
Meridian 1

Option 11C

2.0 Mb DTI/PRI Administration and Maintenance guide

Book one of two

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About this guide

This guide describes Integrated Services Digital Network (ISDN) for the Option 11C. It is written for Option 11 ISDN planning, installation, and maintenance staff. It provides an overview of ISDN features and interfaces, including network clocking and hardware and software installation.

Note: This document is intended to support the Meridian 1 Option 11. When only the term “Meridian 1” is referenced, it is meant to include all members of the Meridian 1 family (Meridian 1 Option 11 through Option 81C).

ISDN is comprised of two interfaces:

- Primary Rate Interface (PRI)
- Basic Rate Interface (BRI)

This guide describes Primary Rate Access (PRA) only, or Primary Rate Interface (PRI), as it is known in Northern Telecom. ISDN Basic Rate Interface is described in *ISDN BRI Administration and Maintenance* 553-3011-311.

In addition to ISDN Primary Rate Interface, this guide describes the following:

- Digital Trunk Interface (DTI)
- ISDN Signaling Link (ISL)

- Integrated Digital Access (IDA) which is used in the United Kingdom and includes:
 - Digital Private Network Signalling System (DPNSS1)
 - Digital Access Signaling Scheme (DASS2)
 - Analogue Private Network Signaling System (APNSS)

Chapter 1 — ISDN and DTI Interfaces

Overview

Integrated Services Digital Network (ISDN) is a global networking strategy that provides a single, intelligent, universal network for voice and data communications. ISDN is a set of international standards that define the next generation of digital networking services.

Some of the benefits of ISDN are as follows:

- faster, more accurate, noise-free transmission, as well as greater security and reliability
- ability to carry multiple information types, including speech, data, image, and video
- standard interfaces to allow multi-vendor compatibility
- configuration flexibility when combining public and private networking facilities.

ISDN differs from traditional communication in that it uses “out-of-band” signaling. Signals are sent and received on a dedicated D-channel, leaving the remaining channels, called B-channels, free for voice and data.

Signaling and trunks

The terms *signaling* and *trunks* are used throughout this document. The following paragraphs briefly explain what they mean.

Trunks

Trunks are transmission channels that carry voice, data and signaling information between switching systems in the network. They can be classified based on two parameters:

- 1 the type of service supported on the trunk, for example Wide Area Telephone Service (WATS), Direct Inward Dial (DID), Foreign Exchange (FX) or tie trunks.
- 2 the nature of the signal that is transmitted, either analogue or digital.

Analogue trunks are provided on voice-frequency (VF) facilities that are operated in either 2-wire or 4-wire mode. They support speech or voice-grade data, both of which require a bandwidth of about 3.1 kilohertz (kHz).

Digital trunks are 64 kbps data channels that carry voice, data, or signaling information.

Signaling

Signaling is the process of transferring information between two parts of a network in order to control call connections and related operations. Some examples of signaling information are call set-up messages, call take-down messages, calling-line identification, and feature-activation information.

Prior to ISDN and common channel signaling, each trunk carried its own signaling together with the voice or data portion of the call. This scheme is referred to as per-trunk, or inband signaling.

With ISDN and out-of-band signaling, signaling is performed on a single D-channel. This leaves the other channels, called B-channels, clear for voice and data transport, and maximizes the efficiency of the carrier.

Network interfaces

A network interface is physically composed of a collection of trunks, each of which carries signaling and/or user information.

In a Meridian Network, four basic elements must be in place and functioning in order to support the networking services desired. These are:

- network interfaces
- network numbering plan
- network route selection
- network control

Network interfaces physically and logically connect the switches in a network. One of the initial steps in building a network, therefore, is to define and implement the types of interfaces that are required.

The prime focus of this document is the implementation, on an Option 11 system, of six types of network interfaces:

- 2 Mb Digital Trunk Interface (2 Mb DTI) that supports the E-1 carrier
- 2 Mb Primary Rate Interface (2 Mb PRI), a CCITT-specified interface that supports E-1 carrier
- ISDN Signaling Link (ISL), a Northern Telecom proprietary interface that supports analogue and/or digital trunks
- ISDN Signaling Link (ISL) to Analogue Private Network Signaling Scheme (APNSS)
- Digital Private Network Signaling System (DPNSS1)
- Digital Access Signaling Scheme (DASS2) to Q.931

The implementation of each of the above-named interfaces is described in this document.

Digital Trunk Interface (DTI)

Digital Trunk Interface (DTI) pre-dates ISDN and Primary Interface (PRI). DTI is a networking scheme that connects an Option 11 to another Private Branch Exchanges (PBXs) or central office (CO). Digital carriers (E-1s) are used to carry voice and data information instead of using dedicated trunks.

Thirty B-channels are available on the 2 Mb DTI for voice and data transmission. Each B-channel is assigned a 2 Kbps signaling channel within the CEPT multiframe, which consists of 16 frames. This networking arrangement, like PRI, makes use of a digital carrier instead of using dedicated trunks. In DTI, “A” and “B” bits are used to imitate traditional trunk signaling (ground start, loop start, and so on).

Table 1 shows the impact that out-of-band (PRI) signaling versus in-band (DTI) signaling has on data transmission.

Table 1
DTI and PRI data transmission

Data Transmission Types	DTI	PRI
Asynchronous	up to 19.2 kbps	up to 19.2 kbps
Synchronous	up to 64 kbps	up to 64 kbps

ISDN interfaces

Meridian 1 Networking supports three ISDN interfaces:

- Primary Rate Interface (PRI): 30B+D
- Primary Rate Interface (PRI): nB+D
- ISDN Signaling Link (ISL): nB+D
- 2B+D Basic Rate Interface (2B+D BRI)

Note: As noted earlier, this document does not contain information on ISDN BRI - this information is contained in the *ISDN BRI Administration and Maintenance* 553-3011-311.

Primary Rate Interface (PRI)

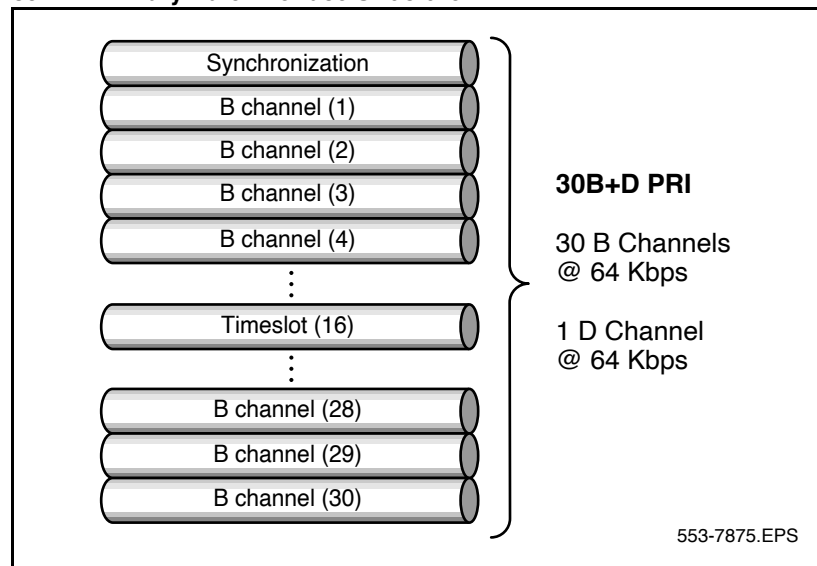
Primary Rate Interface is characterized by digital carriers and out-of-band signaling. All signaling is done over a D-channel, leaving the remaining B-channels free for voice and data transmission. The D-channel carries all the call set-up, call tear-down, calling line identification, and feature activation information.

PRI is referred to as either “30B+D” or “nB+D”. This distinction is based on the fact that one D-channel can support multiple digital carriers. In other words, the choice exists to have one D-channel per digital carrier, or to have one D-channel support a number of carriers.

30B+D Primary Rate Interface

In the 30B+D PRI arrangement, each digital carrier (E-1) has its own D-channel. There are 32 timeslots per E-1 carrier: timeslot zero is reserved for framing information, timeslot 16 is used as the D-channel, leaving 30 channels for voice and data. The B-channels are “clear” channels, meaning all 64 K are used for voice and data. The 30 B channels plus one D-channel structure conforms to the CCITT specification (I.431) for PRI layer 1 protocol.

Figure 1
30B+D Primary Rate Interface Structure



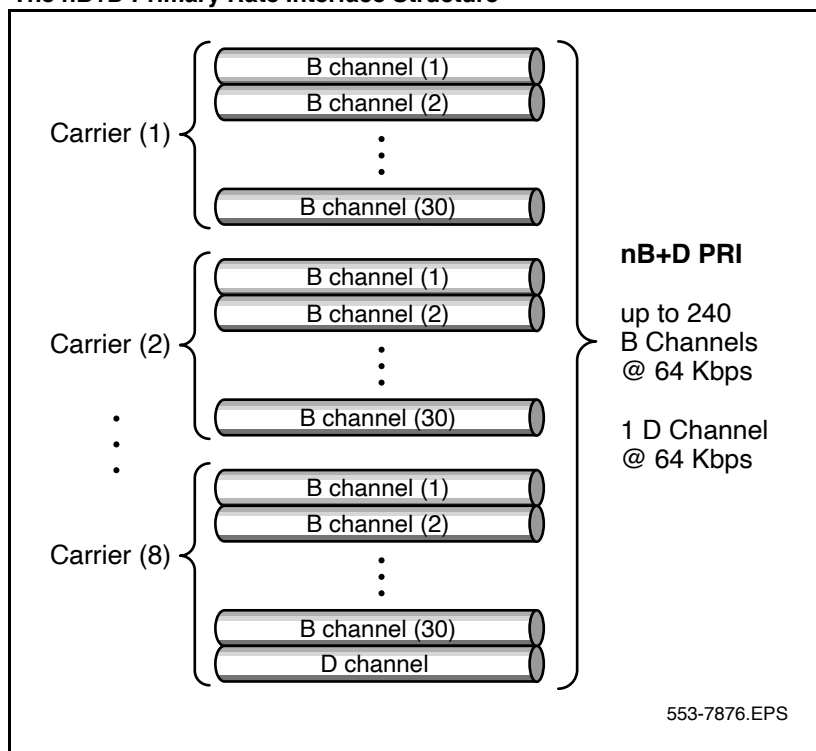
nB+D Primary Rate Interface

In an nB+D PRI arrangement, one D-channel supports multiple digital carriers. A back-up D-channel is usually recommended when one D-channel is supporting a large number of channels.

CCITT currently specifies the PRI layer 1 protocol as a 30 + B structure. Layer 3 (Q.931), however, supports a larger PRI made up of multiple digital carrier facilities.

On the Option 11 system, a maximum of eight NTAK79 or NTBK50 2 Mb PRI cards can be installed in the main cabinet. The maximum number of B-channels is 240 (8 X 30).

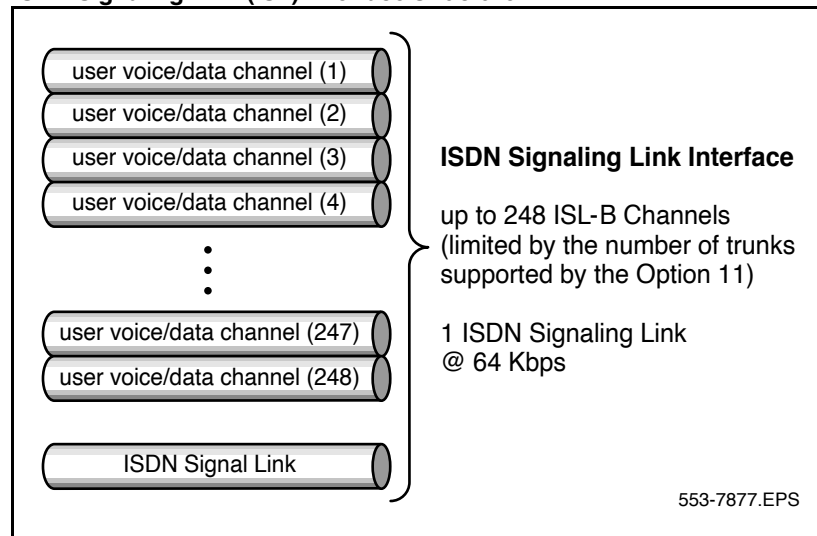
Figure 2
The nB+D Primary Rate Interface Structure



ISDN Signaling Link Interface (ISL)

ISDN Signaling Link allows Meridian 1 systems with digital and analog trunks to replace in-band signaling with out-of-band signaling. Analog or digital tie trunks are used: one trunk is designated as the D-channel while the remaining trunks act as B-channels.

Figure 3
ISDN Signaling Link (ISL) interface structure



D-channel

The D-channel can be a separate synchronous data circuit established over a leased line, or it can be an existing trunk circuit connected to standard communication equipment (modems or Meridian 1 data adapters).

The former is referred to as shared mode of operation, while the latter is known as dedicated mode.

Reverting to conventional signaling

If the ISL D-channel fails, trunk operation reverts back to in-band signaling. This is known as reverting to conventional signaling. A backup D-Channel can also be used, in which case “reverting to conventional signaling” would take place only when both D-Channels are out-of-service.

Possible ISL network configurations

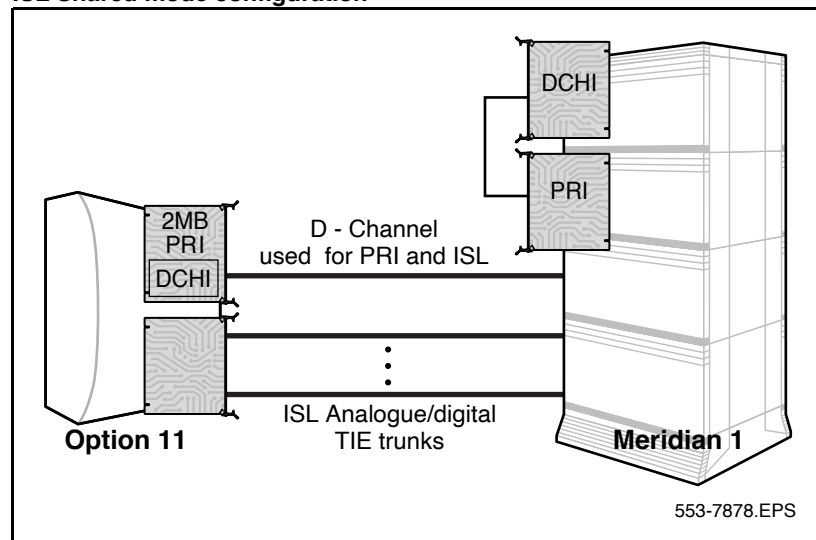
ISL is supported for Meridian 1/SL-1 to Meridian 1/SL-1 connections only.

Shared mode of operation

Shared mode requires PRI between two Meridian 1/ SL-1 switches. The PRI can be either a 30B+D or an nB+D arrangement.

One of the analog or digital Tie trunks is used as a D-channel and is shared between the two switches. The shared D-channel transmits signaling information for the non-ISDN Tie trunks.

Figure 4
ISL Shared mode configuration

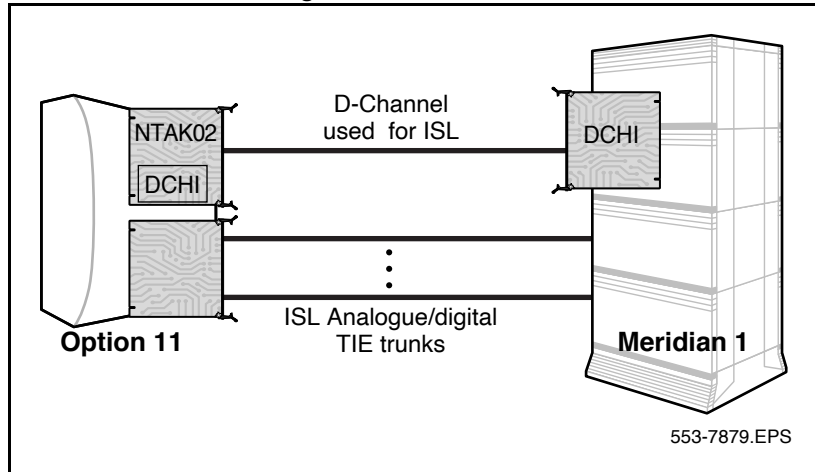


Dedicated mode

Dedicated mode is used when no PRI exists between the two Meridian 1/SL-1 switches, or when it is not desirable to share a D-channel.

A dedicated signaling (ISL) link is established and used to transport the signaling information for the non-ISDN trunks. The D-Channel can communicate with the far end through of a pair of dedicated line modems, dial-up modems, or Meridian 1 data adapters.

Figure 5
ISL Dedicated mode configuration



Integrated Digital Access (IDA)

IDA is used only in the United Kingdom and includes the following networking schemes:

- digital private network signaling system (DPNSS1)
- analogue private network signaling system (APNSS)
- digital access signaling scheme (DASS2)

Note: The NTBK50 2.0 Mb PRI card will not support IDA if the NTBK51 DDCH daughterboard is installed on the card. IDA is supported, however, if the NTAK93 DCHI daughterboard is used instead.

United Kingdom networking schemes require unique interfaces for compatibility with ISDN services. The APNSS, DPNSS1 and DASS2 gateway supports the following functions:

- basic call service (circuit switch voice calls)
- call line identification
- connected line identification
- call division
- coordinated dialing plan
- network ring again, between DPNSS1 and Q.931 protocols

Note: The NTBK50 2.0 Mb PRI card will support IDA only if the NTAK93 DCHI daughterboard is equipped. The NTBK51 Downloadable D-channel daughterboard does not support IDA.

DPNSS1 and DASS2

DPNSS1 and DASS2 are digital interfaces to UK ISDN networks. They conform to CCITT specification Q.931, but always use the 30B+D configuration. (Within the terminating exchanges, it is not necessary to configure all 30 B-channels for use).

DPNSS1 is used in private networks between PBX systems, while DASS2 is used between a PBX and a PSTN (Public Switched Telephone Network).

APNSS

APNSS is an analogue networking scheme that uses E&M trunk circuits on Meridian 1. The quantity of channels varies depending on traffic requirements. B-channels can be carried on a 2 Mb Digital Trunk Interface (DTI) or on a DPNSS1 B-channel in special cases.

The D-channel is normally established using modem equipment and a leased line, but it can be a digital channel in a suitable carrying medium. The D-channel is capable of running at any speed and can be over a digital or analogue link. If the D-channel is carried over a 64 Kbit digital link, a Data Line card must be provided for this function.

An analogue D-channel can also be established using leased line modems and connected by using dial-up modems, a 500 line card and any trunk circuit.

DPNSS1 and APNSS virtual channels

When using DPNSS1 or APNSS, and certain information is passed between two nodes that is not directly related to a call on a B-channel, a B-channel reference is required. An example of this is during the set up of the Call Back When Free feature.

The channel referenced in the message is referred to as a Virtual Channel. Virtual channels for DPNSS1 are the same as real B-channels, and have no effect on normal calls over B-channels.

Virtual channels for APNSS are programmed on an unused loop within the Option 11 system.

Channel configuration

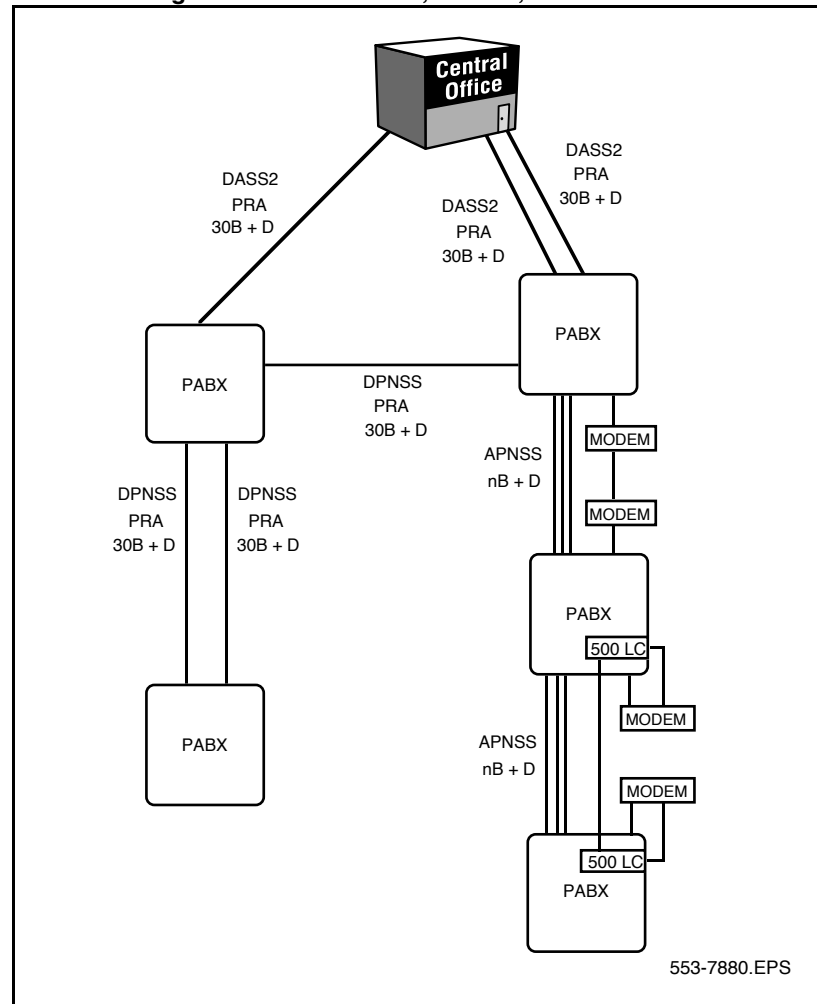
DPNSS1 and DASS2 are message-based signaling systems which use a common signaling channel in timeslot 16. Each traffic channel has an associated Link Access Protocol (LAP). The LAPs operate in parallel over the signaling channel. Each traffic channel and its LAP represent one trunk.

A D-channel handler interface is informed of the configuration of its LAPs each time it is enabled. If a discrepancy between the hardware and software configurations is detected during call processing, the software attempts to correct the configuration of the DCHI.

UK gateway limitations

The UK gateway to ISDN is limited in that Public Switched Telephone Network (PSTN) incoming trunks are not allowed access to PSTN outgoing trunks.

Figure 6
Channel configurations for DPNSS1, APNSS, DASS2



Chapter 2 — ISDN feature descriptions

Overview

ISDN allows us to create an intelligent network based on its out-of-band signaling and intelligent messaging. It makes possible the introduction of additional features, and provides network transparency to many of the existing features.

For example, features which were previously limited to internal Meridian 1 use, such as Call Redirection (Call Forward No Answer, Hunt), are now provided across the entire ISDN network. Features as Call Party Name Display and Calling Line Identification are now network wide features and can be sent to and received from other Meridian 1 systems and central offices.

The Option 11 system uses a series of software cartridges, each providing a number of features. Depending on your particular market, the features offered on any of the cartridges may vary. This chapter outlines some useful features including descriptions, interactions and limitations, and package requirements, as appropriate.

For reference purposes, a full list of feature packages, including their names, mnemonic and dependencies is provided at the end of this chapter. Also refer to the “ISDN Feature Implementation” chapter of this document for programming information on selected ISDN features.

510 Trunk Route Member Expansion

The 510 Trunk Route Member Expansion feature allows a customer to configure a maximum of 510 trunk route members per route (range 1-510). The previous maximum was 254 (range 1-254).

The need for this expansion was particularly evident for a Meridian 1 to DMS Central Office connectivity, where the full potential of an E-1 nB+D configuration (480 B-Channels) was not able to be realized. The DMS supports one D-Channel per route. With a limit of 254 route members per D-Channel, this meant that the maximum offered by nB+D could not be utilized.

With the 510 Trunk Route Member Expansion feature implemented, when connecting a Meridian 1 to another Meridian 1, or a Meridian 1 to a DMS Central Office PBX, users now have greater flexibility when configuring their systems.

For a Meridian 1 to Meridian 1 connectivity, a user is now able to configure, on one trunk route, one D-Channel with 480B-Channels, a Backup D-Channel, and a second D-Channel with another 30 B-Channels. If desired, another Backup D-Channel may be configured. Refer to Figure 7. For a Meridian 1 to DMS connectivity, a user may now configure the full complement of nB+D, which is one D-Channel with 480 B-Channels, and a Backup D-Channel. Refer to Figure 8.

This feature applies to the following trunk mode configurations:

- PRI2 mode
- DTI2 mode
- Shared mode
- Virtual Network Services (VNS)
- ISL mode
- Analog mode
- ISDN Basic Rate Interface (BRI) trunk mode

The following trunk types are supported:

- COT (Central Office)
- TIE

- DID (Direct Inward Dial)
- FEX (Foreign Exchange)
- FGDT (Feature Group D)
- WATS (Wide Area Telecommunications Service)

The overlay programs that are affected are:

- LD 14, the Trunk Data Block
- LD, 16 the Route Data Block
- LD 27, for ISDN BRI trunking

Maintenance routines are also affected. For the Call Trace LD 80, the TRAC c r m (trace calls, customer c, member m) is modified to allow “member” input from 1 to 510.

For the Automatic Trunk Maintenance LD 92, the ATMR c r m (Test customer c, route r, member m) is modified to allow “member” input from 1 to 510.

For traffic measurements, in the Customer Traffic Measurement Output report TFC002, the total number of trunks for the “Trunks Equipped” and “Trunks Working” fields has been increased to a maximum of 510.

In the CDR output format, the Originating Identification (ORIGID) and Terminating Identification (TERID) fields now displays the trunk route member field in the range 1-510.

Figure 7
510 trunk members expansion configured for a Meridian 1 to Meridian 1 connectivity

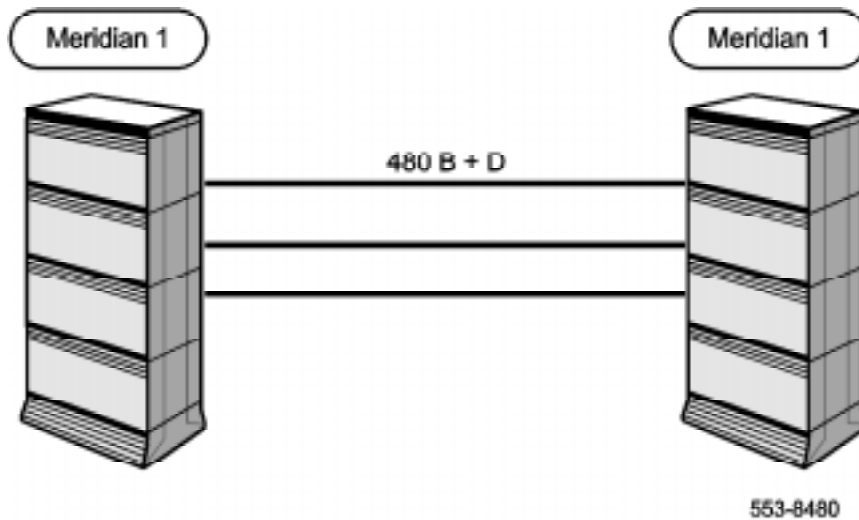
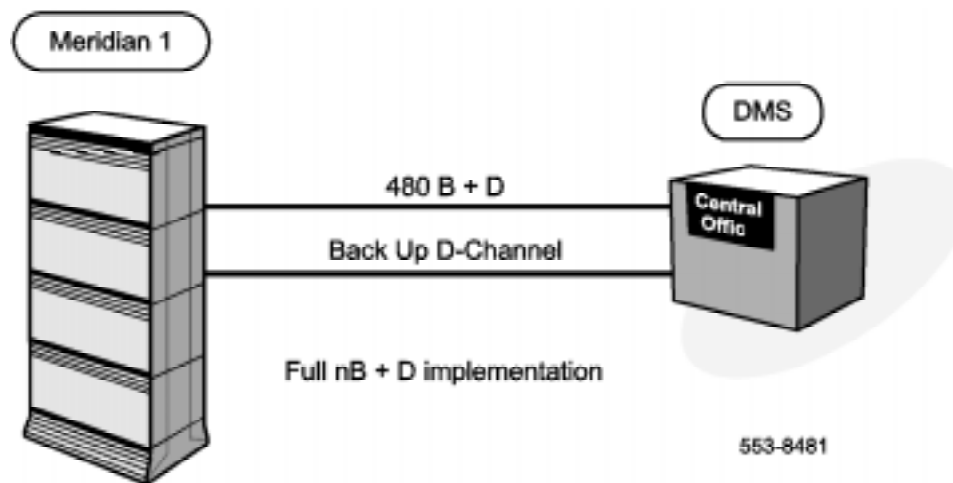


Figure 8
510 trunk members expansion configured for a Meridian 1 to DMS connectivity



Operating parameters

There are no operating parameters for this feature.

Feature interactions

Preference Trunk Usage

The Preference Trunk Usage Threshold (PTUT) in LD 16 may be configured at a value from 0-510.

Meridian MAX

The Meridian MAX configuration does not support the 510 trunk member expansion. The Meridian 1 system software maps all trunk members which are greater than 254 (from 254-510) to trunk number 0, in the High Speed Link protocol.

For the Meridian MAX report, the MAX software collects all call statistics associated with trunk number 255 to trunk number 510, and displays the pegged value in the trunk number 0 field.

Feature packaging

The following package is required:

- Basic (BASIC) package 0

Feature operation

No specific operating procedures are required to use this feature.

Advice of Charge

This section describes the Advice of Charge feature as it applies to the interface types supported by ISDN PRI.

Introduction

Prior to the introduction of the Advice of Charge supplementary service, the cost of calls to Central Offices in most European markets was calculated by counting and recording the periodic pulses provided by the Central Office during a call, and then calculating the cost of each call. The collection and storage of these charging pulses from the Central Office is referred to as Periodic Pulse Metering (PPM).

However, with the deployment of ISDN PRI in certain European public exchanges, PPM is no longer offered in its traditional form. Metering of calls for these interfaces is offered through the supplementary service called Advice of Charge (AOC)

Depending on the Central Office interface, AOC may be offered:

- at the end of a call;
- during a call; **or**
- at call set-up, during a call, and at the end of call.

AOC is summarized below for the following connectivities:

- AXE-10 Australia (at end of call)
- EuroISDN (at call set-up, during a call, end-of-call)
- INS1500 (Japan D70) (at end of call)
- Numeris (at end of call)
- Numeris (during a call)
- SwissNet (during a call)
- 1TR6 (at end of call)

Storing charging information

Charging information is stored using meters, which may be assigned to sets, attendant consoles, trunk routes, and customers as follows:

- one meter is assigned to each set, via Metered Assigned (MRA) Class of Service, on a TN basis. The charge unit count is stored in the meter of the station TN from which the call is made;
- one meter, classified as “ATTN”, is provided to collect charges for all metered calls made by attendant consoles within the same customer;
- one meter, classified as “ROUTE”, is provided for each route;
- at each node, one meter classified as “CUST” is provided for each customer to accumulate any charges that cannot be charged to any other meter for the customer.
- one temporary meter, classified as TRK, is provided for each trunk; this temporary meter is used to accumulate charges for each call. At the end of the call, the charge is added to the permanent meter associated with the caller.

Note: The same station meters are used to store charging units for PPM and AOC. To handle a possible difference in unit costs between PPM and AOC, charge units received for an AOC metered call may be converted into internal charge units using a “Route Unit Conversion Factor” (RUCF) configured in Overlay 16.

The charging information may also be recorded in the Call Detail Recording (CDR) record, as provided by the CDR Enhancement capability (for more information on the CDR Enhancement, refer to *X11 Features and Services*). This information is printed at the end of call, and appears in the same location of the CDR ticket as the PPM call charge information would appear, immediately following the PPM counts.

Reading and changing meters

Meters of Multiple Appearance DN (MADNs) can be changed using the SET METER DN or SET METER TN command at a Background Terminal TTY, or using the MTR key on a Meridian 1 telephone. To change only one of the meters in a MAPDN group, only the Background Terminal TTY can be used. For Automatic Call Distribution (ACD) stations, the meters can only be changed using the POSN-ID prompt at the TTY.

Displaying charging information

Charging information may be displayed on display-equipped Meridian Modular terminals, M2317, and M3000 sets. This display capability is provided by the Charge Display at End of Call feature. This feature appends the charge information to existing information on the display and retains the information displayed for 10 seconds. Depending on the CO interface, the information may be displayed at call set-up, during a call, and/or at end of a call (for more information on the Charge Display at End of Call feature, refer to *X11 Features and Services*).

Printing charging information

The charging information may be printed on a Background Terminal TTY.

Printing charging information on the TTY

Groups of consecutive DNs or a group of TNs can be printed, as well as all meters assigned to a Tenant or Tenants. To print only the sum of all meters, if a range of meters are selected, the option SUM is entered at the TTY. If the Hotel/Motel option is not selected, then the ROOM and ADMN prompts are suppressed in the printout.

For Multiple Appearance PDNs (MADNs), only the sum of all appearances is printed.

For Automatic Call Distribution (ACD) stations, the meters can only be printed using the POSN-ID prompt. The keyword ACD is shown in the meter count printout.

The contents of terminal meters can be automatically or manually printed. Automatic printing is done by establishing a daily, weekly, or monthly schedule. The call charge units in each terminal meter are printed on the TTY according to the schedule. With automatic printing, the terminal meters may be cleared after printing.

Manual printing allows a customer to request the printing of the contents of any meter or block or meters, as required. After printing, the meters may be cleared on command (it is not done automatically, as in automatic printing).

Handling overflow charging information

The charging information meters can store a maximum of 32,767 Periodic Pulse units. If this total is exceeded, an overflow counter is incremented. Whenever this counter reaches three, a message is output to the background terminal and the counter is reset to zero.

The charge field of the CDR record has a maximum value of 32767. If this value is overflowed, a charge overflow indication (“CHXOVF”) is printed in place of the charge count field.

Note: When the meter overflows, the meter can still have stored charge information equal to less than the maximum. Therefore, it may sometimes appear to a user that the charge is small for the length of time spent on the telephone. Therefore, it is important to have a background terminal in order not lose all of the charge information related to a call. To get the total charge for a call, it may be necessary to add the information in the CDR record to the background terminal overflow message(s).

Operating parameters

The following operating parameters apply to the Advice of Charge service, for the AXE-10 Australia, INS1500 (Japan D70), Numeris, SwissNet, EuroISDN, and ITR6 Central Office interfaces.

- There are limitations with respect to the size of the charge that can be displayed in either the CDR record or on the set display. In addition, currency symbols and decimal places cannot be displayed on sets or CDR records.
- Central Offices can send the charge with an accuracy of up to one decimal place. Smaller amounts are rounded up to the nearest integer, and thus rounding errors may have a minor affect on charging accuracy.
- The Advice of Charge information is not transported between a Meridian 1 to Meridian 1 connection. Therefore, if a tandem call is made from one Meridian 1 PBX to another Meridian 1 PBX, and then to a Central Office, the originator’s display is not updated with the charge information. However, the charge is printed in the CDR record at the switch that made the connection to the Central Office.

- Charging information received on incoming calls, as may happen for reverse charging, is not supported. The existing error indication is given if an incoming call is received. This does not happen in Australia because the AOC information must be requested for each call and the Meridian 1 does not support requesting AOC information for incoming calls.

Feature interactions

The following feature interactions apply to the Advice of Charge service, for the AXE-10 Australia, INS1500 (Japan D70), Numeris, SwissNet, EuroISDN, and 1TR6 Central Office interfaces.

Attendant Calls

If the attendant originates an external call using an ISDN CO trunk providing AOC, the call charging information received from the network will be saved in the trunk's temporary meter at the end of the call. The contents of the temporary meter will be added to the contents of the attendant meter for the customer.

If the attendant is the last controlling party (i.e., does not release the call), who gets charged depends on who initiated the call. If the attendant made the call with a party on the source, it is the source who gets charged. If the attendant made the chargeable call and then dialed an internal party (station or TIE trunk) on the destination side of the loop without releasing, the attendant gets charged.

Attendant operation of metered calls

If an attendant desires billing information immediately upon the completion of a long distance call, this call can be flagged as a metered call. When a metered call is terminated, the attendant is recalled and the calculated charge for this call is displayed on the console.

The attendant activates this feature by requesting the call charge on any outgoing ISDN CO trunk providing AOC by pressing the METER (MTR) key after making the call. When the MTR key is pressed, the METER lamp is lit and all metered outgoing ISDN calls connected to the active console loop are marked as "metered". Additional ISDN CO trunks added to the conference are also marked as "metered" automatically.

When a “metered” call is disconnected, a meter recall is presented to the same attendant who originated the call. If that attendant is in position busy, the meter recall is presented to the next idle attendant console in the same node (whether or not it is equipped with a meter key). If all attendants are in Night Service or position busy, the recall is saved in the attendant queue until one of the attendants in the same node becomes idle.

In the case of a station being extended to a toll call originated by the attendant, and if the station recalls to the attendant with the toll call attached, and the attendant accepts the call and releases the station, then the attendant is the last controlling party. The charge goes to the attendant meter.

Automatic Call Distribution

Since ACD calls are incoming only, AOC cannot apply to them; if a reverse charge is sent, it is rejected. Personal calls made by an agent are metered.

Barge-in /Busy Verification

When either of these features is operated a conference is established. If the trunk party disconnects first, the charge information is received and assigned to the originator of the call. If the originator of the call disconnects first, the charge information is received and assigned to the attendant when the trunk party releases.

Break-in

When this feature is operated a conference is established. The “desired” party can either disconnect and be rerung or the party on the trunk connection initiates the disconnection. In either case, the charge information is received and charged to the originator of the call (the “desired” party).

BRI Terminal

The AOC feature does not support AOC being sent to BRI terminals. A meter can be assigned to a Digital Subscriber Link (DSL). All chargeable calls made by a BRI terminal attached to this DSL are charged against the DSL’s meter.

BRI Trunk Access for Japan

AOC information can be received over ISDN BRI trunks using the Japan D70 signaling.

Call Forward All Calls, Call Forward Busy, Call Forward No Answer

Chargeable calls transferred to a station and redirected to another station via Call Forward are charged against: the “Call Forwarded to Station”, if it answers and the transferring party completes the transfer; or the transferring party if the call is abandoned or the transfer is not completed.

Call Park

When a station user parks an AOC trunk call, the calling party will continue to be assigned the charge until the call is answered by another station user.

Call Pickup

Chargeable calls which are extended to a station and answered at another station via Call Pickup are charged against: the “Pick up Station”, if the “Transferring Station” completes the call; or the transferring party if the transfer is not completed.

Call Transfer

If a station transfers a call to another internal station, while the dialed station is still ringing, the call charging information is assigned to the transferring station until the call is answered or abandoned by the external party. When the call is answered, call charging information received is stored against the station to which the call is transferred. If the station user consults with the dialed station user by using the transfer feature, the call charging information is still assigned to the transferring party until the call is actually transferred to the consultation party.

Call Waiting

If a call is transferred to a busy station with “Call Waiting Allowed”, the transferring station will be charged until the call is answered.

Conference

Whenever an ISDN CO trunk providing AOC is added into a conference, the charge for the call continues to be assigned to the station user who initiated the conference. If the person who adds the call to the conference disconnects while the conference is ongoing, the DN to be charged is changed to the one that has been in the conference the longest. Since this feature only provides charge information at the end of the call, the originator of the call who drops out will not incur any charge.

Consultation

If a consultation call is established over a CO trunk providing AOC, the call is charged to the station to which the call is transferred.

Data Calls

Meters can be assigned to data terminals and a terminal can be charged for the call, but the information cannot be displayed on the terminal side because the data terminal does not have its own display.

Hospital Management

AOC information is used in the “Paid Calls Restriction” subfeature of Hospital Management. This is not fully supported as the charge appears at the end of the call. The person with the paid call restriction may actually talk longer than allowed without any indication, since the charge is not known until the call is finished.

Hunting

Chargeable calls transferred or extended to a station and redirected to another station via Hunting are charged against: the “Hunted to Station”, if the “Hunted to Station” answers and the transferring party completes the transfer; or the transferring party if the call is abandoned or the transfer is not completed.

In-band Tones and Announcements

In some ISDN cases, In-band Tones or Announcements are required. In these cases, the disconnect sequence is delayed to allow the user to hear the tones or announcements. The charge information is not displayed, printed in the CDR record or added to the permanent meter of the set until the timer expires or the user releases the call.

Malicious Call Trace (Enhanced)

In Australia, the ISDN version of this feature introduces a disconnect delay to sets with MCT class of service. This delay may be up to 30 seconds. In cases of sets having MCT class of service, the charge information is not delayed, printed in the CDR records, or added to the permanent meter of the set until the delay timer expires or the user releases the call.

Meridian Mail

The same interaction applies as for Call Transfer.

Message Registration

Due to the change in the packaging for MR/PPM, anyone using MR must ensure that the prompt in LD 17 is set to MR before setting up the associated data. This is the default for sites configured with Release 20 and later software.

Multiple Appearance DN

Calls made on a multiple appearance DN are charged to the MADN if there is only one MADN. If there is more than one MADN, the charge is assigned to the MADN that was configured last. This operation is the same as in the PPM/MR feature. In order to identify which prime DN the CDR record is referring to, the Auxiliary ID (AXID) must be set to “Yes” in the CDB. This enables the TN to be printed in the CDR record.

Multiple Party Operation

The same interaction applies as for Call Transfer.

Network Call Transfer

Advice of Charge is not supported for Network Call Transfers.

Periodic Pulse Metering (PPM)

The following PPM capabilities and treatments are supported by AOC:

- Multiple Appearance DN
- Attendant Recall
- Automatic Call Distribution
- Barge-in
- Break-in
- Busy Verification
- Recording the accumulated call charging information for each call on the CDR record (if CDR is equipped)
- Calculating the total charge for each call, based on the assigned unit cost and the accumulated call charging information

- Allowing the attendant to mark a call as being metered, and to display the charge for the call, and
- Allowing a Meridian 1 telephone set equipped with an MRK key and digit display to access meters.

Radio Paging

In the case of party “A” making a toll call and then attempting to transfer or conference in a user who is forwarded to a pager, party “A” is charged. If the paged party answers the call, the charge is assigned to the station where the call is picked up.

Recovery of Misoperation of Call Transfer

In the case of the call being forwarded to the attendant after misoperation and being answered by the attendant, the charge is assigned to the attendant’s meter. All the other misoperation options are handled as described in the Call Transfer section.

Advice of Charge for Central Office Connectivity

The following sections describe the AOC capability as applied to AXE-10 Australia, INS1500 (Japan D70), Numeris, SwissNet, EuroISDN, and 1TR6 connectivities.

AXE-10 Australia (end of call)

This capability provides AOC call charging information sent from an AXE-10 Australia CO to a Meridian 1 PBX over an ISDN PRI connection. Information is received, displayed, and recorded in the CDR when the call is taken down.

AOC for Australia AXE-10 supports PBX control of AOC, which means that the call charging information must be requested for each outgoing call (instead of expecting it for every call once AOC has been configured).

Feature packaging

The following packages are required for Advice of Charge:

- Call Detail Recording (CDR) package 4;
- Call Detail Recording Teletype Terminal (CTY) package 5;
- Controlled Class of Service (CCOS) package 81;
- Background Terminal (BGD) package 99;
- Periodic Pulse Metering/Message Registration (MR) package 101;
- Integrated Services Digital Network (ISDN) package 145;
- 2.0 Mbit Primary Rate Interface (PRI2) package 154; and
- International Primary Rate Access (IPRA) package 202.

EuroISDN (call set-up, during call, end of call)

This feature provides the Advice of Charge supplementary service for ISDN PRI to Public Exchanges/Central Offices complying with the European Telecom Standard Institute (ETSI) standard specification ETS 300 102. This standard is also known as EuroISDN.

AOC for EuroISDN offers three subservices, which may be configured in Overlay 16 (information at call setup – AOC-S; during the call – AOC-D; and at the end of the call – AOC-E).

Operating parameters

The countries AOC supports on EuroISDN are Austria, Germany, Norway, Italy, Switzerland, Finland, Holland, and Portugal.

Requesting the AOC supplementary service on a per call basis is not supported by this development.

ISDN Primary Rate Interface (PRI) requires the following hardware:

- 2.0 Mbit Primary Rate Access card (MTBK50AA) for the layer 1 interface on the Option 11
- The downloadable D-channel Handler daughter board (NTBK51AA) card is required for the Option 11
- XPRI card (NTBK50AA) and a daughter board (NTBK57AA) supporting the downloadable functionality
- The clock controller card (NTAK20BA/BB)

Feature packaging

The following packages are required for Advice of Charge:

- EuroISDN (EURO) package 261.
- Call Detail Recording (CDR) package 4;
- Call Detail Recording Teletype Terminal (CTY) package 5;
- Controlled Class of Service (CCOS) package 81;
- Background Terminal (BGD) package 99;

- Periodic Pulse Metering/Message Registration (MR) package 101; and
- International Supplementary Features (SUPP) package 131.

For PRI connectivity, the following features are also required:

- Integrated Services Digital Network (ISDN) package 145;
- 2.0 Mbit Primary Rate Access (PRI2) package 154;
- ISDN Supplementary Features (ISDNS) package 161; and
- International Primary Rate Access (IPRA) package 202.

INS1500 (Japan D70) (end of call)

This capability provides AOC call charging information sent from an INS1500 (Japan D70) CO to a Meridian 1 PBX over an ISDN PRI connection. Information is received, displayed, and recorded in the CDR when the call is taken down.

Prior to the introduction of the AOC feature for Japan D70, the method used to determine which feature was equipped was to check if the International Supplementary Features software package 131 was equipped. If it was, PPM was required. Since the Supplementary Features package is not available in Japan and AOC requires PPM software, a new method of determination has been introduced. This method uses a system-wide flag to allow the customer to select between MR and PPM. This flag is set by a prompt in LD 17.

Special handling for call charges that exceed normal capacity

The largest charge that can be accepted from the Public Switched Telephone Network (PSTN) is 3,999,999,999. In Japan, however, the largest possible charge is 99,999,999,999 Yen. The Meridian 1 system algorithm for processing the call charge has been set up so that if a number larger than 3,999,999,999 is received by the system, the number is stored as 4,000,000,000 and handled as any smaller number; the rest of the charge is dropped (it is highly unlikely that a charge of this value will ever be encountered).

Feature packaging

The following packages are required for Advice of Charge:

- Call Detail Recording (CDR) package 4;

- Call Detail Recording Teletype Terminal (CTY) package 5;
- 1.5 Mbit Digital Trunk Interface (PBXI) package 75;
- Controlled Class of Service (CCOS) package 81;
- Background Terminal (BGD) package 99;
- Periodic Pulse Metering/Message Registration (MR) package 101;
- Integrated Services Digital Network (ISDN) package 145;
- 1.5 Mbit Primary Rate Access (PRA) package 146; and
- International Primary Rate Access (IPRA) package 202.

Numeris (end of call)

This capability provides AOC call charging information sent from a Numeris CO to a Meridian 1 PBX over an ISDN PRI connection. Information is received, displayed, and recorded in the CDR when the call is taken down.

Feature packaging

The following packages are required for Advice of Charge:

- Integrated Services Digital Network (ISDN) package 145;
- 2.0 Mbit Primary Rate Access (PRI2) package 154; and
- International Primary Rate Access (IPRA) package 202.

Numeris and SwissNet (during call)

Advice of Charge (AOC) Real Time is provided through this feature as part of the connectivity between the Meridian 1 and both Numeris for France and SwissNet for Switzerland.

Each country offers the AOC service in a specific manner. For Numeris Connectivity for France, and SwissNet Connectivity for Switzerland, AOC service is configured on an ISDN Route basis, and provides the total cost of the call at call tear down. This information can be displayed on the user's set, as well as stored in Call Detail Recording (CDR) records. Advice of Charge (AOC) Real Time supplementary service uses cumulative charging information during a call to provide "real-time" updates of the charging information. To do this, the feature uses the same meters as PPM and follows specifications defined in the PPM feature.

Advice of Charge is provided as follows:

- for a simple call, the total cost for the call is produced in the CDR record
- for a call which has been modified by transfer or conference, the cost of each part of the call is provided in each CDR record associated with each extension, and
- a call is redirected, the CDR records show all extensions associated with a particular call together with the call costs associated with each extension.

Operating parameters

The display of call charge during the call is not supported, but the display of call charge at the end of the call is provided by the feature.

For Numeris, a user must be subscribed to the AOC supplementary service in order to get the charging information sent to the Meridian 1.

For SwissNet both services are given by the Swiss PTT without previous subscription.

Feature packaging

The following packages are required for Advice of Charge:

- Periodic Pulse Metering/Message Registration (MR) package 101.

For PRI connectivity, the following packages are required:

- Integrated Services Digital Network (ISDN) package 145;
- 2.0 Mbit Primary Rate Access (PRI2) package 154;
- International Primary Rate Access (IPRA) package 202; and
- ISDN Supplementary Features (ISDNS) package 161.

1TR6 (end of call)

This capability provides AOC call charging information sent from a 1TR6 CO to a Meridian 1 PBX over an ISDN PRI connection. Information is received, displayed, and recorded in the CDR when the call is taken down.

Feature packaging

The following packages are required for Advice of Charge:

- Periodic Pulse Metering/Message Registration (MR) package 101.

For PRI connectivity, the following packages are required:

- Integrated Service Digital Network (ISDN) package 145;
- 2.0 Mbit Primary Rate Interface (PRI2) package 154;
- International Primary Rate Access (IPRA) package 202; and
- ISDN Supplementary Features (ISDNS) package 161.

Attendant and Network Wide Remote Call Forward

Prior to Release 20, the Remote Call Forward (RCFW) feature allowed a user to administer call forward from a remote set within the Option 11 or from outside the Option 11 through the Direct Inward System Access (DISA) number. The RCFW feature was not available on a network wide basis, nor was it applicable to Attendant Consoles. This enhancement introduces the RCFW feature across the Meridian Customer Defined Network (MCDN), while also providing the attendant with RCFW capabilities. New ISDN FACILITY messages are used to facilitate the RCFW feature operation in an MCDN.

The feature capabilities of the set-based (FFC activated) network wide application of the RCFW feature match those of the current standalone RCFW feature.

The attendant RCFW functionality is controlled by a new flexible Attendant key (RFW). The attendant has the capability to view the current call forward number and determine the call forward status of any station. It is also possible for an attendant to activate or deactivate call forward for a particular station. This functionality is applicable both locally within the Meridian 1 and network wide.

A new optional customer-based password is introduced for attendant RCFW operation. This password is configured in LD 15 and is the only password required for attendant RCFW operation. The station control password previously used by the Flexible Feature Code (FFC) set-based RCFW is not required when the attendant activates RCFW by pressing the RFW key

Operating parameters

The network wide application of this feature is only applicable to nodes in an MCDN. The nodes in the network must be Meridian 1 switches, running a minimum of Release 20 software. No other Central Office (CO) or PBX type is supported for this feature operation.

For set-based network operation of the Remote Call Forward feature, the Station Control Password Length (SCPL) must be configured to be the same length for all nodes in the network. Attempts to operate RCFW with different SCPLs will result in overflow tone being presented to the user.

For network operation of the RCFW feature, the Private Network Identifier (PNI) must be configured consistently for all nodes in the network.

The Attendant and Network Wide RCFW features use the existing RCFW code to activate or deactivate call forward on stations. As such, all limitations applicable to the local RCFW feature are applicable to the network and attendant operation of the feature.

As the Swedish CD Attendant Console does not support alpha characters, the “PWD” prompt is not displayed on the console’s digit display when a password is required. The indication that a password is required is limited to the winking RFW key lamp.

No new hardware is required for this feature.

Feature interactions

Outpulsing of Asterisk and Octothorpe (OAPO)

If the OAPO package is equipped, the “#” will be treated as any other dialed digit and will not be used to signal end of dialing. The end of dialing digits to be used are defined in LD 15.

BRI

Since ISDN BRI sets do not support Flexible Feature Codes (FCCs), Remote Call Forward is not applicable to BRI sets.

Multiple Appearance DNs

The RCFW feature only applies to the primary appearances of Multiple Appearance DNs, and it is recommended that only one appearance of a Multiple Appearance DN be configured as the prime DN.

For the case of multiple stations with the same prime DN and SCPW, the RCFW operation will apply to the station that has the Multiple Appearance Redirection Prime (MARF) assigned to it.

If none of the stations having the DN and SCPW assigned are configured as the MARF TN for that DN, the RCFA and RCFD will apply to all stations matching the DN and SCPW.

The attendant-based RCFW feature will only apply remote call forward operation to the prime DN with MARP status. If the DN is not the prime DN or does not have MARP status, overflow tone will be received by the user.

Call Forward Activation from any Feature/Call Forward and Busy Status

There are no direct conflicts with either of these features and the RCFW feature.

Preventing Reciprocal Call Forward

When Preventing Reciprocal Call Forward Allowed (PVCA) is defined in LD 15, a set within the same customer configuration cannot be call forwarded to a set that is call forwarded back to it. Thus, CFW loops are prevented.

This feature applies when the CFW DN is changed by Remote Call Forward. For network operation of the set- and attendant-based RCFW features, entering an invalid CFW DN (under the rules of the PRCF feature) results in overflow tone being returned and the CFW DN being ignored.

Phantom TN

A Phantom TN does not physically exist, but can be configured with limited hardware associated with it (i.e., no sets or line cards); however, all required data blocks are configured.

The Phantom TN feature uses the RCFW feature to configure and activate/deactivate the CFW DN on the Phantom TNs.

As the data blocks associated with Phantom TNs match those of standard PBX sets configured within the Meridian 1, the operation of the RCFA and RCFD features on Phantom TNs is transparent to the RCFW feature. As such, the set-based local and network RCFW features can be used to configure and activate/deactivate the CFW DN of Phantom TNs.

The Phantom TN feature uses a Default Call Forward (DCFW) DN. If call forward is not active on the Phantom TN, all calls to the Phantom TN DN are routed to the DCFW DN.

The Phantom TN feature modifies the set-based RCFW feature so that if CFW is not active on the Phantom TN, and the CFW DN entered in the RCFV operation matches the DCFW DN, confirmation tone is returned to the RCFV user; if the CFW DN entered does not match the DCFW DN, overflow is returned.

This change to the set-based RCFV operation is applicable to the network RCFV operation. The operation of this feature network wide requires no changes to the ISDN message passing for the set-based network RCFV operation.

There is no Attendant RCFW operation which interacts with the DCFW DN of Phantom TNs.

Traffic Measurements

The peg count, for the attendant RFW key, will be generated on the first RFW key press of the RCFW operation. While the RFW key may be pressed multiple times during a single RCFW function, the peg count will only be implemented once.

The RFW key peg count will be included in the TFC005 feature key usage traffic report.

Feature packaging

The Attendant Remote Call Forward (ARFW) package 253 must be provisioned to activate the Attendant-based RCFW feature.

For network operation the following software packages are required: Integrated Services Digital Network (ISDN) package 145; Network Alternate Route Selection (NARS) package 58; and any other trunk or dialing plan packages as required by the customer's configuration.

The following are prerequisites for set-based RCFW: Optional Features (OPFT) package 1; Flexible Feature Codes (FCC) package 139; and Controlled Class of Service (CCOS) package 81.

The following are prerequisites for implementation on PBX sets: Special Service for 2500 Sets (SS25) package 18; and 500 Set Dial Access to Features (SS5) package 73.

Attendant Blocking of DN

Attendant Blocking of Directory Number (DN) allows a person to dial the attendant DN and request an external (long distance) call, and then disconnect while waiting for the call to be processed by the attendant. The requesting DN is idle and can receive and make calls.

When the attendant is ready to make the external (long distance) call, the Attendant Blocking of DN feature provides the attendant with the ability to block the DN while the external call request is being processed. The line appears busy to any caller attempting to contact the blocked DN. The blocked DN cannot be used to originate a call and will be connected to the attendant if it goes off hook. When the attendant has completed the external call, the blocked DN can be rung and the call extended. The attendant is guaranteed that the requesting DN is not busy and is available to take the call when the processing has been completed. This feature works in both standalone and Meridian Customer Defined Network (MCDN) environments.

Previously, this feature had been available on Swedish A345 PBXs, but now is also available on the Meridian 1. Although developed for Telia Sweden, the feature is applicable to all marketplaces desiring Attendant Blocking of DN functionality.

Operating parameters

The Attendant Blocking of DN feature can only be activated as the source party of the Loop key on the Attendant Console.

The attendant has the ability to use the Attendant Blocking of DN feature only for the following types of DNs: sets with ordinary Single Call Arrangement Non Ringing key (SCN)/Single Call Arrangement Ringing key (SCR) DNs and analog (500/2500 type) telephones. HOT DNs, Multiple Call Arrangement None Ringing key (MCN)/Multiple Call Arrangement Ringing key (MCR) DNs, Automatic Call Distribution (ACD) DNs, Primary Line DNs (PLDNs), any trunk access code, Flexible Feature Codes (FFCs), Basic Rate Interface (BRI) and all other types of extensions are considered to be invalid for the Attendant Blocking of DN feature.

When the Attendant Blocking of DN feature is activated for a DN, it is only the DN dialed that is blocked. Other DNs assigned to the set will be idle.

If Attendant Blocking of DN is attempted on a Multiple Appearance Single Call Arrangement DN, all idle appearances of the DN will be blocked.

The Attendant Blocking of DN will only be supported on the following Attendant Console types: M2250; M1250; QCW4; QCW8; or M345CD.

No new hardware is required for this feature.

Feature interactions

ACD

It is not possible to activate the Attendant Blocking of DN feature for an ACD DN. If an attempt to block an ACD DN is made, the attempt will be canceled and overflow tone will be returned. However, individual DNs on ACD sets can be blocked.

Attendant Hold

An Attendant Blocking of DN call can be put on hold by the attendant and will in this case be subject to normal Attendant Hold treatment. The Semi-automatic Camp-on (SACP) key lamp will be dark while on hold and be lit again when taken off hold. The same applies to Automatic Hold on the Loop key.

Attendant Break-in

The Attendant Blocking of DN and the source side Predial Break-in features are mutually exclusive for the same call. If the SACP key lamp is lit when the Break-in key is pressed to start a Predial Break-in attempt, the Break-in key is ignored. On the contrary, if the Break-in key lamp is lit and no call attempt is made on the source side when the SACP key is pressed to start an Attendant Blocking of DN, the SACP key is ignored.

If a Break-in attempt is made for an Attendant Blocking of DN call, the Break-in attempt will be considered to be temporarily denied.

It will be possible to Break-in on the destination side with an Attendant Blocking of DN call on the source side of the Attendant Console. The same limitations to Break-in will apply as if the source side call is a normal call.

Busy Lamp Field/Enhanced Busy Lamp Field

When a DN is blocked due to the Attendant Blocking of DN feature, the Busy Lamp Field/Enhanced Busy Lamp Field lamp corresponding to this DN displays the busy status of the DN as for ringing calls.

Busy Verify

The Attendant Blocking of DN and source-side Busy Verify are mutually exclusive for the same call. If the SACP key lamp is lit when the Busy Verify key is pressed to start a Busy Verify attempt, the Busy Verify key is ignored. On the contrary, if the Busy Verify lamp is lit when the SACP key is pressed to start an Attendant Blocking of DN attempt, the SACP key is ignored.

If a Busy Verify attempt is made on an Attendant Blocking of DN call, it will be denied.

Call Detail Recording Time to Answer

If the CDR Time to Answer feature is active, the time registration before answer will be started when the SACP key is pressed to ring the blocked DN and not when the DN is blocked.

Call Forward All Calls/Internal Calls/Call Forward and Busy Status

The Attendant Blocking of DN feature will override these Call Forward features. If the dialed DN of the set is idle, the DN can be blocked; if the DN is busy, busy tone will be heard.

Call Forward No Answer

The Attendant Blocking of DN feature will override the Call Forward No Answer feature. If the blocked DN of the set has the Call Forward No Answer feature active when the SACP key is pressed to ring the DN, the DN will ring until answered or disconnected. No Call Forward No Answer will be done for the Attendant Blocking of DN call.

Call Park

It is not possible to park an Attendant Blocking of DN call. If a Call Park call recalls to a blocked DN, the recall will be treated as if the DN is in a ringing state.

Call Waiting

If a set that has the Station-to-station Call Waiting feature active (CLS SWA and a Call Waiting (CWT) key for SL-1 and digital sets) is idle when an Attendant Blocking of DN attempt is made, the Attendant Blocking of DN attempt will be allowed and processed as normal. If the DN is idle and there is an active call on the Call Waiting key, the Attendant Blocking of DN attempt will be allowed.

If a set has the Station-to-station Call Waiting feature active and the DN to be blocked is busy when an Attendant Blocking of DN attempt is made, the Attendant Blocking of DN attempt will be canceled and busy tone will be returned.

For a set that has the Call Waiting (or Station-to-station Call Waiting) feature active and a DN is blocked due to the Attendant Blocking of DN feature, any incoming call to the blocked DN will receive busy tone.

Camp-on

Camp-on will be denied for a DN that is blocked due to the Attendant Blocking of DN feature.

Directory Number Delayed Ringing

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.

Do Not Disturb

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.

FFC Boss Secretary Filtering

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.

Flexible Feature Codes

If a Flexible Feature Code is dialed after pressing the SACP key to initiate an Attendant Blocking of DN attempt, overflow tone will be provided and the attempt canceled.

Group Hunting

It is not possible to activate the Attendant Blocking of DN feature for a Pilot DN (PLDN). 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.

HOT DN

It is not possible to activate Attendant Blocking of DN for a HOT DN extension. If an attempt to block a HOT DN extension is made, the attempt will be canceled and overflow tone will be returned.

Hunting

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).

ISDN Basic Rate Interface (BRI) Trunk Access

It is possible to use the Attendant Blocking of DN feature in an ISDN MCDN based on BRI TIE trunks if Network Attendant Service (NAS) is configured in the network.

Line Lockout

If an Attendant Blocking of DN attempt is made on a set in Line Lockout state, busy tone will be returned.

Make Set Busy

The Attendant Blocking of DN feature will override the Make Set Busy feature. If the dialed DN of the set that has the Make Set Busy feature is idle, the DN will be blocked and if the DN is busy, busy tone will be heard.

Multiple Appearance Multiple Call Arrangement

It is not possible to activate the Attendant Blocking of DN feature for a Multiple Appearance Multiple Call Arrangement DN (MCA DN (MCN/MCR)). If an attempt is made to block an MCA DN (MCN/MCR), the attempt will be canceled and overflow tone will be returned.

Multiple Appearance Single Call Arrangement

If Attendant Blocking of DN is attempted on a Multiple Appearance Single Call Arrangement DN (SCA DN (SCN/SCR)), all appearances of the DN will be blocked.

New Flexible Code Restriction

When the attendant has a blocked DN on the source side and dials on the destination side, any New Flexible Code Restriction active for the set of the blocked DN will be overridden. This is the same as if the attendant had a normal established call to the DN on the source side and dials the destination side.

Phantom TN

DNs on Phantom TNs will not be overridden by the Attendant Blocking of DN feature.

Radio Paging

If a transferred Radio Paging call recalls to a blocked DN, the recall will be treated as if the DN is busy.

Ring Again

It is possible to activate Ring Again towards a DN that is blocked due to the Attendant Blocking of DN feature.

Ring Change Key

When the SACP key (or Signal Source) key is pressed to ring a blocked SCR where the Ring Change feature is activated, an audible ring signal will always be given. This is independent of the Ring Change status.

Ring/Hold LED Status

When a DN is blocked, the status of the DN lamp will be the same as for a call put on hold.

Semi-automatic Camp-on

The Attendant Blocking of DN feature uses the SACP key to activate a blocking attempt, but the Attendant Blocking of DN feature is only valid on the source side of the Attendant Console. The Semi-automatic Camp-on feature is only valid on the destination side of the Attendant Console.

To have the Attendant Blocking of DN feature available and not the Semi-automatic Camp-on feature, a new response to the SACP prompt has been introduced in LD 15. Prompt SACP = NO means the Semi-automatic Camp-on feature is not available even if the SACP package is equipped and an SACP key exists on the Attendant Console. To have the Semi-automatic Camp-on feature available the SACP prompt must be answered with SNGL or ALL which have the same meanings as before.

Signal Source

The Signal Source key can be used to notify a blocked DN. Using the Signal Source key for an Attendant Blocking of DN call will give the same response as the SACP key when the DN is blocked (i.e., ring the blocked DN and darken the SACP key lamp). The Signal Source key cannot be used to initiate an Attendant Blocking of DN call.

Single Call No Ringing DN

When the SACP (or Signal Source) key is pressed to ring a blocked Single Call No Ringing DN (SCN), an audible ring signal will be given.

Source Included when Attendant Dials

The Attendant Blocking of DN feature will follow the current Source Included when Attendant Dialing handling.

Vacant Number Routing

The Attendant Blocking of DN feature will work across an ISDN network if the call is routed due to the Vacant Number Routing feature.

Feature packaging

The Attendant Blocking of DN feature is packaged under Semi-automatic Camp-on (SACP) package 181.

To work in an ISDN network environment, Network Attendant Services (NAS) package 159 is required.

Feature operation

To block a DN

- 1** The attendant presses an idle Loop key.
- 2** The attendant presses the Semi-automatic Camp-on (SACP) key.
- 3** The SACP key lamp lights.
- 4** The attendant dials the source DN that is to be blocked. If the dialed DN is idle, the DN lamp will have the same state as a call put on hold, but the DN will not ring. If the DN is busy, the attendant hears busy tone and the SACP key lamp darkens.

If the dialed DN is idle, it is blocked. The DN lamp indicates a call on hold state, although the DN will not ring (for analog [500/2500 type] telephones, there is no indication that the DN is blocked). On the Attendant Console, the SACP key lamp remains lit and the Source key lamp begins blinking.

If the dialed DN is busy, the attendant presses the Release (or Release SRC) key to Release the call.

To place an outgoing call for the blocked DN

- 1** The attendant establishes a call to the desired destination in the normal way.
- 2** The attendant presses the SACP (or Signal SRC) key. The ringback tone is heard.
- 3** If the source DN answers, the attendant presses the Release key to extend the call between the destination to the source.

To Release a blocked DN

The attendant presses either the SACP key or Signal Source key to ring the DN

—or—

The attendant presses either the Release key or the Release Source key to Release the source DN, which then becomes idle.

To notify a blocked DN of an established call

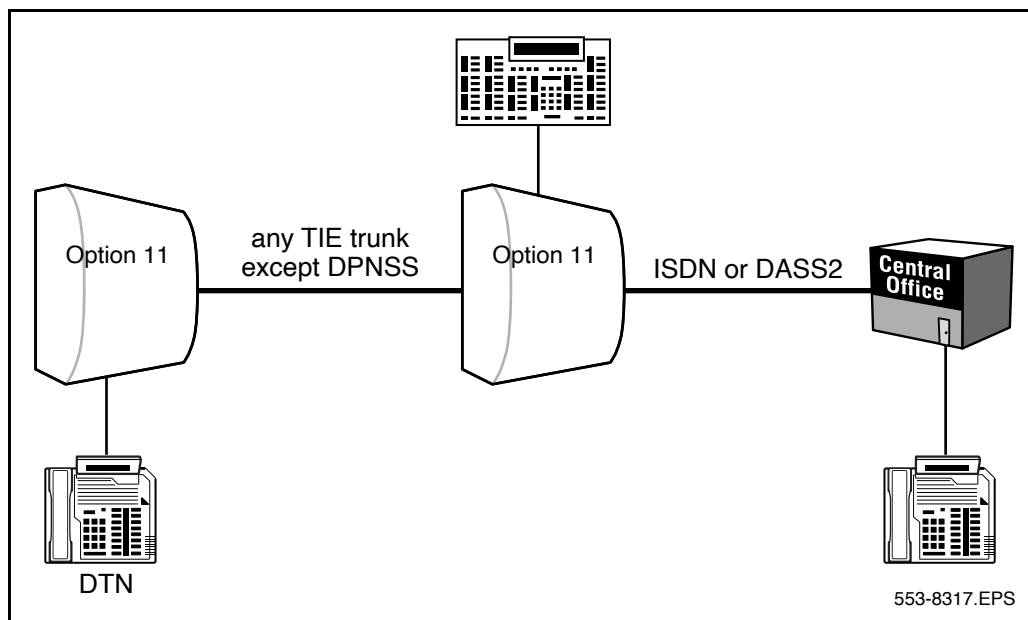
The attendant presses the SACP or Signal Source key.

Attendant Through Dialing Networkwide

Attendant Through Dialing Networkwide extends the functionality of through dialing via an attendant to any Integrated Services Digital Network (ISDN) or DASS2 outgoing trunk. This feature allows an attendant to seize an outgoing Integrated Services Digital Network (ISDN) or DASS2 trunk for a calling party located on the same or another node.

Figure 9 illustrates Attendant Through Dialing Networkwide.

Figure 9
Attendant Through Dialing Networkwide



In the existing standalone capacity, Attendant Through Dialing allows internal callers to request an outgoing trunk except DPNSS from an attendant. In the existing network capacity, Attendant Through Dialing allows callers linked by any TIE trunk to request an analog or DTI2 trunk from the attendant.

When requested, the attendant dials a specific code and extends the call once the Destination (DEST) lamp lights. When the attendant accessed the trunk the caller was free to dial out. However, with standalone Attendant Through Dialing, the outgoing trunk seized must be either an analog or digital trunk. Attendant Through Dialing Networkwide enhances the through dialing networkwide capability to ISDN or DASS2 outgoing trunks.

When this feature is provisioned, an attendant seizes the outgoing trunk by pressing the Release (RLS) key. Following this, the call is extended back to the calling party who receives dial tone and dials the remaining digits.

This feature is applicable in situations where the calling party is not permitted to dial a defined code that provides access to a public or international network or other costly telecom services. In these situations, the calling party requests that the attendant dial a numbering plan for the calling party, seize an external trunk and extend the call back to the calling party. Table 2 shows situations when the attendant is allowed to press the Release (RLS) key depending on the type of numbering plan implemented by a customer.

Table 2
Numbering Plans and Attendant Release of external trunk

Numbering Plan used to seize external trunk	Destination (DEST) becomes lit
Route Access Code	After Route Access Code
Flexible Numbering Plan	After Special Number
Coordinated Dialing Plan	After Trunk Steering Code

Operating parameters

This feature supports the all ISDN trunk types on Basic Rate Interface (BRI) and Primary Rate Interface (PRI). Attendant Through Dialing Networkwide is also supported over analog, DTI and DIT2 trunks.

The Attendant Through Dialing Networkwide feature is not supported over DPNSS. Therefore, an established link cannot be a DPNSS trunk if the outgoing trunk is ISDN or DASS2.

Attendant Through Dialing Networkwide is configured to override/bypass Access Restrictions configured as New Flexible Code Restrictions. Other access restrictions such as Access Restrictions, Schedules Access Restrictions and Trunk Barring are not affected by Attendant Through Dialing Networkwide.

This feature is not supported on sets configured with Dial Pulse (CLS = DIP). Attendant Through Dialing Networkwide is only supported on set configured with Digitone (CLS = DTN).

Attendant Through Dialing Networkwide is available on all types of dialing configurations on ISDN routes, Enbloc or Overlap Signaling. However, if the attendant dials a Trunk Steering Code or Special Number, the outgoing ISDN trunk must support Overlap Signaling.

If an attendant dials a Trunk Steering Code or Special Number over an ISDN trunk connected to a Central Office/Public Exchange, the outgoing trunk must support Overlap Signaling.

Attendant Through Dialing Networkwide allows a caller to bypass all trunk access restrictions at the set level. Once a caller begins dialing an external number, the digits dialed are not analyzed for Access Restrictions, Call Connection Restrictions.

An attendant cannot extend a call back to a caller after dialing an Electronic Switched Network (ESN) access code (AC1/AC2) even if a tone is detected. The route being used is unknown at this time. Therefore, if the access code to the public network is defined as AC1 or AC2, the attendant must dial additional digits, such as a Special Number, before being allowed to press the Release key.

The Attendant Through Dialing Networkwide feature is not supported if the outgoing trunk on the attendant's node is Virtual Network Service (VNS) trunk.

When a calling party requests through dialing, their set display is updated. The called party's display receives the attendant's name or number and maintains this information throughout the duration of the call.

Feature interactions

Autodial

Attendant Through Dialing Networkwide supports Autodial provided that the stored Autodial number excludes the digits previously dialed by an Attendant.

Call Detail Recording

The record on the outgoing trunk node shows the outgoing trunk in the terminating ID field.

No record is output on the Attendant's node for the Destination (DEST) side during call extension. This occurs regardless of the configuration for the outgoing trunk. All other records are produced according to configuration.

If the Calling Line Identification (CLID) option is activated in the Call Detail Recording, the calling party's Directory Number (DN) is printed in the Attendant's node.

If End-to-End Signaling is used to establish a link, the ECDR prompt in Overlay 15 can be used to print End-to-End Signaling digits in the CDR record.

ISDN QSIG/EuroISDN Call Completion

The Call Completion to Busy Subscriber and the Call Completion on No Reply functionalities are not supported if an external call is initiated by the Attendant Through Dialing Networkwide feature.

Last Number Redial

Last Number Redial is not supported when the attendant extends a call back and the caller begins dialing digits.

Network Attendant Service

Network Attendant service can be used on the Meridian Customer Defined Network (MCDN) to automatically locate an attendant from one node to another.

When Attendant Through Dialing Networkwide is provisioned, the Attendant's Destination (DEST) lamp is updated after dialing Route Access Code, Trunk Steering Code or Special Number rather than waiting for the ALERTING message.

Pretranslation

Pretranslation is supported during the attendant dialing phase. The attendant dials a pretranslated digit in the Trunk Steering Code, Route Access Code or Special Number to seize an external trunk. Pretranslation is not supported in the through dialing phase. Therefore, once the attendant extends the call back to the caller, the first digit the calling party dials is not pretranslated even if the calling party has pretranslation configured.

Recovery on Misoperation of the Attendant Console

The Attendant Through Dialing feature allows the attendant to press the RLS (Release) key or another Loop key when the called party is ringing without misoperating the console.

Speed Call

Speed Call is only supported in the attendant dialing phase. Speed Call is not supported once the caller begins dialing an external number. Once an external call is established, the caller cannot press the SCU (Speed Call User) key.

Stored Number Redial

Digits dialed by the caller using End-to-End Signaling are not retained by the Stored Number Redial feature.

Feature packaging

Attendant Through Dialing Networkwide requires the following packages:

- End-to-End Signaling (EES) package 10
- Integrated Services Digital Network (ISDN) package 145
- Overlap Signaling (OVLP) package 184
- New Format Call Detail Recording (FCDR) package 234

Attendant Through Dialing Networkwide also requires **one** of the following dialing plans:

- Flexible Number Plan (FNP) package 160
- Coordinated Dialing Plan (CDP) package 59
- Basic Alternate Route Selection (BARS) package 57
- Network Alternate Route Selection (NARS) package 58

Feature operation

- 1** Calling party dials an attendant that is located either on the same node as the caller or another node.
- 2** Calling party requests the attendant to seize an outgoing external trunk. This external trunk is located on either the same node or on another node.
- 3** The attendant dials a Trunk Steering Code, Special Number or Route Access Code to access the public network and waits for the lighting of the DEST lamp on the console. If the attendant dials either a Trunk Steering Code or a Special Number and the external trunk is an ISDN trunk, it must support Overlap Signaling. If the attendant dials a Route Access Code and the outgoing external is an type ISDN trunk then any type of dialing is supported.
- 4** When the DEST lamp is lit, then the attendant presses the Release (RLS) key or another loop key to extend the call back to the calling party requesting an outgoing external trunk.
- 5** The calling party hears dial tone and dials external number.

Australia ETSI

The Australia ETSI feature supports 2.0 MBit/s ISDN Primary Rate Interface and Basic Rate Interface Trunk connectivity for the Australian Central Office, in compliance with the Australia ETSI specification (Telstra).

This feature uses the existing EuroISDN software packages to provide the basic ISDN capabilities and supplementary services listed below (EURO is configured as the interface in the overlay programs when configuring PRI2 and BRI).

Basic ISDN services:

- 2.0 MBit/s PRI and BRI Basic Call Service
- Circuit-mode bearer capabilities (speech, 3.1 kHz audio, 64 kbps digital, and adapted 56 kbps to 64 kbps digital)
- COT, DID, DOD, and TIE trunk call types
- Calling Line Identification (public and private)
- Enbloc Sending
- Overlap Sending
- Channel Negotiation

Supplementary services:

- Calling Line Identification Presentation (CLIP)
- Calling Line Identification Restriction (CLIR)
- Connected Line Identification Presentation (COLP)
- Connected Line Identification Restriction (COLR)
- Malicious Call Trace
- Advice of Charge (AOC), during call set-up, during the call, and at end of call
- Sub-addressing (SUB)
- Direct Dial In (DDI)

Operating parameters

This feature requires downloadable D-Channel handling, for both the large systems (Options 51C, 61C, 81, 81C) and the small system (Option 11C). Therefore, the NTAK79AA PRI2 card is not supported on Option 11C because it has an on-board DCHI, and does not use the NTBK51downloadable D-Channel daughterboard.

Overlap Receiving is not supported.

Basic Alternate Route Selection (BARS) is not supported. Network Alternate Route Selection (NARS) is.

In a Meridian Customer Defined Network (MCDN), receiving Calling Party Name Display (CPND) and sending a CPND are not supported

MCDN Call Redirection (Call Forward, Call Forward No Answer, Hunt) is not supported.

MCDN Call Modification (Conference, Transfer) is not supported.

Network Call Redirection, Network Call Forward, and Network Call Forward No Answer (MCDN Component) are not supported.

Network Attendant Service (NAS) features are not supported across the Australia ETSI interface; however, incoming calls may be NAS routed from another node.

Trunk Route Optimization is not supported across the Australia ETSI interface.

All operating parameters apply to feature as for the EuroISDN Advice of Charge and Malicious Call Trace functionalities.

The Advice of Charge functionality is supported on a system basis only. It is not supported on a per call basis.

Reverse Charging is not supported, nor is requesting charging information from the user's side.

Tandeming of Advice of Charge charging information across a Meridian 1 network is not supported.

The display of charges is not supported on BRI sets and terminals.

Packet data handling is not supported for the BRI component of this feature.

Feature interactions

Calling Line Identification Enhancements

Prior to the CLID Enhancements feature introduced in Release 22, the Customer Data Block (Overlay 15) contained the prompts PFX1 and PFX2 (for Prefix 1 and Prefix 2) that were used to construct the CLID. The combination of PFX1, PFX2 and the originating DN were used to construct a correct number for the called party to dial in order to reach the calling party. If no digits are configured for either of the prefixes, then that part of the number will not be included in the Calling Party Number. Essentially, this meant that the CLID could only be built from key 0 of a set. Regardless of what key was used to make a call, it was the CLID for key 0 that was sent. Also, only one office code and one location code could have been assigned in the CLID for a customer.

With the introduction of the ISDN CLID Enhancements feature, PFX1 and PFX2 are no longer used to construct the CLID. CLID is now table-driven (when Overlay 15 is loaded, a customer can configure a CLID table), and virtually any of the information contained in the fields of the CLID table can now be programmed against any DN or DN key, on a per set basis. This means that the CLID that is sent from a set is now predicated on what is in the CLID table, rather than the LDN or PDN. That is, a CLID for any key is now built by taking the information contained in a particular field in the CLID table and adding that information to the key's DN. A multi-line set can now have DN keys that each has their own CLID. Or, the CLID of any one key on a set could be programmed to use the CLID of any other key on the set.

The construction of CLID is based on the CPFXS prompt in Overlay 16. If CPFXS = NO, then when constructing the Calling Number, the prefixes are retrieved from the Route Data Block via the responses to the HNTN and HLCL prompts. If CPFXS = YES, which is the default response, then CLID is built depending upon the prefixes HNTN and HLCL retrieved from the Customer Data Block (Overlay 15) via the entries in the CLID table (refer to the paragraph above for more details).

Also, the Meridian 1 can now support multiple office codes, location codes and steering codes in CLID. This means that any telephone set on one Meridian 1 can send a CLID that will have calls returned to another Meridian 1. This type of configuration is typically used in cases where a customer wants calls to be returned to only one central location.

How a CLID table is built

Prompts have been added to Overlay 15 that create a CLID table for a customer. This table contains up to 4,000 CLID “entries.” Each entry contains unique information pertaining to CLID, as explained in the following sections.

For users of an International Numbering Plan, the Meridian 1 now supports multiple Prefix 1 (PFX1) and Prefix 2 (PFX2) contents, and multiple Home Location Codes (HLOCs) and Local Steering Codes (LSCs), on a DN or DN key basis.

For an International Numbering Plan, each CLID entry can now contain the following:

- a 1 -6 digit national code for a home national number (HNTN), which is the equivalent of PFX1
- a 1 -12 digit local code for a home local number (HLCL), which is the equivalent of PFX2, or a one-12 digit Listed Directory Number for a switchboard
- a 1- 7 digit Home Location Code (HLOC)
- a 1 -7 digit Local Steering Code (LSC)

Another new capability pertains to how the HLCL is constructed. A new prompt, DIDN (which signifies “use DN as a DID number”) in Overlay 15, allows the HLCL to be built either using the digits in the HLCL plus the digits of the active key (if DIDN is set to YES, the DN is considered to be a DID number and is included in the CLID), or only the digits in the HLCL (if DIDN is set to NO, the DN is not included in the CLID since it is not a DID number), or based on a search on the DN keys, beginning from key 0, to find the CLID to be used (DIDN is set to SRCH).

For more detailed information, please refer to the “ISDN CLID Enhancements” feature module in *International ISDN PRI Feature description and administration* 553-2901-301.

Connected Line Identification Presentation and Restriction (COLP and COLR)

The Connected Line Identification Restriction (COLR) supplementary service takes precedence over the COLP supplementary service. The COLP service can take precedence over COLR service if the calling user has an override category.

The same Class of Service is used to control both Connected Line Identification Restriction and Calling Line Identification Restriction (CLIR). Thus, if a user has presentation restricted configured, their number is sent to the other party for both incoming and outgoing calls with the presentation flag set to restricted.

Coordinated Dialing Plan (CDP)

A Coordinated Dialing Plan (CDP) can be used to access an Australia ETSI trunk. However, neither the CDP private plan nor the CDP numbering type is supported. They get converted to unknown plan and type, respectively.

Virtual Network Services (VNS)

It is not possible to configure an Australia ETSI D-channel as a VNS D-channel. However, the voice connection through the Public Exchange of a VNS call may use a PRI/BRI COT or DID as a virtual TIE trunk.

Feature packaging

There are no new software packages required for this feature. However, the following packages are necessary in order to connect the Meridian 1 over an Australia ETSI PRI2 interface to a Central Office.

For PRI2 connectivity:

- Integrated Services Digital Network (ISDN) package 145
- 2.0 Mbps Primary Rate Interface (PRI2) package 154
- Overlap Signaling (OVLP) package 184
- International Primary Rate Access (IPRA) package 202
- Multi-Purpose Serial Data Link (MSDL) package 222

If the call is to interwork with any other trunk, the Universal ISDN Gateway (UIGW) package 283 is required.

For the Advice of Charge capability:

- Controlled Class of Service (CCOS) package 81
- Background Terminal (BGD) package 99
- Periodic Pulse Metering/Message Registration (MR) package 101
- International Supplementary Features (SUPP) package 131

For the Malicious Call Trace capability:

- Controlled Class of Service (CCOS) package 81
- Malicious Call Trace (MCT) package 107
- International Supplementary Features (SUPP) package 131
- Flexible Features Code (FFC) package 139
- Network Attendant Service (NAS) package 159
- ISDN Supplementary Features (ISDN INTL SUP) package 161

Feature operation

No operating procedures are required to use this feature.

Automatic Redial

Automatic Redial (ARDL) extends the redialing capabilities of the Ring Again and Network Ring Again features. The redialing capabilities of this feature reside at the system level. The system generates redialing attempts that allow the calling party to redial a busy public network subscriber using analog or digital trunks.

This feature is applicable when a calling party dials a public network subscriber number and receives a busy indication. Instead of attempting repeated redial efforts, the calling party can activate ARDL by pressing the Ring Again (RGA) key.

Once activated, the ARDL feature requests the Meridian 1 to automatically redial the attempted dialed number until a successful call termination is completed or until the configured number of redial attempts is reached. A successful call termination is determined when one of the following occurs: a tone detector attached to the call detects a ringback tone, an answer signal is received or an ISDN signaling trunk indicates call termination.

When a successful call termination is detected from the far end, the calling party hears the called party through the set's loudspeaker. The calling party must accept the redialed call within a specified time limit. If not, the redialed call is dropped and not redialed.

Multi-Automatic Redial permits simultaneous activation of the Automatic Redial feature on several RGA keys. This allows more than one number to be redialed in succession. Each Automatic Redial call is attempted once and then another number is attempted. Multi-ARDL numbers are dialed in order of activation.

One set of ARDL calls can be associated with one DN key. Another set of ARDL calls can be associated with a different DN key. This option facilitates the use of ARDL feature by a secretary who works for several managers. Each manager's DN could be on the secretary's set. A secretary activates the ARDL feature to call different calling groups on both DNs. After a successful call termination, the accepted call is easily accessed by the appropriate manager.

All ARDL requests are associated with the calling party's DN key. Therefore, when the called party is being redialed the calling party's DN key is busy. If the calling party is busy on another DN, the ARDL attempts are redialed on hold. When a successful call termination is completed, the Meridian 1 alerts the calling party by buzzing the set. While ARDL is activated, the calling party's set can be used for incoming/outgoing calls.

ARDL can be activated on a call that has originated from a Single Call Ringing (SCR), Single Call Non Ringing (SCN), Multiple Call Ringing (MCR), Multiple Call Non Ringing (MCN), Private Line key or Hot Line key. The ARDL request is associated with the key from which the call was made. If this key is free, the system attempts to dial the number until a successful call termination is detected and provided a free trunk is available.

Operating parameters

The Ring Again feature must be enabled to operate the ARDL feature.

This feature is only supported on Central Office (CO) and TIE trunks.

The ARDL is supported on Meridian 1 proprietary sets, excluding the M3000 and M2317 sets. It is recommended that sets be equipped with display, handsfree and loudspeaker. Analog (500/2500) type sets do not support this feature.

The ARDL feature cannot be activated on data calls.

ARDL can only redial if the Directory Number (DN) key on the calling party's set is idle. For this feature application, only a single external number can be stored against the Ring Again key.

Network Ring Again features do not interfere with the ARDL feature. ARDL is only activated after all Network Ring Again attempts have failed. When ARDL is activated, redial attempts continue with the ARDL feature. ARDL does not support the failure of a DPNSS1 call attempt.

The ARDL feature does not impact the operation of the Ring Again feature on internal calls.

The tone detector is not allocated to detect non-busy tones for off network trunks that have on-board busy tone detectors such as an Extended Flexible Central Office Trunk (XFCOT). Only a busy tone is detected. Accordingly, an Automatic Redial call is considered a successful call even though an overflow tone is sent from the far end.

With the exception of trunks that have on-board busy detectors or an end-to-end Integrated Services Digital Network (ISDN) call, a tone detector is required for all ARDL calls.

If a trunk is not equipped with answer supervision, an ARDL call is redialed once only and then the redial request is cancelled.

The busy tone detector capability is limited to the current tone detector hardware.

This feature introduces the following three timers that control the operation of ARDL:

- The Automatic Redial Acceptance Timer is the maximum allotted time that the calling party has to respond to an ARDL call;
- The Automatic Redial Retry Timer controls the time between successive ARDL retries; and
- The Tone Detector Response Timer controls the tone detector response and is defined in LD 16.

Feature interactions

Access Restrictions

Trunk Group Access Restrictions

The Access Restriction/Trunk Group Access Restrictions of an ARDL redialed call are those restrictions that were applied when the call was initiated. These initial restrictions are not changed.

Attendant Barge-In

Attendant Barge In is not allowed to a trunk that is currently used for the ARDL call redialing. This is done to avoid creating a conference when the tone detector is involved.

Attendant Break-In
Attendant Busy Verify

Attendant Break-In and Attendant Busy Verify are not permitted on a Meridian 1 proprietary set that is used for an ARDL call. These restrictions avoid creating a conference when the tone detector is involved in the call.

Attendant Blocking of Directory Number

An ARDL redialed call is blocked from the calling party if an attendant uses the Attendant Blocking of Directory Number feature on the calling party's DN.

Attendant Recall
Call Park
Call Transfer
Conference
No Hold Conference
Privacy Release

When an Automatic Redial (ARDL) call is not accepted by the calling party, the following keys are ignored if pressed: Attendant Recall (ARC), Call Park (PRK), Call Transfer (TRN), Conference (A03 or A06), No Hold Conference (NHC) and Privacy Release (PRS).

Autodial

ARDL can be activated on a dialed number using the Autodial (ADL) key.

Automatic Line Selection

Manual Line Selection, Outgoing Line Selection or Prime Line Selection is interpreted as accepting the ARDL by the calling party.

Automatic Preselection of Prime Directory Number

If a call is processed on key 0 and the calling party lifts the handset and selects the Prime Directory Number (PDN), this is interpreted as accepting a redialed call.

Automatic Set Relocation

If the calling party's set is relocated, the ARDL request is cancelled.

Call Detail Recording

The calling party's DN is charged even though a call is not accepted. This occurs because the resources are booked for ARDL attempts.

