group members. To prevent someone from picking up a Hot Line call, do not put the Hot Line user into a Call Pickup group.

China - Flexible Feature Codes - Busy Number Redial

Busy Number Redial cannot be used on Enhanced Hot Line sets.

Controlled Class of Service

When a Hot Line DN is on a telephone that has Controlled Class of Service activated, Hot Line calls ignore the imposed Class of Service if the System Speed Call (SSC) package is present and the Hot Line list is given an adequate Network Class of Service (NCOS) for the override.

Dial Intercom

The analog (500/2500 type) Hot Line telephones cannot be members of Dial Intercom Groups (DIGs).

Digit Display

A Display key on a telephone with a Hot Line appearance will display the Hot Line target DN data stored for that key.

Display of Calling Party Denied

Display information on sets in a Hot Line call is based on the individual Class of Service of each set.

Enhanced Flexible Feature Codes - Busy Number Redial

The Busy Number Redial feature cannot be used on Enhanced Hotline sets.

Group Call

Hot Lines can be members of a Group Call. They cannot, however, have a Group Call key.

HOT

EHOT and HOT are mutually exclusive. A telephone cannot have both MNL and EHTA Classes of Service.

Internal Call Detail Recording

Hot Line stations can be assigned the appropriate Class of Service that allows Call Detail Recording records to be printed for calls originating on that telephone.

Make Set Busy

Make Set Busy is overridden by the Hot Line feature. If a Meridian 1 proprietary telephone is in Make Set Busy mode, incoming Hot Line calls still terminate (ring) on the telephone.

Important:

The Conference-Hot Line key overrides Make Set Busy only when the terminating key is HOT.

Override

A Hot Line call can be entered using the Override feature.

Permanent Hold

Analog (500/2500 type) telephones with EHTA cannot have Permanent Hold.

Prime Directory Number

If the Hot Line key is assigned to key 0 on a Meridian 1 proprietary telephone, it acts as the prime DN. When the user goes off-hook without selecting a DN key, the Hot Line is activated and the call is placed without further user action.

Private Line

A Hot Line key cannot be a Private Line, as this would defeat the benefits of Private Line service.

Room Status

The Room Status feature is incompatible with any telephone for which going off-hook activates Hot Line.

Speed Call, System

When the System Speed Call (SSC) package is equipped, Hot Line lists have the characteristics and limitations of SSC lists. If the package is not equipped, Hot Line lists function like standard Speed Call lists.

User Selectable Call Redirection

An analog (500/2500 type) telephone with a Hot Line feature cannot use User Selectable Call Redirection, because it cannot access any features through SPRE or FFC.

Voice Call

The terminating DN of a Voice Call arrangement may be the incoming DN of a two-way Hot Line.

When engineering call-modification paths (such as Hunting and Call Forward No Answer), the Hot Line Restriction option will cancel the normal call-modification operation for internal non-Hot Line calls.

Feature packaging

Enhanced Hot Line (HOT) package 70 requires:

- Network Class of Service (NCOS) package 32, and
- System Speed Call (SSC) package 34.

Feature implementation

Task summary list

The following is a summary of the tasks in this section:

1. Table 163: LD 17 - Assign the number of Speed Call lists, including Hot Line lists. on page 489

   Assign the number of Speed Call lists, including Hot Line lists.
2. Table 164: LD 15 - Add or change Enhanced Hot Line for a customer. on page 490

   Add or change Enhanced Hot Line for a customer.
3. Table 165: LD 18 - Prompt sequence on page 490

   Use this prompt sequence to determine if there are enough memory and disk records for new Speed Call lists. Compare the output with the MEM AVAIL and DISK AVAIL values output before the REQ prompt.
4. Table 166: LD 18 - Add or change a Hot Line Speed Call list. on page 490

   Add or change a Hot Line Speed Call list.
5. Table 167: LD 10 - Add Enhanced Hot Line for analog (500 and 2500 type) telephones. on page 491

   Add Enhanced Hot Line for analog (500/2500 type) telephones.
6. Table 168: LD 11 - Allow or deny Enhanced Hot Line for Meridian 1 proprietary telephones. on page 491

   Allow or deny Enhanced Hot Line for Meridian 1 proprietary telephones.

Table 163: LD 17 - Assign the number of Speed Call lists, including Hot Line lists.

Prompt	Response	Description
REQ	CHG	Change.
TYPE	CFN PARM	Configuration Record. System Parameters.
MSCL	0-8191	Maximum number of Speed Call lists.

Table 164: LD 15 - Add or change Enhanced Hot Line for a customer.

Prompt	Response	Description
REQ:	CHG	Change.
TYPE:	FTR	Features and Options.
CUST		Customer number
	0-99	Range for Large System and Avaya Communication Server 1000E (Avaya CS 1000E) system.
OPT	(HTU) HTR	Hot Line (unrestricted) or restricted. This program determines whether the call is going to a Hot Line DN or to any available DN. HTR restricts Hot Line calls to Hot Line DNs, but HTU does not.

Table 165: LD 18 - Prompt sequence

Prompt	Response	Description
REQ	COMP	Compute disk and memory.
TYPE	SCL	Speed Call lists.
NOLS	1-8191	Number of lists to be added.
DNSZ	4-16-31	Maximum length of DN allowed for Speed Call list.
SIZE	1-1000	Maximum number of DN entries in Speed Call list.

Table 166: LD 18 - Add or change a Hot Line Speed Call list.

Prompt	Response	Description
REQ	NEW CHG OUT	Add, change, or remove a Speed Call list.
TYPE	HTL	Hot Line List.
CUST	xx	Customer number, as defined in LD 15
LNSO	0-8190	Hot Line List number (only one Hot Line List per customer).
NCOS	0-99	NCOS to be assigned to calls accessing the list.
DNSZ	xx	Maximum number of digits in a list entry (4, 8, 12, 16, 20, 24, 28, or 31).
SIZE	1-1000	Maximum number of entries in the Speed Call list.
STOR	xxx yy...y	xxx = list entry number (0-9, 0-99, or 0-999). yy...y = digits to be stored against the entry (must be equal to or less than DNSZ).
- WRT	(YES) NO	Data (is) is not correct and list (can) cannot be updated.

Prompt	Response	Description
		The WRT prompt follows SIZE and STOR prompts asking for confirmation of the data just entered. If data is correct, enter YES or <CR>. A response of NO to WRT after SIZE returns the REQ prompt. A response of NO to WRT after STOR causes the data just entered to be ignored and a restart message (SCH3213) to be generated. A response of **** aborts the program. The last STOR value is lost but all other values for which WRT was YES are saved. The following information is output with the WRT prompt: ADDS: MEM: xxxxx DISK: yy.y (xxxxx is the amount of protected memory; yy.y is the number of disk records required for the new speed call list. Check the MEM AVAIL and DISK REC AVAIL values output before the REQ prompt).