If Call Detail Recording (CDR) is configured on external calls, additional CDR records are produced. This occurs because each redial attempt produces a CDR record.

Calling Party Privacy

The calling party and called party have the same Calling Party Privacy considerations.

Digit Display

Dialed numbers are displayed when the 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.

Directory Number - Multiple Appearance

An ARDL call from a Single Call Ringing (SCR) or Single Call Non Ringing (SCN) is only redialed when all sets that have the same DN are free.

An ARDL call from a Multiple Call Ringing (MCR) or Multiple Call Non Ringing (MCN) is only redialed when the originating key is free.

Enhanced Hot Line

An 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.

Last Number Redial

An ARDL call can be activated on a number dialed using the Last Number Redial (LNK) key or by pressing the DN key twice. The ARDL number is saved as the last number redialed.

Line Load Control

ARDL attempts are controlled and restricted by Line Load Control.

**Network Alternate Route Selection
Network Speed Call**

ARDL can be activated on a Network Alternate Route Selection DN or Network Speed Call.

New Flexible Code Restriction

ARDL calls must pass New Flexible Code Restriction (NFCR) checks. If the redialed number is restricted, the ARDL request is cancelled.

Override

An ARDL call cannot be overridden. This is done to avoid creating a conference when a tone detector is involved.

Pretranslation

ARDL can be activated on a number that has passed the Pretranslation process. However, on an ARDL call the Pretranslation process is not used.

Privacy

If the ARDL call is redialed on a number that is shared with any single line telephone, the ARDL call is accepted when the single line telephone goes off-hook.

Privacy Override

When the Privacy Override feature is activated on the MADN key and the one set activates ARDL, this call can be accepted by other sets.

Private Line Service

ARDL call can be activated on a Private Line Service key. The call can only be redialed when the calling party's PVR or PVN key is free.

R2 Multifrequency Compelled Signaling

A successful ARDL call dialed through a R2 Multifrequency Compelled Signaling (MFC) trunk is determined by the tone detector (TDET) and MFC. If MFC signaling detects that the call has failed, the ARDL call is cancelled in the same manner as a TDET. If R2 MFC does not detect a call failure a TDET is connected to the call as a regular ARDL call.

Scheduled Access Restrictions

The Scheduled Access Restrictions (SAR) on ARDL redialed calls are set when the call is initiated. If restrictions are changed later, the prior restrictions still apply.

Speed Call

System Speed Call

Stored Number Redial

The Automatic Redial (ARDL) feature can be activated on a call using Speed Call (SCL), System Speed Call (SSU/SSC) or Stored Number Redial (RDL) keys.

Speed Call on Private Lines

The ARDL feature is activated on a number dialed using the Private Line (PVR/PVN) key and then making a speed call by pressing the Speed Call (SCL) key.

Feature packaging

Automatic Redial (ARDL) requires the following packages:

- Automatic Redial (ARDL) package 304;
- Ring Again (RGA) package 1; and
- Tone Detector (TDET) package 65.

Outpulsing of Asterisk and Octothorpe (OPAO) package 104 and Automatic Redial (ARDL) package 304 are mutually exclusive. The ARDL package is turned off automatically if both packages are equipped.

Feature operation

Enabling Automatic Redial

- 1** Press an idle DN key, handsfree key or go off-hook. Dial desired public network number. The DN lamp is lit.
- 2** The calling party hears busy, overflow, ringback tone, etc.
- 3** Press the Ring Again (RGA) key.
 - For a non-ISDN call, the RGA key lamps lights, the tone stops and the DN lamp goes dark.
 - For ISDN call, the RGA key lamps initially flashes while the system attempts to activate the Network Ring Again features. If these features cannot be activated, the RGA lamp is steadily lit and the ARDL feature is activated.

Automatic Redialing - External Number

After ARDL is activated the following possibilities can occur:

- 1** The call is answered by the called party and accepted by the calling party. In this case, both parties are connected.
- 2** The call is answered by the called party, but the calling party does not accept the call within the predefined time limit. The call is dropped and not redialed again.
- 3** The called party's number is occupied. The call is dropped and redialed later according to the timer configuration.
- 4** The call is blocked for some reason such as trunks or network congestion. The call is dropped and redialed later according to the timer configuration.

If the calling party presses another DN key while ARDL is activated, the ARDL attempt continues redialed on hold. If the Release (RLS) key is pressed when an ARDL call is attempted and is the active call, the redial attempt is dropped and redialed later after the predefined allotted time.

Automatic Redial - Originating Telephone is Idle on Another Number

- 1 The calling party's DN is idle.
- 2 The RGA key lamp winks. The DN lamp is lit and the called number appears on the set's display. A connection is made to the loudspeaker of calling party's set.
- 3 One of the following occurs:
 - If the called party's number is busy, the call is dropped when a busy tone is detected. The DN goes dark, the RGA lamp is steadily lit and the loudspeaker is deactivated.
 - Otherwise, the calling party hears ringback or answer through the loudspeaker.
- 4 If the calling party accepts the call by going off-hook, pressing DN key or pressing the handsfree, both parties are connected and the RGA lamp goes dark.
- 5 If the calling party does not accept the call within the predefined time limit, the DN key goes dark and one of the following occurs:
 - If the called party answers the call the following occurs: the RGA key goes dark, the redial call is disconnected and the ARDL request is cancelled. The display returns to an idle state.
 - If the called party does not answer the call, the following occurs: the call disconnected and the RGA lamp is steadily lit. The ARDL request is not cancelled and is ready for another redial attempt. The display returns to an idle state.

Accepting an Automatic Redial Call

To accept an ARDL call, the calling party performs one of the following:

- 1 If the ARDL call is the active call on the set, then the calling party must lift handset, press handsfree or press the key on which the ARDL call is active; or
- 2 If the ARDL call is dialed on hold the calling party must press the associated DN key.

Cancelling Automatic Redial

Automatic Recall is canceled in the following cases:

- The calling party presses the lit or winking RGA key (the cancellation request can be activated between ARDL attempts or during a redial attempt);
- The calling party accepts the ARDL call;
- The ARDL call is redialed the predefined number of call retries; or
- The calling party does not accept a successful ARDL call within the predefined time limit.

At cancellation of the ARDL request, the RGA key lamp goes dark.

Backup D-channel

In order to increase the reliability of the D-channel and enhance the serviceability of the Primary Rate Interface, a second or "backup" D-channel has been implemented.

The Backup D-channel provides redundancy for the D-channel Handler Interface (DCHI). The DCHI provides the signaling and protocol for call set-up, tear down and feature activation. The B-channels can either be PRI B-channels or virtual B-channels using analog or digital trunks with the ISDN Signaling Link (ISL) feature. Because the DCHI is so important to trunking requirements, an additional DCHI can be configured so that automatic switchover to a back-up occurs in case of failure. This configuration requires coordination with the far end to ensure that both ends have backup D-channels configured.

When Back-up D-channel is configured, one D-channel is active and the other one acts as a backup. Should the active D-channel fail, the auto-recovery software first attempts to recover the primary D-channel. If the recovery is successful, the D-channel goes back in operation. If the recovery does not take place, Meridian 1 software switches the D-channel processing to the backup D-channel on another link. If the active back-up D-channel fails after the problem with the primary D-channel has been resolved, the auto recovery software automatically switches back to the primary D-channel. During the switchover procedure, active calls remain intact, transient calls, however, may be dropped.

As an option, when the primary D-channel is brought from the "Released" to the "established" state, automatic changeover back to the primary D-channel can be activated.

The Backup D-channel requires a separate PRI card on another carrier link. For analog ISL applications, a separate NTAK02 circuit card, modem, and NTAK19BA cable are required.

Note: The backup D-channel must be the same D-channel type as the primary D-channel. That is, both must be configured as either DCHI or MSDL in software.

Interactions and Limitations

In X11 Releases 14 through 16, the Back-Up D-Channel feature is only supported between Meridian 1/SL-1 systems, and can be configured for either ISL or PRI applications. With X11 Release 18 and later software, the backup D-channel provides connections from Meridian 1 to the following systems:

- Meridian 1
- DMS-250 (BCS 32 and later)
- SL-100
- AT&T 4ESS
- DMS-100 (BCS32 and later)

In X11 Release 18 and later, the backup D-channel conforms to the Meridian Customer Defined Networking (MCDN) protocol and AT&T's TR41449 and TR41459.

Package Requirements

This option requires the following feature packages:

Feature	Package Number
ISDN	145
PRI2 or ISL	154 or 147

Basic Call Service

Description

ISDN basic call service transmits the ISDN call and consists of call progress signaling, voice and data transmission, and dialing plans.

Call Progress Signaling

PRI supports 64 Kbps out-of-band signaling on the D-channel for:

- call set-up
- call tear down
- feature activation
- local busy and reorder tones

Messages to generate tones are transmitted on the D-channel. Both out-of-band messages and in-band tones are provided for ringback.

Voice and Data transmission

Voice and high speed (64 kbps) data are transmitted on the B-channel and assigned on a per call basis. The following modes of transport are available:

- 64 Kbps circuit switched voice and data transmission
- 64 Kbps packet data transmission

Dialing plans

The following dialing plans are supported:

- Coordinated Dialing Plan (CDP) of 3 to 10 digits
- North American 10-digit numbering plan (E.164)
- Uniform Dialing Plan (UDP) which includes the Electronic Switched Network (ESN) 7-digit private numbering plan with 3-digit NARS location codes.

Typically, the numbering plan for a private network consists of a 3-digit location code (such as the ESN number) and a 4-digit extension. This allows the same extension to be used for private networks and for Direct Inward Dialing (DID) from the public network.

The following variations apply in the United Kingdom:

- group dialing, a hybrid of the coordinated and uniform plans
- mixed-length CDP network

Interactions and Limitations

The basic call service capability is supported for the following configurations:

- Meridian 1 Option 11 to Meridian 1/SL-1
- Meridian 1 Option 11 to AXE-10 central office switch
- Meridian 1 Option 11 to SYS-12 central office switch
- Meridian 1 Option 11 to SwissNet central office switch
- Meridian 1 Option 11 to 1 TR 6 D-channel signal protocol

Package Requirements

This option requires the following feature packages:

Feature	Package Number
ISDN	145
PRI2 or ISL	154 or 147

Bearer Capability in Call Detail Recording

The Bearer Capability in Call Detail Recording feature allows the operator to use bearer capability information to apply different rates based on the type of Call Detail Recording (CDR) calls for trunks.

ISDN calls provide information on the bearer capabilities used for each trunk call. Bearer capability information also gives the trunk characteristics for each call such as rate, voice or data dedicated, and packet or circuit mode transfer.

The Bearer Capability in CDR feature applies for CDR tickets involving trunks only. For non-trunk calls, such as set-to-set calls, blank characters are output. For all trunks not listed below, 99 is output in the CDR.

The Bearer Capability in CDR feature supports the following trunks:

- EuroISDN trunks
- Asia Pacific trunks
- MCDN trunks
- QSIG trunks
- DTI and DTI2 trunks
- DASS2 trunks and DPNSS1 trunks
- North American ISDN trunks

Table 3 shows the new bearer capability codes printed in the CDR ticket and their meaning.

Table 3
Bearer capability information printed in the CDR

Code	Meaning
01	Circuit mode speech
02	Circuit mode 3.1 kHz
03	Circuit mode unrestricted 64 kbit/s digital information transfer
04	Circuit mode unrestricted 64 kbit/s digital information transfer rate adapted from 56 kbit/s
05	Packet mode unrestricted digital information transfer
06	Circuit mode 7 kHz audio or videotelephony
07	Circuit mode restricted 64 kbit/s digital information transfer
08	Circuit mode video
99	Unknown or non-existent

Bearer Capability in the Call Detail Recording for ISDN

For ISDN protocols, the bearer capability is contained in the BCAP Information Element (IE). The IE is a mandatory field of the SETUP message sent to initiate a call. The correspondence between the ISDN BCAP IE and the code for CDR is shown in Table 4. When the received BCAP fields do not match the values shown in Table 4, 99 is output.

Table 4
Correspondence between the ISDN BCAP Information Element and the CDR code

Code	Byte 3 bits 4-0 information transfer capability	Byte 4 bits 6-5 transfer mode	Byte 5 bits 4-0 user information layer 1 protocol	Meaning of CDR code
01	00000	00	/	Circuit mode speech
02	10000	00	/	Circuit mode 3.1 kHz audio
03	01000	00	/	Circuit mode unrestricted 64 kbit/s digital information transfer
04	01000	00	00001 or 01000 <i>Note:</i> See byte5a, bits 4-0, user rate must be 01111 (56 kbit/s).	Circuit mode unrestricted 64 kbit/s digital information transfer rate adapted from 56 kbit/s
05	01000	01	/	Packet mode unrestricted digital information transfer
06	10001	00	/	Circuit mode 7 kHz audio or videotelephony
07	01001	00	/	Circuit mode restricted 64 kbit/s digital information transfer
08	11000	00	/	Circuit mode video

Bearer Capability in the Call Detail Recording for DASS/DPNSS

For DPNSS and DASS2 protocols, the bearer capability is contained in the Service Indicator Code (SIC). The SIC is a mandatory field of the Initial Service Request Message (ISRM) message sent to initiate a call. It contains at least one byte and a maximum of two bytes. Table 5 shows the correspondence between the DPNSS SIC and the CDR code. Table 6 shows the correspondence between the DASS SIC and the CDR code. When the received SIC fields do not match the values shown in Tables 5 and 6, 99 is output.

Table 5
Correspondence between the DPNSS Service Indicator Code and the CDR code

Code	Byte 1 bits 6-4	Byte 1 bits 3-0	Byte 2 bits 2-0	Byte 2 bits 6-4	Meaning of CDR code
01	001	/	/	/	Circuit mode speech
02	Not supported				Circuit mode 3.1 kHz audio
03	010	0000	/	/	Circuit mode unrestricted 64 kbit/s digital information transfer
04	Not supported				Circuit mode unrestricted 64 kbit/s digital information transfer rate adapted from 56 kbit/s
05	010 or 011	/	100	x1x	Packet mode unrestricted digital information transfer
06	Not supported				Circuit mode 7 kHz audio or videotelephony
07	Not supported				Circuit mode restricted 64 kbit/s digital information transfer
08	Not supported				Circuit mode video

Table 6
Correspondence between the DASS2 Service Indicator Code and the CDR code

Code	Byte 1 bits 6-4	Byte 1 bits 3-0	Byte 2 bits 2-0	Byte 2 bits 6-4	Meaning of CDR code
01	000	/	/	/	Circuit mode speech
	001	0000 or 0010	/	/	
02	001	>=1000 not 1111	/	/	Circuit mode 3.1 kHz audio
03	010	0000 if 2 bytes	/	/	Circuit mode unrestricted 64 kbit/s digital information transfer
04	Not supported				Circuit mode unrestricted 64 kbit/s digital information transfer rate adapted from 56 kbit/s
05	010 or 011	/	100	xlx	Packet mode unrestricted digital information transfer
06	Not supported				Circuit mode 7 kHz audio or videotelephony
07					Circuit mode restricted 64 kbit/s digital information transfer
08					Circuit mode video

Bearer Capability in the Call Detail Recording for DTI/DTI2

For DTI and DTI2 trunks, the bearer capability is set during the call processing according to the value entered at prompt DSEL of the route data block in overlay 16.

For both DTI and DTI2 trunks:

- If it is a voice call, 01 is output in the CDR.
- If it is a data call, 03 is output in the CDR.

Additionally, for DTI2 trunks, if the call is voice and data, 99 is output.

Bearer Capability in the CDR ticket

The new CDR format is composed of three lines. Each printed item is assigned a line number and a location on that line. If features are restricted or not equipped, the corresponding field in the CDR record is filled with spaces. The first line is 87 characters long; the second line is 86 characters long.

The bearer capability information is introduced at the end of the third line. Prior to the introduction of the Bearer Capability in CDR feature, this line had 48 characters. With the Bearer Capability in CDR feature, the length is increased by three characters. The bearer capability information is located at position 49-50.

Table 7 shows the new format of the third line of CDR tickets with the BCAP field in **bold**.

The BCAP field is filled with blank characters if:

- the calls involves sets only, or
- the new BCAP prompt is set to NO (In LD 15, Type = CDR; CDR = YES; BCAP = NO.)

The bearer capability information is present in all CDR tickets involving trunks.

Table 7
Third line of the CDR new format

Position	Content	Description
1	blank	
2	&	Third line of TTY
3 - 13	time to answer	Time To Answer ringing/total waiting
14	blank	
15 - 30	blid	Billing Line IDentification
31	blank	
32 - 38	cab	M911 Call Abandon tag
40 - 42	cdr100h	100-Hour Duration
43	blank	
44 - 45	npi	CLID Numbering Plan Identification
46	blank	
47	ton	CLID Type of Number
48	blank	
49 - 50	BCAP	Bearer Capability
51	blank	

Operating parameters

The Bearer Capability in CDR feature is supported on X11 Release 24 and later. It requires the new CDR format (that is, FCDR = NEW in LD 17).

If both the originating and terminating sides are trunks, the bearer capability information is taken from the incoming trunk.

If an initialization (INIT) occurs during an established trunk call, the bearer capability information is lost and 99 is output.

Feature interactions

ISDN Bearer Capability Information Element

The Bearer Capability in CDR feature does not change the bearer capability functionality; the bearer capability information in the CDR record is read from the message Call Register where it is stored. If a particular bearer capability is not supported by Meridian 1, it will not be displayed in the bearer capability field in the CDR.

DASS/DPNSS Service Indicator Code

The Bearer Capability in CDR feature does not change the Service Indicator Code (SIC) functionality; the bearer capability information in the CDR record is read from the channel block where it is stored. If a particular SIC is not supported by the Meridian 1, it will not be displayed in the bearer capability field in the CDR.

Feature packaging

Bearer Capability in CDR requires these packages:

- Call Detail Recording (CDR) package 4
- Call Detail Recording on Teletype Terminal (CTY) package 5
- New Call Detail Recording (FCDR) package 234

Feature operation

There are no specific operating procedures required to use this feature.

Break-in with secrecy

Description

The Break-in With Secrecy (BKIS) feature enhances the operation of the Break-in (BKI) feature.

When the break-in conference is established between the attendant and the desired and undesired parties of a call, the attendant may press the BKI key again to exclude the undesired party and talk directly to the desired party. Once the undesired party is excluded, intrusion tone is no longer provided.

Interactions and Limitations

This feature has the following limitations:

- All conditions for existing NAS Break-in must be satisfied.
- All nodes within the network must have compatible software with identical Multi-Party Operation (MPO) options. (Otherwise, MPO options in the desired party's node take precedence.)
- Where the undesired party resides on a different node from the desired party, MPO treats the undesired party as an external caller on the desired party's node.
- All nodes connected to the attendant must have Network Break-in with Secrecy equipped, for the BKIS feature to operate properly.
- Where only one party is excluded from a conference, intrusion tone is provided this last party, but music is not.

Break-in indication and prevention

Description

For reasons of confidentiality, some customers may not want to allow attendant break-in on external call connections (that is, call connections involving off-network trunks). Alternately, they may require that the attendant be aware that a call connection is external before performing a break-in.

This feature provides two options, described below. Either option can be selected for a customer, but not both.

Break-in indication

With the Break-in Indication option, if an attendant dials a busy extension during a break-in operation the attendant display provides one of two customer-defined indications:

- three dashes, appended to the end of a set of displayed digits, if the busy station is involved in an external call (this is the BBIN option), or
- a mode digit, appended to the end of a set of displayed digits (this is the EBIN option)

In an ISDN PRI environment, the mode digit indicates one of these states:

- 1 = Station is busy on an off-net call, or involved in a conference call.
- 2 = Station is busy with on-net call, and is not involved in a conference call.
- 3 = Station is busy on a non-established call; for instance, dialing, ringing, or announcement.
- 4 = Station is in line lockout.

The meanings of mode digits in a non-ISDN environment are described in 553-2301-105, Appendix 2.

Break-in prevention

With the Break-In to External Call Denied option (BIXD option), break-in to a party involved in an external call is temporarily denied. This applies to both pre-dial and post-dial break-in operations.

Interactions and Limitations

This feature has the following limitations:

- All conditions for existing NAS Break-in must be satisfied.
- All nodes within the network must have compatible software with identical Multi-Party Operation (MPO) options. (Otherwise, MPO options in the desired party's node take precedence.)
- Where the undesired party resides on a different node from the desired party, MPO treats the undesired party as an external caller on the desired party's node.
- All nodes connected to the attendant must have Network Break-in with Secrecy equipped, for the BKIS feature to operate properly.
- Where only one party is excluded from a conference, intrusion tone is provided this last party, but music is not.
- Indication of station status on a call to a busy station is available only on attendant consoles.

Feature interactions for the Break-In and Network Attendant Service Break-In features also apply to this feature.

With the Break-In to Line Lockout Set product improvement, the appropriate busy indication is supported. If the Break-In to Line Lockout Set Denied option (option BLD) was set for the customer, attempts at break-in results in Temporarily Denied—1 status. If the Break-In to Line Lockout Set Allowed option (option BLA) was set then break-in proceeds normally.

If the Attendant Busy Display Allowed option of the First-Second Degree Busy Indication feature is chosen in overlay 15 (option ABDA), display of digits could be changed. When the attendant dials a second-degree busy station, the attendant digit display shows —0. In combination with the Break-in Indication and Prevention feature, the display may show —0— — or —0 followed by another dash and a mode digit.

Package Requirements

The feature options can be selected for a customer by way of overlay 15. Respond to the OPT prompt with the desired option:

BIXA Break-in to external calls allowed

BIXD Break-in to external calls denied

BIND Break-in indication denied

BBIN Basic break-in indication allowed

EBIN Extended break-in indication allowed

Options BIXD, BBIN and EBIN are mutually exclusive. Only one of these options can be selected for a customer.

Call charging under 1 TR 6 connections

Description

Accumulated charging information for an ISDN PRI call is interfaced into the Meridian 1 periodic pulse metering function, to provide a transparent call-charging feature to the customer.

On a node with 1 TR 6 connection, call charge information is received by the SL-1 from the network as part of the connect data. This information is temporarily stored by SL-1. Further charge information from the network is added to the charges being stored. When the call has been completed, the information is used to add call charges to the calling user's meter. (The calling user could be an analog (500/2500 type) telephone, an attendant console or a trunk in a tandem call.)

Call charging under 1 TR 6 supports:

- recording of accumulated call charging information for each call in the CDR (if equipped)
- calculation of total charge for each call based on assigned unit cost, and the accumulated call charging information received over the network. (This information is also recorded on CDR.)
- attendant access to the accumulation of call charge units, on a per call basis, by way of call marking

- SL-1 telephone set access to MR data (on a set with digit display and an MRK key)
- CDR on Multiple Call Transfer for outgoing calls

Interactions and Limitations

Call transfer

Call charging information is counted against a station connected to a 1 TR 6 central office (CO) trunk. If that station transfers the call while the call is still ringing, call charging information is counted against the transferring station until the call is answered or abandoned. Once the call is answered, call charging information begins to count against the station the call was transferred to.

If the transferring station consults with the station receiving the transfer first, call charging information is only counted against the transferring station until the transfer actually takes place. From that point on, continuing call charge information is counted against the station receiving the transfer.

If the transferred call is redirected (for instance, call forwarded or call hunted) call charges accumulate against the transferring station until the transfer is complete and the call is answered. After that point, new call charge information is counted against the station at which the call has been answered.

Call forward all calls

In the transfer or extension of a metered call, charges are counted against the last station at which the call the answered and the controlling station Releases.

The last station to forward a call onto a 1 TR 6 trunk receives the charge for that call.

Call forward no answer

In the transfer or extension of a metered call, by way of Call Forward No Answer, charges are counted against the last station at which the call the answered and the controlling station Releases.

Call pickup

A station which picks up a metered call transferred or extended to another station receives the charge for the call.

Call park

A station which parks a metered call will continue to be charged for that call until it is answered by another station.

Hunting

If a call is transferred or extended to a station and redirected by hunting to another station, the answering station receives the charge for the call.

Camp on

Calls camped on to a busy station by an attendant are charged against the attendant until the call is answered and the attendant Releases.

Conference 3-party/6-party

When a 1 TR 6 trunk is added to a conference call, call charges from the network are counted against the station which initiated the trunk call.

If the station which originated the trunk call then disconnects from the conference, call charges for the 1 TR 6 connection cease against that station. Charges for the trunk connection begin to count against whichever remaining member of the conference has been in the longest.

Conference—attendant

If an attendant dials a 1 TR 6 CO trunk as a source, while establishing a conference call, call charging information received from the network is counted against the attendant.

If the attendant holds the conference at the console by pressing the **RLS** key then the call charging information continues to count against the attendant.

If the attendant Releases the conference from the console by pressing the **RLS SRC** key and then the **RLS** key, continuing call charge information is counted against the station which has been in conference the longest.

If an attendant is connected to an internal station (or a TIE trunk) first, any call charging information incurred from a 1 TR 6 trunk connection established afterwards counts against the internal station (or TIE trunk), including a conference connection.

Attendant originated calls

When the attendant originates an external call using a 1 TR 6 CO trunk, call charging information from the network is held temporarily by the SL-1. If the attendant transfers the call to another station, the call charges are counted against the station receiving the transfer. If the attendant disconnects, thus effecting a transfer, the call charges count against the attendant.

Attendant extended calls

If an attendant dials an external call through a 1 TR 6 CO trunk while connected to an internal station or TIE trunk, call charging information received from the network is held by the SL-1. Cumulative call charges are counted against the internal station or TIE trunk for the entire length of the call. However, these charges are not placed against the station until that station either disconnects or modifies the call. (Attendant extension of the call does not initiate the assignment of charges.)

If the attendant dials the 1 TR 6 trunk before connection with an internal station, call charges are counted against the attendant. Call charges continue to count against the attendant until a dialed station DN or TIE trunk access code answers (or end of dialing occurs on the TIE trunk), and the attendant Releases. At that point the accumulated charges are assigned to the attendant's meter. (If the attendant does not have a meter defined, the charges count against the customer meter.)

Attendant operation of metered calls

An attendant can flag a long-distance call as metered, in order to receive immediate call-charge information at the completion of the call. At call completion, the attendant is recalled and the calculated call charge is displayed on the console.

If the call is transferred, a meter recall routes to the attendant for each part of the trunk connection.

Marking metered calls

A meter (MTR) key on the attendant console allows the attendant to mark as a metered call any call using any outgoing 1 TR 6 CO trunk. Additional 1 TR 6 CO trunks added to the conference after that are automatically marked as "metered". To cancel metering the attendant pressed the meter key again (removing metering from all connected calls). The meter key has no effect on trunks other than 1 TR 6 trunks.

Meter recall

When a metered call is modified or disconnected a meter recall is presented to the originating attendant. The meter recall ICI lamp lights, the source side of an idle loop is lit, the destination lamp remains off and the information below appears on the console display:

- station DN or TIE trunk access code
- call charge

Consultation

A station creating a consultation connection between one or more metered trunks receives call charges for all the involved trunks, until that call is transferred.

Reading and changing call charge units

An SL-1 telephone set equipped with a digit display and an MRK key/lamp pair can read or change meter contents. A background terminal assigned “Meter” access can manually or automatically read, change or print meter contents.

CDR records

Call charge units for metered calls appear on CDR records (when both CDR and MR features are enabled for the customer). If the charge option is enabled (that is, the unit cost set to a non-zero value), then the call charge is calculated and recorded on CDR. (Zeros are printed when the charge option is disabled.)

Origid, terid, auxid in CDR records

An outgoing 1 TR 6 CO-trunk call made from a non-primary station DN is recorded in CDR with the primary DN as the ORIGID (or TERID after call modification). Where the metered call is made from an ACD agent, the position ID of that agent is printed as the ORIGID.

If the PDDN is a Multiple Appearance DN (and AXID has been set to YES in the customer data block) then the AUXID is also be printed.

CDR for answered calls

CDR records are not printed unless calls are answered.

Call Connection Restriction

Description

The Call Connection Restriction product improvement allows limiting conditions to be placed on call connections across the ISDN. Call configurations which would degrade transmission integrity or network performance are prevented.

Interactions and Limitations

The following conditions are placed on network call connections:

- No more than one trunk without disconnect supervision can be used in a call connection. Otherwise, trunk lock-up could occur within the network. (This also applies to call-joined connections. Two call connections cannot be joined if each makes use of a trunk without disconnect supervision.)
- Tandem nodes are limited, to prevent potential transmission problems. The maximum number of tandem nodes to be allowed in a call connection can be set between 0 and 31, by way of service change.
- PSTNs can be limited. If so specified, only a single PSTN is permitted in a call connection; or, an unlimited number of PSTNs can be allowed.
- μ /A-Law conversions are limited, to prevent potential transmission problems. The maximum number of conversions to be allowed in a call connection can be set between 0 and 31, by way of service change.
- Satellite delays are limited. The maximum number of Satellite delays to be allowed in a call connection can be set between 0 and 5, by way of service change.

These call limitations only apply within an ISDN environment. For this product improvement to be effective, ISDN connectivity must be available between the originating and terminating nodes.

With the use of multiple call transfers, conferencing and other manipulations, it is possible to bypass this product improvement's control of the number of unsupervised trunks in a call connection. In these situations, other ISDN call-connection limitations may also be overcome.

The Call Connection Restriction product improvement overrides the Satellite Link Control feature. Whereas the Satellite Link Control feature limited the number of Satellite access lines or intermachine trunks to one, the Call Connection Restriction product improvement allows this limit to be service-changeable. When calls are joined, it is possible to produce a call connection which violates some of the call-connection restrictions. Under these conditions, it is possible to exceed the limits on tandem nodes, μ /A-Law conversions and network call redirections.

Call Forward All Calls/No Answer

Description

Call Forward All Calls/No Answer enables callers to manually forward or forward on a no answer to any other station on the ISDN network.

Using this feature, the receiving location is provided with the dialed number, the calling number/name (if optioned) plus the reason for the redirection. The caller's display is also updated to show the name and number of the person the call was redirected to plus the reason for the redirection.

This feature is applicable to the following sets for Name and Number display:

- M3000 Touchphone
- M2317
- M2008
- M2616
- M2216

Package Requirements

This option requires the following feature packages:

Feature	Package Number
ISDN	145
PRI2 or ISL	154 or 147
DDSP	19
CPND	95

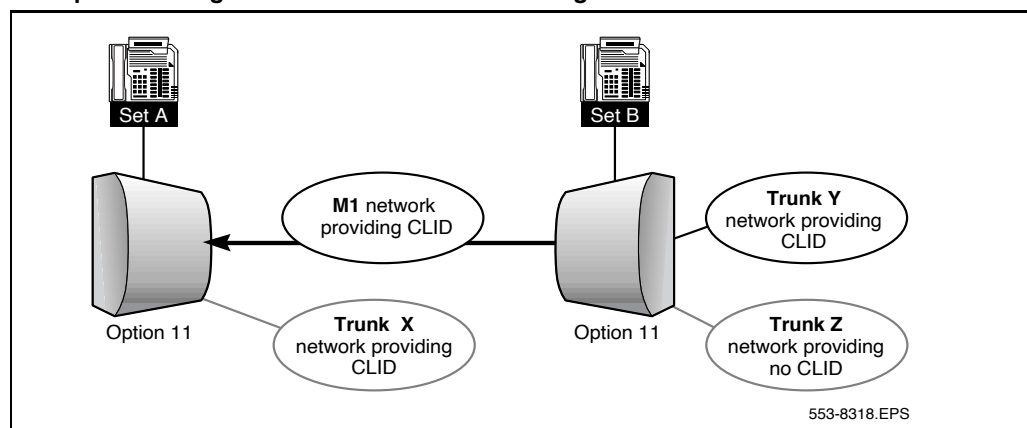
Call Forward, Break-In and Hunt Internal or External Network Wide

The Call Forward, Break-In and Hunt Internal or External Network Wide feature determines whether a call is treated as internal or external on a network wide basis. A call is treated as internal if it terminates and originates within a private network. A call is treated as external if it terminates or originates outside a private network. The definition of an incoming internal or external call is determined by the Call Forward by Call Type feature. The definition of an outgoing internal or external call is determined by the Break-In feature.

A network-wide call receives internal treatment if the Numbering Plan Identifier (NPI) in the Calling Line Identification (CLID) is private. Conversely, if the NPI is not present, the Network Attendant Services (NAS) information will be used if it is configured.

If neither the CLID, NAS, QSIG or DPNSS information is present, the following occurs. The route class mark defined in the route data block will be used to collect information necessary to determine whether a call should be given internal or external treatment. The network wide definition can be superseded by the route class. This can be configured on a route basis by entering local data at the “LOC” for internal treatment the at the new prompt “IDEF” prompt in the customer data block configuration (LD 15).

Figure 10
Example of Calling Line Identification for Incoming Calls



An incoming call is always treated as internal provided the originating party is not a trunk. However, if the originating party is a trunk, Calling Line Identification is present and NPI is private the call is treated as internal.

Figure 10 illustrates several examples of incoming call treatments. When Set B dials Set A and receives no answer the Call Forward by Call Type feature is activated from Set A's node. This type of call is treated as an internal call because the Number Plan Identification (NPI) is private in the Calling Line Identification (CLID).

When Trunk Y calls Set A and Set A does not answer, the Call Forward by Call Type feature is activated from Set A's node. The calling CLID initially received is NPI public or unknown. This is transported to Set A's node and the call receives internal treatment.

When Trunk X calls Set A and Set A does not answer, the Call Forward by Call Type feature is activated at from Set A's node. The calling CLID initially received is NPI public or unknown. The call receives external treatment.

With a private network inter-exchange signaling protocol QSIG, the CLID and NPI are used if they are present. If this information is not present, then the QSIG specific data giving information on the far end of the call is used. Since no NPI is provided by the DPNSS protocol, the Calling Line Category is used.

Operating parameters

The only features impacted by this feature are Call Forward/Hunt by Call Type, Internal Call Forward and Attendant Break-In. The Attendant Break-In feature will continue to treat conference calls as external. Network side conference is not considered.

This feature operates using the available information that is associated with a call. This information is decoded to determine different treatments for Break-In Indication, Call Forward/Hunt by Call Type or Internal Call Forward.

For outgoing calls, this feature functions by the information created from the terminating node back to the originating node.

QSIG calls are treated depending on the equivalent information to the NPI or to NAS type information transported.

Digital Private Network Signaling System #1 (DPNSS) does not support NPI. Alternatively, DPNSS supports NAS type information when connected to an attendant on the originator and the route (called Calling Line Category). This information will be considered when available. With Digital Access Signaling System #2 (DASS), Calling Line Category is not supported and calls will always be treated as external.

Internal Call Forward considers the transferred party and not the transferring party (both attendant and set) when the transferring party is on the treating node. Information transported across the network by NAS, DPNSS or QSIG from a network side transferred call is not considered. The information on the transferring party only will determine the type of treatment given to a call.

Feature interactions

Attendant Break-In

Attendant Break-In Indication and Prevention

Call Forward by Call Type

If the Internal/External definition in LD 15 is set to YES, a call is treated as internal or external on a network wide basis.

Call Forward, Internal

If a treated call is a transfer call and the transferring call is on the treating node, the transferred party will be considered. However, when the transferring party is not on the treating node, the transferring party will determine the treatment given.

Call Transfer

Network Call Transfer

Network Call Redirection

The treatment of a call following a call transfer (Call Forward/Hunt by Call Type) is based on the transferring set and the call originator's set. The set display on network call modification or redirection does not change.

Network Attendant Service

When a call is transferred, a new facility message is sent to the transferred party's node to transport the terminal indicator parameter or the access trunk information parameter.

DPNSS1
QSIG

Call Forward, Break-In and Hunt Internal or External Network Wide uses the NAS equivalent information that is transported on protocols such as Party Category and Progress Indicator for QSIG and Calling Line Category for Digital Private Network Signaling System No. 1 (DPNSS).

Feature packaging

Call Forward, Break-In and Hunt Internal or External Network Wide is included in base X11 system software.

NAS messaging requires Network Attendant Services (NAS) package 159.

DPNSS requires 2Mbit Primary Rate Interface (PRI2) package 154, Integrated Digital Access (IDA) package 122 and Digital Private Network Signaling System No. 1 (DPNSS) package 123.

QSIG requires QSIG package 263.

Feature operation

There are no operating procedures specified for this feature.

Call Forward/Hunt Override via Flexible Feature Code

Call Forward Override provides all telephone users (having a specific Class of Service) and attendants with the ability to override Intercept Computer Call Forward (ICP-CFW), Call Forward All Calls, Call Forward No Answer, Hunting and Make Set Busy by entering a Flexible Feature Code. In order to use this feature, the originating set must have Call Forward Hunt Allowed (CFHA) Class of Service. When this feature is enabled if override is attempted, and the called party is idle, the set is rung regardless of any diversion. If the dialed set is busy and has Hunt active, the calling party will terminate on the wanted set and receive a busy indication. Sets without Call Forward/Hunt Override denied (CFHD) Class of Service will not be able to use the Call Forward/Hunt Override (CFHO) Via Flexible Feature Code (FFC) feature.

Call Forward/Hunt Override Via FFC works in network environments with Meridian 1 nodes and Meridian Customer Defined Network (MCDN) links.

Operating parameters

The Call Forward/Hunt Override FFC can only be used in predial mode from a set (i.e., it has to be dialed before dialing the DN that has Call Forward All Calls, Intercept Call Forward, Call Forward No Answer, Internal Call Forward, Hunt, or Make Set Busy active).

The Call Forward/Hunt Override FFC can only be dialed from its own node (i.e., it has to be dialed before any trunk access code).

On an ABCD set the Call Forward/Hunt Override FFC can only be configured as a predial FFC (ABCD sets are a type of German telephone).

Call Forward/Hunt Override FFC can only be used against extensions with one of the following type: HOT/MCN/MCR/SCN/SCR/Basic Rate Interface (BRI) DNs and PBX sets.

It is not possible for BRI extensions to dial a Call/Forward Hunt Override FFC.

The Call Forward/Hunt Override via FFC feature can only be used in standalone and MCDN environments. If no MCDN links are involved, no information about Call Forward/Hunt Override will be passed on to other nodes.

To get the functionality of Call Forward/Hunt Override Via FFC in an MCDN environment these enhancements must be integrated in the originating node, terminating node and any intermediate nodes.

Feature interactions

Attendant Blocking of DN

Using Call Forward/Hunt Override FFC after activation of ABDN is not allowed. Any attempt will be canceled and overflow tone will be returned.

Automatic Call Distribution (ACD)

ACD DNs are not overridden by Call Forward/Hunt Override Via FFC. Any attempt will be canceled and access denied treatment will be returned.

Individual DNs on an ACD set are overridden by Call Forward/Hunt Override Via FFC with the same limitations as for other sets.

Barge-in
Busy Verify
Break-in

Using Call Forward/Hunt Override Via FFC after activation of Barge-in, Busy Verify or Break-in is not allowed. Attempts will be canceled and overflow tone will be returned.

Using post-dial Break-in after dialing the Call Forward/Hunt Override FFC is possible after encountering a busy set, if Break-in is enabled.

Basic Rate Interface (BRI)

BRI sets are not supported; any attempt to dial Call Forward/Hunt Override from a BRI set will be ignored and access denied treatment will be returned.

BRIT

BRI TIE trunks in a Meridian Customer Defined Network (MCDN) are supported.

Boss Secretary Filtering
Call Forward All Calls
Call Forward No Answer
Call Forward and Busy Status
Internal Call Forward/Make Set Busy

These features are overridden by the Call Forward/Hunt Override Via FFC feature, but there are no changes to the features themselves.

Call Transfer

A set can activate Call Forward/Hunt Override Via FFC when initiating a transfer. If the transfer is completed while ringing, the Call Forward/Hunt Override will still be active and passed on to the transferred party.

Call Waiting

Call Waiting can be used even if the Call Forward/Hunt Override Via FFC feature has been activated. When a busy set with Call Waiting configured is encountered, it will terminate on Call Waiting.

Call Waiting Redirection

There is no interaction with the Call Waiting treatment component of the Call Waiting Redirection feature. However, Call Forward/Hunt Override via Flexible Feature Code does override CFNA, and thus the CFNA treatment given to unanswered Call Waiting calls by the Call Waiting Redirection feature is overridden by the CFHO feature. The incoming call will continue to be given Call Waiting treatment as if the Call Waiting Redirection feature is disabled when the CFHO feature is enabled by the calling party.

Camp-on

When a busy set is encountered, it is possible to Camp-on to the set, even if Call Forward/Hunt Override Via FFC has been activated.

Digital Private Network Signaling System One (DPNSS1)

DPNSS1 is only supported as an incoming trunk transferred to a MCDN environment using Call Forward/Hunt Override Via FFC.

Direct Inward System Access (DISA)

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

Do Not Disturb (DND)

This feature is not overridden by the Call Forward/Hunt Override Via FFC feature.

Flexible DN (FDN)**External Flexible DN (EFD)**

It is not possible to store the Call Forward/Hunt Override FFC as an FDN or EFD.

Group Call

It is not possible to use the Call Forward/Hunt Override FFC as a Group Call DN.

Group Hunt

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.

Hunt

This feature is overridden by the Call Forward/Hunt Override Via FFC feature. If a set activating Call Forward/Hunt Override Via FFC encounters a busy set no hunt steps will be performed; the call will terminate on the DN and busy tone will be returned.

Hunt DN/External Hunt (EHT) DN

It is not possible to store the Call Forward/Hunt Override FFC as a Hunt or EHT DN.

Idle Extension Notification (IEN)

This feature can be used even if the Call Forward/Hunt Override Via FFC feature is activated. When a busy set is encountered, it is possible to place an IEN request against the set.

Intercept Computer (ICP) Call Forward

This feature is overridden by the Call Forward/Hunt Override Via FFC feature. The Call Forward/Hunt Override FFC replaces the ICP Override FFC.

Last Number Redial

The Call Forward/Hunt Override FFC and the dialed DN are stored under Last Number Redial.

Multiple Appearance Multiple Call Arrangements (MCAs)

Multiple Appearance Single Call Arrangements (SCAs)

If the Call Forward/Hunt Override FFC is used against an MCA (MCR/MCN) or SCA (SCR/SCN) DN it will override any active forward and terminate on all idle appearances. If all appearances are busy, busy treatment will be returned.

Primary Line Directory Number (PLDN)

It is not possible to store the Call Forward/Hunt Override FFC as a PLDN.

Phantom DN

This feature is not overridden by the Call Forward/Hunt Override Via FFC feature. If Call Forward/Hunt Override Via FFC is used against a phantom TN the call will be canceled and overflow tone will be given.

Phantom TN

This feature is not overridden by the Call Forward/Hunt Override FFC. If a Call Forward/Hunt Override FFC is used against a Phantom TN, the call will be canceled and overflow will be given.

Priority Override

It is possible to use Priority Override after using the Call Forward/Hunt Override FFC and encountering a busy set.

Radio Paging

If Radio Paging is activated in a call where Call Forward/Hunt Override has been used, the Call Forward/Hunt Override feature will be deactivated.

Ring Again**Network Ring Again**

Using the Ring Again feature is possible after using the Call Forward/Hunt Override FFC and encountering a busy signal. Ring Again can be placed against the set for which the Call Forward/Hunt Override FFC was used (i.e., the set with CFW active should be rung by the Ring Again feature).

Ring Again No Answer**Network Ring Again No Answer**

Using the Ring Again No Answer feature is possible after using the Call Forward/Hunt Override FFC and encountering an idle set that does not answer. Ring Again No Answer can be placed against the set for which the Call Forward/Hunt Override FFC was used (i.e., the set should be rung by the Ring Again No Answer feature).

Single Digit Access (SDA)

It is not possible to store Call Forward/Hunt Override FFCs in an SDA list.

Semi-automatic Camp-on (SACP)

This feature can be used even if the Call Forward/Hunt Override Via FFC feature is activated. When encountering a busy set, it is possible to activate SACP, if it is applicable.

Speed Call

The Call Forward/Hunt Override FFC cannot be stored in a speed call list.

Feature packaging

In a standalone environment the Flexible Feature Codes (FFC) software package 139 must be provisioned to activate this feature.

For network environments, Network Attendant Service (NAS) package 159 must also be provisioned. Attendant Overflow Position (AOP) package 56 must be restricted, as it is mutually exclusive with Network Attendant Service.

Feature operation

The user gets the same functionality in a Meridian Customer Defined Network (MCDN) as in standalone environments. The Call Forward/Hunt Override information is transmitted from the originating node to the terminating node using the Network Attendant Service (NAS) feature.

Activation of the service is call dependent; network-wide Call Forward/Hunt Override is part of the NAS feature.

To activate the Call Forward/Hunt Override feature, the user dials the FFC for Call Forward/Hunt Override and the DN of the wanted party. If the set is idle, the set is rung regardless of any diversion (e.g., Call Forward All Calls, Intercept Call Forward, Call Forward No Answer, or Hunt) or Make Set Busy on the set.

If the set(s) have displays, the display(s) are updated. If the display on the originating set is updated when the call is answered, the Call Forward/Hunt Override FFC will no longer be displayed.

If the dialed set is busy and Hunt is active, the calling party will terminate on the wanted set and will receive busy indication.

If the dialed set is idle, but does not answer within the defined number of ringing cycles for CFNA, the call is not forwarded (i.e., it continues to ring).

If the dialed set is busy, the attendant can activate Camp-on, if Camp-on is applicable. In addition, Ring Again can be placed against a set for which Call Forward/Hunt Override was used and a busy set was encountered.

Call Page Network Wide

Prior to X11 Release 22, call paging was only accessible if both the attendant or user and the paging trunk route were located on the same node. The Call Page Network Wide (PAGENET) feature expands call paging capabilities by allowing an attendant or user to access a paging trunk route located on a different node.

The PAGENET feature controls external paging access privileges with the following levels of access: Restricted, Controlled and Uncontrolled. On the paging trunk, trunks are assigned a level of access on a trunk route basis. On other network nodes, access privileges can be assigned to attendant consoles and sets.

If the paging trunk route at the paging node is configured as PAGENET Restricted, all external users are prevented from accessing the paging trunk route. Access attempts from an external location are given a defined intercept treatment by the paging node.

PAGENET Controlled allows limited access to external users meeting the following criteria:

- the Attendant Console or set is programmed with PAGENET allowed in the Class of Service;
- the point-to-point D-channel is configured with remote capability (e.g RCAP=NAC).

With PAGENET Uncontrolled, all external users can access the paging route provided that the call paging request to the paging node is incoming via a TIE or a VNS Bearer trunk.

When the call paging request has been accepted and established by the paging node, the originator does not receive any special indication when the call is connected to the paging trunk.

Operating parameters

If the NTAK02BB SDI/DCH circuit card is used on systems with a software vintage earlier than Generic X11 Release 21.40, it must be ensured that the power is removed from the card before it is used for the operation of Call Page Network Wide. This will normally take place during software upgrade; otherwise, the card must be removed and the power must be reset.

To support Call Page Network Wide within a dedicated ISL/VNS configuration, a NTAK02 (minimum vintage **BB**) SDI/DCH circuit card must be used.

Call Page Network Wide will only be supported on the NTBK51BA D-Channel Daughterboard, for D-channel handling with PRA.

A user may experience a delay between the time of dialing the last digit and the actual call termination. The Call Page Network Wide feature does not change this operation. Therefore, with external paging calls the user may not realize when a connection is actually established unless the paging equipment provides audible notification.

External PAGENET uncontrolled calls are supported on Integrated Services Digital Network (ISDN) and non-ISDN networks provided that the incoming trunk into the Paging node is a TIE Trunk.

External PAGENET Controlled is only supported on Virtual Network Services (VNS) and ISDN networks, provided that the caller is using an Attendant Console or PAGENET allowed set and the point-to-point D-channel connection between the nodes has remote capability of network access (e.g. RCAP = NAC).

Feature interactions

Attendant Barge-In

Attendant Break-In

For external PAGENET uncontrolled calls, Attendant Barge-In is blocked at the Paging node, per existing operation.

For external PAGENET controlled calls, Attendant Barge In is blocked at both the originating and Paging node.

Attendant Call Extension

If an attendant's source (SRC) or destination (DEST) Key is connected to an external PAGENET uncontrolled trunk, Attendant Call Extension is not blocked. However, if an attendant's SRC or DEST Key is connected to an external PAGENET controlled route, Attendant Call Extension is blocked.

**Call Forward All Calls
Call Forward No Answer
Hunt**

PAGENET does not block a station set from being programmed to Call Forward All Calls, Call Forward No Answer or Hunt 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 Park
Call Transfer
Conference
No Hold Conference**

A station set or Attendant Console that parks, transfers or conferences an external PAGENET uncontrolled call is not blocked. However, an external PAGENET controlled call is blocked.

Originator Routing Control/Remote Virtual Queuing

This is supported for an incoming call to a Paging trunk when all the trunk members of the dialed Paging route are busy.

Feature packaging

Call Page Network Wide (PAGENET) requires package 307.

Feature operation**Internal Paging Call**

No specific operating procedures are required to use this feature.

External Paging Call

When a user makes an external Paging call via dial access, one of the following dialing plans must be used:

- 1** Route Access Code (ACOD) that connects user to Paging node and Paging route ACOD;
- 2** BARS/NARS; or
- 3** Coordinated Dialing Plan.

Call Park Network Wide

Prior to X11 Release 22, a call in a parked state on a System Park Directory Number (DN) or station set park DN could only be retrieved by an attendant console or a station set located within the same node. The Call Park Network Wide (CPRKNET) feature builds on the existing functionality of Call Park and introduces the following capabilities: Network Call Park, Call Park Expansion and External Call Park Access.

Network Call Park enables an attendant or a station set to park a call on System DN or DN within a Meridian Customer Defined Network (MCDN). This networking capability requires users at parking, parked at, tandem and accessing nodes be equipped with Network Attendant Services (NAS). The parking node refers to the location of the attendant or station set parking the call, the parked at node is the location of the parked call, the tandem node is the routing bridge for the parked call and the accessing node is the location of the user retrieving the parked call.

Call Park Expansion increases the amount of Call Park blocks and number of System Park DNs. With CPRKNET configured, a user can define up to five different Call Park blocks. Each call park block can be separately configured with programmable parameters (such as recall timers). The maximum recall timer is expanded from 240 to 480 seconds and the maximum number of System Park DNs in each block is increased from 50 to 100 directory numbers.

External Call Park Access permits an external caller to retrieve a call in parked stated through either a Direct Inward Dial (DID) trunk or a TIE trunk. The external user must initially be informed that a call has been parked. The external party is informed and can only retrieve the parked call if they know the accessing DN. It is important to note that if a system administrator utilizes the enhancements offered by Call Park Expansion and configures five call park blocks then these System Park DN ranges must be known by all users attempting to access this feature.

Operating parameters

The existing Call Park feature packaging is expanded in X11 Release 22. With Release 22, recall timer and number of System Park DNs expansions are included in base system software. However, the new CPRKNET package is required to access the multiple call park blocks and networking capabilities.

Network Call Park does not support Centralized Attendant Service.

The existing Trunk Barring feature ensures that only an appropriate incoming trunk can be connected to a parked party.

The existing Trunk Group Access Restriction (TGAR) feature checks the incoming accessing trunk. A parked call can only be retrieved if the TGAR and Trunk Access Restriction Group (TARG) of the trunks correctly match.

Call Park Expansion

The Primary Call Park Block must be defined for the customer.

Network Call Park

Network Call Park is supported if the network has all nodes connected by MCDN ISDN links with NAS signaling configured. All the current limitations of the NAS feature apply to CPRKNET.

The CPRKNET package must be enabled on both the parking and parked nodes. The package is not required on the tandem node.

The parking node and the parked at node must have a Primary Call Park Block defined.

When the call park recall timer expires on a parked call at another node, the call recalls to the attendant at the parking node regardless of the configuration of the recall park call to attendant (RECA) timer of the associated call park block. This recall to an attendant occurs even if the call was parked by a station set.

A remotely parked call recalls to an attendant, or Network Attendant if configured, at the parking node when the timer expires. This recall to the attendant occurs even if the parking party is a station set.

Parked Call External Access

Only a call that is parked on a System Park DN can be retrieved by an incoming trunk. A call parked on a station DN cannot be retrieved by an external caller.

Only incoming Direct Inward Dial (DID) or TIE trunks can retrieve a parked call.

A user does not receive any special indications when retrieving a parked call. The user is connected to the parked call immediately.

Feature interactions

Answer Supervision

If a parked call is connected to an incoming trunk with Answer Supervision, the appropriate messages on the status of the call are sent through the incoming trunk to the far end.

Basic Alternate Route Selection

Network Alternate Route Selection

An Electronic Switched Network (ESN) number can be assigned as the System Park or Station Set DN to a Network Call Park call. A parked call on a System Park DN can be retrieved by a caller outside the parked node via BARS/NARS dialing.

Camp-On

When an attendant attempts to extend a call to a busy station across the network and the busy station returns a Camp-On allow signal, an attendant has the option of camping on a call or continuing with Network Call Park.

Coordinated Dialing Plan

A Coordinated Dialing Plan number can be assigned to a Network Call Park call that the attendant or station set is attempting to park.

Trunk Anti-Tromboning

The Trunk Anti-Tromboning feature is invoked if programmed at all interim Private Branch Exchanges (PBXs) in the call.

Recall to Same Attendant, Network Wide

Network Call Park supports this feature. However, all limitations and restrictions associated with Network Wide Recall to Same Attendant are applicable.

Feature packaging

The Network Call Park and External Call Park Access capabilities of the Call Park Network Wide (CPRKNET) requires package 306 and Network Attendant Services (NAS) package 159. With Release 22, expansions to the recall timer and the number of System Park DNs are included in Call Park (CPRK) package 33.

Feature operation

Call Park Expansion

Attendant Console with Park Key

Attendant Console via Dial Access

Meridian 1 Proprietary Set with Display and Park Key

- 1 Press PRK key, or dial SPRE + 71 or Call Park FFC.
- 2 An available System Park DN is displayed. To override this DN, dial another Park DN.
 - If number dialed is an available Park DN, the digit display is cleared and only the dialed Park DN is displayed (without SPRE + 71 or CPRK FFC).
 - If number dialed is invalid or unavailable station Park DN an overflow tone is heard.
 - If number dialed is an unavailable System Park DN, overflow tone is heard. If another System Park DN is available in the Call Park Block, the system assigns this DN. The available System Park DN is displayed and silence is returned.
- 3 Press PRK key or if using SPRE or FFC press Release RLS key to complete call park.

Meridian 1 Proprietary Set with Display via Dial Access

- 1 Press the CONF or TRN key.
- 2 Dial SPRE + 71 or CPRK FFC.

- 3 An available Primary System Park DN is displayed. To override this DN dial another Park DN.
 - If number dialed is available, digit display is cleared and dialed Park Access ID is displayed (without SPRE + 71 or CPRK FFC).
 - If number dialed is an invalid System Park DN or an unavailable station park DN, overflow tone is heard.
 - If number dialed is an unavailable System Park DN, but another available System Park DN exists in the Call Park Block, the available System Park DN is displayed. Silence is returned.
- 4 Press the CONF or TRN key to complete call park.

***Meridian 1 Proprietary without Display
Analog (500/2500) type telephones***

The Call Park operation on these types of telephones is not affected.

Network Call Park

The existing Call Park operation is modified to accommodate the Network Call Park capabilities of the CPRKNET feature. A user must enter a Park DN that is either a Station Park DN or a System Park DN that is located at another node, within the attendant's or station set's MCDN network.

***Attendant Console
Meridian 1 Proprietary Set with Display with Park Key
Attendant Console via Dial Access***

- 1 Press the PRK key or dial SPRE + 71 or CPRK FFC.
- 2 An available Primary System Park DN from parking node is assigned and displayed.

- 3 To override the displayed System Park DN with another Park DN, dial the ESN number and remaining digits.
 - At the remote node, if there is an available Park DN, silence is heard. The digit display of the set is cleared and the dialed Park DN is displayed without the SPRE + 71 or CPRK FFC.
 - At the remote node, if numbers dialed are invalid or unavailable, overflow tone is heard.
 - At the remote node, if the numbers dialed are an unavailable System Park DN but another System Park DN is available in the Call Park Block, the system assigns this DN. The available System Park DN is displayed and silence is returned. If no DN is available and there is no other available DN in the Call Park Block, overflow tone is heard.
- 4 If using an Attendant Console, press the RLS key, or if using Meridian 1 Proprietary Set with Display, press the PRK key to complete call park.

Attendant Parking an Extended Call

- 1 Press the PRK key.
- 2 Extended party is released and ringback or busy tone is removed. SPRE + 71 + ESN number of the extended call are displayed.
 - At the remote node, if there is an available Park Access ID, silence is heard.
 - At the remote node, if numbers dialed are an invalid or unavailable Park DN, overflow tone is heard.
- 3 Press the RLS key to complete call park.

Meridian 1 Proprietary Set with Display via Dial Access

- 1 Press the CONF or TRN key.
- 2 Dial SPRE + 71 or CPRK FFC.
- 3 An available Primary System Park DN from the parking node is displayed.

- 4 To override this with another Park DN, dial the ESN number and digits.
 - At the remote node, if there is an available Park Access ID silence is heard. The digit display is cleared and the dialed Park Access ID is displayed without the SPRE + 71 or CPRK FFC.
 - At the remote node, if the ESN number is an invalid or unavailable Park DN, overflow tone is heard.
 - At the remote node, if dialed System Park DN is not available but there is another available System Park DN in its Call Park Block, the available System Park DN is displayed. Silence is returned.
- 5 Press CONF or TRN key to complete call park.

Meridian 1 Proprietary Set without Display with PRK key
Meridian 1 Proprietary Set without Display via Dial Access
Analog (500/2500 type) set

- 1 Press the PRK, CONF, TRN or perform switchhook flash depending on type of set and key assignment.
- 2 Dial SPRE + 71 or CPRK FFC.
- 3 Dial the ESN Number and digits.
 - At the remote node, if there is an available Park DN, silence is heard.
 - At the remote node, if numbers dialed are invalid or unavailable, overflow tone is heard;
- 4 Press same key in Step 1 or go on-hook to complete call park.

Canceling Call Park Network Wide during Operation

The procedure for canceling the Call Park Network Wide feature follows the existing operation of canceling Call Park. However, once an overflow tone is heard, the attendant or station set must cancel the Call Park attempt and restart the Call Park process. The operation of canceling Call Park on different terminals is described below.

Attendant Console

Press the RLS DEST Key. Call Park is canceled and the original call is reconnected to the attendant console.

Meridian 1 Proprietary Set

Press the flashing DN Key. Call Park is canceled and the original call is reconnected to the attendant console or the station set.

Analog (500/2500 type) Set

Complete a switchhook flash. Call Park is canceled and the original call is reconnected to the attendant console or the station set.

External Call Park Access

External Call Park Access allows a parked call on a System Park DN to be retrieved by an external user via an incoming DID or TIE trunk. This capability requires that the node where the call is parked (parked at node) is equipped with the CPRKNET feature and the Primary Call Park Block is defined. To enable external access capabilities, calls must be parked against a System Park DN.

- 1** Call Parked.
- 2** External access caller must be notified or know where the parked call is located.
- 3** Depending on specific dialing plan configuration, the external access caller must dial one of the following to access the call in parked state:
 - for Coordinated Dialing Plan, dial System Park DN;
 - for ESN, dial Access Code + Location Code + System Park DN;
 - for DID, dial area code + Local Exchange + System Park DN; or
 - for DISA, dial DISA number + (Authorization Code + DISA Security Code) + System Park DN.

Call Pickup Network Wide

The Call Pickup Network Wide feature enables the following functionalities to be extended over a Meridian Customer Defined Network ISDN network:

- Ringing Number Pickup
- Directed Call Pickup by Group Number
- Directed Call Pickup by DN, and
- Display Call Pickup.

features remains unchanged. To display Call Pickup, press the Display key, followed by the Call Pickup key. Display Call Pickup is modified so that the Ringing Number Pickup (RNP) key winks for five seconds once a local or remote ringing DN is found and displayed. During this time, the user can press the RNP key to initiate a Call Pickup directed to the displayed DN.

With Call Pickup Network Wide, users must be assigned to the same Call Pickup group regardless of network location. Each Ringing Number Pickup Group may be linked to a Speed Call List (SCL) which is used when there is not an applicable local set to pick up. Different groups may be linked to different SCLs.

To be able to route calls through the network from one originating node to a destination node, an ISDN Private Integrated Services Network Exchange (PINX) DN is defined for each node in the network. The ISDN PINX DN is a DN taken from the customer's numbering plan used to aid with the routing of network calls. It does not correspond to a real terminal on the node, so can never be busy. Each SCL contains a list of PINX DNs which correspond to the remote nodes or customers to be searched. Thus, the purpose of the Speed Call List is to let the system know where to look in the network to pick up the call. If a pickup group is linked to a Speed Call List, this group is considered as being network wide

The search is conducted in an ascending order as programmed in the Speed Call List (i.e., entry 0 first). This Speed Call list is used when there is not an applicable local set to pick up.

When a network search is performed, a slight delay occurs before a call is connected or rejected, during which time the set will receive silence. This delay is traffic dependent. If for any reason the call cannot be rerouted to the requesting party, the call will ring again at the originally dialed DN and the requesting party will receive overflow tone.

Operating parameters

Call Pickup Network Wide is not supported over a Virtual Network Services (VNS) or QSIG link.

A Speed Call List must be configured with the PINX DNs of the remote switch. The local PINX DN must be configured in the Customer Data Block (LD15).

If on one single node two calls are ringing at the same time, the call with the higher priority will be picked up. But if on two nodes two calls are ringing at the same time (one call on each node), the call on the node searched first will be picked up (i.e., a normal ringing call on the local node is picked up before a priority ringing set on a remote node).

Feature interactions

Access Restrictions

All access restrictions applicable to Network Alternate Route Selection (NARS)/Basic Alternate Route Selection (BARS) calls (including Class of Service, Network Class of Service, Trunk Barring (TBAR), and New Flexible Code Restriction (NFCR) restrictions based on digit manipulation) apply to a redirected call from the receiving node to the requesting node. This means that there are no limitations added to the access restriction checks for calls being redirected by the Call Pickup Network Wide feature.

If the call is blocked because of any of these access restrictions on either the receiving, tandeming, or requesting node, the originally called party is re-rung and the party attempting to pick up the call receives overflow tone.

Attendant-Extended Calls

The Call Pickup Network Wide feature can be used to pick up attendant-extended calls to a remote station.

Automatic Call Distribution (ACD)

The Call Pickup Network Wide feature cannot be used to pick up a call to an ACD DN. Calls to ACD DNs will be skipped and queue scanning will search for another ringing call.

A call to a non ACD DN on an ACD set can be picked up as a normal call. A call transferred by an ACD agent can be picked up as a call transferred by a normal set. In addition, it is possible to pick up as a normal call a call originated by an ACD set on a non ACD DN key.

ISDN Basic Rate Interface (BRI)

An ISDN BRI terminal cannot initiate a call pickup, and a call ringing at an ISDN BRI terminal cannot be picked up.

Call Back Queuing

A call redirected by the Call Pickup Network Wide feature cannot be subject to Call Back Queuing.

Call Collision (Glare)

A call redirected by the Call Pickup Network Wide feature will be treated as for normal ISDN calls. A new trunk will be found if possible; otherwise, the requesting party will receive overflow tone.

Call Park

The Call Pickup Network Wide feature cannot be used to pick up parked calls. A recall of a parked call can be picked up, in which case the call is unparked and answered by the requesting party.

Called Party Name Display

Network Call Party Name Display information will be exchanged during Call Pickup Network Wide calls if the sets involved in the call would normally exchange the information for calls over the routes that have been used for the original call and the call pickup. Conversely, if Network Called Party Name Display would not operate for a normal call from the originating party to the terminating party, the service will not be supported when Call Pickup Network Wide is involved.

Name displayed on the requesting party

If a remote ringing station is picked up, the originating party's name is always displayed, independent of the requesting party's Class of Service.

Call Redirection**Network Call Redirection**

Call redirection only has an interaction if the call being picked up has already been redirected. In this case, the original redirecting number will be passed on instead of the DN of the set from which the call is being picked up. The redirection reason displayed will remain as the previous displayed redirection reason if the Called Party Name Display is involved.

Call Transfer**Network Call Transfer**

A call may be picked up before or after the transferring party has completed the transfer.

For pickup before transfer completion, the transferring party is displayed updated information by the Call Pickup Network Wide feature when the call is picked up. Then, when the transfer is completed, normal call transfer information is exchanged by each party involved in the final call.

For pickup after call transfer completion, everything happens as if the call had been made directly from calling to ringing party. After pickup is performed, displays are updated as for normal Call Pickup.

Call Waiting

Call Waiting calls cannot be picked up.

Call Waiting Redirection

A call that is redirected by the Call Waiting Redirection feature to the active set's Call Forward No Answer DN may be picked up.

Calling Party Number Privacy

The "Privacy Indicator" provided by the Calling Party Number Privacy feature is respected when the requesting party's display is updated after the ringing call has been picked up successfully (i.e., if the originating party has specifically requested privacy), the originating party's name and/or DN is not displayed on the requesting party's display.

Conference

Call Pickup Network Wide may be used to pickup an enquiry call from a conference, subject to the same limitations as apply to Call Transfer.

Data Calls

The Call Pickup Network Wide feature cannot be used to pick up data calls.

Dial Intercom

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.

Display of Calling Party Denied

The Class of Service DDGA/DDGD for digit display and NAMA/NAMD for name display are respected when the requesting party's display and the originating party's display are updated after the ringing call has been picked up successfully.

Group Call

The Group Call feature does not allow having 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.

Multi-Tenant Service

The Call Pickup Network Wide is not supported in a multi-tenant environment.

Network HOT Type D Intercom

It is not possible to pick up a HOT type I call; however, it is possible to pick up a HOT type D call.

Network Routing Restrictions

NARS Anti-Tromboning

At the receiving node where the set is ringing, NARS Anti-Tromboning does not apply to a call being redirected by the Call Pickup Network Wide feature.

Network Attendant Service/ISDN Call Connection Limitations

ISDN Call Connection Limitations (ICCL) apply to Call Pickup Network Wide calls. If the call being redirected by the Call Pickup Network Wide feature is blocked due to one of the ICCL limitations (e.g., Tandem Threshold, Call Redirection Threshold, Number of MU/A Law Conversions, Disconnect Supervision), the originating call will re-ring the originally called party and overflow tone is given to the requesting party.

Night DN

The Call Pickup Network Wide feature can be used to pick up a call to the night number if it is ringing an ordinary station (i.e., analog (500/2500 type) telephone, 16-button Dual-tone Multifrequency, or proprietary set).

Off-Hook Queuing

A call redirected by the Call Pickup Network Wide feature cannot be the subject of Off-Hook Queuing.

Path Optimization***Network Attendant Services (NAS)***

NAS Anti-Tromboning is supported by the Call Pickup Network Wide feature.

Trunk Route Optimization Before Answer

The Call Pickup Network Wide feature does not support Trunk Route Optimization Before Answer.

Secondary DN

For calls to, from, or picked up from a secondary DN, all line IDs displayed after a successful pickup will follow the current rules for Calling Line ID and Connected Number. The exception is for Display Call Pickup; if the call is ringing at a secondary DN, the line ID on the requesting user's display will be the ringing DN (that is, the secondary DN). Once the pickup is completed, the displays will be as for normal pickup.

Vacant Number Routing

The Call Pickup Network Wide feature fully supports Vacant Number Routing (VNR) if the route or the set of routes given by the customer for VNR contains at least one MCDN link.

Virtual Network Services

The Call Pickup Network Wide feature will not work in conjunction with the Virtual Network Services feature.

Feature packaging

The Call Pickup Network Wide feature is included in Integrated Services Digital Network (ISDN) package 145.

The following packages must also be provisioned to activate this feature:

- Directed Call Pickup (DCP) package 115
- Advanced ISDN Network Services (NTWK) package 148, and
- 2.0 Mbps Primary Rate Interface (PRI2) package 154.

Feature operation

Display Call Pickup

This feature is only supported on Meridian Digital Telephones (with the exception of the M2317, and M3000).

- 1 Press the DSP key.
 - The DSP key is lit.
- 2 Press the RNP key.
 - The RNP key is flashing.
- 3 The display will show: Hbbbb1 Haaa2

Where 1 = The DN of the originating party if it is local or available from CLID. If no CLID is available for a remote, originating party, either the route access code and the route member number are displayed if the ringing set is local, or dashes (“----”) are displayed if the ringing set is remote. This is because the route access code of a route has no meaning on the requesting node. The “H” is displayed if the DN can be determined and it is located on another node.

Where 2 = The DN of the originally called (ringing) party. The “H” is displayed if the DN is located on another node than the requesting party.

- The RNP key is winking for five seconds.

- 4 Press the DN key.
 - The DN key is lit.
- 5 Press the RNP key. If the RNP key is pressed during the five seconds it is winking, Directed Call Pickup by DN is attempted. If the RNP key is dark when pressed, normal Ringing Number Pickup treatment takes place.
 - The RNP key is flashing.
- 6 You are connected to the call.
 - The RNP key becomes dark.
 - For sets with display the following is shown:
"Name"1
Hcccc2 Hbbbb3 "P"4

Where 1 = Either the originating party's name or the originally called party's name is displayed depending on the requesting party's Class of Service.

Where 2 = The DN of the originating party if it is local or available from CLID. If no CLID is available for a remote, originating party, either the route access code and the route member number are displayed if the ringing set is local, or dashes ("----") are displayed if the ringing set is remote. This is because the route access code of a route has no meaning on the requesting node. The "H" is displayed if the DN can be determined and it is located on another node.

Where 3 = The DN of the originally called party. The "H" is displayed if the DN is located on another node than that of the requesting party.

Where 4 = The string defined in LD 95 for Call Pickup. The default is "P"

Ringing Number Pickup

To answer a call in your Call Pickup group from a Meridian Digital Telephone:

- 1 Lift the handset, or press a DN key.
 - The DN key is lit (if pressed).

- 2 Press the RNP key or dial SPRE + 3 or dial PURN FFC.
 - The RNP key is flashing.
- 3 You are connected to the call.
 - The RNP key becomes dark.
 - For sets with display, refer to setp 3 under “Display Call Pickup”.

To answer a call in your Call Pickup group from an analog (500/2500 type) telephone:

- 1 Lift the handset.
- 2 Dial SPRE + 3 or PURN FFC.
- 3 You are connected to the call.

Directed Call Pickup by Group Number (Group Pickup)

To answer a call in another Call Pickup group from a Meridian Digital Telephone:

- 1 Lift the handset, or press a DN key.
 - The DN key is lit.
- 2 Press the GPU key or dial SPRE + 94 or dial PUGR FFC.
 - The GPU key is lit (if pressed).
- 3 Dial the pickup group number.
 - Once the group number is dialed completely, the GPU key is flashing to indicate that a search or scan is in progress.
- 4 You are connected to the call.
 - The GPU key becomes dark.
 - For sets with display, refer to Step 3 under “Display Call Pickup”.

To answer a call in another Call Pickup group from an analog (500/2500 type) telephone:

- 1 Lift the handset.
- 2 Dial SPRE + 94 or PUGR FFC.
- 3 Dial the pickup group number.
- 4 You are connected to the call.

Directed Call Pickup by Directory Number (DN Pickup)

To answer a call on a specified DN from a Meridian Digital Telephone:

- 1 Lift the handset, or press a DN key.
 - The DN key is lit.
- 2 Press the DPU key or dial SPRE + 95 or dial PUDN FFC.
 - The DPU key is lit (if pressed).
- 3 Dial the extension number.
 - Once the group number is dialed completely, the DPU key is flashing to indicate that a search or scan is in progress.
- 4 You are connected to the call.
 - The DPU key becomes dark.
 - For sets with display, refer to Step 3 under “Display Call Pickup”.

To answer a call on a specified DN from an analog (500/2500 type) telephone:

- 1 Lift the handset.
- 2 Dial SPRE + 95 or PUDN FFC.
- 3 Dial the extension number.

You are connected to the call.

Calling Line Identification

Description

Calling Line Identification (CLID) is a feature that delivers the calling party's telephone number to the called party's telephone set across the ISDN network. The CLID display lasts for the duration of the call.

CLID on Outgoing Calls

A set's Prime Directory Number (PDN) is assembled and sent out with the call SETUP message on the D-channel. For private network calls in a NARS network, the calling party number is the PDN which consists of the Home Location Code (HLOC) plus the 4 digit directory number. In a Coordinated Dialing Plan (CDP) Network, the CLID is the CDP number. For public network calls, the CLID is the 10-digit (E.163) Northern American Standard Number.

The Listed Directory Number (LDN) for a customer is sent instead of the Prime Directory Number (PDN) if the set has the LDN class of service assigned in the set data block (see Feature Implementation for details).

CLID on Incoming Calls

For incoming calls across the ISDN network, the CLID received over the D-channel is displayed instead of the trunk route access code and trunk member number.

The following devices, if equipped with a digit display, can receive and display a CLID number:

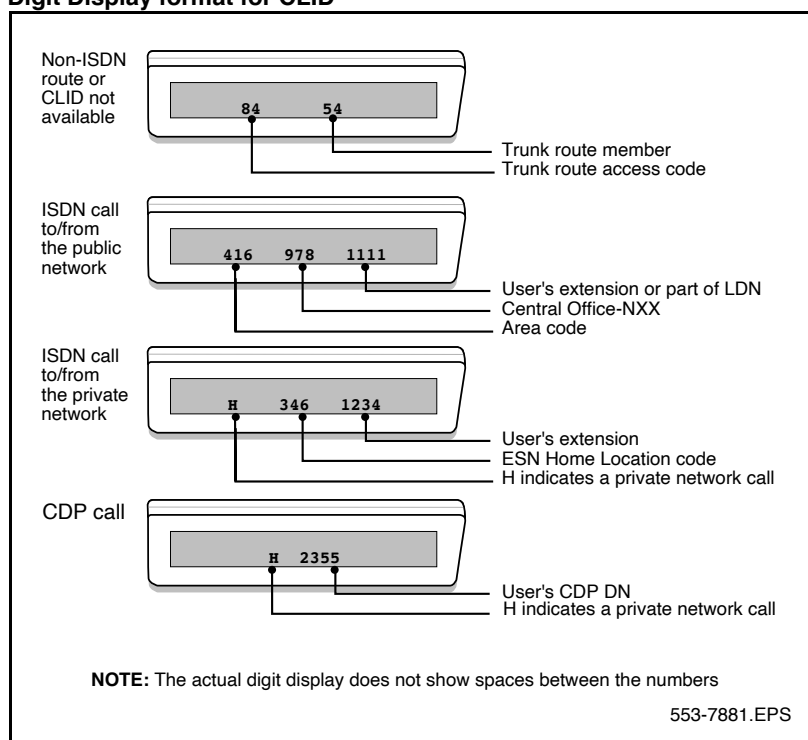
- Attendant Consoles
- Digital sets equipped with digit display
- SL-1 sets with digit display
- ASCII terminals with data adapter such as ADM or programmable data adapter.

The digit display contents are summarized in Figure 11 and described below:

- If the CLID display is unavailable because the call was not routed on ISDN routes, the trunk route access code and the trunk route member number are displayed. Note that Dialed Number Identification Service (DNIS) digits may be displayed as well if proper software is equipped.

- For calls from a public network, the CLID displays the public-exchange number
- For a private network (with NARS), the CLID display is preceded by the letter "H" followed by the ESN Home Location Code and the user's extension number (or the customer's Listed Directory Number).
- For a private network (with CDP), the CLID display is preceded by the letter "H" followed by the user's CDP directory number (or the customer's Listed Directory Number)

Figure 11
Digit Display format for CLID



Package Requirements

This option requires the following feature packages:

Feature	Package Number
ISDN	145
PRI2 or ISL	154 or 147
DDSP	19

Calling Line Identification in Call Detail Records

Description

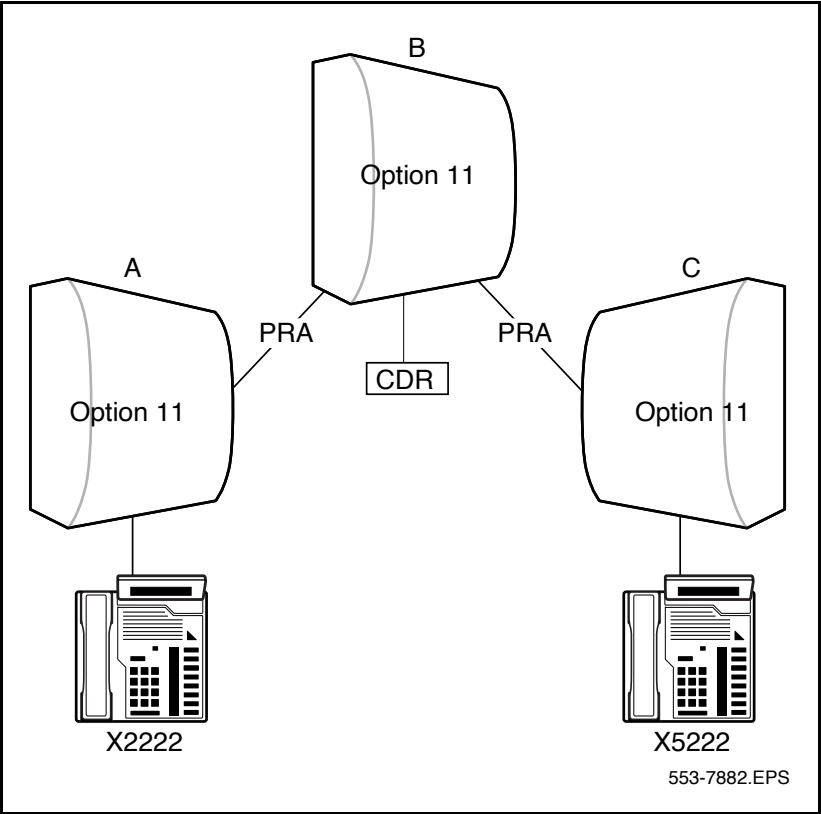
This service provides the addition of a Calling Line Identification (CLID) field in the Call Detail Record (CDR).

The addition of the CLID field allows customers to charge back the calling party for services rendered in connection with their incoming calls. For example, calls placed to a service centre may be charged back to departments receiving the service, or calls placed to a consultant for the time spent with the client.

Another use of CLID in CDR feature is to capture the actual calling DN at the tandem PBX. Figure 12 illustrates a network with three Meridian 1 switches. When a user on PBX "A" calls PBX "C" through PBX "B", the caller's CLID from PBX "A" can be captured on the CDR at PBX "B".

In the following example DN 2222 on PBX A is calling DN 5222 on PBX C where PBX B is used as a tandem PBX. PBX B's CDR captures the actual extension (X2222) of the caller.

Figure 12
CDR in Multi-site Configuration



Package Requirements

This option requires the following feature packages:

Feature	Package Number
CDR	4
CTY	5
ISDN	145
PRI2 or ISL	154 or 147
CLID in CDR	118 This is only required at the collecting location (i.e. site B in Figure 12).

Calling Party Privacy

The Calling Party Privacy (CPP) feature enables the Meridian 1 to support the blocking of a Calling Party's Number and Name from being displayed at the terminating set on a per-call basis. Users can dial a Calling Party Privacy code (e.g., *67 from a Meridian 1 proprietary set or 1167 from an analog (500/2500 type) set) to prevent their telephone number and name from being displayed on a receiving telephone across the Public Switched Telephone Network (PSTN). Internal calls within the Meridian 1 will have originating numbers or names displayed even though the originating call has requested privacy.

This feature also allows a per-line blocking Class of Service to be programmed for station sets for public network calls. This relieves the user from having to dial the Flexible Feature Code (FFC) for every call, but in every other way is equivalent to the per-call blocking.

Depending on the trunk route configuration, public network numbers which tandem over the Meridian 1 Meridian Customer Defined Network (MCDN), prior to exiting to the PSTN, Privacy Indicator will be passed along if dialed by the originator. This means that users can be sure that their privacy wishes are respected whether the call exits directly at the originating node, or is given alternate routing through a private network.

However, if private network nodes are connected by non Integrated Services Digital Network (ISDN) Electronic Switched Network (ESN) trunks, the complexity of the signaling precludes the transmission of the Privacy Indicator. To compensate for this, outgoing Central Office (CO), Foreign Exchange (FEX), Wide Area Telephone Services (WATS), and Direct Inward Dialing (DID) trunks, may be configured to automatically generate a Privacy Indicator for calls received from incoming non-ISDN trunks.

A Privacy Indicator is used to signify that a call is a Calling Party Privacy call. For an outgoing non-Integrated Services Digital Network (ISDN) trunk call, the Privacy Indicator is defined in the outgoing trunk route as a digit string (e.g., *67). No Privacy Indicator is expected for an incoming non-ISDN trunk call. For an ISDN call between two Meridian 1 switches, the Privacy Indicator is represented by setting the Presentation Indicator field to “Presentation Restricted” in the Calling Party Number Information Element (IE) and the Calling Party Name Display (CPND) Indicator to “Presentation Denied” in the Display IE. For an outgoing ISDN call to the Public Exchange/Central Office, the Privacy Indicator is represented by setting the Presentation Indicator field to “Presentation Restricted” in the Calling Party Number IE and excluding the Display IE with the CPND information. An incoming ISDN call is marked as a CPP call (i.e., carries the Privacy Indicator) if the Presentation Indicator field is set to “Presentation Restricted” in the Calling Party IE or the CPND Indicator is set to “Presentation Denied” in the Display IE.

Calling Party Privacy Enhancement

The X11 Release 23 Calling Party Privacy Enhancement (CPPE) feature provides a route option to ignore the Calling Party Privacy Indicator on incoming calls received from the North American public ISDN network.

When the Privacy Indicator Ignore (PII) prompt is set to YES in Overlay 16, the Calling Line Identification (CLID) Presentation Indicator and the Calling Party Name Display (CPND) Indicator (if it exists) are changed from restricted/denied to allowed. If the CLID and CPND information is available, it is displayed on the terminating set, and the CLID is passed to the Auxiliary processor. It is recommended that the PII prompt be set to YES for 800, 888, 900, and 911 call types. When PII is set to NO (default) in Overlay 16, the Calling Party Privacy Indicator is honored, and the existing functionality is maintained.

The following parameters apply to the PII route option:

- The PII route option is applicable to the CLID Presentation Indicator in the Calling Number Information Element (IE) and the CPND Indicator in the Display IE in incoming SETUP messages only.
- The PII route option is not applicable to the CLID Presentation Indicator in the Redirecting Number IE of a SETUP message for call redirection.
- The PII route option is not applicable to the CLID Presentation Indicator in the Connected Number IE of a NOTIFY message for call modification.

The Calling Party Privacy Enhancement is included in Calling Party Privacy (CPP) package 301.

Operating parameters

The code to be dialed by the user can be flexibly defined, although *67 will be usual in North America. Multiple codes can be defined allowing a different code (e.g., 1167) to be used for rotary sets.

The code which is outpulsed on non-ISDN analog or digital trunks can also be flexibly defined on a per-route basis. Different codes can be programmed for routes which mix Digitone (DTN) and Dial Pulse (DIP) Classes of Service.

Frequently, the codes outpulsed on trunks will be the same as those dialed from station sets, but there is no reason why they cannot be different.

A non-ISDN trunk route will not be able to provision the CPP feature if Outpulsing of Asterisk and Octothorpe (OPAO) package 104 is equipped on the switch. During SYSLOAD, the CPP database will be removed from the non-ISDN trunk routes if the OPAO package is equipped.

The CPP feature is not supported on Digital Private Network Signaling System #1 (DPNSS1), Digital Access Signaling System #2 (DASS2) or R2 Multifrequency Compelled Signaling gateways.

Central Office Trunk (COT), Foreign Exchange (FEX), Wide Area Telephone Service (WATS), and Direct Inward Dialing (DID) are the only trunk route types allowed to outpulse the Privacy Indicator for an outgoing non-ISDN call.

ISDN implementations include DSM-100/250, SL-100, AT&T4, AT&T5, TR-1268 (NI-2) and Meridian Customer Defined Network (MCDN) private networks, QSIG, EuroISDN and Basic Rate Interface (BRI) trunks.

CPP is not formally supported on the International ISDN PRI connectivities, since CPP is primarily a North American feature. However, existing Calling Line Identification (CLID)/Calling Line Identification Restriction (CLIR) operations will continue to work.

The Privacy Indicator defined for a non-ISDN trunk route (dial pulse or digitone) consists of any arbitrary digit sequence (0-9) up to four digits in length. The asterisk “*” or octothorpe “#” is not allowed in the Privacy Indicator for an outgoing dial pulse trunk route. The asterisk is only allowed as the first digit of the Privacy Indicator (e.g., *67) for an outgoing digitone trunk route; the octothorpe is not allowed in any Privacy Indicator on an outgoing digitone trunk route.

If as user requests privacy by dialing the Flexible Feature Code (FFC) defined for the CPP feature, and CPP is not provisioned in the outgoing trunk route, the call will proceed without carrying the Privacy Indicator.

No Privacy Indicator is expected to be received from the Public Exchange/Central Office on non-ISDN DID trunks. This would be treated as a misdial.

The CPP feature will not inhibit the Calling Party Number and Name from being displayed for an internal call within a local Meridian 1 customer group.

A common number defined for the Special Prefix (SPRE) code in the database is “1”. Thus, “1167” will not be accepted as an FFC for CPP due to the conflict with existing DNs. The technician should either change the SPRE code, or define a new FFC for CPP to be used by a rotary telephone.

Feature interactions

Autodial

An outgoing trunk call initiated by pressing the Autodial key will carry the Privacy Indicator if the Calling Party Privacy (CPP) code followed by the normal dialing sequence is stored against the Autodial key. The CPP code is counted against the maximum number of digits (currently 23) stored against the Autodial key.

A user can also store the CPP code against the Autodial key. An outgoing CPP call can be initiated by pressing the Autodial key, followed by manually dialing the digits.

An outgoing CPP call can also be initiated by dialing the CPP code, followed by pressing the Autodial key against which the normal dialing sequence of digits have been stored.

Automatic Call Distribution

A call placed by means of Enhanced Automatic Call Distribution (ACD) Routing, Enhanced Interflow, Enhanced Night Call Forward, Enhanced Network Routing, or Network ACD will respect the CPP request of the originator.

Automatic Call Distribution MAX (ACD MAX)

The Calling Line Identification (CLID) is still included in ACD MAX reports, even if the caller has requested CPP.

Automatic Redial

The calling party and called party have the same Calling Party Privacy considerations.

Call Detail Recording

The current Call Detail Recording (CDR) records which include the Calling Party Number will continue to do so even if the caller has requested CPP. The FFC for CPP dialed by the user will be included in the dialed digits field when generating a CDR record.

An outgoing non-ISDN trunk call outputting the Privacy Indicator will include the Privacy Indicator in the outputted digits field when generating the CDR records if the outgoing non-ISDN trunk route has Outputted Digit Option (DPD) activated.

Call Forward All Types Hunt

If an incoming ISDN trunk call with the Privacy Indicator is forwarded, the Privacy Indicator will be tandemmed 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 CPP 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 CPP 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 (i.e., at the tandem node) from being displayed on the terminating set. In this case, 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 Hunt and Network Hunt.

Call Hold, Deluxe Call Hold, Permanent

When a user takes an incoming trunk call with the Privacy Indicator off of hold, no Calling Party Number or Name will be displayed on the set.

Call Pickup

If an incoming trunk call with the Privacy Indicator is picked up locally, the display of the calling Party Number and Name are not displayed on the terminating set.

Call Pickup Network Wide

If an incoming trunk call with the Privacy Indicator is picked up by a remote set (requesting party), the display of the calling Party Number and Name are not displayed on the requesting set.

Call Party Name Display (CPND)

In current operations, if the International Supplementary Features (SUPP) package 131 is not equipped in the system, an incoming ISDN call with the Call Party Name Display (CPND) Indicator field set to “Presentation Denied” still displays the Calling Party Name. If package 131 is equipped in the system, the current operations will inhibit the Calling Party Name for an incoming ISDN call with the CPND Indicator field set to “Presentation Denied”.

The CPP feature will inhibit the display of the Calling Party Name for an incoming ISDN call with the CPND Indicator field set to “Presentation Denied” if package 131 is not equipped.

Call Transfer

If an incoming non-ISDN call is being transferred or an incoming ISDN call is transferred to a non-ISDN trunk, the Calling Party Name and Number will not be passed on to the terminating set. The CPP feature will not change this operation.

For cases where an incoming call with the Privacy Indicator is transferred over an MCDN trunk, or to a local station, the name and/or number of the originating party will not be displayed on the set of the final terminating party.

Calling Line Identification Restriction (CLIR)

The Flexible Feature Code is not supported on BRI sets. Calling Party Privacy can only be requested by setting the soft key “ID PRES” (if it exists) to “Denied” or the “PRES” prompt to “NO” in LD 27. If the Calling Party Number IE with the Presentation Indicator set to “Presentation Allowed” is included in the SETUP message generated by the BRI terminal, this BRI terminal will not allow Calling Party Privacy, as the Presentation Indicator generated by the BRI terminal always overwrites the CLIR service option.

Conference

The CPP feature will pass the Privacy Indicator to the terminating set to inhibit the display of the Calling Party Name and Number if the Conference feature is used for the purpose of performing a transfer.

Calling Party Name Display Denied

For outgoing calls, if the CPP package is equipped, the CPP feature will take precedence over the Calling Party Name Display Denied feature for restricting the Calling Party Name and Number. For example, if an outgoing ISDN call is marked as a CPP call, the outgoing SETUP message will include the Calling Party Number IE with the Presentation Indicator set to “Presentation Restricted” and the Display IE with the CPND Indicator set to “Presentation Denied”, to inhibit both the Calling Party Number and Name being displayed on the terminating set, regardless of whether or not the Calling Party Name Display Denied feature allows the display of the Calling Party Name and/or Number.

The Calling Party Name Display Denied feature takes precedence over the CPP feature for displaying an incoming ISDN call. If International Supplementary Features (SUPP) package 131 is equipped, an incoming ISDN call with the Presentation Indicator set to “Presentation Restricted” in the Calling Party Number IE will be marked as a CPP call, and will display “ACOD + Member” or “XXXX” as for the Calling Party Name Display Denied feature.

Display Calling Party Denied

If the Calling Party Privacy (CPP) package is equipped, the CPP feature will take precedence over the Display Calling Party Denied (DPD) feature. The CPP feature also takes precedence over the DPD feature for displaying an incoming ISDN call if the CPP package is equipped. No “----” or “XXX” will be displayed, as for the DPD feature.

EuroISDN Trunk - Network Side

If a number presentation for a call is blocked by the Calling Party Privacy feature, the Calling Line ID, sent over a EuroISDN Trunk - Network Side connectivity, will have the presentation flagged as restricted.

Feature Group D

If an incoming Feature Group D (FGD) call terminates at a Meridian 1 switch locally, the received 10-digit Automatic Number Identification (ANI) will be displayed on the terminating set if the Show ANI Digits on Terminal Displays (SHAN) field is set to “YES” in the FGD data block associated with the incoming trunk route. If the originator requests CPP, the end office will not send the 10-digit ANI to the PBX.

If an incoming FGD call is routed to another switch via ISDN Primary Rate Interface (PRI) or Integrated Service Link (ISL), the outgoing SETUP message will include the 10-digit ANI (if it exists) as the Calling Party Number with the Presentation Indicator set to “Presentation Restricted” if the outgoing trunk route has the TCPP option on. The TCPP option takes precedence over the SHAN field defined in the FGD data block associated with the incoming trunk route to restrict the 10-digit ANI display.

Hot Line

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.

Incoming Trunk Programmable CLID

If the incoming trunk route is a non-ISDN route, the billing number assigned by the incoming trunk route will be passed to the Public Exchange/Central Office with the Presentation Indicator field set to “Presentation Restricted” if the outgoing ISDN trunk route has the TCPP prompt set to “YES”. If the TCPP prompt is set to “NO”, the Presentation Indicator is set to “Presentation Restricted” only if the BDSP (Billing Display) prompt in the incoming trunk route is set to “NO”.

If the incoming trunk route is an ISDN route, the “Restricted” Presentation Indicator will be tandemed to the outgoing trunk route. If the Presentation Indicator is set to “Presentation Allowed” or no Calling Party Number IE is received in the incoming trunk route, the billing number assigned by the incoming trunk route will be passed to the Public Exchange/Central Office with the Presentation Indicator field set to “Presentation Restricted” only if the incoming trunk route has the BDSP prompt set to “NO”.

ISDN QSIG Name Display

Calling Party Privacy (CPP) takes precedence over the ISDN QSIG Name Display feature.

Last Number Redial

The Last Number Redial (LNR) feature will store the CPP code in the LNR data space if the CPP code was included in the last number dialed by the user. Any subsequent outgoing redialed call will send the Privacy Indicator to the far end.

Malicious Call Trace

Incoming calls to stations having the Malicious Call Trace feature enabled will continue to include the Terminal Number (TN) of the calling party in the Malicious Call Trace record, even if the caller has requested CPP.

Meridian Link

The CLID is still included in the Application Module Link (AML) messages sent to the Meridian Link Module even if the call has requested CPP.

Meridian Mail

When an incoming ISDN call with the Privacy Indicator terminates on Meridian Mail, the Calling Party Name and Number will not be passed to Meridian Mail to be recorded. When the called party retrieves the messages, no Calling Party Number or Name will be played, and the called party will not be able to initiate the Call Sender feature either, since no CLID is recorded.

Calls placed by means of Through Dial will be able to request privacy. These are calls where the person accessing Meridian Mail can dial 0 followed by any telephone number. The caller will be able to dial the CPP code as part of the number following 0.

Meridian MAX

The CLID is still sent to the Meridian MAX even if the caller has requested CPP.

Meridian 911

An incoming 911 call with ANI information will always display the ANI digits on the terminating set or pass the ANI information to the Meridian 911 application.

Network Message Services

An incoming trunk call with the Privacy Indicator will not display the Calling Party Name and Number on the Message Center operator's terminal.

Network Ring Again

A call placed by means of the Network Ring Again feature will respect the CPP requested when the call was originally dialed.

Private Line Service

The Private Line Service feature will outpulse the Privacy Indicator only if it is dialed by the originator. An asterisk will be outpulsed to the far end only if it is an OPAO call, otherwise the asterisk signals a three-second pause.

R2MFC CNI/CDR Enhancements

If the Calling Line ID is received with presentation denied, it is not mapped to the Call Number Information (CNI). Instead, the CNI is composed of the CNI DN and the Trunk ID. Optionally, the CNI request may set to ECNI (the CNI End-of-CNI R2MFC level 1 forward signal).

Ring Again – Busy Trunk

A call automatically redialed by the Ring Again – Busy Trunk feature will respect the CPP requested when the call was originally dialed.

Speed Call

System Speed Call

An outgoing trunk call initiated by dialing the Speed Call code will carry the Privacy Indicator if the CPP code followed by the normal dialing sequence is stored in the Speed Call Entry represented by the Speed Call code. The CPP code will be counted against the maximum number of digits (currently 31) allowed per Speed Call list entry.

A user can also store the CPP code in the Speed Call Entry (or Speed Call key). An outgoing CPP call can then be initiated by dialing the Speed Call code (or pressing the Speed Call key), followed by manually dialing the digits. However, existing Speed Call limitations do not allow a user to dial *67 (or anything else) before accessing a Speed Call list entry.

Stored Number Redial

During Stored Number Redial (SNR) programming, a user can store the CPP code followed by the normal dialing sequence in the SNR data space. Outgoing calls originated by the SNR feature will send the Privacy Indicator to the far end. The CPP code will be counted against the maximum number of digits (currently 31) allowed by the SNR feature.

During an active call on a Meridian 1 proprietary telephone, the Stored Number Redial feature will store the CPP code in the SNR data space if the CPP code was included in the number dialed by the originator. The outgoing redialed calls will send the Privacy Indicator to the far end.

Trunk Optimization Before Answer

An optimized call due to Trunk Optimization Before Answer will respect the CPP requested by the originator.

Feature packaging

The following packages are required for Calling Party Privacy:

- Calling Party Privacy (CPP) package 301, which is dependent on
 - Flexible Feature Codes (FFC) package 139.

Note: Non ISDN trunks must restrict the Outpulse Asterisk and Octothorpe (OPAO) package 104 to provide for the CPP feature.

Feature operation

Any outgoing call initiated from a set with Calling Party per-line blocking (CLBA) Class of Service will request Calling Party Privacy.

If the originating party has CLBD Class of Service, the Calling Party Privacy feature can only be activated on a per-call basis; if standard dialing procedures are used, no CPP is requested, and the call will proceed as usual. The user must do one of the following to request CPP:

- 1 Precede any dialing of a call with a new Flexible Feature Code defined for the CPP feature. This operates from all telephone types, except BRI sets.
- 2 The Flexible Feature Code is not supported on BRI sets; however, Calling Party Privacy can be requested by setting the softkey “ID PRES” (if it exists) to “Denied” state or the “PRES” prompt to “NO” in LD 27.

Note: If the Calling Party Number ID with the Presentation Indicator set to “Presentation Allowed” is included in the SETUP message generated by the BRI terminal, this BRI terminal will not allow Calling Party Privacy, as the Presentation Indicator generated by the BRI terminal always overwrites the CLIR service option.

Data Packet Network Access X.25

Description

To access a Public Data Pack Network.

PRI trunks may be configured to access a Public Data Packet Network (DPN - X25) through the Central Office ISDN node, provided that this service is available. On a per call basis, any 'B' channel can be used to access the packet network.

Interactions and Limitations

No high-speed data is currently available on Option 11.

Package Requirements

This option requires the following feature packages:

Feature	Package Number
ISDN	145
PRI2	154

DID-to-Network Calling

Description

This feature facilitates Direct Inward Dialing into the private ISDN.

The Direct Inward Dial (DID) call is treated as though the entire ISDN is a large PBX. It is available on Group E (Phase 5) of Generic X11 software with Supplementary Features.

DID calls entering the network at the local node (that is, the node on which the destination set resides) are unaffected by the feature's operation. In this case, treatment of the DID call is the same as with the stand-alone configuration. The DID-to-Network Calling feature affects only those DID calls which enter the network at a node other than the destination set's node.

Routing of the DID call across the ISDN is the same as the routing of a network call originated from within the network. (An additional information element is sent with the call set-up message, to indicate that the network call originated from a DID trunk.)

A DID call which must be routed across the network, receives a treatment similar to that given to a call terminating within the local node. The DID call receives intercept treatment, if the dialed DN is fully restricted or has DID-restricted class of service. (It also receives intercept treatment if the DN is maintenance busy, vacant, or if routing failure/PABX congestion is encountered.)

Interactions and Limitations

This feature is subject to all network access restrictions.

Package Requirements

The DID-to-Network Calling feature requires no software administration.

Digit Key Signaling at console

Description

This Digit Key Signaling enhancement provides attendants with a limited set of Meridian Mail functions at the console. It allows attendants to enter command digits during certain call states. These digits are sent to Meridian Mail over the ISDN/AP link.

The functionality offered by this feature could allow attendants to help callers operate the features offered by Meridian Mail (for instance, playing voice messages from an external rotary dial set).

The digits are sent to Meridian Mail by way of ISDN/AP KEY messages (and not by way of End to End Signaling). Dialed digits are not saved by Meridian 1 and are not displayed at the attendant console.

While connected to Meridian Mail, other attendant functions continue to operate as before.

Extending source calls to Meridian Mail

When extending a source call to Meridian Mail, Digit Key Signaling operates under the following conditions:

- a call is present on the SRC key of the active loop,
- a call is established on the DEST key to a Meridian Mail agent, and
- the DEST call to Meridian Mail is not a conference call

Once the attendant has reached Meridian Mail and entered the necessary digits to begin playback of messages, the SRC call can be extended to Meridian Mail to allow the caller to hear voice messages.

Direct calls to Meridian Mail

Digit Key Signaling also operates when the attendant dials Meridian Mail directly, under the following conditions:

- no call is present on the DEST key of the active loop,
- a call is established on the SRC key to the Meridian Mail agent (CLS of VMA), and
- the SRC call to Meridian Mail is not a conference call

The attendant cannot extend the call to a destination party using dialed digits. (Key pad input is treated as Digit Key Signaling and not as dialing digits.)

Interactions and Limitations

The attendant can send keypad digits (0-9, * and #) under the following conditions:

- while extending of source calls to Meridian Mail
- during direct calls to Meridian Mail

Package Requirements

This option requires the following feature packages:

Feature	Package Number
ISDN	145
DKS	180

Digital Trunk Interface and Primary Rate Interface Time Slot Reuse

This feature eliminates call blocking due to time slots being unavailable. By allowing a time slot reserved for a primary function to be reused for any subsequent requirements during call processing, all channels on Digital Trunk Interface (DTI) or Primary Rate Interface (PRI) loops will be available for use.

Time slots reserved for a particular DTI or PRI channel are stored in an unprotected data structure allowing easy access, such as a call register. Each time a new path is required for the same DTI or PRI channel, the data structure can be accessed to determine if there is a reserved time slot that can be reused. If the new path requirements is intra-group (single-group switch), then the time slot can be reused. If the new path requirement is inter-group (multi-group switch), the reserved time slot can be reused if a matching junctor slot is available.

A count is kept of the number of times that a time slot has been reused. The time slot is not idled until all reservations have been canceled.

This feature applies to the following loop types on single-group switches:

- 1.5 Mbit 24-channel DTI
- 2 Mbit 30-channel DTI
- 1.5 Mbit 23 B+D PRI
- 2 Mbit 30 B+D PRI, and
- 2 Mbit Japan Digital Multiplex Interface (DMI).

Operating parameters

This feature applies only to network enhanced machines.

Feature interactions

None.

Feature packaging

None.

Display of Access Prefix on CLID

The Display of Access Prefix on Calling Line Identification (CLID) feature enhances the set display by adding the Local, National or International prefix to the CLID display.

The four digit access prefix is a combination of the access code and the National or International prefix. The access code defines the best outgoing route for an external call and the prefix code defines National or International calls. If the trunk requires the International prefix of two digits, the access code (ACOD) is a maximum of two digits.

The Access Prefix Display on CLID feature supports:

- Meridian 1 proprietary sets: M2008, M2216, M2016, and M2317
- Meridian 1 Attendant Consoles: M2250 and M1250

The Display of Access Prefix on CLID feature supports the following Integrated Services Digital Networks (ISDN) interfaces:

- Basic Rate Interface (BRI) trunks
- Meridian Customer Defined Network (MCDN)
- Q-reference Signaling Point (QSIG)

Operating parameters

If CLID is not available, the trunk access code and the prefix number are displayed according to the existing feature operation.

The Display of Access Prefix on CLID feature supports the Coordinated Dialing Plan (CDP) or the Uniform Dialing Plan (UDP).

The Display of Access Prefix on CLID feature does not support analog (500/2500) sets, CLASS sets, or Integrated Services Digital Network (ISDN) Basic Rate Interface (BRI) sets.

Feature interactions

Automatic Set Display

The feature provides enhanced display of CLID of an incoming trunk call at a Set when the Set is busy. The Automatic Set Display can be configured to display both the Access Prefix and CLID.

Euro-ESDN continuation phase III

This feature is an enhancement for the Add Prefix Prompt (ADDP) in the Route Data Block (RDB) sub-feature implemented as a part of Euro-ISDN continuation Phase III.

Meridian Communication Adaptor

The pass through feature on the set modifies the Display of Access Prefix on CLID.

ORION ‘Call Log’

The Display of Access Prefix on CLID feature development did not make any changes to applications such as ORION ‘Call Log’, enabling them to understand the extra digits of the Access Prefix display

Selectable Conferee Display and Disconnect

The Selectable Conferee Display and Disconnect feature displays and allows the disconnect of conferenced sets.

Feature packaging

The Display of Access Prefix on CLID feature requires the Integrated Services Digital Network (ISDN) package 145.

Feature operation

No specific new operating procedures are required to use this feature.

Display of Calling Party Denied

Display of Calling Party Denied (DPD) permits Analog (500/2500 type) and Meridian 1 proprietary sets to either allow or deny associated name and number from being displayed on other sets when involved in a call. This feature is supported on internal calls (same node) and calls placed over a Meridian Customer Defined Network (MCDN) Integrated Services Digital Network (ISDN). Display of Calling Party Denied Class of Service options are programmed on a per set basis.

DPD uses the following Class of Service options in LD 10 and LD 11: Display Digits Allowed/Denied (DDGA/DDGD) and Name Display Allowed/Denied (NAMA/NAMD).

Display of Digits Denied (DDGD) Class of Service restricts digits from being displayed on another set when involved in a call. However, on the calling party's set, the dialed digits are always displayed regardless of the Class of Service assigned to the called party's set. Display of Digits Allowed (DDGA) allows the number associated with a set to be displayed when that set is involved in a call.

Display of Name Denied (NAMD) Class of Service restricts the name associated with a set from being displayed on another set when involved in a call. Display of Name Allowed (NAMA) allows the name associated with a set to be displayed when that set is involved in a call.

Table 8 outlines the possible configurations and the resulting display on both the calling and called parties' displays.

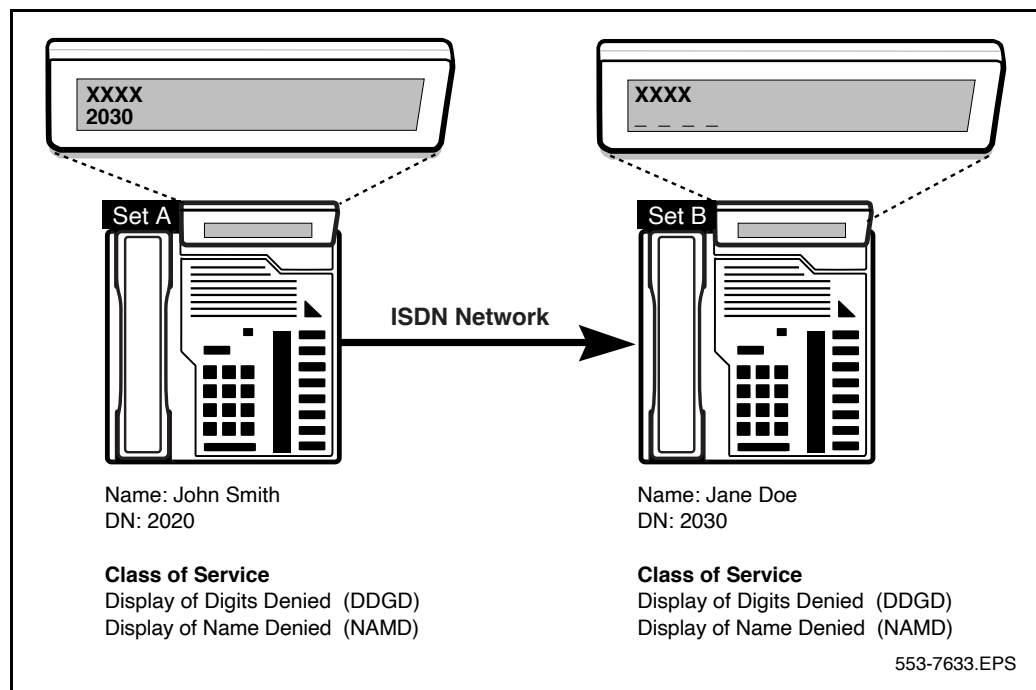
Table 8
Display Scenarios on Calling and Called Party's Sets

CALLING PARTY'S CLASS OF SERVICE	CALLED PARTY'S CLASS OF SERVICE	DISPLAY OF CALLING PARTY	DISPLAY OF CALLED PARTY
NAMA DDGA	NAMA DDGA	Name of Called Party Called Party's Number	Name of Calling Party Calling Party's Number
NAMA DDGA	NAMD DDGD	X X X X Called Party's Number	Name of Calling Party Calling Party's Number
NAMD DDGD	NAMA DDGA	Called Party's Name Called Party's Number	X X X X -----
NAMD DDGD	NAMD DDGD	X X X X Called Party's Number	X X X X -----
NAMA DDGD	NAMD DDGA	X X X X Called Party's Number	Name of Calling Party -----
NAMD DDGA	NAMA DDGD	Called Party's Name Called Party's Number	X X X X Calling Party's Number

Figure 13 illustrates the functionality of Display of Calling Party Denied (DPD) on a nodal basis when both the calling and called parties have enabled DPD Class of Service options. The calling party has configured a Digit Display Denied (DDGD) and Name Display Denied (NAMD) Class of Service. The called party has the same Class of Service options programmed. The display of the calling party's associated name and number is restricted on the called party's set. The calling party's number is replaced by a string of dashes (— — —), where each "—" represents a suppressed digit. The associated name is replaced by a string of four Xs (XXXX).

On the calling party's set, the name associated with the called party is replaced by a string of four Xs (XXXX). However, the dialed digits remain displayed on the calling set.

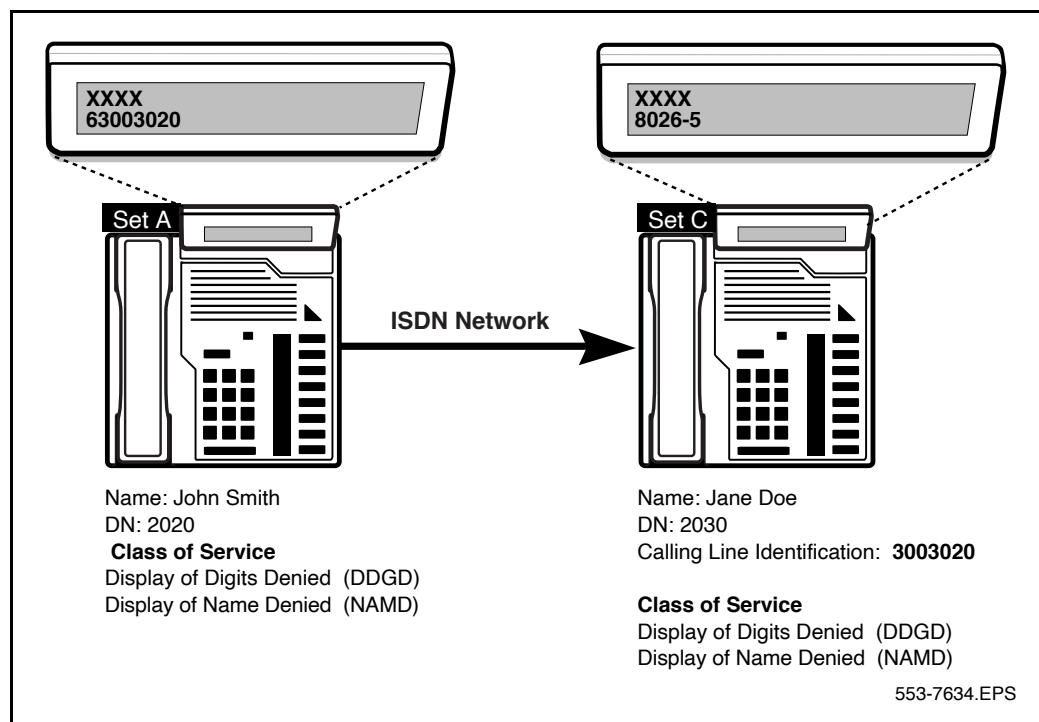
Figure 13
Calling Party and Called Party on the Same Node



As illustrated in Figure 14, for calls placed over a MCDN ISDN network, Display Digits Denied (DDGD) Class of Service on the calling party's set prevents the Calling Line Identification (CLID) from being displayed on the called party's set. In this case, the calling party's CLID is replaced by the ISDN route access code and route member number. If the calling party has configured Name Display Denied (NAMD), the calling party's associated name is not displayed on the called party's set. The associated name is replaced by a string of four Xs (X X X X).

On the calling party's set, the name associated with the called party is replaced by a string of four Xs (X X X X). However, the dialed digits remain displayed on the calling set.

Figure 14
ISDN Network Calls



Operating parameters

Display of Calling Party Denied (DPD) is only supported on internal calls, Meridian Customer Defined Network (MCDN) ISDN networks and the interface to the public network.

This feature does not support calls over DASS/DPNSS1 trunks.

Display of Calling Party Denied is not supported on ISDN BRI sets.

The dialed digits always appear on the display of the calling party's set. This occurs whether or not the called party has a Digit Display Denied (DDGD) Class of Service. However, if the dialed digits represent a Multiple Appearance Directory Number (MADN), the calling party's display reflects the Class of Service of the answering set.

Feature interactions

Attendant Console

An Attendant Console's display is not impacted by this feature. Display is provided to an attendant regardless of the Class of Service configured on a set. In order to allow calls placed over an ISDN network to be displayed on the Attendant Console, Network Attendant Service (NAS) must be configured.

Call Detail Recording

For internal calls, calling and called party's Directory Numbers are included in Call Detail Recording (CDR) records regardless of the set's Class of Service. For MCDN ISDN network calls, the calling party's or connected party's number is included in the CDR regardless of the set's Class of Service.

Call Forward All Type

When a set activates any of the call forwarding features, the displays given on the calling set and the terminating set are in accordance with the Class of Service of the sets involved in the call.

If the terminating set has Dialed Name Display Denied (DNDD), the display on the terminating set reflects the name and number of the calling party and the name and the number of the forwarding set.

If the terminating set has Dialed Name Display Allowed (DNDA), the display on the terminating set reflects the number of the calling party and the name and number of the forwarding set. In both cases, the terminating set's display is in accordance with the DPD Class of Service options of the calling and forwarding sets.

For a MCDN ISDN call, the calling party's Calling Line Identification (CLID) is replaced with the ISDN route access code (ACOD) and the route member number, and the calling party's name is replaced by a string of four Xs (X X X X).

The display given on the calling set of an internal call, which has been forwarded to a set within the same switch, includes the name and number of the terminating set along with the number of the forwarding set. If the DPD Class of Service options, which are specified for the terminating set, indicate that the display of the name and number of the terminating set be denied, then on the calling set, the name of the terminating set is replaced by a string of four Xs (X X X X). The number is replaced by dashes (- - -). If the number of the terminating set is blocked from being displayed on the calling set, the number of the forwarding set is also blocked from being displayed on the calling set, regardless of the DPD Class of Service options of the forwarding set. Conversely, if the display of the terminating set's number is allowed in the calling set, then the number of the forwarding set is also displayed on the calling set, irrespective of the DPD Class of Service options of the forwarding set.

Call Hold

When a call is retrieved from hold, the calling and called parties' displays reflect their individual DPD Class of Service options.

Call Park

When the Call Park timer expires on a parked call, a set's display reflects the Directory Number the call is parked against. The display does not include the name and DN of the calling party. When a parked call is retrieved by another set, display information is based on the DPD Class of Service of the individual sets.

Call Pickup

When a call is picked up from another set, the terminating set's display is in accordance with the Class of Service of the dialed and calling sets. The calling party's display includes the dialed DN, the terminating DN and the name of the terminated set. However, if the terminating set has Digit Display Denied (DDGD), then both the dialed and terminating sets' DNs are blocked from the calling party's display. The same occurs when Digit Display Allowed (DDGA) is configured on the terminating set. Both the dialed and terminating sets' DNs are displayed on the calling party's set, regardless of the Class of Service of the dialed set.

Call Transfer

When a set transfers a call, display information is updated according to the Class of Service of the respective sets. This occurs for both internal and ISDN network calls.

If an unsupervised call transfer occurs on an internal call, the DN of the terminating set is displayed to the calling party regardless of the DPD Class of Service options that are configured on the terminating set.

Calling Party Name Display

If Calling Party Name Display Denied (CNDD) is configured for a set, then the name associated with that set is not displayed when it is involved in a call. This is so regardless of the DPD Class of Service Options of that set.

Calling Party Privacy

Calling Party Privacy takes precedence over Display of Calling Party Denied on outgoing calls over a MCDN ISDN network. This precedence pertains to the display of the calling party's name and number on the called party's set.

Calling Party Privacy does not affect internal calls using Display of Calling Party Denied.

Conference

For internal conference calls, display information is not provided on any of the conferee's sets. When setting up a conference call, by conferencing one set at a time, the display on the conferee's set is in accordance with the Class of Service of the individual sets.

For network conference calls, display is provided. This is in accordance with the DPD Class of Service options of the individual sets.

Dial Intercom

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 (– –).

Group Call

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.

Hot Line

Display information on sets in a Hot Line call is based on the individual Class of Service of each set.

Meridian 911

An incoming M911 call with Automatic Number Identification (ANI) information always displays ANI digits on the terminating set regardless of the calling party's DPD Class of Service.

Meridian Mail

A calling party's name and/or DN within Meridian Mail are not impacted by this feature. When a Digit Display Denied (DDGD) call is forwarded to Meridian Mail, the calling party's DN is provided to Meridian Mail and is available when reviewing messages.

Multiple Appearance Directory Number

When a Multiple Appearance Directory Number (MADN) is dialed and ringing, the calling party's display does not show name information, provided that at least one of the MADN TNs has Name Display Denied (NAMD) Class of Service. Once a call is answered, the calling party's set reflects the DPD Class of Service options of the answering set.

The called party's display reflects the DPD Class of Service options of the calling party's set.

Multi Party Operations - Call Join

When three parties are joined using the Call Join capabilities of the Multi Party Operations feature, display information is not provided on any of the conferee's sets. When setting up a conference call, by conferencing one set at a time, the display on the conferee's set is in accordance with the individual set's Class of Service. If one set leaves a three party conference, display information on the remaining sets is based on the individual Class of Service of each set.

Network Attendant Services

When a call is placed to an attendant over an ISDN network with Network Attendant Services (NAS) configured and Calling Party Name Display (CPND) equipped, the Attendant Console displays the calling party's DN and name. The same occurs when an attendant places a call over the network; however, an Attendant Console's display shows the connected party name and number.

If NAS is not configured, the name and calling/connected party's number is displayed to the attendant provided that the calling/connected party has the Display of Calling Party Denied (DPD) Class of Service options configured to Allowed. If Class of Service options are set to Denied, then this information is not displayed to the attendant.

No Hold Conference Voice Call

Display information on sets involved in a No Hold Conference and Voice call is based on the individual Class of Service of each set.

Feature packaging

For internal calls, Display of Calling Party Denied (DPD) requires Calling Party Name Display (CPND) package 95.

For ISDN Network calls the following packages and their dependencies are also required:

- Calling Party Name Display (CPND) package 95;
- Integrated Service Digital Network (ISDN) package 145; and
- Network Attendant Services (NAS) package 159.

Note: The NAS package is required to display a calling party's name and DN on the attendant console, when using the Display of Calling Party Denied feature within an ISDN PRI private network. With NAS configured and Calling Party Name Display equipped, the attendant console will display the calling party's name and DN, regardless of the Display of Calling Party Denied Class of Service of the calling party's set. Without NAS, if the Display of Calling Party Denied Class of Service of the calling party's set is Denied, the name and DN of the calling party will not be displayed on the attendant console.

Feature operation

There are no operating procedures specified for this feature.

E.164/ESN Numbering Plan Expansion

The E.164/ESN Numbering Plan Expansion feature provides the capabilities to meet the International Telegraph and Telephone Consultative Committee (CCITT) recommendation E.164 for Integrated Services Digital Network (ISDN) and Public Switching Telephone Network (PSTN) dialing.

Generally, the following enhancements are offered by this feature:

- the support of the numbering plan for ISDN and PSTN dialing has been increased to a 15-digit maximum, from a 12-digit maximum;
- the removal of the leftwise-unique restriction imposed on ESN Supplemental Digit Restriction and Recognition (SDRR) entry codes. For example, if 555 is an existing entry, a new entry of either 55 or 5551212 can be entered;
- the addition of “allow” (ALLOW) as a new Supplemental Digit Restriction and Recognition (SDRR) entry type in LD 90. There are now nine entry types for Supplemental Digit Restriction and Recognition. One entry type (DENY), denies a call from going through if the digits in the SDRR entry match the dialed digits. There are seven other entry types that provide special treatments, if the digits in the SDRR entry match the dialed digits. The new ALLOW entry allows a call to go through, if the dialed digits did not match any entry within the Supplemental Digit Restriction and Recognition table;
- the Special Number Translation (SPN) screening digits scheme is changed; and
- the capability of the Meridian 1 base features to store digits is increased to 31 digits.
- the capability to send up to 32 digits to the Extended Conference and Tone & Digit Switch card, allowing an International Number to be sent in a single Automatic Number Identification (ANI) message. Previously, only 16 digits could have been outpulsed in a single ANI message, which included the start digit, the end digit, and an optional information digit. This left only 13 or 14 digits of the actual International Number to be outpulsed in a single ANI message. This meant that the International Number may have been sent in two separate ANI messages.

The specific enhancements delivered by the E.164/ESN Numbering Plan Expansion feature are provided in the tables that follow.

Summary of new ESN functionalities

Table 9 summarizes the changes made to the ESN functionality. Please note that the components that are not affected are not listed.

Table 9
Summary of ESN changes

Component	Old	New
Maximum number of digits for a Special Numbering Translation screening.	11	19
Maximum number of Digit Manipulation Index (DMI) deletion digits.	15	19
Maximum number of Digit Manipulation Index (DMI) insertion digits.	24	31
Maximum number of Supplemental Digit Restriction and Recognition (SDRR) tables, with BARS.	256	1500
Maximum number of Supplemental Digit Restriction and Recognition (SDRR) tables, with NARS.	512	1500
Maximum number of words per Supplemental Digit Restriction and Recognition (SDRR) entry.	3	4
Maximum number of digits in each Supplemental Digit Restriction and Recognition (SDRR) entry.	7	10
Maximum length of Flexible Numbering Plan (FNP) Flexible Digit Number Length (FLEN) numbers for Special Number Translation (SPN).	16	24
Maximum length of Flexible Numbering Plan (FNP) Flexible Digit Number Length (FLEN) numbers for Trunk Steering Codes (TSCs).	16	24
Maximum number of digits per Free Special Number Screening (FSNS) Special Number.	11	19
Total number of digits for screening under Free Special Number Screening (FSNS).	14	22

Maximum number of possible Supplemental Digit Restriction and Recognition (SDRR) entry types.	8	9 ('ALLOW' is added). ALLOW is a new entry type in LD 90. This entry allows a call to go through, as if the dialed digits did not match any entry within the Supplemental Digit Restriction and Recognition table.
Restriction imposed on Supplemental Digit Restriction and Recognition (SDRR) entry codes.	Leftwise Unique.	None. The leftwise-unique restriction, imposed on SDRR entry codes, is removed. For example, if 555 is an existing entry, a new entry of either 55 or 5551212 can be entered.

Summary of Meridian 1 base features enhancements

Table 10 summarizes the changes made to the base Meridian 1 features functionality. Please note that the components that are not affected are not listed.

Table 10
Summary of Meridian 1 base features changes

Component	Old	New
Maximum number of digits for an Autodial entry. Note that this enhancement does not apply to M3000 sets.	23	31
For the No Hold Conference functionality, the maximum number of digits for a Conference Autodial target DN entry.	23	31
Maximum number of digits for the Call Forward All Calls DN. Note that this enhancement does not apply to M2317 and M3000 sets.	23	31
Maximum number of digits for the Internal Call Forward DN.	23	31
For the Phantom DN functionality, the maximum number of digits for the Default Call Forward DN.	23	31
Maximum number of digits for the Customer Call Forward DN.	16	23
Maximum number of digits sent in an ANI message.	16	32

Summary of Automatic Call Distribution (ACD) enhancements

Table 11 summarizes the changes made to the ACD functionality. Please note that the components that are not affected are not listed.

Table 11
Summary of Meridian 1 ACD changes

Component	Old	New
Maximum number of digits for the ACD Night Forward DN.	23	31
Maximum number of digits for the ACD Interflow DN.	23	31

Summary of Customer Controlled Routing (CCR) enhancements

Table 12 summarizes the changes made to the Customer Controlled Routing functionality. Please note that the components that are not affected are not listed.

Table 12
Summary of CCR changes

Component	Old	New
Maximum number of Termination DN digits in the CCR ITR To-Route-A-Call message.	24	31

Summary of ISDN features enhancements

Table 13 summarizes the changes made to the ISDN functionality. Please note that the components that are not affected are not listed.

Table 13
Summary of ISDN changes

Component	Old	New
For ISDN BRI basic call service line access (BRIL), the maximum number of Called Party Information Element digits.	24	31
For the ISDN QSIG interface, the maximum number of Called Party Information Element digits.	24	31
For the EuroISDN interface, the maximum number of Called Party Information Element digits.	20	31
For the ISDN BRI NI-1 Call Forward All Calls feature, the maximum number of digits for the forward DN.	20	31
For the ISDN BRI ETSI Call Forward All Calls feature, the maximum number of digits for the forward DN.	20	31
For the Overlap Signaling feature, the maximum number of digits that a user may dial before a SETUP message is sent over the D-channel.	16	24

Operating parameters

All existing operating parameters that apply to the Meridian 1 ESN functionality, base system features and ISDN PRI and ISDN BRI networking features apply to the E.164/ESN Numbering Plan Expansion feature.

The enhancements introduced by the E.164/ESN Numbering Plan Expansion feature, pertaining to Called Party Information Element digits (CDPN), and forwarded and redirected DNs do not pertain to the M5317 and M5209 ISDN BRI terminals. This is because these terminals follow the BRI ETSI standard, placing a maximum limit of 24 digits on Called Party Information Element digits and redirected DNs.

The 31-digit expansion, pertaining to the Call Forward All Calls DN, does not apply to Meridian M2317 and M3000 sets, since their firmware limits entering a CFAC DN from the set to a maximum of 23 and 28 digits, respectively.

An Autodial key cannot be configured on a Meridian M3000 set; instead, the M3000 sets are equipped with a private directory, which allows automatic dialing of a stored DN entry by touching the corresponding name on the set's display screen. Since the firmware of the M3000 limits the length of a DN entry to a maximum of 28 digits, the 31-digit expansion pertaining to the Autodial does not apply to Meridian M3000 sets.

Since the Network Call Trace feature can outpulse a maximum of 21 digits to be printed on the TTY, the 31-digit expansion does not apply to Network Call Trace.

The 16 digit maximum, pertaining to the Calling Party Number, Connected Party Number, Redirecting Number and Redirection Number, has not been expanded by the E.164/ESN Numbering Plan Expansion feature.

If all the digits of a DN are dialed automatically, such as for a Speed Call or for any redial feature, only the first 31 digits are processed. This means that any digits that are inserted by, for example, network signaling, pretranslation or digit manipulation, are counted as part of these 31 digits. Therefore, even fewer than the automatically dialed digits may be processed.

Flexible Feature Codes (FFC) DNs may have a length of 1-7 digits. Call Forward All Calls (CFAC), Internal Call Forward All Calls (ICF) and Remote Call Forward (RCFW) are supported by FFCs, and can be activated, deactivated and verified using FFCs. When activating CFAC, using the Remote Call Forward FFC (RCFA), or when verifying the Call Forward DN, using the Call Forward All Call FFC (CFWV), the Remote Call Forward FFC (RCFV), or the Internal Call Forward All Calls FFC (ICFV), the FFC DN is kept, using eight of the available 31 digits. For the CFAC DN or ICF DN, this means that only 23 possible digits can be stored or verified using FFCs. As a result, this limitation does not accommodate the 31 digit CFAC DN and ICF DN expansion, introduced by the E.164/ESN Numbering Plan Expansion feature.

Calling Party Privacy (CPP) allows a caller to block his or her name from appearing on the called party display. The call is marked as a CPP call by entering a CPP FFC, up to seven digits long. This means that up to seven additional digits may have to be stored for automatic dialing. These digits are stored in sequence, and subsequent digits may not be processed.

In the case of CPP outpulsing, the CPP outpulsed digits (typically *67 or 1167) are stored after CPP has been activated. If this call goes over a trunk, the CPP outpulsed digits are sent to the far end to indicate that the call is a CPP call. For ISDN calls, a CPP flag is marked in the ISDN message. For non-ISDN CPP calls, digits that are already stored, such as for Call Forward, Autodial, and Speed Call, may be pushed out when the CPP digits are inserted, since only 31 digits can automatically be processed. This means that up to four digits may be removed from the digits that are outpulsed.

The SPN expansion, provided by the E.164/ESN Numbering Plan Expansion feature, applies to the DPNSS1/DASS2 UDP Interworking feature.

The System Access Enhancement (SAE) feature impacts the accepted values for the Call Forward All Calls DN, and the Internal Call Forward DN, when entered in LD 10 and LD 11. This feature allows all values between 4-31 to be accepted, rather than the normal accepted values of 4, 8, 12, 16, 20, 24, 28, 31. Also, the default is 4, rather than 16.

The Called Party Information Elements expansion, provided by the E.164/ESN Numbering Plan Expansion feature, applies to ISDN QSIG GF Transport feature.

Feature interactions

The E.164/ESN Numbering Plan Expansion feature does not interact with the Meridian 1 base system features and ISDN PRI and BRI networking features, other than as described in Table 9 to Table 13 and in the operating parameters.

Feature packaging

The E.164/ESN Numbering Plan Expansion feature does not introduce a new package. However, the following packages are required to support the new expansions.

For ESN expansions:

- Basic Alternate Route Selection (BARS) package 57;
- Network Alternate Route Selection (NARS) package 58; and
- Flexible Numbering Plan (FNP) package 160.

For ISDN expansions:

- Digit Display (DDSP) package 19;
- Integrated Services Digital Network (ISDN) package 145;
- 2.0 Mbit Primary Rate Interface (PRI2) package 154;
- Integrated Service Digital Network Signaling Link (ISL) package 147;
- Overlap Signaling (OVLP) package 184;
- International Primary Rate Access (IPRA) package 202;
- Meridian 1 Extended Peripheral Equipment (XPE) package 203;
- Basic Rate Interface (BRI) package 216;
- Multi-purpose Serial Data Link (MSDL) package 222;
- Integrated Services Digital Network Basic Rate Interface Trunk Access (BRIT) package 233;
- Basic Rate Interface Line Application (BRIL) package 235;
- EuroISDN (EURO) package 261; and
- QSIG Interface (QSIG) package 263.

For base system features expansions:

- Optional Features (OPTF) package 1;
- 2500 Telephone Features (SS25) package 18;
- Automatic Call Distribution (ACD) package 40;
- Automatic Call Distribution Advanced (ACD) package 41;
- Automatic Call Distribution Basic (ACD) package 45;
- 500 Telephone Features (SS5) package 73;
- Controlled Class of Service (CCOS) package 81;
- Background Terminal (BGD) package 99;
- Flexible Feature Codes (FFC) package 139; and
- Phantom Terminal Numbers (PHTN) package 254.

Feature operation

There are no operating procedures specified for this feature.

Electronic Lock Network Wide/Electronic Lock on Private Lines

The basic Electronic Lock feature has been enhanced to provide the following capabilities:

- The feature can be implemented network wide.
- The capability to define (for the locked sets), on a per customer basis, a Controlled Network Class of Service (CNCS) different from the Network Class of Service (NCOS), which is defined on a per set basis.
- Locking can be implemented for Private DNs.

In a Meridian Customer Defined Network (MCDN) environment, Electronic Lock Network Wide can be used to change the Class of Service of a set in a remote location. Electronic Lock Network Wide is activated or deactivated from any node by dialing the Electronic Lock Flexible Feature Code (FFC), the password, and the DN of the set to be changed. Since the password length defined at the destination node is not known at the originating node, the Station Control Password (SCPW) length (defined in LD 15) must be defined the same for all network nodes.

If the originating node has the FFC Confirmation Tone option selected, a confirmation tone is given when the feature is successfully activated or deactivated. Overflow tone is given if the operation is unsuccessful. There is no FFC verify code for Electronic Lock.

When a locked set makes an outgoing trunk call, if Controlled Network Class of Service is defined, the Network Class of Service (NCOS) defined by CNCS is used instead of the NCOS defined in LD 10 or 11 for the set. If network signaling is configured for the trunk that normally transmits the NCOS of the set between Electronic Switched Network (ESN) nodes, the CNCS is transmitted instead of the NCOS. This prevents a locked set from reaching the exchange network by tandeming through a TIE trunk using ESN.

A new prompt (PELK) is introduced in the Customer Data Block to implement Electronic Lock on private lines. If this option is enabled, an outgoing call on a private line of a locked set is subject to the same restrictions as all other DNs on the set. The same intercept treatment would be given as for a regular DN. The restrictions for private lines, as well as other DN keys on the set, are controlled by the Controlled Class of Service (CCOS), and by the CNCS if defined. Therefore, for outgoing calls, the Class of Service restrictions and/or New Flexible Code Restriction (NFCR) apply to private line keys on locked sets. Only outgoing calls are affected. The Class of Service of a non-locked set has no affect on private lines.

Operating parameters

The Network Dialing Plan must be either a Coordinated Dialing Plan (CDP) or a Uniform Dialing Plan (UDP).

The set password lengths must be equal for all nodes in the network.

Network wide operation is only supported through an MCDN ISDN network.

The Electronic Lock feature must be equipped on both originating and remote nodes.

The FFC used is defined on the node from which Network Electronic Lock is being activated. To activate or deactivate Network Electronic Lock from any remote node, the user has to use the FFC Electronic Lock Activate (ELKA) or Electronic Lock Deactivate (ELKD) code defined on that remote node.

ISDN Basic Rate Interface (BRI) sets cannot be used to lock another set, nor can they be locked themselves.

An analog (500/2500 type) telephone with a private line DN or a Meridian Digital Telephone set with a private line on the Prime DN cannot be locked.

The following hardware is required:

- Primary Rate Interface – D-channel Handler Interface (DCHI)/ Multipurpose Serial Data Link (MSDL) and PRI/PRI2 cards.
- Integrated Services Digital Network Signaling Link (ISL) – DCHI cards and TIE trunks.
- Virtual Network Services (VNS) – DCHI cards and any trunks.

Feature interactions

Automatic Call Distribution (ACD)

An ACD set cannot be locked.

Call Forward (CFW)

Call Forward No Answer (CFNA)

For Call Forwarding, the COS and NCOS used for the forwarding call can be taken from either the forwarding set or from the forwarded set, depending on the option defined in the Customer Data Block.

For example, set B call forwards all calls to an external trunk. Set A calls set B. If OPT = CFF in LD 15 (Call Forward forwarded to party's COS and NCOS), the COS and NCOS of set B are used for forwarding the call to the trunk. If OPT = CFO (Call Forward originating party's COS and NCOS), the COS and NCOS of set A are used for forwarding the call to the trunk.

Direct Inward System Access (DISA)

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

Digital Private Network Signaling System (DPNSS1)

Digital Access Signaling System (DASS2)

Analog Private Network Signaling System (APNSS)

Electronic Lock Network Wide is not supported on DPNSS1, DASS2, or APNSS trunks.

Electronic Switched Network Authcode

If a station user enters an authcode on the set, regardless of the status of the set being locked or not, the NCOS defined for the authcode is used. The ESN Authcode feature overrides the Electronic Lock Network Wide feature.

Flexible Numbering Plan

If a network is equipped with a flexible numbering plan (i.e., not all the network DNs are the same length), hanging up before the usual end-of-dialing timeout cancels the request for activation or deactivation of Electronic Lock. Dialing an octothorpe (#) after the network DN will cause the request for activation or deactivation of the Electronic Lock to be sent immediately, instead of waiting for the usual end-of-dialing timeout to send it.

Multiple Appearance DN

The same locked or unlocked state applies to all Terminal Numbers with the same primary DN and the same SCPW. Terminal Numbers with the same DN, but not having the same SCPW, cannot be locked or unlocked.

New Flexible Code Restriction (NFCR)

With NFCR, toll denied stations are allowed or denied calling privileges according to the Facility Restriction Level (FRL) assigned to the NCOS defined in the protected line block. For a locked set, NFCR uses the FRL assigned to the CNCS to determine its calling privileges if one is defined; if no CNCS is defined, the NCOS of the locked set will be used.

Scheduled Access Restrictions (SAR)

The SAR feature overrides Electronic Lock.

Virtual Network Services (VNS)

Electronic Lock can function in a VNS environment.

Feature packaging

Electronic Lock Network Wide/Electronic Lock on Private Lines is packaged under Flexible Feature Codes (FFC) package 139.

Use of this feature requires the following additional packages:

- Controlled Class of Service (CCOS) package 81
- Network Class of Service (NCOS) package 32, and
- Integrated Services Digital Network (ISDN) package 145.

Error Handling on ISDN Signaling Link Analog E&M TIE Trunks

The Integrated Services Digital Network Signaling Link (ISL) Analog E&M TIE Trunks feature identifies error conditions that are generated by external equipment and ensures that faulty trunks cannot be used. When the error condition is removed the trunk is put back into service.

When an error condition is identified while the trunk is idle, the trunk is put into a maintenance busy mode and a trunk error message is printed. If an error occurs while a call is being sent, an error message is printed and the call proceeds; the trunk unit is put into maintenance busy mode after the call is disconnected. Outgoing calls cannot be made on units that are in maintenance busy and incoming calls are rejected.

Once the external equipment removes the error condition, a message is printed and a restart message is sent on the D-channel to indicate that the unit is usable.

Operating parameters

Error Handling on ISL Analog E&M TIE Trunks only functions on ISL Analog E&M TIE Trunks and not on other types of ISL TIE trunks. This function only operates when the D-channel is established.

E&M types applicable to this feature include two-wire and four-wire with TY1, TY2 and British Post Office E&M signaling types. The feature applies to immediate start, delay dial start and wink start arrangement types.

This feature is designed to operate only when ISL E&M TIE trunks are used in conjunction with transmission equipment.

To implement Error Handling on ISL Analog E&M TIE trunks the FALT prompt response in the Route Data Block (LD 16) must be YES. If FALT=NO, then no error handling will occur on the route.

Error handling exists on ISDN Signaling Link Digital Trunk Interface (2 Mbps).

Feature interactions

There are no interactions with other features.

Feature packaging

Error Handling on ISL Analog E&M Tie Trunks is ISDN Signaling Link (ISL) package 147 and is included in ISDN Signaling (ISDN) package 145.

Feature operation

Overlay Changes

The enabling/disabling commands in the maintenance overlay is changed so that if the trunk unit is in error state the following occurs:

- The trunk unit cannot be enabled until the external error condition is removed. The message TRK533 is printed to warn the user that the enable command is cancelled because an error exists on the ISL E&M TIE Trunks. When this occurs, the trunk unit will remain in maintenance busy mode.

These overlay changes ensure that error states on both sides of the link are not affected by enabling/disabling the unit.

Error Applied to Unit

- An error condition can be generated by the external equipment when a unit is idle, during call setup or when a call is established.

Error applied, Unit Idle

- If an error is detected the unit is put in maintenance busy mode and the error message TRK531 is printed.

Error applied, Call Setup

- If an error is detected during call setup the error message TRK531 is printed.
- After the call is disconnected, the trunk is put into maintenance busy mode.

Error applied, Call Established

When an error is applied while a call is established, it is detected after the call is disconnected. If the Release message is not received within the time defined by the supervision timer then the E lead remains grounded and causes an error condition. The following occurs:

- The unit is put into maintenance busy mode and the error message TRK531 is printed.

Error State

When the unit is in an error state then it is not available for incoming or outgoing calls. The error state is maintained until it is cleared by external equipment.

Outgoing Calls

Normal call processing selects the next available channel for outgoing calls.

Incoming Calls

Far end attempts to make calls will send a SETUP message. If the unit is in an error state, a Release Complete message is sent. The far end trunk will be put into a LOCKOUT state.

Restart Message

During an error state, a RESTART message is received from the far end. This message is ignored so that the error state of the unit is preserved.

Error Removed

If the error condition is removed by external equipment, the trunk is marked IDLE, the error message TRK532 is printed and a RESTART One Channel message is sent on the D-channel. The far end trunk unit changes from LOCKOUT to IDLE state.

No specific operating procedures are required to use this feature, however the following system messages will occur throughout feature operation.

Message Number	Events
TRK531 tn	An error has been detected on ISL E&M TIE, unit will be put to Maintenance Busy.
TRK532 tn	Error Condition on ISL E&M TIE removed.
TRK533 tn	Enable command is cancelled on this unit because an error exists on the ISL E&M TIE trunks.

ESN on PRI/ISL

Description

Generic X11 Release 16 provides most of the current ESN services over ISDN DTI/PRI with the exception of the Network Call Transfer (NXFER), the Coordinated Call Back Queuing (CCBQ), the CBQ to Conventional Main (CBQCM) and Off-Hook Queuing (OHQ) features.

Generic X11 Release 16 provides Network Class of Service (NCOS) information which extends ESN functionality to DTI/PRI facilities. NCOS is an ESN software capability which provides the means to control:

- a** User access to routes
- b** Eligibility of queuing for a user
- c** Eligibility for user to receive Expensive Route Warning Tone (ERWT)
- d** Eligibility to access Network Speed Call (NSC)

NCOS

A NCOS is assigned to every user on the Meridian 1/SL-1 PBX. Depending on the NCOS assigned, users are either allowed or denied the placing of local, long distance and/or International calls. During a tandem call, DTI/PRI sends the NCOS of the user across the DTI/PRI link prefixed to the dialed number to the other Meridian 1/SL-1. This allows network access control to be based on a consistent set of NCOSs for calls originating anywhere on the network.

NARS

Northern Telecom's ISDN/PRI networking feature development is based on the signaling and functionality within the Network Automatic Route Selection (NARS) and/or Coordinated Dialing Plan (CDP) features.

The NARS feature provides the special digit processing to distinguish between private and public network calls within the first 2 to 3 digits dialed. This allows the NARS equipped switch to process the call immediately upon dialing the last digit thus resulting in instantaneous sending of the call setup message.

CDP

CDP uses Distant Steering Codes (DSC) and Trunk Steering Codes (TSC). DSC codes are recognized in software and also provide instantaneous cut-through. However, TSC and the number that follows varies in length, therefore end-of-dialing timers are used.

ISDN with CDP or NARS also provides the exact display of the CLID across the network. However, when connecting a number of switches with some switches using CDP and the other switches using NARS, the correct display of the CLID may not make sense because the NARS switches will be sending the location code plus the extension number, whereas the CDP switches will be sending only the CDP number. In some networks, the CDP number does not contain enough information to be able to dial back the calling number if necessary.

NRAG

Option 148 (Advanced Network Features) is the option that provides Network Ring Again (NRAG). Because of NRAG and other features (NACD, NMS, TRO) planned in future X11 software Releases; as well as the complexity of inserting and deleting digits in the number dialed in an all CDP or NARS or combination CDP/NARS network, Northern Telecom requires that every switch in the network be equipped with either NARS or CDP software.

With the CDP Expansion enhancement, X11 Release 15 and later ISDN networks can support a 3 to 10 digit Coordinated Dialing Plan (CDP). Prior to X11 Release 15, a 3 to 6 digit ISDN CDP was supported.

INAC

The Automatic Insertion of ESN Access Codes feature (INAC) provides greater flexibility in designing network numbering plans. This enhancement allows either AC1 or AC2 to be used as the access code for private network and long distance calls. Previously only AC1 could be used for these types of calls. Advanced Networking Features such as NACD and NMS require that INAC be implemented, rather than using Digit Manipulation Tables to insert the appropriate ESN Access Code.

Package Requirements

This option requires the following feature packages:

Feature	Package Number
ISDN	145
PRI2 or ISL	154 or 147
NARS	Option 58 and/or CDP Option 59 and NSIG Option 37 or Multi-Site 1 or 2 (see details on Multi-Site packaging 1 or 2 at the end of this section).

Equi-distribution routing, network attendant service

This feature provides a new algorithm for Network Attendant Service (NAS) routing. The algorithm attempts to distribute routed calls evenly among network attendants, based on loading levels. Calls are routed across a network according to NAS routing tables. These tables define the primary attendant node and alternatives in case the attendant at the primary node is busy and the call waiting threshold has been exceeded. In the case of all attendants busy, as routing is attempted to each alternate defined in the NAS routing table, the Equi-distribution NAS Routing feature reads the load level at each location. The call is then routed to the alternate with the lowest load level.

The load level is a value defined as the customer-defined efficiency factor, multiplied by the number of calls waiting for an attendant, divided by the number of attendants in service. The efficiency factor is a value between 0-8064. The lower the efficiency factor, the more efficient the attendants are considered to be at a specific location (however, if a value of 0 is entered, then the Equi-distribution NAS Routing feature is not activated, and normal NAS routing occurs).

Another capability of the Equi-distribution NAS Routing feature deals with an attendant at a node going into Night Service. All calls in the attendant queue are routed to another location, based on the NAS routing schedule, instead of to the local night DN.

Operating parameters

This feature applies only to Meridian 1 systems using Meridian Customer Defined Networking (MCDN) signaling over Integrated Services Digital Network Signaling Link (ISL)/ISDN TIE links.

All nodes in the Meridian 1 network should have Equi-distribution NAS Routing feature.

This feature may not be compatible with the Console Presentation Group (CPG) or Network Tenant Service (NTS) features.

Feature functionality takes into account a maximum number of four alternate locations per time period.

Feature interactions

When the attendant goes into Night Service, calls presented to the attendant receive NAS routing in an attempt to reach another attendant that is in day service, rather than being routed to the local night DN.

Feature packaging

Network Attendant Service (NAS) package 159, Network Alternate Route Selection (NARS) package 58, Integrated Services Digital Network Signaling (ISDN) package 145 or ISDN Signaling Link (ISL) package 147.

EuroISDN 7kHz/Videotelephony Teleservices

The 7 kHz/Videotelephony teleservices feature supports the 7 kHz teleservice, and the Videotelephony teleservice.

The **7 kHz teleservice** is an enhanced telephony teleservice offering better voice quality. It is a real-time teleservice in which a voice call uses one circuit-mode 64 kbit/s connection with 7 kHz bandwidth.

The **Videotelephony teleservice** is a real-time, audiovisual teleservice in which voice and moving pictures are interchanged by means of one or two 64 kbit/s circuit-mode connections.

The 7 kHz/Videotelephony teleservices are supported over EuroISDN PRI2 and Basic Rate Trunk interfaces, and on ETSI Basic Rate Interface sets.

This feature also supports the **fall-back** capability. Fall-back allows a request for the telephony 7 kHz teleservice or for the videotelephony teleservice, to use the alternate teleservice. In the case of the 7 kHz teleservice, the alternative is the 3.1 kHz teleservice. In the case of the Videotelephony teleservice, the alternative is the 7 kHz teleservice (if supported), or the 3.1 kHz teleservice.

This feature supports the following communications:

- videotelephone terminal to videotelephone terminal
- 7 kHz terminal to 7 kHz terminal
- videotelephone terminal to 7 kHz terminal or 3.1 kHz terminal, if fall-back allowed
- 7 kHz terminal to 3.1 kHz terminal, if fall-back allowed

Originating and terminating terminals can be local or remote, but calls using one of these teleservices with at least one remote terminal must be made over a EuroISDN PRI2 or EuroISDN BRI link.

When a user originates a call requesting a 7 kHz or videotelephony teleservice, the user can choose if fall-back is allowed or not, that is, whether or not the call is to be established with the alternative teleservice if the requested teleservice is not available. Such request can be made either from a local ETSI BRI set supporting the teleservice, or from a remote set whose request is received over a EuroISDN link. The terminating party may accept the call, if the teleservice is supported end to end, or reject the call, if the requested teleservice is not supported.

Note: Notice that a videotelephone terminal may not support the telephony 7 kHz teleservice.

7 kHz or videotelephony teleservice request with fall-back not allowed

If fall-back is not allowed, the call may be either established with the requested teleservice, or rejected.

If the call request reaches a non EuroISDN link, or if the requested set is a non-ETSI BRI set, the call is rejected. For analog (500/2500) sets and Meridian proprietary sets, the call is rejected. The call is also rejected for calls over a EuroISDN link and for ETSI BRI sets not configured to support the 7 kHz/Videotelephony teleservices (the 7VOD or 7DTA teleservice EuroISDN for routes, or the VID7 supplementary service for ETSI BRI sets).

7 kHz or videotelephony teleservice request with fall-back allowed

If fall-back is allowed, the call is established with the preferred teleservice or with the alternate teleservice.

If the call request reaches a non EuroISDN link, or if the destination set does not support the requested teleservice, the call is established with the alternate teleservice (3.1 kHz teleservice). The originator is notified that fall-back has occurred.

Establishment of a second connection for videotelephony calls

When a call is established with the videotelephony teleservice, a second connection can be established to provide a better video quality. Only the calling party can initiate a request for a second connection, and this only after the first connection is established.

Operating parameters

This feature is not supported over QSIG, DPNSS1, or MCDN networks. If a EuroISDN call gets routed over a QSIG, DPNSS1, or MCDN link, the call is not established with 7 kHz or videotelephony teleservice. If fall-back is allowed, call is established with the alternate (3.1 kHz) teleservice, or else the call is dropped.

This feature requires that the EuroISDN ETS 300-403 compliance update feature be implemented (IFC = E403 in LD 16).

Feature interactions

Call Forward All Calls

Fall-back may be applied, if Call Forward All Calls has been activated. If fall-back is not allowed, the call is treated as per normal operation.

If fall-back has already occurred, a call request for the alternate (3.1 kHz) teleservice is sent to the forwarded-to set or trunk. If fall-back has not already occurred, the call is treated as per normal operation for the forwarded-to set or trunk.

Call forward Busy

If Call Forward Busy feature has been activated on a busy set, the call is established with the alternate (3.1 kHz) teleservice, if fall-back is allowed. If fall-back is not allowed, the call is rejected.

Call Forward No Answer

When Call Forward No Answer has been activated on a set, the call is treated as per normal for the forwarded-to set or trunk, if fall-back is not allowed.

If fall-back is allowed and has already occurred, a call request for the alternate (3.1 kHz) teleservice is sent to the forwarded-to set or trunk. If fall-back is allowed and has not already occurred, the call is treated as per normal for the forwarded-to set or trunk.

Hunting

If Hunting has been applied to a busy destination, destination set is busy, and if Hunting feature is configured, the call is treated as per normal for the hunted-to set or trunk.

DISA call

For a A DISA call requesting 7 kHz or videotelephony teleservice, fall-back may occur, if allowed, or the else the call is rejected.

ACD DN

For request for 7 kHz or videotelephony teleservice terminating on an ACD DN, fall-back may occur, if allowed, or the else the call is rejected.

Advice On Charge

If a second connection is established for a videotelephony teleservice, the charge information is received separately for the two connections, since there is no way to identify that the two connections relate to the same videotelephony call.

Multiple Appearance Directory Number

An incoming call to a Multiple Appearance DN associated to several sets (non BRI sets) is presented to all sets. For an incoming call to a Multiple Appearance DN requesting the 7 kHz or videotelephony teleservice, fall-back may occurs if allowed, or else the call is rejected.

Network Alternate Route Selection

NARS route selection is done by looking at the Data Selection (DSEL) values of the different routes. An outgoing call requesting the 7 kHz or videotelephony teleservice can only use route defined with 7DTA or 7VOD at DSEL prompt, whether fall-back is allowed or not.

If fall-back is allowed and if no route has been found for the preferred teleservice, then fall-back occurs and a new route selection is started for the alternate teleservice.

If fall-back is allowed and occurs further in the network, a voice call may be established using a route defined for data (DSEL = 7DTA).

Customer Controlled Routing

For a call request for the 7 kHz or videotelephony teleservice terminating on a CCR Control DN (same as an ACD DN), fall-back occurs if allowed, or else the call is rejected.

Symposium Call Center Server

For a call request for the 7 kHz or videotelephony teleservice terminating on a Symposium Call Center Server, fall-back occurs if allowed, or else the call is rejected.

Meridian IVR

For a call request for the 7 kHz or videotelephony teleservice terminating on a Meridian IVR application, fall-back occurs if allowed, or else the call is rejected.

Meridian Link

For a call request for the 7 kHz or videotelephony teleservice terminating on a Meridian link application, fall-back occurs if allowed, or else the call is rejected.

Meridian Mail

For a call request for the 7 kHz or videotelephony teleservice terminating on Meridian Mail, fall-back occurs if allowed, or else the call is rejected.

Feature packaging

The following packages are required for the 7 kHz/Videotelephony teleservices feature:

For ISDN PRI2:

- EuroISDN (EURO) package 261
- Multi-Purpose Serial Data Link (MSDL) package 222
- International Primary Rate Access (IPRA) package 202
- ISDN Supplementary Features (ISDN_SUPP) package 161
- 2.0 Mbps Primary Rate Interface (PRI2) package 154
- Integrated Services Digital Network (ISDN) package 145

For ISDN Basic Rate trunking:

- EuroISDN (EURO) package 261
- Basic Rate Interface Trunking (BRIT) package 233
- Basic Rate Interface (BRI) package 216

- International Primary Rate Access (IPRA) package 202
- ISDN Supplementary Features (ISDN_SUPP) package 161
- Integrated Services Digital Network (ISDN) package 145

For ISDN Basic Rate line application:

- Basic Rate Interface Line (BRIL) package 235

Maintenance and diagnostics guidelines

Call Trace (LD 80)

LD 80 is used to gather information pertaining to a call. If the call originated or terminated over a ISDN trunk, the following information is printed, in addition to the basic call information:

- call reference number
- bearer capability
- call state
- calling number
- called number
- Virtual Network Services Directory Number, if applicable

New bearer capability print information

The new printed bearer capability information which is printed for the 7kHz/Videotelephony teleservices feature is 'UDI-TA', which pertains to the 'Unrestricted Digital Information with Tones/Announcement' Information Transfer Capability Information Element (IE).

Bearer capability print format when fall-back is not allowed

If fall-back is not allowed, there is only one bearer capability Information Element in the received SETUP message. The print format is not changed. The format is 'BEARER CAP = aaaaaa', where 'aaaaaa' = one of 'VOICE, 64KCLR, 64KRES, 56KRES, 3.1KHZ, DATAUN, UDI-TA, or DATARS'.

Bearer capability print format when fall-back is allowed

If fall-back is allowed, there are two bearer capability IE contained in the received SETUP message. There are two print formats for the bearer capability.

When the used teleservice has not yet been selected, both values are printed in the order that they are received. When the used teleservice has been selected (and whether fall-back occurs or not), only the selected value is printed.

Examples follow:

BEARER CAP = VOICE UDI-TA (if a bearer capability has yet to be selected)

BEARER CAP = UDI-TA (if fall-back has not occurred)

BEARER CAP = VOICE (if fall-back has occurred)

High Layer Compatibility print information

If a bearer capability 'UDI-TA' IE is received, then the High Layer Compatibility (HLC) information is printed on the line below the bearer capability information, whatever the value of the bearer capability.

The HLC print format is 'HLC = aaaaa', where aaaaa = VIDEO or TELE.

For other values, nothing is printed, since these values do not define a teleservice when the bearer capability receives a UDI-TA IE. If two HLC IEs are received, in the case of fall-back, the print format is the same as for the bearer capability print format.

No correlation between two connections for the same call

There is no messaging information within a EuroISDN network to correlate two connections established for the same call. The only way to gather information pertaining to two connections for the same call, is to analyze the calling and called number information associated with each connection.

Feature operation

No operating procedures are required to use this feature.

EuroISDN Continuation

EuroISDN provides Primary and Basic Rate Interfaces to Central Offices/Public Exchanges that comply with the specifications for Layer 3, as described in ETS 300-102. The EuroISDN Continuation feature expands the scope of the original EuroISDN feature to include application support for Switzerland, Spain, Belgium and the United Kingdom.

In addition, the following functionalities are provided by the EuroISDN Continuation feature:

- Intercept treatment upon reception of an invalid or incomplete called party number;
- Interception to an attendant for EuroISDN voice calls terminating on a data device;
- The capability to listen to tones and announcements provided by the Central Office on call clearing (T306 supported);
- The capability of transferring outgoing EuroISDN calls after completion of dialing (only for Italy);
- Calling Line Identification and Connected Line Identification transparency to or from EuroISDN to or from a Basic Rate Interface (BRI) set;
- The configuration of Connected Line Presentation (COLP) on a per D-channel basis (remote capability);
- 3.1 KHz audio bearer capability for outgoing fax calls, based on a Class of Service assigned to the Analog (500/2500 type) telephone;
- The capability of defining the bearer as “Voice” or “3.1 KHz” on a system basis;
- The addition and display of national or international prefix in front of the received Calling/Connected Line Identification on incoming/outgoing EuroISDN calls;
- Flexible national and local prefixes in addition to the Calling/Connected Line Identification on incoming/outgoing calls based on the route configuration;
- User-to-User information transparency in call control messages;

- Capability of mapping a PROGRESS message or Progress Indicator in a CALL PROCEEDING message into an ALERT or a CONNECT message on a per configuration basis;
- Capability of using different options to build the Calling Line Identification (CLID); and
- Called party number size increase to 31 octets.

This development also provides a subset of the ETSI Generic functional protocol for the support of supplementary services. Implementation relies on the Generic Functional (GF) Transport platform introduced in the X11 Release 22 ISDN QSIG GF Transport feature. The ETSI GF subset provides two types of supplementary service control entities as follows:

- **Bearer-related transport with a point-to-point transport mechanism**
This service is used to transport supplementary service protocol information in association with a basic call.
- **Bearer-independent transport with a point-to-point (connection-oriented) mechanism**
This service is used to transport supplementary service protocol information, which is entirely independent of any existing basic call.

The ETSI GF provides a generic transport platform that supports ETSI-compliant supplementary services. The ETSI GF protocol is implemented on the ETSI, Swiss, German and Danish EuroISDN interfaces.

Operating parameters

Layer 1 and Layer 2 compliance with ETSI requirements are also supported.

ISDN Primary Rate Interface (PRI) – The NT8D72BA PRI card provides the 30B+D physical interface to the Central Office/Public Exchange.

ISDN BRI – The NT6D70BA card provides a 2B+D trunk interface for Central Office/Direct Inward Dialing connectivity.

The EuroISDN feature is not supported on all downloadable D-channel cards (e.g., the NT6D11).

If more than one Channel Identification Information Element (IE) is received in a SETUP message, only the first one is used by the Meridian 1. If the Information Element is not available, the call is processed as per channel negotiation configuration. If no negotiation is allowed, the call is released.

The transfer of an unanswered EuroISDN call to a remote ringing set requires disconnect supervision from a TIE trunk. If disconnect supervision is not available when an external user goes on-hook, the trunk could be locked out. Therefore, it is important to ensure that trunks used for a EuroISDN call have disconnect supervision.

With ETSI GF, DN address translation requires association with a customer number. In an ETSI basic call establishment, the customer number association is located through the B-channel, identified in the channel identification IE. If the DN address is not associated with a basic call, the customer number is determined through other means.

For Connected Line Identification Presentation (COLP) supplementary service, sending or restricting connected line identification is not supported on ISDN BRI sets on a per call basis.

Feature interactions

Auto terminate Call

This feature is not supported.

Call Back Queuing and Off-hook Queuing

This feature is not supported with Overlap Signaling.

Call Completion Supplementary Service

Call Completion interacts with Call Transfer Over EuroISDN SN3. Call Completion Free Notification can only be presented to the Call Completion originating station and cannot be transferred to any other station. However, once the Call Completion call is established, it behaves as a normal call with respect to Call Transfer.

End-to-End Signaling

End-to-End Signaling is supported on all outgoing EuroISDN routes as soon as the CALL PROCEEDING message with a Progress Indicator is received.

Flexible Hotline

This feature is not supported with Overlap Signaling.

Incoming Digit Conversion Enhancement

The Incoming Digit Conversion Enhancement (IDC) feature converts incoming digits from a DID route. This feature is supported on the incoming EuroISDN DID routes. Digits received as a called party number are converted if the IDC feature is activated on the route. Digit analysis is then performed on the converted digits by the Meridian 1.

ISDN Basic Rate Interface Connected Line Presentation/Restriction

The EuroISDN Continuation capability adds National and Local prefixes to the connected number being sent. This is programmed on a route basis and is applicable to connected numbers received from a ISDN BRI terminal and sent over a ISDN trunk.

ISDN Calling Line Identification Enhancements

The EuroISDN Continuation feature allows Home National Numbers and Home Local Numbers to be configured on a route. When an ISDN call is made from a set to a EuroISDN interface, the Calling Line Identification (CLID) constructed by EuroISDN, based on the outgoing route, takes precedence over the CLID constructed for the calling station set.

KD3 Signaling

Interworking with KD3 signaling is not supported.

Integrated Service Access (ISA)

Integrated Service Access is not supported.

Interworking Notification

Direct Interworking

When “direct interworking” occurs, the EuroISDN trunk terminates directly on the analog trunk. As a result, the CALL PROCEEDING message is sent to indicate that no further called party information is required. No further call control message is initiated by the analog trunk until answer supervision is provided. If the trunk is a supervised trunk, the resulting CONNECT message will permit the Public Switched Telephone Network (PSTN) to start charging the caller. However, an unsupervised trunk may also be called. In this case, the CONNECT message is sent upon the expiry of the pseudo answer supervision timer.

The destination non-ISDN trunk may be busy instead of ringing; therefore, it is not effective to provide an ALERTING message, since the PSTN provides the caller with ringback. However, the PSTN will be running a T310 timer. Expiry of this timer causes call clearing, and a message must be provided.

The following three alternatives can be configured in LD 16 by responding to the PROG prompt to stop the network T310 timer:

- 1 A PROGRESS message is sent to the network after the CALL PROCEEDING message.
- 2 An ALERT message is sent to the network after the CALL PROCEEDING message.
- 3 A CONNECT message is sent to the network after the CALL PROCEEDING message.

Tandem Interworking

When tandem interworking occurs, the EuroISDN trunk terminates on an ISDN link terminating on an analog trunk. When the tandem SETUP message is sent, the CALL PROCEEDING message is sent to the PSTN to indicate that no further called party information is required. The tandem ISDN link returns a PROGRESS message, indicating interworking with a non-ISDN trunk.

Since the PSTN does not implement Annex N, proposed by ETSI, it runs a T310 timer. A message must be provided to stop the T310 timer.

The following three alternatives can be configured in PROG prompt LD 16 to stop the network T310 timer:

- 1 A PROGRESS message is sent to the network after the CALL PROCEEDING message.
- 2 An ALERT message is sent to the network after the CALL PROCEEDING message.
- 3 A CONNECT message is sent to the network after the CALL PROCEEDING message.

When interworking with an analog trunk without answer supervised signaling, the Meridian 1 should be configured to send a CONNECT message rather than an ALERT message. An ALERT or CONNECT message stops the receipt of digits, which implies that all digits have been received.

Networking Features

Some networking features currently exist on more than one ISDN interface implemented on the Meridian 1. These features are listed in Table 14. Any networking feature that does not appear in the table is only supported on one ISDN interface and is rejected by the gateway when the service is requested. That is the case for all Meridian Customer Defined Network (MCDN) features that are not supported over the EuroISDN or QSIG interface.

Table 14
Networking features supported by more than one ISDN interface

	Euro-ISDN	MCDN	QSIG	BRI sets	DPNSS	MFC	DTI2	Analog
Calling Line ID	Yes	Yes	Yes	Yes	Yes	No	No	No
Connected Number	Yes	Yes	Yes	No	Yes	No	No	No
Transit Counter	No	Yes	Yes	No	Yes	No	No	No

Special Dial Tone after Dialed Numbers

Special dial tones after dialed numbers are not supported for incoming calls.

Transfer of Unanswered Call

The EuroISDN Continuation feature supports the transfer of a call from a set dialing an external number, before the external set answers, to both local and remote sets upon ringing.

Virtual Network Services

A EuroISDN link can be used as a B-Channel for the Virtual Network Services feature.

R2MFC Calling Number Identification/Call Detail Recording Enhancements

The outgoing Calling Line Identification (CLID) element of the EuroISDN Continuation feature is mutually exclusive with the R2MFC CNI/CDR Enhancements feature. If the CLID is to be composed from the EuroISDN Continuation feature, it will not contain the Calling Number Identification (CNI). If the CLID is to be composed from the CNI, no prefixes will be added to the number.

Feature packaging

EuroISDN Continuation is included in EuroISDN (EURO) package 261.

For Primary Rate Interface (PRI), the following software packages are required:

- Digit Display (DDSP) package 19;
- Integrated Services Digital Network (ISDN) package 145;
- 2.0 Mbit Primary Rate Interface (PRI2) package 154;
- Integrated Services Digital Network Supplementary Features (ISDN INTL SUP) package 161;
- International ISDN PRI (IPRA) package 202; and
- Multi-purpose Serial Data Link (MSDL) package 222.

For Basic Rate Interface (BRI), the following software packages are required:

- Digit Display (DDSP) package 19;
- Integrated Services Digital Network (ISDN) package 145;
- Integrated Services Digital Network Supplementary Features (ISDN INTL SUP) package 161;
- International ISDN PRI (IPRA) package 202;
- Meridian 1 Extended Peripheral Equipment (XPE) package 203;
- Basic Rate Interface (BRI) package 216; and
- Integrated Services Digital Network Basic Rate Interface Trunk Access (BRIT) package 233.

Overlap Signaling is included in (OVLP) package 184. For the Uniform Dialing Plan (UDP), the following packages are also required:

- Basic Routing (BRTE) package 14;
- Network Call of Service (NCOS) package 32;
- Digit Display (DDSP) package 19;
- Basic Alternate Route Selection (BARS) package 57; and
- Network Alternate Route Selection (NARS) package 58.

For the Coordinated Dialing Plan (CDP), package 59 and the following packages are required:

- Integrated Services Digital Network (ISDN) package 145;
- 2.0 Mbit Primary Rate Interface (PRI2) package 154;
- Flexible Numbering Plan (FNP) package 160;
- Integrated Services Digital Network Supplementary Features (ISDN INTL SUP) package 161; and
- International Primary Rate Access (IPRA) package 202.

The following packages are required for EuroISDN to interact with Multifrequency Compelled Signaling (MFC), Multifrequency Compelled Signaling for Socotel (MFE), Direct Inward Dialing (DID), Digital Access Signaling System #2 (DASS2), and Digital Private Network Signaling System #1 (DPNSS1):

- Universal ISDN Gateway (UIGW) package 283; and
- Enhanced DPNSS1 Gateway (DPNSS189I) package 284.

Feature operation

There are no operating procedures specified for this feature.

EuroISDN Continuation Phase III

The EuroISDN feature was initially introduced in Release 20 to provide ISDN PRI and ISDN BRI connectivity between the Meridian 1 and Public Exchanges that comply to the European Telecom Standards Institute (ETSI) specification ETS 300 102 for the Layer 3. The interfaces provided by this feature also comply with the country-specific Application Documents for Austria, Denmark, Finland, Germany, Holland, Ireland, Italy, Norway, Portugal, Sweden, and Switzerland (only ISDN BRI connectivity).

The EuroISDN Continuation feature, offered in Release 22, expanded the scope of the original EuroISDN feature to provide application support for Belgium, Spain, Switzerland (ISDN PRI connectivity added to the ISDN BRI connectivity introduced in Release 20), and the United Kingdom.

EuroISDN Continuation Phase III adds France, Russia and the Ukraine to the list of countries having country-specific Application Documents compliance for EuroISDN.

In addition to providing all of the functionalities provided by the original introduction of EuroISDN and the EuroISDN Continuation, EuroISDN Continuation Phase III introduces the following new functionalities for all EuroISDN interfaces:

- optional sending of last forwarding DN as CLID
- Trunk Route Optimization - before Answer applied to incoming EuroISDN trunks
- Numbering Plan Identification (NPI) and Type of Number (TON) included in CDR tickets for EuroISDN calls

These new functionalities are described in the sections that follow.

Optional sending of last forwarding DN as CLID

In a call redirection scenario (Call Forward All Calls, Call Forward No Answer, Hunt, and the ACD redirections, Night Call Forward and Interflow) occurring at a gateway node, the (optional) capability, on a per system basis, of sending the last forwarding DN as Calling Line Identification (CLID) information. This functionality is controlled by a new prompt, the OCLI (Outgoing CLID) prompt, that has been introduced in Overlay 15 and which has three possible responses (NO, EXT, or ALL).

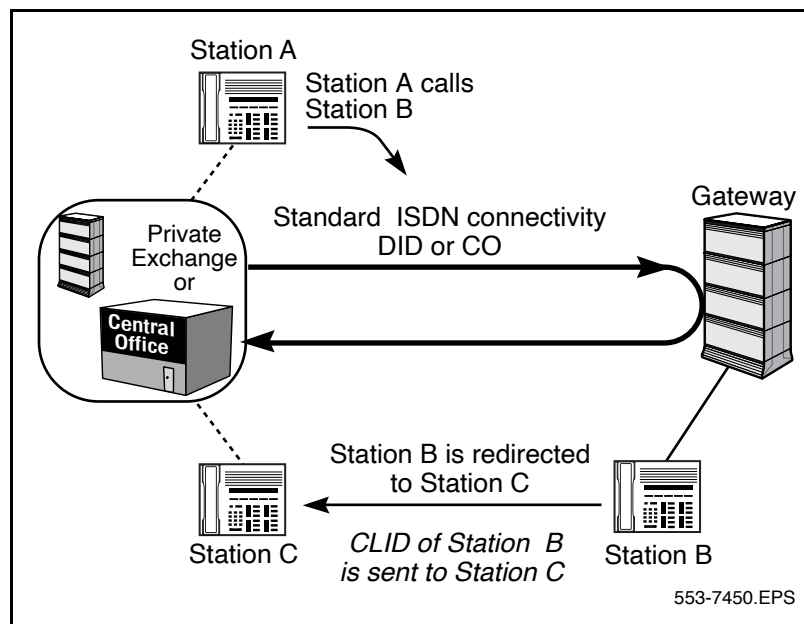
This new functionality applies to the following types of calls being redirected at a gateway node, as described in the pages that follow.

Note: There is a limitation on the functionality if the last redirection is due to ACD NCFW or Interflow, and if the redirection is preceded by one or more redirections on the same node. In such a case, the called number on the node is sent as CLID rather than the ACD DN.

- Incoming EuroISDN DID or Central Office calls coming from a Public Network Central Office or a Private Exchange over a standard ISDN connectivity¹, are forwarded back to the Public Network.
 - Station A, located at a Public Network Central Office or a Private Exchange, calls Station B at a Gateway node. Station B is redirected to Station C, located at the same or another Public Network Central Office or Private Exchange as Station A. If OCLI = EXT or ALL, the CLID of Station B is sent to Station C. Refer to Figure 15.

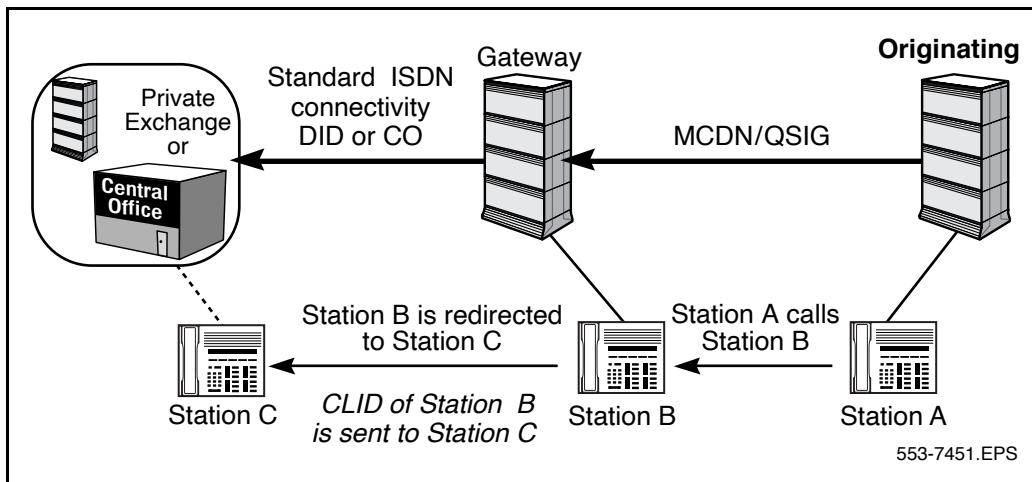
1. By a standard ISDN connectivity, we mean all supported ISDN connectivities including, and since, Release 20, namely, EuroISDN, Asia Pacific, and NI-2.

Figure 15
Incoming EuroISDN call redirected back to the Public Network/PBX



- MCDN/QSIG calls from an originating node, are being redirected to a Public Network Central Office or a Private Exchange over a standard ISDN connectivity.
- Station A, located at an originating node, calls Station B located at a gateway node. Station B is redirected to Station C, located at a Public Network Central Office or a Private Exchange. If OCLI = ALL, the CLID of Station B is sent to Station C. Refer to Figure 16.

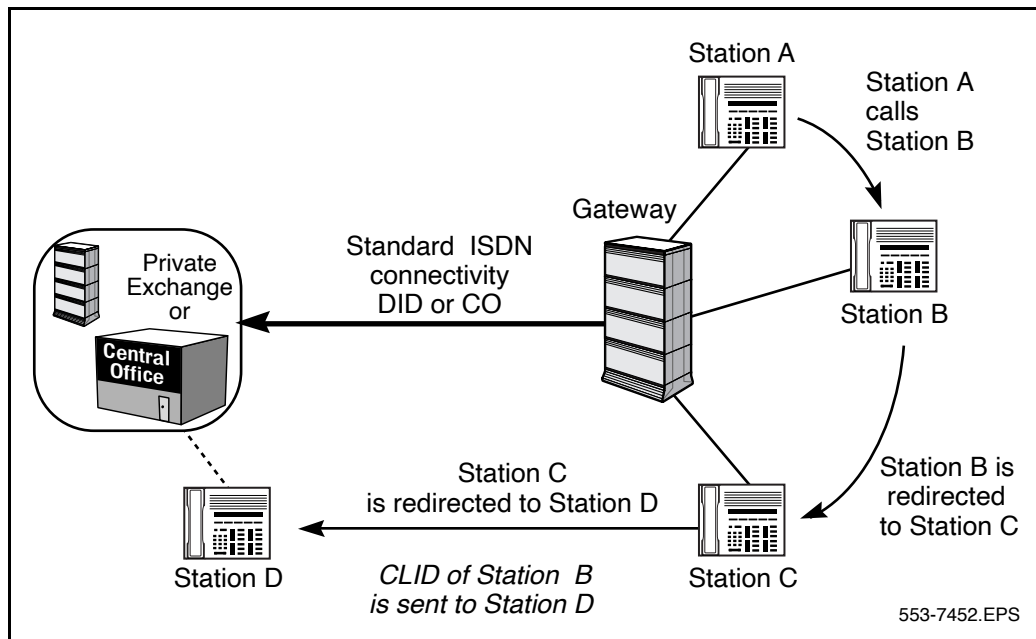
Figure 16
MCDN/QSIG call redirected to the Public Network/PBX



- Internal calls (in a stand-alone scenario) are being redirected to a Public Network Central Office or a Private Exchange over a standard ISDN connectivity.
- Station A, Station B, and Station C are all located on the same gateway node. Station A calls Station B, which is redirected to Station C. Station C is redirected to Station D located at a Public Network Central Office or a Private Exchange. If OCLI = ALL, the CLID of Station C is sent to Station D. Refer to Figure 17.

Note: If Station B is redirected to an ACD queue in via ACD NCFW or Interflow across a Standard ISDN connectivity, then the CLID sent to Station C will be that of the number called by Station A rather than that of the ACD DN. This is due to the limitation on ACD redirection described in the Note on page 190.

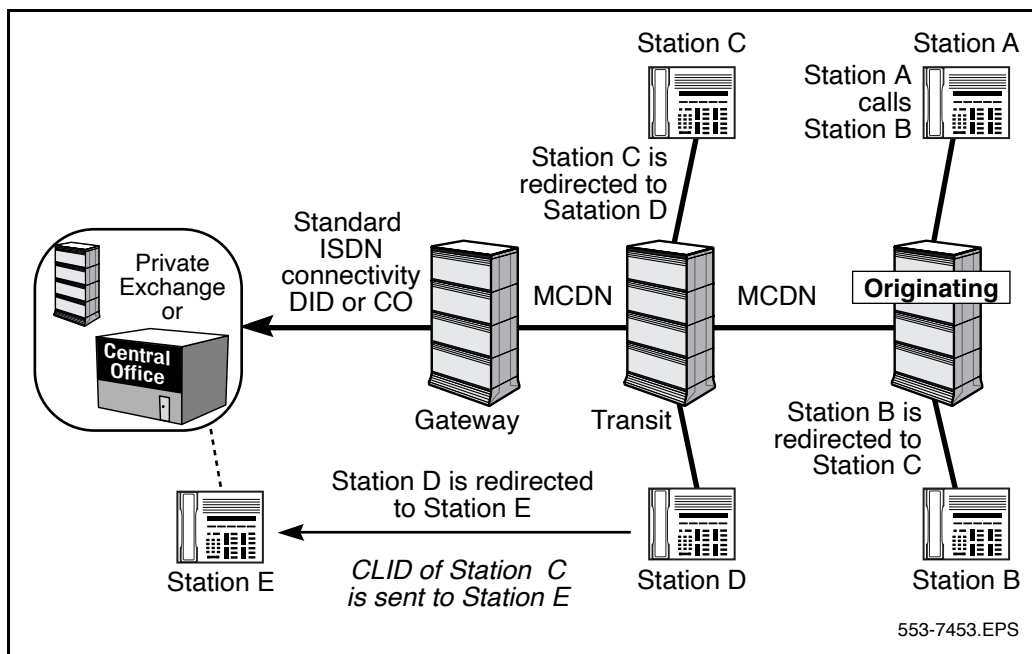
Figure 17
Internal call (stand-alone case) redirected to the Public Network/PBX



When the redirection does not occur at the gateway node for calls redirected to a Public Network Central Office or a Private Exchange over a standard ISDN connectivity, the outgoing CLID is that of the redirecting number information included in the SETUP message and received at the gateway node. The redirecting number information is the first redirecting DN on the last redirecting node.

- Station A and Station B are located at the same originating node. Station A calls Station B, which is redirected to Station C located at a transit node. Station C is redirected to Station D on the same transit node. Station D is redirected through a gateway node, to Station E located at a Public Network Central Office or a Private Exchange. If OCLI = ALL, the CLID of Station C is sent to Station E, since the redirecting number information is the first redirecting DN (that of Station C) on the last redirecting node (at the transit node). Refer to Figure 18.