Table 167: LD 10 - Add Enhanced Hot Line for analog (500 and 2500 type) telephones.

Prompt	Response	Description
REQ:	CHG	Change.
TYPE:	500	Telephone type.
TN		Terminal number
	l s c u	Format for Large System, Avaya MG 1000B, and CS 1000E system, where l = loop, s = shelf, c = card, u = unit.
CLS	DTN DIP	Digitone or dial pulse service (manual service is not allowed).
	EHTA (LDN) LNA (XFD) XFA (CWD) CWA (XRD) XRA	Enhanced Hot Line allowed. Last Number Redial (denied) allowed – optional. Call Transfer (denied) allowed – optional. Call Waiting (denied) allowed – optional. Ring Again (denied) allowed – optional.
FTR	HOT D nn x...x	Direct Hot Line DN. nn = number of digits (1-31) for target DN x...x.
	HOT L 0-999	Hot Line List entry number defined in LD 18.

Table 168: LD 11 - Allow or deny Enhanced Hot Line for Meridian 1 proprietary telephones.

Prompt	Response	Description
REQ:	CHG	Change.

Prompt	Response	Description
TYPE:	a...a	Telephone type. Type ? for a list of possible responses.
TN		Terminal number
	l s c u	Format for Large System, MG 1000B, and CS 1000E system, where l = loop, s = shelf, c = card, u = unit.
KEY	nn HOT D cc x...x nn HOT L aaa nn HOT D cc x...x xxxx nn HOT L aaa xxx...x nn CH D cc x...x	One-way Hot Line key. One-way Hot Line List key. Two-way Hot Line key. Two-way Hot Line List key. Combined No Hold Conference and Direct Hot Line feature
	nn CH L aaa	Combined No Hold Conference and Hot Line List feature nn = key number. cc = number of digits for target DN (1-31). x...x = target DN (up to 31 digits). aaa = Hot Line List entry defined in LD 18. xxx...x = DN for Hot Line key.

Feature operation

To make an EHOT call on an analog (500/2500 type) telephone:

- Lift the handset. The Hot Line number is automatically dialed.
- To transfer or conference an EHOT call on analog (500/2500 type) telephones:
- Flash the switchhook (or press Link) and dial the third-party extension.
- To make an EHOT call on a Meridian 1 proprietary telephone:
- Press Hotline.
- To answer an incoming Hot Line call on a Meridian 1 proprietary telephone:
- Press the flashing Hotline key.

To end an Enhanced Hot Line call:

- Hang up or press Rls.

Chapter 73: Hot Line, Flexible

Contents

This section contains information on the following topics:

Feature description

Flexible Hot Line (HOT) allows designated analog (500/2500 type) telephones to place calls to a predetermined destination simply by lifting the handset. The destination may be internal or external to the system, and the call does not require attendant intervention.

Flexible Hot Line (HOT) is provided to designated analog (500/2500 type) telephones on a Class of Service basis. A telephone is assigned the Hot Line feature through Service Change and a Manual Line (MNL) Class of Service. Address digits must be stored for the predetermined destination. If no digits are defined, the call will route to the attendant console.

When the user lifts the handset, no dial tone is returned. The system translates the stored digits and performs one of two operations:

- It rings an internal Directory Number (DN), then returns ringback tone.
- It translates to an external Trunk Access Code (TRC) and DN, then returns external call-progress tones or announcements.

Flashing the switchhook at any time during call setup or during the call will be ignored.

If the caller is a Hot Line, the prime Directory Number of the calling telephone is displayed on the terminating telephone, if equipped with a display.

Operating parameters

Flexible Hot Line applies to analog (500/2500 type) telephones only.

Feature interactions

Autodial

Flexible Hot Line and/or Enhanced Hot Line are mutually exclusive with the Autodial feature.

Calling Party Privacy

A Hot Line call will carry the Privacy Indicator if the Calling Party Privacy (CPP) code followed by the normal dialing sequence is stored in the Hot Line DN. The CPP will count against the maximum number of digits (currently 31) allowed for the Hot Line DN.

China - Flexible Feature Codes - Busy Number Redial Enhanced Flexible Feature Codes - Busy Number Redial

Busy Number Redial cannot be used on Flexible Hot Line sets.

Conference

A Flexible Hot Line (non-enhanced) telephone cannot place conference calls, but an Enhanced Hot Line telephone can activate the conference feature. If the Hot Line restriction option is set, the conference call can terminate only to other Hot Line telephones. If the restriction option is not set, the conference call can terminate to any type of telephone

Enhanced Hotline

Flexible Hotline and Enhanced Hotline are mutually exclusive; a telephone cannot have both Manual Line (MNL) and Enhanced Hot Line Allowed (EHTA) Classes of Service.

EuroISDN Continuation

Flexible Hotline does not support EuroISDN Continuation.

Flexible Feature Code Boss Secretarial Filtering

Flexible Feature Code Boss Secretarial Filtering takes precedence over Private Line and Hot Line.

Hunting

Calls will hunt before being routed to the attendant. Any Hot Line telephone can be assigned Hunting (excluding Short Hunt) Class of Service, but it applies only to the two-way Hot Line capability.

ISDN QSIG and EuroISDN Call Completion

Call Completion cannot be used in conjunction with the Hot Line feature.

Make Set Busy

Make Set Busy is overridden by the Hot Line feature. If a Meridian 1 proprietary telephone is in Make Set Busy mode, incoming Hot Line calls still terminate (ring) on the telephone.

Important:

The Conference-Hot Line key overrides Make Set Busy only when the terminating key is HOT.

No Hold Conference

The Conference-Hot Line key supports only one-way Hot Line calls.

On Hold on Loudspeaker

It is possible to program Hot Line with a loudspeaker DN, but operation will be the same as for direct dial to a loudspeaker DN.

Override

A Hot Line call can be entered using the Override feature.

Phantom Terminal Numbers

Hot Line does not support Phantom Terminal Numbers.

Private Line Service

A Hot Line key cannot be a Private Line, as this would defeat the benefits of Private Line service.

Room Status

The Room Status feature is incompatible with any telephone for which going off-hook activates Hot Line.

Speed Call, System

When the System Speed Call package is equipped, Hot Line lists have the characteristics and limitations of SSC lists. If the package is not equipped, Hot Line lists function like standard Speed Call lists.