Figure 18
MCDN call redirected to the Public Network/PBX from a non-gateway node

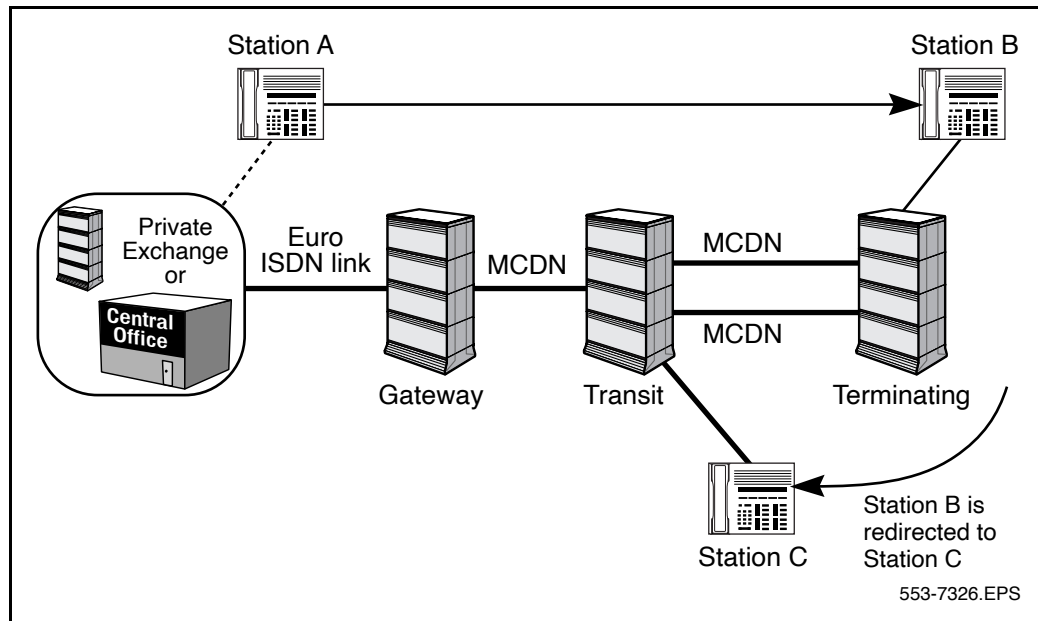


Trunk Route Optimization - before Answer on EuroISDN trunks

Trunk Route Optimization before Answer (TRO-BA) is enhanced to allow TRO-BA to be applied to EuroISDN trunks. The trunk optimization occurs within an MCDN network to eliminate redundant trunks due to a call redirection scenario (Call Forward All Calls, Call Forward No Answer, Call Forward Busy, and Hunt).

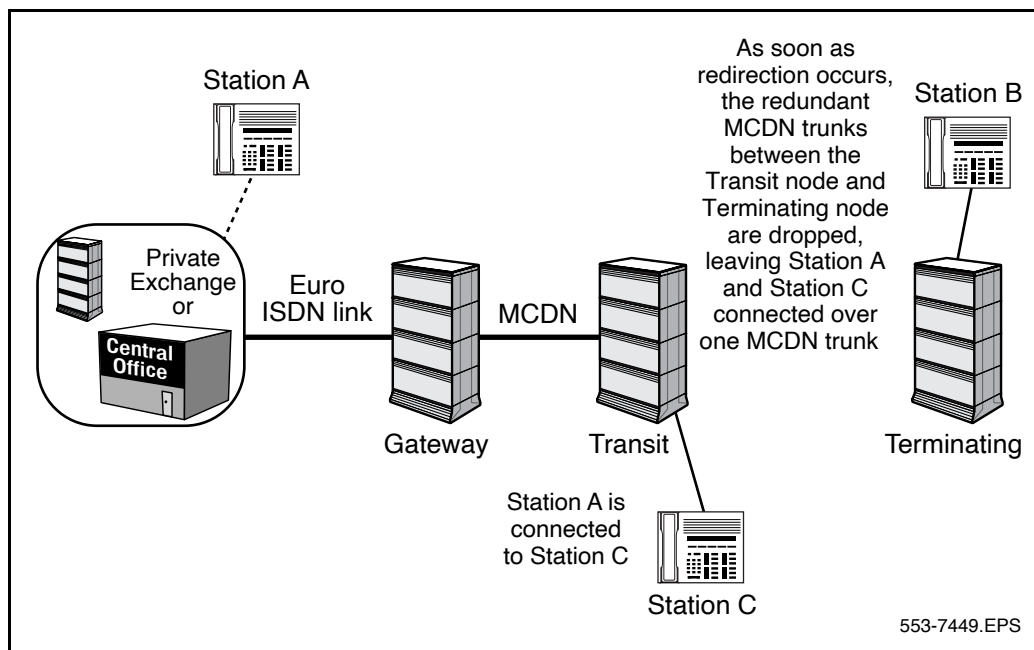
- Station A, located at a Public Network Central Office or a Private Exchange, calls Station B at a terminating node over an MCDN trunk. Station B is redirected to Station C, located at a transit node, over another (redundant) MCDN trunk (refer to Figure 19).

Figure 19
EuroISDN calling scenario before Trunk Optimization is applied



- As soon as the redirection occurs at the redirecting node (the terminating node), trunk optimization is invoked, the redundant trunks between the terminating node and the transit node are dropped, and Station C and Station A connected over one MCDN trunk. Refer to Figure 20.

Figure 20
EuroISDN calling scenario after Trunk Optimization has been applied



Numbering Plan Identification and Type of Number in CDR tickets for EuroISDN calls

The Numbering Plan Identification (NPI) and Type of Number (TON) fields may be optionally included in the CDR tickets, for EuroISDN calls. The purpose is to allow billing the charge of a EuroISDN call to the call originator. By entering NEW against the FCDR (Format for Call Detail Recording) prompt, and YES against the CLID prompt in Overlay 17, the Numbering Plan Identification (NPI) and Type of Number (TON) fields are printed as part of the CDR ticket in addition to the CLID field. This functionality applies only for incoming calls over ISDN trunks, since the NPI and TON are derived from the incoming trunk.

The NPI and TON fields are left blank if the calls are internal, are over outgoing ISDN trunks, are over incoming non-ISDN trunks, or if the FCDR prompt is set to OLD or the CLID prompt is set to NO (note that the FCDR prompt takes precedence over the CLID prompt, that is, if FCDR is set to OLD and the CLID prompt is set to YES, the fields are left blank.)

The NPI and TON are both printed on the third line of the CDR ticket, at characters 44-45 and 47, respectively.

Operating parameters

EuroISDN is not supported on non-downloadable D-channel cards.

Interworking with KD3 signaling is not supported.

Integrated Service Access (ISA) is not supported.

Advice of Charge for EuroISDN calls is supported with the Advice of Charge for EuroISDN feature.

The EuroISDN to CIS MF Shuttle gateway is supported, as provided by the CIS MF Shuttle feature introduced in Release 23.

The Auto Terminate Call feature is not supported.

The Call Back Queuing and Off-hook Queuing feature and the Flexible Hotline feature are not supported with Overlap Signaling.

End-to-End Signaling is supported on all outgoing EuroISDN routes as soon as the CALL PROCEEDING message with a Progress Indicator is received.

Special dial tones after dialed numbers are not supported for incoming calls.

For Connected Line Identification Presentation (COLP) supplementary service, sending or restricting connected line identification is not supported on a per call basis on non-ISDN BRI sets.

For the Optional Sending of Last Forwarding DN as CLID functionality, if more than one ACD redirection takes place when an ACD set is involved, the DN of the called set is sent as the CLID rather than the DN of the last forwarding set. Refer to page 193 for an example.

Feature interactions

Call Completion Supplementary Service

Call Completion interacts with Call Transfer Over EuroISDN SN3. Call Completion Free Notification can only be presented to the Call Completion originating station and cannot be transferred to any other station. However, once the Call Completion call is established, it behaves as a normal call with respect to Call Transfer.

Call Detail Recording

The Optional Sending of Last Forwarding DN as CLID functionality interacts with the CDR feature. The CLID field in the CDR ticket for the redirected outgoing EuroISDN call is modified according to the CLID information in the SETUP message, for the various scenarios described in the section “Optional sending of last forwarding DN as CLID” on page 190.

CLID Enhancements

The Optional Sending of Last Forwarding DN as CLID functionality interacts with the ISDN CLID Enhancements feature introduced in Release 22, in that the CLID entry programmed against the prime DN of the forwarding set at a gateway node is used to build the outgoing CLID.

Incoming Digit Conversion Enhancement

The Incoming Digit Conversion Enhancement (IDC) feature converts incoming digits from a DID route. This feature is supported on the incoming EuroISDN DID routes. Digits received as a called party number are converted if the IDC feature is activated on the route. Digit analysis is then performed on the converted digits by the Meridian 1.

Incoming Trunk Programmable CLID

The Optional Sending of Last Forwarding DN as CLID functionality interacts with the Incoming Trunk Programmable CLID feature. This feature, introduced in Release 21, provides the capability of assigning a billing number to incoming trunks.

In the scenario described on page 194, whereby the redirection does not occur at the gateway node, the incoming redirecting number information is used as the outgoing CLID on the standard ISDN connectivity DID or CO trunk, even though the Incoming Trunk Programmable CLID feature is configured to send a billing number as CLID information.

Intercept Treatment on Invalid or Partial Dialing

The Partial Dial Timing (PRDL), Vacant Number Routing (VNR), and Intercept features handle dialing irregularities on the Meridian 1.

The Intercept feature allows calls with dialing irregularities to be routed to an attendant or to a Recorded Announcement, or to receive overflow or busy tone. The VNR feature allows calls to a vacant number to be routed to another node, where the call is treated as a vacant number, given intercept treatment, or terminated. The PRDL feature allows incoming non ISDN DID calls to be routed to an attendant after a configured amount of time.

With respect to EuroISDN calls with dialing irregularities, the PRDL feature has been enhanced so that when the End of Dial (EOD) timer for PRDL has expired, the calls, instead of being automatically routed to an attendant, now receive the intercept treatment provided by the VNR and Intercept features. A proviso is that the EOD timer used as the PRDL timer must be configured to be at least two seconds less than the ISDN interdigit timer (T302). Otherwise, the calls are rejected if the T302 times out first.

Interworking Notification

Direct Interworking

When “direct interworking” occurs, the EuroISDN trunk terminates directly on the analog trunk. As a result, the CALL PROCEEDING message is sent to indicate that no further called party information is required. No further call control message is initiated by the analog trunk until answer supervision is provided. If the trunk is a supervised trunk, the resulting CONNECT message will permit the Public Switched Telephone Network (PSTN) to start charging the caller. However, an unsupervised trunk may also be called. In this case, the CONNECT message is sent upon the expiry of the pseudo answer supervision timer.

The destination non-ISDN trunk may be busy instead of ringing; therefore, it is not effective to provide an ALERTING message, since the PSTN provides the caller with ringback. However, the PSTN will be running a T310 timer. Expiry of this timer causes call clearing, and a message must be provided.

The following three alternatives can be configured in Overlay 16 by responding to the PROG prompt to stop the network T310 timer:

- 1 A PROGRESS message is sent to the network after the CALL PROCEEDING message.
- 2 An ALERT message is sent to the network after the CALL PROCEEDING message.
- 3 A CONNECT message is sent to the network after the CALL PROCEEDING message.

Tandem Interworking

When tandem interworking occurs, the EuroISDN trunk terminates on an ISDN link terminating on an analog trunk. When the tandem SETUP message is sent, the CALL PROCEEDING message is sent to the PSTN to indicate that no further called party information is required. The tandem ISDN link returns a PROGRESS message, indicating interworking with a non-ISDN trunk.

Since the PSTN does not implement Annex N, proposed by ETSI, it runs a T310 timer. A message must be provided to stop the T310 timer.

The following three alternatives can be configured in PROG prompt Overlay 16 to stop the network T310 timer:

- 1 A PROGRESS message is sent to the network after the CALL PROCEEDING message.
- 2 An ALERT message is sent to the network after the CALL PROCEEDING message.
- 3 A CONNECT message is sent to the network after the CALL PROCEEDING message.

When interworking with an analog trunk without answer supervised signaling, the Meridian 1 should be configured to send a CONNECT message rather than an ALERT message. An ALERT or CONNECT message stops the receipt of digits, which implies that all digits have been received.

Networking Features

Some networking features currently exist on more than one ISDN interface implemented on the Meridian 1. These features are listed in Table 15. Any networking feature that does not appear in the table is only supported on one ISDN interface and is rejected by the gateway when the service is requested. That is the case for all Meridian Customer Defined Network (MCDN) features that are not supported over the EuroISDN or QSIG interface.

Table 15
Networking features supported by more than one ISDN interface

	Euro-ISDN	MCDN	QSIG	ETSI BRI sets	DPNSS1	MFC	DTI2	Analog
Calling Line ID	Yes	Yes	Yes	Yes	Yes	No	No	No
Connected Number	Yes	Yes	Yes	No	Yes	No	No	No
Transit Counter	No	Yes	Yes	No	Yes	No	No	No

Transfer of Unanswered Call

The EuroISDN Continuation Phase III feature supports the Transfer of an Unanswered Call feature. This allows a station (an attendant or set) to transfer an unanswered outgoing EuroISDN call, after dialing has been completed and before the transferred-to set answers (while the set is ringing). The transferred-to set may be local, or may be remote if the link is an MCDN link equipped with Network Attendant Services (NAS), or if the link is a QSIG link equipped with Slow Answer Recall.

Note: The trunk type for the EuroISDN link must be either CO, DID, TIE, FEX, or WATTS for this functionality to work. Otherwise, the transferred-to set must answer before the transfer can be done.

The transfer of an unanswered EuroISDN call interacts with the CDR feature. When the originating party completes the call transfer after the transferred-to set answers, a CDR Start (S) is generated when the transferred-to set answers. When one side releases the call, a CDR End (E) ticket is generated. If the originating party completes the transfer while the transferred-to set is ringing, an S ticket will be generated only when the transferred-to set answers.

Virtual Network Services

A EuroISDN link can be used as a B-Channel for the Virtual Network Services feature.

Feature packaging

EuroISDN Continuation is included in EuroISDN (EURO) package 261.

For Primary Rate Interface (PRI), the following software packages are required:

- Digit Display (DDSP) package 19
- Integrated Services Digital Network (ISDN) package 145
- 2.0 Mbit Primary Rate Interface (PRI2) package 154
- Integrated Services Digital Network Supplementary Features (ISDN INTL SUP) package 161
- International ISDN PRI (IPRA) package 202
- Multi-purpose Serial Data Link (MSDL) package 222

For Basic Rate Interface (BRI), the following software packages are required:

- Digit Display (DDSP) package 19
- Integrated Services Digital Network (ISDN) package 145
- Integrated Services Digital Network Supplementary Features (ISDN INTL SUP) package 161
- International ISDN PRI (IPRA) package 202
- Meridian 1 Extended Peripheral Equipment (XPE) package 203
- Basic Rate Interface (BRI) package 216
- Integrated Services Digital Network Basic Rate Interface Trunk Access (BRIT) package 233

Overlap Signaling is included in (OVLP) package 184. For the Uniform Dialing Plan (UDP), the following packages are also required:

- Basic Routing (BRTE) package 14
- Network Call of Service (NCOS) package 32
- Digit Display (DDSP) package 19
- Basic Alternate Route Selection (BARS) package 57 or Network Alternate Route Selection (NARS) package 58 (for the Uniform Dialing Plan).

For the Coordinated Dialing Plan (CDP), package 59 and the following packages are required:

- Integrated Services Digital Network (ISDN) package 145
- 2.0 Mbit Primary Rate Interface (PRI2) package 154
- Flexible Numbering Plan (FNP) package 160
- Integrated Services Digital Network Supplementary Features (ISDN INTL SUP) package 161
- International Primary Rate Access (IPRA) package 202

The following packages are required for EuroISDN to interact with Multifrequency Compelled Signaling (MFC), Multifrequency Compelled Signaling for Socotel (MFE), Direct Inward Dialing (DID), Digital Access Signaling System #2 (DASS2), and Digital Private Network Signaling System #1 (DPNSS1):

- Universal ISDN Gateway (UIGW) package 283
- Enhanced DPNSS1 Gateway (DPNSS189I) package 284

The following packages are required for the Numbering Plan Identification (NPI) and Type of Number (TON) fields in CDR:

- Call Detail Recording (CDR) package 4
- Call Detail Recording on Teletype Terminal (CDR_TTY) package 5
- Calling Line Identification in Call Detail Recording (CLID_CDR_PKG) package 118
- New Format for Call Detail Recording (CDR_NEW) package 234

Feature operation

No specific operating procedures are required to use this feature.

EuroISDN ETS 300 403 Compliance Update

The EuroISDN feature was initially introduced in Release 20 to provide ISDN PRI and ISDN BRI connectivity between the Meridian 1 and Public Exchanges that comply to the European Telecom Standards Institute (ETSI) specification ETS 300 102 for the Layer 3. The interfaces provided by this feature complied with the country-specific Application Documents for Austria, Denmark, Finland, Germany, Holland, Ireland, Italy, Norway, Portugal, Sweden, and Switzerland (only ISDN BRI connectivity).

The EuroISDN continuation feature, offered in Release 22, expanded the scope of the original EuroISDN feature to provide application support for Belgium, Spain, Switzerland (ISDN PRI connectivity added to the ISDN BRI connectivity introduced in Release 20), and the United Kingdom.

EuroISDN Continuation Phase III, offered in Release 23, added France, Russia and the Ukraine to the list of countries having country-specific Application Documents comply with ETS 300 102 for EuroISDN.

With Release 24, the EuroISDN ETS 300 403 Compliance Update feature has been introduced to provide ISDN Primary Rate Interface and Basic Rate Interface to Central Offices that comply with the ETS 300 403-1 European Telecom Standards Institute (ETSI) standard.

Note: In order to support countries that have not yet upgraded to the ETS 300 403 standard, the Meridian 1 still interworks with Central Offices conforming to the ETS 300 102 standard. So, when programming the D-Channel for PRI2 trunks (in LD 17) or a PRI2 route for ISDN trunks (in LD 16) for an ETS 300 403 interface, the following applies:

- If IFC = E403 and CNTY = ETSI (ETS 300 403 for the user side) or NET (ETS 300 403 for the network side), then the interface is fully compliant with ETS 300 403.
- If IFC = E403 and CNTY = any of the supported country entries except ETSI and NET, then the interface behaves like an ETS 300 102 extended version, that is, in addition to the existing ETS 300 102 capabilities, the bearer capability and High Layer Compatibility selection procedures (fall-back mechanism), and the basic telecommunication service identification are also implemented in order to take advantage of new teleservices being offered in Release 24, such as 7kHz telephony and Videotelephony.

Also note that a user may still configure an interface fully compliant with the ETS 300 102 standard if IFC = EURO and CNTY = any of the supported country values.

With the EuroISDN ETS 300 403 Compliance Update feature, all of the functionalities provided by the previous offerings of the EuroISDN features are supported, along with the following enhancements:

- Support for signalling procedures for bearer capability and High Layer Capability selections by, providing repeatable Bearer capability, High layer compatibility Information Elements (IEs) in the SETUP message. ALERT, CALL PROCEEDING, CONNECT, and PROGRESS messages can be used to confirm the selected Bearer Capability/High Layer compatibility for the called user.

This enhancement provides high-quality bearer services or teleservices, with alternate bearer capability or high layer compatibility in case of fall-back.

- Support for Basic telecommunication service identification. Each basic telecommunication service has the required Bearer capability IE encoding and, if applicable, the required High Layer compatibility IE encoding defined for that service. The requested teleservice is identified by taking the presented Bearer capability and High layer compatibility information elements in all combinations. If there is no valid combination, the presented Bearer Capability IE is considered in order to identify a bearer service.

The following section summarizes the existing EuroISDN functionalities that are supported by the EuroISDN ETS 300 403 Compliance Update feature.

Existing EuroISDN functionalities supported by the EuroISDN ETS 300 403 Compliance Update feature

The ETSI GF subset provides the following two types of functions to the supplementary services control entities:

- Bearer-related transport with a point-to-point transport mechanism. This service is used for the transport of supplementary service protocol information in association with a basic call.
- Bearer-independent transport with a point-to-point facility. This service is used for the transport of supplementary service protocol information which are entirely independent of any existing basic calls.

Note: The ETSI GF does not by itself control any supplementary service but rather provides a generic transport platform that will support ETSI compliance supplementary services. The ETSI Generic Functional protocol is implemented for all ETSI interfaces.

The following call services are supported by the Meridian 1 on the EuroISDN connectivities:

- Basic call service (3.1 kHz, speech, unrestricted digital information).
- 64 kbps bearer capability.
- DID, DOD, COT and TIE call types.
- Channel negotiation.
- Enbloc dialing.
- Overlap sending and Overlap receiving.

- Flexible numbering plan.
- Calling Line Identification Presentation and Calling Line Identification Restriction.
- Connected Line Presentation and Connected Line Restriction.
- MCID (Malicious Call Identification).
- Call Completion to a Busy Subscriber (CCBS).
- 7 kHz telephony (with fall-back to 3.1 kHz).
- Videotelephony.
- Advice of Charge.

Table 16 summarize the networking features supported at gateways with the EuroISDN connectivity.

Any feature that is not listed in this table is not supported at gateways with EuroISDN connectivity. A “YES”/ “NO” indicates that the gateway is/is not supported. A “NS” indicates the service is not supported on the specified interface.

Table 16
Networking features supported by more than one ISDN interface

	Euro-ISDN	MCDN	QSIG	ETSI BRI sets	DPNSS1	MFC	DTI2	Analog	CIS	KD3
Calling Line ID	Yes	Yes	Yes	Yes	Yes	No	No	No	No	No ⁴
Connected Number	Yes	Yes	Yes	No	Yes	No	No	No	No	No ⁴
Transit Counter	No	Yes ₁	Yes ₂	No	Yes ₃	No	No	No	No	No ⁴

Note 1: This is supported using the ICCL tandem count feature existing over the NAS feature.

Note 2: This is only supported for ETSI version of QSIG. For the ISO version, it is discarded.

Note 3: This is supported using the Loop Avoidance supplementary service.

Note 4: The basic call is not supported at the gateway.

The following functionalities are available for EuroISDN interfaces. When not specified these functionalities are networkwide:

- Intercept Treatment on reception of invalid or incomplete called party number.
- Interception to an Attendant for incoming EuroISDN voice calls terminating on a data set.
- Capability to listen to Tones and Announcements provided by the Central Office on call clearing.
- Capability of Transferring outgoing EuroISDN calls after completion of dialling (applies only for Italy).
- Calling Line Identification and Connected Line Identification transparency to/from EuroISDN to/from a BRI set.
- Connected Line Presentation service configurable on a per D-Channel basis as a remote capability.
- Capability of providing 3.1KHz audio bearer capability for outgoing fax calls, based on a Class of Service assigned to a 500/2500 set.
- Capability of defining bearer capability “Speech” or “3.1KHz” on a system basis (this functionality is also provided on ISDN interfaces).
- Addition and display of National or International Prefix on top of the received Calling/Connected Line Identification on incoming/outgoing EuroISDN calls.
- Flexible National and Local prefixes addition in the Calling/Connected Line Identification for outgoing/incoming calls, based on the route configuration.

- User to User information transparency in call control message (only for EuroISDN trunk interfaces).
- Called party number size increase to 31 octets.
- Capability of mapping a PROGRESS message or a Progress Indicator in the CALL PROCEEDING message, into an ALERT or a CONNECT message on a per configuration basis.
- Capability of using different options to build the CLID.
- Capability of sending a RELEASE message to a BRI set, with cause and Progress Indicator (PI) Number 8. The user is provided with display of cause in addition of tone, for the following scenarios:
 - internal BRI set originating call to busy or invalid DN.
 - when receiving a disconnect message from the Central Office via EuroISDN (if a Progress Indicator (PI) Number 8 is not present in the disconnect message, tone is provided locally. Otherwise, inband tone is remotely provided).
 - when receiving a disconnect message over an MCDN network.
- Capability of immediately releasing rejected BRI data calls without providing tone provision to the BRI terminal.
- Capability, on a per configuration basis, of sending last forwarding DN as a CLID information, for the following types of calls:
 - incoming EuroISDN calls being forwarded back to the public network via EuroISDN.
 - incoming MCDN/QSIG calls forwarded to the public network via EuroISDN.
 - internal calls (stand-alone case) being forwarded to the public network via EuroISDN.
- Trunk optimization before call establishment (TRO) is enhanced for incoming EuroISDN calls.

- The Type Of Number (TON) and Numbering Plan Identification (NPI) are included in CDR tickets, when the configuration record parameter CLID is set to YES and when the configuration record parameter FCDR is set to NEW. This allows incoming calls to be billed to different accounts according to the originator of the call. The TON and NPI are required, since the CLID provides a sequence of digits that could be the same for a local, national or international number.
- The EuroISDN interface also provides interworking with other ISDN or non-ISDN interfaces, such as existing ISDN Central Office connectivities (AXE10, 1TR6, SYS12, Numeris and Swiss Net 2), MCDN (1.5 and 2.0 Mbps), QSIG, DPNSS1, DASS2, R2MFC, MFE, DTI, DTI2 and analog trunk interfaces.
- Layer 1 and layer 2 compliance with ETSI requirements are also supported.

Operating parameters

The EuroISDN feature does not support non-Downloadable D-Channel cards (an NT6D80 MSDL or NTBK51AA Downloadable D-Channel card is required on Options 51C, 61C, 81, 81C, and the NTBK51BA Downloadable D-Channel card is required on the Option 11C).

Operating parameters pertaining to Call Control Procedures:

- If more than one Channel Identification IE is received in a SETUP message, only the first one is used by the Meridian 1. If it is not available, the call is processed as per the channel negotiation configuration. The call is released if no negotiation is allowed. The transfer of a unanswered EuroISDN call to a remote ringing set requires disconnect supervision on the TIE trunk. If the disconnect supervision is not available, after the external user hangs up, the trunk could be locked out. It is the craftsman's responsibility to ensure that the trunks used for this type of call actually have disconnect supervision.

Operating parameters pertaining to ETS 300 102:

- The user-to-user compatibility checking, by the means of the Low Layer compatibility IE and/or the High Layer compatibility IE, is not supported. The LLC IE and the HLC IE are tandemed by the Meridian 1, but this information is not used to perform any checking on a Meridian 1 node.

- Transit network selection is not supported. This IE is normally used by the user to identify a selected transit network in the SET-UP message. As no European country specifies the coding to use this IE, this service is not supported by the Meridian 1. As a result, this IE is never sent by the Meridian1.
- Extension for symmetric call operation is not supported. This is normally used to implement a private network application.
- Network specific facility selection procedures are not supported.
- D-Channel backup procedures are not supported by EuroISDN.
- Message segmentation procedures are not supported. These are normally used to split messages that are too long.
- Low Layer Information coding principle is not checked by the Meridian 1. No LLC is generated by the Meridian 1, but this information is tandemed if received (from an ISDN BRI set for example).
- Low layer compatibility negotiation procedures are not supported.
- The USER INFORMATION message is not implemented in the Meridian 1 software.

Operating parameters pertaining to ETSI GF:

- ETSI GF gateways to and from other signalling systems, such as DPNSS1, QSIG, and MCDN, are not supported.

- The following ETSI GF procedures are not implemented:
 - control of supplementary services using the separate message approach (HOLD/RETRIEVE).
 - bearer-related broadcast transport mechanism (multipoint configuration).
 - bearer-independent point-to-point connectionless transport mechanism.
 - bearer-independent broadcast transport mechanism.
 - generic notification procedures.
 - network-side channel reservation function.
 - generic procedures for supplementary service management.
 - generic status request procedure.
 - support of the Extended facility information element.
- DN address translation requires the association with a customer number. For an ETSI basic call establishment, the customer number association is found through the B-Channel identified in the channel ID IE. For DN address translation that is not associated with a basic call, the customer number association needs to be determined through other methods.

For a BRI trunk DSL interface, there is a customer number association with the D-Channel. For a PRI interface, a prompt is added for an ETSI D-Channel configuration to create a customer number association with a given D-Channel. This implies that bearer independent messages on a primary rate D-Channel are associated with a single customer as configured in overlay 17. For example, in a multi-customer configuration, if every customer on the switch wishes to use the bearer-independent transport service over ETSI PRI interfaces, then each customer requires a separate D-Channel.

- The Facility Information Element (FIE) is a repeatable IE, and its length is application dependent. However, due to system capacity considerations, such as call register usage and real time usage, there are two types of limitations enforced by the ETSI GF transport platform (actually by the ISDN GF transport platform provided by the ISDN QSIG GF development):
 - the Meridian 1 ETSI GF platform supports up to a maximum of eight ROSE components in one message. The eight components may be included in one Facility Information Element (FIE), or multiple FIEs. In addition, the inclusion of components in a message is also limited by the “available message length”. The available message length is the difference between the maximum message length (260 octets), and the maximum message length taken up by other mandatory and optional IEs supported in the given message.
 - when a supplementary service requests the ISDN ETSI GF transport to send a component which exceed the available message length or the number of components supported, the supplementary service will be notified.

Operating parameters pertaining to ETS 300 403:

- The operating parameters pertaining to ETS 300 102 are applicable to ETS 300 403, except for the user-to-user compatibility checking, which is partially supported in some cases. For instance, as in the case of the EuroISDN 7 kHz/Videotelephony teleservices.
- Codeset 4, which is reserved for use by the ISO/IEC standards, has been added to the ETS 300 403 standard. This codeset is currently not used by the Meridian 1. Therefore, it is not supported.
- The fall-back capability for multirate services is not supported.

Feature interactions

Incoming Digit Conversion

The IDC feature provides the availability to convert the incoming digits from a DID route. This feature is supported on the incoming EuroISDN DID routes. Digits received as a called party number are converted if the IDC feature is activated on the route. Digit analysis is then performed on the converted digits by the Meridian 1.

Intercept Treatment on Invalid or Partial Dialing

There are three features that are involved in the treatment of calls with dialling irregularities. These are the Partial Dial Timing (PRDL), the Vacant Number Routing (VNR), and the Intercept features.

The Intercept feature allows calls with dialling irregularities to be routed to a Recorded Announcement, an attendant, or to receive overflow or busy tone. Separate treatments can be specified for DID and TIE trunks, for CDP/UDP calls, or for non CDP/ non UDP calls.

The PRDL feature allows to route non-ISDN DID incoming calls to the attendant after a configurable amount of time has expired, if the received digits cannot lead to the completion of the routing.

The VNR feature allows a call to a vacant number to be routed to another node, where the call is either treated as a vacant number, terminated, or given intercept treatment.

In order to provide as much flexibility as possible in the treatment of calls from a EuroISDN interface with dialling irregularities, the PRDL feature has been enhanced. Once the PRDL feature is activated, instead of automatically routing the call to an Attendant, the call will be given the intercept treatment provided by the VNR and Intercept features.

For the PRDL feature, when applied to EuroISDN trunks, the EOD timer used as the PRDL timer must be at least two seconds shorter than the ISDN interdigit timer ($T_{302} = 15$ seconds) used for the VNR and Intercept treatments. The call is rejected if T_{302} times out first. The VNR feature uses another timer which is the NIT timer. Its value is configurable between 3 - 8 seconds, on a per customer basis.

Every time a digit is received, if the number received is an invalid number, the VNR, and then potentially the Intercept treatment feature, are activated.

A number is known to be invalid under the following conditions:

- If Enbloc receiving is configured and an incomplete or invalid number has been received.
- If overlap receiving is configured, and if:
 - an invalid number has been received.
 - an incomplete number with a Sending complete IE has been received.
- If the number received is not completed and not yet invalid, one of the following timer is started:
 - EOD timer (if PRDL = YES)
 - NIT timer for UDP/CDP calls (If PRDL = NO)
 - T302 timer

Calls within a CDP or UDP Numbering plan

If an invalid number has been received, the PRDL feature is not involved as no timer has been started.

If an incomplete number has been received, and if PRDL is configured, the NIT timer has no effect. If the EOD timer times out before T302 timer, the PRDL feature is activated, otherwise the call is rejected.

Non CDP and Non UDP Calls

During enbloc receiving, if an invalid number is received in the SETUP message, the call is treated by the intercept feature.

During Overlap Receiving, if the SETUP message contains an incomplete/empty Called Party Number with the Sending complete IE, the call is treated by the intercept feature. If PRDL is configured, the EOD timer is started.

Transfer of an Unanswered Call

This improvement of the Call Transfer feature (for Italy only) allows an attendant, and an analog or a digital set to transfer an outgoing EuroISDN unanswered call, after dialling has been completed. completion of dialling is indicated by the receipt of a CALL PROCEEDING or ALERT message. The Transfer can be performed before the internal set (set to which the outgoing call is transferred) answers.

The transfer is allowed if NAS is configured on the MCDN link or if the link is QSIG with Slow Answer Recall configured (RTIM prompt in the CDB). If the trunk type of the EuroISDN link is neither a Central Office, DID, TIE, FEX, or WATTS, the completion of the transfer is possible only after the called party on the remote node has answered the call.

The transfer of an unanswered EuroISDN outgoing call interacts with the CDR feature. When the originating party completes the transfer, a CDR ticket type “S” (Start) is generated as soon as the transferred party answers. If one side releases the call, a second CDR ticket E is generated corresponding to the end of the communication.

If the originating party completes the transfer and both parties are still ringing, the CDR ticket type “S” (Start) will only be generated when the transferred party answers. If the external party answers when the transferred party is still ringing and then disconnects the call, the CDR ticket ‘S’ and CDR ticket ‘E’ (End) are generated: in this case the orig ID is the transferring set and the Ter ID is the outgoing trunk.

The Recovery of Misoperation during Call Transfer function is applicable, as per normal operation.

For the Break-in feature, the same restrictions as for normal Call Transfer are apply. During the process of transferring a call, the attendant is not able to reach one of the set involved in the transfer operation.

The changes introduced to enable the transfer before answer have an impact on the conference feature. With the introduction of transfer before answer, it is possible to add additional parties to a call before the first party called over a EuroISDN link has answered the call.

The transfer of unanswered call has no impact on the Call Forward No Answer feature.

The transfer of unanswered call has no impact on the Slow Answer Recall feature.

End to End Signalling

End to End Signalling is supported on all outgoing EuroISDN DID routes as soon as the CALL PROCEEDING message with a Progress Indicator is received.

Integrated Services Access

The Integrated Service Access (ISA) is not supported in Europe.

Advice Of Charge

AOC is implemented as part of the AOC for EuroISDN feature.

Call Completion Supplementary Service

Call Completion interacts with Call Transfer over EuroISDN. This interaction is similar to the current Network Ring Again/Network Ring Again on No Answer, and Call Back When Free/Call back When Not in Use implementations. Call Completion Free Notification can only be presented to the Call Completion Originating station and cannot be transferred to any other station. Once the Call Completion call is established, it behaves as a normal call with respect to call transfer.

Call Back Queuing and Off-hook Queuing

This feature is not supported with Overlap Signalling.

Flexible Hotline

This feature is not supported with Overlap Signalling.

Auto Terminate Call

This feature is supported.

Special Dial Tone after Dialed Numbers

Special dial tones after dialed number feature is not supported for incoming calls.

Trunk Route Optimization

The purpose of this functionality is to provide trunk optimization before call establishment to incoming EuroISDN calls. The trunk optimization occurs when, on the MCDN network, the call has been redirected for one of the following reasons: Call Forward All Calls, Call Forward Busy (CFB), Call Forward No Answer (CFNA), or Hunting. This function will be automatically triggered as soon as TRO is configured on the MCDN network.

If the call is not routed inside MCDN Network before T310 expiry (that is, an ALERT or a CONNECT message is sent back to the Central Office), the call will be released by the Central Office, with cause #102 “Recover on Timer Expiry”. This situation will hardly occur since the optimization operation has to last more than T310 seconds (30 to 40 seconds).

Note: When TRO is invoked on the redirecting node, a 2s timer is armed and if no response is received from the M1 Gateway node before expiry, TRO invocation is cancelled and the call is normally routed through the redirecting node.

Optional Sending of Forwarding CLID

The Meridian 1 is able, on a per configuration basis, to send the last forwarding DN as CLID information for different scenarios of forwarded calls when the redirection occurs on a gateway node. The functionality is activated for the following redirections: Call Forward All Calls, Call Forward No Answer, Hunt and the two ACD redirections, Night Call Forward and Interflow.

When the redirection does not occur on the outgoing gateway node for calls coming from a private network (MCDN/QSIG) and routed to the public network via DID or Central office, the outgoing CLID is the redirecting number information included in the SETUP message received on the outgoing gateway node and coming from the private network.

This redirecting number information is the first redirecting DN on the last redirecting node.

When no redirecting IE is present, no manipulation is done and the original CLID is propagated on the DID link.

When the functionality is activated, everything occurs as if the redirecting set had initiated the forwarded call. This means that information such as Presentation Indicator, Screening Indicator, Type Of Number and Numbering Plan are modified accordingly.

The Outgoing CLID (OCLI) prompt is defined in LD15, and can take three different values:

- **NO.** No manipulation is done on outgoing CLID for calls forwarded to UIPE DID or Central Office.
- **EXT.** The last forwarding DN is sent as CLID information for incoming UIPE DID or Central Office calls landing on gateway node and forwarded back to EuroISDN link.
- **ALL.** The scenario described above still applies. Moreover, last forwarding DN is sent as CLID for internal calls (MCDN/QSIG/set) redirected from gateway node to the DID or Central Office link. If the redirection does not occur on gateway node, redirecting information number is sent as CLID if present in the incoming SETUP message received on gateway node from private network.

Note: An incoming DID/Central Office call, routed through a private ISDN network and then redirected back to the PSTN via the DID/Central Office link, is considered as an internal call. Hence, OCLI prompt has to be set to ALL for the outgoing CLID to be manipulated.

The redirecting information number sent over the MCDN link is not the Last Forwarding DN, but the first Redirecting Number on the last node where redirections occur. Therefore, the DN sent as CLID over the DID or Central Office link will be that DN which is not the Last Forwarding DN.

ISDN CLID Enhancement

The ISDN CLID enhancement interacts with the Release 23 product improvement “Optional Sending of Forwarding CLID”. The CLID entry programmed against the prime DN key of the forwarding set attached to the gateway node is used to build the outgoing CLID.

Incoming Trunk Programmable CLID

The “Optional Sending of Forwarding CLID” interacts with the Release 21 feature “Incoming Trunk Programmable CLID”. This feature provides the capability to assign a billing number for incoming trunk routes.

Even though “Incoming Trunk Programmable CLID” is configured to send a billing number (such as the ACOD of the MCDN/QSIG route) as the CLID information, the incoming redirecting information number, if known, is used for the outgoing CLID on the DID trunk.

Type of Number and Numbering Plan Identification in CDR tickets

The Type Of Number and Numbering Plan Identification fields are included in CDR tickets.

Numbering Plan Identification will be printed out on the third line of CDR at characters 44 and 45. Type Of Number will be printed out on the third line of CDR at character 47. Even if seven values are actually used by the Meridian 1, as NPI information is coded on four bits in the Calling Party Number Information Element, it is advisable to reserve a two characters printout.

Both fields only appear if the system configuration parameter CLID is set to YES, if the system configuration parameter FCDR is set to NEW and if there is an incoming ISDN trunk involved in the call. This means that both NPI and TON information are taken from the incoming trunk.

For non ISDN trunks as well as for sets, two blank characters are output for the NPI field and one blank character is output for the TON field. Blank characters are also output if the system configuration parameter CLID is set to NO. If the system configuration parameter FCDR is set to OLD, the fields do not appear, whatever the value of CLID parameter.

Customer Controlled Routing

A call request for telephony 7 kHz or videotelephony teleservice terminating on a Control DN (same as an ACD DN) follows the same rules as for a Meridian proprietary set, that is, fall-back occurs if allowed, otherwise the call is rejected.

Call Detail Recording

The “Optional Sending of Forwarding CLID” interacts with the CDR feature. The CLID field in the generated CDR ticket for the forwarded outgoing EuroISDN call is modified exactly the same way as the CLID information in the SETUP message.

Meridian Link

The CLID generated for DISA and incoming trunks is sent to the Meridian Link application. The prefixes 0/00 added to the display of the CLID are not sent to the Meridian Link.

Meridian Mail

The CLID generated for DISA and incoming trunks is sent to the Meridian Mail application. The prefixes 0/00 added to the display of the CLID are not sent to the Meridian Mail.

Virtual Network Service

VNS provides ISDN features when no MCDN link is available between two Meridian 1 switches. This can be used when analog links are available, or when ISDN links that provide no supplementary services are used. The existing link is used as a B-Channel, and a separate D-Channel handles the MCDN signalling between the two end switches.

A EuroISDN link can be used as a B-Channel for the VNS feature.

Commonwealth of Independent States Phase II, MF Shuttle

The gateway between EuroISDN and the Release 23 CIS MF Shuttle is supported.

Interworking with KD3

No interworking with KD3 is supported.

Advice Of Charge***Network Call Transfer, Network Call Redirection***

Advice of Charge is not supported network wide. If a call is transferred to another Meridian 1 node, the charging information is stored against the Route meter. Relevant CDR S, X and E records are output on the TTY.

Call Completion on Busy Subscriber

The CCBS service is developed on EuroISDN interfaces as an ETSI service. The service is supported for all EuroISDN countries compliant with the ETSI CCBS protocol. As a result, each country requires detailed study to insure that the CCBS service offered is compliant with the ETSI CCBS protocol.

EuroISDN Malicious Call Identification

Disconnection of the basic call

On reception of a DISCONNECT message for an incoming EuroISDN trunk which has MCID configured, clearing of the call is delayed until the MCTM timer expires or until the called user goes on-hook.

The MCTM timer will only be started depending on the class of service of the set connected to the EuroISDN trunk. It can also be started if the MTND prompt is configured in the route data block in case of a MCDN to EuroISDN gateway. Otherwise, the connection will not be held.

In band timer (T306)

For an incoming EuroISDN call, a DISCONNECT message is sent from the network side with a Progress Indicator (PI) Number 8, so that the T306 timer should delay the sending of the release message. If for the same call the Malicious Call Trace Timer (MCTM in the Route Data Block, LD 16) is applicable, then it is the T306 timer that takes precedence in order to avoid any in-band information to be cut.

Note: It is recommended that the MCTM timer be configured to a lower value than the T306 timer to avoid any timeout of the T306 timer on the network side.

EuroISDN 7 kHz/Videotelephony teleservices

Telephony 7 kHz and videotelephony teleservices are supported on ISDN BRI Line and EuroISDN interfaces. For any other interface, such as QSIG, MCDN, or DPNSS1, fall-back occurs if allowed. Otherwise, the call is rejected.

Feature packaging

The following packages are required for the EuroISDN ETS 300 403 Compliance Update feature:

- EuroISDN (EURO) package 261
- Multi-purpose Serial Data Link (MSDL) package 222
- International Primary Rate Access (IPRA) package 202
- ISDN Supplementary Features (ISDN INTL SUPP) package 161
- 2.0 Mbps Primary Rate Access (PRI2) package 154

- Integrated Services Digital Network (ISDN) package 145
- Digit Display (DDSP) 19

For ISDN Basic Rate Interface Trunking (BRIT), the following additional packages are required:

- ISDN Basic rate Access (BRI) 216
- ISDN Basic Rate Trunk Access (BRIT) package 233
- Meridian 1 XPE (XPE) package 203

For EuroISDN with Overlap Signalling, the following additional packages are required:

- Overlap Signalling (OVLP) package 184
- Flexible Numbering Plan (FNP) package 160
- Coordinated Dialing Plan (CDP) package 59, **or**, if the Uniform Dialing Plan is used, **either**
- Network Alternate Route Selection (NARS) package 58 **or**
- Basic Alternate Route Selection (BARS) package 57
- Network Class of Service (NCOS) package 32
- Basic Routing (BRTE) package 14

For EuroISDN to interwork with MFC, MFE, DID, and DASS2/DPNSS1 gateways, the following additional packages are required:

- New Format Call Detail Recording (FCDR) package 234
- Calling Line Identification in Call Detail Recording (CCDR) package 118
- Call Detail Recording on Teletype Terminal (CTY) package 5
- Call Detail Recording (CDR) package 4

Feature operation

No specific operating procedures are required to use this feature.

EuroISDN Malicious Call Identification

The EuroISDN Malicious Call Identification feature (MCID) allows the source of an incoming call to be identified and recorded by the network side (the Central Office) of a EuroISDN PRI/BRI connection.

This request is initiated by the activation of the Malicious Call Trace feature (MCT) from analog (500/2500) type sets using a SPRE code and a two-digit Flexible Feature MCT access code or from Meridian 1 proprietary sets and attendant consoles using the TRC feature key.

EuroISDN Malicious Call Identification (MCID) can be invoked during the active phase of a malicious call or after the active phase of a malicious call for a limited period of time.

Figure 21 illustrates a situation where EuroISDN MCID is initiated during the active phase of a malicious call.

Figure 21
EuroISDN MCID invoked during the active phase of a malicious call

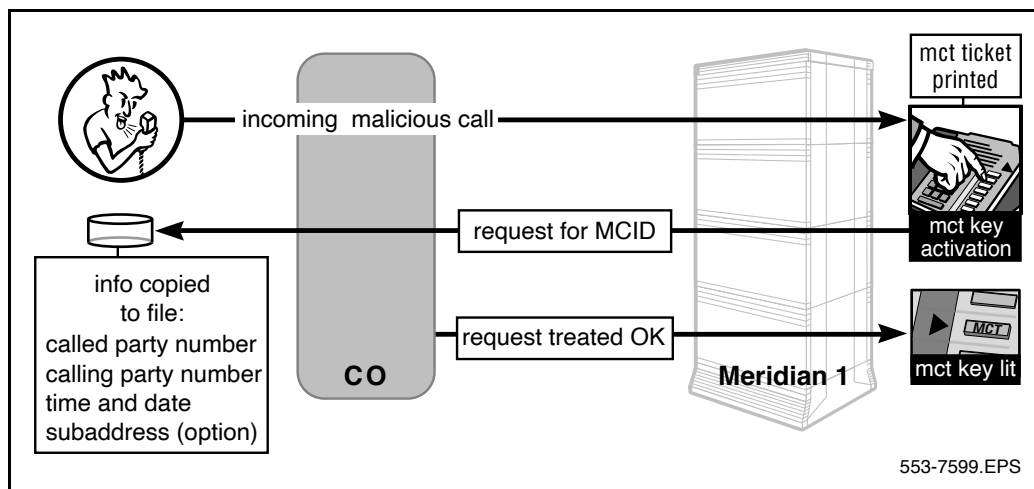
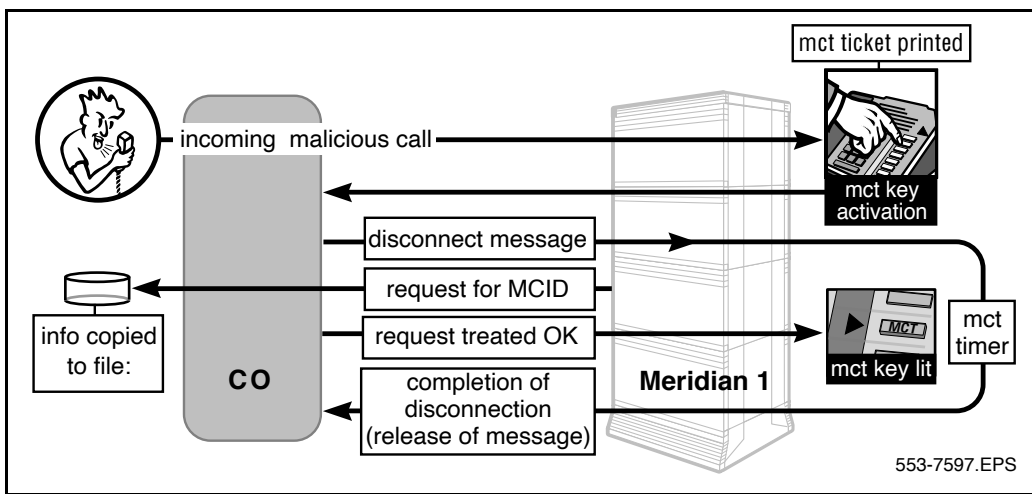


Figure 22 illustrates a situation where EuroISDN MCID is initiated after the active phase of a malicious call.

Figure 22
EuroISDN MCID invoked after the active phase of a malicious call



A report showing the results of MCID request is printed on any system Teletype Terminal (TTY) designated as a maintenance terminal. The report includes an enhanced Malicious Call Trace (MCT) ticket that shows the Number Plan Identifier (NPI) and the Type of Number (TON). A description of these output fields are provided in the section “Malicious Call Trace Record output”.

The required system components of the MCID feature functionality are as follows:

- A Central Office switch supporting EuroISDN PRI/BRI connectivity (This CO is referred to as the network side.)
- A Meridian 1 switch supporting EuroISDN PRI/BRI connectivity
- For tandem applications, another Meridian 1 switch connected to the other over an MCDN link supporting the Malicious Call Trace feature and with Network Attendant Services enabled. In this case, in order for the MCID request to be processed, the gateway switch must be located between the MCDN link supporting MCT and with Network Attendant Service (NAS) enabled, and the EuroISDN Central Office interface.

When the MCID request is initiated via activation of the Malicious Call Trace feature, a MCID request message is sent to the Central Office from the called party node. The Central Office processes the MCID request and returns a message to indicate the successful operation. It registers the following relevant MCID information that is stored by the network operator:

- the called party’s directory number
- the calling party’s directory number
- the local time and date of invoking call trace on the network serving the called party
- the calling party’s sub-address, if provided

Malicious Call Trace Record output

The NPI and TON fields have been added to the print output format, to show the Numbering Plan Identifier (NPI) and the Type of Number (TON) of the Calling Line Identifier field. The MCID ticket printed when the malicious call occurs. The output format is as follows:

```

MCI/MCE CUST# STAR1 ORIGTN STAR2 TERTN ATTPTY TIME
CNI#
STAR1ORIGDN      STAR2TERDN      NPI TON

```

Note: The second line is printed only if MCDC = YES in Overlay 15, the Customer Data Block. Also, for ISDN trunks, the ORIGDN/TERDN fields in line two are replaced by the CLID# field. Therefore, for ISDN trunks, the output would be as follows:

```

MCI/MCE CUST# STAR1 ORIGTN STAR2 TERTN ATTPTY TIME
CNI#
STAR1CLID#      STAR2CLID#      NPI TON

```

where, for line 2:

Field	Description
NPI	For ISDN trunks. The Numbering Plan Identifier of the CLID printed above.
TON	For ISDN trunks. The Type of Number of the CLID printed above.

Operating parameters

Malicious Call Trace (MCT) feature is only activated against one call at one time from either a set or an attendant. If a set activates MCT against a call, then the traced call is put on hold and a second call is activated. A trace against the second call cannot be completed until the first call is disconnected. If multiple TRC keys are defined, then this still occurs. The TRC key reflects the status of the MCT activation.

This feature is not supported on Basic Rate Interface (BRI) sets. However, it is supported over ISDN BRI trunks.

Soft keys cannot be defined as a TRC key on Meridian 1 2317, 3000 or 5317 digital sets.

For Options 51C-81C, the EuroISDN Malicious Call Identification requires the following hardware:

- For Primary Rate Access, the NT6D72 PRI card or the NTCK43 Dual PRI card
- For D-Channel processing, NT6D80 the Multi-Serial Data Link (MSDL) card, or the NTBK51 two-port Downloadable D-Channel Daughterboard (DDCH) with the NTCK43 DPRI instead of an MSDL card
- For Basic Rate Access, the NT6D73AA MISP card and the NT6D70BA SILC card for trunk access

For Option 11C, the EuroISDN Malicious Call Identification requires the following hardware:

- For Primary Rate Access, the NTBK50AA PRI card
- For D-Channel processing, the NTBK51BA Downloadable D-Channel Daughterboard
- For Basic Rate Access, the NTBK22AA MISP card and the NT6D70BA SILC card for trunk access

When EuroISDN Malicious Call Identification is configured on a set, the Attendant Administration feature cannot modify this feature.

Feature interactions

Call Detail Recording

If a DISCONNECT message is received from an incoming EuroISDN call and the Malicious Call Trace time is started, then this timer delays the output of the Call Detail Recording (CDR) record. This will not occur if the called party's set goes on-hook.

Malicious Call Trace

EuroISDN Malicious Call Identification (MCID) is based on the existing Malicious Call Trace (MCT) feature. The existing MCT activation triggers MCID treatments.

Feature packaging

The following packages are required for EuroISDN Malicious Call Identification:

- Euro Supplementary Service (ETSI_SS) package 323
- Euro ISDN (EURO) package 261
- International Supplementary Features (SUPP) package 131
- Multipurpose Serial Data Link (MSDL) package 222
- Integrated Services Digital Network (ISDN) package 145
- Malicious Call Trace (MCT) package 107
- Universal ISDN Gateway (UIGW) package 283
- Flexible Feature Code (FFC) package 139 for 2500 type sets

Primary Rate Interface (PRI) connectivity requires the following packages:

- International Primary Rate Access CO (PRA) package 146
- Integrated Services Digital Network Supplementary Features (ISDN INTL SUP) package 161
- 2.0 Mb/s Primary Rate Interface (PRI2) package 154
- International Primary Rate Access (IPRA) package 202

Basic Rate Interface (BRI) connectivity requires:

- Basic Rate Interface (BRI) package 216
- ISDN Basic Rate Interface Trunk Access (BRIT) package 233

Feature operation

There are no operating procedures specified for this feature. However, the Malicious Call Identification feature is activated by one of the following methods, depending on the station:

- From an analog (500/2500) type set, a switch-hook flash is performed, and then a SPRE code and two-digit Flexible Feature MCT access code are dialed.
- From Meridian 1 proprietary sets, the TRC feature key is pressed.
- From an attendant console, the TRC feature key is pressed.

The following section describes the lamp status associated with the TRC key on Meridian 1 proprietary sets and attendant consoles.

Trace Number (TRC) key lamp status

The TRC key lamp status indicates the progress and success of the Malicious Call Identification request signaling to the CO. The following are the lamp states:

Lamp Winking

When the TRC key is activated, its associated lamp changes from dark to winking if the trunk involved in the call requires the signaling to be done. The lamp remains winking until the call identification request signaling to the CO has been completed.

In a Meridian Customer Defined Network (MCDN) tandem scenario, the lamp on the set which originated the call identification remains winking until a Facility message is received from the node nearest to the Central Office. The user cannot invoke MCT again while the lamp is in the winking state.

Lamp Lit

The lamp state changes from winking to lit if the call identification request to the CO is successful.

In an MCDN tandem scenario, the lamp changes from winking to lit if a Facility message received from the node nearest to the CO indicates that the MCID request was successful. Any further activation of the TRC key during this state is ignored.

Lamp Dark

This lamp state indicates an idle TRC key or failure of the call identification request to the CO.

In an MCDN tandem scenario, the lamp goes from winking to dark if a Facility message received from the node nearest the CO indicates that the MCID request was unsuccessful.

During this lamp state, the TRC key may be activated to initiate the call identification request again.

EuroISDN Trunk - Network Side

EuroISDN Trunk - Network Side refers to the behaviour of EuroISDN as a network interface (generally referred to as “network side”), where the EuroISDN interface delivered with Release 20 has the behaviour of a terminal interface (generally referred to as the “user side”).

As part of the EuroISDN Trunk - Network Side feature, the following capabilities are provided:

- ETSI EuroISDN network side compliance to go along with the ETSI EuroISDN user side compliance delivered by Generic X11 Release 20 software. The ETSI EuroISDN network side interface (EuroISDN Trunk - Network Side) provides the capability to connect the ETSI EuroISDN user side interface of a Common Peripheral Equipment (CPE) to the Meridian 1. Another arrangement could be the connection of key systems to a Meridian 1, to form part of the private network.

Similarly, this product also offers a trunk interface for the connection of terminal equipment, such as fax servers, routers and multiplexers, which would normally connect to the public network, but which now can be connected to the Meridian 1 to achieve a greater public network connection efficiency. Such access to the Meridian 1 may be via ISDN Primary Rate Interface (ISDN PRI) or ISDN Basic Rate Interface (ISDN BRI) trunks.

The network side interface provides all feature operations, interactions and gateways that are supported by the user side. It supports the following EuroISDN capabilities:

- Basic call;
- Direct Inward Dialing;
- Sub-addressing;
- Calling Line Identification Presentation (CLIP)/Calling Line Identification Restriction (CLIR);
- Connected Line Identification Presentation (COLP)/Connected Line Identification Restriction(COLR); and
- CCBS.

- User-to-User Signaling 1 (UUS1) information exchange transparency, which allows a calling and a called party to exchange small amounts of data over an ISDN PRI or ISDN BRI trunk's D-Channel. This data is contained in the USER_USER Information Element (IE) within the call control messages.
- Bearer Capability-Based Routing, which allows outgoing calls to be selectively routed (over Central Office, Direct Inward Dialing, or TIE trunks for ISDN routes, and Integrated Digital Access trunks for DPNSS1 and DASS2 routes) based on the its Bearer Capability. Any ISDN PRI or ISDN BRI route can be configured (in overlay 16) to be dedicated to handle voice calls only, data calls only, voice with 3.1 KHz, data with 3.1 KHz, or both voice and data.

The following protocols are supported by Bearer Capability-Based Routing:

- EuroISDN
- Asia Pacific
- Numeris
- 1TR6
- SWISSNET2
- AXE10 (for Sweden and Australia)
- SYS12
- D70
- QSIG
- DPNSS1
- DASS2
- MCDN
- All North American ISDN connectivities (Meridian 1 to Meridian 1, Meridian 1 to SL-100, Meridian 1 to DMS-100/250, Meridian 1 to AT&T 4ESS, Meridian 1 to AT&T 4ESS/5ESS).

Operating parameters

The User-to-User transparency capability is only supported between EuroISDN trunks.

The following specific hardware is required on Meridian 1 Options 51C, 61C, 81 and 81C.

For ISDN PRI access:

- The NT8D72BA PRI2 card or the NTCK43 dual-port PRI2 card;
- For D-Channel handling, the NT6D80 MSDL card or the NTBK51BA Downloadable D-Channel Daughterboard (if the NTCK43 dual-port PRI2 card is used);
- QPC414 Network Interface card, if the NT8D72BA PRI2 card; and
- QPC775E (and later vintages) Clock Controller card.

For ISDN BRI access:

- The NT6D73AA MISP card and the NT6D70BA SILC card for trunk access;
- Optionally, the NT6D71 UILC line card (the NT6D71 UILC line card requires ANSI 2B1Q line encoding and a Network Termination 1 (NT1) configuration); and
- The QPC775E (and later vintages) Clock Controller card.

The following specific hardware is required for the Option 11C.

For ISDN PRI access:

- The NTBK50AA PRI2 card;
- The NTBK51BA Downloadable D-Channel Daughterboard; and
- The NTAK20 (vintage BB or later) Clock Controller daughterboard.

For ISDN BRI access:

- The NTBK22AA MISP card and the NT6D70BA SILC card for trunk access.

Feature interactions

Basic Call Gateways

The gateways that are supported for the EuroISDN Trunk - Network Side connectivity are the same as the ones supported for the EuroISDN connectivities. Table 17 lists these gateways and the associated support for basic call functionality.

Table 17
Basic Call Gateways

Gateway	Basic Call Support
Analog (End-of-Signaling)	Basic Voice Call
DTI2 (End-of-Signaling)	Basic Voice Call Data Call (64K unrestricted)
MFE	Basic Voice Call
KD3	Not supported
R2MFC	Basic Voice Call Calling Line ID transmission (provided by the Tandem Call/CDR feature)
DPNSS1	Basic Voice Call Data Call (64K unrestricted) Calling Line ID transmission
ISDN COs (North American and non-EuroISDN interfaces)	Basic Voice Call Data Call (64K unrestricted) Calling Line ID transmission
EuroISDN and Asia Pacific ISDN	Basic Voice Call Data Call (64K unrestricted) Calling Line ID transmission
NI2	Basic Voice Call Data Call (64K unrestricted) Calling Line ID transmission
QSIG	Basic Voice Call Data Call (64K unrestricted) Calling Line ID transmission

Table 17
Basic Call Gateways

Gateway	Basic Call Support
MCDN	Basic Voice Call Data Call (64K unrestricted) Calling Line ID transmission
CIS	Basic Voice Call Calling Line ID transmission
Note: For MFE, MFC, and CIS using DTI2, data calls are possible.	

Call Completion to Busy Subscriber

This feature is supported on a EuroISDN Trunk - Network Side connectivity interface.

Call Completion on No Reply

This feature is supported on QSIG and DPNSS1 (as Call Back When Next Used) interfaces, corresponding to the MCDN Network Ring Again on No Answer feature. It is not supported on a EuroISDN Trunk - Network Side connectivity interface.

Calling Line Identification Presentation (CLIP)/Calling Line Identification Restriction (CLIR)

The EuroISDN Trunk - Network Side connectivity can generate, tandem or receive a Calling Line Identification (CLID) with presentation allowed (CLIP) or restricted (CLIR.) A CLID that is generated is constructed in the same manner as the EuroISDN user mode connectivity. A CLID that is received is displayed on the called user's display, if call presentation is allowed.

Even though the EuroISDN Trunk - Network Side connectivity acts as the network side of the Central Office connectivity, it does not provide the network functions (screening and validation) for the Calling Line Identification service.

Connected Line Identification Presentation (COLP)/Connected Line Identification Restriction (COLR)

The EuroISDN Trunk - Network Side connectivity can generate, tandem or receive a Connected Line Identification with presentation allowed (COLP) or restricted (COLR). A Connected Line Identification that is generated or tandemed is constructed in the same manner as the EuroISDN user mode connectivity. A Connected Line Identification that is received is displayed on the called user's display, if call presentation is allowed.

Even though the EuroISDN Trunk - Network Side connectivity acts as the network side of the Central Office connectivity, it does not provide the network functions (screening and validation) for the Connected Line Identification service.

Calling Party Privacy

If a number presentation for a call is blocked by the Calling Party Privacy feature, the CLID, sent over a EuroISDN Trunk - Network Side connectivity, will have the presentation flagged as restricted.

End-to-End Signaling

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.

Incoming Digit Conversion (IDC) Enhancement

This feature is supported on the incoming EuroISDN Trunk - Network Side connectivity DID routes. If IDC is equipped, digits received as a called party number are converted, and digit analysis is then performed on the converted digits.

ISDN CLID Enhancements

The EuroISDN Trunk - Network Side connectivity supports all of the user side ISDN CLID enhancements introduced by Generic X11 Release 22 software.

Integrated Services Access (ISA)

ISA is not supported on a EuroISDN Trunk - Network Side connectivity. ISA is currently implemented for the North American ISDN interfaces. ISA allows a B-channel to be configured as a universal trunk. For example, the same B-channel trunk can be used once as a Central Office trunk and the next time as a Direct Inward Dial trunk.

Network Alternate Route Selection (NARS)/Basic Alternate Route Selection (BARS)/Coordinated Dialing Plan (CDP)

For NARS, the Numbering Plan Area (NPA) code and Central Office Code (NXX) cannot be used on the ETSI network side interface, since the codes are not supported by the European public network. Special Numbers (SPNs) get converted to “unknown”, since SPNs are used only in North America. Also, when using Location Codes (LOCs), the networking features do not accept a Digit Manipulation Index (DMI) used to insert an ESN access code. Therefore, the trunks are treated as though they were non-ISDN.

For BARS, the Numbering Plan Area (NPA) code and Central Office Code (NXX) cannot be used on the ETSI network side interface, since the codes are not supported by the European public network. Special Numbers (SPNs) get converted to “unknown”, since SPNs are used only in North America.

For CDP, a CDP call can access a trunk on the EuroISDN Network Side. However, since neither the private number nor a CDP number is supported by the ETSI EuroISDN Trunk - Network Side, they get converted to a type that is supported by public network. This applies to both the called and calling number plan and type. Also, normal usage of steering codes with Distant Steering Codes (DSCs) and Trunk Steering Codes (TSCs) is supported, as is the use of Digit Manipulation Indexes (DMIs).

Network Attendant Service

Network Attendant Service signaling is not supported on a EuroISDN Trunk - Network Side connectivity. However, NAS will interwork with an incoming call from the EuroISDN Trunk - Network Side (routing and call handling).

Network Call Redirection (Call Forward, Call Forward No Answer, Hunt) and Network Call Modification (Conference, Transfer)

It is possible to have a set Call Forward, Call Forward No Answer or Hunt to an external number over a EuroISDN Trunk - Network Side connectivity ISDN PRI or ISDN BRI trunk. It is also possible to transfer or conference a call to an external number over a EuroISDN Trunk - Network Side connectivity ISDN PRI or ISDN BRI trunk. Access restrictions may block some transfers from being completed.

Notices of call redirection or call modification are not transmitted over a EuroISDN Trunk - Network Side connectivity.

Network Call Party Name Display

This feature is not supported on a EuroISDN Trunk - Network Side connectivity.

Name Display

The transport of the name information, which is supported on QSIG interfaces and which corresponds to the Network Calling Party Name Display feature, is not supported on a EuroISDN Trunk - Network Side connectivity interface.

Networking Supplementary Features

The supplementary features that are supported on a EuroISDN Trunk - Network Side connectivity interface are Call Completion to Busy Subscriber, Calling Line Identification Presentation/Restriction, Connected Line Identification Presentation/Restriction and User-to-User Signaling 1 (UUS1) information exchange transparency.

Special Dial Tone After Dialed Numbers

This feature is not supported for incoming calls on the ETSI network side, but it is supported for outgoing calls.

Trunk Optimization

Trunk Optimization is not supported across a EuroISDN Trunk - Network Side interface. Trunk Optimization Before Answer is not supported within an MCDN network, if the call originated from a EuroISDN Trunk - Network Side connectivity interface.

Virtual Network Services

Virtual Network Services is supported on a EuroISDN Trunk - Network Side connectivity, meaning that a EuroISDN Trunk - Network Side trunk can be used as a VNS bearer trunk.

Feature packaging

To support the network side EuroISDN Trunk - Network Side connectivity, EuroISDN Trunk - Network Side feature (MASTER), package 309, is required.

The following software packages are required as dependencies to the MASTER package or are required to support the full capabilities described in this feature module:

- 2.0 Mbit Primary Rate Access (PRI2) package 154;
- Integrated Services Digital Network Supplementary Features (ISDNS) package 161;
- International Primary Rate Access (IPRA) package 202;
- Basic Rate Interface (BRI) package 216;
- Multi-purpose Serial Data Link (MSDL) package 222;
- Basic Rate Interface Trunk Access (BRIT) package 233; and
- EuroISDN (EURO) package 261.

Feature operation

There are no operating procedures specified for this feature.

Flexible Numbering Plan

Description

This feature, available on Group E (Phase 5) of Generic X11 software with Supplementary Features, allows a private SL-1 network to operate with location codes and directory numbers of varying lengths. The Flexible Numbering Plan (FNP) feature meets Global Numbering Plan requirements by modifying the ESN dialing plan.

The FNP feature can be used with a Co-ordinated Dialing Plan, Universal Dialing Plan or Group Dialing Plan to create a number of different network configurations. The dialing plans are divided into two categories of operation: dialing to a station within the private network; and off-network dialing, to a station which is not part of the private network (typically a call through the Public Numbering Plan).

Fast connect key. The use of flexible numbering (that is, telephone numbers of different lengths) introduces the problem of deciding when to attempt call termination.

If, during digit dialing, the length of the longest available number (the expected number of digits, administered by way of the FLEN prompt) has been reached, the system automatically tries to terminate the call. However, if the call is destined for a station whose number is shorter than the expected number of digits, automatic termination does not occur until the network interdigit timer has expired.

To speed up call connection to stations with short numbers, a Fast Connect key is provided. This is the octothorpe (#) key on the dial pad of the originating station. After the digits of the desired number have been entered, pressing the Fast Connect key forces an immediate termination, thus avoiding the wait for NARS interdigit timeout.

Local network dialing

Local network dialing is the dialing required to reach a location within the private ISDN.

FNPN allows any station within the network to be represented by a flexible number of digits (up to 10, maximum). It also allows flexibility in the length of location codes from node to node (up to 7 digits, maximum). The total number of digits that need be dialed to terminate on a given station can be different for different stations.

Co-ordinated dialing plan. Under the Co-ordinated Dialing Plan (CDP) with FNP, stations on any node have unique 1- to 10-digit numbers. Within the CDP group, no access codes or pauses for dial tone are required; dialing that 1- to 10-digit number terminates directly upon the desired station. FNP allows DNs of different lengths even within the same node, so that number length becomes completely flexible across the entire network.

Universal numbering plan. With the Universal Numbering Plan (UNP), 3- to 7-digit DNs are provided. These DNs are transferable across the network.

Vacant-number routing is used to effect transferable DNs. When a vacant number on any node is dialed, the call is rerouted to a (central) common location. At this location, which serves the entire network, a list is consulted to determine if there is an alternate location for that called DN, or if the DN is truly vacant. Routing to the station's location takes place; or, if appropriate, vacant-number treatment is given at the terminating node.

Group dialing plan. Under the group dialing plan, CDP groups can exist independently within a network. The number used to dial a particular station will vary, depending on the location of the calling station. Dialing a station from within that station's node requires only the use of the DN. Dialing the station from another station on another node within the same CDP group requires the use of the distant steering code followed by the DN. And dialing in from outside the station's CDP group requires the use of the location code followed by the DN.

Options are provided for digit display in Group Dialing Plan configurations, where display consistency can suffer. Options of 'home location code with DN' display, 'local steering code with DN' display, or simply display of the DN alone, can be selected for each distant steering code over an ISDN PRI facility.

Off network dialing

Off-network dialing is the dialing required to reach a location which is not a part of the network. (The location could be a public exchange station, or a part of another private network.)

Special number treatment. With FNP enabled (that is, FLEN not equal to 0), an outgoing trunk is not seized immediately upon the dialing of a Special Number (SPN). Trunk seizure does not occur until either the expected number of digits (maximum of 16) have been dialed, the Fast Connect key has been pressed, or the NARS interdigit timer has expired.

FNP provides a Free Special Number Screening (FSNS) capability which allows the screening of 1- to 11-digit SPNs against 3-digit routing codes, in order to either allow or restrict calls proceeding by way of particular routing codes.

Universal numbering plan. With FNP, the maximum number of route-list blocks and digit-manipulation tables allowed is 1000. This supports the requirements of Global Networking and the Universal Numbering Plan, in which more routes and digit manipulations become necessary.

The dialing of Numbering Plan Area and Office Translation codes are unaffected by FNP.

Interactions and Limitations

Transferable DNs are supported with the 1- to 7-digit Co-ordinated Dialing Plan (CDP). They are not supported by the 8- to 10-digit CDP.

The maximum number of digits permitted for either the location code and DN, the local steering code and DN, or the distant steering code and DN cannot exceed 10 digits when the Group Dialing Plan is used.

On a route going to a location with flexible-length DNs, the expected number of digits is the number of digits in the longest DN at that location. Calls to DN of shorter length terminate either when the Fast Connect (octothorpe) key is pressed on the originating set, or when the NARS interdigit timer expires.

When the FNP feature is equipped, the Supplemental Digit Restriction and Recognition (SDRR) feature is modified to allow a variable number of digits (up to 7) in the digit restriction table. This is independent of the number (of digits) entered in response to the NPA, NXX or LOC prompts in service change.

Flexible Numbering Plan Enhancement

The Flexible Numbering Plan Enhancement (FPE) allows the Flexible Numbering Plan (FNP) feature to be enabled/disabled through service change at a Customer level. FNP is enabled/disabled in Overlay 15 by defining the FNP prompt.

Operating parameters

All operating parameters that apply to the existing FNP feature also apply to the FNP Enhancement.

Feature interactions

Flexible Numbering Plan

The Flexible Numbering Plan Enhancement does not change the functionality of the FNP feature.

Vacant Number Routing

Vacant Number Routing (VNR) is available only when FNP is enabled (FNP = YES). Therefore, VNR is only prompted in Overlay 15 when FNP = YES. When FNP is disabled, VNR is also disabled.

Feature packaging

The Flexible Numbering Plan Enhancement feature requires Flexible Numbering Plan (FNP) package 160, which depends on Network Alternate Route Selection (NARS) package 58.

Feature operation

No specific operating procedures are required to use this feature.

Idle Extension Notification

Description

This feature provides the attendant with the ability to be notified when a busy extension becomes idle. A typical application of this feature is in the case of the attendant wanting to leave an urgent message to someone who is on their extension, but does not want to break-in to their call.

Interactions and Limitations

The Idle Extension Notification feature is not supported on ACD DN's.

An Idle Extension Notification call cannot be placed on hold by the attendant. If the attendant attempts to place an Idle Extension Notification call on hold before the Release key is pressed, the Idle Extension Notification request is canceled and the call is disconnected. An answered Idle Extension Notification recall cannot be placed on hold until after the SACP key is pressed to ring the extension.

When an attendant has requested an Idle Extension Notification, it can be canceled by then pressing the Position Busy key. This will also cancel all other Idle Extension Notification requests.

Idle Extension Notification and pre-dial Break-In are mutually exclusive. It is possible to press the Break-In key, the DN of a busy extension (to override call forward, for example), and then press the SACP key to request Idle Extension Notification. While the SACP lamp is lit, it is not possible to press the Break-In key to complete the Break-In. Conversely, if a Break-In conference is underway, it is not possible to activate Idle Extension Notification by pressing the SACP key.

If the SACP lamp is lit during the pre-dial Break-In sequence, the only available possibilities are: to press the Release key to request for Idle Extension Notification; to press the SACP key again to remove the request; or to press the Release SRC key to Release the source without requesting Idle Extension Notification. If the Idle Extension Notification request is removed by pressing the SACP key a second time (or due to any other limitation of Idle Extension Notification), Break-In is then possible.

It is not possible to do a Post-Dial Break-In on the source side of the attendant, therefore it is not possible to request for Idle Extension Notification in conjunction with Post-Dial Break-In. This means that it is not possible to a Break-In after an Idle Extension Notification recall is answered if the wanted extension becomes busy again.

When an extension that is being supervised for an Idle Extension Notification to the attendant becomes idle, it is kept busy from receiving any incoming calls. The lamp on the attendant console for that DN will display a busy status, according to the parameters of the Busy Lamp Field/Enhanced Busy Lamp Field feature.

It is not possible to request for Idle Extension Notification if the Busy Verify feature has been activated after the Busy Verify key is pressed.

When an extension that is being supervised for Idle Extension Notification becomes idle, it has the ability to make outgoing calls. If Call Forward All Calls or Intercept Call Forward are activated at the extension before the attendant presses the SACP key to ring that extension, the attendant's call will be forwarded to the Call Forward destination. The attendant display will show both the call forward DN as well as the original extension's DN.

If the Call Forward DN is busy, SACP can be activated towards the Call Forward DN, if all the requirements for allowing Idle Extension Notification are met by this DN.

If an extension has Call Waiting configured, it is not possible to request Idle Extension Notification. Call Waiting has precedence over the Idle Extension Notification feature.

When an extension is blocked for receiving calls due to the Idle Extension Notification feature, camp-on is not possible.

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 for 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.

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 BRI extensions always have the Call Waiting feature equipped, therefore Idle Extension Notification is not possible.

The Idle Extension Notification feature is available for ISDN Meridian Customer Defined Networking (MCDN) networks using BRI TIE trunks if Network Attendant Service is configured.

It is not possible to request for Idle Extension Notification towards an extension that has the Make Set Busy feature activated.

If Idle Extension Notification is requested for a Multiple Call Arrangement DN, the first extension with this DN that becomes idle will cause the recall. This extension will also be blocked from receiving calls.

An Idle Extension Notification recall will always recall to the same attendant, regardless of the configuration of the Recall To Same Attendant (RTSA) feature.

During the time that an extension is supervised or temporarily blocked from receiving calls due to the Idle Extension Notification feature, it is possible to activate Ring Again towards that extension. It is also possible to request for Idle Extension Notification on an extension that is supervised for Ring Again. When the extension becomes idle, the Idle Extension Notification will be served first.

Package Requirements

This option requires the following feature packages:

Feature	Package Number
ISDN	145
NAS	159
SACP	181
SUPP	131

BKI	127

In-Band ANI (Automatic Number Identification)

Description

In-Band ANI is a Northern Telecom Meridian 1 capability which allows an Inter-Exchange Carrier to send the Meridian 1 the calling parties phone number over standard digital trunks. The Meridian 1 can then terminate the number to the following:

- The digit display of an auto terminating ACD set
- A Meridian Link connection
- A CDR record

A three or four digit Dialed Number Identification Service (DNIS) can be delivered instead of the ANI digits if required.

Interactions and Limitations

In-Band ANI is only supported over MCI using the DMS-250 Inter-exchange switch.

Package Requirements

This option requires the following feature packages:

Feature	Package Number
ISDN	145
PRI2	154
IEC	149
DNIS	98
BACD	40

Incoming Trunk Programmable Calling Line Identification

The Incoming Trunk Programmable Calling Line Identification feature allows a billing number to be assigned to incoming trunk routes. The billing number length can be from one to 16 digits, and is only used when the incoming trunk terminates to a PRI trunk connected to a Public Exchange/Central Office. The billing number is inserted into the Calling Line Identification (CLID) field of the outgoing PRI/BRI trunk route.

An option is also provided that allows the billing number to have the “Presentation Allowed” field in the outgoing route CLID to be allowed or denied. This will allow or deny the display of the billing number when the call terminates on a set off the Central Office.

In addition, incoming routes that support CLID can have the Billing Number replace the CLID for tandeming only.

Operating parameters

This feature is not supported on the following outgoing interface types:

- SL-1 interfaces, and
- DPNSS1 interfaces.

M911 trunks do not support the billing number.

If the route defined is both incoming and outgoing, the billing number will only be used for incoming calls to support outgoing Integrated Services Digital Network (ISDN) routes.

The billing number is not unique; the same number can be programmed for multiple routes.

The billing number is only inserted on tandem calls for supporting trunks to the Central Office. Therefore, the display of billing number on sets is not provided, and the CDR output will not show the billing number.

There is no control over how the Central Office uses the billing number. The presentation indicator is configurable to either allow or deny the displaying of the billing number. This is only used for the outgoing call to the Central Office which means when the call terminates on a set off the Central Office the billing number will be displayed at that set if the option is set (BDSP = YES).

Call redirection is not supported (e.g., if the incoming call terminates on the Meridian 1 and is transferred, or call forwarded to the Central Office, the billing number is not passed on to the Central Office).

Feature interactions

Not applicable.

Feature packaging

The Incoming Trunk Programmable Calling Line Identification feature is part of basic X11 system software. The following software option package is required to activate this feature:

- Integrated Services Digital Network (ISDN) package 145.

For ISDN and networking, some or all of the following packages may be required:

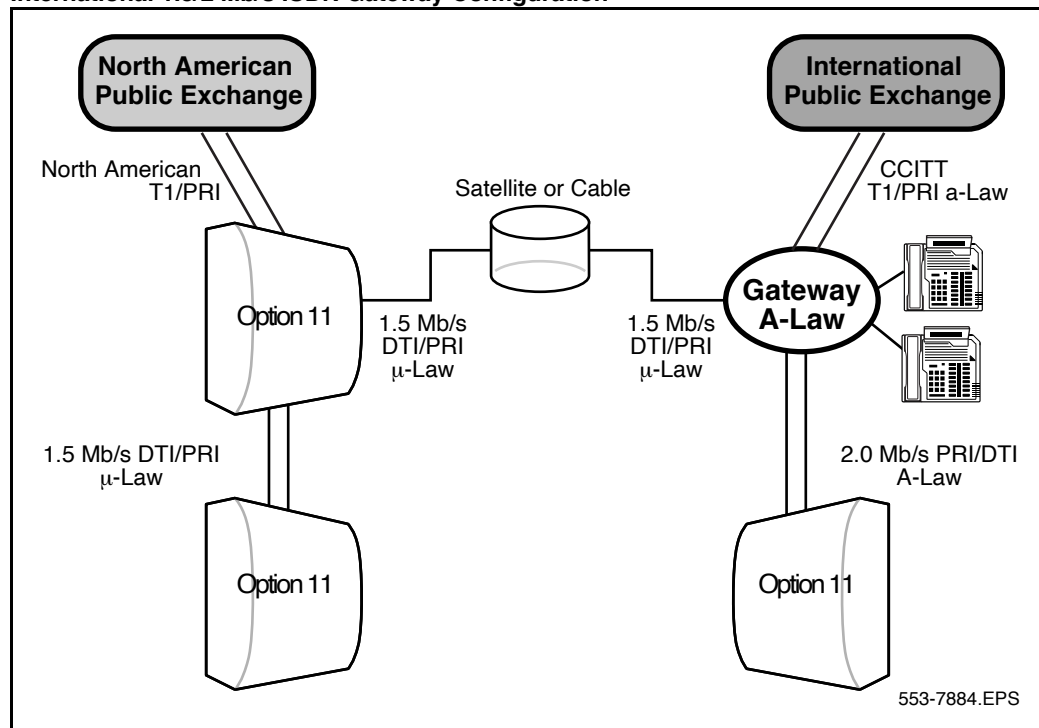
- Basic Alternate Route Selection (BARS) package 57
- Network Alternate Route Selection (NARS) package 58
- Coordinated Dialing Plan (CDP) package 59
- Basic Routing (BRTE) package 14, and
- Network Class of Service (NCOS) package 32.

ISDN International 1.5/2 Mb/s Gateway

This feature provides digital gateways between North American and International Telegraph and Telephone Consultative Committee (CCITT) standards. The international gateway functions as a network node by providing A-Law/ μ -Law conversion and loss of level adjustments to underwork between 1.5 Mb/s and 2 Mb/s Primary Rate Interface/Digital Trunk Interface.

Figure 23 depicts the standard configuration of an X11 International 1.5/2 Mb/s gateway. The gateway switch can function either as an a-Law or μ -Law switch.

Figure 23
International 1.5/2 Mb/s ISDN Gateway Configuration



This feature makes the following software, firmware, and hardware provisions:

A-Law/ μ -Law conversion

The following software, firmware, and hardware modifications have been made to accommodate A-Law/ μ -Law conversion for the International 1.5/2 Mb/s gateway:

- A-Law/ μ -Law conversion has been provided in the QPC720 card.
- Firmware modifications have been made to the PROM/look-up table.
- The Meridian 1 software has been modified to send an SSD message to the hardware on a per-call basis.

The QPC472 DTI card should not be used in gateway configuration because it does not support A-Law/ μ -Law conversion.

Clock Synchronization

The international gateway switch uses the stratum 4 QPC775 CC card synchronized to an international Public Exchange. The North American side of the gateway also uses a stratum 3 QPC775 CC card synchronized to a high-quality clock on the North American Public Exchange.

Loss of Level Adjustments

The following software, firmware, and hardware modifications have been made to accommodate Loss of Level Adjustments for the International 1.5/2 Mb/s gateway:

- Since each country has its own loss-level plan, 14 new pad values have been introduced on the QPC720 card. These values are +0.6db, +7db, +9db, +10db, +11db, +12db, +13db, +14db, -4db, -5db, -7db, -8db, -9db, and -10db.
- In order to implement the new pad values, the pad PROM/look-up table has been modified.
- The Meridian 1 software has been changed to send an SSD message to set the new pad values in the QPC720 card. Also, a new SSD message, to account for all possible pad pairs, has been created.
- The number of pad values supported on the PRI2 has been expanded. The following 11 new pad values have been introduced: +7db, +9db, +11db, +13db, +14db, -4db, -5db, -7db, -8db, -9db, and -10db.

These values are not supported on 2 Mb/s Digital Trunk Interface.

Feature interactions

The following Integrated Services Digital Network (ISDN) features will function transparently across the International Gateway if the dialing plans are consistent on either side of the gateway:

- Calling Line Identification
- Network Call Redirection
- Network Call Party Name Display
- Network Ring Again
- 64 kbps Clear Data
- Overlap Signaling
- International PRA (30b+d) and DTI2
- Meridian MS-1 Audio Teleconferencing
- ISDN Call Connection Limitations
- International Private Exchange to Public Exchange Primary Rate Interface (PRI) Connectivity

Only those features supported by these connections will function transparently across the gateway.

The following features are not supported by the International Gateway:

- 64 kbps Restricted Data
- 56 kbps Clear Data
- Virtual Network Services

Feature packaging

The International 1.5/2 Mb/s Gateway (GPRI) feature is available in package 167. The following packages are also required.

PBXI	option 75
ISDN	option 145
PRA	option 146
PRA2	option 154

ISDN PRI Central Office connectivity

This feature allows the interconnection of the Meridian 1 with public exchanges in the international markets that support the following protocols:

- Asia Pacific, consisting of:
 - Australia (private or alternative carrier)
 - China
 - Hong Kong
 - India
 - Indonesia
 - Japan
 - Malaysia
 - New Zealand
 - Philippines
 - Singapore
 - Taiwan
 - Thailand
- Australia ETSI
- AXE-10 Australia (non-Asia Pacific ISDN Connectivity)
- AXE-10 (Sweden)
- EuroISDN
- Japan D70 (non-Asia Pacific ISDN Connectivity)
- NEAX-61 (New Zealand) (non-Asia Pacific ISDN Connectivity)
- Numeris VN3 (France)
- SwissNet (Switzerland)
- SYS-12 (Norway)
- 1TR6 (Germany)

These connectivities are described in the sections which follow.

Asia Pacific ISDN PRI Central Office Connectivity

The ISDN PRI Meridian 1 to Asia Pacific connectivity provides ISDN Primary Rate Interface (PRI) connectivity between the Meridian 1 and Public Exchange/Central Offices in the following Asia Pacific markets:

- Australia (private or alternative carrier)
- China
- Hong Kong
- India
- Indonesia
- Japan
- Malaysia
- New Zealand
- Philippines
- Singapore
- Taiwan
- Thailand

The Asia Pacific connectivities support the following ISDN features:

- Basic Call Service
- Back-up D-Channel, for Hong Kong
- Advice of Charge, for Japan (considered a basic service)
- Malicious Call Trace, for Australia
- Advice of Charge (AOC) at End of Call, for Australia
- Incoming Trunk Programmable CLID for analog trunks, for Australia. This feature is available for use in a private or alternative carrier network, as required in Australia.
- nB+D, for Japan (up to 215 B-channels/nine interfaces), for New Zealand (up to 120 B-channels/four interfaces), for Malaysia (up to 120 B-channels/four interfaces), and for Hong Kong

- Calling Line Identification Presentation and Restriction (CLIP and CLIR)
- Connected Line Identification Presentation and Restriction (COLP and COLR), for India, Philippines, Taiwan, and Indonesia
- Circuit switched voice and data on the B-channel
- Direct Dialing Inward (DDI/DID), for Indonesia
- Overlap Sending (supported by all interfaces except Japan, Philippines)
- Overlap Receiving, for India, Indonesia, China, Malaysia, and Thailand
- COT, DID, DOD, and TIE trunk call types, as applicable
- 64 kbps clear digital information
- Flexible Numbering Plan
- Sub-addressing (supported only when information is received from the Asia Pacific ISDN interfaces and passed through a tandem node)
- Channel Negotiation (for all countries except Singapore. See the note which follows).

Note: As part of the Singapore enhancement introduced with X11 Release 24, Channel Negotiation is not supported for Singapore. The CNEG option must be set to 1 (the default) in LD 17.

Interworking is provided with the following other ISDN or non-ISDN interfaces.

- existing ISDN CO connectivities (e.g., NUMERIS, Swiss Net 2)
- Taiwan R1 Modified Signaling
- private networks (e.g., Meridian Customer Defined Network, and QSIG)
- other digital signaling systems (e.g., Digital Private Network Signaling System #1 (DPNSS1), and Digital Access Signaling System #2 (DASS2), and 1.5/2.0 Mbps DTI)
- analog signaling systems (Inter Register Multifrequency Compelled signaling (R2/MFC), Multifrequency Compelled Signaling for Socotel (MFE), dial pulse (DP), and Digitone (DTN))

Operating parameters

The Singapore interface complies with the updated national ISDN specification issued by the Telecommunication Authority of Singapore (TAS). This latest specification for Singapore requires that the Channel Negotiation option no longer be supported. Therefore, the CNEG option must be set to 1 (the default) in LD 17 for the Singapore interface type. Also pertaining to the Singapore interface, PROGRESS messages are supported only for BRI (not supported for PRI), and messages with Global Call Reference are supported only for PRI.

The Meridian 1 only supports point-to-point data link connections since PRI only allows a single-point terminal configuration on a user-to-network ISDN interface. BRI supports only point-to-point data link connections.

The ISDN PRI Meridian 1 to Asia Pacific connectivity feature does not support a backup D-channel, except for Hong Kong.

Network Attendant Service (NAS) is not supported across the Asia Pacific interface; however, incoming calls may be routed by NAS from another node.

In a Meridian Customer Defined Network (MCDN), receiving Calling Party Name Display (CPND) from an Asia Pacific ISDN and sending a CPND from an Asia Pacific ISDN are not supported.

Network Call Redirection, Network Call Forward, and Network Call Forward No Answer (MCDN Component) are not supported across an Asia Pacific ISDN interface.

For all of the Asia Pacific interfaces, User-to-User Signalling and Multiple Subscriber Number, Calling Party Name Display, and Call Diversion are not supported.

For all of the Asia Pacific interfaces, the SUSPEND, SUSPEND ACKNOWLEDGE, SUSPEND REJECT, RESUME, RESUME ACKNOWLEDGE, RESUME REJECT call rearrangement messages are not supported.

The Asia Pacific interface does not support the Meridian 1 Packet Handler (MPH) across the CO.

nB+D Primary Rate Access is supported only for Hong Kong, Japan, New Zealand, and Malaysia.

For the Japan interface, Overlap Signaling (Sending and Receiving) is not supported.