User Selectable Call Redirection

An analog (500/2500 type) telephone with a Hot Line feature cannot use User Selectable Call Redirection, because it cannot access any features through SPRE or FFC.

Voice Call

The terminating DN of a Voice Call arrangement may be the incoming DN of a two-way Hot Line. When engineering call-modification paths (such as Hunting and Call Forward No Answer), the Hot Line Restriction option will cancel the normal call-modification operation for internal non-Hot Line calls.

Feature packaging

The Flexible Hot Line feature is contained in Enhanced Hot Line (HOT) package 70. There are no feature package dependencies.

Feature implementation

Table 169: LD 10 - Add or change Flexible Hot Line for analog (500 and 2500 type) telephones at the FTR prompt.

Prompt	Response	Description
REQ	CHG	Change.
TYPE	500	Telephone type.
TN		Terminal number
	l s c u	Format for Large System, Avaya CS 1000 Media Gateway 1000B (Avaya MG 1000B), and Avaya Communication Server 1000E (Avaya CS 1000E) system, where l = loop, s = shelf, c = card, u = unit.
CLS	MNL	Manual signaling – requires transfer denied (XFD) Class of Service.
FTR	HOT D 1-31 xxx...x yyy...y	Add Flexible Hot Line. 1-31 = maximum digits for Hot Line DNs. xxx...x = Flexible Hot Line DN. yyy...y = Phantom DN for a two-way Hot Line.

Feature operation

To make a Flexible Hot Line Call, follow these steps:

1. Lift the handset. The Hot Line number is automatically dialed.
2. To end the call, hang up.

Chapter 74: Hunting

Contents

This section contains information on the following topics:

Feature description

Hunting allows calls encountering a busy Directory Number (DN) to route automatically to another DN. Hunting continues along a predefined path, known as the hunt chain, until reaching an idle DN, the end of the hunt chain, or the maximum number of hunt steps. Hunting is specified on a DN basis. DNs in the hunt chain can be consecutive or nonconsecutive numbers.

The four types of hunt chains provided by the system are:

- Circular hunting
- Linear hunting
- Secretarial hunting
- Short hunting

The following pages describe and illustrate each of these ways to hunt.

Circular Hunting

Circular Hunting begins at the dialed DN and travels through every DN in the hunt group. The chain can begin at any point in the circle. The call goes around the circle until answered, or until returned to the initial DN. If all the DNs in the chain are busy, the caller hears busy tone. Figure 24: Example of Circular Hunting on page 500 shows an example of circular hunting.

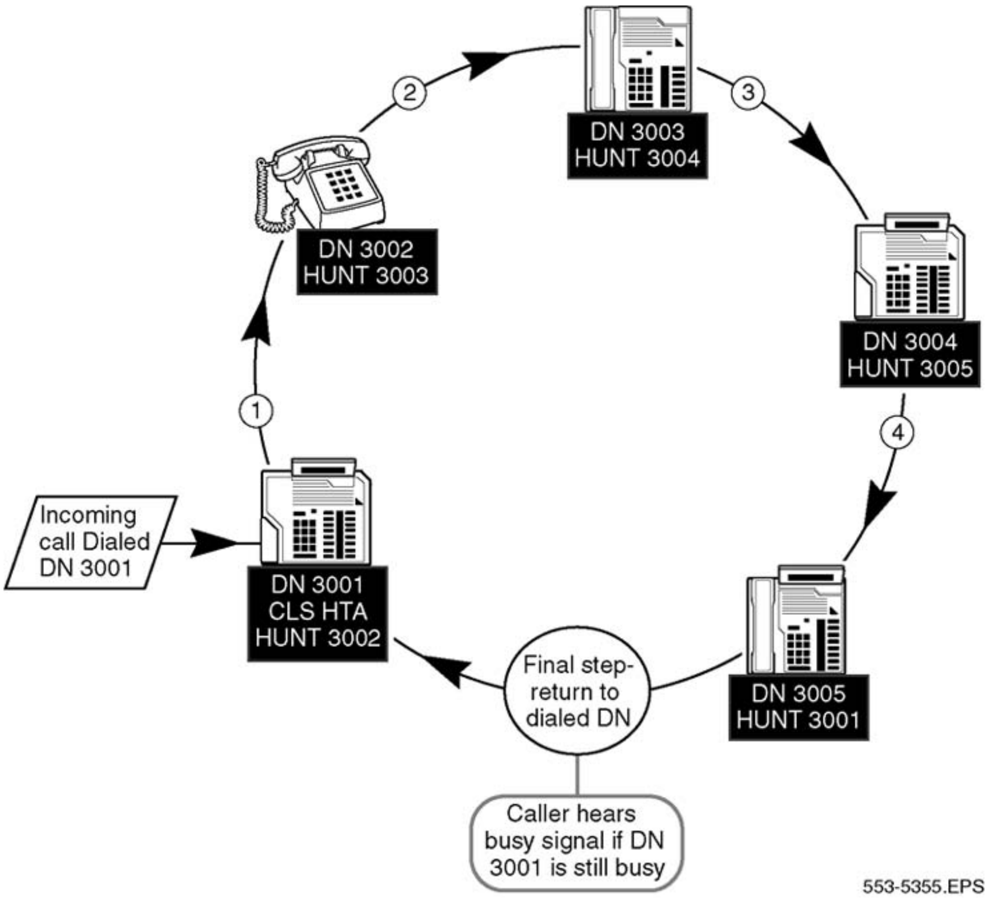


Figure 24: Example of Circular Hunting

Linear Hunting

Linear Hunting begins at the dialed DN. The call travels in one direction only when hunting along a linear chain. If a call comes into the second DN of a four-DN chain, it hunts to the third and fourth DNs only. If all the DNs are busy, the caller hears busy tone. Figure 25: Example of Linear Hunting on page 501 shows an example of Linear Hunting.

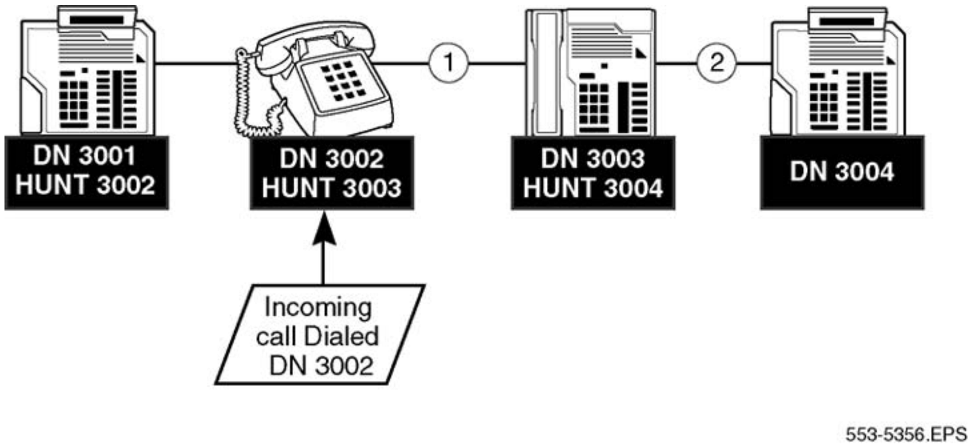


Figure 25: Example of Linear Hunting

Secretarial Hunting

Secretarial Hunting sends calls to a single Hunt DN, typically a secretary or Voice Mail. When a call comes in to a busy DN, it travels to the central location. Figure 26: Example of Secretarial Hunting on page 501 shows an example of Secretarial Hunting.

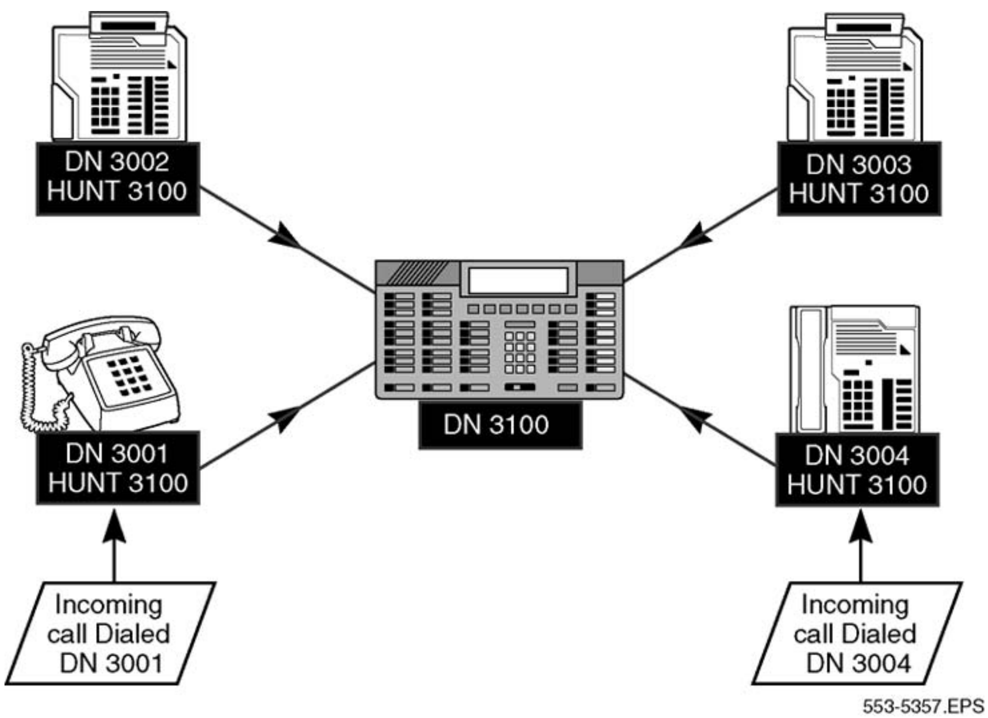


Figure 26: Example of Secretarial Hunting

Short Hunting

Short Hunting takes place along the key strip of any Meridian 1 proprietary telephone. The hunt chain begins on a DN on the key strip. The call hunts up the keys until it reaches a feature key, an unassigned key, or the Last Hunt Key (LHK, defined in LD 11). If the call cannot reach an available DN, the caller hears busy tone. When a call hunts to a Multiple Appearance DN, all appearances with ringing are allowed.

For a TN with Hunting Control enabled, Short Hunt takes precedence over normal Hunting (Circular, Linear, or Secretarial). If the Hunting search selects a TN for a digital telephone, Short Hunt redirects the call before attempting to use the Hunt TN. The hunt chain might become Hunt DN A, Hunt DN B, Short Hunt Sequence C, Short Hunt Sequence D, or Hunt DN E.

Figure 27: Example of Short Hunting on page 502 shows an example of Short Hunting.

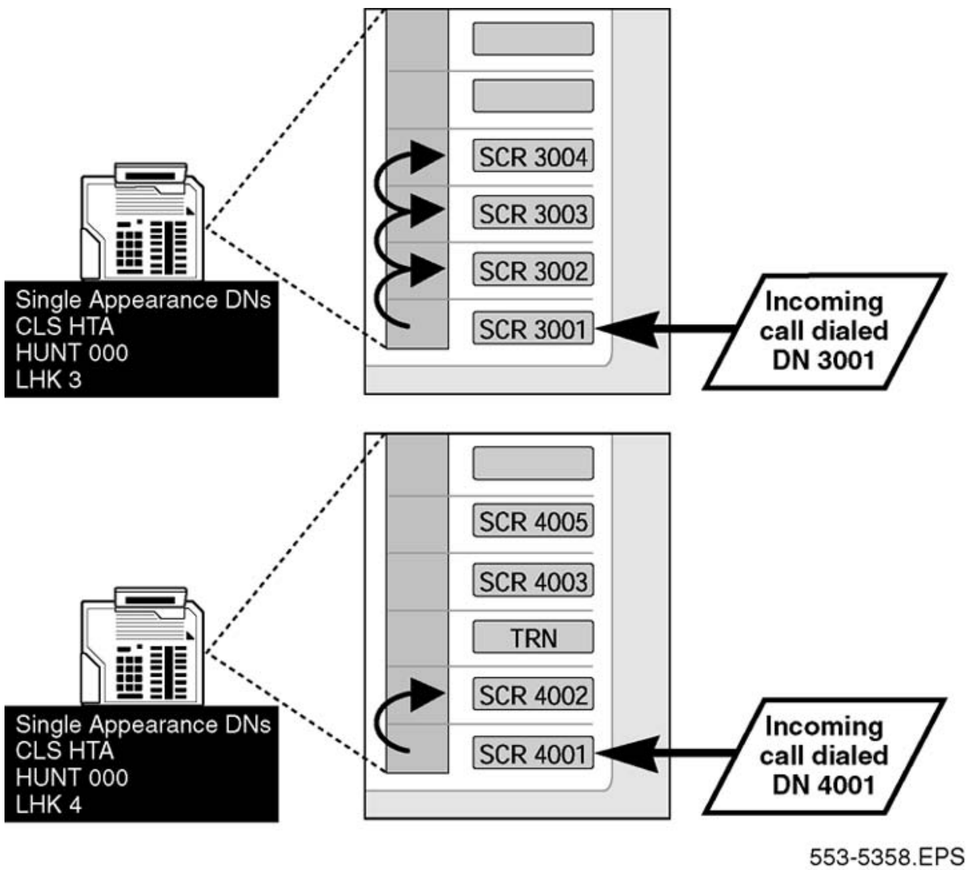


Figure 27: Example of Short Hunting

Operating parameters

There are no operating parameters associated with this feature

Feature interactions

Advice of Charge for EuroISDN

Calls charged with Advice of Charge that are either extended, transferred or redirected to another set using Hunting, are charged against the last station that answers the call and the controlling station releases.