For the Hong Kong, Singapore, Australian, and New Zealand interfaces, Overlap Receiving is not supported.

For the Japan interface, Advice of Charge is not provided on BRI terminals.

The following features and capabilities are not supported across any Asia Pacific interface:

- PSTN Equal Access Service - North America Service
- Integrated Service Access (ISA)
- FEX and WATS call types
- Network Wide Ring Again (NRAG)
- Network Calling Party Name Display (NCPND)
- CLID Enhancements (Redirecting Number)
- Network Call Redirection, Network Call Forwarding, Network Call Forward No Answer
- Trunk Optimization/Anti-Tromboning
- ISDN Signaling Link
- RESTART message and INSERVICE message for BRI Trunk Access
- ESN Signaling on PRA (NCOS)
- Virtual Network Services (VNS)

The Malaysia interface does not support RESTART and RESTART ACKNOWLEDGE messages for BRIT.

Hardware requirements

For Option 11C, the following hardware is required:

For Japan, Taiwan, and Hong Kong interfaces:

- For Primary Rate Access, the 1.5 NTAK09AA (DTI/PRI) circuit card is required along with the NTBK51BA Downloadable D-Channel (DCHI) card.
- For Basic Rate Access, the NTBK22AA MISP card and the NT6D70BA SILC card.

For Australia, China, Indonesia, Malaysia, New Zealand, Singapore, Thailand, India, and the Philippines:

- For Primary Rate Access, the NTBK50AA PRI2 card is required along with the NTBK51BA Downloadable D-Channel (DCHI) card
- For Basic Rate Access, the NTBK22AA MISP card and the NT6D70BA SILC card.

Feature interactions***Call Detail Recording***

For a simple call, the total cost for the call is produced in the Call Detail Record.

For the Japan connectivity, if a call is redirected, the Call Detail Recording (CDR) records show all extensions associated with the particular call, however, it is not possible to show call costs associated with each extension. This is because the CDR reports which pertain to the Japan protocol (used for the Japan interface), charge information only at the point when the call is disconnected.

Calling Line Identification Enhancements

Prior to the CLID Enhancements feature introduced in Release 22, the Customer Data Block (Overlay 15) contained the prompts PFX1 and PFX2 (for Prefix 1 and Prefix 2) that were used to construct the CLID. The combination of PFX1, PFX2 and the originating DN were used to construct a correct number for the called party to dial in order to reach the calling party. If no digits are configured for either of the prefixes, then that part of the number will not be included in the Calling Party Number. Essentially, this meant that the CLID could only be built from key 0 of a set. Regardless of what key was used to make a call, it was the CLID for key 0 that was sent. Also, only one office code and one location code could have been assigned in the CLID for a customer.

With the introduction of the ISDN CLID Enhancements feature, PFX1 and PFX2 are no longer used to construct the CLID. CLID is now table-driven (when Overlay 15 is loaded, a customer can configure a CLID table), and virtually any of the information contained in the fields of the CLID table can now be programmed against any DN or DN key, on a per set basis. This means that the CLID that is sent from a set is now predicated on what is in the CLID table, rather than the LDN or PDN. That is, a CLID for any key is now built by taking the information contained in a particular field in the CLID table and adding that information to the key's DN. A multi-line set can now have DN keys that each has their own CLID. Or, the CLID of any one key on a set could be programmed to use the CLID of any other key on the set.

The construction of CLID is based on the CPFXS prompt in Overlay 16. If CPFXS = NO, then when constructing the Calling Number, the prefixes are retrieved from the Route Data Block via the responses to the HNTN and HLCL prompts. If CPFXS = YES, which is the default response, then CLID is built depending upon the prefixes HNTN and HLCL retrieved from the Customer Data Block (Overlay 15) via the entries in the CLID table (refer to the paragraph above for more details).

Also, the Meridian 1 can now support multiple office codes, location codes and steering codes in CLID. This means that any telephone set on one Meridian 1 can send a CLID that will have calls returned to another Meridian 1. This type of configuration is typically used in cases where a customer wants calls to be returned to only one central location.

How a CLID table is built

Prompts have been added to Overlay 15 that create a CLID table for a customer. This table contains up to 4,000 CLID “entries.” Each entry contains unique information pertaining to CLID, as explained in the following sections.

For users of a North American Numbering Plan, the Meridian 1 now supports multiple Home Central Office Codes (HNXXs), Home Numbering Plan Area (HNP) codes, Home Location Codes (HLOCs) and Local Steering Codes (LSCs), on a DN or DN key basis.

For a North American Numbering Plan, each CLID entry can contain the following:

- a 1 - 6 digit national code for a home national number (HNTN)
- a 1 - 12 digit local code for a home local number (HLCL), or a one-12 digit Listed Directory Number for a switchboard
- a 1 - 7 digit Home Location Code (HLOC)
- a 1 - 7 digit Local Steering Code (LSC)

For users of an International Numbering Plan, the Meridian 1 now supports multiple Prefix 1 (PFX1) and Prefix 2 (PFX2) contents, and multiple Home Location Codes (HLOCs) and Local Steering Codes (LSCs), on a DN or DN key basis.

For an International Numbering Plan, each CLID entry can now contain the following:

- a 1 -6 digit national code for a home national number (HNTN), which is the equivalent of PFX1
- a 1 -12 digit local code for a home local number (HLCL), which is the equivalent of PFX2, or a one-12 digit Listed Directory Number for a switchboard
- a 1- 7 digit Home Location Code (HLOC)
- a 1 -7 digit Local Steering Code (LSC)

Another new capability, which applies to both the North American Numbering Plan and the International Numbering Plan, pertains to how the HLCL is constructed. A new prompt, DIDN (which signifies “use DN as a DID number”) in Overlay 15, allows the HLCL to be built either using the digits in the HLCL plus the digits of the active key (if DIDN is set to YES, the DN is considered to be a DID number and is included in the CLID), or only the digits in the HLCL (if DIDN is set to NO, the DN is not included in the CLID since it is not a DID number), or based on a search on the DN keys, beginning from key 0, to find the CLID to be used (DIDN is set to SRCH.)

For more detailed information, please refer to the “ISDN CLID Enhancements” feature module in *International ISDN PRI Feature description and administration*.

Connected Line Identification Presentation and Restriction (COLP and COLR)

Connected Line Identification Presentation and Restriction (COLP and COLR) are applicable only for Indonesia, India, Taiwan, and Philippines.

The Connected Line Identification Restriction (COLR) supplementary service takes precedence over the COLP supplementary service. The COLP service can take precedence over COLR service if the calling user has an override category.

The same Class of Service is used to control both Connected Line Identification Restriction and Calling Line Identification Restriction (CLIR). Thus, if a user has presentation restricted configured, their number is sent to the other party for both incoming and outgoing calls with the presentation flag set to restricted.

Electronic Switched Network (ESN)

Basic Alternate Route Selection (BARS)

BARS interacts with the Asia Pacific ISDN interface as follows:

- NPA and NXX cannot be used on the Asia Pacific ISDN interface, as the public network cannot accept them
- Special Numbers (SPNs) are converted to “unknown” type and plan, as the Electronic Switched Network (ESN) SPN is a North American concept

Network Alternate Route Selection (NARS)

All of the BARS concerns apply to NARS.

In addition, LOC codes must be used with care, since the networking features do not accept a Digit Manipulation Index (DMI) used to insert an Electronic Switched Network (ESN) access code. Therefore, the only supported usage has the trunks effectively being treated as though they are non-ISDN.

Coordinated Dialing Plan

A Coordinated Dialing Plan (CDP) call can access an Asia Pacific trunk. However, since neither the private plan nor the CDP numbering type is supported by the Asia Pacific ISDN, these are converted to a type and plan supported by the public network.

DPNSS Gateway

The Asia Pacific interface interacts with the DPNSS gateway in a manner consistent with other PSTN interfaces. The Information Elements (IEs) supported on the Asia Pacific interface, which have equivalents in DPNSS interwork are translated on an IE by IE basis. IEs which have no matching DPNSS element are discarded. The same is true for DPNSS to Asia Pacific. Supported IEs with DPNSS equivalents pass through with appropriate mapping.

Incoming Digit Conversion

The Asia Pacific interface supports Incoming Digit Conversion ISDN DID routes.

The Incoming Digit Conversion (IDC) feature provides the availability to convert the incoming digits from a DID route. Digits received as a called party number are converted if the IDC feature is activated on the route. Digit analysis is then performed on the converted digits by the Meridian 1.

MCDN Call Redirection (Call Forward, Hunt, Call Forward No Answer)

These features operate in the same manner as when a call to a set is redirected over any other Central Office trunk; however, notices of redirection or modification are not transferred.

nB+D

For the Malaysia and New Zealand connectivities, the channel identification scheme used must be changed from implicit to explicit.

Outpulsing of Asterisk and Octothorpe

The Outpulsing of Asterisk and Octothorpe is not supported for any Asia Pacific interface.

PRI Channel Negotiation

This interaction is applicable to the Japan interface only. When a D-channel has one or more secondary PRIs associated with it while using the Japan interface, channel negotiation requires that the PRI interface be identified (implicitly) in the channel interface IE for all B-channels not on the primary interface whether secondary loops are configured or not.

Trunk Optimization

Trunk Optimization is not supported across the Asia Pacific interface, but interworks transparently. In other words, trunk optimization will not be observed at the Asia Pacific ISDN. The call will appear to be static, not undergoing any modifications.

T309 Timer

The Meridian 1 supports the T309 timer on PRI trunks for all Asia Pacific interfaces.

Virtual Network Services (VNS)

It is not possible to configure an Asia Pacific D-channel as a VNS D-channel. However, the voice connection through the Public Exchange of a VNS call may use a PRI COT or DID as a virtual TIE trunk.

Feature packaging

There are no new software packages required for this feature. However, the following packages are necessary in order to connect the Meridian 1 over an Asia Pacific ISDN PRI interface to a Central Office:

- Digit Display (DDSP) package 19
- 1.5 MBit/s Digital Trunk Interface (PBXI) package 75
- Integrated Services Digital Network (ISDN) package 145
- For Japan, Taiwan and Hong Kong, 1.5 Mbps Primary Rate Access (PRA) package 146
- For Australia, China, Indonesia, Malaysia, New Zealand, Singapore, Thailand, India, and the Philippines, 2.0 Mbps Primary Rate Interface (PRI2) package 154, which has the following dependencies:
 - 2.0 MBit/s Digital Trunk Interface (DTI2) package 129
 - Integrated Services Digital Network (ISDN) package 145
- International Primary Rate Access (IPRA) package 202, which has the following dependencies:
 - Digit Display (DDSP) package 19
 - 2.0 Mbps Primary Rate Interface (PRI2) package 154
 - Integrated Services Digital Network (ISDN) package 145
- Meridian 1 Extended Peripheral Equipment (XPE) package 203
- Multi-Purpose Serial Data Link (MSDL) package 222, which requires the Integrated Services Digital Network (ISDN) package 145
- New Format CDR (FCDR) package 234
- Basic Rate Interface (BRI) package 216
- Basic Rate Interface Trunk (BRIT) package 233

If the call is to interwork with any other trunk, the Universal ISDN Gateway (UIGW) package 283 is required.

For nB+D, the International nB+D (INBD) package 255 is required.

The Overlap Signaling feature requires:

- Network Class of Service (NCOS) package 32
 - Basic Routing (BRTE) package 14
 - Basic Alternate Route Selection (BARS) package 57, Network Alternate Route Selection (NARS) package 58, and/or Coordinated Dialing Plan (CDP) package 59
 - Flexible Numbering Plan (FNP) package 160
 - Overlap Signaling (OVLP) package 184
- Note:*** This package should not be enabled for the Philippines
- Basic Queuing (BQUE) package 28

Feature operation

No operating procedures are required to use this feature.

Australia ETSI

The Australia ETSI feature supports 2.0 MBit/s ISDN Primary Rate Interface and Basic Rate Interface Trunk connectivity for the Australian Central Office, in compliance with the Australia ETSI specification (Telstra).

This feature uses the existing EuroISDN software packages to provide the basic ISDN capabilities and supplementary services listed below (EURO is configured as the interface in the overlay programs when configuring PRI2 and BRI).

Basic ISDN services:

- 2.0 MBit/s PRI and BRI Basic Call Service
- Circuit-mode bearer capabilities (speech, 3.1 kHz audio, 64 kbps digital, and adapted 56 kbps to 64 kbps digital)
- COT, DID, DOD, and TIE trunk call types
- Calling Line Identification (public and private)
- Enbloc Sending
- Overlap Sending
- Channel Negotiation

Supplementary services:

- Calling Line Identification Presentation (CLIP)
- Calling Line Identification Restriction (CLIR)
- Connected Line Identification Presentation (COLP)
- Connected Line Identification Restriction (COLR)
- Malicious Call Trace
- Advice of Charge (AOC), during call set-up, during the call, and at end of call
- Sub-addressing (SUB)
- Direct Dial In (DDI)

Operating parameters

This feature requires downloadable D-Channel handling, for both the large systems (Options 51C, 61C, 81, 81C) and the small system (Option 11C). Therefore, the NT6D11 DCHI is not supported on large systems. The NTAK79AA PRI2 card is not supported on Option 11C because it has an on-board DCHI, and does not use the NTBK51 downloadable D-Channel daughterboard.

Overlap Receiving is not supported.

Basic Alternate Route Selection (BARS) is not supported. Network Alternate Route Selection (NARS) is.

In a Meridian Customer Defined Network (MCDN), receiving Calling Party Name Display (CPND) and sending a CPND are not supported

MCDN Call Redirection (Call Forward, Call Forward No Answer, Hunt) is not supported.

MCDN Call Modification (Conference, Transfer) is not supported.

Network Call Redirection, Network Call Forward, and Network Call Forward No Answer (MCDN Component) are not supported.

Network Attendant Service (NAS) features are not supported across the Australia ETSI interface; however, incoming calls may be NAS routed from another node.

Trunk Route Optimization is not supported across the Australia ETSI interface.

All operating parameters apply to feature as for the EuroISDN Advice of Charge and Malicious Call Trace functionalities.

The Advice of Charge functionality is supported on a system basis only. It is not supported on a per call basis.

Reverse Charging is not supported, nor is requesting charging information from the user's side.

Tandeming of Advice of Charge charging information across a Meridian 1 network is not supported.

The display of charges is not supported on BRI sets and terminals.

Packet data handling is not supported for the BRI component of this feature.

Feature interactions

Calling Line Identification Enhancements

Prior to the CLID Enhancements feature introduced in Release 22, the Customer Data Block (Overlay 15) contained the prompts PFX1 and PFX2 (for Prefix 1 and Prefix 2) that were used to construct the CLID. The combination of PFX1, PFX2 and the originating DN were used to construct a correct number for the called party to dial in order to reach the calling party. If no digits are configured for either of the prefixes, then that part of the number will not be included in the Calling Party Number. Essentially, this meant that the CLID could only be built from key 0 of a set. Regardless of what key was used to make a call, it was the CLID for key 0 that was sent. Also, only one office code and one location code could have been assigned in the CLID for a customer.

With the introduction of the ISDN CLID Enhancements feature, PFX1 and PFX2 are no longer used to construct the CLID. CLID is now table-driven (when Overlay 15 is loaded, a customer can configure a CLID table), and virtually any of the information contained in the fields of the CLID table can now be programmed against any DN or DN key, on a per set basis. This means that the CLID that is sent from a set is now predicated on what is in the CLID table, rather than the LDN or PDN. That is, a CLID for any key is now built by taking the information contained in a particular field in the CLID table and adding that information to the key's DN. A multi-line set can now have DN keys that each has their own CLID. Or, the CLID of any one key on a set could be programmed to use the CLID of any other key on the set.

The construction of CLID is based on the CPFXS prompt in Overlay 16. If CPFXS = NO, then when constructing the Calling Number, the prefixes are retrieved from the Route Data Block via the responses to the HNTN and HLCL prompts. If CPFXS = YES, which is the default response, then CLID is built depending upon the prefixes HNTN and HLCL retrieved from the Customer Data Block (Overlay 15) via the entries in the CLID table (refer to the paragraph above for more details).

Also, the Meridian 1 can now support multiple office codes, location codes and steering codes in CLID. This means that any telephone set on one Meridian 1 can send a CLID that will have calls returned to another Meridian 1. This type of configuration is typically used in cases where a customer wants calls to be returned to only one central location.

How a CLID table is built

Prompts have been added to Overlay 15 that create a CLID table for a customer. This table contains up to 4,000 CLID “entries.” Each entry contains unique information pertaining to CLID, as explained in the following sections.

For users of an International Numbering Plan, the Meridian 1 now supports multiple Prefix 1 (PFX1) and Prefix 2 (PFX2) contents, and multiple Home Location Codes (HLOCs) and Local Steering Codes (LSCs), on a DN or DN key basis.

For an International Numbering Plan, each CLID entry can now contain the following:

- a 1 -6 digit national code for a home national number (HNTN), which is the equivalent of PFX1
- a 1 -12 digit local code for a home local number (HLCL), which is the equivalent of PFX2, or a one-12 digit Listed Directory Number for a switchboard
- a 1- 7 digit Home Location Code (HLOC)
- a 1 -7 digit Local Steering Code (LSC)

Another new capability pertains to how the HLCL is constructed. A new prompt, DIDN (which signifies “use DN as a DID number”) in Overlay 15, allows the HLCL to be built either using the digits in the HLCL plus the digits of the active key (if DIDN is set to YES, the DN is considered to be a DID number and is included in the CLID), or only the digits in the HLCL (if DIDN is set to NO, the DN is not included in the CLID since it is not a DID number), or based on a search on the DN keys, beginning from key 0, to find the CLID to be used (DIDN is set to SRCH).

For more detailed information, please refer to the “ISDN CLID Enhancements” feature module in *International ISDN PRI Feature description and administration* 553-2901-301.

Connected Line Identification Presentation and Restriction (COLP and COLR)

The Connected Line Identification Restriction (COLR) supplementary service takes precedence over the COLP supplementary service. The COLP service can take precedence over COLR service if the calling user has an override category.

The same Class of Service is used to control both Connected Line Identification Restriction and Calling Line Identification Restriction (CLIR). Thus, if a user has presentation restricted configured, their number is sent to the other party for both incoming and outgoing calls with the presentation flag set to restricted.

Coordinated Dialing Plan (CDP)

A Coordinated Dialing Plan (CDP) can be used to access an Australia ETSI trunk. However, neither the CDP private plan nor the CDP numbering type is supported. They get converted to unknown plan and type, respectively.

Virtual Network Services (VNS)

It is not possible to configure an Australia ETSI D-channel as a VNS D-channel. However, the voice connection through the Public Exchange of a VNS call may use a PRI/BRI COT or DID as a virtual TIE trunk.

Feature packaging

There are no new software packages required for this feature. However, the following packages are necessary in order to connect the Meridian 1 over an Australia ETSI PRI2/BRI interface to a Central Office.

For PRI2 connectivity:

- Integrated Services Digital Network (ISDN) package 145
- 2.0 Mbps Primary Rate Interface (PRI2) package 154
- Overlap Signaling (OVLP) package 184
- International Primary Rate Access (IPRA) package 202
- Multi-Purpose Serial Data Link (MSDL) package 222

If the call is to interwork with any other trunk, the Universal ISDN Gateway (UIGW) package 283 is required.

For the Advice of Charge capability:

- Controlled Class of Service (CCOS) package 81
- Background Terminal (BGD) package 99
- Periodic Pulse Metering/Message Registration (MR) package 101
- International Supplementary Features (SUPP) package 131

For the Malicious Call Trace capability:

- Controlled Class of Service (CCOS) package 81
- Malicious Call Trace (MCT) package 107
- International Supplementary Features (SUPP) package 131
- Flexible Features Code (FFC) package 139
- Network Attendant Service (NAS) package 159
- ISDN Supplementary Features (ISDN INTL SUPP) package 161

For ISDN Basic Rate Interface Trunking connectivity:

- Basic Rate Interface (BRI) package 216
- Basic Rate Interface Trunk (BRIT) package 233

AXE-10 (Australia), non-Asia Pacific connectivity

This feature allows ISDN PRI connectivity between the Meridian 1 and the AXE-10 for Australia, using switched-circuit mode connections. The design is based on the international 30B + D PRI configuration.

Basic ISDN PRI connectivity between the Meridian 1 and the AXE-10 will support:

- Basic call service
- Connected number delivery
- Calling Line Identification, both public and private
- Increased CDR accuracy
- COT, DID, DOD, and TIE call types
- 64 kb/s clear-bearer capability
- Channel Negotiation
- Overlap Sending (but not overlap receiving)
- Flexible Numbering Plan
- Advice of Charge at end of call

Operating parameters

The only connection that is supported is the switched circuit mode connection.

The supplementary service control messages not relating to a call are not supported.

The Meridian 1 does not send the High Layer Compatibility Information Element in the SETUP message to the AXE-10, nor does it send the Restart Request and Restart Global Call States.

Overlap Receiving is possible only if the AXE-10 conforms to the guidelines contained in CCITT's preliminary Q.931 section 5.0.

Feature interactions

There are no feature interactions specified for this feature.

Feature packaging

To support all of the call services, the following packages must be equipped:

- Integrated Services Digital Network (ISDN) package 145;
- 2.0 Mbit Primary Rate Interface (PRI2) package 154;
- International Primary Rate Access (IPRA) package 202;
- Digit Display (DDSP) package 19;
- Flexible Numbering Plan (FLEN) package 160; and
- Overlap Signaling (OVLP) package 184.

Feature operation

There are no operating procedures specified for this feature.

AXE-10 (Sweden), non-Asia Pacific connectivity

This feature allows ISDN PRI connectivity between the Meridian 1 and the Ericsson AXE-10 for Sweden, using switched-circuit mode connections. The design is based on the international 30B + D PRI configuration.

Basic ISDN PRI connectivity between the Meridian 1 and the AXE-10 will support:

- Basic call service
- Connected number delivery
- Calling Line Identification, both public and private
- Increased CDR accuracy
- COT, DID, DOD, and TIE call types
- 64 kb/s clear-bearer capability
- Channel Negotiation
- Overlap Sending (but not overlap receiving)
- Flexible Numbering Plan
- Advice of Charge at end of call

Operating parameters

The only connection that is supported is the switched circuit mode connection.

The supplementary service control messages not relating to a call are not supported.

The Meridian 1 does not send the High Layer Compatibility Information Element in the SETUP message to the AXE-10, nor does it send the Restart Request and Restart Global Call States.

Overlap Receiving is possible only if the AXE-10 conforms to the guidelines contained in CCITT's preliminary Q.931 section 5.0.

Feature interactions

There are no feature interactions specified for this feature.

Feature packaging

To support all of the call services, the following packages must be equipped:

- Integrated Services Digital Network (ISDN) package 145;
- 2.0 Mbit Primary Rate Interface (PRI2) package 154;
- International Primary Rate Access (IPRA) package 202;
- Digit Display (DDSP) package 19;
- Flexible Numbering Plan (FLEN) package 160; and
- Overlap Signaling (OVLP) package 184.

Feature operation

There are no operating procedures specified for this feature.

NEAX-61 (New Zealand), non-Asia Pacific connectivity

This feature provides an ISDN Primary Rate Interface (PRI) between the Meridian 1 and New Zealand NEAX-61. The design is based on the international 30B + D PRI configuration.

Meridian 1 to New Zealand NEAX-61 ISDN PRI Connectivity supports the following:

- Basic Call Service
- Calling Line Identification (public and private)
- Overlap Sending
- COT, DID, DOD, and TIE trunk call types
- 64 kbps clear bearer capability
- Channel Negotiation
- nB+D, up to 120 B-channels/four interfaces
- Public Switched Telephone Network (PSTN) three-party conferencing
- Flexible Numbering Plan, and
- Malicious Call Trace.

Note that Electronic Switched Network (ESN) cannot be supported across the interface for private network applications. Coordinated Dialing Plan (CDP) and Network Automatic Route Selection/Basic Automatic Route Selection (NARS/BARS) can access the public network, but private network features are not supported. The NEAX-61 does not support the required numbering plans to allow CDP or UDP (location code dialing).

New Zealand Supplementary ISDN Services

Malicious Call Trace

This subscription supplementary service allows the Meridian 1 to inform the public network in New Zealand that the current call is malicious. The NEAX-61 automatically records all calls to a number subscribing to the MCT feature. If an MCT is activated, the public network marks the call as being malicious.

This service is activated by ISDN messaging. It is a user-triggered feature, requiring the called party to press a key or dial codes.

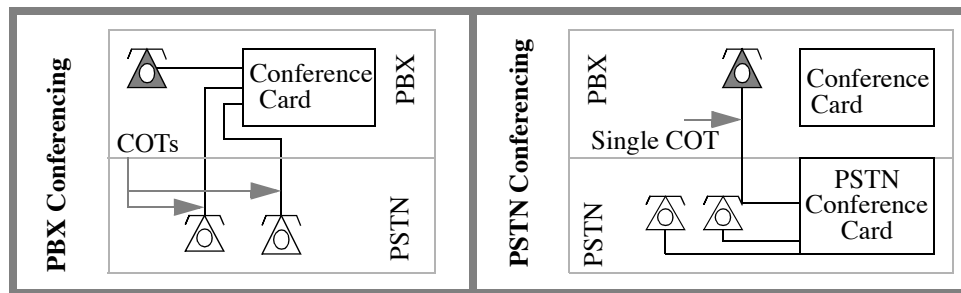
PSTN Three Party Conference

General

This subscription supplementary service allows the New Zealand NEAX-61 to provide a Three Party Conference. Rather than having the Meridian 1 provide the conference capability and the two trunks, it allows a caller to conference in two external callers using one trunk. This supplementary feature is activated using the Trunk Hook Flash (THF) or Centrex Switch Hook Flash operation, and is user invoked.

To illustrate this operation, the PBX supported conference and PSTN supported conference is illustrated in Figure 24.

Figure 24
PBX and PSTN supported conference



This operation is a stand-alone feature as the network wide trunk hook flash is not supported. Note also that from the New Zealand NEAX-61 side, MCT and PSTN conferencing are mutually exclusive features.

Call Detail Recording for PSTN Three Party Conferencing

PSTN Three Party Conferencing does not have the capability of providing precise Call Detail Recording (CDR) records. The NEAX-61 does not send messages to indicate that one of the two external parties has released, or that the attempt to complete the conference via THF was unsuccessful, due to the third party not being established. Therefore, if party “A” on the Meridian 1 called party “B” in the public network, and later called party “C”, no notice is given whether party “C” answered. As well, if either of “B” or “C” releases, the Meridian 1 will not be informed.

As a result, no attempt is made to make the CDR records match the normal “S” and “E” records of conference calls with two trunks. Instead of two “S” records at the start of the conference, a single “Y” record is issued to indicate a PSTN feature is invoked. This “Y” record includes all digits dialed to access the third party. The final record, produced on call clearing, remains as an “N” record, and includes information from the original call.

Operating parameters

The following hardware is used to implement the interface:

- For Option 11, the layer one function is provided by the Downloadable D-channel Primary Rate motherboard, circuit pack NTBK50AA. Layer two and three are provided by the Downloadable D-channel daughterboard, circuit pack NTBK51AA. The Clock Controller daughterboard NTAK20BB is used to provide clock synchronization.

Feature interactions**Calling Line Identification**

This feature is provided via the Calling Party Number. It will have the following format when the Meridian 1 is connected to a NEAX-61:

ZZZZ + YYYY +DDDDDDD

The Customer Data block contains the prompts PFX1 and PFX2 for the responses “ZZZZ” and “YYYY” respectively. “DDDDDDD” is a Meridian 1 internal (set) DN up to seven digits. The prompts PFX1 and PFX2 must be defined by the craftsperson so that the combination of PFX1, PFX2, and the originating DN provides a correct number for the terminating party to dial in order to reach the originating party. If no digits are configured for either of the prefixes, that part of the number will not be included in the Calling Party Number Information Element (IE).

Call Redirection (Call Forward, Hunt, CFNA) and Modification (Conference, Transfer)

All of the following actions apply to the call itself; a notice of redirection or modification (changes in CLID, etc.) is not transmitted. It is possible to Call Forward a set on a Meridian 1 to an external number over an NEAX-61 Primary Rate Interface (this operates exactly the same as over any other Central Office trunk).

A set on the Meridian 1 can Hunt or CFNA to an external number over an NEAX-61 Primary Rate Interface (this operates exactly the same as over any other Central Office trunk).

It is possible to transfer or conference a call on the Meridian 1 to an external number over an NEAX-61 Primary Rate Interface (this operates exactly the same as over any Central Office trunk). Access restrictions may block some transfers from being completed, but this is consistent with other interfaces.

Outpulsing of Asterisk “*” and Octothorpe “#” (OPAO)

As the outpulsed digits can include the octothorpe “#” and asterisk “*”, this feature is used to provide the function. The impact on other applications of OPAO is negligible.

Centrex Switchhook Flash

The Centrex Switchhook Flash feature provides an analog trunk line break to initiate public network features. This capability is expanded to include ISDN signaling to the New Zealand NEAX-61. As the ISDN equivalent is carried out by D-channel signaling, only the triggering of the flash by a set is shared with the analog Centrex flash; signaling via ISDN messaging is independent.

Feature interactions Interworking within the Meridian Customer Defined Network

The following group of features are supported in part by Meridian 1 to New Zealand NEAX-61 ISDN PRI Connectivity. These cannot be supported across the NEAX-61 interface, but will interwork with a call coming in from the PSTN. The portion of the function local to the Meridian Customer Defined Network (MCDN) is supported there.

BRI Sets

The NEAX-61 interface will support calls from BRI sets within the MCDN. These will be able to call out to and receive calls from the Public Switched Telephone Network (PSTN). Operation will be as though the call was a tandem trunk call, with all IEs supported on the interface passed transparently.

BRI Trunks

BRI trunks are supported by the NEAX-61 interface.

Calling Party Name Display

This feature is offered within the MCDN. Names can be assigned to various items, including route access codes. An incoming call across this route will have the name stored against the route displayed. This will continue, however, CPND received from the NEAX-61 is not currently supported, and sending CPND to the NEAX-61 is also not supported at this time.

Calling Line Identification (CLID) Enhancements

MCDN Component CLID enhancements for a redirected call will be seen within the MCDN part of the call. If an incoming PSTN call is redirected, this will be displayed on the terminating device as applicable. If a call is redirected off the MCDN, information up to and including the redirection off of the network will be provided. This feature cannot be offered across the NEAX-61 interface as it depends on an IE not supported by the NEAX-61 (the Redirecting Number IE).

DPNSS Gateway

The NEAX-61 interface interacts with the DPNSS gateway consistently as with other PSTN interfaces. The IEs supported on the NEAX-61 interface which have equivalents in DPNSS interwork on an IE-by-IE basis. IEs that have no matching DPNSS element are discarded. Supported IEs with DPNSS equivalents pass through with appropriate mapping.

Coordinated Dialing Plan (CDP)

A CDP call can access the trunk. However, since neither the private plan nor the CDP numbering type is supported by the NEAX-61, these get converted to unknown type and plan. This applies to both the called and calling number plan and type.

Since the normal use of CDP does not provide a number that can be dialed when calling the public network, the NEAX-61 will replace the CLID with the default number for the system. Normal use of steering codes with Distant Steering Codes (DSCs) and Trunk Steering Codes (TSCs) will be maintained. Use of a Digital Manipulation Index (DMI) to convert the number as required can be done.

Basic Automatic Route Selection (BARS)

This feature has no interactions with the NEAX-61 with the exceptions of: the NPA and NXX cannot be used on this interface, as the public network cannot accept them; and SPNs get converted to “unknown numbering plan and type, as the ESN SPN is a North American concept.

Network Automatic Route Selection (NARS)

All BARS concerns exist for NARS as well. In addition, location codes should be used with care, since the networking features do not accept a Digit Manipulation Index (DMI) used to insert an ESN access code. Therefore, the only supported usage has the trunks effectively treated as though they were non-ISDN.

Network Attendant Service (NAS)

NAS is not supported across the interface except in a few simple ways. An incoming call from the NEAX-61 can be routed using NAS. NAS routing will allow a call to the attendants in night service to be rerouted across the NEAX-61, but no NAS function is possible. The call will be treated as though the call was placed directly, rather than via NAS. Thus, NAS attendant control, signaling, and recall and so forth are not possible.

Network Call Redirection, Network Call Forward, Network Call Forward No Answer – MCDN Component

These features cannot be offered to the NEAX-61 or across its interface. However, they are offered within the MCDN network, and the aspects applicable within the network still apply.

If an incoming call from the NEAX-61 is MCDN call redirected, all signaling information within the Meridian 1 network will be consistent with current operations, but redirection information will be discarded at the Meridian 1 to NEAX-61 interface. If a call originating within the MCDN is redirected to the PSTN, all redirection information up to and including the redirection to the public network will be provided through the network.

Trunk Optimization – MCDN Component

Trunk Optimization within the MCDN will be supported by the appropriate interfaces. It will not be supported across the NEAX-61 interface, but will interwork transparently. In other words, trunk optimization will not be observed at the NEAX-61 interface; the call will appear to be static, not undergoing any modifications.

Virtual Network Services (VNS)

VNS uses a dedicated, MCDN level of service DCH to control a number of temporary B-channels, seized and used on demand. The ISDN signaling requests a connection (SETUP), and a returned FACILITY has a DN that is to be recognized as a VNS virtual DN on the terminating switch. On this VNS DN being received, the original call is completed, since a B-channel has now been obtained.

An NEAX-61 trunk can act as a virtual B-channel.

Feature packaging

There is no new software package for this feature.

The following packages are required for customers who choose to have software that allows them to connect the Meridian 1 over ISDN PRI to the NEAX-61 (New Zealand) interface:

- Digit Display (DDSP) package 19 for Calling Line Identification on ISDN PRI;
- Outputting of Asterisk “*” and Octothorpe “#” (OPAO) package 104;
- Integrated Services Digital Network (ISDN) package 145;
- 2.0 Mbit Primary Rate Access (PRI2) package 154;
- International Primary Rate Access (IPRA) package 202; and
- New Format Call Detail Recording (FCDR) package 234.

To carry out overlap sending to the PSTN, the following packages and their prerequisites are required:

- Overlap Signalling (OVLP) package 184; and
- One or more of:
 - Basic Alternate Route Selection (BARS) package 57;
 - Network Alternate Route Selection (NARS) package 58; or
 - Coordinated Dialing Plan (CDP) package 59.

If the user subscribes to the following services on the PSTN, the ISDN International Supplementary (ISDNS) package 161 is required. The packages as listed below for each feature are also required:

- **PRA Groups (nB+D)** – the International nB+D (INBD) package 255.
- **Malicious Call Trace (MCT)** – the Malicious Call Trace (MCT) package 107.
- **PSTN Three Party Conferencing** – the Trunk Hook Flash (Centrex) (THF) package 157. For FFC access to this feature, the Flexible Feature Codes (FFC) package 139, Controlled Class of Service (CCOS) package 81, and International Supplementary Services (SUPP) package 131 are required.

Feature operation

There are no operating procedures specified for this feature.

Numeris VN3 (France)

This feature provides ISDN PRI connectivity between the Meridian 1 software and the ISDN access software developed by France Telecom (Numeris). The connectivity is based on International 30B+D ISDN Primary Rate Access development.

The following basic services are provided for this initial offering of the ISDN PRI Meridian 1 to Numeris connectivity:

- Basic Call Service
- Calling Line Identification (public and private)
- Confidentiality of Calling Line Identification
- COT, DID, DOD, TIE trunks call types
- 64 kbps clear bearer capability
- Channel Negotiation
- Call Restriction
- Flexible Numbering Plan
- Advice of Charge at end of call, and during call

Operating parameters

The only connection that is supported is the switched circuit mode connection. The X.25 packet mode connection, via the B-channel, the packet mode temporary signaling connection, and the semi-permanent circuit-mode connection are not supported.

The NUMERIS supplementary service control messages not relating to a call are not supported.

The Calling Party Number Information Element may have two occurrences, with two different contents, in a SETUP message sent from the network to the user. For CLID display processing, the Meridian 1 interprets only the contents of the first Calling Number Information Element.

For direct trunk access code dialing, digit analysis is not performed for the enbloc dialing. A time-out rather than an “#” is used as an indication that dialing has been completed.

Overlap Receiving has not been implemented for NUMERIS.

The maximum delay to send an RAI signal upon detection of an LOS, LFAS, LMAS, and AIS is 100 milliseconds. The same applies to the termination of RAI transmission when the alarm condition clears up.

No incoming/outgoing calls can be initiated during an RAI, LFAS, LMAS, AIS, LOS, or ERR alarm condition.

Incoming/outgoing calls must be allowed within 100 milliseconds of cessation of an RAI, AIS, LOS, LMAS, or LFAS alarm condition.

Established calls must not be disconnected by layer 1 processing during an RAI, AIS, LOS, LMAS, LFAS, or ERR alarm condition.

An RAI signal must be sent if an ERR condition of 10^{-3} lasts more than four to five seconds.

The RAI transmission must cease after an ERR condition of 10^{-4} lasts for more than four to five seconds.

The PRI2 link must not be placed in Out-of-service condition due to an RAI, LFAS, LMAS, AIS, LOS, or ERR alarm condition.

If the PRI2 pack is manually disabled, it must go into the Out-of-service state.

Feature interactions

There are no feature interactions specified for this feature.

Feature packaging

To support all of the call services, the following packages must be equipped:

- Integrated Services Digital Network (ISDN) package 145;
- 2.0 Mbit Primary Rate Interface (PRI2) package 154;
- International Primary Rate Access (IPRA) package 202; and
- Digit Display (DDSP) package 19 (required for CLID).

Feature operation

There are no operating procedures specified for this feature.

SwissNet 2 (Switzerland)

This feature provides basic ISDN PRI connectivity between the Meridian 1 and a Central Office meeting SwissNet 2 specifications in Switzerland.

Following is the set of call services provided in the initial ISDN PRI offering for the Meridian 1 to SwissNet basic connectivity:

- Basic Call Service (3.1kHz, speech)
- Calling Line Identification (CLID) and Calling Line Restriction (CLIR)
- COT, DID, DOD, TIE call types.
- 64 kbps clear bearer capability.
- Channel Negotiation
- Flexible Numbering Plan
- Overlap Sending
- Advice of Charge during call

Operating parameters

The only connection that is supported is the switched circuit mode connection. The X.25 packet mode connection, via the B-channel, the packet mode temporary signaling connection, and the semi-permanent circuit-mode connection are not supported.

For Channel Negotiation to operate correctly, customers on the Meridian 1 cannot share a PRI to the SwissNet interface.

For Calling Line Identification to work correctly across a network, the nodes should be connected using a Coordinated Dialing Plan, with DN's of the same fixed lengths on each node.

Overlap Receiving has not been implemented for the SwissNet interface.

ISDN PRA to SwissNet Connectivity is not supported on DPNSS1.

Feature interactions

Calling Line Identification

The CLID will have the following format for Meridian 1 ISDN PRA to SwissNet Connectivity: `zzzz+yyyy+ddddddd` where `zzzz` and `yyyy` are entered in response to prompts PFX1 and PFX2 respectively, in LD 15, and `ddddddd` is an internal DN up to seven digits long. If no digits are configured

for either PFX1 and PFX2, then the Calling Party Number IE will not contain the entire CLID number.

Call Detail Recording (CDR)

For calls made over an ISDN PRI to SwissNet interface equipped with the Advice of Charge at End of Call feature, all Start and Transfer records will show “OVF99” in the overflow count field. This is done to indicate that there will be charge units to be shared among successive call controllers at the end of a call.

Virtual Network Services (VNS)

VNS is supported over an ISDN PRI to SwissNet interface.

Feature packaging

To support all of the call services, the following packages must be equipped:

- Call Detail Recording, Teletype Terminal (CTY) package 5;
- PPM/Message Registration (MR) package 101;
- International Supplementary Features (SUPP) package 131;
- Integrated Services Digital Network (ISDN) package 145;
- 2.0 Mbit Primary Rate Interface (PRI2) package 154;
- ISDN Supplementary Features (ISDNS) package 161;
- International Primary Rate Access (IPRA) package 202;
- Digit Display (DDSP) package 19 (required for CLID); and
- Multi-purpose Serial Data Link (MSDL) package 222 (if required).

Feature operation

There are no operating procedures specified for this feature.

SYS-12 (Norway)

This feature allows ISDN PRI connectivity between the Meridian 1 and the Norwegian SYS-12, using switched-circuit mode connections. The ISDN PRI connectivity between the Meridian 1 and the SYS-12 supports:

- Basic call service
- Connected number delivery
- Calling Line Identification, both public and private
- Increased CDR accuracy
- COT, DID, DOD, and TIE call types
- 64 kb/s clear-bearer capability
- Channel Negotiation
- overlap sending (but not overlap receiving)
- Flexible Numbering Plan

Operating parameters

There are no operating parameters specified for this feature.

Feature interactions

There are no feature interactions specified for this feature.

Feature packaging

To support all of the call services, the following packages must be equipped:

- Periodic Pulse Metering/Message Registration (MR) package 101.

For PRI connectivity, the following packages are required:

- Integrated Service Digital Network (ISDN) package 145;
- 2.0 Mbit Primary Rate Access (PRI2) package 154;
- International Primary Rate Access (IPRA) package 202; and
- ISDN Supplementary Features (ISDNS) package 161.

Feature operation

There are no operating procedures specified for this feature.

1TR6 (Germany)

This feature will allow the interconnection of the Meridian 1 with the 1TR6 protocol. Basic connectivity between the Meridian 1 and the 1TR6 DCH supports:

- Basic call service
- Connected number delivery
- Calling Line Identification, both public and private
- Increased CDR accuracy
- COT, DID, DOD, and TIE call types
- 64 kb/s clear-bearer capability
- Channel Negotiation
- Overlap sending (but not overlap receiving)
- Flexible Numbering Plan
- Advice of Charge at end of call
- Call restrictions to tandem calls

Operating parameters

There are no operating parameters specified for this feature.

Feature interactions

There are no feature interactions specified for this feature.

Feature packaging

To support all of the call services, the following packages must be equipped:

- Periodic Pulse Metering/Message Registration (MR) package 101.

For PRI connectivity the following packages are required:

- Integrated Service Digital Network (ISDN) package 145;
- 2.0 Mbit Primary Rate Access (PRI2) package 154;
- International Primary Rate Access (IPRA) package 202; and
- ISDN Supplementary Features (ISDNS) package 161.

Feature operation

There are no operating procedures specified for this feature.

EuroISDN

The original offering of the EuroISDN feature provided ISDN PRI connectivity between the Meridian 1 and Public Exchanges that comply to the European Telecom Standards Institute (ETSI) specification ETS 300 102 for the Layer 3. The interfaces provided by this feature also comply with the country-specific Application Documents for Austria, Denmark, Finland, Germany, Holland, Ireland, Italy, Norway, Portugal, Sweden, and Switzerland; other countries must comply with ETS 300 102.

The following capabilities were introduced for the Meridian 1 to EuroISDN connectivity:

- Basic call service
- Connected number delivery
- 64 kb/s clear-bearer capability
- Calling Line Identification Presentation,
- Calling Line Identification Restriction, Connected Line Identification Presentation and
- Connected Line Identification Restriction are provided for the above countries where Application Documents are available.
- Channel negotiation
- Advice of Charge (at call set-up, during call, and end of call) is supported in some countries
- interworking with other ISDN or non-ISDN interfaces, including MCDN, QSIG, R2 Multifrequency Compelled Signaling (R2 MFC), 2.0 Mbit Digital Trunk Interface (DTI2), DPNSS1 and DASS2 links, ETSI-NET3 compliant BRI sets, and analog trunk interfaces.

The **EuroISDN Continuation** feature expanded the scope of the original EuroISDN feature to include application support for Switzerland, Spain, Belgium and the United Kingdom.

In addition, the following functionalities are provided by the EuroISDN Continuation feature:

- Intercept treatment upon reception of an invalid or incomplete called party number;
- Interception to an attendant for EuroISDN voice calls terminating on a data device;
- The capability to listen to tones and announcements provided by the Central Office on call clearing (T306 supported);
- The capability of transferring outgoing EuroISDN calls after completion of dialing (only for Italy);
- Calling Line Identification and Connected Line Identification transparency to or from EuroISDN to or from a Basic Rate Interface (BRI) set;
- The configuration of Connected Line Presentation (COLP) on a per D-channel basis (remote capability);
- 3.1 KHz audio bearer capability for outgoing fax calls, based on a Class of Service assigned to the Analog (500/2500 type) telephone;
- The capability of defining the bearer as “Voice” or “3.1 KHz” on a system basis;
- The addition and display of national or international prefix in front of the received Calling/Connected Line Identification on incoming/outgoing EuroISDN calls;
- Flexible national and local prefixes in addition to the Calling/Connected Line Identification on incoming/outgoing calls based on the route configuration;
- User-to-User information transparency in call control messages;
- Capability of mapping a PROGRESS message or Progress Indicator in a CALL PROCEEDING message into an ALERT or a CONNECT message on a per configuration basis;

- Capability of using different options to build the Calling Line Identification (CLID); and
- Called party number size increase to 31 octets.

This development also provides a subset of the ETSI Generic functional protocol for the support of supplementary services. Implementation relies on the Generic Functional (GF) Transport platform introduced in the X11 Release 22 ISDN QSIG GF Transport feature. The ETSI GF subset provides two types of supplementary service control entities as follows:

- **Bearer-related transport with a point-to-point transport mechanism**
This service is used to transport supplementary service protocol information in association with a basic call.
- **Bearer-independent transport with a point-to-point (connection-oriented) mechanism**
This service is used to transport supplementary service protocol information, which is entirely independent of any existing basic call.

The ETSI GF provides a generic transport platform that supports ETSI-compliant supplementary services. The ETSI GF protocol is implemented on the ETSI, Swiss, German and Danish EuroISDN interfaces.

EuroISDN Continuation Phase III added France, Russia and the Ukraine to the list of countries having country-specific Application Documents compliance for EuroISDN.

In addition to providing all of the functionalities provided by the original introduction of EuroISDN and the EuroISDN Continuation, EuroISDN Continuation Phase III introduces the following new functionalities for all EuroISDN interfaces:

- optional sending of last forwarding DN as CLID
- Trunk Route Optimization - before Answer applied to incoming EuroISDN trunks
- Numbering Plan Identification (NPI) and Type of Number (TON) included in CDR tickets for EuroISDN calls

With Release 24, the **EuroISDN ETS 300 403 Compliance Update** feature has been introduced to provide ISDN Primary Rate Interface and Basic Rate Interface to Central Offices that comply with the ETS 300 403-1 European Telecom Standards Institute (ETSI) standard.

Note: In order to support countries that have not yet upgraded to the ETS 300 403 standard, the Meridian 1 still interworks with Central Offices conforming to the ETS 300 102 standard. So, when programming the D-Channel for PRI2 trunks (in LD 17) or a PRI2 route for ISDN trunks (in LD 16) for an ETS 300 403 interface, the following applies:

- If IFC = E403 and CNTY = ETSI (ETS 300 403 for the user side) or NET (ETS 300 403 for the network side), then the interface is fully compliant with ETS 300 403.
- If IFC = E403 and CNTY = any of the supported country entries except ETSI and NET, then the interface behaves like an ETS 300 102 extended version, that is, in addition to the existing ETS 300 102 capabilities, the bearer capability and High Layer Compatibility selection procedures (fall-back mechanism), and the basic telecommunication service identification are also implemented in order to take advantage of new teleservices being offered in Release 24, such as 7kHz telephony and Videotelephony.

Also note that a user may still configure an interface fully compliant with the ETS 300 102 standard if IFC = EURO and CNTY = any of the supported country values.

With the EuroISDN ETS 300 403 Compliance Update feature, all of the functionalities provided by the previous offerings of the EuroISDN features are supported, along with the following enhancements:

- Support for signalling procedures for bearer capability and High Layer Capability selections by, providing repeatable Bearer capability, High layer compatibility Information Elements (IEs) in the SETUP message. ALERT, CALL PROCEEDING, CONNECT, and PROGRESS messages can be used to confirm the selected Bearer Capability/High Layer compatibility for the called user.

This enhancement provides high-quality bearer services or teleservices, with alternate bearer capability or high layer compatibility in case of fall-back.

- Support for Basic telecommunication service identification. Each basic telecommunication service has the required Bearer capability IE encoding and, if applicable, the required High Layer compatibility IE encoding defined for that service. The requested teleservice is identified by taking the presented Bearer capability and High layer compatibility information elements in all combinations. If there is no valid combination, the presented Bearer Capability IE is considered in order to identify a bearer service.

Operating parameters

The EuroISDN feature does not support non-Downloadable D-Channel cards (an NT6D80 MSDL or NTB51AA Downloadable D-Channel card is required on Options 51C, 61C, 81, 81C, and the NTB51BA Downloadable D-Channel card is required on the Option 11C).

Interworking with KD3 signaling is not supported.

Integrated Service Access (ISA) is not supported.

Advice of Charge for EuroISDN calls is supported with the Advice of Charge for EuroISDN feature.

The EuroISDN to CIS MF Shuttle gateway is supported, as provided by the CIS MF Shuttle feature introduced in Release 23.

The Auto Terminate Call feature is not supported.

The Call Back Queuing and Off-hook Queuing feature and the Flexible Hotline feature are not supported with Overlap Signaling.

End-to-End Signaling is supported on all outgoing EuroISDN routes as soon as the CALL PROCEEDING message with a Progress Indicator is received.

Special dial tones after dialed numbers are not supported for incoming calls.

For Connected Line Identification Presentation (COLP) supplementary service, sending or restricting connected line identification is not supported on a per call basis on non-ISDN BRI sets.

For the Optional Sending of Last Forwarding DN as CLID functionality, if more than one ACD redirection takes place when an ACD set is involved, the DN of the called set is sent as the CLID rather than the DN of the last forwarding set.

The EuroISDN feature does not support non-Downloadable D-Channel cards (an NT6D80 MSDL or NTB51AA Downloadable D-Channel card is required on Options 51C, 61C, 81, 81C, and the NTB51BA Downloadable D-Channel card is required on the Option 11C).

Operating parameters pertaining to Call Control Procedures:

- If more than one Channel Identification IE is received in a SETUP message, only the first one is used by the Meridian 1. If it is not available, the call is processed as per the channel negotiation configuration. The call is released if no negotiation is allowed. The transfer of an unanswered EuroISDN call to a remote ringing set requires disconnect supervision on the TIE trunk. If the disconnect supervision is not available, after the external user hangs up, the trunk could be locked out. It is the craftsman's responsibility to ensure that the trunks used for this type of call actually have disconnect supervision.

Operating parameters pertaining to ETS 300 102:

- The user-to-user compatibility checking, by the means of the Low Layer compatibility IE and/or the High Layer compatibility IE, is not supported. The LLC IE and the HLC IE are tandemed by the Meridian 1, but this information is not used to perform any checking on a Meridian 1 node.
- Transit network selection is not supported. This IE is normally used by the user to identify a selected transit network in the SET-UP message. As no European country specifies the coding to use this IE, this service is not supported by the Meridian 1. As a result, this IE is never sent by the Meridian1.
- Extension for symmetric call operation is not supported. This is normally used to implement a private network application.
- Network specific facility selection procedures are not supported.
- D-Channel backup procedures are not supported by EuroISDN.

- Message segmentation procedures are not supported. These are normally used to split messages that are too long.
- Low Layer Information coding principle is not checked by the Meridian 1. No LLC is generated by the Meridian 1, but this information is tandemed if received (from an ISDN BRI set for example).
- Low layer compatibility negotiation procedures are not supported.
- The USER INFORMATION message is not implemented in the Meridian 1 software.

Operating parameters pertaining to ETSI GF:

- ETSI GF gateways to and from other signalling systems, such as DPNSS1, QSIG, and MCDN, are not supported.
- The following ETSI GF procedures are not implemented:
 - control of supplementary services using the separate message approach (HOLD/RETRIEVE).
 - bearer-related broadcast transport mechanism (multipoint configuration).
 - bearer-independent point-to-point connectionless transport mechanism.
 - bearer-independent broadcast transport mechanism.
 - generic notification procedures.
 - network-side channel reservation function.
 - generic procedures for supplementary service management.
 - generic status request procedure.
 - support of the Extended facility information element.

- DN address translation requires the association with a customer number. For an ETSI basic call establishment, the customer number association is found through the B-Channel identified in the channel ID IE. For DN address translation that is not associated with a basic call, the customer number association needs to be determined through other methods.

For a BRI trunk DSL interface, there is a customer number association with the D-Channel. For a PRI interface, a prompt is added for an ETSI D-Channel configuration to create a customer number association with a given D-Channel. This implies that bearer independent messages on a primary rate D-Channel are associated with a single customer as configured in overlay 17. For example, in a multi-customer configuration, if every customer on the switch wishes to use the bearer-independent transport service over ETSI PRI interfaces, then each customer requires a separate D-Channel.

- The Facility Information Element (FIE) is a repeatable IE, and its length is application dependent. However, due to system capacity considerations, such as call register usage and real time usage, there are two types of limitations enforced by the ETSI GF transport platform (actually by the ISDN GF transport platform provided by the ISDN QSIG GF development):
 - the Meridian 1 ETSI GF platform supports up to a maximum of eight ROSE components in one message. The eight components may be included in one Facility Information Element (FIE), or multiple FIEs. In addition, the inclusion of components in a message is also limited by the “available message length”. The available message length is the difference between the maximum message length (260 octets), and the maximum message length taken up by other mandatory and optional IEs supported in the given message.
 - when a supplementary service requests the ISDN ETSI GF transport to send a component which exceed the available message length or the number of components supported, the supplementary service will be notified.

Operating parameters pertaining to ETS 300 403:

- The operating parameters pertaining to ETS 300 102 are applicable to ETS 300 403, except for the user-to-user compatibility checking, which is partially supported in some cases. For instance, as in the case of the EuroISDN 7 kHz/Videotelephony teleservices.

- Codeset 4, which is reserved for use by the ISO/IEC standards, has been added to the ETS 300 403 standard. This codeset is currently not used by the Meridian 1. Therefore, it is not supported.
- The fall-back capability for multirate services is not supported.

Feature interactions

Call Completion Supplementary Service

Call Completion interacts with Call Transfer Over EuroISDN SN3. Call Completion Free Notification can only be presented to the Call Completion originating station and cannot be transferred to any other station. However, once the Call Completion call is established, it behaves as a normal call with respect to Call Transfer.

Call Detail Recording

The Optional Sending of Last Forwarding DN as CLID functionality interacts with the CDR feature. The CLID field in the CDR ticket for the redirected outgoing EuroISDN call is modified according to the CLID information in the SETUP message.

CLID Enhancements

The Optional Sending of Last Forwarding DN as CLID functionality interacts with the ISDN CLID Enhancements feature introduced in Release 22, in that the CLID entry programmed against the prime DN of the forwarding set at a gateway node is used to build the outgoing CLID.

Incoming Digit Conversion Enhancement

The Incoming Digit Conversion Enhancement (IDC) feature converts incoming digits from a DID route. This feature is supported on the incoming EuroISDN DID routes. Digits received as a called party number are converted if the IDC feature is activated on the route. Digit analysis is then performed on the converted digits by the Meridian 1.

Incoming Trunk Programmable CLID

The Optional Sending of Last Forwarding DN as CLID functionality interacts with the Incoming Trunk Programmable CLID feature. This feature, introduced in Release 21, provides the capability of assigning a billing number to incoming trunks.

If the redirection does not occur at the gateway node, the incoming redirecting number information is used as the outgoing CLID on the standard ISDN connectivity DID or CO trunk, even though the Incoming Trunk Programmable CLID feature is configured to send a billing number as CLID information.

Intercept Treatment on Invalid or Partial Dialing

There are three features that are involved in the treatment of calls with dialling irregularities. These are the Partial Dial Timing (PRDL), the Vacant Number Routing (VNR), and the Intercept features.

The Intercept feature allows calls with dialling irregularities to be routed to a Recorded Announcement, an attendant, or to receive overflow or busy tone. Separate treatments can be specified for DID and TIE trunks, for CDP/UDP calls, or for non CDP/ non UDP calls.

The PRDL feature allows to route non-ISDN DID incoming calls to the attendant after a configurable amount of time has expired, if the received digits cannot lead to the completion of the routing.

The VNR feature allows a call to a vacant number to be routed to another node, where the call is either treated as a vacant number, terminated, or given intercept treatment.

In order to provide as much flexibility as possible in the treatment of calls from a EuroISDN interface with dialling irregularities, the PRDL feature has been enhanced. Once the PRDL feature is activated, instead of automatically routing the call to an Attendant, the call will be given the intercept treatment provided by the VNR and Intercept features.

For the PRDL feature, when applied to EuroISDN trunks, the EOD timer used as the PRDL timer must be at least two seconds shorter than the ISDN interdigit timer (T302 = 15 seconds) used for the VNR and Intercept treatments. The call is rejected if T302 times out first. The VNR feature uses another timer which is the NIT timer. Its value is configurable between 3 - 8 seconds, on a per customer basis.

Every time a digit is received, if the number received is an invalid number, the VNR, and then potentially the Intercept treatment feature, are activated.

A number is known to be invalid under the following conditions:

- If Enbloc receiving is configured and an incomplete or invalid number has been received.
- If overlap receiving is configured, and if:
 - an invalid number has been received.
 - an incomplete number with a Sending complete IE has been received.
- If the number received is not completed and not yet invalid, one of the following timer is started:
 - EOD timer (if PRDL = YES)
 - NIT timer for UDP/CDP calls (If PRDL = NO)
 - T302 timer

Interworking Notification

Direct Interworking

When “direct interworking” occurs, the EuroISDN trunk terminates directly on the analog trunk. As a result, the CALL PROCEEDING message is sent to indicate that no further called party information is required. No further call control message is initiated by the analog trunk until answer supervision is provided. If the trunk is a supervised trunk, the resulting CONNECT message will permit the Public Switched Telephone Network (PSTN) to start charging the caller. However, an unsupervised trunk may also be called. In this case, the CONNECT message is sent upon the expiry of the pseudo answer supervision timer.

The destination non-ISDN trunk may be busy instead of ringing; therefore, it is not effective to provide an ALERTING message, since the PSTN provides the caller with ringback. However, the PSTN will be running a T310 timer. Expiry of this timer causes call clearing, and a message must be provided.

The following three alternatives can be configured in Overlay 16 by responding to the PROG prompt to stop the network T310 timer:

- 1 A PROGRESS message is sent to the network after the CALL PROCEEDING message.
- 2 An ALERT message is sent to the network after the CALL PROCEEDING message.
- 3 A CONNECT message is sent to the network after the CALL PROCEEDING message.

Tandem Interworking

When tandem interworking occurs, the EuroISDN trunk terminates on an ISDN link terminating on an analog trunk. When the tandem SETUP message is sent, the CALL PROCEEDING message is sent to the PSTN to indicate that no further called party information is required. The tandem ISDN link returns a PROGRESS message, indicating interworking with a non-ISDN trunk.

Since the PSTN does not implement Annex N, proposed by ETSI, it runs a T310 timer. A message must be provided to stop the T310 timer.

The following three alternatives can be configured in PROG prompt Overlay 16 to stop the network T310 timer:

- 1 A PROGRESS message is sent to the network after the CALL PROCEEDING message.
- 2 An ALERT message is sent to the network after the CALL PROCEEDING message.
- 3 A CONNECT message is sent to the network after the CALL PROCEEDING message.

When interworking with an analog trunk without answer supervised signaling, the Meridian 1 should be configured to send a CONNECT message rather than an ALERT message. An ALERT or CONNECT message stops the receipt of digits, which implies that all digits have been received.

Networking Features

Some networking features currently exist on more than one ISDN interface implemented on the Meridian 1. These features are listed in Table 18. Any networking feature that does not appear in the table is only supported on one ISDN interface and is rejected by the gateway when the service is requested. That is the case for all Meridian Customer Defined Network (MCDN) features that are not supported over the EuroISDN or QSIG interface.

Table 18
Networking features supported by more than one ISDN interface

	Euro-ISDN	MCDN	QSIG	ETSI BRI sets	DPNSS1	MFC	DTI2	Analog
Calling Line ID	Yes	Yes	Yes	Yes	Yes	No	No	No
Connected Number	Yes	Yes	Yes	No	Yes	No	No	No
Transit Counter	No	Yes	Yes	No	Yes	No	No	No

Transfer of Unanswered Call

The EuroISDN Continuation Phase III feature supports the Transfer of an Unanswered Call feature. This allows a station (an attendant or set) to transfer an unanswered outgoing EuroISDN call, after dialing has been completed and before the transferred-to set answers (while the set is ringing). The transferred-to set may be local, or may be remote if the link is an MCDN link equipped with Network Attendant Services (NAS), or if the link is a QSIG link equipped with Slow Answer Recall.

Note: The trunk type for the EuroISDN link must be either CO, DID, TIE, FEX, or WATTS for this functionality to work. Otherwise, the transferred-to set must answer before the transfer can be done.

The transfer of an unanswered EuroISDN call interacts with the CDR feature. When the originating party completes the call transfer after the transferred-to set answers, a CDR Start (S) is generated when the transferred-to set answers. When one side releases the call, a CDR End (E) ticket is generated. If the originating party completes the transfer while the transferred-to set is ringing, an S ticket will be generated only when the transferred-to set answers.

Virtual Network Services

A EuroISDN link can be used as a B-Channel for the Virtual Network Services feature.

Feature packaging

EuroISDN Continuation is included in EuroISDN (EURO) package 261. The following packages may also be required.

For Primary Rate Interface (PRI), the following software packages are required:

- Digit Display (DDSP) package 19
- Integrated Services Digital Network (ISDN) package 145
- 2.0 Mbit Primary Rate Interface (PRI2) package 154
- Integrated Services Digital Network Supplementary Features (ISDN INTL SUP) package 161
- International ISDN PRI (IPRA) package 202
- Multi-purpose Serial Data Link (MSDL) package 222

For Basic Rate Interface (BRI), the following software packages are required:

- Digit Display (DDSP) package 19
- Integrated Services Digital Network (ISDN) package 145
- Integrated Services Digital Network Supplementary Features (ISDN INTL SUP) package 161
- International ISDN PRI (IPRA) package 202
- Meridian 1 Extended Peripheral Equipment (XPE) package 203

- Basic Rate Interface (BRI) package 216
- Integrated Services Digital Network Basic Rate Interface Trunk Access (BRIT) package 233

Overlap Signaling is included in (OVLP) package 184. For the Uniform Dialing Plan (UDP), the following packages are also required:

- Basic Routing (BRTE) package 14
- Network Call of Service (NCOS) package 32
- Digit Display (DDSP) package 19
- Basic Alternate Route Selection (BARS) package 57 or Network Alternate Route Selection (NARS) package 58 (for the Uniform Dialing Plan).

For the Coordinated Dialing Plan (CDP), package 59 and the following packages are required:

- Integrated Services Digital Network (ISDN) package 145
- 2.0 Mbit Primary Rate Interface (PRI2) package 154
- Flexible Numbering Plan (FNP) package 160
- Integrated Services Digital Network Supplementary Features (ISDN INTL SUP) package 161
- International Primary Rate Access (IPRA) package 202

The following packages are required for EuroISDN to interact with Multifrequency Compelled Signaling (MFC), Multifrequency Compelled Signaling for Socotel (MFE), Direct Inward Dialing (DID), Digital Access Signaling System #2 (DASS2), and Digital Private Network Signaling System #1 (DPNSS1):

- Universal ISDN Gateway (UIGW) package 283
- Enhanced DPNSS1 Gateway (DPNSS189I) package 284

The following packages are required for the Numbering Plan Identification (NPI) and Type of Number (TON) fields in CDR:

- Call Detail Recording (CDR) package 4

- Call Detail Recording on Teletype Terminal (CDR_TTY) package 5
- Calling Line Identification in Call Detail Recording (CLID_CDR_PKG) package 118
- New Format for Call Detail Recording (CDR_NEW) package 234

Feature operation

There are no operating procedures specified for this feature.

Japan D70, non-Asia Pacific connectivity

This feature provides ISDN PRI connectivity between the Meridian 1 and the INS1500 D70 for Japan. The design is based on the 23 B+D PRI configuration.

Meridian 1 to Japan D70 ISDN PRI connectivity supports the following:

- Basic Call Service
- Calling Line Identification (public and private)
- Overlap Sending
- COT, DID, DOD, and TIE trunk call types
- 64 kbps clear bearer capability
- Channel Negotiation
- nB+D, up to 120 B-channels/four interfaces
- Public Switched Telephone Network (PSTN) three-party conferencing
- Flexible Numbering Plan
- Malicious Call Trace

Operating parameters

For the ISDN layer 1 interface, the circuit pack QPC720 (Digital Trunk Interface) is used for the Meridian 1 to D70 connectivity.

The QPC757E version of the D-channel Handler (DCH) circuit pack provides the Layer 2 functions and incoming Layer 3 preprocessing for the Meridian 1 to D70 connectivity.

The NTAK09AA (DTI/PRI) circuit pack is used for Option 11 systems, along with the NTAK93 D-channel Interface (DCHI) and NTAK20 Clock Controller.

A Multi-purpose Serial Data Link (MSDL) can be used in place of the DCH circuit pack (NT6D80AA).

Feature interactions

PRI Channel Negotiation

When a D-channel has one or more secondary PRIs associated with it while using the D70 interface, channel negotiation requires that the PRI interface be explicitly defined for all B-channels not on the primary interface.

Feature packaging

For Japan D70 ISDN PRI connectivity, the following software packages are required:

- 1.5 Mbit Digital Trunk Interface (PBXI) package 75;
- Integrated Services Digital Network (ISDN) package 145;
- 1.5 Mbit Primary Rate Access (PRA) package 146;
- International Primary Rate Access (IPRA) package 202;
- Digit Display (DDSP) package 19 for Calling Line Identification on ISDN PRI; and
- Multi-purpose Serial Data Link (MSDL) package 222 for use of the MSDL card in place of the DCH card.

Feature operation

There are no operating procedures specified for this feature.

Intercept Computer Dial from Directory

An Intercept Computer (ICP) is an external information system that can be added to enhance attendant operation. Whenever an attendant answers an internal direct call, or any redirected call due to ICP CFW, the Intercept Computer Terminal (ICT) screen is lit up with information regarding either the caller (for internal calls), or the “called” party (for redirected calls). This information is presented to the attendant who can then give the appropriate information to the caller. For an external call, no information is displayed on the ICT screen.

With the Intercept Computer Dial from Directory feature (ICPD), the attendant does not need to dial the DN from the attendant console; pressing a single key on the ICT keyboard connects the call to the DN, thereby saving the attendant time.

An ICP can be programmed with a directory of the internal Directory Numbers (DNs) in the system. From the ICT, the attendant can search the ICP database for a specific person by name, in order to find that person’s DN, according to a coordinated dialing plan (CDP). Again, pressing a single key on the ICT keyboard connects the call to the corresponding DN.

Feature requirements

This feature is not available for either of the following intercept positions: ACD Agents, or the ICP Answering Machine.

It is only possible to dial from the ICT if the active loop is idle, or has only one part established in a call with the attendant (on SRC or DEST side).

This feature does not support any dialing plan, other than CDP (since this is an already existing limitation of the networking part of the ICP feature).

A maximum of seven digits per incoming message can be received by the PBX.

Feature interactions

Pre-dial Operations

Attendant Barge-in – It is possible for an attendant to Barge-in the following manner:

- Press an idle loop key, and press the Barge-in key from the attendant console.
- Dial a Route Access code and Route member from the ICT (which must be configured in such a way that it is possible to dial the Route access code and Route member from the dialing key).

Pre-dial Break-in – It is possible for an attendant to override call forward on a set in the following manner:

- Press an idle loop key, and press the Break-in key on the attendant console.
- Dial an extension DN from the ICT.

Call Forward Override – This feature is introduced in Release 20 and replaces ICP CFW Override, adding the functionality to override other call forwarding features. It is possible for an attendant to override CFWAC/DND/MSB on a set by dialing the Call Forward Override Flexible Feature Code (FFC) from any set that is configured with a new COS.

Call Park – An attendant can park a call in the following manner:

- Press the Call Park key on the attendant console.
- Dial a DN from the ICT.
- Terminate Call Park operation by pressing the Release key.

Radio Paging Pre-dial Selection – It is possible to start automatic paging in the following manner:

- Dial the pre-dial selection RPA FFC on the Attendant console.
- Dial a DN from the ICT.

Manual radio paging is started as follows:

- Dial the pre-dial selection RPA FCC on the attendant console.
- Dial a DN from the ICT.
- Dial a mode digit, information and octothorpe “#” sign.

Post-dial Operations

Automatic Wake-up – This feature can be requested as follows:

- Press the Wake-up key on the attendant console.
- Dial a DN from the ICT.
- Dial an octothorpe sign “#”, and terminate by dialing the requested wake-up time from the attendant console.

The same approach is used to cancel auto wake-up.

Break-in – An attendant can break-in to a call by:

- Dialing an extension DN from the ICT.
- Pressing the Break-in key on the attendant console.

RPA Post-dial Selection – To start radio paging an extension DN:

- Dial a DN from the ICT.
- Press the RPA post-dialing paging (RPAG) key on the attendant console.

Stored Number Redial – An attendant can dial an extension from the ICT, and then press the Stored Number Redial key to store the called number (following the rules of the Stored Number Redial feature).

Other feature interactions

Attendant Recall with Splitting – If a set transfers a call to the attendant, or a Meridian Digital Telephone set presses the Attendant Recall (ARC) key and the transferring party has not yet completed the transfer before the attendant answers, it is not possible to dial from the ICP (since the transferred party is connected to SRC, and the transferring party is connected to DEST).

Autodial – It is possible to press the ADL key (in which some digits are stored such as an ESN code or FCC), and then dial a DN from the ICP.

Do Not Disturb – This feature can be activated for an extension DN as follows:

- Press an idle loop key, and press the 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 canceling 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 ICT.
- Press the DND IND key on the attendant console.

Message Waiting Indication – To activate the message waiting lamp:

- Press the loop key and the Message Indication (MSG INDIC) key on the attendant console.
- Dial the set's DN from the ICT.
- Press the Message Indication key and the Release key on the attendant console.

The same approach can be used for darkening a Message Waiting lamp by using the Message Cancel key instead of the MSG INDIC key.

Multi-tenant Service (MTS) – The ICP Dial from Directory feature only works at the customer level. If several tenants are configured for a customer, they will all be affected by the ICTD prompt in LD 15.

Network Tenant Service (NTS) – The ICP Dial from Directory feature only works at the customer level and for a single node. If several tenants are configured in a network situation, they will all be affected by how the ICTD prompt in LD 15 has been configured for the customers on different nodes.

Night Key/Position Busy – If the attendant console has the night key activated (i.e., is busy or in Night Service), it is still possible to dial from the ICT.

Slow Answer Recall – If the attendant extends an SRC party to a DEST party on the local node, but slow answer recall occurs since DEST does not answer, dialing a new DN from the ICP is ignored by the PBX (the DEST is still ringing as per slow answer recall operation).

Slow Answer Recall Enhancement – If the attendant extends an SRC party to a DEST party on the local node, but slow answer recall occurs since DEST does not answer, it is possible to dial a new DN from the ICP (the DEST is disconnected when the attendant answered).

Transfer to Attendant – If a set transfers a call to the attendant, and the transferring party has not yet completed the transfer before the attendant has answered, dialing from the ICP is ignored (the transferred party is connected to SRC, and the transferring party is connected to DEST due to the Attendant Recall with Splitting feature).

Feature packaging

The following packages are required:

- Intercept Computer Interface (ICP) package 143
- Basic Automatic Call Distribution (BACD) package 40
- Automatic Call Distribution Package A (ACDA) package 45
- Message Center (MWC) package 46
- Auxiliary Processor Link (APL) package 109
- International Supplementary Features (SUPP) package 131
- Flexible Feature Codes (FCC) package 139, and
- Flexible Tones and Cadences package 125.

To use the ICP Flexible DN length, DN Expansion (DNXP) package 150 is required.

To be able to use ICP in a network environment (depending upon customer requirements) the following packages are needed: Integrated Services Digital Network (ISDN) package 145; 1.5 Mbit Primary Rate Access (PRA) package 146; and Network Attendant Service (NAS) package 159.

ISDN Calling Line Identification enhancements

The capabilities introduced by the ISDN Calling Line Identification (CLID) Enhancements greatly increases the functionality of CLID within any ISDN network.

These enhancements replace the previous method of constructing CLID, and allow much more flexibility when programming CLID for any telephone set on a Meridian 1. Prior to Release 22, a CLID was defined as either a Listed Directory Number (LDN) or Primary Directory Number (PDN). The contents of these data fields were used to generate the CLID for a call. The CLID could only be built from key 0 of a set. This meant that regardless of what key was used to make a call, it was the CLID for key 0 that was sent. Also, only one office code and one location code could have been assigned in the CLID for a customer.

CLID is now table-driven (when LD 15 is loaded, a customer can configure a CLID table), and virtually any of the information contained in the fields of the CLID table can now be programmed against any DN or DN key, on a per set basis. This means that the CLID that is sent from a set is now predicated on what is in the CLID table, rather than the LDN or PDN. That is, a CLID for any key is now built by taking the information contained in a particular field in the CLID table and adding that information to the key's DN. A multi-line set can now have DN keys that each has its own CLID. Or, as explained in "How CLID entries are assigned to a set" on page 315, the CLID of any one key could, on a set, be programmed to use the CLID of any other key on the set.

Also, the Meridian 1 can now support multiple office codes, location codes and steering codes in CLID (this is more fully explained in the section "How a CLID table is built", which follows.) This means that any telephone set on one Meridian 1 can send a CLID that will have calls returned to another Meridian 1. This type of configuration would typically be used in cases where a customer would want calls to be returned to only one central location.

Note: Since the system does not perform verification of CLID entries that are defined in a CLID table, it is the responsibility of the system administrator to ensure that DN keys are programmed correctly.

How a CLID table is built

New prompts have been added to LD 15 that create a CLID table for a customer. This table contains CLID ‘entries’ (up to 4,000.) Each entry contains unique information pertaining to CLID, as explained in the following sections.

North American Numbering Plan

For users of a North American Numbering Plan, the Meridian 1 can now support multiple Home Central Office Codes (HNXXs), Home Numbering Plan Area (HNPA) codes, Home Location Codes (HLOCs) and Local Steering Codes (LSCs), on a DN or DN key basis.

For a North American Numbering Plan, each CLID entry can contain the following:

- a one-six digit national code for a home national number (HNTN);
- a one-12 digit local code for a home local number (HLCL), or a one-12 digit Listed Directory Number for a switchboard;
- a one-seven digit Home Location Code (HLOC); and
- a one-seven digit Local Steering Code (LSC).

International Numbering Plan

For users of an International Numbering Plan, the Meridian 1 can now support multiple Prefix 1 (PFX1) and Prefix 2 (PFX2) contents, and multiple Home Location Codes (HLOCs) and Local Steering Codes (LSCs), on a DN or DN key basis.

For an International Numbering Plan, each CLID entry can now contain the following:

- a one-six digit national code for a home national number (HNTN), which is the equivalent of PFX1;
- a one-12 digit local code for a home local number (HLCL), which is the equivalent of PFX2, or a one-12 digit Listed Directory Number for a switchboard;
- a one-seven digit Home Location Code (HLOC);
- a one-seven digit Local Steering Code (LSC).

Another new capability, which applies to both the North American Numbering Plan and the International Numbering Plan, pertains to how the HLCL is constructed. A new prompt, DIDN (which signifies “use DN as a DID number”) in LD 15, allows the HLCL to be built either using the digits in the HLCL plus the digits of the active key (if DIDN is set to “YES” - the DN is considered to be a DID number and is included in the CLID), or only the digits in the HLCL (if DIDN is set to “NO” - the DN is not included in the CLID since it is not a DID number), or based on a search on the DN keys, beginning from key 0, to find the CLID to be used (DIDN is set to “SRCH”). Please refer to Examples one to six in the “Examples of CLID generation” section starting on page 316.

How CLID entries are assigned to a set

Once the CLID table has been built for a customer, any CLID entry can be assigned to any set, on a per DN or DN key basis (any DN or DN key can be programmed against the information defined for any CLID entry in the CLID table.) This means that for a multi-line set, DN key 0 may send one CLID (the information contained in one CLID entry), DN key 1 may send a different CLID (the information contained in another CLID entry), and DN key 2 may send yet another CLID (the information contained in a third CLID entry). Therefore, a customer may now send different CLIDs from the same set.

Another enhancement pertains to Meridian 1 proprietary sets. When configuring the CLID for a DN key in LD 11, the value entered against the KEY prompt may be a CLID table entry number (0-3999), which corresponds to a CLID entry in the CLID table or ‘D’. If ‘D’ is entered, the Meridian 1 initiates a search on the set for a DN key, starting from key 0. The first found CLID is then sent as the CLID of the active DN key. This means that a call can be made on one key, and the CLID of another key is sent. This configuration is typically used in an ACD or Hotline DN key application, where, for example, the CLID for a particular key is not desired to be sent. Please refer to Examples seven, eight and nine in the “Examples of CLID generation” on page 316.

Prior to the CLID enhancement, for feature keys, such as the Call Transfer key (TRN), the Conference key (AO3 and AO6) and the No-hold Conference key (NHC), the DN of key 0 was used as the CLID DN when a call transfer or conference occurred over ISDN trunks. With the CLID enhancement, the CLID DN is now associated with the active DN key, rather than key 0.

Examples of CLID generation

The examples that follow show CLID generation using different combinations of CLID entries for DN keys, for Private ESN UDP/CDP calls and for Public Local and Public National calls.

Most of the examples use a North American Numbering Plan, since the construction of CLID entries in a CLID is the same for both a North American Numbering Plan and an International Numbering Plan. In some cases, however, examples are provided that pertain to calls using an International Numbering Plan.

Therefore, unless otherwise stated in the headings, the examples provided in the following sections pertain to calls made using a North American Numbering Plan.

The CLID construction for a CLID call type is as follows:

- Private ESN UDP Number - HLOC + DN
- Private ESN CDP Number - LSC + DN
- Public Local Number - HLCL + DN
- Public National Number - HNTN + HLCL + DN

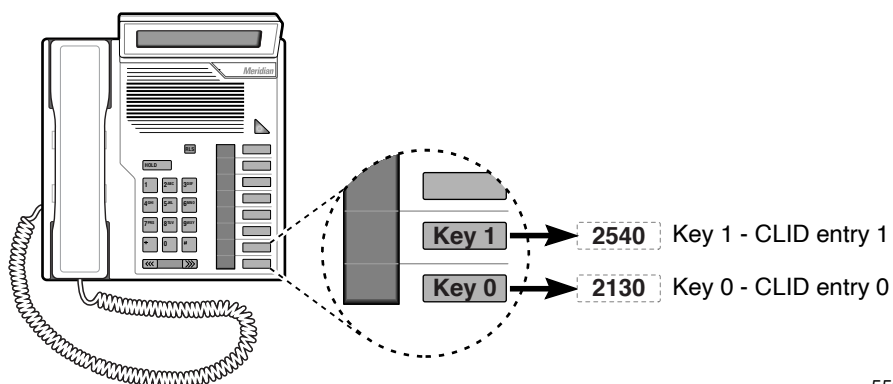
Please note that for calls made using the International Numbering Plan, the CLID for a Public Number is a National Number. For the examples of using the International Numbering Plan, the construction of a Public Local Number is for reference purposes only.

Example 1 - Set has DID number on DN key 0

In this example, the calls are made on DN key 0, which has a DID number. Key 0 is configured against CLID table entry 0, and key 1 is configured against CLID table entry 1.

Table 19
Set has DID number on DN key 0

CLID entry	HNTN (code for the home national number)	HLCL (code for the home local number), or LDN	DIDN (use the DN as the DID?)	HLOC (home location code)	LSC (local steering code)
Entry 0	415	940	YES	646	5
Entry 1	415	9699170	NO	646	5
Entry 2			SRCH	646	5



553-7655.EPS

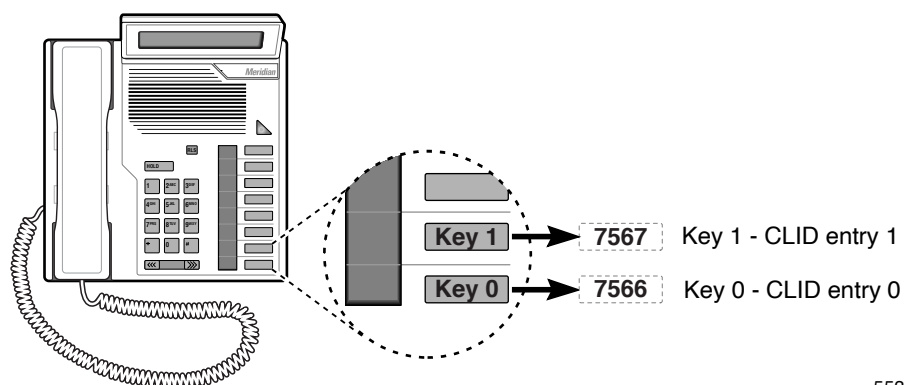
- 1 For a private ESN UDP call made from Key 0 (DN 2130) on the set, the CLID would be built as follows:
 - HLOC from CLID entry 0 (646) plus the DN of the active key (2130).
 - That is, the CLID that is sent would be 646 2130.
- 2 For a private ESN CDP call made from Key 0 (DN 2130) on the set, the CLID would be built as follows:
 - LSC from CLID entry 0 (5) plus the DN of the active key (2130).
 - That is, the CLID that is sent would be 5 2130.
- 3 For a public local call made from Key 0 (DN 2130) on the set, the CLID would be built as follows:
 - HLCL from CLID entry 0 (940) plus the DN of the active key (2130).
 - That is, the CLID that is sent would be 940 2130.
- 4 For a public national call made from Key 0 (DN 2130) on the set, the CLID would be built as follows:
 - HNTN from CLID entry 0 (415) plus the HLCL from CLID entry 0 (940) plus the DN of the active key (2130).
 - That is, the CLID that is sent would be 415 940 2130.

Example 2 - Set has DID number on DN key 0, International Numbering Plan.

In this example, the calls are made on DN key 0, which has a DID number. Key 0 is configured against CLID table entry 0, and key 1 is configured against CLID table entry 1.

Table 20**Set has DID number on DN key 0, for an International Numbering Plan.**

CLID entry	HNTN (code for the home national number, equivalent to PFX1)	HLCL (code for the home local number, equivalent to PFX2), or LDN	DIDN (use the DN as the DID?)	HLOC (home location code)	LSC (local steering code)
Entry 0	2	6476	YES	578	6
Entry 1	2	64767676	NO	578	6
Entry 2			SRCH	578	6



553-7656.EPS

- 1 For a private ESN UDP call made from Key 0 (DN 7566) on the set, the CLID would be built as follows:
 - HLOC from CLID entry 0 (578) plus the DN of the active key (7566).
 - That is, the CLID that is sent would be 578 7566.
- 2 For a private ESN CDP call made from Key 0 (DN 7566) on the set, the CLID would be built as follows:
 - LSC from CLID entry 0 (6) plus the DN of the active key (7566).
 - That is, the CLID that is sent would be 6 7566.
- 3 For a public local call made from Key 0 (DN 7566) on the set, the CLID would be built as follows:
 - HLCL from CLID entry 0 (6476) plus the DN of the active key (7566).
 - That is, the CLID that is sent would be 6476 7566.
- 4 For a public national call made from Key 0 (DN 7566) on the set, the CLID would be built as follows:
 - HNTN from CLID entry 0 (2) plus the HLCL from CLID entry 0 (6476) plus the DN of the active key (7566).
 - That is, the CLID that is sent would be 2 6476 7566.

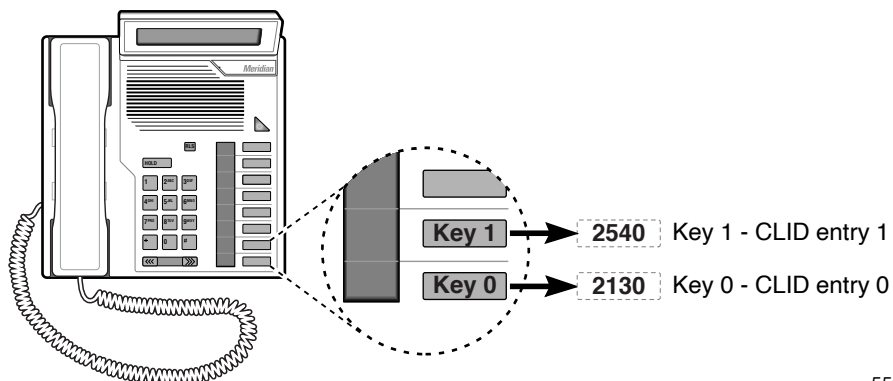
Example 3 - Set does not have a DID number on DN key 1 (DIDN set to NO)

In this example, the calls are made on DN key 1, which does not have a DID number. Key 0 is configured against CLID table entry 0, and key 1 is configured against CLID table entry 1, which has the DIDN option set to NO. This means that the CLID is built using only the digits defined in the HLCL field of table entry 1 (the DN of key 1 would not be sent in the CLID after the HLCL digits.)

Table 21

Set does not have a DID number on DN key 1 (DIDN set to NO).

CLID entry	HNTN (code for the home national number)	HLCL (code for the home local number), or LDN	DIDN (use the DN as the DID?)	HLOC (home location code)	LSC (local steering code)
Entry 0	415	940	YES	646	5
Entry 1	415	9699170	NO	646	5
Entry 2			SRCH	646	5



553-7655.EPS

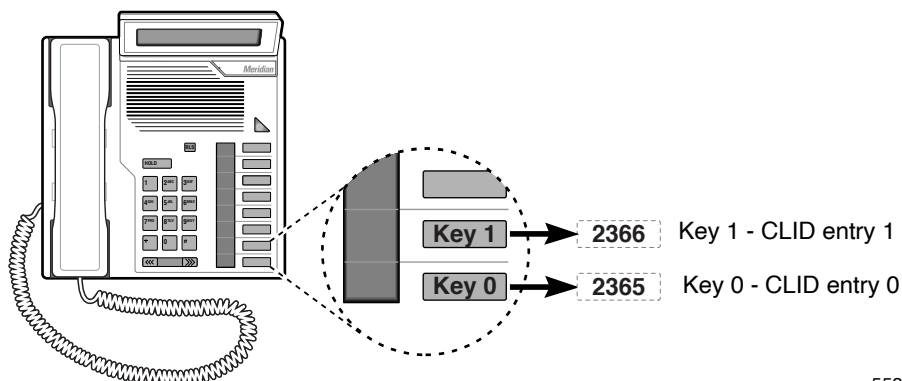
- 1 For a private ESN UDP call made from Key 1 (DN 2540) on the set, the CLID would be built as follows:
 - HLOC from CLID entry 1 (646) plus the DN of the active key (2540).
 - That is, the CLID that is sent would be 646 2540.
- 2 For a private ESN CDP call made from Key 1 (DN 2540) on the set, the CLID would be built as follows:
 - LSC from CLID entry 1 (5) plus the DN of the active key (2540).
 - That is, the CLID that is sent would be 5 2540.
- 3 For a public local call made from Key 1 (DN 2540) on the set, the CLID would be built as follows:
 - HLCL from CLID entry 1 (9699170) - the DN of key 1 is not sent.
 - That is, the CLID that is sent would be 9699170.
- 4 For a public national call made from Key 1 (DN 2540) on the set, the CLID would be built as follows:
 - HNTN from CLID entry 1 (415) plus the HLCL from CLID entry 1 (9699170).
 - That is, the CLID that is sent would be 415 9699170.

Example 4 - Set does not have a DID number on DN key 1 (DIDN set to NO), International Numbering Plan.

In this example, the calls are made on DN key 1, which does not have a DID number. Key 0 is configured against CLID table entry 0, and key 1 is configured against CLID table entry 1, which has the DIDN option set to NO. This means that the CLID is built using only the digits defined in the HLCL field of table entry 1 (the DN of key 1 would not be sent in the CLID after the HLCL digits.)

Table 22**Set does not have a DID number on DN key 1 (DIDN set to NO).**

CLID entry	HNTN (code for the home national number)	HLCL (code for the home local number), or LDN	DIDN (use the DN as the DID?)	HLOC (home location code)	LSC (local steering code)
Entry 0	89	3505	YES	581	6
Entry 1	89	35052020	NO	581	6
Entry 2			SRCH	581	6



553-7657.EPS

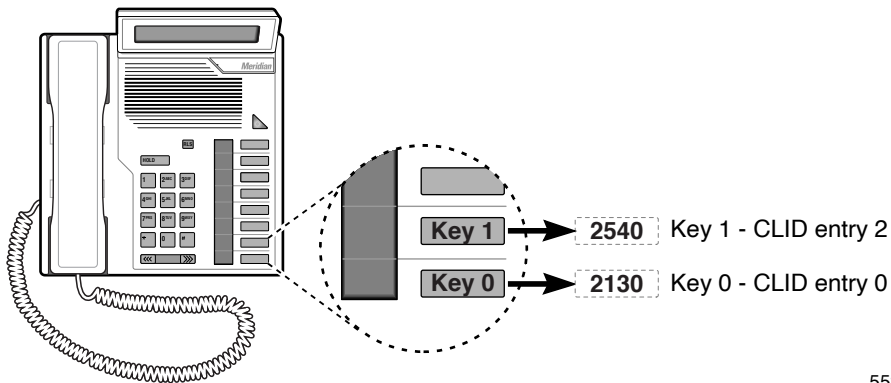
- 1 For a private ESN UDP call made from Key 1 (DN 2366) on the set, the CLID would be built as follows:
 - HLOC from CLID entry 1 (581) plus the DN of the active key (2366).
 - That is, the CLID that is sent would be 581 2366.
- 2 For a private ESN CDP call made from Key 1 (DN 2366) on the set, the CLID would be built as follows:
 - LSC from CLID entry 1 (6) plus the DN of the active key (2366).
 - That is, the CLID that is sent would be 6 2366.
- 3 For a public local call made from Key 1 (DN 2366) on the set, the CLID would be built as follows:
 - HLCL from CLID entry 1 (35052020) - the DN of key 1 is not sent.
 - That is, the CLID that is sent would be 35052020.
- 4 For a public national call made from Key 1 (DN 2366) on the set, the CLID would be built as follows:
 - HNTN from CLID entry 1 (89) plus the HLCL from CLID entry 1 (35052020).
 - That is, the CLID that is sent would be 89 35052020.