Attendant Alternative Answering

Calls directed to a busy Attendant Alternative Answering (AAA) DN with Hunt defined are routed down the Hunt chain as defined for the AAA DN.

A Pilot DN for a hunting group can be defined as an AAA DN. Calls forwarded to a Pilot DN are directed to the next DN in the group.

Attendant Blocking of Directory Number

If Attendant Blocking of DN is attempted on a busy DN having the Hunting feature active, busy tone will be returned (overriding the Hunting feature).

Attendant Break-In

If the destination DN is in a Hunting chain with some idle DNs, the Break-In request goes to the first idle DN in the chain. To prevent this occurrence, the attendant can press the Break-In key prior to dialing the destination DN.

Attendant Busy Verify

Attendant Busy Verify does not affect Hunting.

Automatic Set Relocation

Calls will not hunt to a telephone that is being relocated.

Basic Rate Interface

Uses ISDN BRI Special Hunting. For information on the implementation of the Special Hunting feature, refer to the ISDN BRI Special Hunting feature description in *Avaya ISDN Basic Rate Interface Feature Fundamentals, NN43001-580*.

Call Detail Recording on Redirected Incoming Calls

The Call Detail Recording on Redirected Incoming Calls feature does not affect how the Hunting feature operates; however, it does provide information about the answering party in the CDR ID field if incoming calls have been redirected by any one of these features.

Call Forward All Calls Call Forward, Internal Calls

Call Forward All Calls and Internal Call Forward takes precedence over Hunting.

Call Forward Busy

Hunting takes precedence over Call Forward Busy for Direct Inward Dialing (DID) calls. When the station receiving a DID call has both Call Forward Busy and Hunting Allowed (HTA) Class of Service, the call is routed along the hunt chain. If all stations in the hunt chain are busy, the call is forwarded to the attendant.

Call Forward and Hunt Override Via Flexible Feature Code

A hunt can be overridden by the Call Forward/Hunt Override Via Flexible Feature Code feature, through the use of a Flexible Feature Code.

Call Forward No Answer

Hunting may result in the presentation of a call to a telephone that is different from the dialed DN. In this case, if the call is eligible for Flexible Call Forward No Answer, it is forwarded to the DN specified for the dialed DN, not the ringing DN.

Suppose that party A calls to busy party B, and party B has programmed Hunting to party C. Flexible Call Forward No Answer forwards a No Answer call at party C to the FDN associated with party B, the dialed DN.

After the call is forwarded, the MARP TN for the dialed DN controls the call redirection for Flexible Call Forward No Answer.

Call Forward No Answer, Second Level

A forwarded call may be modified by Hunting if the Call Forward No Answer DN is busy. This call is eligible for Second Level Call Forward No Answer if the SFA Class of Service is allowed and a Call Forward No Answer DN has been defined for the last rung DN.

If Group Hunting is active, Second Level CFNA is not applied.

Call Page Network Wide

Call Page Network Wide (PAGENET) does not block a station set from being programmed to Hunting to an external Paging trunk. At call termination time, calls that are forwarded to an external PAGENET uncontrolled trunk are not blocked. However, calls forwarded to an external PAGENET controlled trunk are given access denied intercept treatment at the Paging node.

Call Redirection by Time of Day

When Call Redirection by Time of Day (CRTOD) is enabled and an incoming call reaches a busy Directory Number, the time is checked against the Alternate Redirection Time Option range defined on the telephone.

Call Waiting Station-to-Station Call Waiting

If a call comes into a busy DN, it begins the hunting route defined from the called DN. If there are idle DNs on the hunting route, the call becomes a Call Waiting call on the called DN.

Hunting takes precedence over Call Waiting. If all steps in the hunt chain are busy, Call Waiting is activated.

Call Waiting Redirection

If Call Forward and Hunt by Call Type (CFCT) is enabled with Call Forward No Answer and Call Waiting Redirection, "no answer" internal calls receiving Call Waiting treatment are routed for CFNA treatment to the Flexible CFNA DN (FDN) or Hunt DN, and "no answer" external calls are routed for CFNA treatment to the External Flexible CFNA DN (EFD) or External Hunt DN (EHT).

Calling Party Privacy

When an incoming trunk call with the Privacy Indicator is forwarded, the Privacy Indicator will be tandemed to the far end to inhibit the display of the Calling Party Name or Number, provided that the tandem node also has Calling Party Privacy (CCP) provisioned.

If an incoming ISDN trunk call with the Privacy Indicator is forwarded, the Privacy Indicator will be tandemed to the far end to inhibit the display of the Calling Party Name or Number provided that the outgoing trunk route on the tandem node also has CCP provisioned.

If an incoming non-ISDN trunk call is forwarded to a trunk, the outgoing trunk call from the tandem node will carry the Privacy Indicator if the outgoing trunk route on the tandem node has the TCPP option set.

The CCP code can also be stored on the forwarding DN. If the CPP is requested on the forwarding DN, the Privacy Indicator will be outpulsed to the terminating node to inhibit the number of the forwarding set (that is, at the tandem node) from being displayed on the terminating set. In this scenario, the forwarding station must include the CPP in the forwarding DN (such as *67 + ACOD + the DN on the terminating node).

The above scenario also applies to Network Hunt.

Camp-On Camp-On, Station

Hunting takes precedence over Camp-On and Station Camp-On.

Capacity Expansion

If more than 16 appearances of the same Directory Number (DN) are configured, each hunt step is counted as two, to avoid running out of time slots.

China - Toll Call Loss Plan

Toll pad switching is also provided after call hunting has been completed. When the toll call is diverted, the diverted party's pad level is switched back to its original value (unless it is an OPS station using dynamic switching). The Toll Loss Plan is applied again for the new call as if it is a direct call. For Call Transfer, it is provided after the transferring party completes the transfer and drops out. For Call Forward or Hunting, it is provided when the forwarding or hunting call is answered.

Digital Private Signaling System 1 (DPNSS1) Executive Intrusion

If Executive Intrusion is attempted against an extension with a Hunt DN configured, an attempt will be made to reroute the call to the hunt DN provided the Hunt DN is on the same node. If the Hunt DN is busy, this rerouting process is repeated. If all DNs in the Hunt chain are busy, Executive Intrusion is attempted against the wanted extension originally dialed. Otherwise, the call will terminate as a simple call on the first idle extension in the Hunt chain.

Direct Inward Dialing Call Forward No Answer Timer

Hunting takes precedence over the Message Center feature.

Do Not Disturb

If activated, Hunting takes precedence over Do Not Disturb busy indication.

Flexible Feature Code Boss Secretarial Filtering

A boss set with filtering activated is passed over by Hunting; the next hunt sequence is to the secretary set.

Group Call Dial Access to Group Calls

Hunting cannot be applied to a Group Call.

Group Hunt

Group Hunting has priority over Hunting. If the DN attempted for termination by Group Hunting has HTA COS, and if it is busy, Group Hunting continues with the next DN in the group instead of following the DN's hunting configuration.

Hot Line

Any Hot Line telephone can be assigned Hunting (excluding Short Hunt) Class of Service, but it applies only to the two-way Hot Line capability.

ICP Network Screen Activation Flexible DN Meridian Mail

When a call redirected by Call Forward All Calls, Call Forward No Answer, Call Forward Busy, or Hunt terminates on an Intercept Computer (ICP) position, a redirected message identification "50" is sent to the ICP computer, when the call is answered.

Idle Extension Notification

If the attendant dials a busy extension that has Hunting configured and where all the DNs in the hunt chain are busy, Idle Extension Notification may be requested towards the dialed extension.

ISDN QSIG Name Display

When an incoming QSIG call with name display presentation allowed is hunted locally, the calling party's name information is displayed on the destination set. With presentation restriction, the calling party's name information is not displayed.

Lockout, DID Second Degree Busy, and MFE Signaling Treatments Multiple Appearance Directory Numbers

Hunting is controlled by the MADN Redirection Prime (MARP) Terminal Number (TN). If the MARP system option is disabled, Hunting proceeds as if MARP did not exist.

If all the telephones in the Multiple Appearance Directory Number (MADN) group are Meridian 1 proprietary telephones, ringing telephones are placed at the top of the DN list, and non-ringing telephones are placed at the bottom.

If a Multiple Appearance Directory Number appears in a group with several telephone types, the telephone type affects the position of the TN in the list. The analog (500/2500 type) telephones are listed at the top, and Meridian 1 proprietary telephones are listed in numerical TN order at the bottom of the list. A service change to an analog (500/2500 type) telephone moves its TN to the top of the list. A service change to a Meridian 1 proprietary telephone moves it to the bottom of the list. Call redirection follows the TN order from top to bottom.

The MARP TN is always checked to determine if and how the call is to be redirected by Hunting, regardless of where the MARP TN resides in the TN list of the DN block. No searching of the TN list of the DN block is needed. Hunting will follow the hunt chain based on the originally dialed DN. The actual functioning and requirements for Hunting are not changed by the MARP feature. The basic change introduced by the MARP feature is to always have a designated TN, the MARP TN, as the TN supplying the call redirection parameters.

If the MARP TN does not have Hunting control enabled, no Hunting is attempted. Other features for redirecting calls to busy DNs may be attempted based on the MARP TN.

A Short Hunting sequence begins when the MARP TN of a busy DN can perform Short Hunting. When a Short Hunt begins, it completes on that telephone before going to the Hunt DN. The precedence of Short Hunting over normal Hunting is maintained. Once a Short Hunting sequence is started on a digital TN, all the DNs in the Short Hunt sequence on that TN are attempted before redirecting the call to the TN's Hunt DN. Thus, a Hunt Chain connects Short Hunting sequences through Hunt DNs only.

Multiple Appearance Directory Number Redirection Prime

The Multiple Appearance Directory Number Redirection Prime (MARP) TN always controls the call redirection for Hunting. Short Hunting takes precedence over Hunting and MARP. The MARP TN is referred to until Short Hunting is encountered. Short Hunting is in control until it expires. When short hunting expires, the MARP TN for the first DN in the Short Hunt sequence takes control.

Network Individual Do Not Disturb Recovery on Misoperation of Attendant Console

Hunting takes precedence over the Network Individual Do Not Disturb and the Misoperation feature.

On Hold on Loudspeaker

Hunting to a loudspeaker DN can be programmed, but will receive intercept treatment as for direct dial to the loudspeaker DN.

Recorded Announcement for Calls Diverted to External Trunks

Recorded Announcement for Calls Diverted to External Trunks (RANX) is activated if the call is forwarded to an outgoing external CO trunk with the RANX feature active.

Recovery on Misoperation of Attendant Console

Hunting takes precedence over the Misoperation feature.

Ring Again on No Answer

If Ring Again on No Answer has been applied to a station going through a Hunt sequence, Ring Again is applied to that station and not the ringing station.

Total Redirection Count

Hunt redirections is limited to the value defined in the Total Redirection Count limit (if greater than 0). If this limit is exceeded, intercept treatment is given.

User Selectable Call Redirection

User Selectable Call Redirection permits a user to alter the HUNT DNs or EHT from a telephone.

Feature packaging

This feature is included in base system software.

Feature implementation

Task summary list

The following is a summary of the tasks in this section:

1. Table 170: LD 10 - Add or change Hunting for analog (500 and 2500 type) telephones. on page 512

   Add or change Hunting for analog (500/2500 type) telephones.
2. Table 171: LD 11 - Add or change Hunting for Meridian 1 proprietary telephones. on page 512

   Add or change Hunting for Meridian 1 proprietary telephones.

Table 170: LD 10 - Add or change Hunting for analog (500 and 2500 type) telephones.

Prompt	Response	Description
REQ:	CHG	Change.
TYPE:	500	Telephone type.
TN		Terminal number
	l s c u	Format for Large System, Avaya CS 1000 Media Gateway 1000B (Avaya MG 1000B), and Avaya Communication Server 1000E (Avaya CS 1000E) system, where l = loop, s = shelf, c = card, u = unit.
HUNT	xxx...x	Hunt DN. xxx...x removes the DN from the hunt chain.
CLS	(HTD) HTA	(Deny) allow hunting.

Table 171: LD 11 - Add or change Hunting for Meridian 1 proprietary telephones.

Prompt	Response	Description
REQ:	CHG	Change.
TYPE:	a...a	Telephone type. Type **?** for a list of possible responses.
TN		Terminal number
	l s c u	Format for Large System, Avaya MG 1000B, and CS 1000E system, where l = loop, s = shelf, c = card, u = unit.