Example 5 - Set does not have DID number on DN key 1, SRCH option is used (DIDN is set to SRCH)

In this example, the calls are made on DN key 1, which does not have a DID number. Key 0 is configured against CLID table entry 0, and key 1 is configured against CLID table entry 2, which has the DIDN option set to SRCH. This means that the CLID assigned for key 1 is constructed based on a search starting from key 0. When a CLID entry is found, which has DIDN set to YES (that would be key 0), the HLCL for that key is used in the CLID for key 1. Therefore, the CLID that is sent for key 1 would contain the HLCL of key 0 plus the DN of key 1.

Table 23**Set does not have a DID number on DN key 1, SRCH is used (DIDN is set to SRCH).**

CLID entry	HNTN (code for the home national number)	HLCL (code for the home local number), or LDN	DIDN (use the DN as the DID?)	HLOC (home location code)	LSC (local steering code)
Entry 0	415	940	YES	655	6
Entry 1	415	9699170	NO	655	6
Entry 2			SRCH	655	6



553-7655.EPS

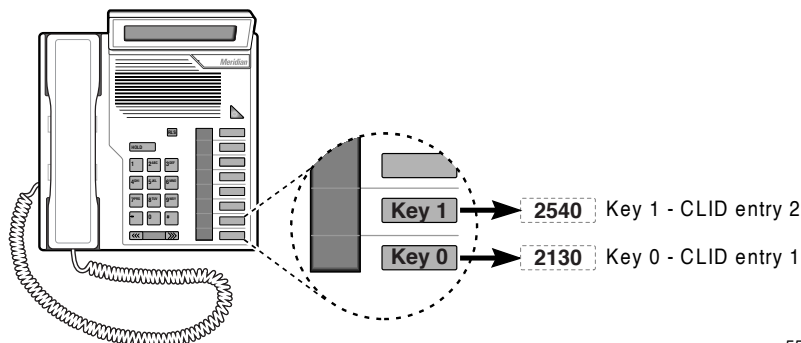
- 1 For a private ESN UDP call made from Key 1 (DN 2540) on the set, the CLID would be built as follows:
 - HLOC from CLID entry 2 (655) plus the DN of the active key (2540).
 - That is, the CLID that is sent would be 655 2540.
- 2 For a private ESN CDP call made from Key 1 (DN 2540) on the set, the CLID would be built as follows:
 - LSC from CLID entry 2 (6) plus the DN of the active key (2540).
 - That is, the CLID that is sent would be 6 2540.
- 3 For a public local call made from Key 1 (DN 2540) on the set, the CLID would be built as follows:
 - HLCL of CLID entry with DIDN set to YES (the HLCL of key 0, which is 940) plus the DN key 0 (2130.)
 - That is, the CLID that is sent would be 940 2130.
- 4 For a public national call made from Key 1 (DN 2540) on the set, the CLID would be built as follows:
 - HNTN from CLID entry 0 (415) plus the HLCL from CLID entry 0 (940) plus the DN of key 0.
 - That is, the CLID that is sent would be 415 940 2130.

Example 6 -Set does not have DID number on DN key 1, SRCH option is used but does not find a DID

In this example, the calls are made on DN key 1, which does not have a DID number. Key 0 is configured against CLID table entry 1, and key 1 is configured against CLID table entry 2, which has the DIDN option set to SRCH. This means that the CLID assigned for key 1 is constructed based on a search starting from key 0. Key 0 is configured against table entry 1. Since table entry 1 has DIDN set to NO, the search next takes place on key 1, which is configured against table entry 2. Since table entry 2 does not have DIDN set to YES either, the search fails to find a CLID entry which has DIDN set to YES. This means that the CLID for key 1 is constructed using only the HLCL of the active DN (that is, the HLCL for key 1, which is the HLCL defined in table entry 2.)

Table 24**Set does not have a DID number on DN key 1, SRCH option fails to find a DID.**

CLID entry	HNTN (code for the home national number)	HLCL (code for the home local number), or LDN	DIDN (use the DN as the DID?)	HLOC (home location code)	LSC (local steering code)
Entry 0	415	940	YES	646	5
Entry 1	415	9699170	NO	646	5
Entry 2	408	9885560	SRCH	655	6



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- 1 For a private ESN UDP call made from Key 1 (DN 2540) on the set, the CLID would be built as follows:
 - HLOC from CLID entry 2 (655) plus the DN of the active key (key 1, 2540).
 - That is, the CLID that is sent would be 655 2540.
- 2 For a private ESN CDP call made from Key 1 (DN 2540) on the set, the CLID would be built as follows:
 - LSC from CLID entry 2 (6) plus the DN of the active key (2540).
 - That is, the CLID that is sent would be 6 2540.
- 3 For a public local call made from Key 1 (DN 2540) on the set, the CLID would be built as follows:
 - HLCL of CLID entry 2 (9885560) - since the search failed to find a DID number, only the HLCL of the active key (key 1, which is configured against table entry 2) is sent in the CLID. Since the DN of Key 1 (2540) is not a DID number, it is not sent in the CLID.
 - That is, the CLID that is sent would be 9885560.
- 4 For a public national call made from Key 1 (DN 2540) on the set, the CLID would be built as follows:
 - HNTN from CLID entry 2 (408) plus the HLCL from CLID entry 2 (9885560) - since the DN of Key 1 is not a DID number, the DN (2540) is not sent..
 - That is, the CLID that is sent would be 408 9885560.

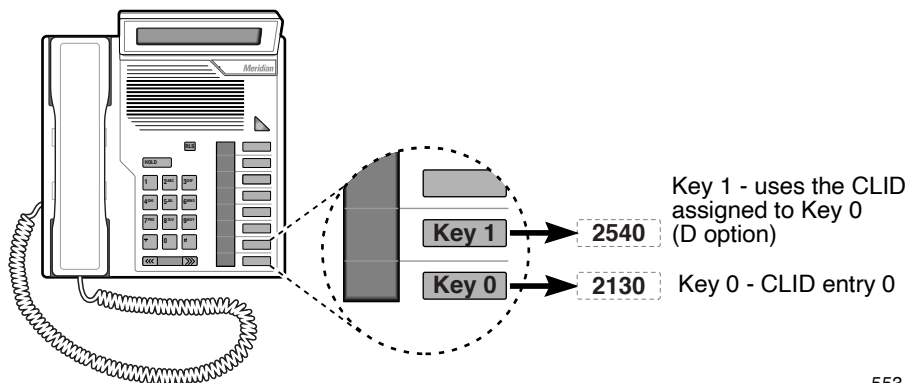
Example 7 - Set with DN KEY prompt of “D” on key 1

In this example, the calls are made on DN key 1, which has the KEY prompt set to “D” in LD 11. This means that a search is done on the set for a DN key, starting from key 0. The CLID of the found DN (key 0 in this case) key is used as the CLID of the active DN key; that is, the CLID for key 0 is used as the CLID for key 1.

Key 0 is configured against CLID table entry 0, and key 1 uses the CLID assigned to key 0.

Table 25
Set with DN KEY prompt of “D” on key 1.

CLID entry	HNTN (code for the home national number)	HLCL (code for the home local number), or LDN	DIDN (use the DN as the DID?)	HLOC (home location code)	LSC (local steering code)
Entry 0	415	940	YES	646	5
Entry 1	415	9699170	NO	646	5
Entry 2			SRCH	646	6



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- 1 For a private ESN UDP call made from Key 1 (DN 2540) on the set, the CLID would be built as follows:
 - HLOC from CLID entry 0 (646) plus the DN for key 0 (2130) - since the Key option has been set to 'D', a search is done from key 0 to find a DN key. The CLID for the found DN key is used as the CLID of active DN key - therefore, CLID for key 0 (the found DN) is used in the CLID for key 1. This also applies to cases 2, 3 and 4 which follow.
 - Therefore, the CLID that is sent would be 646 2130.
- 2 For a private ESN CDP call made from Key 1 (DN 2540) on the set, the CLID would be built as follows:
 - LSC from CLID entry 0 (5) plus the DN of key 0 (2130).
 - That is, the CLID that is sent would be 5 2130.
- 3 For a public local call made from Key 1 (DN 2540) on the set, the CLID would be built as follows:
 - HLCL of CLID entry 0 (940) plus the DN of key 0 (2130).
 - That is, the CLID that is sent would be 940 2130.
- 4 For a public national call made from Key 1 (DN 2540) on the set, the CLID would be built as follows:
 - HNTN from CLID entry 0 (415) plus the HLCL from CLID entry 0 (940) plus the DN of key 0 (2130).
 - That is, the CLID that is sent would be 415 940 2130.

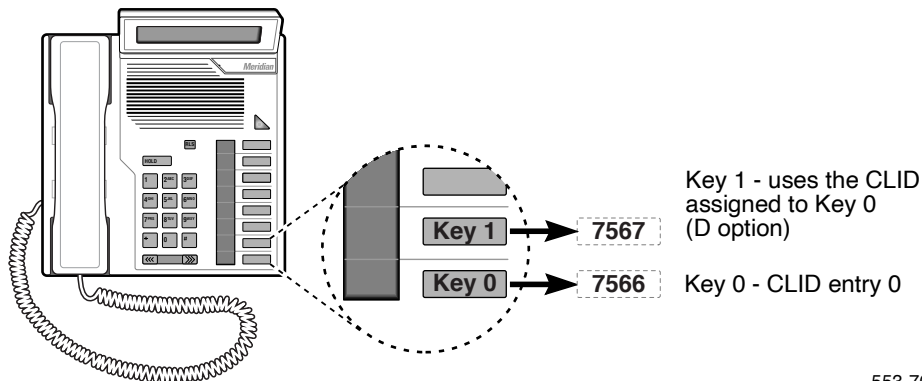
Example 8 - Set with DN KEY prompt of “D” on key 1, International Numbering Plan

In this example, the calls are made on DN key 1, which has the KEY prompt set to “D” in LD 11. This means that a search is done on the set for a DN key, starting from key 0. The CLID of the found DN (key 0 in this case) key is used as the CLID of the active DN key, that is, the CLID for key 0 is used as the CLID for key 1.

Key 0 is configured against CLID table entry 0, and key 1 uses the CLID assigned to key 0.

Table 26
Set with DN KEY prompt of “D” on key 1.

CLID entry	HNTN (code for the home national number)	HLCL (code for the home local number), or LDN	DIDN (use the DN as the DID?)	HLOC (home location code)	LSC (local steering code)
Entry 0	2	6476	YES	578	6
Entry 1	2	64767676	NO	578	6
Entry 2			SRCH	578	6



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- 1 For a private ESN UDP call made from Key 1 (DN 7567) on the set, the CLID would be built as follows:
 - HLOC from CLID entry 0 (578) plus the DN for key 0 (7566) - since the Key option has been set to 'D', a search is done from key 0 to find a DN key. The CLID for the found DN key is used as the CLID of active DN key - therefore, CLID for key 0 (the found DN) is used in the CLID for key 1. This also applies to cases 2, 3, and 4 which follow.
 - Therefore, the CLID that is sent would be 578 7566.
- 2 For a private ESN CDP call made from Key 1 (DN 7567) on the set, the CLID would be built as follows:
 - LSC from CLID entry 0 (6) plus the DN of key 0 (7566).
 - That is, the CLID that is sent would be 5 2130.
- 3 For a public local call made from Key 1 (DN 7567) on the set, the CLID would be built as follows:
 - HLCL of CLID entry 0 (6476) plus the DN of key 0 (7566).
 - That is, the CLID that is sent would be 6476 7566.
- 4 For a public national call made from Key 1 (DN 7567) on the set, the CLID would be built as follows:
 - HNTN from CLID entry 0 (2) plus the HLCL from CLID entry 0 (6476) plus the DN of key 0 (7566).
 - That is, the CLID that is sent would be 2 6476 7566.

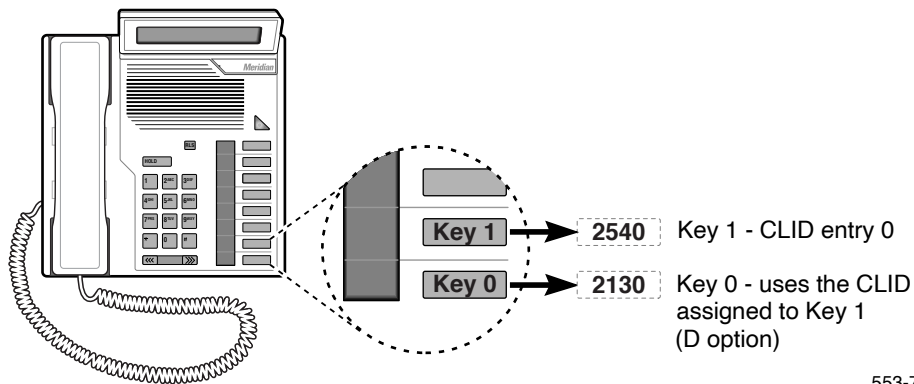
Example 9 - Set with DN KEY prompt of “D” on key 0

In this example, the calls are made on DN key 0, which has the KEY prompt set to “D” in LD 11. This means that a search is done on the set for a DN key, starting from key 0. The CLID of the found DN key (key 1 in this case) is then used as the CLID of the active DN key, that is, the CLID assigned to key 1 is used as the CLID for key 0.

Key 1 is configured against CLID table entry 0, and key 0 uses the CLID assigned to key 1.

Table 27
Set with DN KEY prompt of “D” on key 0.

CLID entry	HNTN (code for the home national number)	HLCL (code for the home local number), or LDN	DIDN (use the DN as the DID?)	HLOC (home location code)	LSC (local steering code)
Entry 0	415	940	YES	646	5
Entry 1	415	9699170	NO	646	5
Entry 2			SRCH	655	6



553-7655.EPS

- 1 For a private ESN UDP call made from Key 0 (DN 2130) on the set, the CLID would be built as follows:
 - HLOC from CLID entry 0 (646) plus the DN for key 1 (2540) - since the Key option has been set to 'D', a search is done from key 0 to find a DN key. The CLID for the found DN key is used as the CLID of active DN key - therefore, CLID for key 1 (the found DN) is used in the CLID for key 0. This also applies to cases 2, 3, and 4 which follow.
 - Therefore, the CLID that is sent would be 646 2540.
- 2 For a private ESN CDP call made from Key 0 (DN 2130) on the set, the CLID would be built as follows:
 - LSC from CLID entry 0 (5) plus the DN of key 1 (2540).
 - That is, the CLID that is sent would be 5 2540.
- 3 For a public local call made from Key 0 (DN 2130) on the set, the CLID would be built as follows:
 - HLCL of CLID entry 0 (940) plus the DN of key 1 (2540).
 - That is, the CLID that is sent would be 940 2540.
- 4 For a public national call made from Key 0 (DN 2130) on the set, the CLID would be built as follows:
 - HNTN from CLID entry 0 (415) plus the HLCL from CLID entry 0 (940) plus the DN of key 1 (2540).
 - That is, the CLID that is sent would be 415 940 2540.

Operating parameters

In order for CLID to be properly delivered, a CLID entry must be defined in the CLID table for a customer. If a CLID entry or table is not defined, the active DN is sent as CLID.

The ISDN Calling Line Identification (CLID) Enhancements feature only pertains to calls that are made over ISDN routes.

The CLID that is sent from the Meridian 1 may be subject to any restrictions which may be imposed by the serving Telco.

This feature does not change the operation of CLID for attendant consoles. The codes for different call types are extracted from CLID entry 0 in LD 15, the customer data block. If the call type is private, the attendant DN is used. If the call type is public, LDN0 is used.

There is no system validation of digits entered in the CLID entry via a service change. Therefore, caution must be taken that any digits that are entered are valid and correct.

There is no cross-validation of the CLID entry associated with a set against the CLID entry entered in LD 15. If a CLID entry is deleted from the CLID table and the CLID entry remains assigned to a DN key, the active DN is sent as CLID.

The maximum number of digits for a CLID in an ISDN message is 16.

The restriction remains of constructing a seven-digit local number or a 10-digit national number, for the North American Numbering Plan. Also, a seven-digit maximum is maintained for an ESN Uniform Dialing Plan number, if the Flexible Numbering Plan package is not equipped. The seven and 10 digit restriction for a public number does not apply to International ISDN interfaces.

The CLID is constructed using the information in the CLID table associated with the active DN key for the calling party in the Setup message and the connected party in the Connected and Notify messages. This feature does not change the Information Elements pertaining to any other ISDN network DNs, such as for Redirecting and Original Called Number.

For ISDN supplementary services, the ISDN CLID enhancements do not change how a Network DN is constructed. For a Local Number, the Network DN is formed using the Home Central Office Code (HNXX)/Prefix 2 (PFX2). For ESN Numbers, the Network DN is formed using the Home Location Code (HLOC) for a Uniform Dialing Plan number or the Local Steering Code (LSC) for a Coordinated Dialing Plan number.

Feature interactions

Attendant Administration

Administration of a CLID entry, for a set from an attendant console, is not supported.

Automatic Call Distribution

The ACD DN is sent as the CLID for a call made by an ACD agent using a DN key on a key other than Key 0. With the feature enhancement, the CLID is constructed using the CLID entry associated with the active DN key. The ACD agent ID is not designed to be sent as the CLID.

If an ACD agent has an active call on Key 0 and if a call transfer or conference is initiated by the ACD agent, the CLID entry associated with the ACD DN Key 0 is used as the CLID (for remote calls only.)

There is no CLID entry for an analog (500/2500 type) ACD DN. The CLID associated with the analog (500/2500 type) DN is used when a call transfer or conference is initiated by an ACD agent on an analog (500/2500 type) set.

Call Detail Recording

The CLID in the CDR records, including the X records, contains the DN of the key from which the call is made, not the DN of Key 0.

Call Pickup Network Wide

The Private Integrated Services Network Exchange (PINX) DN in the customer data block is used for Call Pickup Network Wide. The network DN for the PINX DN is constructed using the existing Home Location Code (HLOC) or Local Steering Code (LSC) in LD 15. The DN of the originating party is constructed using the CLID associated with the active DN key. The DN of the originally called (ringing) party is constructed using the existing HLOC or LSC in LD 15.

For calls picked up from a secondary DN, the redirection DN in the Notify message is formatted with the CLID entry 0 of LD 15.

Calling Party Name Display

If a call transfer or conference is initiated on a multiple appearance DN programmed on a key other than Key 0, the Call Party Name Display associated with the DN of the active key is used, rather than the Call Party Name Display for Key 0.

CLID for an ISDN BRI set

For an internal call terminating on an ISDN BRI set, the calling set's Dialed Digits Denied (DDGD)/Dialed Digits Allowed (DDGA) Class of Service is used to determine whether to send Calling Party Number to the terminating ISDN BRI set for display purposes.

Connected Number

When a call is modified, such as by a call transfer, this feature enhancement will try to use the CLID entry associated with the active DN key if available, otherwise the connected number will be constructed using CLID entry 0.

EuroISDN Continuation for UK/Spain/Belgium/SN3

The EuroISDN Continuation feature allows Home National Numbers and Home Local Numbers to be configured on a route. When an ISDN call is made from a set to a EuroISDN interface, the CLID constructed by EuroISDN, based on the outgoing route, takes precedence over the CLID constructed for the calling station set.

EuroISDN Master Mode

The EuroISDN Master Mode connectivity supports all of the user side ISDN CLID enhancements introduced by Generic X11 Release 22 software.

Network Attendant Service

If Network Attendant Service is equipped, CLID entry 0 is used for incoming trunks.

Network Call Redirection

Network Call Redirection constructs Redirecting Number and Redirection Number. The feature enhancement does not change the construction of the Redirecting Number. However, the Redirection Number of the Notify message is constructed using the CLID entry 0 of LD 15.

Network Message Services

Message Waiting Indication with DMS

When a user leaves a voice message, from a multiple appearance DN, on a key other than Key 0 (such as Key 1), the caller's recorded number will be the multiple appearance DN on Key 1, rather than the primary DN of Key 0. This means that when the user returns the call, he/she will ring the DN of Key 1 on all the sets that have the appearance of the DN.

When a user retrieves messages using a multiple appearance DN key other than Key 0, the user now retrieves the messages on the other DN key.

The mail box number for a Network Message Service user is based on the original called number, in the form of a Coordinated Dialing Plan (CDP) or Uniform Dialing Plan (UDP) number. The construction of the original called number is not changed. If a user's DN is using a different Home Location Code or Local Steering Code for CLID, the Meridian Mail database adds the user's CLID as the mail box number. For example, if the first called number for a user is 6462300, a UDP number, and the user's CLID is 6472300, using CLID entry 'x', the user's mail box numbers are 6462300 and 6472300.

In general, with the enhancement of sending the DN associated with an active key to make a call to Meridian Mail, the secondary key's DN is included in the Meridian Mail User's definition (for either a local or remote user.)

Network Ring Again

The Network Ring Again feature remains operational when a user uses a multiple appearance DN on a key other than Key 0 to activate Network Ring Again, since Network Ring Again saves the Terminal Number of the set that initiates Network Ring Again.

Remote Virtual Queuing

The Remote Virtual Queuing (RVQ) feature remains operational when a user uses a multiple appearance DN on a key other than Key 0 to activate RVQ.

Transaction Capabilities Application Part (TCAP)

The network DN for TCAP messages is constructed using the existing Home Location Code (HLOC), Local Steering Code (LSC) or Home Numbering Plan Area (HNPA)/Home Central Office Code (HNXX) prompts in LD 15. The feature enhancement does not change the construction of the TCAP messages.

Feature packaging

The ISDN Calling Line Identification (CLID) Enhancements feature does not introduce any new package. However, Integrated Services Digital Network (ISDN) package 145 is required as a dependency.

Feature operation

There are no operating procedures specified for this feature.

ISDN QSIG

The European Computer Manufacturer's Association (ECMA) has defined an ISDN protocol that specifies the Layer 3 signaling requirement for support of circuit switched call control at the "Q" reference point between Private Telecommunications Network Exchanges (PTNXs) connected within a Private Telecommunications Network (PTN). This protocol has been adopted by the European Telecommunications Standards Institute (ETSI) and the International Standards Institute (ISO). Most of the major European PTNX manufacturers will be supporting ISDN connectivity based on this standard.

QSIG is oriented towards signaling and services that occur between two switches. For example, two PBXs, or a PBX and a Centrex switch could exchange signaling for services across a "Q" reference point.

The QSIG interface will support the following services:

- Call Establishment and Tear Down
- ETSI or ISO version of basic call
- 64 kbps clear data
- Overlap Sending/Receiving
- Channel Negotiation
- Calling and Connected Parties Information (CLIP/COLP)
- Calling and Connected Parties Restriction (CLIR/COLR)
- Flexible Numbering Plan
- TIE call types

- nB+D with n ranging from 1 to 480 for PRI, and
- Transit Count information transmitted when ISDN Call Connection Limitation (ICCL) is present.

This interface is supported on PRI, and BRI, as well as ISDN Signaling Links (ISLs). The QSIG interface does not have any transit capability for supplementary services, nor does it support any supplementary services besides the CLIP/COLP and CLIR/COLR.

QSIG development underscores Northern Telecom's commitment to global standards. QSIG will provide a greater interworking of voice, image, video, and data services in multi-vendor environments. Users will also be able to combine ISDN PRI with ISDN BRI to deliver enhanced services via end-to-end ISDN networks.

Operating parameters

The QSIG Basic Call feature requires the Multi-serial Data Link (MSDL) circuit pack NT6D80AA, as well as the 2.048 Mbits PRA Access card NT8D72AA for the Layer 1 interface for large systems. The Downloadable D-channel Daughterboard NTBK51 is needed on the Option 11 to handle the Layer 2 function and the Layer 3 protocol control function. For Option 11, the NTBK50 XPRI2 is used for the Layer 1 interface.

Feature interactions

Networking Features

Some networking features currently exist on more than one ISDN interface implemented on Meridian 1. These features are listed in the following table. Any networking feature that does not appear in the table is only supported on one ISDN interface and is rejected by the gateway when the service is requested. That is the case for all Meridian Customer Defined Network (MCDN) features that are not supported over the QSIG interface.

	Euro-ISDN	MCDN	QSIG	BRI sets	DPNSS	MFC	MFE	DTI2	Analog	1TR6	NUMERIS	AXE10
Calling Line ID	Y	Y	Y	Y	Y	Y	N	N	N	Y	Y	Y
Connected Number	Y	Y	Y	N	Y	N	N	N	N	Y	N	N
Transit Counter	N	Y	Y	N	Y	N	N	N	N	N	N	N
Call Charge	Y	N	N	N	N	N	N	Y	Y	Y	Y	N

Trunk Route Optimization

Trunk Route Optimization (TRO) is supported within the Meridian 1 network only. When a redirecting node sends a message to the originating node and the TRO request is accepted, the new call can go through the QSIG interface as a normal basic call. However, TRO signaling will not be able to operate on the QSIG interface.

Network Call Redirection

The existing Network Call Redirection limitation on unsupported interfaces applies to the QSIG interface. When a call is terminated on the Meridian 1 and network call redirection is active, the feature can still be operated but the Original Called number and the Redirection number Information Elements (IEs) which are used by the Network Call Redirection feature will not be sent on by the QSIG interface.

ISDN Signaling Link (ISL)

The existing ISL operation is supported on the QSIG interface on the PRI/PRI2 interface only.

Network Attendant Service (NAS)

The interaction with NAS is as if the call is going to a route without NAS equipped. The only information exchanged between NAS and the QSIG interface is the transit count. Other information contained in the Progress IE will be handled by a future development.

Virtual Network Services (VNS)

VNS provides ISDN features when no MCDN link is available between two switches. This can be done when analog lines are available, or when ISDN links that provide no supplementary services are used. The existing link is used as a B-channel, and a separate D-channel handles the MCDN signaling between the two end switches.

A QSIG link can be used as a B-channel for the VNS feature over a private network. All VNS services are then supported as normal; the QSIG link is only used as a speech bearer.

Feature packaging

A new QSIG Interface (QSIG) package 263 must be provisioned to activate this feature.

The following software packages are also required: International Primary Rate Access (IPRA) package 202, Digit Display (DDSP) package 19; Integrated Services Digital Network (ISDN) package 145; 2Mbit Primary Rate Access (PRI2) package 154; and Overlap Signaling (OVLP) package 184.

ISDN QSIG Alternate Routing

The QSIG Alternate Routing feature provides a solution to calls encountering congestion due to high traffic situations within a QSIG network. Both the QSIG interface protocols, European Telecommunications Standard Institute (ETSI), and International Standards Organization (ISO), are supported by this feature.

This feature uses the routing capability of Network Alternate Route Selection (NARS) to re-route a congested call. For each QSIG call translated at a Meridian 1 node, NARS selects one route from up to 512 routes to complete the call. These routes are programmed in a route list. Each route in the list is called an entry. There may be up to 64 entries in each route list. Any combination of trunks (such as public exchange, TIE) can be specified in a route list. QSIG Alternate Routing may be configured for each of the 512 different routes.

Congestion occurs when all trunks of a route are busy. With the introduction of the QSIG Alternate Routing feature, each entry of a route list on one node can be configured to take an alternate entry (route) from the route list of that node (Private or Public Exchange), if congestion is encountered.

Using LD 86, the Electronic Switched Network (ESN) administration overlay, the option defined for the Step Back On Congestion (SBOC) prompt determines the type of alternate routing available to calls over a particular route. These options may be as follows:

- NRR (no alternate routing is performed. The call receives congestion treatment).
- RRO (reroute if a call encounters congestion at the originating node. If congestion is encountered at a transit node, the call drops back to the originating node, so that the originating node decides if re-routing is required. The drop-back functionality, which is part of the ISDN Drop Back Busy feature, is also configured in LD 86, using the IDBB prompt).
- RRA (reroute the call at any node, whether congestion is encountered at the originating or transit node).

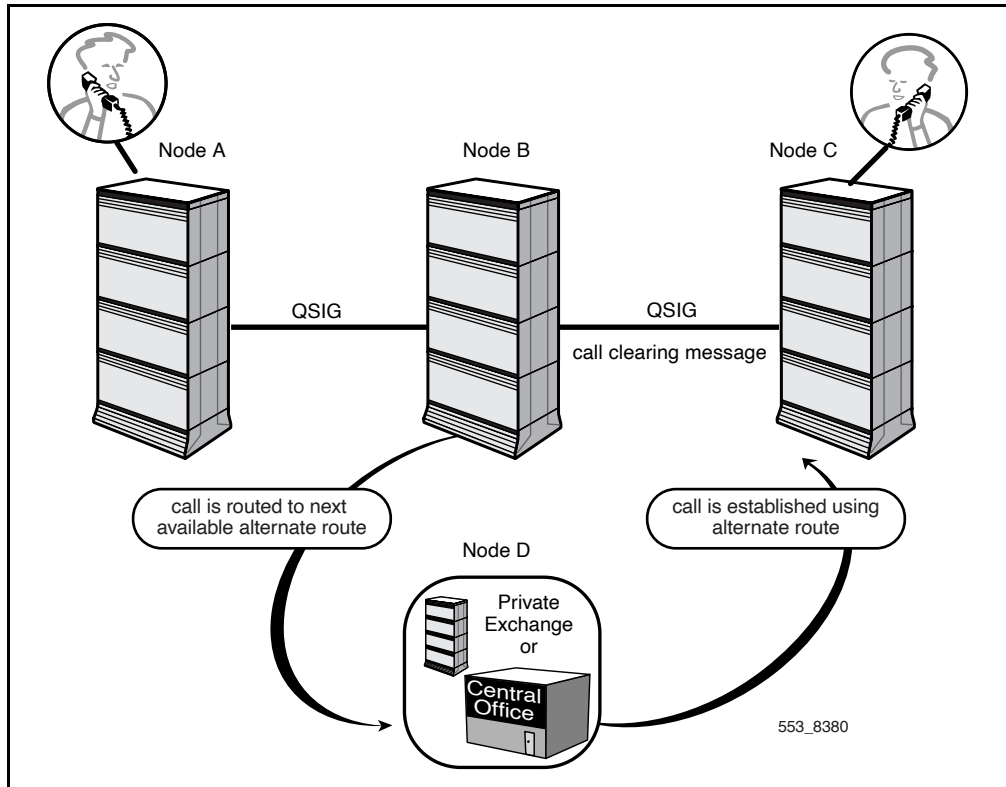
QSIG Alternate Routing is triggered at the controlling node when a Call Clearing message (DISCONNECT or a REJECT) is received, and the cause value is to activate QSIG Alternate Routing. The cause values are defined using LD 86.

Transit node operation

Consider the following calling scenario for a transit node operation (refer to Figure 25). An attempt is being made to establish a call over a QSIG link, from originating node A to terminating node C, via transit node B. All the trunks pertaining to the call attempt at node C are busy. Node C sends congestion message information, along with a supported QSIG Alternate Routing cause value, back to node B.

At node B, the SBOC option, as defined in LD 86, is checked to determine the routing treatment. If SBOC = RRA, the next free alternate route is tried. In our example, the first free alternate route that is found is between node B and node D. Node D can either be a Private or Public Exchange. If the Network Class of Service access checks are passed, the direct leg between the congested node (node C) is released, and a indirect new leg is created to node C, through node D. The call would then be established from node A, to node B, to node D, to node C.

Figure 25
Transit node operation



An attempt is made to find a free alternate route until all of the alternate routes, as defined in LD 86, are tried. If no free alternate route is available, the congestion message information is sent from node B to the originating node A, where the QSIG Alternate Routing functionality is activated. If no alternate routes are found, the call receives network blocking treatment, as defined by prompt NBLK in LD 15, as part of the initial customer configuration.

If SBOC = RRO, the congestion information is passed back from transit node B to the originating node A. At node A, QSIG Alternate Routing is activated in an attempt to find an alternate route to set up the call from node A, to node B, to node D, to node C.

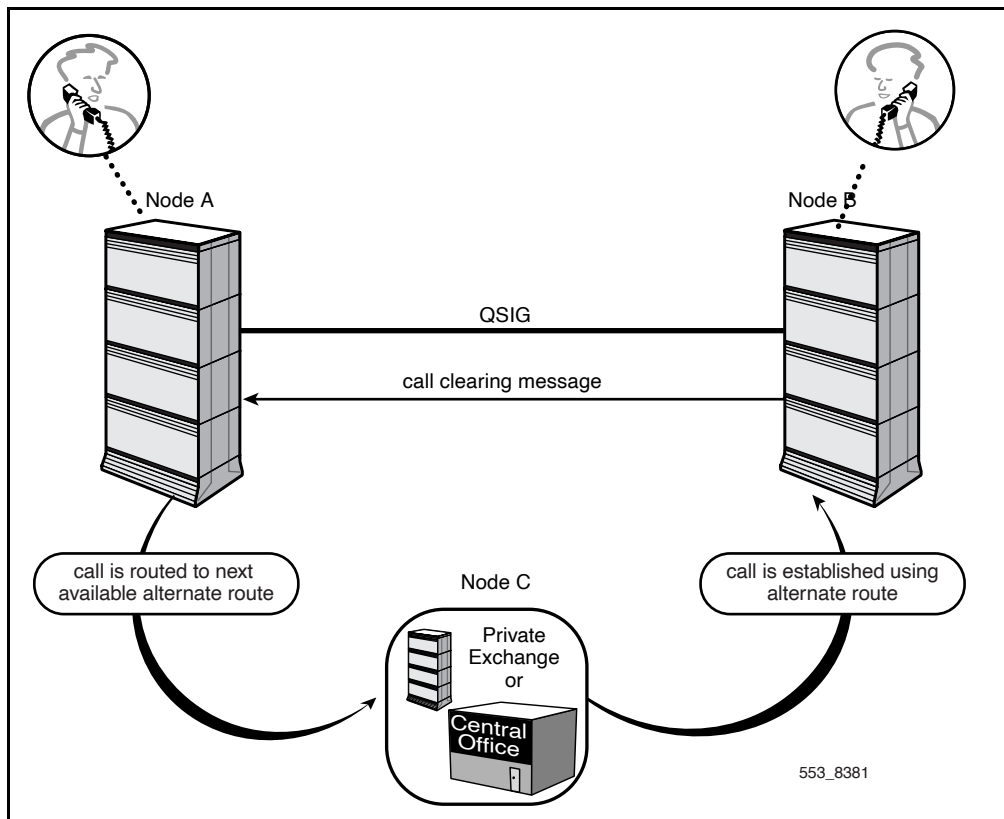
If SBOC = NRR, no alternate routing is performed. The call receives network blocking treatment, depending on the congestion cause value and the type of treatment defined by prompt NBLK in LD 15.

Originating node operation

Consider the following calling scenario for an originating node operation (refer to Figure 26). An attempt is being made to establish a call over a QSIG link, from originating node A to terminating node B. All the trunks All the trunks pertaining to the call attempt at node B are busy, so node B sends congestion message information, along with a supported QSIG Alternate Routing cause value, back to node A.

At node A, the SBOC option, as defined in LD 86, is checked to determine the routing treatment. If SBOC = RRA or RRO, the next free alternate route is tried. In our example, the first free alternate route that is found is between node A and node C. Node C can either be a Private or Public Exchange. If the Network Class of Service access checks are passed, the direct leg between the congested node (node B) is released, and a new indirect leg is created to node B, through node C. The call would then be established between node A, to node B, to node C.

Figure 26
Originating node operation



An attempt to find a free alternate is made until all of the alternate routes, as defined in LD 86, are tried. If no alternate routes are found, the call receives network blocking treatment, as defined by prompt NBLK in LD 15, as part of the initial customer configuration.

If SBOC = NRR, no alternate routing is performed. The call receives network blocking treatment, depending on the congestion cause value and the type of treatment defined by prompt NBLK in LD 15.

Operating parameters

There are two forms of QSIG Call Transfer - QSIG Call Transfer by the “rerouting” method, and QSIG Call Transfer by the “join” method. QSIG Alternate Routing and the QSIG Call Transfer by the “re-routing” method are mutually exclusive.

Optional Information Elements (IEs) may be lost at a rerouting node. Only the following optional Information Element (IEs) are tandemmed in the SETUP of the alternate call, if they are present in the congested call:

- Calling number
- Called and calling subaddress
- High layer compatibility
- Low layer compatibility

On a QSIG network, optional IEs (like progress IEs) carried in the SETUP messages are lost when the alternate routing occurs at a transit node. Mandatory IE are always tandemmed transparently.

GF facility information is lost at a transit node. If QSIG Alternate Routing is performed, Facility IEs contained in the SETUP message are lost, so services like QSIG Name Display, QSIG Call Diversion (rerouting and by join method) are not delivered to the end-user. QSIG Alternate Routing is exclusive with QSIG Call Transfer, if the rerouting method is used. This does not have a major impact, since QSIG Call Transfer by rerouting is optional, whereas QSIG Call Transfer by join is mandatory. It is therefore advised that if a third-party PBX is part of a QSIG network, QSIG Call Transfer by join method is used.

Missing progress IEs due to QSIG Alternate Routing activation may create a situation where a call fails involving an ISDN BRI set. This may occur when a Meridian proprietary set calls an ISDN BRI set over a QSIG link, and QSIG Alternate Routing is triggered. In this case the Progress Indicator Number 3 “Originator is not ISDN” is lost.

Route Access codes are not supported. A Coordinated Dialing Plan (CDP) or a Uniform Dialing Plan (UDP) is required for the QSIG Alternate Routing feature.

Feature interactions

Drop Back Busy and Off-Hook Queuing

QSIG Alternate Routing takes precedence over ISDN Drop Back Busy, if both are configured on an MCDN-to-QSIG gateway node.

Intercept treatment

If QSIG Alternate Routing fails to find an alternate route for a call encountering congestion at a transit node, intercept treatment is not applied at the transit node. The call is dropped back to the originating node, with the appropriate congestion IE information.

Overlap Sending

QSIG Alternate Routing is supported over both the enbloc and overlap signaling methods.

QSIG Name Display

When QSIG Alternate Routing is triggered at a tandem node, no calling name display information is sent in the call SETUP message for the new call over the alternate route. However, the calling name display information is sent from an originating node or at an MCDN-to-QSIG gateway node, if it is available.

QSIG Path Replacement

QSIG Alternate Routing cannot be applied to a QSIG Path Replacement call set-up. Therefore, QSIG Path Replacement is rejected if congestion is encountered.

Virtual Network Services

QSIG Alternate Routing can function over all QSIG-based bearer interfaces supporting Virtual Network Services.

MCDN End to End Transparency

When QSIG Alternate Routing is triggered at a tandem node, the Facility IE messages containing the Network Attendant Services (NAS) and Network Attendant ACD (NACD) information are not included in the SETUP message for the new call over the alternate route.

Feature packaging

The following packages are required for the QSIG Alternate Routing feature:

- Basic Routing (BRTE) package 14
- Network Class of Service (NCOS) package 32

One of the following:

- Coordinated Dialing Plan (CDP) package 59, or
- Network Alternate Route Selection (NARS) package 58, or
- Basic Alternate Route Selection (BARS) package 57

Feature operation

No operating procedures are required to use this feature.

ISDN QSIG/EuroISDN Call Completion

The ISDN QSIG/EuroISDN Call Completion feature expands Ring Again functionalities on QSIG and EuroISDN interfaces. This feature provides Call Completion to Busy Subscriber (CCBS) and Call Completion on No Response (CCNR) supplementary services.

The Call Completion to Busy Subscriber (CCBS) supplementary service allows the calling party to apply a Ring Again request when encountering a busy Directory Number (DN). The Meridian 1 alerts the calling party when occupied DN is available to receive a call. The calling party has the option of completing this call without making a new call attempt. CCBS is supported on QSIG and EuroISDN signalling protocols.

Call Completion on No Response (CCNR) supplementary service allows the calling party to apply a Ring Again request to an unanswered DN. With this service, the Meridian 1 alerts the calling party when the dialed DN becomes idle after a period of activity. The calling party has the option of completing the call without making a new call attempt. CCNR is only supported on QSIG signalling protocols.

Operating parameters

ISDN QSIG/EuroISDN Call Completion supports Uniform Dialing, Customer Dialing and Group Dialing plans.

Since the Ring Again feature is configured on set level basis, either by Class of Service or Ring Again key, a set cannot be programmed with CCBS on EuroISDN denied and Meridian Customer Defined Network (MCDN) Network Ring Again allowed. Ring Again is allowed or denied locally or network wide in the same manner.

ISDN QSIG/EuroISDN is supported in ETSI compliant countries.

This feature uses QSIG GF and ETSI GF platforms. With the development of the QSIG Generic Transport (GF) feature, the Call Completion on No Response (CCNR) and Call Completion to Busy Subscriber service on QSIG cannot be involved in any gateway.

Call Completion to Busy Subscriber and Call Completion on No Response are not supported on ISDN BRI sets.

Gateways between EuroISDN CCBS, MCDN Network Ring Again on Busy and DPNSS1 Call Back When Free are supported.

A PINX (Private Integrated Services Network Exchange) DN must be configured in the Customer Data Block (LD 15). The PINX DN is used for routing free notification on the MCDN or DPNSS1 if the EuroISDN network does not provide a calling number for the service. The DN configured should be consistent with the type of number plan used. PINX DN is also used for incoming Direct Inward Dialing (DID) calls in the same manner as Listed Directory Number 0 (LDN0) is used for a basic call.

Existing limitations, applicable to standalone or network wide Ring Again operation, apply to CCNR and CCBS on QSIG and CCBS on QSIG and EuroISDN.

Feature interactions

Access Restrictions

Call Restriction

Trunk Group Access Restrictions

ISDN QSIG/EuroISDN Call Completion does not override Access, Call Restriction or Trunk Group Access restrictions. When Call Completion is activated, the second call has the same restrictions as the initial call that received either no answer or a busy indication.

Advice of Charge for EuroISDN

Advice of Charge applies to the initial and call completion (second call). The initial call receives the same charging information as a normal busy call. The call completion receives the same charging information as the first call.

Attendant Consoles

Call Completion requests cannot be directed to or from an Attendant Console.

Automatic Call Distribution

Call Completion requests cannot be directed to or from Automatic Call Distribution (ACD) Directory Numbers (DNs).

An ACD set that uses a normal DN key (not the ACD key) can activate the Ring Again key when encountering a busy or no answer situation.

Call Detail Recording

A Call Detail Recording (CDR) is not produced for Call Completion signaling. The second call receives a CDR as a normal call.

Call Forward All Calls

When the Call Forward feature is activated on a local basis and an incoming Call Completion request is received, the Call Completion request is registered against the forwarded DN.

Call Transfer

Call Completion notification is only presented to the Call Completion originating set. This notification cannot be transferred to another station. Once the second call is completed, the call can be transferred.

If a user encounters a busy or no answer situation during a transfer operation, Call Completion can be activated.

Call Waiting

On an Analog (500/2500 type) set, Call Completion notification waits until the set has finished an active call. If Call Waiting is configured on a set, notification is presented after the Call Waiting call. If an additional call is queued while Ring Again free notification is waiting on a set, the waiting call takes precedence over the Call Completion notification. An established Call Completion call is also queued if a set has Call Waiting feature equipped and is occupied on another call.

Conference

A Call Completion request cannot be made on a conference call attempt.

Direct Inward System Access

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.

Do Not Disturb

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.

Flexible Feature Codes

Analog (500/2500 type) set can use Flexible Feature Codes (FFCs) to activate Call Completion to Busy Subscriber requests.

Group Call

Call Completion cannot be applied to a Group Call.

Group Hunt

Call Completion to Busy Subscriber cannot be applied to Pilot DN when no idle set is located during a Group Hunt call.

Hot Line

Call Completion cannot be used in conjunction with the Hot Line feature.

Maintenance Busy

Call Completion on Busy Subscriber is not accepted against a set in Maintenance Busy state.

Make Set Busy

Sets that have Make Set Busy (MSB) activated can request Call Completion to another DN, as the free notification overrides the MSB feature. Incoming Call Completion to Busy Subscriber (CCBS) requests do not override the MSB feature. A set is considered busy while MSB is active. A CCBS request is registered against a busy set, but only advances when the MSB feature is deactivated and the set remains free.

Permanent Hold

Analog (500/2500 type) sets with Permanent Hold cannot use the Ring Again functionalities.

Ring Again

Ring Again No Answer

Analog (500/2500 type) sets can have only one Call Completion to Busy Subscriber request at a given time. Meridian 1 proprietary sets can make Ring Again requests based on the number of Ring Again keys programmed on a set.

Feature packaging

ISDN QSIG/EuroISDN Call Completion (QSIG-SS) is package 316. The package dependencies for QSIG-SS are:

- Optional Features (OPTF) package 1; and
- ISDN QSIG GF Transport (QSIGGF) package 305.

Depending on the application, the following packages are also required.

For the QSIG ISDN PRI interface:

- Digit Display (DDSP) package 19;
- Coordinated Dialing Plan (CDP) package 59;
- Integrated Services Digital Network (ISDN) package 145;
- 2.0 Mbit Primary Rate Access (PRI2) package 154;
- Flexible Numbering Plan (FNP) package 160;
- Overlap Signaling (OVLP) package 184 (optional);
- International Primary Rate Access (IPRA) package 202;
- Multipurpose Serial Data Link (MSDL) package 222; and
- QSIG Interface (QSIG) package 263.

For the QSIG ISDN BRI interface:

- Digit Display (DDSP) package 19;
- Integrated Services Digital Network (ISDN) package 145;
- Overlap Signaling (OVLP) package 184 (optional);
- Basic Rate Interface (BRI) package 216;
- ISDN BRI Trunk Application (BRIT) package 233; and
- QSIG Interface (QSIG) package 263.

The EuroISDN Call Completion Supplementary Service (ETSI-SS) is package 323.

For the EuroISDN ISDN PRI interface:

- Digit Display (DDSP) package 19;
- Coordinated Dialing Plan (CDP) package 59;
- Integrated Services Digital Network (ISDN) package 145;
- 2.0 Mbit Primary Rate Access (PRI2) package 154;
- Flexible Numbering Plan (FNP) package 160;
- ISDN Supplementary Features (ISDN INTL SUPP) package 161;
- Overlap Signaling (OVLP) package 184 (optional);
- International Primary Rate Access (IPRA) package 202;
- Multipurpose Serial Data Link (MSDL) package 222; and
- EuroISDN (EURO) package 261.

For the EuroISDN ISDN BRI interface:

- Digit Display (DDSP) package 19;
- Integrated Services Digital Network (ISDN) package 145;
- ISDN Supplementary Features (ISDN INTL SUPP) package 161;
- Overlap Signaling (OVLP) package 184 (optional);
- Basic Rate Interface (BRI) package 216;
- ISDN BRI Trunk Application (BRIT) package 233; and
- EuroISDN (EURO) package 261.

Feature operation

Call Completion on Busy Subscriber

Meridian 1 Proprietary Sets

- 1 Set A calls Set B on an outgoing QSIG/EuroISDN route and receives a busy indication.
- 2 Before releasing this call, Set A presses the Ring Again key (RGA) to activate Ring Again functionality. The DN lamp flashes.
- 3 When the far end receives a Call Completion on Busy Subscriber (CCBS) request, Set A's RGA lamp key is steadily lit, and the DN lamp darkens.
- 4 Set A awaits notification that Set B is free. Set A can make or receive calls while waiting for notification.
- 5 When Set B goes on-hook, Set A is notified in the form of a one second burst of ringing and the RGA key lamp fast flashes.
- 6 The recall is accepted by selecting the DN key and pressing the RGA key. Set A has thirty seconds to accept the recall before it is cancelled.
- 7 If Set A accepts the recall and Set B is free, the call is rung. However, if Set B has either received or originated another call, Set A receives a busy indication. Set A has the option of requesting CCBS again.

Note: If Set A presses the RGA key before receiving notification then the request is cancelled.

Analog (500/2500 type) sets without RGA key

- 1 Set A calls Set B on an outgoing QSIG/EuroISDN route and receives a busy indication.
- 2 Set A performs a switch-hook flash and a special dial tone is heard.
- 3 Set A dials RGA Activate Flexible Feature Code (FFC) or SPRE +1 to activate CCBS feature. This request cancels and replaces previous RGA Activate requests.
- 4 If an FFC is dialed, an FFC confirmation tone if programmed indicates that the CCBS request is registered.
- 5 Set A goes on-hook and is free to make or receive other calls.

- 6 When Set B goes on-hook, Set A receives a burst of tone and has thirty seconds to accept the recall before it is cancelled.
- 7 If Set B has originated or received another call, Set A receives a busy indication and has the option of requesting the CCBS again.

Note: If Set A presses the RGA key before receiving notification then the request is cancelled.

Call Completion No Response

Meridian 1 Proprietary Sets

- 1 Set A calls Set B on an outgoing QSIG route. The call rings but is not answered. Set A receives a ringback tone.
- 2 Set A presses RGA key to activate Ring Again on No Answer prior to releasing the call. The DN key lamp darkens.
- 3 Set A goes on-hook and is free to make or receive calls.
- 4 When Set B makes a call and goes on-hook. Set A is notified by a one second burst of tone and the RGA lamp fast flashes.
- 5 Set A can accept the call by selecting a DN key and pressing the RGA key. Set A has thirty seconds to accept this call.
- 6 If Set A accepts the call and Set B is free, the call goes through. However, if Set B receives or originates another call, Set A receives a busy indication. Set A has the option of requesting CCBS.

Note: If Set A presses the RGA key before receiving notification then the request is cancelled.

ISDN QSIG/EuroISDN Call Completion Enhancement

The ISDN QSIG/EuroISDN Call Completion was introduced in Release 22, to provide Call Completion to a Busy Subscriber (CCBS) over QSIG and EuroISDN networks, and Call Completion on No Response (CCNR) over QSIG networks.

- **Call Completion to a Busy Subscriber (CCBS).** This service allows a calling party to apply a Ring Again request to a called party that is busy. When the busy party becomes free, the calling party receives notification and can complete the call without making a new call attempt.
- **Call Completion on No Response (CCNR).** This service allows a calling party to apply a Ring Again request to a called party that is idle, and does not answer. The calling party receives notification after the Meridian 1 notices subsequent activity on the called extension, that has again become idle.

With Release 24, the QSIG/EuroISDN Call Completion Enhancement feature introduces the following functionalities:

- allows the method of Operation Coding for the CCBS and CCNR Supplementary Services to be configured by Object Identifier, used for European Telecommunications Standard Institute (ETSI) interfaces, or by Integer Value, used for International Standards Organization (ISO) interfaces. The Operation Coding method is configured as a remote capability using the RCAP prompt for an ISDN PRI D-Channel for ISDN PRI, or for an ISDN BRI route for ISDN BRI trunking.
- supports all ETSI/QSIG to MCDN and DPNSS1 gateways.
- allows all EuroISDN countries to fully support the QSIG/EuroISDN Call Completion services.

Operating parameters

Call Completion to a Busy Subscriber (CCBS) is available only on EuroISDN interfaces supporting the ETSI QSIG Generic Functional Protocol Transport.

Call Completion to a Busy Subscriber (CCBS) is not supported on ISDN BRI sets.

The QSIG/EuroISDN Call Completion Enhancement feature cannot use route access codes. The feature can only use a Coordinated Dialing Plan or Uniform Dialing Plan. Also, this feature does not support digit manipulation tables.

Since Ring Again is configured on a per-set basis, using Class of Service for analog (500/2500) sets or defining a Ring Again key on Meridian proprietary sets, it is not possible to configure a set so that Call Completion to a Busy Subscriber is denied for EuroISDN interfaces, and MCDN Network Ring Again is allowed.

Call completion on QSIG uses two signaling types, **Connection Retention**, where the signaling connection remains established until call completion is achieved, or the call completion attempt is cancelled, and **Connection Release**, where the signaling connection is cleared after each phase of call signaling, and a new signaling connection is established for each subsequent phase of call independent signaling. At the terminating side of the call connection, both signaling methods are mandatory, and are supported by the Meridian 1. At the originating side, although both methods are possible, the Meridian 1 supports only the Connection Retention type, since it matches the EuroISDN signaling.

A call completion recall over a QSIG interface uses two methods of establishing the recall, **Path Reservation**, where a bearer connection between the originating and terminating sides is established before presenting the recall to the originating party, in order to avoid network congestion, and **Non-path Reservation**, where a bearer connection between the originating and terminating sides is established after the originating party answers the recall, and the recall attempt is cancelled if network congestion is encountered. At the terminating side of the call connection, both methods of recall are mandatory, and are supported by the Meridian 1. At the originating side, although both methods are possible, the Meridian 1 supports only the Non-path Reservation method, since it is used by MCDN and EuroISDN. Also, the Path Reservation method is not supported over QSIG to MCDN or DPNSS1 gateways.

For a QSIG or EuroISDN call completion attempt, if the called party is found to be again busy after the originating party responds to a call completion recall, there are two methods of behavior that the Meridian 1 may take. With the **Request Retention** method, the call completion request remains in place at both the originating and terminating side. The terminating side begins monitoring the terminating party. With the **Service Cancellation** method, the call completion request is cancelled at both the originating and terminating sides. The Request Retention method is not supported on the Meridian 1.

Busy name information sent with a DISCONNECT message is only supported for Overlap Receiving (the OVLR prompt is set to YES in LD 17).

Feature interactions

Access Restrictions/trunk Group Access Restrictions

QSIG/EuroISDN Call Completion cannot override access restrictions placed on a call. The second call due to either CCBS or CCNR is under the same access restrictions as the first call.

Advice of Charge for EuroISDN

The Advice of Charge for EuroISDN applies to both the first call and second call due to CCNR. The second call receives the same call charging information as the first call (that is, as a normal call).

Attendant Console

A CCBS or CCNR cannot originate from, or terminate to, an attendant console.

Automatic Call Distribution

A CCBS or CCNR cannot originate from, or terminate to, an ACD Directory Number.

Call Connection Restriction

QSIG/EuroISDN Call Completion cannot override call connection restrictions placed on a call. The second call due to either CCBS or CCNR is under the same call connection restrictions as the first call.

Call Detail Recording

There is no CDR information created for a CCBS attempt. A new call that is established as part of CCBS is subject to CDR, as per a normal call.

Call Forward

When a CCBS or CCNR request is received on a set which has been call forwarded, the CCBS or CCNR request is registered against the DN to which the set has been forwarded.

Call Transfer

Call Completion free notification can only be presented to the originating station, and cannot be transferred to another station. However, once the call has been established, it behaves as a normal call, with respect to call transfer.

During a call transfer operation, a caller that encounters a busy or no answer situation, can still activate the call transfer.

A call cannot be completed that is being transferred to a busy destination, as part of the EuroISDN Continuation feature, or to a no answer destination.

Call Waiting

A Call Completion free notification is presented to an analog (500/2500) set after an active call has been completed. If there is a call waiting on the set, the Call Completion free notification is presented to the set after the call waiting call has been completed. If another call waiting occurs while a Ring Again free notification is waiting on the set, the new call waiting presentation takes precedence.

A Call Completion call may wait to be established on a set that is busy on another call, and has the Call Waiting feature activated.

Conference

A Call Completion request cannot be made as part of a conference call attempt.

Customer Controlled Routing

A Call Completion attempt request be applied to a Customer Controlled Routing DN.

Direct Inward System Access

CCBS cannot be applied to DISA calls when the terminating party is busy. CCNR cannot be applied to DISA calls when the terminating party does not answer.

Directory Number - Multiple Appearance Directory Number Redirection Prime

MARP does not apply to a Call Completion attempt.

Do Not Disturb

Stations with the Do Not Disturb feature active may make Call Completion requests against other stations, subject to the same restrictions as if the feature were not active. The incoming Call Completion free notification overrides the Do Not Disturb state.

Flexible Numbering Plan

A Flexible Numbering Plan may be used with QSIG/EuroISDN Call Completion, as long as the numbering plan is consistent over the network.

Group Call

A Call Completion request cannot be applied to a group call.

Group Hunt

CCBS cannot be applied to a PLDN when no idle set is found during a group hunt call.

Hot Line

A Call Completion request cannot be used in conjunction with the Hot Line or Enhanced Hot Line feature.

Initialize

If a system initialize occurs at a Meridian 1, all Call Completion requests at the Meridian 1 are deleted. If the requesting party is not on the Meridian 1 which has initialized, it is not notified that a Call Completion request has been cancelled.

Maintenance Busy

A Call Completion request cannot be made against a set in maintenance busy state.

Make Set Busy

If a set with Make Set Busy active makes a Call Completion request to another station, then the Call Completion free notification overrides the Make Set Busy state. Incoming CCBS requests, however, do not override the Make Set Busy state. A CCBS request can only be completed once Make Set Busy is deactivated.

Permanent Hold

Analog (500/2500) sets with a call on permanent hold cannot use the Ring Again feature.

Pretranslation

Pretranslation may be used in conjunction with a Call Completion request.

Ring Again/Ring Again No Answer

Analog (500/2500) sets can only have one Call Completion request active at a time. Meridian proprietary sets may make as many Call Completion requests as available keys to support the requests.

Ring Again Inhibition

Ring Again cannot be activated if a limit of 12.5% of all available call registers in the Ring Again queues is exceeded.

DPNSS1 Call Back When Free and Next Used

In a QSIG/EuroISDN to DPNSS1 gateway, a caller on one node can activate Ring Again to any other destination party at another node, using a combination of QSIG/EuroISDN and DPNSS1 links.

Electronic Switched Network

QSIG/EuroISDN Call Completion uses a Uniform Dialing Plan or Coordinated Dialing Plan Numbering Plan.

The use of the Connection Retention procedure (refer to the relevant discussion on page 360, as part of the 'Operating parameters' section) on QSIG does not require that the Numbering Plan be symmetrical (although this approach is not recommended). The called destination, which is being monitored as part of CCBS or CCNR, does not require notification as how to connect to the originating party. This applies only if the Connection Retention procedure is used. If the Connection Release procedure is used, the numbering plan has to be symmetrical, so that the destination party can use the CLID of the originating party, to send the Call Connection free notification.

Hunting and Group Hunting

Hunting and Group Hunting is not supported by the QSIG/EuroISDN Call Completion feature.

ISDN Basic Rate Interface

QSIG/EuroISDN Call Completion is not supported on ISDN BRI sets. CCBS and CCNR cannot neither be activated or received on a ISDN BRI set.

Network Call Redirection - EuroISDN

If a call over a EuroISDN interface has not been locally call forwarded, then a Call Completion request is ignored.

Network Call Redirection - QSIG Diversion***Call Completion to Busy Subscriber***

If CCBS is invoked on a busy destination that has been diverted over a QSIG interface, the originating Private Integrated Services Network Exchange (PINX) stores the diverted-to number issued by the served PINX, to use it for CCBS call treatment. This diverted-to number is used as received, without any digit manipulation.

Call Forward Unconditional and Call Forward Busy take precedence over CCBS.

Call Completion on No Reply

If CCNR is invoked on a destination that does not answer, and that has been diverted over a QSIG interface, the originating Private Integrated Services Network Exchange (PINX) stores the diverted-to number issued by the served PINX, to use it for CCNR call treatment. This diverted-to number is used as received, without any digit manipulation.

Call Forward No Reply takes precedence over CCNR.

Feature packaging

The following packages are required for the **QSIG** component of the QSIG/EuroISDN Call Completion Enhancement feature:

For ISDN PRI2:

- QSIG Supplementary Services (QSIG-SS) package 316
- QSIG Generic Functional Protocol Transport (QSIG-GF) package 305
- Q-reference Signaling Point Interface (QSIG) package 263
- Multi-purpose Serial Data Link (MSDL) package 222
- International Primary Rate Access (IPRA) package 202
- Overlap Signaling (OVLP) package 184 (optional)
- Flexible Numbering Plan (FNP) package 160
- 2.0 Mbps Primary Rate Access (PRI2) package 154
- Integrated Services Digital Network (ISDN) package 145
- Coordinated Dialing Plan (CDP) package 59, and one of the following:
 - Network Alternate Route Selection (NARS) package 58, **or**
 - Basic Alternate Route Selection (BARS) package 57
- Digit Display (DDSP) package 19
- Ring Again (RGA) package 1

For ISDN BRI trunking:

- Q-reference Signaling Point Interface (QSIG) package 263

- ISDN Basic Rate Trunk Access (BRIT) package 233
- ISDN Basic rate Access (BRI) package 216
- Integrated Services Digital Network (ISDN) package 145
- Overlap Signaling (OVLP) package 184 (optional)
- Digit Display (DDSP) package 19

The following packages are required for the **EuroISDN** component of the QSIG/EuroISDN Call Completion Enhancement feature:

For ISDN PRI2:

- ETSI Supplementary Services (ETSI-SS) package 323
- Multi-purpose Serial Data Link (MSDL) package 222
- International Primary Rate Access (IPRA) package 202
- Overlap Signaling (OVLP) package 184 (optional)
- ISDN International Supplementary Features (ISDN SUPP) package 161
- Flexible Numbering Plan (FNP) package 160
- Network Attendant Service (NAS) package 159 (for MCDN to EuroISDN gateways)
- 2.0 Mbps Primary Rate Access (PRI2) package 154
- International Primary Rate Access (IPRA) package 202
- Integrated Services Digital Network (ISDN) package 145
- Coordinated Dialing Plan (CDP) package 59, and one of the following:
- Network Alternate Route Selection (NARS) package 58, **or**
- Basic Alternate Route Selection (BARS) package 57
- Digit Display (DDSP) package 19
- Ring Again (RGA) package 1

For ISDN BRI trunking:

- ETSI Supplementary Services (ETSI-SS) package 323
- ISDN Basic Rate Trunk Access (BRIT) package 233

- ISDN Basic rate Access (BRI) package 216
- International Primary Rate Access (IPRA) package 202
- Overlap Signaling (OVLP) package 184 (optional)
- ISDN International Supplementary Features (ISDN SUPP) package 161
- Integrated Services Digital Network (ISDN) package 145
- Digit Display (DDSP) package 19

Feature operation

No operating procedures are required to use this feature.

ISDN QSIG Call Transfer Notification

The QSIG Call Transfer Notification feature supports the Call Transfer Notification capability over QSIG private networks, and over mixed QSIG/MCDN and QSIG/DPNSS1 networks. This applies to 2.0 Mbps Primary Rate Interface (PRI2), and ISDN Basic Rate Trunking (BRIT) applications.

The QSIG Call Transfer supplementary service, introduced in Release 20, provided the capability of using the Transfer by Call Join function to join two separate QSIG calls, resulting from a call transfer activation, into one call between the originating party and the transferred-to party. However, there was no notification, via protocol message exchange between the originating and transferring nodes, that the transfer had taken place.

With the introduction of the QSIG Call Transfer Notification feature in Release 24, whenever a call transfer occurs over a QSIG link, or over a QSIG/MCDN or QSIG/DPNSS1 gateway, a notification of the transfer is sent to the originating and transferred-to party. The information, conveyed in call transfer messages, includes the redirection and originating number, redirection and originating name, and the redirection reason (transfer). This information is displayed on the sets of originating and terminating party. It is also captured in Call Detail Recording records.

The method of Operation Coding for the Call Transfer Notification feature is configured by Object Identifier, used for European Telecommunications Standard Institute (ETSI) interfaces, or by Integer Value, used for International Standards Organization (ISO) interfaces. The Operation Coding method is configured as a Remote Capability using the RCAP prompt in LD 16 and LD 17).

QSIG Call Transfer Notification at work

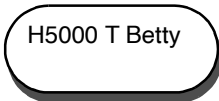
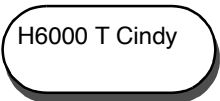
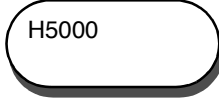
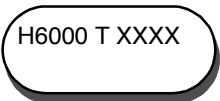
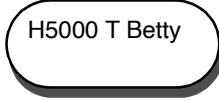
Consider the following application of the QSIG Call Transfer Notification feature being applied to a call within a QSIG Meridian 1 network (the network signaling occurs over QSIG links).

Station A (Betty, DN 5000) calls Station B (Anne, DN 4000), over a QSIG link. Station B transfers to Station C (Cindy, DN 6000), also over a QSIG link. Table 28 summarizes the display options pertaining to the originating set and transferred-to set.

Note: The underlined Class of Service values are in effect.

Table 28
Originating set and transferred-to set display possibilities

Originating set Class of Service	Originating set display	Transferred-to set Class of Service	Transferred-to set display
	Originating set display after receipt of transfer notification from transferred-to set		Transferred-to set display after receipt of transfer notification from originating set
CNDA NAMA DDGA	H6000 T Cindy	CNDA NAMA DDGA	H5000 T Betty
CNDA NAMA <u>DDGD</u>	H6000 T Cindy	CNDA NAMA DDGA	211-4 T Betty
<u>CNDD</u> NAMA DDGA	H6000	CNDA NAMA DDGA	H5000 T Betty
CNDA <u>NAMD</u> DDGA	H6000 T Cindy	CNDA NAMA DDGA	H5000 T XXXX

CNDA NAMA DDGA		CNDA NAMA <u>DDGD</u>	
CNDA NAMA DDGA		<u>CNDD</u> NAMA DDGA	
CNDA NAMA DDGA		CNDA <u>NAMD</u> DDGA	

Note 1: No reason is displayed if the Class Of Service CNDD is configured on the set

Note 2: When presentation of the remote party number is not allowed (CLS DDGD on the remote party), the other party displays the following instead of the DN:

- The trunk access code and the route member, if a trunk is involved, that is, transfer to an other node; or
- Several dashes, if this party is on the transferring node, that is, local transfer to a node.

Operating parameters

The Call Transfer by rerouting method is not implemented.

The subAddressTransfer invoke APDU is not generated by Meridian1:

The maximum allowed size of the name information is 50 characters for QSIG. However, a maximum of 27 characters is allowed with the Calling Party Name Display (CPND) feature. Therefore, name display information might be truncated.

The call transfer notification is sent whatever the numbering plan is. However, the redirection number that is sent has any meaning only if either the Coordinated Dialing Plan (CDP) or Uniform Dialing Plan (UDP) is used to build the redirection number.

Digit manipulation can not be applied to the redirection number sent in the Call Transfer messages.

The Call Transfer Notification for a call extended by an attendant in an MCDN network is provided by the NAS feature. The Call Transfer QSIG/MCDN gateway does not map this NAS information into QSIG messages.

Feature interactions

Calling Line Identification Presentation/Restriction

CLIP/CLIR applies to the originating and to the transferred-to set display.

Connected Line Identification Presentation/Restriction

COLP/COLR applies to the originating and to the transferred-to set display.

Calling Name Identification Presentation/Restriction (CNIP/CNIR)

The Calling, Alerting or Connected name are not included in the ctComplete invoke APDU, at the transferring PINX.

Connected Name Identification Presentation/Restriction (CONP/CONR)

If a name, either from QSIG Name Display Service or QSIG Call Transfer Notification, is received, it will be displayed, if no presentation option forbids it.

On-Hold

The interaction with On-Hold is handled with the same manner as between MCDN Call Transfer Display and On-Hold.

Symposium Call Center Server

If a QSIG call is transferred to a Symposium Call Center DN, the call type is updated in the ICC message for QSIG Transfer.

Meridian Mail

A QSIG call transferred to Meridian Mail may access the different Meridian Mail functionalities (message reception, mailbox interrogation).

QSIG Name Display Supplementary Services (QNDS)***Calling, Alerting & Connected Name***

The Calling, Alerting or Connected name is not included in the ctComplete invoke APDU, at the transferring PINX.

Connected Name

If a name, either from QSIG Name Display Service and QSIG Call Transfer Notification, is received, it will be displayed only if no presentation option forbids it.

QSIG Diversion***Call Forwarding Unconditional/Call Forwarding Busy***

When the call transfer is completed, the diverted-to set displays the Call Transfer primary user DN and the transfer notification reason.

Call Forwarding No Reply

Call Transfer interacts with Call Forward No Reply (CFNR), if CFNR is invoked for an unanswered, transferred call.

Either the Transferring PINX acts as Rerouting PINX (in case of call transfer by join and call forwarding by rerouting, or the Secondary PINX acts as Rerouting PINX (call forwarding by forward switching).

QSIG Path Replacement (QPR)

QSIG Call Transfer as trigger for QPR

QSIG Call Transfer can be configured as a trigger for QSIG Path Replacement. If configured, after the completion of the transfer, QSIG Path Replacement starts to obtain a more efficient connection, if necessary.

Feature packaging

The following packages are required for the QSIG Call Transfer Notification feature:

- Q-reference Signaling Point Interface Supplementary Services (QSIG-SS) package 263
- QSIG Generic Functional Protocol Transport (QSIG-GF) package 305
- QSIG Supplementary Services (QSIG-SS) package 316
- Q-reference Signaling Point Interface (QSIG) package 263
- Multi-purpose Serial Data Link (MSDL) package 222
- International Primary Rate Access (IPRA) package 202
- 2.0 Mbps Primary Rate Access (PRI2) package 154
- Integrated Services Digital Network (ISDN) package 145

For ISDN Basic Rate Interface Trunking (BRIT), the following additional packages are required:

- ISDN Basic rate Access (BRI) 216
- ISDN Basic Rate Trunk Access (BRIT) package 233

For a QSIG/DPNSS1 Gateway, the following additional packages are required:

- DPNSS Network Services (DNWK) package 231
- Digital Private Networking Signaling System No. 1 (DPNSS) 123
- Integrated Digital Access (IDA) package 122

For notification display, the following additional packages are required:

- Call Party Name Display (CPND) package 95

- Digit Display (DDSP) 19

Feature operation

No specific operating procedures are required to use this feature.

ISDN QSIG Call Diversion Notification

ISDN QSIG Call Diversion Notification allows the Meridian 1 to provide notification to the originating calling party and forwarded-to (diverted) party when a call is forwarded over an ISDN QSIG private network. With this feature, call diversion is invoked by the following triggers:

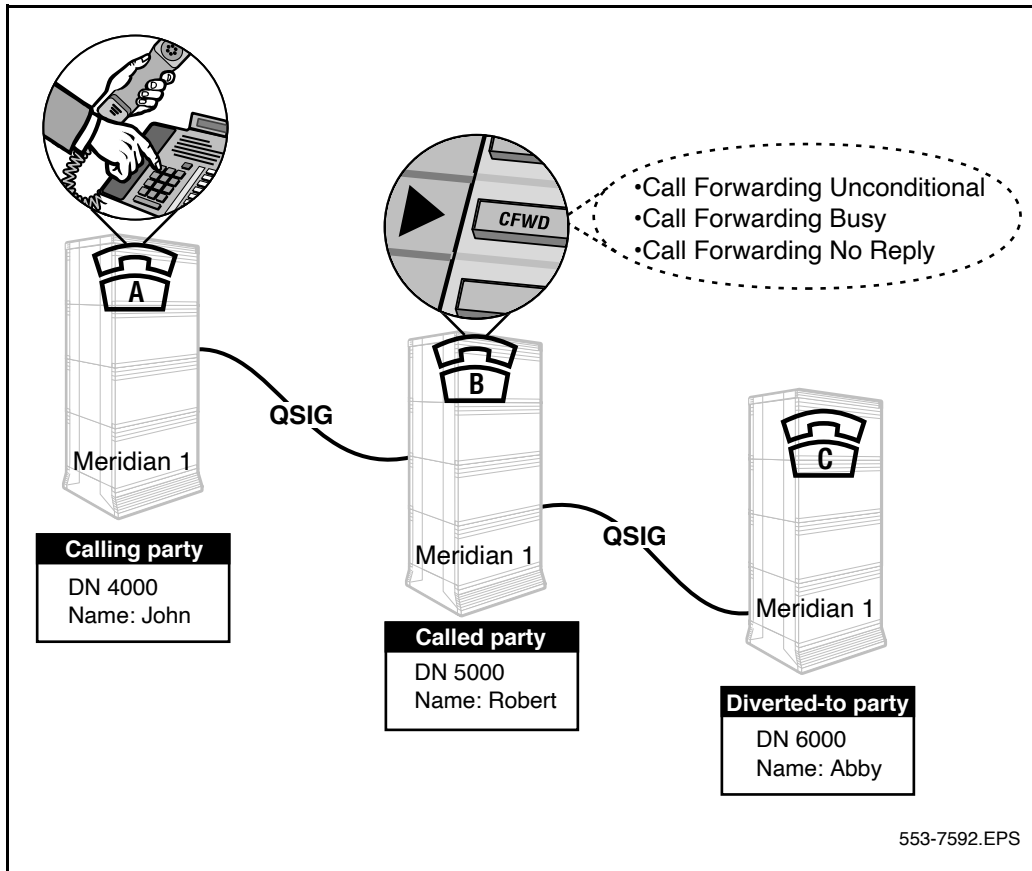
- Call Forwarding Busy (CFB) - the called set is in use
- Call Forwarding No Reply (CFNR) - the called set does not answer after a given number of rings
- Call Forwarding Unconditional (CFU) - the called set is forwarded as soon as the set is reached

With this feature, notification can be provided to the calling party and the forwarded-to (diverted) party that call diversion has occurred. When provisioned, call diversion notification is displayed on the calling party's set and on the forwarded-to (diverted) party's set according to the existing terminal display configuration.

Diversion notification provided to the calling and forwarded-to (diverted) party depends on the subscription options (Class of Service) configured on the called party's set. The called party can configure two diversion options. One Class of Service option is applicable to the calling party and the other option is applicable to the forwarded-to (diverted) party. These new Classes of Service determine if call diversion notification is provided to the calling party and the forwarded-to party and what type of diversion information these parties receive.

Figure 27 shows an example of a Call Diversion environment.

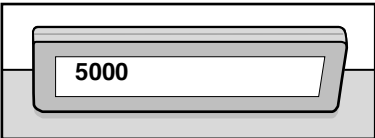
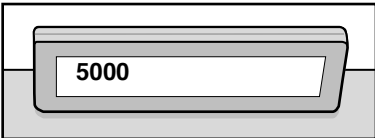
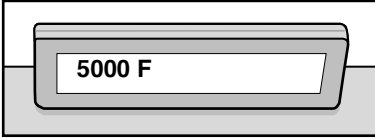
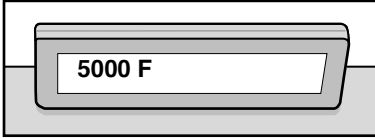
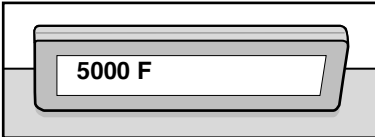
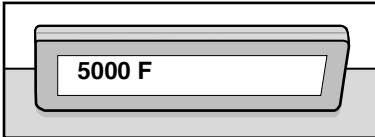
Figure 27
Call Diversion Environment



Calling Party's Notification

The calling party that originates the call receives notification of call diversion depending on the configuration of the called party's set. As shown in Table 29, the called party configures one of the three possible Classes of Service.

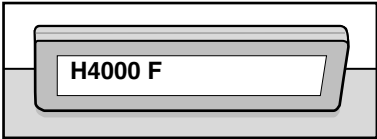
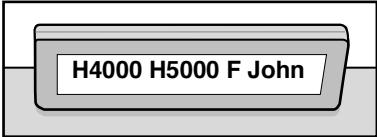
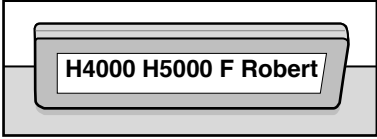
Table 29
Calling Party's Notification

Called Party's Diversion Subscription Options (Class of Service)	Calling Party's Display when call is forwarded.	Forwarded-to Party's Display
CLS = DNO1 No diversion notification provided to calling party.		
CLS = DNO2 Diversion notification provided without forwarded-to (diverted) party's number and name.		
CLS = DNO3 (default) Diversion notification provided with forwarded-to (diverted) party's number and name when available.		

Called Party's Notification

The forwarded-to (diverted) party receives notification of call diversion depending on the configuration of the called party's set. As shown in Table 30, the called party configures one of the two possible Classes of Service.

Table 30
Forwarded-to Party's Notification

Called Party's Diversion Subscription Options (Class of Service)	Forwarded-to Party's Display.	Forwarded-to Party's Class of Service
DNDN = No Called party's number and name are not released to the forwarded-to (diverted) party.		DNDD = Dialed Name Display Denied (default)
DNDY = Yes (default) Called party's number and name are released to forwarded-to (diverted) party.		DNDD = Dialed Name Display Denied (default)
DNDY = Yes (default) Called party's number and name are released to forwarded-to (diverted) party.		DNDA = Dialed Name Display Allowed

On the forwarded-to (diverted) party's set, the Dialed Name Display Allowed/Denied (DNDA/DNDD) Class of Service allows the forwarded-to (diverted) party to choose the name displayed after call diversion. If the forwarded-to (diverted) party configures a Dialed Name Display Allowed (DNDA) Class of Service their display shows one of the following:

- the original called party's name when available; or
- if the original called party's name is not available, the redirecting name when available; or
- if neither of the two previous names are available, then nothing is displayed.

If the forwarded-to (diverted) user has configured a Dialed Name Display Denied (DNDD) Class of Service then the calling party's name is displayed if available. Otherwise, nothing is displayed.

Operating parameters

A Meridian 1 user cannot remotely activate or deactivate QSIG Call Diversion Notification from either a Meridian 1 or on another third party Private Branch Exchange.

As per the current operation of the Call Forward features, the Meridian 1 does not provide any indication, such as an LED, on the called party's set that call diversion has been activated.

The validity of the forwarding number is not supported. If an invalid forwarding directory number is dialed no verification is provided by the system.

The calling party is notified each time a call redirection occurs, provided this information is supplied by the network. If previous call diversion notification is provided, then it is replaced by last received. If call redirection occurs and no call redirection information is provided by the network, then any notification information that has previously been provided remains unchanged.

When a call redirection occurs no call diversion information is provided by the network. However, if a previous notification has been provided then the call diversion information remains unchanged.

The maximum allowable size of QSIG name information is 50 characters. With the Call Party Name Display feature, only 27 characters string length is supported. Accordingly, names processed by the Meridian 1 may be truncated.

Across a network, it is not possible to prevent reciprocal Call Forward from occurring. However, the capabilities of the Flexible Orbiting Prevention Timer and Total Redirection Count features can be used.

QSIG Call Diversion Notification requires Coordinated Dialing Plan (CDP) or Uniform Dialing Plan (UDP). Digit manipulation is not supported.

Interworkings with Meridian Customer Defined Network (MCDN) call redirection and QSIG Call Diversion Notification is not supported.

Interaction with Meridian Mail and this feature is not supported.

Feature packaging

QSIG Call Diversion Notification is ISDN QSIG Supplementary Service - Call Completion (QSIG-SS) package 316 which has the following dependencies:

- QSIG Interface (QSIG) package 263
- Calling Party Name Display (CPND) package 95
- Digit Display (DDSP) package 19
- ISDN QSIG-GF Transport (QSIG-GF) package 305
- Network Alternate Route Selection (NARS) package 58 (for Uniform Dialing Plan), or Coordinated Dialing Plan (CDP) package 59

2.0 Mbps Primary Rate Interface (PRI2) QSIG Interface requires:

- 2.0 Mbps Primary Rate Interface (PRI2) package 154
- International Primary Rate Access (IPRA) package 202
- Multipurpose Serial Data Link (MSDL) package 222

ISDN Basic Rate Interface Trunk Access (BRIT) QSIG Interface requires:

- Basic Rate Interface (BRI) package 216
- ISDN Basic Rate Interface Trunk Access (BRIT) package 233

Feature operation

No specific operating procedures are required to use this feature.

ISDN QSIG/ETSI GF Enhancement

The Integrated Services Digital Network ISDN (ISDN) Q-reference Signaling Point (QSIG)/European Telecommunications Standards Institute (ETSI) Generic Functional Transport (GFT) was introduced in Release 22, to provide a set of generic transport mechanisms to support QSIG supplementary services over a Private or Public Integrated Services Network.

It must be noted that the ISDN QSIG/ETSI GF platform by itself does not actually support any features, it only provides the platform to be used by the various QSIG supplementary services.

The ISDN QSIG/ETSI Generic Functional (GF) Transport Enhancement feature has been introduced in Release 24 to provide improvements in call control, to allow the supported QSIG signaling protocols to better work seamlessly together. These improvements comprise the following:

For QSIG GFT:

- **Support of Call-Independent gateways between QSIG, and EuroISDN, Meridian Customer Defined Networking MCDN, and Digital Private Networking Signaling No. 1 (DPNSS1).**

The QSIG Cal-Independent Generic Functional Transport - Control capability has been modified, so that at these gateways, the CONNECT message of a QSIG supplementary service is not systematically sent after the receipt of a QSIG SETUP message. Instead, the control of the call processing, such as controlling the expiration of the T303 and T310 timers, is left to the QSIG supplementary service operating over the QSIG GFT.

- Call Related Application Protocol Data Unit (APDU) transport.

Transit Private Integrated Services Network Exchange (PINX) operation:

The Call Related APDU transport has been modified, to ensure that a call that is being optimized during QSIG Path Replacement is using the optimum path (the first available alternate route in a route list index). A transit Private Integrated Services Network Exchange (PINX) may now determine whether or not to intercept a Facility Information Element (IE) containing a Network Facility Extension (NFE) EndPinx message.

This modification also allows a Facility IE carried in a Setup message, that has encountered congestion, to be included in an alternate call, if QSIG Alternate Routing is activated on a transit node.

Calculating the remaining available length for an APDU:

The method of calculating the available length for an APDU in basic QSIG call control messages has been made more accurate. The available length for an APDU is now calculated as the difference between the maximum message length (260 octets) and the exact length of other mandatory and non-mandatory IEs in the message.

For ETSI GFT:

- **Call Related Application Protocol Data Unit (APDU) transport.**

Calculating the remaining available length for an APDU:

The method of calculating the available length for an APDU in basic ETSI call control messages has been made more accurate. The available length for an APDU is now calculated as the difference between the maximum message length (260 octets) and the exact length of other mandatory and non-mandatory IEs in the message.

Operating parameters

The operating parameters that follow are carried over from the ETSI Generic Functional Transport feature that was introduced in Release 22.

The following operating parameters apply to the ISDN ETSI GF component of the feature enhancements:

- The ETSI GF gateways to and from other signaling systems, such as QSIG, Digital Private Networking Signaling No. 1 (DPNSS1), and Meridian Customer Defined Networking (MCDN), are only supported for Call Independent gateways.
- The following ETSI GF functionalities are not supported:
 - the control of supplementary services using the separate message approach (Hold/Retrieve messages)
 - the bearer-related broadcast transport mechanism (multipoint configuration)
 - the bearer-independent point-to-point connections transport mechanism
 - the bearer-independent broadcast transport mechanism
 - the generic notification procedures
 - the network-side channel reservation function
 - the generic procedures for supplementary services management
 - the generic status request procedure
 - the support of the Extended Facility Information Element (FIE)

- During call processing, the translation of a DN address requires that the DN be associated with a customer number. For the processing of an ETSI basic call, this customer number association is found through the B-Channel identified in the Channel Identification Information Element.

For the DN address translation that is not associated with a basic call, the customer number/DN association has to be determined using other methods. For an ISDN BRI trunk Digital Subscriber Loop (DSL) interface, the customer number is associated with a D-Channel. For an ISDN PRI interface, the Private Integrated Services Network Exchange Customer (PINX_CUST) prompt in LD 17 is used to define a customer number to a D-Channel. This implies that bearer-independent messages on an ISDN PRI D-Channel are associated with a single customer, as configured in LD 17.

For example, in a multi-customer configuration, if every customer on a Meridian 1 PBX wishes to use the bearer-independent transport service over an ETSI PRI interface, then each customer requires the use of a separate D-Channel.

- The ISDN ETSI GF platform supports up to eight Remote Operation Service Element (ROSE) components in one message, either in a single Facility Information Element (FIE) or multiple FIEs. If a supplementary service requests that the ISDN ETSI GF Transport send a message exceeding this limit, a notification message is sent to the supplementary service.

The following operating parameters apply to the ISDN QSIG GF component of the feature enhancements:

- The QSIG GF gateways to and from other signaling systems, beyond the Q reference point, such, Digital Private Networking Signaling No. 1 (DPNSS1), EuroISDN, and Meridian Customer Defined Networking (MCDN), are only supported for Call Independent gateways.
- At a QSIG to EuroISDN, DPNSS1, or MCDN gateway, when a call-independent Setup message is received which has a valid gateway destination, and which includes several FIEs that belong to different supplementary services, then the gateway is not supported by this feature.

- The MCDN to QSIG GF “protocol converter” does not actually convert the MCDN features, it only provides the conversion platform. The individual MCDN features have to be ported to the ISDN QSIG GF separately.
- This feature does not implement or enhance any ISDN QSIG supplementary services, or networking applications such as Virtual Network Services (VNS).
- Although the transport of Manufacturer Specific Information (MSI) across a ISDN QSIG interface is used by the MCDN to QSIG GF “protocol converter”, a tandem PINX receiving this information may discard it. The MSI may be passed on to the next node only if the transit PINX is a Meridian 1.
- During call processing, the translation of a DN address requires that the DN be associated with a customer number. For the processing of a QSIG basic call, this customer number association is found through the B-Channel identified in the Channel Identification Information Element. For the DN address translation that is not associated with a basic call, the customer number/DN association has to be determined using other methods.

For an ISDN BRI trunk Digital Subscriber Loop (DSL) interface, the customer number is associated with a D-Channel.

The ISDN QSIG GF transport does not support the use of the MCDN proprietary Private Network Identifier (PNI) numbers for the customer association. For an ISDN PRI interface, the Private Integrated Services Network Exchange Customer (PINX_CUST) prompt in LD 17 is used to define a customer number to a D-Channel. This implies that all call independent messages on an ISDN PRI D-Channel are associated with a single customer, as configured in LD 17.

For example, in a multi-customer configuration, if every customer on a Meridian 1 PBX wishes to use the call-independent transport service over a QSIG PRI interface, then each customer requires the use of a separate D-Channel. Note that this limitation also applies to MCDN features that are converted by the MCDN to QSIG GF “protocol converter”, to function over a QSIG network.

- Basic call setup and call-independent connection setup originating from a QSIG supplementary service may be rejected by the far end, due to the unavailability of system resources, and due to overdo control. A notification message will be sent to the supplementary service, so that it may initiate a retry.
- The call-independent connection-oriented APDU transport service requires that a transit node PINX retain any information pertaining to a call-independent connection, for the duration of the connection. This information may be required by a particular application to properly route subsequent messages associated with the same call-independent connection.

This requirement at the transit PINX implies that system resources, such as call registers, are used by the call-independent transport services at the tandem nodes. Therefore, applications that make use of call-independent transport services may be limited by the amount of available system resources allocated at the transit PINX.

- The Call Independent APDU transport can be supported by two different transport mechanisms, connection-oriented, or without connection. The later method is not supported on the ISDN QSIG GF platform.
- The Association Control Service Element (ACSE) and Dialogue Service Element (DSE) for application association is not supported by the ISDN QSIG GF platform on a Meridian 1. ACSE and DSE protocol data units that are received by the ISDN GF platform facility at a destination Private Integrated Services Network Exchange (PINX) will be discarded. A transit INX will pass on the ACSE and DSE data units transparently to the next node.
- The ISDN QSIG GF platform supports up to eight Application Protocol Data Units (APDUs), that is, Remote Operation Service Element (ROSE) components, in one message, either in a single Facility Information Element (FIE) or multiple FIEs. If a supplementary service requests that the ISDN QSIG GF Transport send a message exceeding this limit, a notification message is sent to the supplementary service.

Feature interactions

There are no feature interactions for this feature.

Feature packaging

The following packages are required for the QSIG Supplementary Service - Name Display Enhancement feature:

- Q-reference Signaling Point Interface (QSIG) package 263
- QSIG Generic Functional Protocol Transport (QSIG-GF) package 305
- Integrated Services Digital Network (ISDN) package 145
- Multi-purpose Serial Data Link (MSDL) package 222
- Primary Rate Access (PRA) package 146 for 1.5 Mbps signaling, **or**
- International Primary Rate Access (IPRA) package 202 (for 2.0 Mbps signaling)
- 2.0 Mbps Primary Rate Access (PRI2) package 154 (for 2.0 Mbps signaling)

For ISDN Signaling Link (ISL) QSIG, the following additional packages are required:

- ISDN Signaling Link (ISL) package 147

For ISDN Basic Rate Interface Trunking (BRIT), the following additional packages are required:

- ISDN Basic rate Access (BRI) 216
- ISDN Basic Rate Trunk Access (BRIT) package 233

Feature operation

No operating procedures are required to use this feature.