Prompt	Response	Description
HUNT	xxx...x 000	Hunt DN. xxxx...x removes the DN from the hunting chain. Allow Short Hunting only.
LHK	xx	Last Hunt Key (LHK) number (default is 0). LHK 0 deactivates Short Hunt.
CLS	(HTD) HTA	(Deny) allow hunting.

Feature operation

No specific operating procedures are required to use this feature.

Chapter 75: Hunting by Call Type

Contents

This section contains information on the following topics:

Feature description

An additional Class of Service is provided for the system which will allow Direct Inward Dialing (DID) calls to hunt using the hunt chain when the dialed extension is busy, and the call's Classes of Service are Hunt by Call Type Deny (HTD) and Hunt by Call Type Allowed (HBTA).

The following rules apply to the call processing:

- If an extension is busy and its Class of Service is HTA, all types of calls to the extension will hunt using the hunt chain, regardless of HBTA/HBTD and FBA/FBD.
- If a busy extension's Class of Service includes HTD and HBTD, internal calls to the extension receive busy tone. Direct Inward Dialing (DID) calls to the extension which have Class of Service FBA are forwarded to the attendant. DID calls to the extension which have a Class of Service of FBD receive busy tone.
- If a busy extension's Class of Service include HTD and HBTA, internal calls to the extension receive busy tone. DID calls to the extension hunt using the hunt chain. If hunting fails, DID calls to the extension which have a Class of Service of FBA are forwarded to the attendant, and DID calls with a Class of Service of FBD receive busy tone.

Operating parameters

There are no operating parameters associated with this feature.

Feature interactions

Call Redirection by Time of Day

If Call Forward by Call Type (CFCT) is enabled with Call Forward No Answer (CFNA) and Call Redirection by Time of Day (CRTOD), unanswered internal calls receiving CFNA are routed to the Flexible CFNA DN, Hunt DN, Alternate Flexible CFNA DN or Alternate Hunt DNs. External calls are routed in the same manner.

If CFNA is enabled with Hunting by Call Type and Call Redirection by Time of Day (CRTOD), unanswered internal calls are redirected to the Hunt DN or Alternate Hunt DN during the alternative time. External calls are routed in the same manner. The alternate time is defined on the called DN's data block.

Feature packaging

This feature is packaged under International Supplementary Features (SUPP), package 131.

Feature implementation

Task summary list

The following is a summary of the tasks in this section:

1. Table 172: LD 10 - Create or modify the analog (500 and 2500 type) telephone data blocks to allow or deny Hunt by Call Type. on page 517

Create or modify the analog (500 and 2500 type) telephone data blocks to allow or deny Hunt by Call Type.

2. Table 173: LD 11 - Create or modify the Meridian 1 proprietary telephone data blocks to allow or deny Hunt by Call Type. on page 517

   Create or modify the Meridian 1 proprietary telephone data blocks to allow or deny Hunt by Call Type.

Table 172: LD 10 - Create or modify the analog (500 and 2500 type) telephone data blocks to allow or deny Hunt by Call Type.

Prompt	Response	Description
...		
CLS	(HBTD) HBTA	Hunt by Call Type (denied) allowed.

Table 173: LD 11 - Create or modify the Meridian 1 proprietary telephone data blocks to allow or deny Hunt by Call Type.

Prompt	Response	Description
CLS	(HBTD) HBTA	Hunt by Call Type (denied) allowed.

Feature operation

No specific operating procedures are required to use this feature.

Chapter 76: Hunting, Data Port

Contents

This section contains information on the following topics:

Feature description

Data Port Hunting improves the Hunting operation for data ports and modem pooling, and improves Ring Again operation for modem pooling.

Up to 255 data ports can be configured as trunks in data port trunk routes. In addition, the route can be programmed to step to another data port route if all members in the route are busy.

A data port serves as the interface between the system and a computer or other data communication device. A data port can be one of the following devices:

- Standalone Add-on Data Module (ADM) in auto-answer mode (no modem)
- Any modem that can recognize ringing and simulate off hook or on hook status
- Standalone ADM in auto-answer mode, connected to a modem
- Data Access Card (DAC), or
- Meridian Communication Adapter (MCA).

The following types of trunk routes are supported for data port hunting:

- ADM Trunk Routes: Add-on Data Module (ADM) data ports that interface through Data Line Cards
- Modem Trunk Routes: Modem data ports that interface through 500/2500 Line Cards
- RS-232 (R232): RS-232 data ports that interface through Data Access Cards (DACs)
- RS-422 (R422): RS-422 data ports that interface through Data Access Cards (DACs), and
- MCA: Meridian Communication Adapter (MCA) data ports that interface through Integrated Services Data Line Cards (ISDLCs) or Data Line Cards (DLCs).

Data ports act only as terminating parties. The user dials the access code of the trunk route to access the data ports.

Operating parameters

All data port trunks within a route must be of a single type. ADM and MDM data ports cannot be mixed in the same data port trunk route.

Only an attendant can extend incoming calls from stations or trunks (CO, FX, WATS, TIE, Direct Inward Dialing [DID], Common Controlled Switching Arrangement [CCSA]) to data port trunk routes. Calls cannot be extended, transferred, or conferenced from a station to a data port group.

In Night Service mode, any station can transfer incoming calls to data port routes.

Trunk access limitations (TARG, TGAR) should be applied to data port trunk routes to prevent stations with co-located ADMs from directly accessing data ports with modems, and vice versa.

Class of Service limitations do not apply to data port trunks.

Ring Again, Basic/Network Alternate Route Selection (BARS/NARS), and trunk access limitations (TARG, TGAR) are the only features that may be applied on calls to data port routes.

Feature interactions

Conference

There are no feature interactions associated with this feature.

Ring Again

When a user activates Ring Again against the data port extension Access Code (ACOI), the system stores the request until a member in the data port route becomes idle. When an idle member is found, the calling party is notified and the member is reserved for eight seconds. If the calling party does not respond to the Ring Again notification within eight seconds, the reservation is dropped.

Feature packaging

This feature is included in base system software.

Feature implementation

Task summary list

The following is a summary of the tasks in this section:

1. Table 174: LD 16 - Add or change a data port trunk route. on page 522

   Add or change a data port trunk route.

2. Table 175: LD 14 - Add or change a data port trunk. on page 523

   Add or change a data port trunk.

Table 174: LD 16 - Add or change a data port trunk route.