ISDN QSIG-BC and QSIG-GF Compliance Update

With X11 Release 24.2x, two European QSIG interfaces are available. One version (called the ISDN QSIG-BC and QSIG-GF Compliance Update) is compliant with latest editions of the standards. The other version is compliant with older (X11 Release 22) versions of the standards.

The latest European standards documents for QSIG are:

- ETS 300-172, fourth edition (1997)
- ETS 300-239, second edition.

In X11 Release 20, support for the ISDN QSIG interface and the basic call capability was introduced. This version of the interface was based on the ETS 300-172, first edition (1990) document.

In X11 Release 22, the QSIG-GF Transport interface was introduced, in accordance with the ETS 300-239 (1993) standard document. The basic call functionality was upgraded to be compliant with the second edition of the ETS 300-172 document.

QSIG Basic Call Compliance Update

Comprehension required IEs

When an incoming Information Element (IE) coded “comprehension required” is not recognized, the system treats the message the same way it treats messages with a mandatory information element missing. A message is sent back with cause 96 “mandatory information element missing.” The messages returned are shown in Table 31.

Table 31
Unrecognized “comprehension required IE” treatment

Incoming message	Message returned
SETUP	RELEASE COMPLETE with cause 96
DISCONNECT	RELEASE with cause 96
RELEASE	RELEASE COMPLETE with cause 96
RELEASE COMPLETE	No message is returned (see note 1)
Other	STATUS with cause 96

Note 1: The message is treated as if it is received with cause 31 “Normal, unspecified.”

Transit counter

The transit counter IE is defined as a codeset 4 IE instead of codeset 5. The transit counter functionality is considered to be a supplementary service.

Restart procedure

When a RESTART message with a restart IE coded “Channel indicated” is received with no Channel ID IE, the message is treated as if a mandatory IE is missing. A STATUS message is returned with cause 96, (mandatory IE missing).

Progress indicator

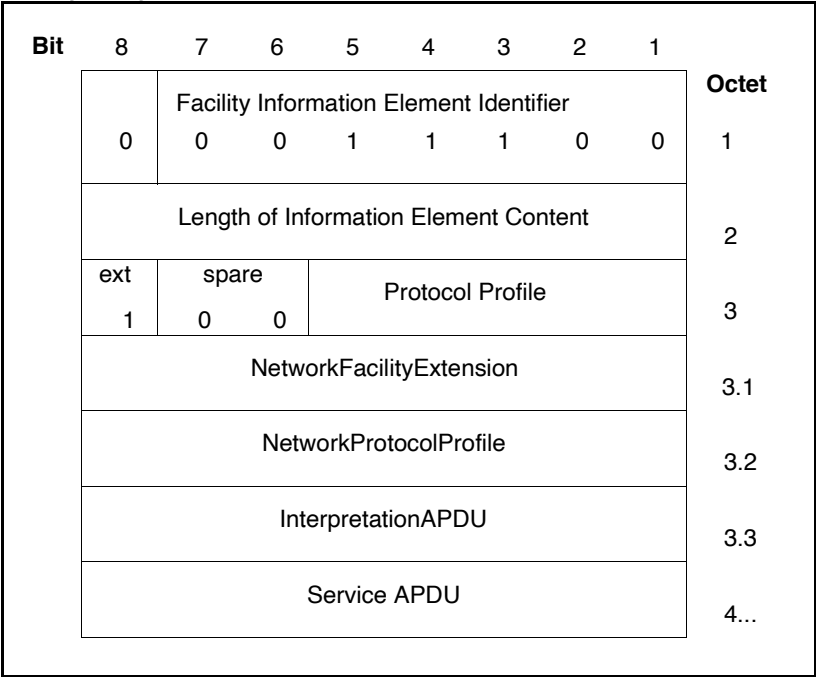
This is not defined.

QSIG GF Compliance Update

Facility IE coding

Figure 28 illustrates the layout of the Facility IE.

Figure 28
Facility IE layout



Protocol profile coding

Figure 29 illustrates three different examples of coding in the Protocol Profile.

Figure 29
Protocol Profile Coding

Bits					
5	4	3	2	1	
1	0	0	0	1	ROSE
1	0	0	1	1	ACSE
1	1	1	1	1	Networking Extension
All other values are reserved.					

On transmission, the Meridian 1 codes the Protocol Profile as “Networking Extension.”

On reception, the Meridian 1 supports Protocol Profile coded as “ROSE” or “Networking Extension.”

Network Protocol Profile coding

The Network Protocol Profile is used to specify the Service APDU coding type. Since the Meridian 1 only supports ROSE (the default coding type), the Network Protocol Profile is always omitted.

NFE and interpretation coding

No change is introduced by this update in the coding of the NFE and of the interpretation APDU.

Service APDU coding

The Service APDU coding is service dependent. It is not affected by this update.

Operating parameters

The X11 Release 22 version of QSIG is still available, to be used in networks where supplementary services are involved.

The ISDN QSIG-BC and QSIG-GF Compliance Update feature is not available on PRI 1.5 Mbps links. These links are usually configured with ISO QSIG GF.

The following optional parts of the protocol are not supported by the ISDN QSIG-BC and QSIG-GF Compliance Update feature:

- Message segmentation
- Multi-rate
- Indication of the channel with a map in the Channel ID IE

These systems require the QPC775 (Release E) or the QPC471 Clock Controller card.

The hardware required for the Option 11C is:

- PRI2 card (NTBK50)
- Downloadable D-channel Daughterboard card (NTBK51)
- Clock Controller (NTAK20)

The hardware required for ISDN BRI is as follows:

- SILC card (NT6D70)
- MISP card (NT6D73) for Option 51C, Option 61C, Option 81, and Option 81C systems
- MISP card (NTBK22) for the Option 11C system

Feature interactions

There are no feature interactions specified for this feature.

Feature packaging

The following software packages are required for ISDN PRI2 QSIG:

- ISDN Signaling (ISDN) package 145
- 2 Mbps PRI (PRI2) package 154
- International PRA (IPRA) package 202
- Multi-Purpose Serial Data Link (MSDL) package 222
- QSIG Interface (QSIG) package 263
- QSIG GF Transport (QSIGGF) package 305

The following software packages are required for ISL QSIG:

- ISDN Signaling (ISDN) package 145
- ISDN Signaling Link (ISL) package 147
- 2 Mbps PRI (PRI2) package 154
- Multi-Purpose Serial Data Link (MSDL) package 222
- QSIG Interface (QSIG) package 263
- QSIG GF Transport (QSIGGF) package 305

The following software packages are required for ISDN BRIT QSIG:

- ISDN Signaling (ISDN) package 145
- ISDN Basic Rate Access (BRI) package 216
- Multi-Purpose Serial Data Link (MSDL) package 222
- ISDN BRI Trunk Access (BRIT) package 233
- QSIG Interface (QSIG) package 263
- QSIG GF Transport (QSIGGF) package 305

Feature operation

There are no operating procedures specified for this feature.

ISDN QSIG Channel ID Coding

PBXs use different kinds of timeslot mapping in ISDN QSIG networks using PRI2 links. When the Meridian 1 interworks with these switches, the QSIG Channel ID Coding feature allows one of two mapping options to be configured.

The QSIG Channel ID Coding feature provides an option for two different mapping schemes for timeslots 17-31 on a PRI2 loop. The mapping scheme for timeslots 1-15 is not affected.

The mapping scheme is applicable for both encoding and decoding of the Channel ID IE.

Prior to the introduction of QSIG Channel ID Coding, the channel number in the Channel ID Information Element (IE) on a Meridian 1 PRI2 QSIG link matched the timeslot number. A summary of the previous mapping is shown in Table 32.

Table 32
Previous implementation of Channel IE and timeslot number

Timeslot number	Channel number for Meridian 1 software	Channel number in Channel ID IE
1-15	1-15	1-15
17-31	16-30	17-31
16 (signaling)	31 (D-channel)	not applicable

CHID IE mapping options

With the QSIG Channel ID Coding feature, the D-channel can be configured with one of two CHID IE mapping options.

Timeslot number coded in CHID IE

With this option, the channel number of the CHID IE matches the timeslot number. Timeslots 17-31 are assigned channel numbers 17-31. This is the same as the method used prior to the introduction of the QSIG Channel ID Coding feature. A summary of the mapping performed is shown in Table 33.

Table 33
Timeslot number is coded in CHID IE

Timeslot number	Channel number for Meridian 1 software	Channel number in Channel ID IE.
1-15	1-15	1-15
17-31	16-30	17-31
16 (signaling)	31 (D-channel)	not applicable

Channel number coded in CHID IE

With this option, the channel number of the CHID IE matches the Meridian 1 software channel number. Timeslots 17-31 are assigned channel numbers 16-30. A summary of the mapping performed is shown in Table 34.

Table 34
Channel number in Meridian 1 software is coded in CHID IE

Timeslot number	Channel number in Meridian 1 software	Channel number in Channel ID IE.
1-15	1-15	1-15
17-31	16-30	16-30
16 (signaling)	31 (D-channel)	not applicable

The option for the QSIG Channel ID Coding feature is configured in Overlay 17. The two choices are:

- QCHID = NO – Timeslot values 17-31 of the PRI2 loop are associated with channel number 16-30 of the Channel ID IE.
- QCHID = YES – Timeslot values 17-31 of the PRI2 loop are associated with channel number 17-31 of the Channel ID IE (same as previous operation).

Operating parameters

QSIG Channel ID Coding is only supported on QSIG interfaces equipped with the GF capability (introduced in X11 Release 22).

These systems require the QPC775 (Release E) or the QPC471 Clock Controller card.

The hardware required for the Option 11C is:

- PRI2 card (NTBK50)
- Downloadable D-channel Daughterboard card (NTBK51)
- Clock Controller (NTAK20)

When a Meridian 1 connects to another Meridian 1 system, the proper setting for QCHID (NO or YES) depends on the software equipped at both switches. If both switches are equipped with X11 Release 24.2x (or later) software, QCHID can be YES or NO, provided it has the same value on both switches. If one switch is equipped with pre-Release 24.2x software, the Release 24.2x switch must be configured with QCHID = YES.

Feature interactions

There are no feature interactions specified for this feature.

Feature packaging

The following packages are required for the QSIG Channel ID Coding feature:

- ISDN Signaling (ISDN) package 145
- 2 Mbps PRI (PRI2) package 154
- International PRA (IPRA) package 202
- Multi-Purpose Serial Data Link (MSDL) package 222
- QSIG Interface (QSIG) package 263
- QSIG GF Transport (QSIGGF) package 305

Feature operation

There are no operating procedures specified for this feature.

ISDN QSIG Generic Functional Transport

This feature provides a generic transport platform that will support QSIG compliant supplementary services and ISDN networking applications on a Private Integrated Services Digital Network (PISN), in accordance with the International Standards Organization (ISO) and the European Telecommunication Standards Institute (ETSI.)

ISDN QSIG GF Transport is supported on the following interfaces with the D-channel configured as International Standards Organization (ISO QSIG) or European Telecommunications Standard Institute (ETSI QSIG):

- ISDN 1.5 Mbps Primary Rate Interface (PRI),
- ISDN 2 Mbps Primary Rate Interface (PRI2),
- ISDN Basic Rate Interface (BRI) Trunk, and
- ISDN Signaling Link (ISL).

Operating parameters

DN address translation requires the association with a customer number. For a BRI trunk Digital Subscriber Loop (DSL) interface, the customer number association is with the D-channel. For PRI, a prompt creates a customer number association with a given D-channel.

Feature interactions

There are no feature interactions specified for this feature.

Feature packaging

The following packages are required for ISDN QSIG GF Transport:

- ISDN QSIG GF Transport (QSIGGF) is package 295.

QSIG on 2.0 Mbit PRI interface requires the following packages:

- QSIG Interface (QISG) package 263;
- QSIG GF Transport (QSIG GF) package 295;
- Integrated Services Digital Network (ISDN) package 145;
- Multi-purpose Serial Data Link (MSDL) package 222;

- 2.0 Mbit Primary Rate Interface (PRI2) package 154; and
- International Primary Rate Access (IPRA) package 202.

ISDN Signaling Link (ISL) interface requires the following packages:

- QSIG Interface (QSIG) package 263;
- QSIG GF Transport (QSSIGF) package 295;
- Integrated Service Digital Network Signaling Link (ISL) package 147; and
- Multi-purpose Serial Data Link (MSDL) package 222.

For ISL shared mode:

- Integrated Services Digital Network (ISDN) package 145; and
- 2.0 Mbit Primary Rate Interface (PRI2) package 154.

ISDN BRIT QSIG interface requires the following packages:

- QSIG Interface (QSIG) package 263;
- QSIG GF Transport (Qsig GF) package 295;
- Integrated Services Digital Network (ISDN) package 145; Multi-purpose Serial Data Link (MSDL) package 222;
- Basic Rate Interface (BRI) package 216; and
- Integrated Services Digital Network Basic Rate Interface Trunk Access (BRIT) package 233.

Feature operation

There are no operating procedures specified for this feature.

ISDN QSIG Name Display

The Integrated Services Digital Network (ISDN) Q Reference Signaling Point (QSIG) Name Display feature provides three supplementary services pertaining to the calling or connected party name display. These supplementary services allow the calling/connected party to either present or restrict the display of name identification on an ISDN PRI or ISDN BRI link. ISDN QSIG Name Display feature supports the following services:

- Calling Name Identification Presentation (CNIP),
- Connected Name Identification Presentation (CONP), and
- Calling/Connected Name Identification Restriction (CNIR).

Calling Name Identification Presentation

Calling Name Identification Presentation (CNIP) service is available to the called/connected party. When this service is enabled, the calling party's name is displayed on the connected party's set. CNIP service is available on a permanent basis only, and requires the calling party to have the Class of Service Name Presentation Allowed, defined in LD 10 or LD 11.

Connected Name Identification Presentation

Connected Name Identification Presentation (CONP) is a service available to the calling party. When this service is enabled, the called/connected party's name is displayed on the calling party's set. CONP is provided on a permanent basis only and requires the called/connected party to have the Class of Service Name Presentation Allowed, defined in LD 10 or LD 11.

Calling/Connected Name Identification Restriction

Calling/Connected Name Identification Restriction (CNIR) prevents the calling/connected party's name from being presented to a called/calling party. CNIR is invoked on either a permanent basis, provided the calling/connected party has a Class of Service Name Presentation Denied defined in LD 10 or LD 11, or on a per-call basis. When CNIR is activated on a per-call basis, the Class of Service Name Presentation Allowed is configured, and Calling Party Privacy (CPP) Flexible Feature Code (FFC) is dialed prior to initiating a call. This supplementary service restricts presentation of the calling/connected party's name during a normal call establishment and also when the possibility of name presentation arises from the operation of other features, such as Call Transfer, Call Forwarding or Hunting.

Operating parameters

ISDN QSIG Name Display feature requires the QSIG General Functional (GF) transport platform. QSIG GF protocols provide name information that is contained in the facility information element. This protocol transports call control messages from one network to another.

Fifty characters string length is the maximum allowable size of name display information for QSIG. The Call Party Name Display (CPND) feature accepts 27 characters string length. If a name display exceeds these parameters, the name is truncated on the receiving set.

Interworking with existing Meridian Customer Defined Network (MCDN) Name Display feature is dependent upon the remote capability of the associated D-channel defined in LD 16 for BRI routes and LD 17 for PRI routes.

Individual service profiles cannot override Calling Name Identification Restriction requests. Therefore, when name display information has been restricted, it is not possible to obtain this information on the display of the set.

For ISDN BRI sets to have the capability to activate Calling/Connected Name Identification Restriction (CNIR) on a permanent basis, the PRES prompt in LD 27 must be defined PRES = NO.

Calling Party Privacy (CPP) Flexible Feature Codes (FFC) is not supported on BRI sets. Therefore, CNIR on a per-call basis is not supported on BRI sets.

Calling/Connected Name Identification Restriction (CNIR) takes precedence over Calling Name Identification Presentation (CNIP) and Connected Name Identification Presentation (CONP) services.

Feature interactions

Call Modification

If an incoming QSIG with name display presentation allowed receives call modification treatment, such as Call Pickup, Call Transfer, Conference, Call Forward All Types or Hunt, the reason for call modification is not displayed on the calling party's set. Name display information is not updated on the calling party's set. However, if a CONNECT message is sent, the calling party's name display is updated on receipt of this message.

Call Pickup

An incoming QSIG call with name display presentation allowed has name information displayed on the set that picks up the call. If the incoming QSIG call has presentation denied, the calling party's name is not displayed on the set picking up the incoming call.

Call Pickup Network Wide

When a QSIG call with name display presentation allowed is picked up on a MCDN, the calling party's name information is displayed on the set that answers the call. If presentation restricted is defined, then name information is not displayed.

Call Transfer

After the completion of a call transfer, an incoming QSIG call with name display presentation allowed has name information displayed on the destination set. If the incoming QSIG call has presentation denied, name information is not displayed to the destination set.

Call Transfer Network Wide

When a QSIG call with name display presentation allowed is transferred over a MCDN, the calling party's name information is displayed on the set that answers the call. If presentation restricted is defined, then name information is not displayed.

Conference

Conference Network Wide

An incoming QSIG call with name display presentation allowed is conferenced locally. When a conferee drops out of the conference, calling party's name information is displayed and is passed on to another conferee. Name display information remains until the last local set remains on the call. With presentation restricted, the calling party's name information is not displayed as conferees leave the call.

Call Forward All Calls

When an incoming QSIG call, with name display presentation allowed Name Display, is forwarded locally, the calling party's name information is displayed on the forwarding set. With presentation restriction, the calling party's name information is not displayed to the destination set.

Call Forward All Calls - Network Wide

When a QSIG call with name display presentation allowed is forwarded over a MCDN, the calling party's name information is displayed on the forwarded set. If presentation restricted is defined, then name information is not displayed on the destination set.

Call Party Name Display**Calling Party Name Display Denied**

Call Party Name Display and Calling Party Name Display Denied interact with ISDN QSIG Name Display, depending on the Name Display configuration in LD 16 for BRI or LD 17 for PRI. When a QSIG network is interacting with an MCDN network providing network capability ND3, both the MCDN and QSIG Name Display feature function on the same level.

Calling Party Privacy

Calling Party Privacy (CPP) takes precedence over the ISDN QSIG Name Display feature.

Hunting

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.

Hunting - Network Wide

When a QSIG call with name display presentation allowed is forwarded over a MCDN, the calling party's name information is displayed on the forwarded set. If presentation restricted is defined, then name information is not displayed on the forwarded set.

Incoming DID Digit Conversion**Incoming DID Digit Conversion Network Wide**

IDC trunk and name information is passed and displayed to the terminating party when no name information is received from the Direct Inward Dial (DID) trunk. The Incoming DID Digit Conversion (IDC) feature is activated, and name information is associated with the converted digit sequence.

Name information received from a DID trunk takes precedence over an IDC trunk name.

Feature packaging

ISDN QSIG Name Display requires the following packages:

- Calling Party Name Display (CPND) package 95;
- Digit Display (DDSP) package 19;
- International Supplementary Features (SUPP) package 131;
- Calling Party Privacy (CPP) package 301;
- Flexible Feature Codes (FFC) package 139.

The following packages are required for ISDN 2.0 Mbit Primary Rate Interface:

- ISDN Basic Call QSIG (QSIG) package 263;
- QSIG GF Transport (QSIGGF) package 305;
- Integrated Services Digital Network (ISDN) package 145;
- Multipurpose Serial Data Link (MSDL) package 222;
- 2.0 Mbit Primary Rate Interface (PRI2) package 154; and
- International Primary Rate Access (IPRA) package 202.

The following packages are required for a Basic Rate Interface QSIG Interface:

- ISDN Basic Call QSIG (QSIG) package 263;
- QSIG GF Transport (QSIGGF) package 305;
- Integrated Services Digital Network (ISDN) package 145;
- Multipurpose Serial Data Link (MSDL) package 222;
- Basic Rate Interface (BRI) package 216; and
- ISDN BRI Trunk Access (BRIT) package 233.

Feature operation

To activate Calling/Connected Name Identification Restriction (CNIR) on a per call basis, the calling party dials the Calling Party Privacy (CPP) Flexible Feature Code prior to initiating the call.

ISDN QSIG Name Display Enhancement

The ISDN QSIG Supplementary Service - Name Display feature, introduced in Release 22, supports the following Name Display supplementary services across an ISDN QSIG network:

- **Calling Name Identification Presentation (CNIP).** This service is offered to the called party, to provide the name of the calling party. Once configured, CNIP is permanently activated.

CNIP is based on the Class of Service of the originating set, as configured in LD 10 or LD 11. If the CLS prompt is configured as Name Presentation Allowed (NAMA), the name of the calling party is displayed on the set of the terminating party. If the CLS prompt is set to Name Presentation Denied (NAMD), or if the calling party name is not available (due, for example, to the name not being registered, or because of call interworking), then CNIP is not able to provide the caller's name to the called party.

Note: To restrict the presentation of a calling party on ISDN BRI sets, the Presentation (PRES) prompt in LD 27 is set to NO. This applies for the other Name Display supplementary services as well.

- **Connected Name Identification Presentation (CONP).** This service applies to an established call. It is offered to the calling party, providing alerting information, or the called or connected party name. Once configured, CONP is permanently activated.

CONP is based on the Class of Service of the terminating set, as configured in LD 10 or LD 11. If the CLS prompt is configured as Name Presentation Allowed (NAMA), the name of the called party is displayed on the set of the calling party. If the CLS prompt is set to Name Presentation Denied (NAMD), or if the called party name is not available (due, for example, to the name not being registered, or because of call interworking), then the called party name is not provided to the calling party.

- **Calling/Connected Name Identification Restriction (CNIR).** This service prevents the name display of a calling party from being presented to the called party.

CNIR may be activated for all calls by setting the CLS prompt in LD 10 or LD 11 to Name Presentation Denied (NAMD). If the CLS has been set to Name Presentation Allowed (NAMA), CNIR may be activated on a per-call basis, by dialing the Calling Party Privacy Flexible Code before making a call.

Note: The capability of activating CNIR on ISDN BRI sets is not supported.

These Name Display supplementary services are provided across an ISDN QSIG network using the QSIG Generic Functional (GF) protocol transport platform introduced in Generic X11 Release 22.

The QSIG Supplementary Service - Name Display Enhancement feature, introduced in Release 24, provides the option of choosing the method of Operation Coding for the Name Display supplementary services described above. The choice may be by Object Identifier, used for European Telecommunications Standard Institute (ETSI) interfaces, or by Integer Value, used for International Standards Organization (ISO) interfaces. The Operation Coding method is configured as a remote capability using the RCAP prompt for an ISDN PRI D-Channel for ISDN PRI, or for an ISDN BRI route for ISDN BRI trunking.

Operating parameters

QSIG allows a maximum of 50 characters to be displayed for Name Display information. However, up to only 27 characters are allowed with the Calling Party Name Display feature. Consequently, if the name display string exceeds 27 characters, it will be truncated.

The QSIG Supplementary Service - Name Display Enhancement feature may interwork with the MCDN Name Display feature, depending on the Network Name Display (ND) method (either ND1, ND2, or ND3) configured as the Remote Capability (using RCAP prompt in LD 17) for the associated D-Channel. ND3 ensured the same level of service between the MCDN Name Display and QSIG Name Display supplementary services.

The user-based service profile cannot be used to override the Calling Name Identification Restriction service.

For Calling/Connected Name Identification Restriction (CNIR), restricting the presentation of name information is supported only for a calling party.

Busy name information sent with a DISCONNECT message is only supported for Overlap Receiving (the OVLRL prompt is set to YES in LD 17).

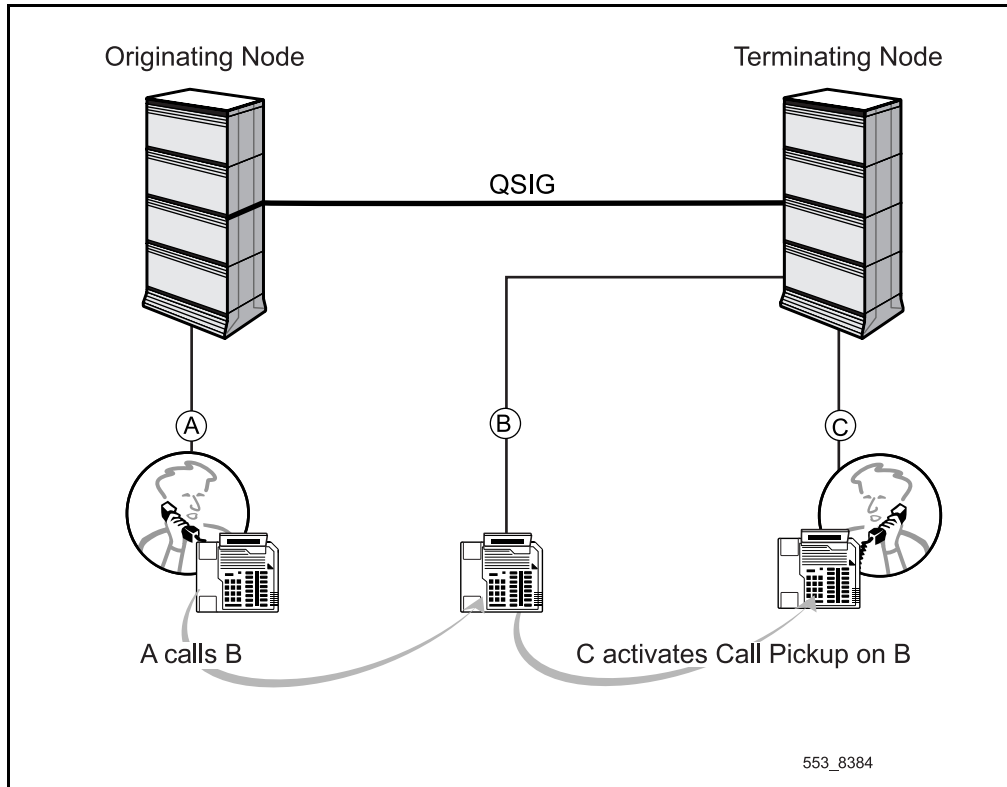
Feature interactions

Call Pickup (over a QSIG link)

Referring to Figure 30, Station A, at the originating node, calls Station B, at the terminating node, over a QSIG link. Station C, also at the terminating node, picks up the call.

If Station A has a Class of Service of Name Presentation Allowed (CLS = NAMA in LD 10 or LD 11), then Station A's name information will be presented to Station C. If Station A has a Class of Service of Name Presentation Denied (CLS = NAMD in LD 10 or LD 11), then Station A's name information will not be presented to Station C. Station A's display will be updated with the connected party (Station C) information, once the call pickup has been established. The reason for the call modification (Pickup) will not be presented to Station A.

Figure 30
Call Pickup over a QSIG link



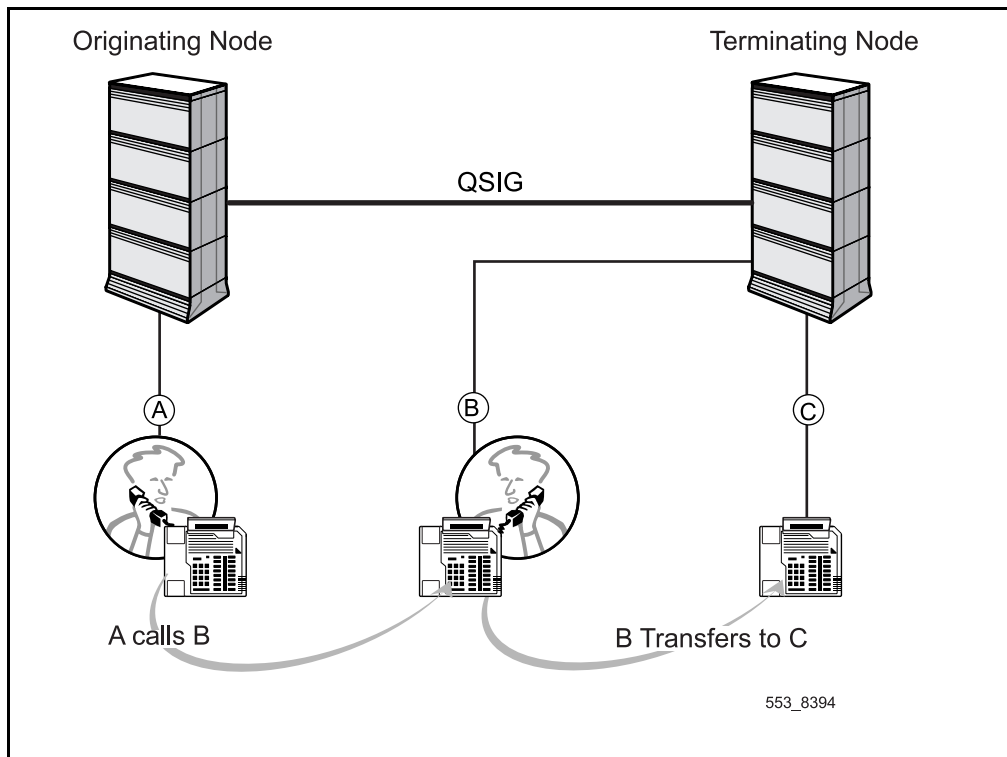
Call Transfer (over a QSIG link)

Referring to Figure 31, Station A, at the originating node, calls Station B, at the terminating node, over a QSIG link. Station B transfers the call to Station C, also at the terminating node.

If Station A has a Class of Service of Name Presentation Allowed (CLS = NAMA in LD 10 or LD 11), then Station A's name information will be presented to Station C, after the transfer has been completed.

If Station A has a Class of Service of Name Presentation Denied (CLS = NAMD in LD 10 or LD 11), then Station A's name information will not be presented to Station C, after the transfer has been completed. Station A's display will not be updated with the connected party (Station C) information, once the call transfer has been completed. The reason for the call modification (Transfer) will not be presented to Station A.

Figure 31
Call Transfer over a QSIG link



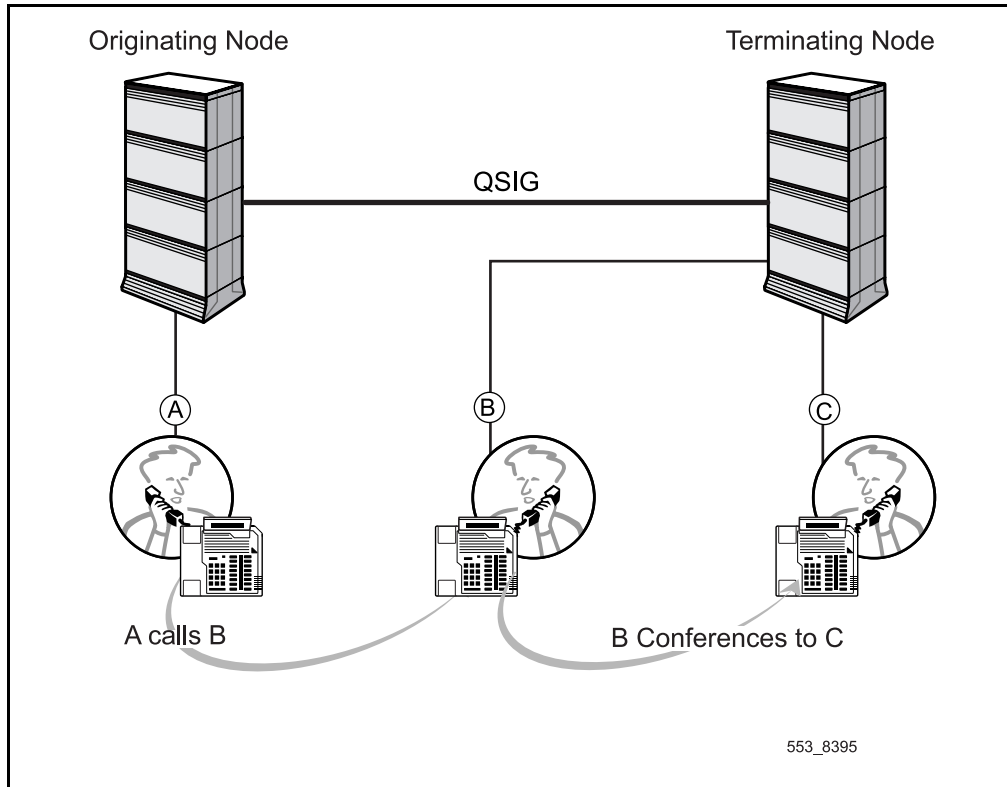
Conference (over a QSIG link)

Referring to Figure 32, Station A, at the originating node, calls Station B, at the terminating node, over a QSIG link. Station B conferences the call to Station C, also at the terminating node.

If Station A has a Class of Service of Name Presentation Allowed (CLS = NAMA in LD 10 or LD 11), then Station A's name information will be presented to Station C, after Station B drops out of the conference.

If Station A has a Class of Service of Name Presentation Denied (CLS = NAMD in LD 10 or LD 11), then Station A's name information will not be presented to Station C, after Station B drops out of the conference. Station A's display will be updated with Station B's name received in alert and connect messages.

Figure 32
Conference over a QSIG link



Call Forward all types (over a QSIG link)

Call Forward all types comprises the following call modifications:

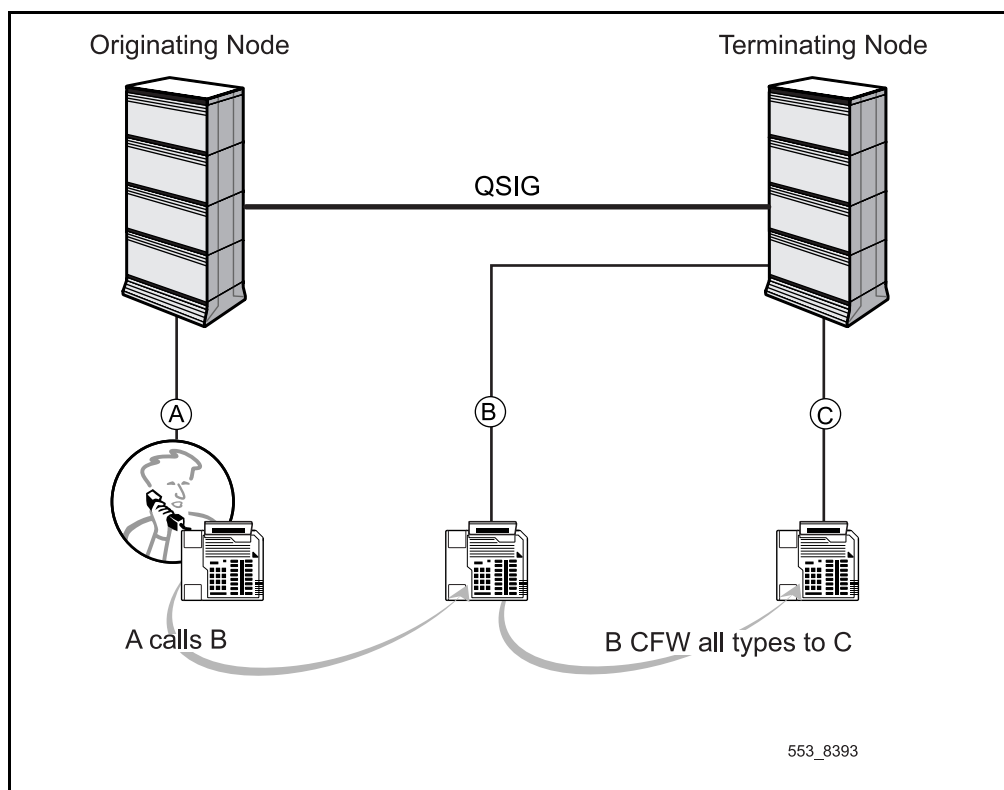
- Call Forward All Calls (CFW)
- Call Forward Busy (CFB)
- Call Forward by Call Type (CFT)
- Call Forward External Deny (CFX)
- Call Forward, Internal Calls
- Call Forward No Answer/Flexible Call Forward No Answer (CFNA)
- Call Forward No Answer, Second Level

Referring to Figure 33, Station A, at the originating node, calls Station B, at the terminating node, over a QSIG link. Station B forwards the call to Station C, also at the terminating node.

If Station A has a Class of Service of Name Presentation Allowed (CLS = NAMA in LD 10 or LD 11), then Station A's name information will be presented to Station C, after Station B completes the call forward. If Station A has a Class of Service of Name Presentation Denied (CLS = NAMD in LD 10 or LD 11), then Station A's name information will not be presented to Station C, after Station B completes the call forward.

In the case where Station B's set rings before Station C's set (during Call Forward No Answer), then Station A's display will be updated with Station C's name information, once Station C's set rings. In the case where Station C's set rings directly (during Call Forward All Calls), then Station A's display will be updated directly with Station C's name information. The reason for the call forward (type of call forward) will not be presented to Station A.

Figure 33
Call Forward All Types over a QSIG link



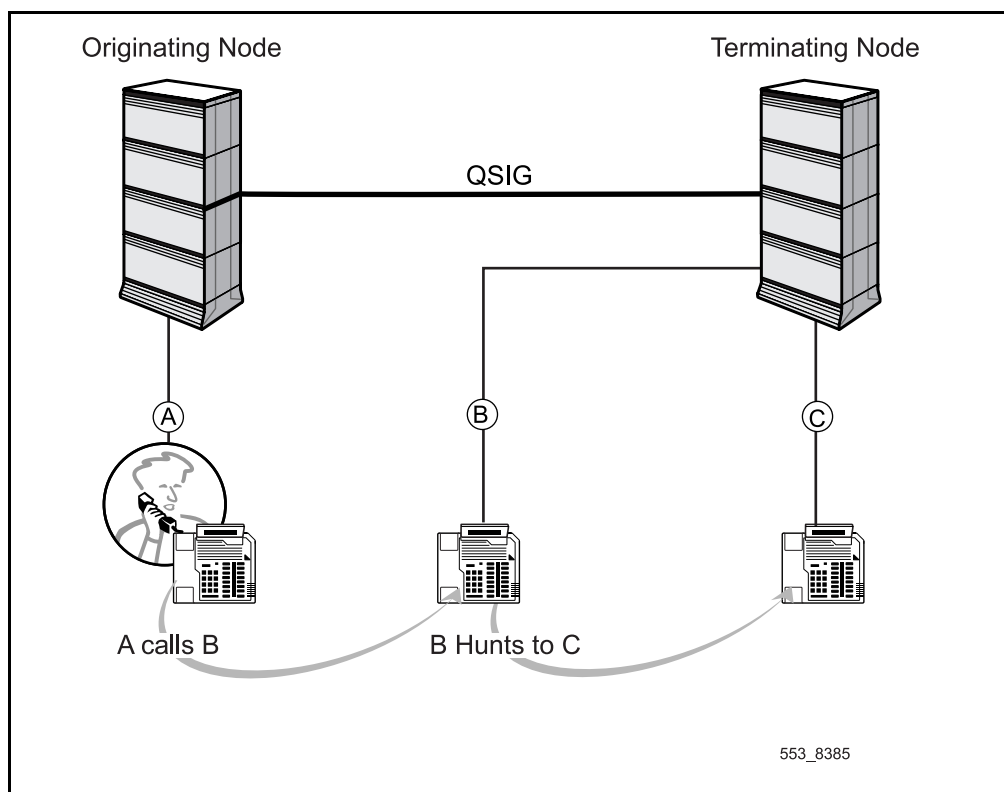
Hunt (over a QSIG link)

Referring to Figure 34, Station A, at the originating node, calls Station B, at the terminating node, over a QSIG link. The call is hunted from Station B to Station C, also at the terminating node.

If Station A has a Class of Service of Name Presentation Allowed (CLS = NAMA in LD 10 or LD 11), then Station A's name information will be presented to Station C, after the call hunt has been completed to Station C (Station C's set is ringing). If Station A has a Class of Service of Name Presentation Denied (CLS = NAMD in LD 10 or LD 11), then Station A's name information will not be presented to Station C, after the call hunt has been completed to Station C (Station C's set is ringing).

Station A's display will be updated with the Station C's name information, while Station C's set is ringing. The reason for the call modification (Hunt) will not be presented to Station A.

Figure 34
Hunt over a QSIG link

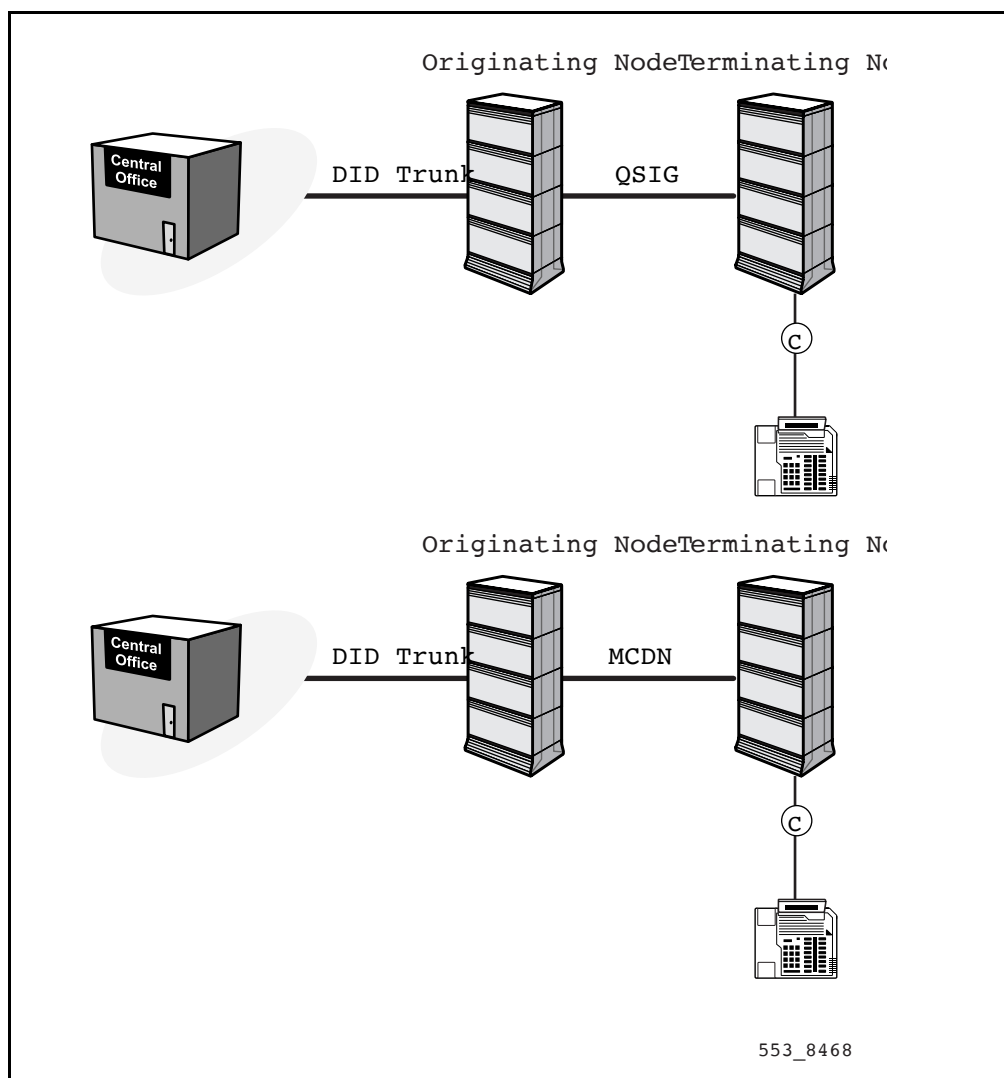


Incoming Digit Conversion (over a QSIG link)

Referring to Figure 35, a Central Office DID call comes over a Direct Inward Dial (DID) trunk to an originating node, and terminates on Station C at a terminating node, over a QSIG link.

If no name information is received from the DID trunk, and Incoming Digit Conversion (IDC) is activated and the incoming name information is associated to the new digit sequence, then this name information is passed to Station C at the terminating node. If name information is received from the DID trunk, this takes precedence over the IDC trunk name.

Figure 35
Incoming Digit Conversion over a QSIG link



Call Forward All Types (QSIG/MCDN link)

Call Forward all types comprises the following call modifications:

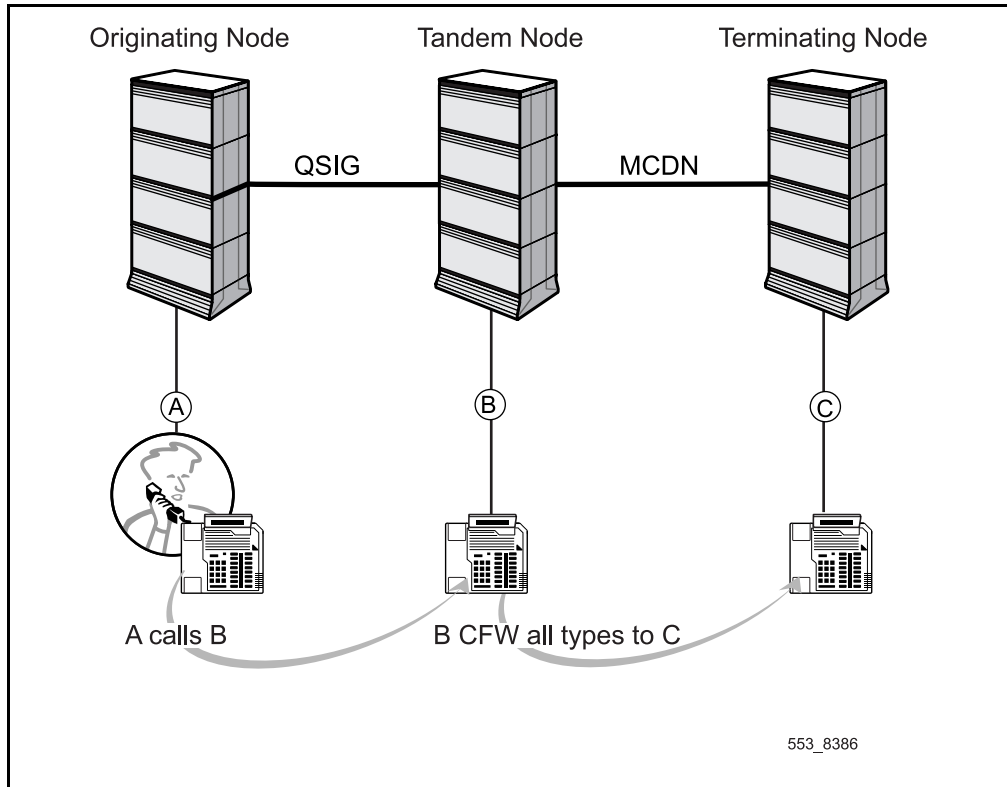
- Call Forward All Calls (CFW)
- Call Forward Busy (CFB)
- Call Forward by Call Type (CFT)
- Call Forward External Deny (CFX)
- Call Forward, Internal Calls
- Call Forward No Answer/Flexible Call Forward No Answer (CFNA)
- Call Forward No Answer, Second Level

Referring to Figure 36, Station A, at the originating node, calls Station B, at a tandem node, over a QSIG link. Station B forwards the call to Station C, at the terminating node, over an MCDN link.

If Station A has a Class of Service of Name Presentation Allowed (CLS = NAMA in LD 10 or LD 11), then Station A's name information will be presented to Station C, after Station B completes the call forward. If Station A has a Class of Service of Name Presentation Denied (CLS = NAMD in LD 10 or LD 11), then Station A's name information will not be presented to Station C, after Station B completes the call forward.

In the case where Station B's set rings before Station C's set (as in the case of Call Forward No Answer), then Station A's display will be updated with Station C's name information, once Station C's set rings. In the case where Station C's set rings directly (as in the case of Call Forward All Calls), then Station A's display will be updated directly with Station C's name information. The reason for the call forward (type of call forward) will not be presented to Station A.

Figure 36
Call Forward All Calls over a QSIG/MCDN link



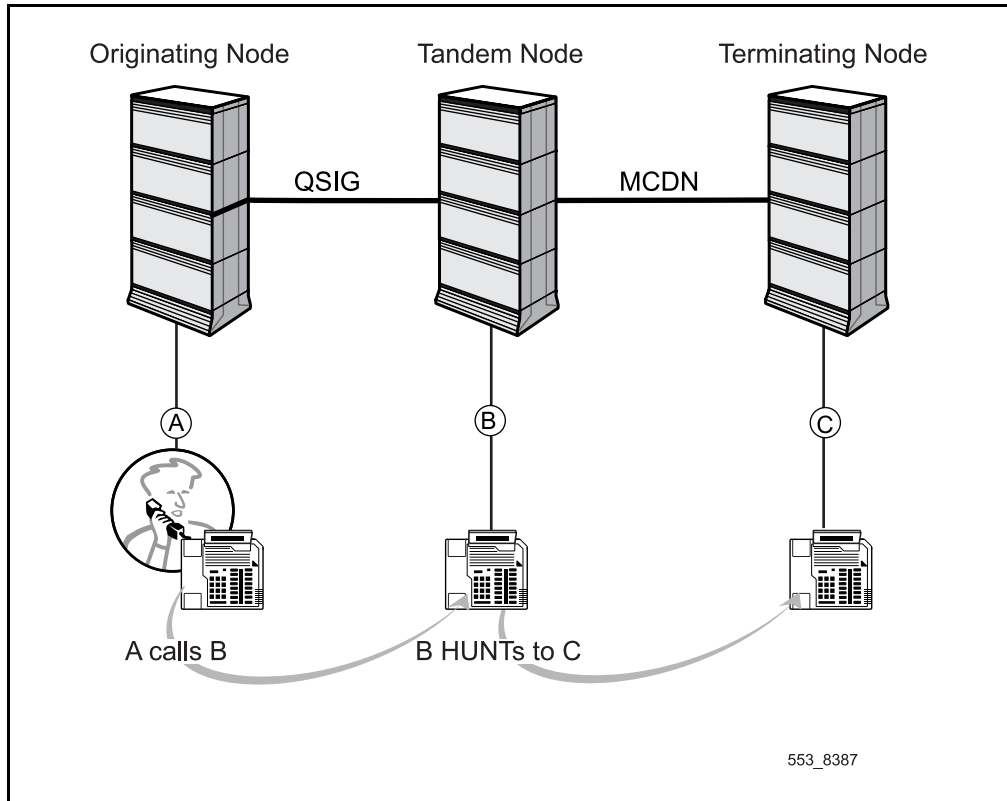
Hunt (QSIG/MCDN link)

Referring to Figure 37, Station A, at the originating node, calls Station B, at the terminating node, over a QSIG link. The call is hunted over an MCDN link from Station B to Station C, at the terminating node.

If Station A has a Class of Service of Name Presentation Allowed (CLS = NAMA in LD 10 or LD 11), then Station A's name information will be presented to Station C, after the call hunt has been completed to Station C (Station C's set is ringing).

If Station A has a Class of Service of Name Presentation Denied (CLS = NAMD in LD 10 or LD 11), then Station A's name information will not be presented to Station C, after the call hunt has been completed to Station C (Station C's set is ringing). Station A's display will be updated with the Station C's name information, while Station C's set is ringing. The reason for the call modification (Hunt) will not be presented to Station A.

Figure 37
Hunt over a QSIG/MCDN link

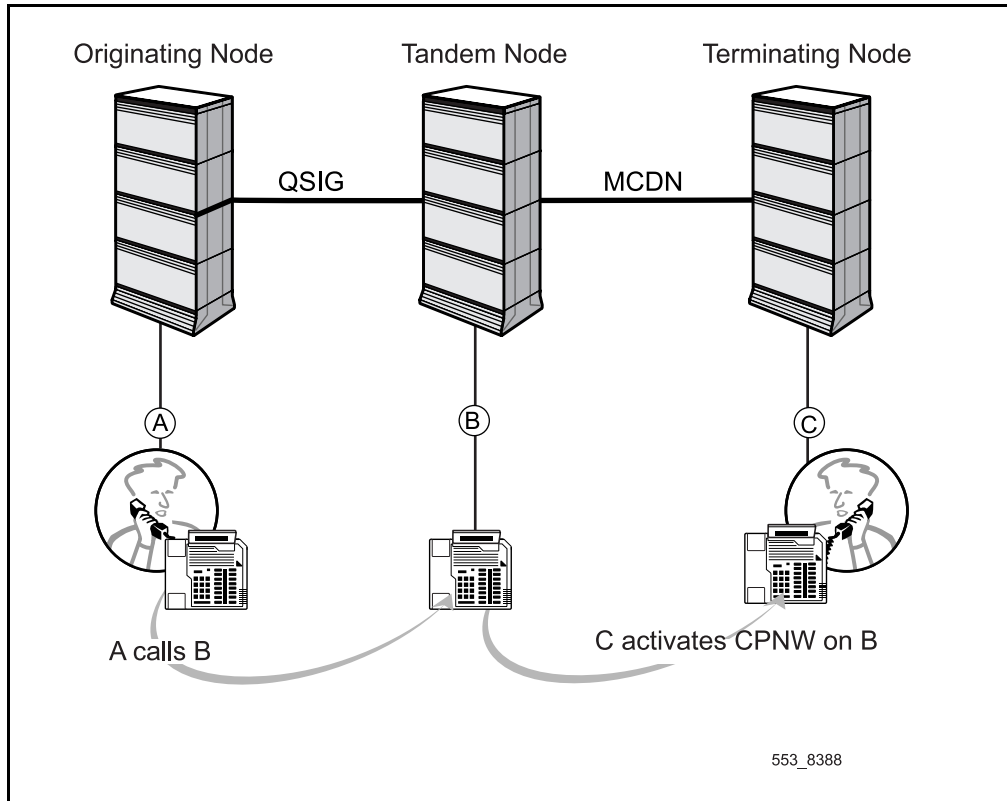


Call Pickup Network Wide (QSIG/MCDN link)

Referring to Figure 38, Station A, at the originating node, calls Station B, at the terminating node, over a QSIG link. Station C, at the terminating node, picks up the call over an MCDN link.

If Station A has a Class of Service of Name Presentation Allowed (CLS = NAMA in LD 10 or LD 11), then Station A's name information will be presented to Station C. If Station A has a Class of Service of Name Presentation Denied (CLS = NAMD in LD 10 or LD 11), then Station A's name information will not be presented to Station C. Station A's display will be updated with the connected party (Station B) information, once the call pickup has been established. The reason for the call modification (Pickup) will not be presented to Station A.

Figure 38
Call Pickup Network Wide over a QSIG/MCDN link



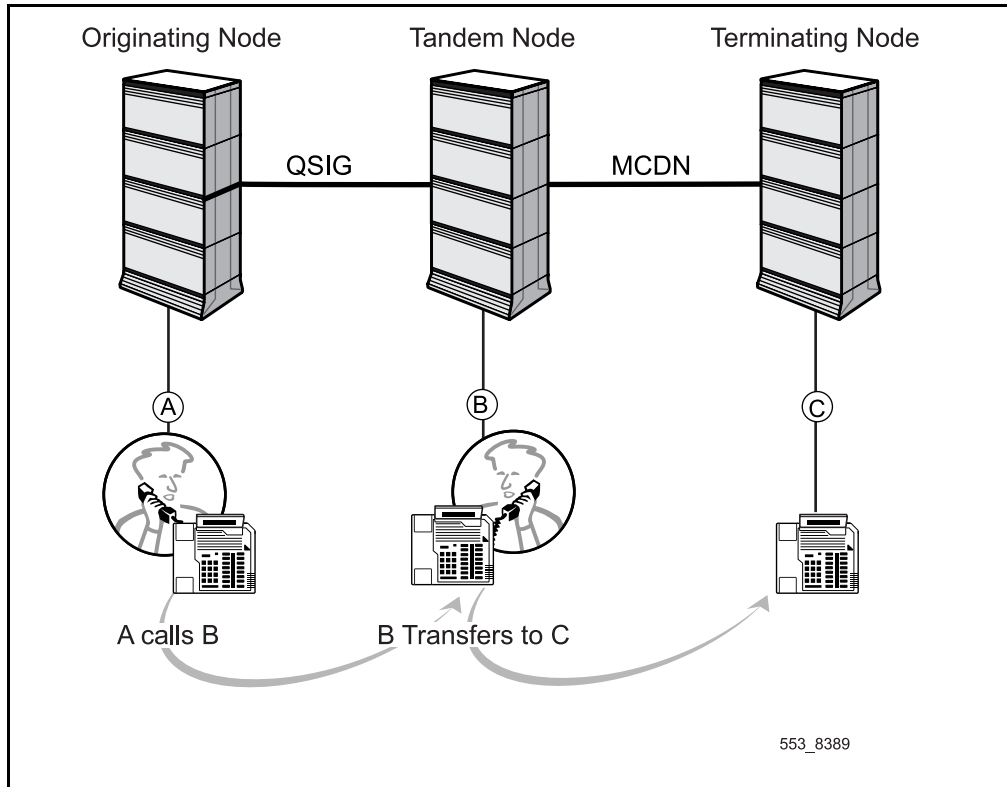
Call Transfer (QSIG/MCDN link)

Referring to Figure 39, Station A, at the originating node, calls Station B, at the terminating node, over a QSIG link. Station B transfers the call to Station C, at the terminating node, over an MCDN link.

If Station A has a Class of Service of Name Presentation Allowed (CLS = NAMA in LD 10 or LD 11), then Station A's name information will be presented to Station C, after the transfer has been completed.

If Station A has a Class of Service of Name Presentation Denied (CLS = NAMD in LD 10 or LD 11), then Station A's name information will not be presented to Station C, after the transfer has been completed. Station A's display will not be updated with the connected party (Station C) information, once the call transfer has been completed. The reason for the call modification (Transfer) will not be presented to Station A.

Figure 39
Call Transfer over a QSIG/MCDN link



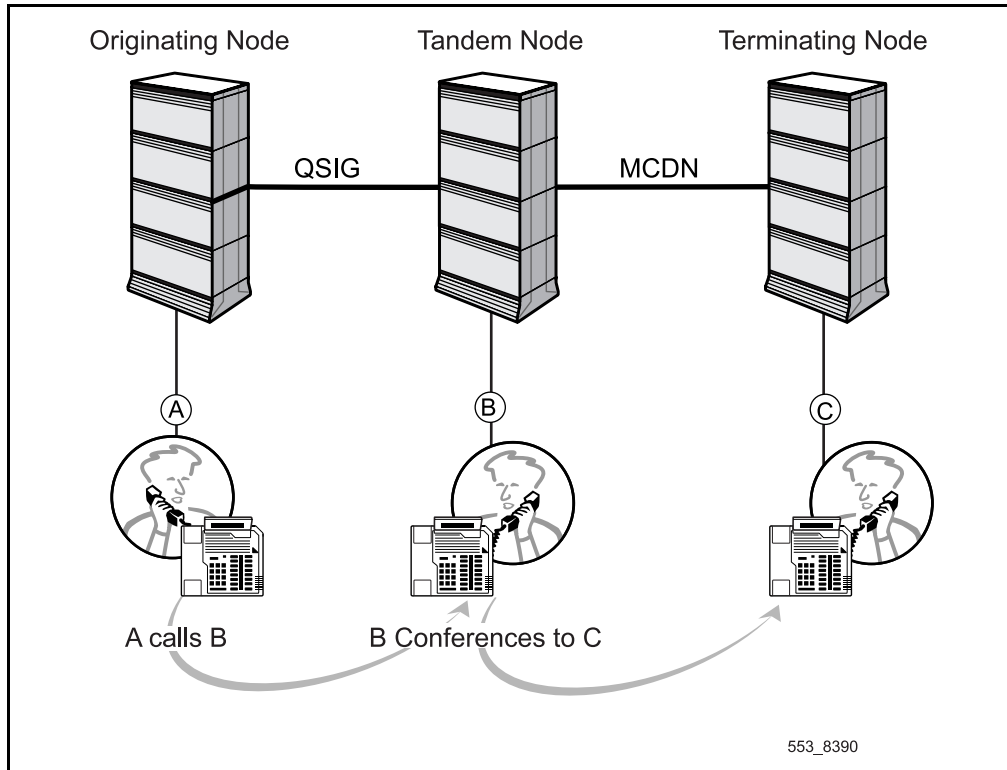
Conference (QSIG/MCDN link)

Referring to Figure 40, Station A, at the originating node, calls Station B, at the terminating node, over a QSIG link. Station B conferences the call to Station C, at the terminating node, over an MCDN link.

If Station A has a Class of Service of Name Presentation Allowed (CLS = NAMA in LD 10 or LD 11), then Station A's name information will be presented to Station C, after Station B drops out of the conference.

If Station A has a Class of Service of Name Presentation Denied (CLS = NAMD in LD 10 or LD 11), then Station A's name information will not be presented to Station C, after Station B drops out of the conference. Station A's display will be updated with Station B's name received in alert and connect messages.

Figure 40
Conference over a QSIG/MCDN link

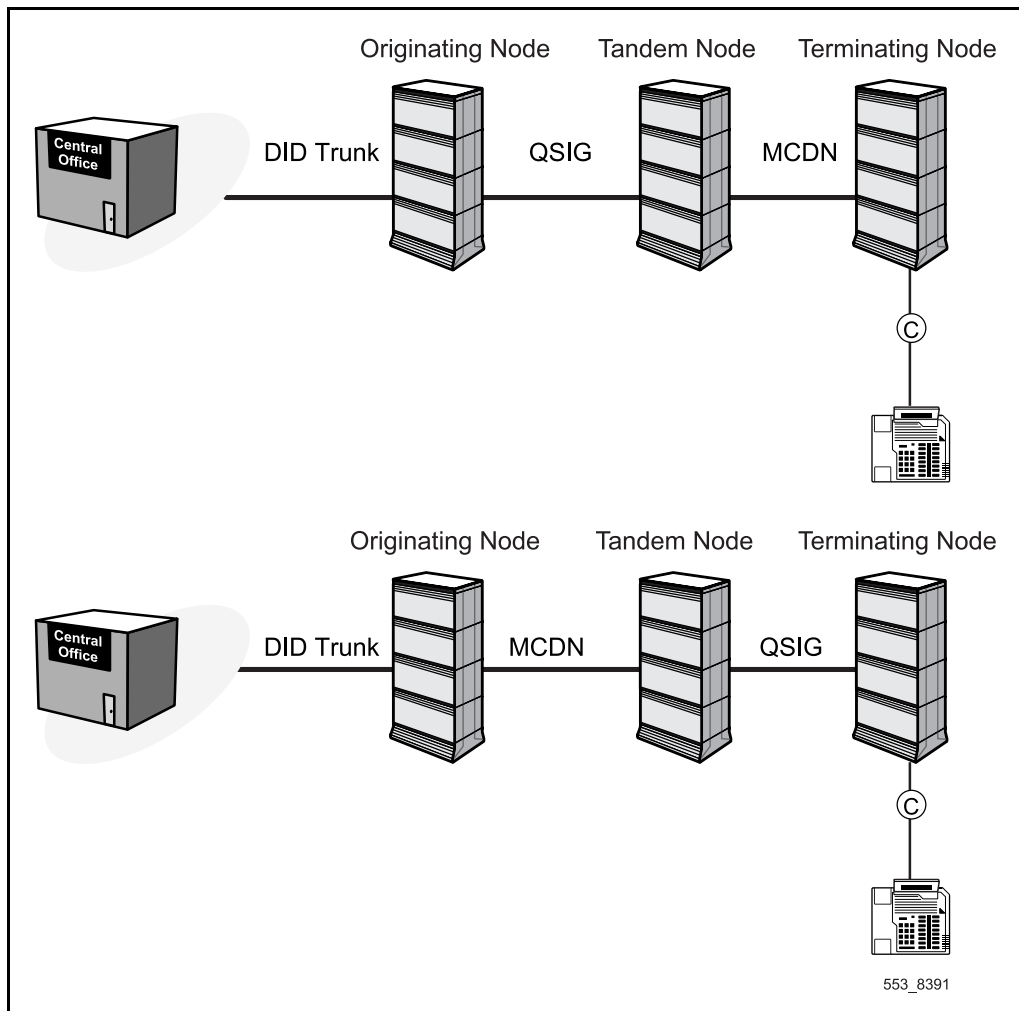


Incoming Digit Conversion (QSIG/MCDN or MCDN/QSIG link)

Referring to Figure 41, A Central Office DID call comes over a Direct Inward Dial (DID) trunk to an originating node. The call is routed to a tandem node over a QSIG link. From the tandem node, it is routed to Station C at a terminating node, over an MCDN link. Or, a Central Office DID call comes over a Direct Inward Dial (DID) trunk to an originating node. The call is routed to a tandem node over an MCDN link. From the tandem node, it is routed to Station C at a terminating node, over a QSIG link.

If no name information is received from the DID trunk, and Incoming Digit Conversion (IDC) is activated and the incoming name information is associated to the new digit sequence, then this name information is passed to Station C at the terminating node. If name information is received from the DID trunk, this takes precedence over the IDC trunk name.

Figure 41
Incoming DID Digit Conversion over a QSIG/MCDN, or MCDN/QSIG link

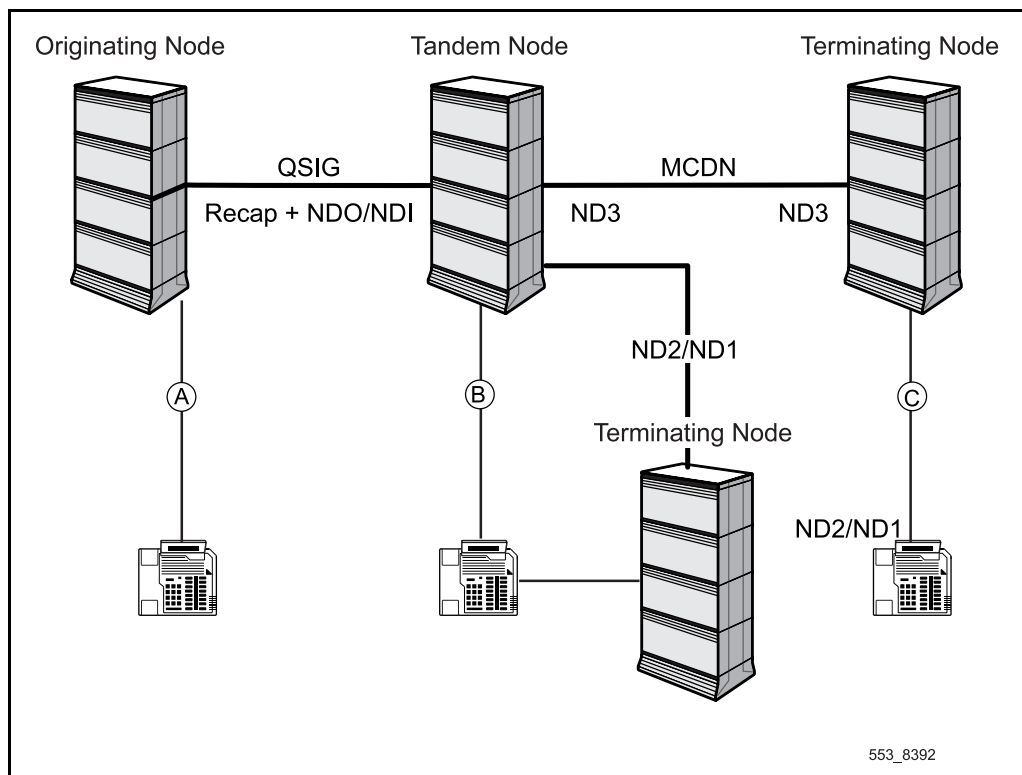


MCDN Name Display features (QSIG/MCDN link)

When a QSIG network interworks with an MCDN network, consistent interworking of the QSIG Supplementary Service - Name Display Enhancement feature and the MCDN Name Display features depends on how the Remote Capability (RCAP) has been configured for the MCDN Name Display feature.

For MCDN, the RCAP prompt is used to configure the Network Name Display (ND) method (either ND1, ND2, or ND3) for the associated D-Channel. ND3 ensures the same level of service between the MCDN Name Display and QSIG Name Display supplementary services. Refer to Figure 42.

Figure 42
MCDN Name Display features over a QSIG/MCDN link



QSIG Call Diversion Notification

Calling Name

A QSIG Call Diversion occurs to a station. After the Diversion Notification Information is received at the diverted-to node, the station's display will be updated with the calling name information, if it has a Class of Service of Dialed Name Display Denied (CLS = DNDD in LD 10 or LD 11). If the set has a Class of Service of Dialed Name Display Allowed (CLS = DNDA in LD 10 or LD 11), then another name, or none, is displayed on the set.

Alerting and Connected Name

When the names from both a QSIG Name Display Service and a QSIG Call Diversion are included in the same message, the name from the QSIG Name Display Service will be ignored (the QSIG Name Display and the QSIG Call Diversion names should be the same).

QSIG Path Replacement

For QSIG Path Replacement, the name conveyed in any message of a new call is not displayed on any set.

QSIG Call Transfer

Calling, Alerting, and Connected Name

For QSIG Call Transfer, the Calling, Alerting, and Connected Name are not provided at the transferring Private Integrated Services Network Exchange (PINX).

Connected Name

If connected name information is received, either as part of a QSIG Name Display Service or QSIG Call Transfer Notification, it will be displayed, unless there are subsequent presentation restriction options that may prevent it from being displayed.

Feature packaging

The following packages are required for the QSIG Supplementary Service - Name Display Enhancement feature:

- Q-reference Signaling Point Interface (QSIG) package 263
- QSIG Generic Functional Protocol Transport (QSIG-GF) package 305
- QSIG Supplementary Services (QSIG-SS) package 316
- Integrated Services Digital Network (ISDN) package 145
- Multi-purpose Serial Data Link (MSDL) package 222
- International Primary Rate Access (IPRA) package 202
- 2.0 Mbps Primary Rate Access (PRI2) package 154

For the QSIG Name Display service, the following packages are required:

- Digit Display (DDSP) 19
- Call Party Name Display (CPND) package 95
- International Supplementary Features (SUPP) package 131, to allow the Digit and Name Party restriction capability (without package 131, Digit and Name presentation will always be allowed)
- Calling Party Privacy (CPP) package 301
- Flexible Feature Code (FFC) package 139

For ISDN Basic Rate Interface Trunking (BRIT), the following additional packages are required:

- ISDN Basic rate Access (BRI) 216
- ISDN Basic Rate Trunk Access (BRIT) package 233

Feature operation

No operating procedures are required to use this feature.

ISDN QSIG Path Replacement

ISDN QSIG Path Replacement allows an active connection through an ISDN QSIG private network to be replaced with more efficient connection. Path replacement service is invoked by the following triggers:

- QSIG Call Diversion (QSIG Call Diversion redirects Call Forwarding Busy, Call Forwarding No Reply and Call Forwarding Unconditional calls to another set over a QSIG network)
- Network congestion on the first Private Integrated Service Network Exchange (PINX) node

Note: Private Integrated Service Network Exchange (PINX) may be used interchangeably with Private Branch Exchange (PBX).

- Detection that the connected number differs from the called number

QSIG Path Replacement does not directly attempt to reduce the number of trunks involved in a call. This feature attempts to replace non-optimum paths across a QSIG network with an optimum path. The optimum path between to PINX nodes is the path which takes the first choice route to all PINXs. The optimum path is not set up until the terminating set answers the call.

ISDN QSIG Path Replacement uses the following replacement mechanisms: triangulation and anti-tromboning. These are described in more detail beginning on page 436.

QSIG Path Replacement is permanently active on the PINX. This service is triggered internally or externally by another PINX. QSIG Path Replacement can be invoked on a mixed network of QSIG and Non-QSIG (such as MCDN). Path replacement uses the PINX_DN in the Customer Data Block as a rerouting number. Only QSIG interfaces are optimized. The Meridian Defined Customer Network (MCDN) is not optimized.

The section that follows discusses the use of Private Integrated Service Network Exchange (PINX) DNs as part of the Path Replacement feature.

ISDN PINX DN

The Path Replacement feature must scan the PINX nodes to find the optimum path to a remote PINX node. To accomplish this trunk optimization function, Path Replacement makes non-call associated requests (D-Channel messages only, no B-Channels are used) to the Path Replacement feature at the remote PINX node. The requests are not directed to any existing DN on the remote PINX node, but to a processing function on that PINX node.

To be able to route those types of calls through the network from one originating node to a destination node, an ISDN PINX DN must be defined for each PINX node in the network.

The ISDN PINX DN is a DN taken from the customer's numbering plan used to aid with the routing of network calls. It does not correspond to a real terminal on the node, so it is never busy. The ISDN PINX DN uses the same number of digits as the numbering plan; either UDP or CDP. It is defined so that the Path Replacement feature on a distant node can make non-call associated calls to the Path Replacement feature on another node when no particular destination exists. Those calls are made using existing routing configurations, e.g. a CDP distant steering code or a UDP location code expecting a fixed number of digits. For example, the far end PINX node has a digit sequence composed of the steering code plus the ISDN PINX DN with the same length and digit sequence used to route normal calls from the distant node to the local node for CDP type calls and location code plus a 4 digit ISDN PINX DN for UDP type calls.

One ISDN PINX DN should be configured per network location. This ISDN PINX DN is used for the Path Replacement feature and other features such as Electronic Lock Network Wide/Electronic Lock for Private Lines and Call Pickup Network Wide features.

No user or feature other than Path Replacement, Electronic Lock Network Wide/Electronic Lock for Private Lines and Call Pickup Network Wide features will be allowed to dial the ISDN PINX DN. If a user dials this DN, they will receive overflow tone.

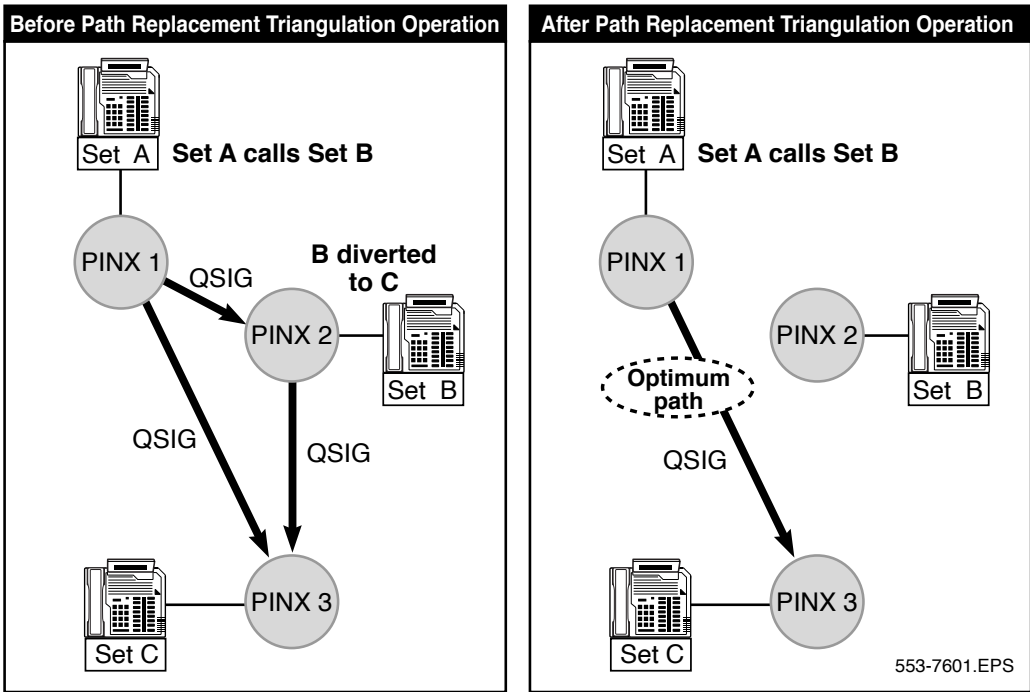
Triangulation Path Replacement

With Triangulation Path Replacement, either the originating or the terminating node involved in a QSIG call recognizes that a call is using a non-optimum path due to call diversion, network congestion or detection of a different number than dialed. If this call is still ringing, the originating/terminating node will wait for an answer signal before initiating triangulation path replacement. If the call is established, then replacement can be performed immediately.

The ISDN QSIG Path Replacement feature replaces non-optimum paths across a QSIG network with optimum paths. An optimum path between two nodes is the first choice route at all nodes. This feature does not attempt to reduce the number of trunks involved in call but does attempt to locate the optimum path.

As shown in Figure 43, Set A on PINX 1 calls Set B on PINX 2. Set B has activated Call Forward All Calls to Set C on PINX 3. Trunk optimization occurs between PINX 1 and PINX 3. Trunks between PINX 1 and PINX 2 and PINX 2 and PINX 3 are torn down after the call is answered by Set C.

Figure 43
Triangulation Path Replacement

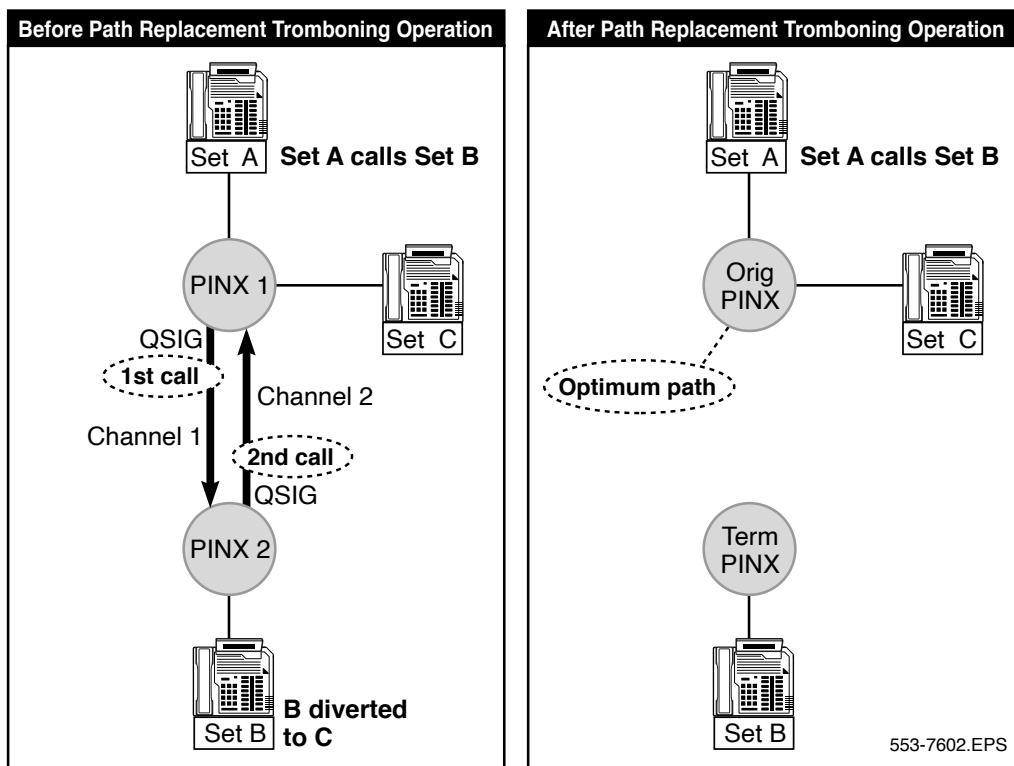


Anti -Tromboning Path Replacement

This feature performs anti-tromboning replacements to eliminate any dual parallel trunking that are caused by call modification. Anti-tromboning path replacement is only invoked after the divert-to caller answer the call. Anti-tromboning is only performed when all the trunks are associated with the same customer.

As shown in Figure 44, Set A on PINX 1 calls Set B on PINX 2. Set B has activated Call Forward All Calls to Set C on PINX 1. Anti-tromboning occurs between PINX 1 and PINX 2 and these trunks are torn down after Set C answers the call.

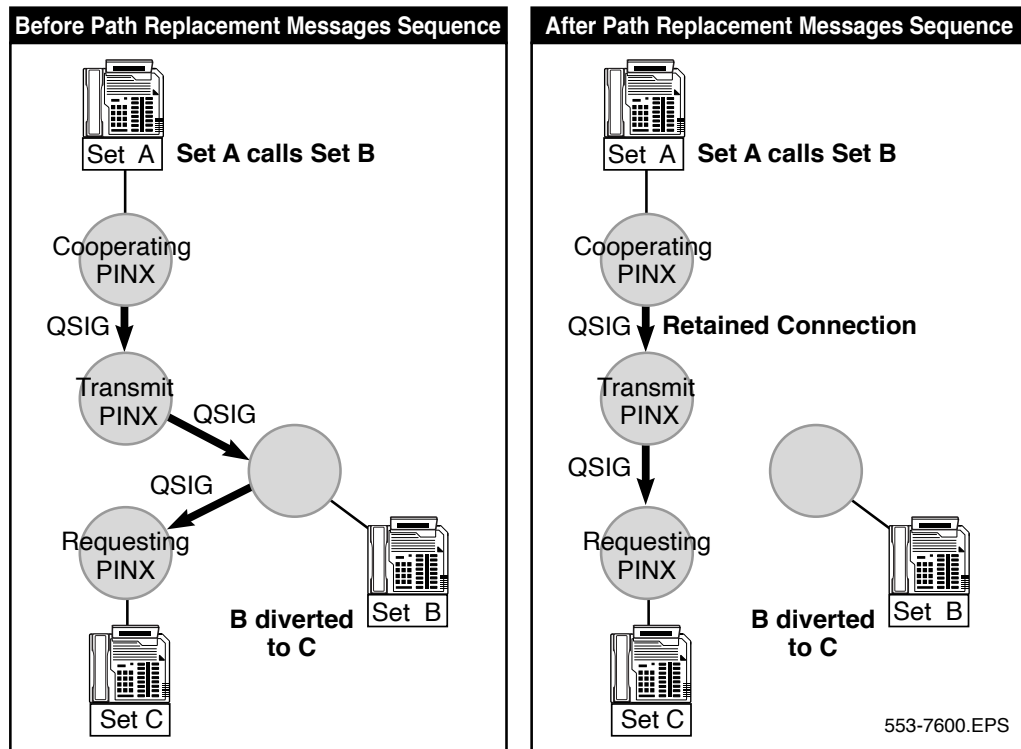
Figure 44
Tromboning Path Replacement



Retaining Path Replacement

When path replacement has been invoked, the cooperating PINX determines whether or not to retain part of the original connection provided that the original connection used the optimum route to reach the re-routed number. As shown in Figure 45, the connection between the cooperating and transit PINX is retained after path replacement has occurred.

Figure 45
Path Replacement Retaining Part of Old Connection



Operating parameters

Network congestion occurs when a non-optimum path from the first PINX node is detected.

When call diversion is not supported by a PINX node, path replacement is triggered when the connected number differs from the called party's number.

ISDN QSIG Path Replacement requires the Multi-Serial Data Link (MSDL) card.

This feature is only supported on Coordinated and Uniform Dialing Plans. Route Access Codes are not supported. Digit manipulation is not supported.

If a path replacement request is triggered and fails with the cause "invalid Rerouting Number" the request is discontinued. No attempt is made with another rerouting number.

When a call has several triggers for path replacement and the first trigger fails, no other attempts are performed using the other triggers. Path replacement triggers are chosen according to the following order:

- 1 QSIG Call Diversion
- 2 Congestion on the first PINX
- 3 Different Connected Number

The configuration of diversion and connected number triggers can generate a lot of path replacement collisions if the QSIG Call Diversion feature is configured on the QSIG network. In this situation, it is recommended that only the diversion trigger be configured. This establishes the optimized call in the same direction as the originating call.

New traffic reports are generated for QSIG Path Replacement. TSF009 reports traffic activity for D-channels and TFS011 reports traffic activity for Multi-Purpose ISDN Signalling processor for QSIG Path Replacement service.

Feature interactions

Attendant Console

Path replacement calls invoked by an attendant are not supported.

Attendant Barge In Attendant Break In

The Attendant Barge In and Attendant Break In features are not supported during path replacement operation.

Automatic Call Distribution

ISDN QSIG Path Replacement calls placed in an Automatic Call Distribution (ACD) queue are initially rejected. Path replacements requests are accepted after the call is answered by an ACD agent.

Call Transfer

When a local is transferred by the Meridian 1 during an ISDN QSIG Path Replacement request, then the replacement request is terminated. After Call Transfer, the Meridian 1 may accept path replacement from another PINX. QSIG Call Transfer is not supported at this time.

Call Hold

Path Replacement cannot be invoked on a held call. Path replacement is only accepted on an active call.

Conference Calls

The Meridian 1 does not invoke path replacement when callers are involved in an established conference. Any replacement requests from a far end PINX are rejected when a call is on an established conference.

Customer Controlled Routing

Following a Customer Controlled rerouting, path replacement is only invoked by the connected number trigger.

Direct Inward System Access

Path replacement does not occur on a Direct Inward System Access (DISA) call. The Meridian 1 does not invoke the Path Replacement feature on DISA calls.

End-to-End Signalling

When End to End Signalling is detected the path replacement attempt is terminated. However, on Analog (500/2500 type) sets, replacement is not terminated because the Digit Tone Multi-Frequency (DTMF) generated by the set is detected.

Music

Radio Paging

Recorded Announcement

The Meridian 1 does not invoke the Path Replacement feature on Music, Paged or Recorded Announcement calls.

Feature packaging

The following packages are required for ISDN QSIG Path Replacement:

- QSIG Supplementary Services (QSIG-SS) package 316
- QSIG-GF Transport (QSIGGF) package 305
- QSIG Interface (QSIG) package 263
- Network Alternate Route Selection (NARS) package 58 (for Uniform Dialing Plan), or Coordinated Dialing Plan (CDP) package 59
- Flexible Numbering Plan (FNP) package 160

2.0 Mbps Primary Rate Interface (PRI) requires:

- 2.0 Mbps Primary Rate Access (PRI2) package 154
- International Primary Rate Access (IPRA) package 202
- Multipurpose Serial Data Link (MSDL) package 222

ISDN Basic Rate Interface Trunk Access (BRIT) QSIG Interface requires:

- Basic Rate Interface (BRI) package 216
- ISDN Basic Rate Interface Trunk Access (BRIT) package 233

Feature operation

No specific operating procedures are required to use this feature.

ISDN Semi Permanent Connections for Australia

The ISDN Semi Permanent Connections (ISPC) feature allows a number of B-channels, through an Australian ISDN Central Office (CO), to be permanently connected between Meridian 1 PBXs (nailed up B-channels). These connections are used transparently by the ISDN Signaling Link (ISL) as if they were leased lines or channels on a TIE trunk.

ISL calls between PBX end users linked by the permanent connections are performed using all capabilities provided by Meridian Customer Defined Network (MCDN) features. Other possibilities provided by this feature are to use ISPC links to convey D-channel signaling between Meridian 1 PBXs.

ISL calls through an established ISPC link use two trunks. The physical trunk is the permanent connection initiated by the Central Office between both PBXs on the B-channel of a PRI2 trunk. The logical trunk is an intermediate data structure without any associated hardware. This trunk is also referred to as the Phantom DTI2 trunk.

The physical trunk allows an ISPC link to be established between the PBXs; whereas, the logical trunk allows end users to perform calls, using established ISPC links.

Semi Permanent Connection Links to convey D-Channel Signals

An ISPC link can be used to convey D-channel signaling. However, D-channel signaling requires a different configuration than the ISPC link used by the ISDN Signaling Link (ISL). Both sides of an ISPC link that convey D-channel signaling must be equipped with either a Meridian 1 Communication Adaptor (MCA) or a Meridian 1 Communication Unit (MCU) data interface.

The Dial Tone Detection feature allows the data interfaces at each end of the ISPC links to synchronize. For synchronization to occur, one side of the ISPC link must be defined as MASTER and the other side must be defined as SLAVE.

Master Side

The data interface, connected to the D-channel, must be configured in autodialing mode. The dialing sequence configured on the MASTER SIDE, rings the data interface on the SLAVE SIDE of the ISPC link through the associated route of the Phantom TN. The Dial Tone Detection feature must be configured for this route so that dial tone is provided by the SLAVE SIDE of the ISPC link. The dial tone indicates to the MASTER SIDE that both legs of the ISPC link have been successfully setup by the Central Office.