Prompt	Response	Description
REQ	NEW CHG	New, or change.
TYPE	RDB	Route Data Block.
CUST	xx	Customer number, as defined in LD 15
ROUT		Route number
	0-511	Range for Large System and Avaya Communication Server 1000E (Avaya CS 1000E) system.
TKTP	ADM MDM R232 R422 MMPM	Trunk route type
STEP	0-511	Alternate trunk route number
TARG	0-31	Trunk Access Restriction Groups (TARGs)
- TOV	0-3	Data port time out 0 = No timeout 1 = 15 minutes 2 = 30 minutes 3 = 60 minutes
- PSEL	(DMDM) TLNK	Protocol selection. T-Link or DM-DM protocol. Prompt offered to MCU (TKTP = MMPM).
- OPE	(NO) YES	(Do not) change data port operating parameters. Prompt offered to MCU (TKTP = MMPM).
- - PSDS	(NO) YES	(Do not) allow PSDS protocol. Prompt offered to MCU (TKTP = MMPM).
- - TRAN	(ASYN) SYN	Port transmission type; if PSDS = YES, then TRAN must be SYN. Prompt offered to MCU (TKTP = MMPM).
- - PAR	(SPAC) EVEN ODD MARK	Parity type, where: SPAC = space parity EVEN = even parity ODD = odd parity MARK = mark parity
- - DTR	(OFF) ON	Forced DTR (if ON) or dynamic DTR (if OFF). Prompt offered to R232, and to MCU (TKTP = MMPM).
- - DUP	(FULL) HALF	Full duplex/half duplex. Prompt offered to MCU (TKTP = MMPM).
- - DCD	(ON) OFF	(ON) = dynamic CD. OFF = forced CD. Prompt offered to R232, and to MCU (TKTP = MMPM).
- - MOD	(NO) YES	Modem, (Network): when TRAN = SYN. Prompt offered to MCU (TKTP = MMPM).
- - INT	(OFF) ON	SL-1/100 Interworking. Prompt offered to MCU (TKTP = MMPM).

Prompt	Response	Description
- - CLK	(OFF) ON	(OFF) = External Clock, ON = Internal, when TRAN = SYN. Prompt offered to MCU (TKTP = MMPM).
- - V25	(NO) YES	V.25 bis offered only when TRAN = SYN. Prompt offered to MCU (TKTP = MMPM).
- - HDLC	(NO) YES	High Level Data Link Control offered only when V25 = YES. Prompt offered to MCU (TKTP = MMPM).
- - DEM	(DCE) DTE	Data Equipment Mode. DCE or DTE mode. Prompt offered to R232.
- - PBDO	(OFF) ON	Port Busy upon DTR off. Presented when DCE, Dynamic DTR. Prompt offered to R232. ON = enabled (OFF) = disabled

Table 175: LD 14 - Add or change a data port trunk.

Prompt	Response	Description
REQ	NEW CHG	New or change.
TYPE	ADM MDM R232 R422 MMPM	Trunk type
TN		Terminal number
	l s c u	Format for Large System, Media Gateway 1000B, and CS 1000E system, where l = loop, s = shelf, c = card, u = unit.

Feature operation

To access a Data Unit (DU), the user dials the Access Code (ACOD) of the route data block. If a DU is available, a connection is made. If a DU is unavailable, the user receives this message on the terminal screen: "ALL PORTS ARE BUSY. ACTIVATE RING AGAIN?" Select Ring Again and wait until a DU port becomes available.

When a user dials a data port, the request is placed in the Ring Again queue until a port becomes idle. When an idle port is located, the calling party is notified and the port is reserved for eight seconds.

Data Port Verification (DVS)

Any applicable telephone with Data Port Verification Allowed (ADV) Class of Service can place a call to a specific Add-on Data Module (ADM) in a route by going off-hook, receiving dial tone, and dialing:

SPRE + 70 + ACOD + mmm

where:

SPRE	=	special prefix
70	=	special access code for the Data Port Verification (DVS) feature
ACOD	=	Access Code for the ADM trunk group, and
mmm	=	three-digit number that is to be seized within the trunk group.

The selected ADM trunk is seized if it is in not busy, maintenance busy, or disabled state. Once the call is established, it is treated as a normal ADM trunk call. If the selected trunk is in busy, maintenance busy, or disabled state, the call originator receives an overflow tone. No tone is returned when keyboard dialing is used.

Chapter 77: Hunting, Trunk

Contents

This section contains information on the following topics:

Feature description

Trunk Hunting provides either Linear Hunting or Round Robin Trunk Hunting for outgoing trunks in a route.

When Linear Hunting is implemented, the system searches for an available trunk in descending order. A station originating an outgoing call is connected to the last available trunk (highest available trunk route member number) of the trunk route accessed. The last trunk route member is always the first choice for outgoing calls and the first trunk route member is always the last choice.

Round Robin Trunk Hunting

Outgoing calls are evenly distributed among the members of a trunk route. When a station originates an outgoing call, the system searches for an available trunk route member in descending order, starting with the next lower member number from the last trunk seized for an outgoing call on the trunk route. If a trunk with a lower member number is not available, the system searches for a trunk starting with the highest member number of the route.

Note for multiple group machines using Round Robin Trunk Hunting:

To minimize system resource usage, the system will attempt to hunt to an available trunk within the same group as the originating TN. For example, if a call is placed from a telephone whose TN is in group 1, the system will first attempt to locate an available trunk within group 1. If there are no available trunks in group 1, the system selects an available trunk from another group.

Each time hunting occurs, the round robin index value, which points to the next route member to be examined, is updated. Because the proximity of a trunk loop to the originating TN loop takes precedence over the order of the trunk route members, the system may be forced to hunt through many route members to locate an available trunk within a given group. This can cause the round robin index to change dramatically, yielding inconsistent trunk usage patterns.

If uniform trunk usage is a prime concern, configure route members with alternating groups. For example, if a given route contains trunk members from different groups, alternate the groups so that route member 1 is a trunk member from group 1, route member 2 is a trunk member from group 2, and so on. This configuration will produce more uniform trunk usage than would occur if trunks of the same group were bunched together within a route.

Operating parameters

The Public Exchange/Central Office (CO) governs incoming trunk hunting. The system has no control over the order of incoming trunks.

Feature interactions

There are no feature interactions associated with this feature.

Feature packaging

This feature is included in base system software.

Feature implementation

Table 176: LD 16 - Implement Linear or Round Robin Trunk Hunting for a trunk route.

Prompt	Response	Description
REQ	NEW CHG	New, or change.
TYPE	RDB	Route data block.
CUST	xx	Customer number, as defined in LD 15
ROUT		Route number
	0-511	Range for Large System and Avaya Communication Server 1000E (Avaya CS 1000E) system.
SRCH	(LIN) RRB	Linear or Round Robin Hunting.

Feature operation

No specific operating procedures are required to use this feature.

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