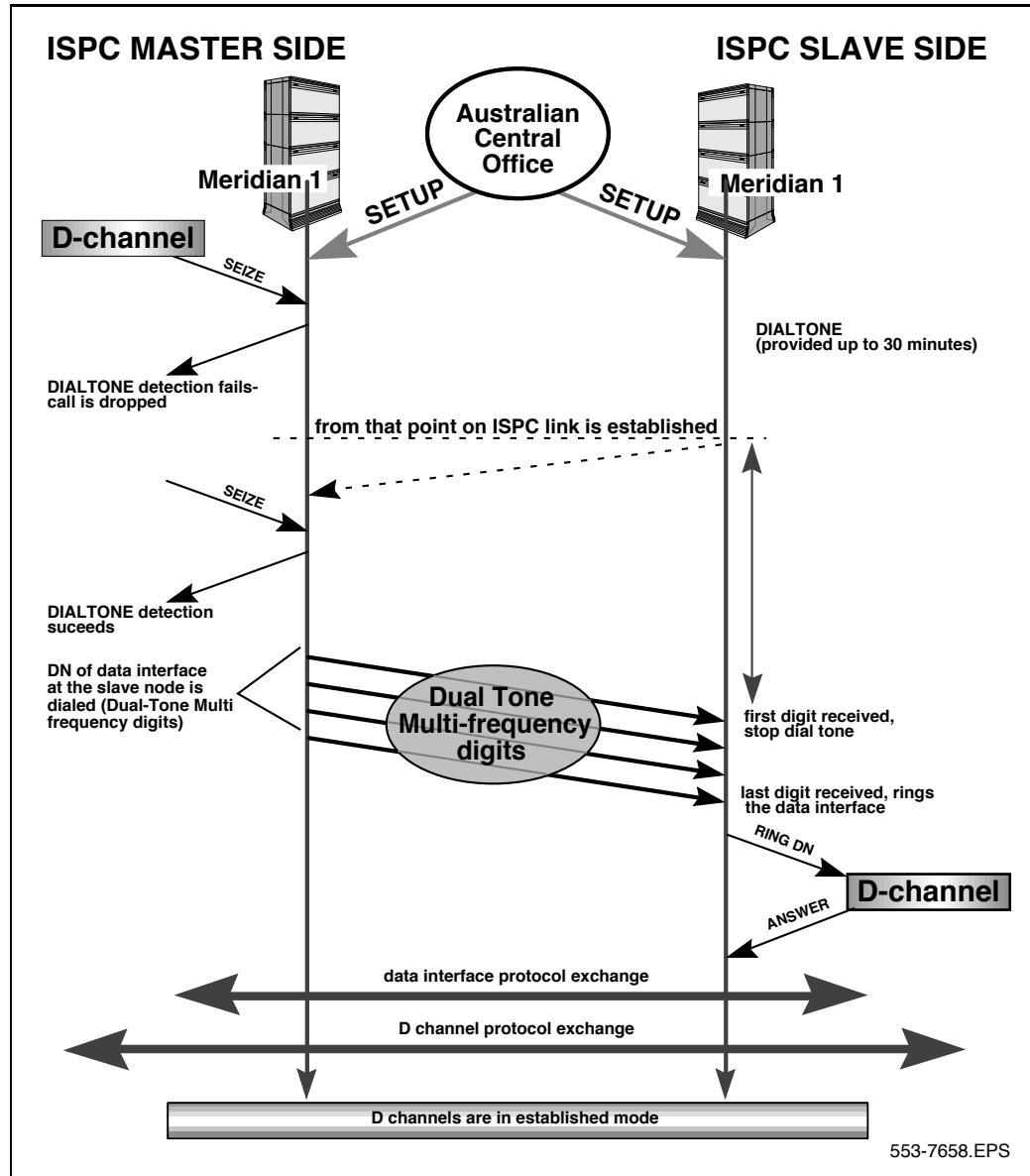
As illustrated in Figure 46, when the data interface initiates its dialing sequence, a path is set between the interface and the ISPC link. If no dial tone is detected, the path is released. Additional attempts are performed by the data interface because it has been configured in autodialing mode. When the dialing tone is successfully detected, the Directory Number associated to the data interface used on the ISPC SLAVE SIDE, is dialed through the ISPC link using Dual Tone Multi-Frequency (DTMF) tones.

Slave Side

The data interface connected to the SLAVE SIDE must be configured to autoanswer mode. When an ISPC link is established, a path is made with a TDS card to provide Dial Tone to the ISPC link, and DTMF digits are expected. The DTMF digits identify the Directory Number where the data interface that is to ring is connected. When the data interface on the SLAVE SIDE is rung, it answers and initiates its own protocol sequence and D-channels can be established.

After 30 minutes of unsuccessful attempts to connect, the path between the ISPC link, the TDS card and the DTR cards is removed, provided that the default mode is set to “MANUAL and 1 attempt”. A technician can change these default parameters in LD 96.

Figure 46
Mechanism Used to Synchronize Data Interfaces



Operating parameters

For this feature, Meridian 1 users must initially subscribe for ISPC services from Telecom Australia. ISDN Semi Permanent Connections (ISPC) service is only available from Telecom Australia on Central Office (CO) and Direct Inward Dialing (DID) trunks.

The ISPC feature is based on the Asia Pacific ISDN protocol for Australia. The ISPC requires PRI2 connections to an Australian Central Office using Asia-Pacific ISDN connectivity for Australia.

ISPC is not supported over Basic Rate Interface (BRI) trunks or AXE-10 Australia Primary Rate Interface connectivity.

Data interfaces, supported when the ISPC links are configured to convey D-channel signaling to forward signals from MSDL, DCHI or DDCHI card ports into the Meridian 1, are Meridian Communication Unit (MCU) and Meridian Communication Adapter (MCA).

Integrated Signaling Links (ISL) in a D-channel configured shared or dedicated mode are supported.

ISPC reference numbers, provided by Telecom Australia, are limited to a maximum of seven digits.

Established ISPC links are only used in conjunction with the ISL feature functionality. Therefore, ISPC links are interpreted as available TIE trunks with an established ISL D-channel.

Outband signaling, through an ISL D-channel, is required to control the usage of established ISPC links. When an ISPC link conveys D-channel signaling, a special mechanism based on the Dial Tone Detection feature is used.

The number of established ISPC links on a PRI2 loop is limited to the number of configured links and the idle B-channels on the loop.

Each ISPC link requires an associated TN defined on a Phantom DTI2 loop.

Established ISPC links cannot be used for Virtual Network Services (VNS) bearer trunks.

ISPC links are established by the Australian Central Office (CO) as a 64kbps cleared data link. The CO and Meridian 1 PBXs do not provide a mechanism for echo cancellation for ISL calls using ISPC links.

The Phantom DTI2 trunks used by this feature must belong to the same group as the real PRI2 trunk.

When establishing an ISL D-channel between two Meridian 1 PBXs, the following hardware is required:

- D-channel (DCH), Dual Density D-channel (DDCHI) or Multi-Serial Data Link (MSDL) card on each Meridian 1 site;
- Meridian Communication Unit (MCU) or Meridian Communication Adapter (MCA) on each site for the exchange of D-channel signaling between sites; and
- Tone Detector Card (XTD or DTD) on one site and a Tone Generation Card (XCT or TDS) at the other site in situations where at least one ISPC link is used to convey D-channel signaling.

Feature interactions

Attendant Barge-In

When an attendant attempts to Barge-In on 2.0 Mbps Primary Rate Interface B-channel used as an ISPC link with the Central Office, a fast busy tone is provided.

Attendant Incoming Call Indicators

Calls using an ISPC link are always presented as calls over TIE trunks.

Call Detail Recording

Calls using an established ISPC link are identified by the calling party, called party, time and duration of call. Call Detail Recording tickets are not generated for the ISPC links that have been established.

Call Forward by Call Type

ISPC calls are handled 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.

Dial Tone Detection

To convey D-channel signaling over an ISPC link, the route associated to the link at the Meridian 1 configured as MASTER must detect a dialtone.

Distinctive New Distinctive Ringing

For 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.

Integrated Service Digital Network Signaling Link

Redirection or optimization of calls will not release established ISPC links.

Trunk Barring

For calls using or requesting an ISPC link, Trunk Barring 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.

Feature packaging

ISDN Semi Permanent Connection for Australia requires the following packages:

- ISDN Semi Permanent Connection for Australia (ISPC) package 313;
- ISDN Signaling Link Package (ISL) package 145; and
- 2.0 Mbit Digital Trunk Interface (DTI2) package 129.

The following packages are required for Asia-Pacific ISDN Primary Rate:

- International PRA (IPRA) package 202;
- Integrated Digital Services Network (ISDN) package 145;
- 2.0 Mbit Primary Rate Interface (PRI2) package 154;
- Intelligent Peripheral Equipment (XPE) package 203; and
- Multi-Purpose Serial Data Link (MSDL) package 222

The following packages are required for an ISPC link to convey D-channel signaling:

- Dial Tone Detector (DTD) package 138 (for MASTER nodes only); and

- International Supplementary Features (SUPP) package 131 (for SLAVE nodes only).

Feature operation

No specific operating procedures are required to use this feature. However, the following steps must be performed to operate ISDN Semi Permanent Connections for Australia.

- 1** Subscribe for ISDN Semi Permanent Connections (ISPC) service from Telecom entity.
- 2** Install hardware for ISPC links usage (if required).
- 3** Configure Meridian 1 PBXs for ISPC links usage.
- 4** Wait for ISPC links to be established.
- 5** Wait for ISL D-channel to be established (if using).
- 6** Perform call via established ISPC links.

ISDN Signaling Link (ISL)

Description

This feature provides out of band ISDN D-channel signaling capability to enhance both digital and analog conventional in-band trunk signaling.

This package allows for the configuration of the NTAK02 as an ISDN Signaling Link (ISL), and for the transport of CLID and ESN/ISDN features.

This link provides ISDN Feature Networking Services to locations in a customer's network where a DTI/PRI link is not available or not economically justified.

The ISL uses a dedicated channel at speeds between 2.4 to 64 Kbps to establish the ISDN "D" channel for transmission of the ISDN Q.931 signaling protocol. The D-channel information is transported to the far end using one of the following methods:

- a PRI channel in the case of ISL shared mode (in the same manner as PRI)
- a data adapter or modem in the case of ISL dedicated mode

The associated voice or data is sent over the existing conventional analog or digital trunks.

Note: Auto recovery for ISL tie trunks to revert back to in-band signaling upon failure of the D-channel Handler Interface is available in Meridian 1 systems, as long as the connected system is X11 Release 14 or later.

Interactions and Limitations

TIE trunk types (including DTI) are supported for Meridian 1 to Meridian 1 connectivity only.

Package Requirements

This option requires the following feature packages:

Feature	Package Number
ISDN	145
ISL	147

Japan D70 nB+D

This feature is an enhancement of the INS1500 Japan D70 interface introduced in Generic X11 Release 17, providing nB+D ISDN Primary Rate connectivity between Meridian 1 and the INS1500 D70 for Japan. The capability of a D-channel is expanded to support multiple PRIs (up to nine), in an nB+D configuration, where $1 \leq n \leq 215$. This enhancement allows for non facility associated D-channels, so that a PRI might consist of 24 B-channels, and the D-channel controlling these B-channels can be shared with another PRI link. The bit rate is 1.544 Mbits.

The nB+D enhancement adheres to current trunk assignment limitations on the number of trunks that can be assigned to an individual trunk group.

Operating parameters

For the ISDN layer 1 interface, the circuit pack QPC720 (Digital Trunk Interface) is used for the Meridian 1 to D70 connectivity.

The QPC757E version of the D-channel Handler (DCH) circuit pack provides the Layer 2 functions and incoming Layer 3 preprocessing for the Meridian 1 to D70 connectivity.

The NTAK09AA (DTI/PRI) circuit pack is used for Option 11 systems, along with the NTAK93 D-channel Interface (DCHI) and NTAK20 Clock Controller.

A Multi-purpose Serial Data Link (MSDL) can be used in place of the DCH circuit pack (NT6D80AA).

Feature interactions

PRI Channel Negotiation

When a D-channel has one or more secondary PRIs associated with it while using the D70 interface, channel negotiation requires that the PRI interface be explicitly defined for all B-channels not on the primary interface.

Japan D70 Connectivity

Interface Identifiers are required on all PRIs, including the primary PRI. This interface is used in channel negotiation.

Feature packaging

For Japan D70 ISDN PRI connectivity, the following software packages are required: 1.5 Mbit Digital Trunk Interface (PBXI) package 75; Integrated Services Digital Network (ISDN) package 145; Primary Rate Access (PRA) package 146; International Primary Rate Access (IPRA) package 202; Digit Display (DDSP) package 19 for Calling Line Identification on ISDN PRI; and the NTBK51 Downloadable D-Channel handler card in place of the DCH card.

All of the software packages required for Japan D70 ISDN PRI connectivity are used to configure Japan D70 ISDN nB+D PRI, with the following addition: International nB+D (INDB) package 255.

Japan TTC Common Channel Signaling

The Japan Telecommunication Technology Committee (JTTC) Common Channel Signaling is the Japanese version of the International Standard Organization (ISO) ISDN QSIG. It specifies the Layer 3 protocol signaling requirement for support of circuit switched call control at the “Q” reference point between a Private Integrated Services Digital Network (PISN) and a Private Telecommunication Network (PTN).

The JTTC Common Channel Signaling feature provides basic call service on Meridian 1 ISDN 1.5 Mbps PRI on TTC connectivity. It also supports other supplementary services.

This feature supports the following call services on the Meridian 1 JTTC interface:

- Basic Call Service
- 64K Clear Bearer capability
- Calling Line Identification Presentation/Calling Line Identification Restriction (CLIP/CLIR)
- Channel Negotiation
- Enbloc Dialing
- User to User Information Element Transparent Transport

- TIE call types
- Traveling Class Mark (TCM) Transparent Transport

Operating parameters

Basic Rate Interface (BRI) is not defined on the JTTC interface.

Connected Line Identification Presentation/Restriction (COLP/COLR) is not defined to JTTC standard.

Backup D Channel, nB+D, Overlap Signaling (Overlap sending/Overlap receiving), Party Category, and Transit Counter are not supported on the JTTC interface.

ISDN QSIG Generic Functional Transport, Network Automatic Call Distribution (NACD), Network Call Trace (NCT), and Remote Virtual Queueing (RVQ) are not supported on the JTTC interface.

Only circuit mode connection is supported on the JTTC interface.

MCDN/JTTC gateway supports only Basic Call and CLIP/CLIR.

For Option 11C, ISDN Signaling Link (ISL) is not supported on the JTTC interface.

The Japan TTC Common Channel Signaling feature requires the following hardware:

- 1.5 Mbps Primary Rate Interface (PRI)/Digital Trunk Interface (DTI) card (NTAK09)
- Downloadable D-Channel Daughterboard (NTBK51)
- Clock Controller (NTAK20)

Feature interactions

Networking Features

Some networking features currently exist on more than one ISDN interface on the Meridian 1 system. These features are listed in Table 1. The columns list the services and the rows list the interfaces.

Networking features that do not appear in Table 35 are only supported on one ISDN interface and are, therefore, rejected by all JTTC gateways when the service is requested. This is the case for all MCDN features that are not supported over the JTTC interface.

In Table 1 when a service is supported (Y), the information related to this service is accepted, decoded, and used according to the service description. When a service is not supported (N), the information related to this service is not sent to the interface. The request for the service is rejected according to the service rejection procedures. When a service is supported on two interfaces, a gateway function exists. This passes the information in order to support the service from one interface to the other.

Table 35
Networking features that exist on more than one ISDN interface implemented on Meridian 1

	JTTC	ETSI QSIG	ISO QSIG	Euro-ISDN	MCDN	ETSI BRI sets	Other BRI sets	Analog	D70
Calling Line ID/Calling Party Subaddress	Y	Y	Y	Y	Y	Y	Y	N	Y
Transit Counter	N	Y	N	N	Y	N	N	N	N
Call Charge	N	N	N	Y	N	N	N	Y	N

ISDN Signaling Link

The existing Integrated Signaling Link (ISL) operation is supported on the JTTC interface on the JDMI 2 Mbps interface only.

Only the first 23 TIE trunks of the JDMI loop are configurable for ISL operation.

Network Attendant Service

JTTC interacts with Network Attendant Service (NAS) as though the call is being sent to a route without NAS being equipped.

Network Call Redirection

The existing Network Call Redirection limitation on unsupported interfaces applies to the JTTC interface.

When a call is terminated on the Meridian 1 and Network Call Redirection is active, Japan TTC Common Channel Signaling can still operate. However, the original called number and the redirection number IEs that are used by the Network Call Redirection feature are not sent on the JTTC interface.

Network Calling Party Name Display

The Network Calling Party Name Display (NCPND) feature is supported within an MCDN network only. When the JTTC interface is involved in the call setup, the existing NCPND operation on unsupported interfaces applies to the JTTC interface also.

Network Message Service

Network Message Service (NMS) is supported within an MCDN network only. The NMS operation on the JTTC interface is the same as that of the existing treatment for unsupported interfaces.

Network Ring Again

Network Ring Again (NRAG) signaling is supported within a Meridian Customer Defined Network (MCDN) only. When a user requests NRAG on the JTTC interface, the request is rejected.

Trunk Route Optimization

Trunk Route Optimization (TRO) is supported within an MCDN network only. When a redirecting node sends a message to the originating node and the TRO request is accepted, the new call goes through the JTTC interface as a normal basic call. However, TRO signaling does not operate on the JTTC interface.

Virtual Network Services (VNS)

A JTTC link can be used as a B-Channel for Virtual Network Service (VNS) over a private network. All VNS services are supported normally. JTTC is used only as a speech bearer.

Feature packaging

Japan TTC Common Channel Signaling is Japan TTC (JTTC) package 335. The following packages are also required for JTTC:

- Integrated Services Digital Network (ISDN) package 145
- Primary Rate Access (PRA) package 146
- Japan Digital Multiplex Interface (JDMI) package 136 for ISL functionality
- International Primary Rate Access (IPRA) package 202
- Multi-purpose Serial Data Link (MSDL) package 222

Feature operation

No specific operating procedures are required to use this feature.

MCDN Alternate Routing

The MCDN (Meridian Customer Defined Network) Alternate Routing feature provides a solution to calls encountering congestion due to high traffic situations within an MCDN network. The MCDN Alternate Routing feature is supported over an MCDN-based ISDN protocol.

The MCDN Alternate Routing feature uses the routing capability of Network Alternate Route Selection (NARS) to re-route a congested call. For each MCDN call translated at a Meridian 1 node, NARS selects one route from up to 512 routes to complete the call. These routes are programmed in a route list. Each route in the list is called an entry. There may be up to 64 entries in each route list. Any combination of trunks can be specified in a route list.

The MCDN Alternate Routing feature may be configured for each of the 512 different routes.

Congestion occurs when channels within the network are not available. With the introduction of the MCDN Alternate Routing feature, each entry of a route list on one node can be configured to take an alternate entry (route) from the route list of that node (Private Exchange or Central Office), if congestion is encountered.

Alternate routing configuration takes place in LD 86, the Electronic Switched Network (ESN) administration overlay. The option defined for the Step Back On Congestion (SBOC) prompt determines the type of alternate routing available to calls over a particular route.

Alternate routing options are:

- NRR: no alternate routing is performed. The call receives congestion treatment.
- RRO: re-route at the originating node if a call encounters congestion. If congestion is encountered at a transit node, the call drops back to the preceding node, so that the preceding node decides if re-routing is required.
- RRA: re-route the call at any node, whether originating or transit node, when congestion is encountered.

The MCDN Alternate Routing feature is triggered at the controlling node when a Call Clearing message (DISCONNECT or RELEASE COMPLETE) is received, and the cause value in the Call Clearing message is supported for the MCDN Alternate Routing feature. The cause values which activate the MCDN Alternate Routing feature are:

- Cause 3 = No route to destination
- Cause 27 = Destination is out of service
- Cause 34 = No channel/circuit available
- Cause 38 = Network out of order
- Cause 41 = Temporary failure
- Cause 42 = Congestion

To ensure that MCDN Alternate Routing occurs for all nodes in a system, define all nodes as SBOC = RRA. Conversely, to prevent MCDN Alternate Routing from occurring on any nodes, define all nodes as SBOC = NRR.

Note: If any of the cause values listed below are received in a Call Clearing message, the non-MCDN Alternate Routing retry operation occurs. That is, an attempt is made to retry the call. The retry first attempts to find another idle channel in the same route. If all the channels in that route are busy, it attempts to find another channel in the next alternate route in the Route List Block.

- Cause 6 = Channel unacceptable
- Cause 44 = Requested circuit/channel not available
- Cause 82 = Identified channel does not exist

Refer to Figure 47, Originating node operation. If Node C sends one of the non-MCDN Alternate Routing cause values, the Call Clearing message is sent back to Node A which attempts non-MCDN Alternate Routing.

Refer to Figure 48, Transit node operation. If Node C receives one of the non-MCDN Alternate Routing cause values:

- Node C attempts to find an alternate route but none is available.
- Node B is a transit node and does not attempt to find an alternate route.
- Node A attempts to find an alternate route but none is available.

Originating node operation

Consider the following calling scenario for an originating node operation. An attempt is being made to establish a call over an MCDN link, from originating node A to terminating node D, through transit nodes B and C. Congestion occurs between nodes C and D.

Whether node C is defined as SBOC = RRA or RRO or NRR, the congestion message, along with a MCDN Alternate Routing feature cause value, is always sent back to node B.

At node B, the SBOC option is checked to determine the routing treatment:

- If **node B is defined as SBOC = NRR**, the Call Clearing message is sent back to the preceding node, A.

- If **node B is defined as SBOC = RRO**, the Call Clearing message is sent back to node A because A is the originating node.
- If **node B is defined as SBOC = RRA**, an attempt is made to find an available alternate route until all of the alternate routes, as defined in the Route List (the RLB prompt in LD 86) are tried. If no alternate route is available, the congestion message information is sent from node B to the preceding node A.

In the originating node operation, node A must be defined as SBOC = RRA or RRO to perform MCDN alternate routing from node A to E to D.

Figure 47 illustrates an originating node operation. With congestion between nodes C and D, the Call Clearing message is sent from node C to B. Because node B is defined as SBOC = RRO or NRR, the Call Clearing message is sent back to node A. With node A defined as RRA or RRO, an attempt is made to find an available alternate route. The first available alternate route is found between nodes A and E. Node E can either be a Private Exchange or Central Office. The Network Class of Service access checks are passed, the direct leg between the congested node, C, is released, and an alternate route is created from node A through node E. The call is then rerouted from nodes A and E to the terminating node, D.

Transit node operation

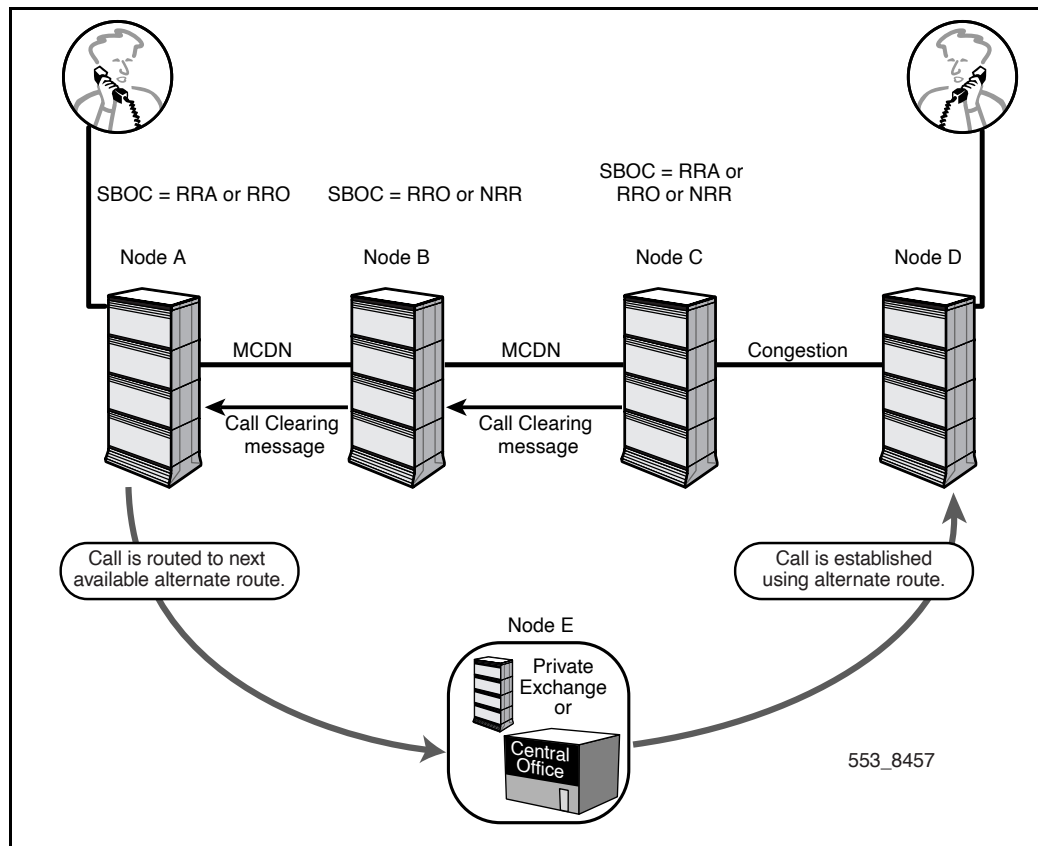
Consider the following calling scenario for a transit node operation. An attempt is being made to establish a call over an MCDN link, from originating node A to terminating node D, through transit nodes B and C. Congestion occurs between nodes C and D.

Whether node C is defined as SBOC = RRA or RRO or NRR, the congestion message, along with a MCDN Alternate Routing feature cause value, is always sent back to node B.

At node B, the SBOC option is checked to determine the routing treatment:

- If **node B is defined as SBOC = NRR**, no rerouting occurs; the Call Clearing message is sent back to the preceding node, A.

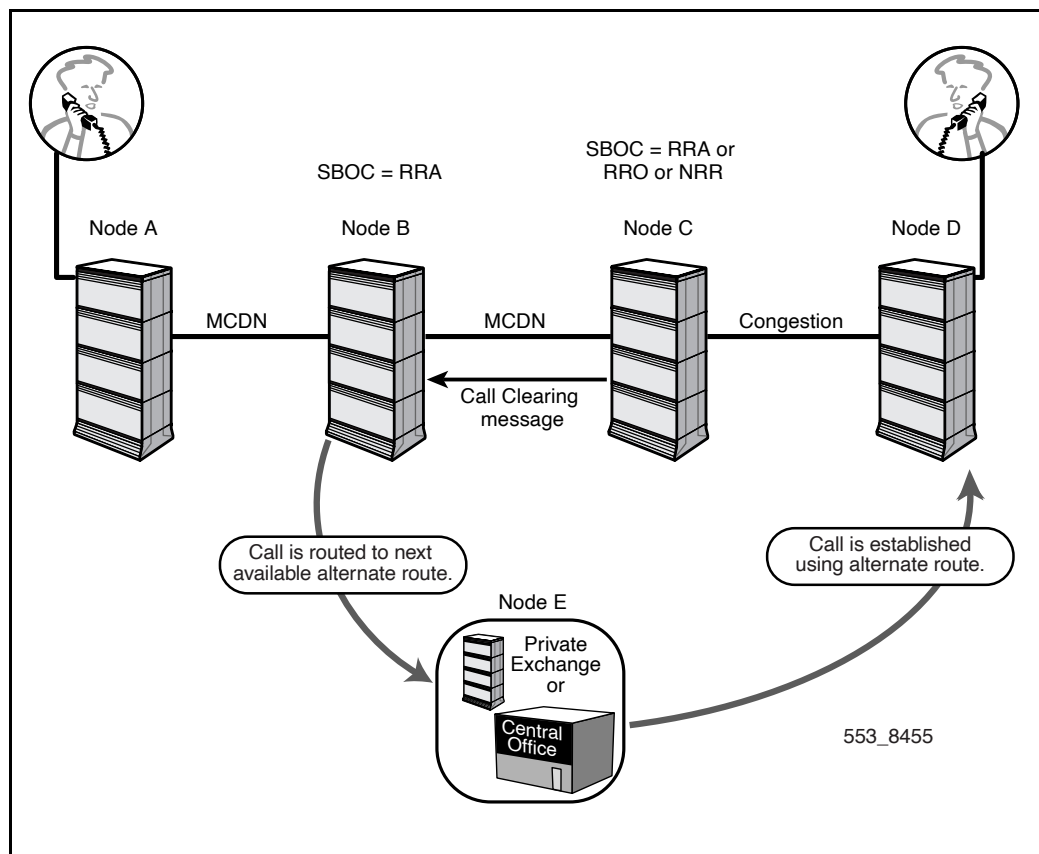
Figure 47
Originating node operation



- If **node B is defined as SBOC = RRO**, the Call Clearing message is sent back to the preceding node, A, because node B is not the originating node.
- If **node B is defined as SBOC = RRA**, an attempt is made to find an available alternate route until all of the alternate routes, as defined by Route List (the RLB prompt in LD 86) are tried.

Figure 48 illustrates a transit node operation with node B defined as SBOC = RRA. The first available alternate route is found between nodes B and E. Node E can either be a Private Exchange or Central Office. The Network Class of Service access checks are passed, the direct leg between the congested node (node D) is released, and an alternate route is created from node B through node E to the terminating node, D.

Figure 48
Transit node RRA operation

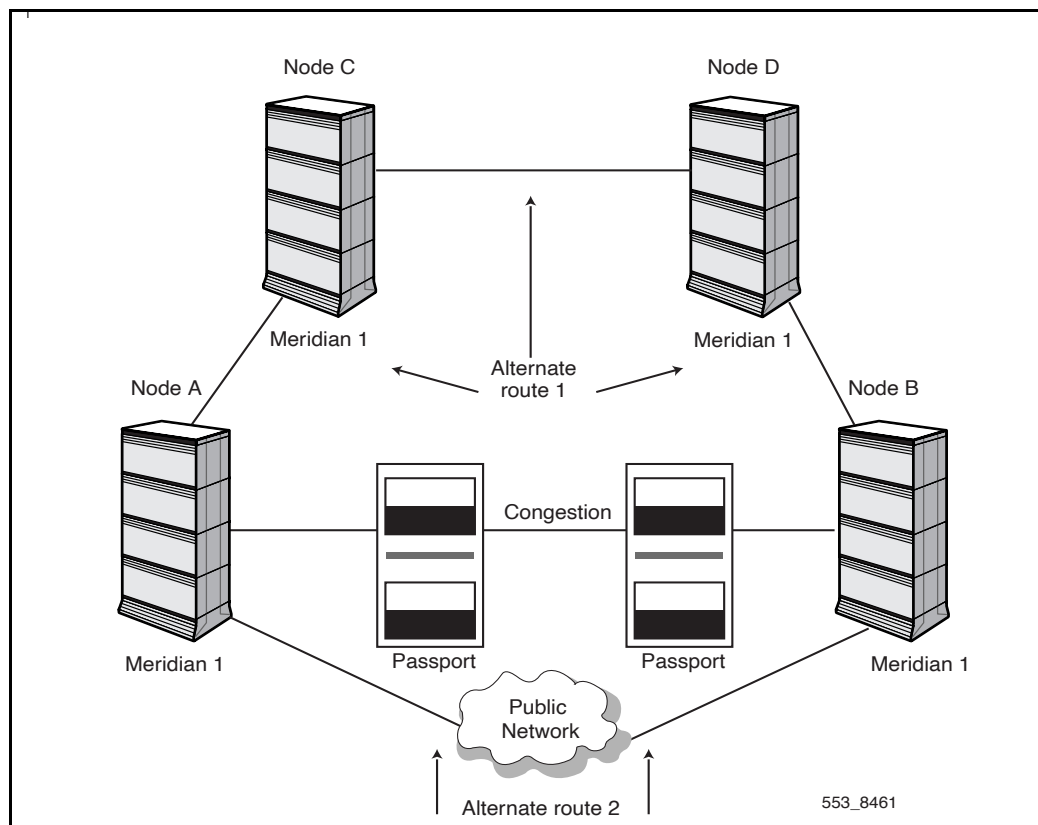


MCDN Alternate Routing examples

Example 1: Meridian 1 networked with Passport

In this scenario (refer to Figure 49), node A attempts to call node B using a primary route through Passport. The primary route is congested. Passport sends the cause value in the Call Clearing message to node A. With node A defined as SBOC = RRA or RRO, the cause value triggers a retry to make the call through the next alternate route. Alternate route 1 is from node A to node B through nodes C and D. If alternate route 1 is unavailable, then an attempt is made to route the call through alternate route 2: node A to node B through the Public Network.

Figure 49
Example of Meridian 1 with Passport network

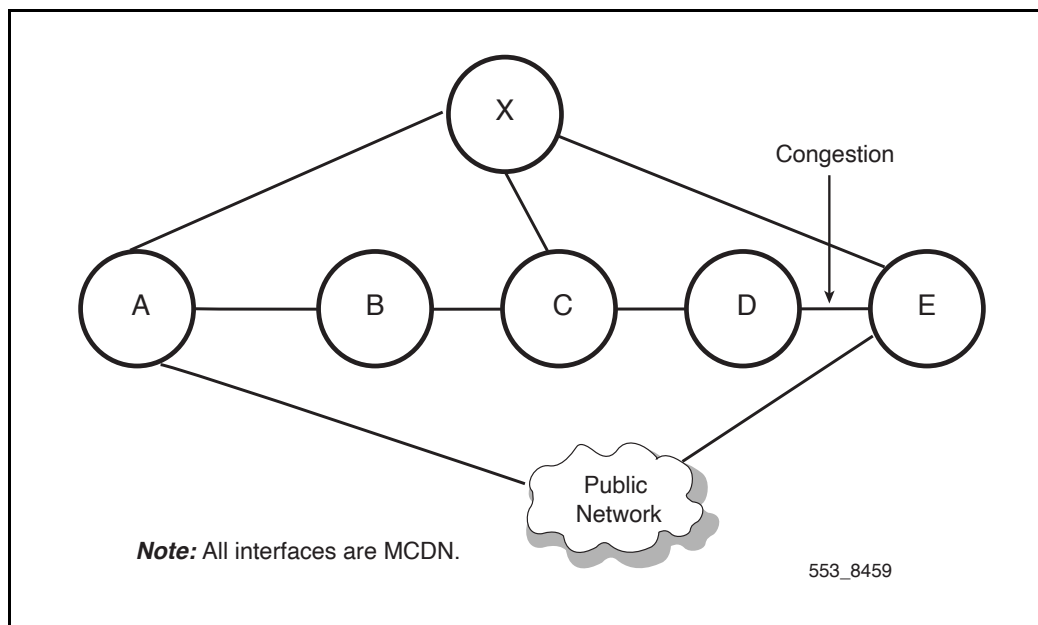


Example 2: Multi node network

Consider a scenario (refer to Figure 50) where a call is made from node A to node E through nodes B, C, and D. If there is congestion between nodes D and E, the Call Clearing message is sent from node D to node C:

- If node C is defined as SBOC = RRA, an attempt is made to find an alternate route. An alternate route is found from node C through node X to node E.
- If node C is defined as SBOC = RRO or NRR, the Call Clearing message is sent back to node B. Whether node B is defined as SBOC = NRR or RRO or RRA, the Call Clearing message is sent back to node A. Whether node A is defined as SBOC = RRA or SBOC = RRO, an attempt is made to find an alternate route. Alternate routes in this case are either from node A through node X to node E or from node A to node E through the Public Network.

Figure 50
Example of multi node network



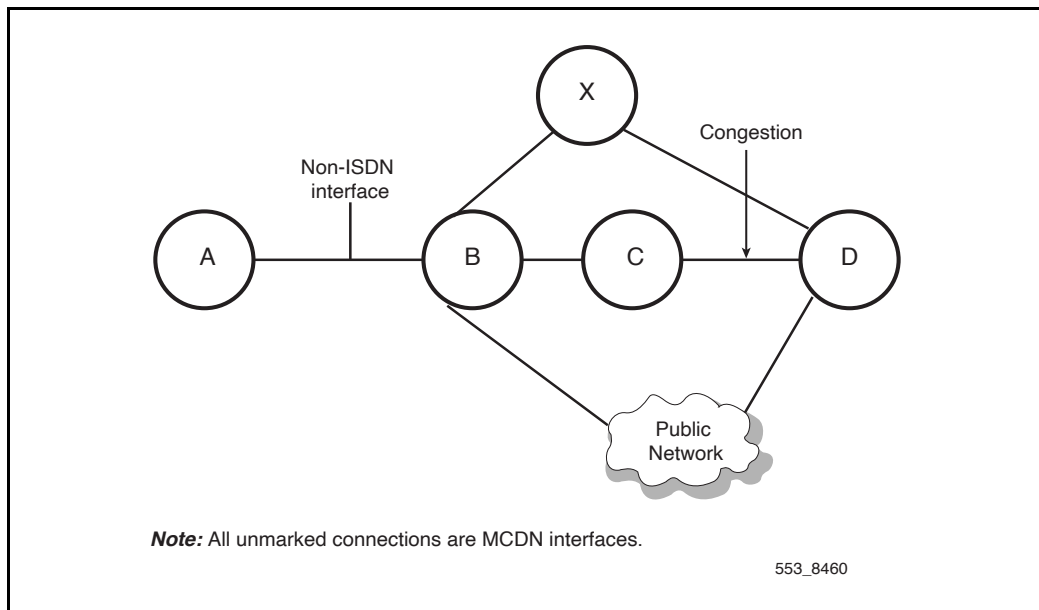
Example 3: Tandem call with non-ISDN originating node

This scenario (refer to Figure 5) shows a tandem call with a non-ISDN originating node. When a call is made from node A to node D, the call is routed through nodes B and C. Congestion is encountered between nodes C and D.

Whether node C is defined as SBOC = RRA or RRO or NRR, the Call Clearing message is always sent back to node B.

- If node B is defined as SBOC = RRA, an attempt is made to find an alternate route. The alternate route through node X or through the Public Network to terminating node D is used to route the call.
- If node B is defined as SBOC = RRO or NRR, no alternate routing is attempted because the link between nodes B and A is non-ISDN. The call is cleared according to non-MCDN Alternate Routing operations.

Figure 51
Example of a tandem call with non-ISDN originating node



Operating parameters

Route Access codes are not supported for MCDN Alternate Routing. The Coordinated Dialing Plan (CDP) or the Uniform Dialing Plan (UDP) is required for MCDN Alternate Routing, but not both. If UDP is used, Network Alternate Route Selection (NARS) or Basic Alternate Route Selection (BARS) must be provisioned.

The MCDN Alternate Routing feature does not support MCDN ISDN BRI trunks.

Feature interactions

Automatic Least Cost Routing

When a Call Clearing message is received, the MCDN Alternate Routing feature attempts a retry through the next alternative route, whether the alternative route is an Initial Set (ISET) or an Extend Set (ESET) route.

Drop Back Busy

If both the MCDN Alternate Routing feature and Drop Back Busy feature are configured in the same MCDN network, the MCDN Alternate Routing feature takes precedence.

Expensive Route Warning Tone

The Expensive Route Warning Tone is provided at the originating node when the MCDN Alternate Routing attempts to re-route the call through an expensive route. The Expensive Route Warning Tone is **not** provided when re-route to an expensive route occurs at a transit node.

Intercept treatment

The intercept treatment for network blocking is not applied at the transit node if the MCDN Alternate Routing feature is active at the transit node. The call is dropped back with the appropriate congestion IE information.

If the MCDN Alternate Routing feature fails to find an alternate route for a call encountering congestion at a transit node, intercept treatment is not applied at the transit node. The call is dropped back to the originating node with the appropriate congestion IE information.

Integrated Services Access

The MCDN Alternate Routing feature is supported for MCDN Integrated Services Access (ISA) routes.

ISDN Signaling Link

The MCDN Alternate Routing feature is supported for MCDN D-Channel ISDN Signaling Link (ISL) routes.

Network Attendant Service

The Network Attendant Service (NAS) feature operation is transparent to the MCDN Alternate Routing. The NAS drop back function takes priority over the MCDN Alternate Routing drop back function.

Off-Hook Queuing

Off-Hook Queuing (OHQ) takes precedence over the MCDN Alternate Routing feature. At the node where congestion is first encountered, if all outgoing routes are busy, the call is cleared back immediately to the preceding node only if there are no queuing features at this node.

The MCDN Alternate Routing feature comes into operation only at the preceding node when a Call Clearing message is received. OHQ waits until the busy route becomes available. If this route does not become available before OHQ times out, a Call Clearing message is sent.

Overlap Signaling

The MCDN Alternate Routing feature is supported over both the enbloc and overlap signaling methods.

Remote Virtual Queuing

The MCDN Alternate Routing feature takes precedence over Remote Virtual Queuing (RVQ) when congestion is encountered at the tandem node: an attempt is made to find an alternative route instead of informing the call originator to activate Ring Again Allowed.

Trunk Barring

If trunk barring prevents any trunk to trunk connection at the tandem node, the MCDN Alternate Routing feature retries on the next available route in the Route List Index (RLI).

Virtual Network Services

The MCDN Alternate Routing feature can function over all MCDN-based bearer interfaces supporting Virtual Network Services (VNS).

Feature packaging

The MCDN Alternate Routing feature requires the following packages:

- Basic Routing (BRTE) package 14
- Network Class of Service (NCOS) package 32
- Coordinated Dialing Plan (CDP) package 59 **or** Uniform Dialing Plan (UDP)
 - UDP requires either Network Alternate Route Selection (NARS) package 58 or Basic Alternate Route Selection (BARS) package 57
- Digital Trunk Interface (DTI) package 75

For an ISDN Primary Rate Interface (PRI), the following package is required:

- ISDN Signaling (ISDN) package 145
- 2.0 Mbit/s Primary Rate Interface (PRI2) package 154 **or**
- International Primary Rate Interface (IPRA) package 202 **or**
- ISDN Signaling Link Interface (ISL) package 147

Feature operation

There are no operating procedures required to use this feature.

MCDN End to End Transparency

Meridian Customer Defined Network (MCDN) services are based on propriety specific Integrated Services Digital Network (ISDN) signaling. The MCDN End to End Transparency feature conveys MCDN proprietary services on a standardized interface, ISDN QSIG. The QSIG gateway supports MCDN features both in a QSIG network and a mixed MCDN/QSIG network. It is the MCDN QSIG conversion tool which provides the basis for support of the selected MCDN applications such as Network Attendant Service (NAS), Network Automatic Call Distribution (NACD), and Network Message Service (NMS). The MCDN End to End Transparency feature supports the following features:

- MCDN to QSIG encapsulation conversion tool
- NAS using the QSIG transport
- NACD using the QSIG transport
- Network Message Service - Message Center (NMS-MC) and Network Message Service - Meridian Mail (NMS-MM) using the QSIG transport

All three MCDN applications, NAS, NACD, and NMS, are supported over the ISDN Primary Rate Interface (PRI) or the ISDN Basic Rate Interface (BRI) using the QSIG generic functional protocol (transport).

Figure 52 shows the MCDN End to End Transparency feature with QSIG links; Figure 53 shows the MCDN End to End Transparency feature in a mixed QSIG/MCDN network.

Figure 52
MCDN End to End Transparency feature with QSIG links

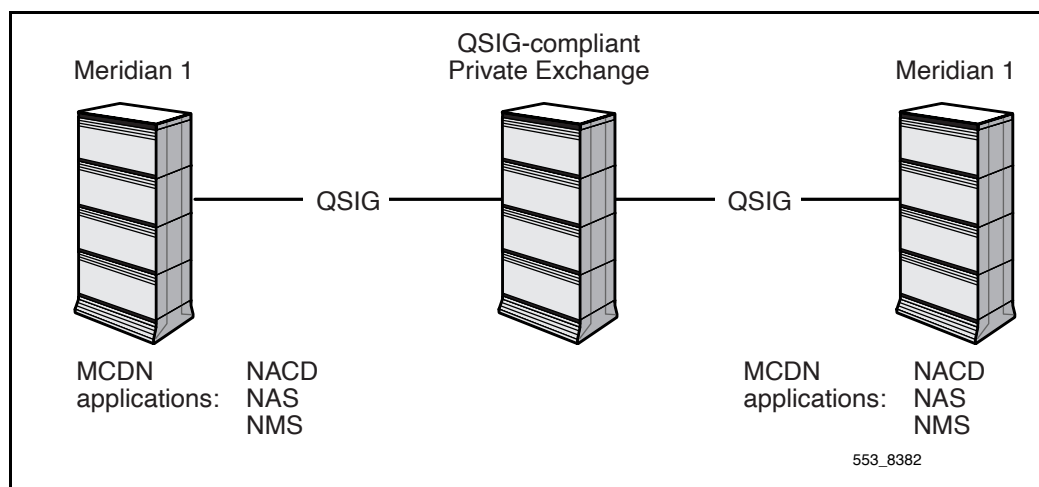
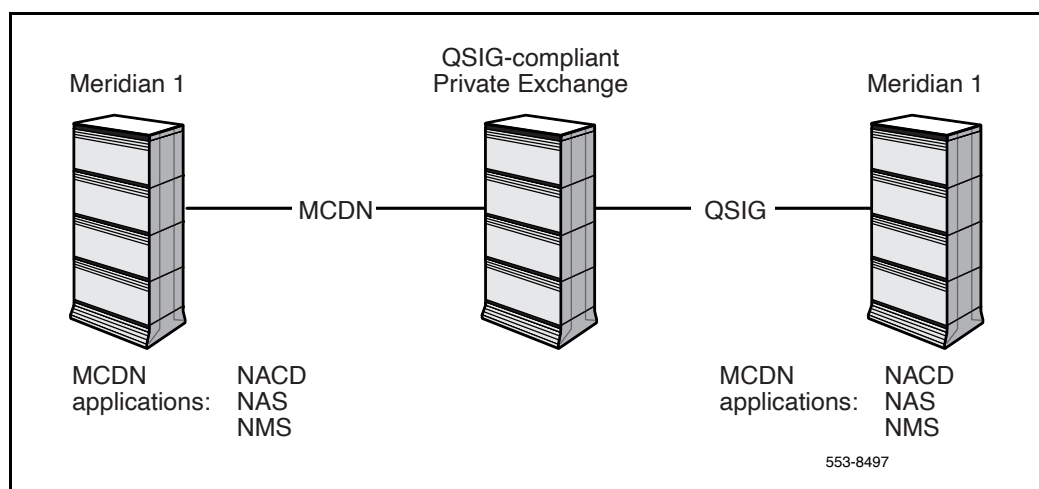


Figure 53
MCDN End to End Transparency feature with QSIG/MCDN links



NAS using the QSIG transport

The MCDN End to End Transparency feature supports NAS on the ISDN Primary Rate Interface (PRI) or the ISDN Basic Rate Interface (BRI) using the QSIG transport. The following NAS functionalities are available on QSIG:

- NAS Routing
- ISDN Call Connection Limitations
- Incoming Call Indication
- Attendant Break-In
- NAS Anti-Tromboning
- Call Extension
- Timed Reminder Recall
- Camp-On
- Call Waiting

For further information about these NAS features, see *ISDN Primary Rate Interface description and administration* (553-2901-101).

NACD using the QSIG transport

The MCDN End to End Transparency feature supports NACD on the ISDN Primary Rate Interface (PRI) or the ISDN Basic Rate Interface (BRI) using the QSIG transport. The ESN Coordinated Dialing Plan (CDP) or Uniform Dialing Plan (UDP) is required but may not be mixed together. The following NACD functionalities are available on QSIG:

- Make Set Busy (MSB) key
- Not Ready (NRD) key
- Individual DN key (IDN)
- Dialed Number Identification Service (DNIS) and DNIS Name Display
- Calling Line Identification (CLID)
- ACD-C and ACD-D reports

NMS-MC and NMS-MM using the QSIG transport

The MCDN End to End Transparency feature supports both NMS-MC and NMS-MM on the ISDN Primary Rate Interface (PRI) or the ISDN Basic Rate Interface (BRI) using the QSIG transport.

NMS-MC using the QSIG transport provides:

- Message Center access (direct or indirect)
- Message Waiting Indication notification

A direct call results from a station dialing the DN of the Message Center or pressing the Message Waiting key. An indirect call is a call network redirected to the Message Center.

NMS-MM using the QSIG transport provides:

- Meridian Mail access (direct or indirect)
- Message Waiting Indication notification
- NMS-MM features except Call Sender and Thru-Dialing

Operating parameters

The Manufacturer Specific Information (MSI) carries the MCDN proprietary information transparently over a QSIG network. The third-party node must support the QSIG generic functional protocol (transport) and MSI.

Directory Number (DN) address translation requires the association with a customer number, as configured using the CUST prompt in LD 15.

Either CDP or UDP is required but they may not be mixed together.

Digit manipulation is not supported for DN transported on QSIG.

Operating parameters for NAS using the QSIG transport

Attendant and users must be located on Meridian 1 nodes to obtain full NAS capabilities.

For NAS Anti-Tromboning capability, both legs must be on the same D-Channel. If the tromboning occurs with one MCDN trunk and one QSIG trunk, the redundant legs are not removed because they are not associated with the same D-Channel.

For a call established through International Standards Organization (ISO) QSIG trunks, the transit counter information is not updated at a QSIG tandem node because the message information is transparent; therefore, the transit counter does not reflect exactly the number of transit nodes the call has gone through.

NAS using the QSIG transport does not support Supervisory Console, Call Park, Charge Account, Do-not disturb and Group Do-not disturb, Barge-in, or Emergency Transfer.

There is no gateway developed between NAS functionalities using the QSIG transport and the corresponding DPNSS services. At a QSIG/DPNSS gateway, if NAS information is present, the NAS information is discarded.

Operating parameters for NACD using the QSIG transport

With the MCDN End to End Transparency feature, NACD continues to work fully with Symposium in an MCDN network. The MCDN End to End Transparency feature does not support NACD with Symposium using the QSIG transport or a mixed MCDN/QSIG network.

Messages which are resent at the first timer expiration on MCDN are sent once only on QSIG. At the first timer expiration on QSIG, the transaction is cancelled. This applies to the following messages: database update, the logical call request, the cancellation (from source or from target and for any reason) and the status exchange.

Operating parameters for NMS-MC and NMS-MM using the QSIG transport

NMS-MC and NMS-MM using the QSIG transport indirect access are based on the QSIG Call Diversion feature. The QSIG Call Diversion feature must be available and configured in the network.

NMS-MC using the QSIG transport is provided for NMS-MC users and operations located on Meridian 1 nodes.

NMS-MM using the QSIG transport supports all NMS-MM features except Call Sender and Thru-Dialing.

Feature interactions

Feature interactions with NAS using the QSIG transport

Interactions specific to NAS using the QSIG transport are discussed in the following section. Refer to *ISDN Primary Rate Interface description and administration* (553-2901-101) for generic NAS feature interactions.

QSIG Transit Counter

The QSIG Transit Counter Information Element (IE) is defined only with the ETSI interface; it carries the number of transit nodes the call has gone through. It is updated when a SETUP is sent on an ETSI QSIG trunk. Similarly, the NAS ISDN Call Connection Limitations (ICCL) information is updated when a SETUP is sent on a NAS MCDN or QSIG trunk.

For a call established using the International Standards Organization (ISO) interface for QSIG trunks, the counter information is not updated at a QSIG tandem node because the message information is transparent; therefore, the transit counter does not reflect exactly the number of transit nodes the call has gone through.

QSIG Attendant Recall

QSIG Attendant Recall allows a call extended on an ETSI QSIG trunk to recall the attendant if it is not answered within a customer-defined time. When ETSI QSIG is used and MQC_FEAT is set to NAS, NAS Timed Reminder Recall is enabled and takes precedence over QSIG Attendant Recall.

QSIG ANF Path Replacement/QSIG Call Transfer

QSIG ANF Path Replacement allows an established connection through a QSIG private network to be replaced by a new connection after a call modification to obtain a more efficient connection. It handles triangulation as well as tromboning mechanisms.

QSIG ANF Path Replacement take precedence on NAS anti-tromboning if the “Call transfer complete” FACILITY message is received before the ‘Call extension complete’ FACILITY message. Otherwise, NAS anti-tromboning applies and the QSIG ANF Path Replacement is not initiated when a ‘Call transfer complete’ FACILITY message is received.

Feature interactions with NACD using the QSIG transport

Interactions specific to NACD using the QSIG transport are discussed in the following section. Refer to *Network ACD description and operation* (553-3671-120) for information on generic NACD feature interactions.

QSIG Name Display

When QSIG Name Display is configured using the QSIG transport D-Channels and the name is allowed to be displayed and if the call is diverted to the target node through NACD routing, the originator's name is displayed on the target agent's set and the name of the target agent's ACD DN is displayed on the originator's set when the target agent answers the call.

Feature interactions with NMS-MC using the QSIG transport

Interactions specific to NMS-MC using the QSIG transport are discussed in the following section. Refer to *Network ACD description and operation* (553-3671-120) for information on generic NACD feature interactions.

NACD and NAS encapsulation within QSIG

For ACD and attendant Message Centers that are NACD or NAS, the ACD agents and attendants may reside at different nodes. With packaging requirements satisfied at the node where the MC call is presented, either the NACD agents or the NAS attendants are treated as a Message Center.

If the Message Center is an attendant, the NMS-MC feature operation remains the same as for a simple NMS-MC set. If the attendant NMS-MC has a specific ICI key defined, when a message Waiting call terminates to the attendant NMS-MC, the ICI lamp changes to lit to indicate a NMS-MC call.

As for MCDN, it is not recommended to use NACD or NAS as a NMS-MC unless the message data base can be made available to all agents or attendants at different nodes.

Feature packaging

The MCDN End to End Transparency feature requires these packages:

- MCDN End to End Transparency (MEET) package 348
- Multi-purpose Serial Data Link (MSDL) package 222
- QM reference signaling point Interface (QSIG) package 263
- QSIG Generic Functional protocol (QSIGGF) package 305

Depending on the application, other packages are also required.

For the QSIG ISDN PRI interface, the following packages are also required:

- ISDN Signaling (ISDN) package 145
- Advanced ISDN Network Services (NTWK) package 148
- 2.0 Mbit/s Primary Rate Interface (PRI2) package 154 **or**
- International Primary Rate Interface (IPRA) package 202

For NAS using the QSIG transport, you must have the ISDN PRI packages or ISDN BRI package and the following packages (the same packages as required for on the MCDN transport):

- Basic Routing (BRTE) package 14
- Basic Queuing (BQUE) package 28
- Network Class of Service (NCOS) package 32
- Network Alternate Route Selection (NARS) package 58
- Coordinated Dialing Plan (CDP) package 59
- Flexible Call Back Queuing (FCBQ) package 61
- Multi-Tenant Service (TENS) package 86
- Attendant Break-In/Trunk Offer (BKI) package 127
- Network Attendant Service (NAS) package 159

- ISDN Supplementary Features (ISDN INTL SUP) package 161
- Remote Virtual Queuing (ORC/RVQ) package 192

Note 2: NAS packages and Attendant Overflow Position (AOP) are mutually exclusive and cannot be equipped on the same system.

Note 3: NAS packages and Centralized Attendant Service (CAS) are mutually exclusive and cannot be packaged together.

For NACD using the QSIG transport, you must have the ISDN PRI packages or the ISDN BRI package and the following packages (the same packages as required for NACD on the MCDN transport):

- Basic Routing (BRTE) package 14
- Digit Display (DDSP) package 19
- Basic Queuing (BQUE) package 28
- Network Class of Service (NCOS) package 32
- Basic Automatic Call Distribution (BACD) package 40
- Automatic Call Distribution, Package B (ACDB) package 41
- Basic Alternate Route Selection (BARS) package 57 **or** Coordinated Dialing Plan (CDP) package 59 **or** Network Alternate Route Selection (NARS) package 58
- ACD Enhanced Flow (EOVF) package 178
- Network Automatic Call Distribution (NACD) package 207

For NMS-MC using the QSIG transport, you must have the ISDN PRI packages or ISDN BRI package and the following packages (the same packages as required for NMS-MC on the MCDN transport):

- Message Waiting Center (MWC) package 46
- Network Message Services (NMS) package 175

In addition, Automatic Call Distribution (ACD) Messages Services require the following ACD packages.

- Basic Automatic Call Distribution (BACD) package 40

- Automatic Call Distribution, Package B (ACDB) package 41
- Automatic Call Distribution, Package C (ACDC) package 42
- Automatic Call Distribution, Package A (ACDA) package 45

For NMS-MM using the QSIG transport, you must have the ISDN PRI packages or ISDN BRI package and the following packages (the same packages as required for NMS-MM on the MCDN transport):

- End-To-End Signaling (EES) package 10
- Integrated Message System (IMS) package 35 (for home node only)
- Basic Automatic Call Distribution (BACD) package 40
- Automatic Call Distribution, Package A (ACDA) package 45
- Message Waiting Center (MWC) package 46
- Command/Status Link (CLS) package 77
- Network Message Services (NMS) package 175

Feature operation

For NAS and NMS-MC and NMS-MM feature operation, see *ISDN Primary Rate Interface description and administration* (553-2901-101).

For NACD feature operation, see *Network ACD description and operation* (553-3671-120).

Malicious Call Trace enhancements

Introduction

Malicious Call Trace (MCT) allows users of selected telephones to activate a call trace that results in a printed report of the calling and called parties. The report is generated on all system TTYs designated as maintenance (MTC) terminals.

Note: For a description of the basic Malicious Call Trace feature, refer to *X11 Features and Services* 553-3001-306.

The Malicious Call Trace feature has been enhanced to offer advanced capabilities across an ISDN PRI network, as explained in the following sections.

Functional enhancements

The following enhancements to the functionality of the basic Malicious Call Trace feature have been introduced:

- For a call (from or to a Central Office on an analog CO or DID trunk) a special signal (hook flash and optional DTMF digit string) is sent to the Central Office if the option configured.
- Alarm operation is maintained; however, the duration of the alarm is now flexible (0-15 minutes), instead of being fixed at 15 minutes.
- The malicious call can be recorded using a recording trunk, if provisioned.
- The call trace record can now be printed on any Serial Data Interface (SDI) port with MCT defined as a user.
- The flexible firmware flash can be transmitted on the EXUT and XFCOT cards. This functionality is used by the Enhanced Malicious Call Trace, Meridian 911 and Autodial Tandem Transfer features
- The MCT feature can be activated from a Meridian 1 set during the established state of the call or during call clearing when interfaced with AXE10 Australia on 2 Mbit PRI trunks.

MCT record

The following enhancements to the Malicious Call Trace record has been introduced:

- Prior to Release 20, during an established call, the user of a set having Malicious Call Trace (MCT) Allowed class of service could invoke a call trace against the call. Activation of the feature resulted in the printing of a call trace record on the maintenance teletype terminal (TTY) of the system. The malicious call trace record did not contain any identifier of whether the call was external or internal; the format of these records is changed to provide information about the type of call (internal or external).

Malicious Call Trace for Saudi Arabia

In Saudi Arabia, from a user's perspective, the Malicious Call Trace feature activation remains the same as it was prior to this enhancement. However, with this enhancement the feature is now available for different types of analog and digital (CO, DID, and DOD) trunks. In order to send the MCT request, a special digit string is transmitted to the CO for an analog or digital trunk interface.

Malicious Call Trace for AXE-10 Australia

In Australia, MCT can be activated during the established state of the call when interfaced with AXE-10 Australia on 2.0 Mbit Primary Rate Interface (PRI) trunks. MCT can also be activated during the call clearing state of the call (within a maximum of 30 seconds from the caller going on hook). When MCT is activated, a special FACILITY message with a Key Pad information element is transmitted to the CO.

Trace Number (TRC) Key Lamp Status

The TRC key lamp status indicates the progress and success of the Malicious Call Trace request signaling to the CO and availability of the recorder. The following are the lamp states:

Lamp Winking

Activation of the TRC key changes its lamp from dark to winking if the trunk involved in the call requires the signaling to be done. The lamp remains winking, indicating a transient state, until the call trace request signaling to the CO has been completed.

In a Meridian Customer Defined Network (MCDN) tandem scenario, the set which originated the call trace remains winking until a Facility message is received from the node nearest to the central office. The user cannot invoke MCT again while the lamp is in the winking state.

Lamp Lit

If the call trace request to the CO is successful and the recorder is conferenced in the call, the lamp state is changed to lit.

In an MCDN tandem scenario, the lamp goes from winking to lit if a Facility message received from the node nearest to the CO indicates that the MCT request was successful. Activation of the TRC key during this state is ignored.

Lamp Flashing

This lamp state indicates that the call trace request to the CO was transmitted successfully, but a recorder could not be conferenced in. Activation of the TRC key during this state re-generates the MCT record, activates the alarm, and again attempts to conference in the recorder. The call trace request signaling to the central office is not transmitted again.

Lamp Dark

This lamp state indicates an idle TRC key or failure of the call trace request to the CO.

In an MCDN tandem scenario, the lamp goes from winking to dark if a Facility message received from the node nearest the CO indicates that the MCT request was unsuccessful.

Activation of the TRC during this state initiates all call trace elements again including: transmission of trunk hook flash; conferencing a recorder (if one is not already hooked in); generating an MCT record; and activating an alarm.

Operating parameters

Any country using flexible firmware flash timing (60-1536 msec.) requires the Generic XFCOT cards NTCK16AE or NTCK16BE; or the EXUT card NT8D14BA. For any country not using either the Generic XFCOT card or the EXUT card, the same functionality is provided by software control.

The Downloadable D-channel must be used to support MCT for AXE-10 Australia (2 Mbit PRI).

The following hardware is required: XUT NT8D14A; TDS/DTR NTAK03AA; CPU/CONF NAK01AA; 2.0 Mbit PRI NTAK79AA; DCH loadware NTBK50, NTBK51; 1.5 Mbit DTI NTAK09AA; 2.0 Mbit DTI NTAK10AA; Recorded telephone trunk NT8D14; and XFCOT card NTCK16AE, NTCK16BE; and EXUT card NT8D14BA.

Feature interactions

Note: For feature interactions pertaining to the basic Malicious Call Trace feature, refer to *X11 Features and Services* 553-3001-306.

Malicious Call Trace enhancements

ACD Emergency Key (EMR)

The Malicious Call Trace feature operates in a similar manner to the ACD emergency key feature when conferencing a recording. In this enhancement, the ACD set can activate both the Malicious Call Trace and ACD EMR features.

Autodial Tandem Transfer

The Trunk Hook Flash functionality is used by Autodial Tandem Transfer and Enhanced Malicious Call Trace.

Centrex Switchhook Flash

Interaction with the Centrex switchhook flash results because the flash range is changed for this feature. Communication to the CO (trunk hook flash) is performed by using the Centrex switchhook flash feature base code. The enhanced range is available for the Centrex switchhook flash.

Called Party Control (CDPC) Option

Prior to this feature, the CDPC option was not supported for conference calls. This is changed; the CDPC option is now supported if the conference contains exactly one recording trunk, one MCT activating party and one other trunk. This is done to make the recorder transparent to the user. The CDPC option remains unsupported for all other conference calls.

Malicious Call Trace Idle Signal

The existing operation of the Malicious Call Trace Idle Signal feature is unchanged.

MCT DN/TN Print

If the option MCDC (in LD 15) is set, a second line is added in the MCT reports to show the DN of both parties of the call. If CLID is available it is printed in the second line.

M911

The Trunk Hook Flash functionality is used by M911 and Enhanced Malicious Call Trace.

Feature packaging

Enhanced Malicious Call Trace is packaged under the existing Malicious Call Trace (MCT) package 107.

For ISDN environments, ISDN packages are required based on the node and network interface applicable to the specific country. In addition, International Supplementary ISDN Features (ISDN INTL SUPP) package 161 is required.

In order to send an MCT message request to a tandem node, Network Attendant Service (NAS) package 159 must be equipped.

Meridian Hospitality Voice Services

Description

This feature provides an enhanced form of Meridian Mail tailored to Hospitality services. It simplifies the use of mailboxes. It also allows more dynamic management of mailboxes as guests check in and out.

Two components of this feature rely on ISDN. These are

- Property Management System Interface Enhancement (message-waiting indication enhancement)
- ISDN/AP Recovery Enhancement

Message waiting indication enhancement

Under the Property Management System Interface enhancement, Text Messaging on a Property Management System (PMS) and Voice Mail on Meridian Mail are integrated. The handling of Message Waiting indications from the ISDN/AP is enhanced to ensure that Meridian Mail controls the Message Waiting status for both voice and text messages. Message Waiting indications will be handled only by way of the ISDN/AP link (when that link is enabled).

Note: If the ISDN/AP link is disabled for any reason the PMS link will be used to update Message Waiting Indication status.

ISDN/AP recovery enhancement

The ISDN/AP link is used to pass all the command and status information between the SL-1 and any Value Added Server (VAS). When the ISDN/AP link fails (due either to faults in the link itself or failure of the VAS), all signaling between the SL-1 and the VAS stops. Calls could have been left in various ringing states indefinitely (until the ISDN/AP link was restored).

The ISDN/AP Recovery Enhancement ensures that callers to any VAS using a particular ISDN/AP link are redirected to some alternate DN, should that ISDN/AP link fail. (This redirection only applies to calls in the ringing state which are being presented to virtual Voice Messaging Service agents.) Calls in the following call states can be recovered under this enhancement:

- calls ringing to a virtual Voice Messaging Service agent, but not yet answered by the VAS

- calls in an ACD queue at the time the failure is detected
- new calls arriving at the queue after the failure is detected, but before the link is restored

Calls are recovered from ACD queues under the following conditions:

- The ACD queue is defined as a Message Waiting Centre.
- The ACD queue uses the ISDN/AP.
- All ISDN/AP links associated with the ACD queue are not active.
- If an ACD agent is involved in the call, then the agents must be defined as Virtual Voice Messages access agents for the enhancement to take effect.

When calls are redirected, they are routed to the Night Call Forward DN (NCFW) for the particular ACD queue involved. The treatment of calls will be identical to that of existing Night Call Forward operation for ACD queues, except that the Night Call Forward DN must be located on the same SL-1 switch as the attendant and room telephones.

When the ISDN/AP link fails, any calls without disconnect supervision which are connected to the VAS are disconnected immediately. Calls with disconnect supervision remain connected, until the originating end disconnects. (In either case, a CDR record is produced when the call is Released from the agent.)

When the ISDN/AP link becomes operational once again (through either automatic or manual recovery) the handling of new calls will return to normal.

Feature interactions

Attendant End-to-end Signaling (AEES)

AEES (which uses Dual-tone Multifrequency signaling) requires an additional attendant (EES) key.

Attendant Overflow Position (AOP)

Attendant Overflow allows unanswered calls to the attendant to be forwarded to a customer-defined DN after a defined time. With AOP equipped in X11 Release 16, overflowed calls can be directed to Meridian Mail. The AOP DN

must be defined as an ACD DN, and the ACD DN must be an ACD agent configured as a virtual Voice Messaging Service (VMS) agent. A call may also be overflowed if all the attendants are in Position Busy.

Centralized Attendant Service (CAS)

The attendant must be located on the same switch as Meridian Mail for the attendant to use Meridian Mail features.

Digit Key Signaling (DKS)

With the DKS package (180) equipped, attendants may assist callers in operating Meridian Mail voice messaging service. The attendant enters the digits for Meridian Mail and extends the call to Meridian Mail. The caller may then access voice messaging. DKS is only supported from the attendant consoles local to Meridian Mail. The attendant may also place direct calls to Meridian Mail.

Digit Key Signaling at Console (DKS)

With DKS equipped, attendants may assist callers in Meridian Mail activities. The attendant may extend source calls to Meridian Mail or direct calls to Meridian Mail.

Do Not Disturb (DND)

Individual DND allows the attendant to place a DN in the Do Not Disturb mode. A DN in this mode is free to originate calls, but appears busy to incoming calls. With DND equipped in Release 16, callers can be redirected to Meridian Mail for voice mail services. A called set must have Hunting Allowed (HTA) Class of Service and the Customer Route Data Block must be set to “YES” in LD 15.

M2317, M3000, and Meridian Modular Soft Key Menus

M2317, M3000, or Meridian Modular soft key menus will not be supported by MHVS. These three telephones with CCSA Class of Service will not be presented with the Meridian Mail soft key menus when connected to Meridian Mail.

Network Automatic Call Distribution

The night number specified for the Automatic Call Distribution (ACD) involved in the ISDN/AP recovery operation must be local to the node.

PMSI, DKS, DND, and Message Waiting Indication

These operations are only supported when PMSI, Meridian Mail, and attendant room telephones are located on the same switch.

Pretranslation

Prior to MHVS, the setup of calls using the ISDN/AP was not supported from telephones using the Pretranslation feature. With MHVS equipped, call setup using the ISDN/AP is supported.

Stripping of Calling Party Name Display (CPND) Blanks

The maximum length of a CPND name sent from the PMSI/Background (BGD) terminal is 27 characters. When the full 27-character length is used, part of the CPND name may scroll off the screen. To avoid this problem, the PMSI/BGD software has been updated to strip all trailing blanks from the CPND name from the screen.

Feature packaging

Meridian Hospitality Voice Services (MHVS) package 179.

The standard Meridian Mail packages must be equipped for the Pretranslation and Do Not Disturb functions to operate properly. These include:

- Recorded Announcement (RAN) package 7
- End to End Signaling (EES) package 10
- Make Set Busy (MSB) package 17
- Integrated Message Services (IMS) package 35
- Basic Automatic Call Distribution (BACD) package 40
- Automatic Call Distribution Package A (ACDA) package 45
- Message Waiting Center (MWC) package 46
- Command Status Link (CSL) package 77
- CSL with Alpha Signaling (CSLA) package 85, and
- Auxiliary Processor Link (APL) package 109.

The PMSI package requires the following packages:

- Controlled Class of Service (CCOS) package 81
- Background Terminal (BGD) package 99
- Room Status (RMS) package 100, and
- Property Management System Interface (PMSI) package 103.

The Attendant Overflow (AOP) package 56 is required for AOP DN enhancement.

The Digit Key Signaling (DKS) package 180 requires that the standard Meridian Mail packages, such as those listed under Meridian Hospitality Voice Services (MHVS) package 179, are equipped.

The site may also require other packages such as PPM/Message Registration (MR) package 101 and Automatic Wake-up (AWU) package 102; however, these packages do not impact MHVS operations.

In addition, any packages required by ISDN are required.

Meridian Mail Trunk Access Restriction

The Meridian Mail Trunk Access Restriction (MTAR) feature prevents direct or indirect call transfer or conference of external calls to Meridian Mail. In this feature, external calls are defined as incoming/outgoing trunk calls that originate or terminate outside a private network.

This definition is applicable to all types of trunks, with the exception of TIE trunk calls. External calls are separated from a transferring/conferencing set on a network using TIE trunks. MTAR operation is dependant on the information sent to the remote node from the node that is attempting to transfer/conference. MTAR is triggered if the network information (such as Network Attendant Service or Calling Line Identification) indicates that an external call and a transfer/conference attempt to Meridian Mail is occurring. MTAR is also triggered if local information, such as Route Class, indicates an external call and a transfer/conference attempt to Meridian Mail is occurring.

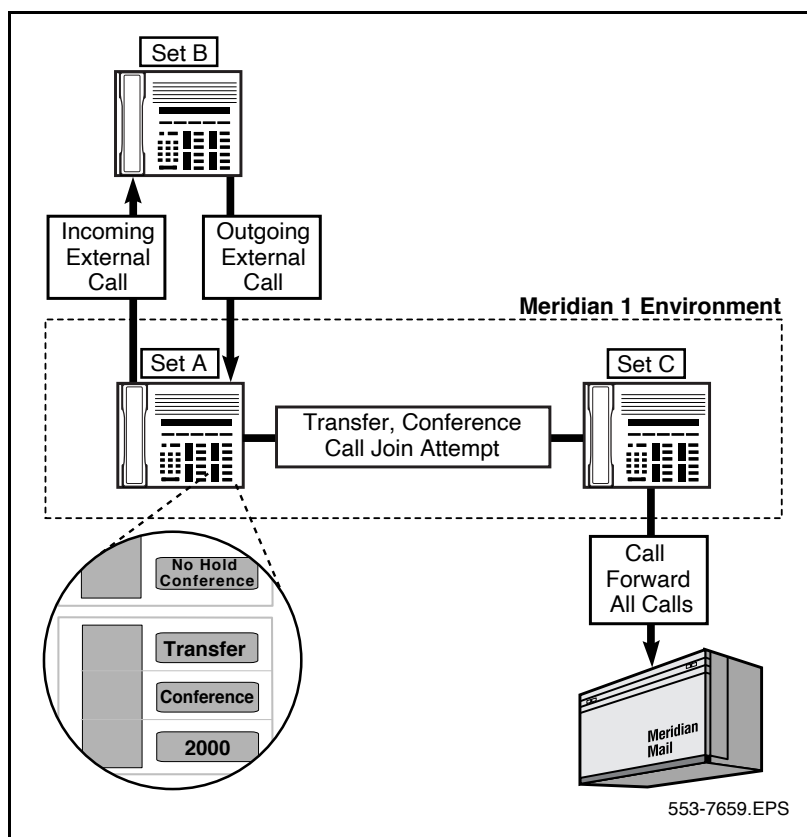
Meridian Mail Trunk Access Restriction averts potential Meridian Mail system abuse by distinguishing between internal and external calls that are directed to Meridian Mail. When activated, Meridian Mail Trunk Access Restriction impacts the operation of the following features:

- Call Transfer
- Conference,
- No Hold Conference, and
- Call Join capabilities of the Multi-Party Operation feature.

Meridian Mail Trunk Access Restriction prevents the completion of any Call Transfer, Conference, No Hold Conference or Call Join attempts on incoming/outgoing external calls to Meridian Mail.

As illustrated in Figure 54, Maters capabilities prevent an established call between Set A, an internal call, and Set B, an external call, from being forwarded to Meridian Mail. When Set A attempts to either Transfer, Conference, No Hold Conference or Call Join Set B to Set C, which is either a direct Meridian Mail DN or has activated Call Forward All Calls (CFAC) to Meridian Mail, the transfer and conference keys are ignored when pressed to complete operation.

Figure 54
Meridian Mail Trunk Access Restriction Call Transfer



Operating parameters

MTAR does not treat Centralized Attendant Position and Night Attendant sets as Attendant Consoles. These sets receive treatment based on their actual set type. For example, if the night attendant is a Meridian 1 proprietary set, then it is treated as Meridian 1 proprietary set.

The operation of an Attendant Console is not affected when this feature is enabled. An attendant can transfer or conference an external line to Meridian Mail directly or indirectly.

MTAR does not affect Automatic Attendant, Customer Controller Routing, Integrated Voice Response or Meridian Link features. However, if a user disallows any of these features from accessing Meridian Mail, the application must be written to take this into account.

Call transfer from ISDN Basic Rate Interface (BRI) set is not supported.

In a networking environment, Meridian Mail must reside on the same node as the transferring/conferencing set.

Feature interactions

Traffic Reporting

Traffic Reporting and Meridian Administration Tool's (MAT) traffic report, TFC005, are modified to report the number of times this feature is requested. A new line is added for the Meridian Mail Trunk Access Restriction which is identified by the feature number "27" and its peg count.

Network Call Transfer

Network Call Conference

Meridian Mail Trunk Access Restriction (MTAR) requires the transferring or conferencing set and Meridian Mail to be located on the same node. If the transferring or conferencing set are located not on the same node as Meridian Mail, the MTAR feature is not provoked because the call transfer/ conference attempt is terminated by a network on Meridian Mail.

However, an external call can be transferred or conferenced over the network, using TIE trunks. This operation is dependant on the type of network information the remote node forwards to the node where the transfer/conference attempt is made. Meridian Mail must be on the transferring/conferencing node. If network information is provided, indicating that an external call is attempting to transfer/ conference to Meridian Mail, the MTAR feature is invoked. When no network information is provided, MTAR is provoked if the local information (Route Class) indicates that an external call to Meridian Mail is being attempted.

Feature packaging

Meridian Mail Trunk Access Restriction requires Message Waiting Center (MWC) package 46.

Feature operation***Call Transfer/Conference******Meridian 1 Proprietary Set***

Set A is a Meridian 1 proprietary set with Transfer Key and Conference Key.

- 1** An incoming/outgoing external call is established between Set A and Set B, an external party. The call between Set A and Set B is active on Key X.
- 2** Set A presses the Transfer/Conference Key that automatically puts Set B on Hold.
- 3** Set A dials Set C. Set C has either Call Forward All Calls to Meridian Mail or is a Meridian Mail DN.
- 4** When Set A attempts to transfer/conference Set B by pressing more than once the Transfer/Conference Key it is ignored.
- 5** Set A recovers Set B by pressing Key X.

No Hold Conference***Meridian 1 Proprietary Set/ISDN BRI Set***

When Meridian Mail Trunk Access Restriction is enabled, direct or indirect no hold conference to an external call is permitted. During direct or indirect no hold conference, the calling party is never put on hold.

Set A is a Meridian 1 proprietary set or an ISDN BRI set with a No Hold Conference Key configured as either No Hold Conference, Conference Autodial, Conference Speed or Conference Hotline.

- 1** An incoming/outgoing external call is established between Set A and Set B, an external party. The call between Set A and Set B is active on Key X.
- 2** Set A presses the No Hold Conference Key.
- 3** Set A dials Set C. Set C is has either Call Forward All Calls to Meridian Mail enabled or is a Meridian Mail DN.

- 4 The conference is set up as normal. However a two party connection between Set B, an external party, and Meridian Mail is not allowed if the call controller releases. If this occurs, the connection between the trunk and Meridian Mail party is dropped.

Transfer

Analog (500/2500 type) Set

Set A is an Analog (500/2500 type) set with a XFA Class of Service (transfer and three/six party conference allowed).

- 1 An incoming or outgoing external call is established between Set A and Set B, an external party.
- 2 Set A performs a switchhook flash that puts Set B on hold.
- 3 Set A dials Set C. Set C has either Call Forward All Calls to Meridian Mail enabled or is a Meridian Mail DN.
- 4 Before or after the Meridian Mail has answered, Set A attempts to transfer Set B to Meridian Mail by going on-hook.
- 5 This attempt is treated as an illegal transfer. Set A is re-rung and reconnected with Set B when going off-hook.

Conference

Analog (500/2500 type) Set

Set A is an Analog (500/2500 type) set with a XFA Class of Service (transfer and three/six party conference allowed).

- 1 An incoming or outgoing external call is established between Set A and Set B, an external party.
- 2 Set A performs a switch hook flash that puts Set B on hold.
- 3 Set A dials Set C. Set C has either Call Forward All Calls to Meridian Mail enabled or is a Meridian Mail DN.
- 4 Before or after the Meridian Mail has answered, Set A attempts to conference Set B to Meridian Mail by performing another switchhook flash.
- 5 The conference is not permitted. Set A is reconnected to Set B. The call to Meridian Mail is disconnected.

Set A is an Analog (500/2500 type) set with a TSA Class of Service (three party service allowed).

- 1** An incoming or outgoing external call is established between Set A and Set B, an external party.
- 2** Set A perform a switch hook flash that puts Set B on hold.
- 3** Set A dials Set C. Set C has either Call Forward All Calls to Meridian Mail enabled or is a Meridian Mail DN.
- 4** Before or after Meridian Mail has answered, Set A attempts to conference Set B to Meridian Mail by dialing the conference control digits.
- 5** The conference is not permitted and Set A is reconnected to Set B. The call to Meridian Mail is disconnected.

Table 36 summarizes how different external calls are handled when Meridian Mail Trunk Access Restriction is enabled.

Table 36
Summary of Meridian Mail Trunk Access Restrictions

Telephone	External Call Type	Operation	Failure Treatment	Result
500/2500	Incoming	Transfer to Meridian Mail (MMail)	Re-ring to transferring set	Not allowed
500/2500	Incoming	Transfer to set with Call Forward All Calls (CFAC) to MMail	Re-ring to transferring set	Not allowed
500/2500	Outgoing	Transfer to MMail	Disconnect external call and Meridian Mail	Not allowed
500/2500	Outgoing	Conference to set with CFAC to MMail	Disconnect external call and Meridian Mail	Not allowed
500/2500	Outgoing/ Incoming	Conference to MMail	Reconnect to external call. Disconnect call to MMail	Not allowed

500/2500	Outgoing/ Incoming	Conference to set with CFAC to MMail	Reconnect to external call. Disconnect call to MMail	Not allowed
Telephone	External Call Type	Operation	Failure Treatment	Result
Meridian 1 Proprietary	Outgoing/ Incoming	Transfer/Conference to MMail	Operation ignored	Not allowed
Meridian 1 Proprietary	Outgoing/ Incoming	Transfer/Conference to set with CFAC to MMail	Operation ignored	Not allowed
Meridian 1 Proprietary	Outgoing/ Incoming	No Hold Conference to Meridian Mail or to set CFAC to MMail	Not applicable	Allow
Meridian 1 Proprietary	Outgoing/ Incoming	No Hold Conference release to make MMail to trunk two-party connection	Disconnect MMail and external trunk	Not allowed
Meridian 1 Proprietary	Outgoing/ Incoming	Call Join of external call to MMail	Operation ignored	Not allowed
Basic Rate Interface	Outgoing/ Incoming	Conference to MMail	Operation ignored	Not allowed
Basic Rate Interface	Outgoing/ Incoming	Conference to set with CFAC to MMail	Operation ignored	Not allowed
Attendant	Outgoing/ Incoming	Transfer/Conference to Meridian Mail	Not Applicable	Allowed
Attendant	Outgoing/ Incoming	Transfer/Conference to set with Call Forward All Calls to Meridian Mail	Not Applicable	Allowed

Meridian Network Services Drop Back Busy and Off-hook Queuing

Description

The Drop Back Busy feature provides a Drop Back capability at a tandem node during call setup. ISDN calls which are blocked at a tandem node are dropped back to the originating node, where the next routing alternative can be used.

The Off-hook Queuing feature allows the originator of an ISDN call control over whether Off-hook Queuing is to be provided at a tandem node. In effect, it provides the originating node with control over network calls.

Drop Back Busy and Off-Hook Queuing options are maintained even if the tandem node does not support this feature. Treatment at the tandem node is based upon the options set at the originating node. These options are transmitted to the tandem node in the call setup message.

The options available at the remote node are:

- Drop back to the originator if all routes in the initial set are busy.
- Off-hook queue if all routes in the initial set are busy, and then drop back to the originating node upon off-hook queuing timeout.
- Off-hook queue if all routes in the initial set are busy. Upon off-hook queuing timeout, attempt routing by way of the extended set. Drop back to the originating node only if all routes in the extended set are busy.
- Drop back to the originator if all routes in the initial set and the extended set are busy. (This is the default option.)

Interactions and Limitations

This feature operates only across an ISDN. Non-ISDN calls cannot receive drop back busy or off-hook queuing treatment across a network.

ESN off-hook queuing must be configured at the tandem node.

ISDN Network Attendant Service (NAS) takes precedence over ISDN Drop Back Busy.

For this feature to operate, ESN off-hook queuing (OHQ) must be operating. The criteria for ESN OHQ are:

- At least one of the trunk routes in the initial set of a route list is defined as being eligible for OHQ.
- The network class of service (NCOS) assigned to the calling party is defined to allow OHQ.
- The probability that the call will timeout in the OHQ before a trunk becomes available is below a specified threshold.
- The OHQ feature is enabled for the customer.

The feature is administered through overlay 86, by responding to the IDBB and IOHQ prompts.

Message Waiting Indication (MWI) Interworking with DMS

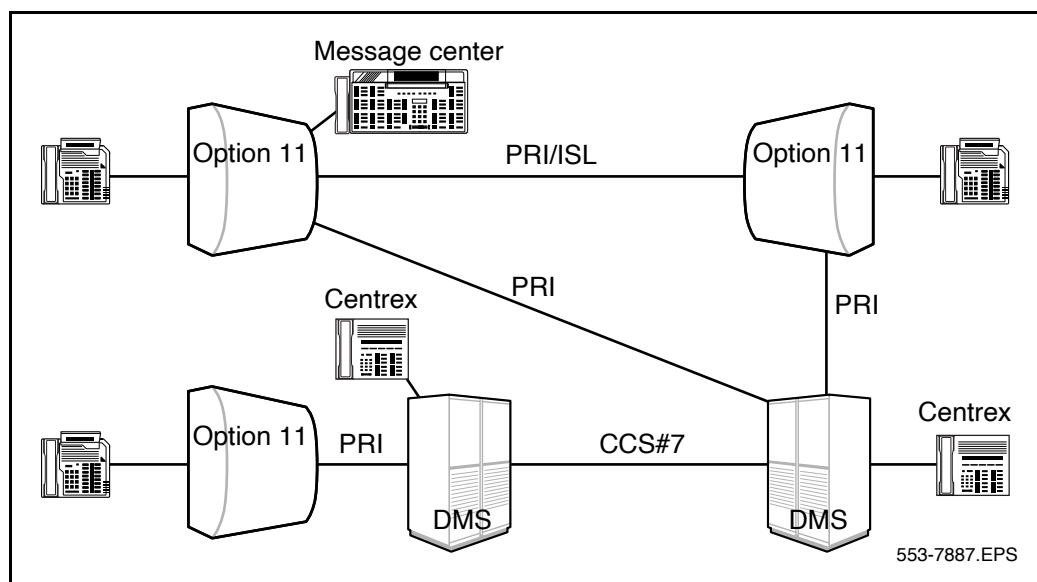
The Message Waiting Indication (MWI) feature allows a Message Center on either Meridian 1 or DMS-100 (Centrex) to exchange Message Waiting Indicator information with phones serviced by either type of system on the network. Calls redirected to the Message Center can be from private or public network callers on either system. A caller served by Meridian 1 can use the Message Center on DMS. A caller served by DMS can use the Message Center (Meridian Mail) on Meridian 1. The Meridian 1 must use a Meridian Mail as its Message Center.

MWI uses the non-call associated Facility messages on ISDN PRI for transporting the TCAP information.

The Meridian 1 and DMS systems can be connected directly or in tandem to whichever system hosts the Message Center. When a caller from either system leaves a message on either system, the Message Waiting Indicator is activated. When the receiver retrieves the message, the Message Waiting Indicator is deactivated. The indicator is either a visual LED (light emitting diode), an icon, or an audible tone in the telephone handset.

Figure 55 shows a network in which Meridian 1 hosts the Message Center that serves both Meridian 1 users and users of a DMS-100 in the same customer group.

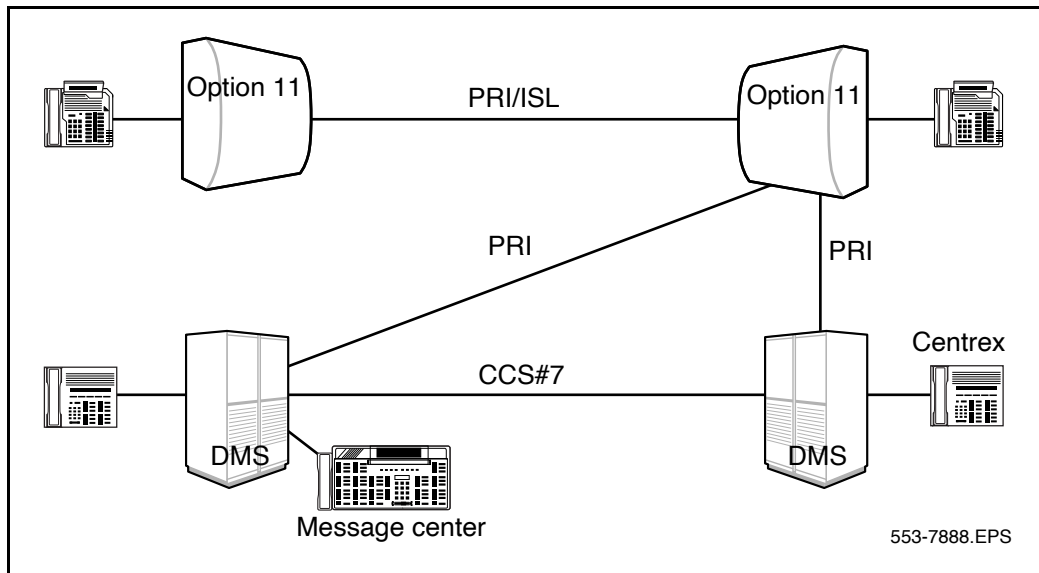
Figure 55
Private corporate network with Meridian 1 hosting the Message Center



Note: CCS#7 is not supported by the Meridian 1; CCS#7 must be a DMS-to-DMS configuration.

Figure 56 shows a network in which DMS hosts the Message Center that serves both Meridian 1 users and DMS users in the same or different enterprise groups.

Figure 56
Private corporate network with DMS hosting the Message Center



Note: CCS#7 is not supported by the Meridian 1; CCS#7 must be a DMS-to-DMS configuration.

Operating parameters

MWI supports DMS-100 and SL-100 as tandem switches, but no non-Northern Telecom switches as tandems.

Only Meridian Mail is supported as the Message Center on Meridian 1.

Both DMS-100 (using BCS 36) and Meridian 1 can transport Facility messages for MWI and can be tandem switches. DMS-100 can tandem Facility messages from Meridian 1 to Meridian 1 for features such as Call Sender activated by Meridian 1 users of Meridian Mail. Other Facility messages that DMS-100 can tandem are for updating the voice mail softkeys of an M2317 or M3000 set.

MWI allows public and private numbers for the Original Called Number (OCN) in the Setup message during call forwarding, and for the Directory Numbers (DN) in the ORIG and DEST Information Elements (IE) of the Facility message to toggle the MWI indicator. The OCN refers to the user's mailbox DN. The OCN must be unique in the Meridian Mail database. If the call forwarding DN to the Message Center is a private number, then the OCN constructed in the setup message will contain a private number. If the call forwarding DN to the Message Center is a public number, then the OCN constructed in the setup message will contain a public number.

If Meridian Mail is on Meridian 1:

- For Meridian 1 users, Meridian Mail provides the same functions as for the X11 Release 16 NMS-MM users, if a private number is used.
- For DMS users, some Meridian Mail features may not be available. Contact your local DMS administrator for available Meridian Mail features.

If the Message Center is on DMS:

- If a set on Meridian 1 does not have a DID number, the call forwarding DN to the Message Center should be private. Thus, a set that cannot receive DID calls will still be able to use the Message Center services provided that the DMS Message Center supports private numbers for the mailbox DNs.

End-to-End in-band signaling is required for accessing Message Center features for a local or remote switch.

The call forwarding connection from the caller to the Message Center on another switch must be with ISDN conductivity between Meridian 1 and DMS. This is for passing the OCN from the caller's switch to the switch that hosts the Message Center for leaving voice messages and for transporting TCAP messages to toggle the MWI indicator.

Feature interactions

- Multiple Customer
Serving multiple customers requires multiple Meridian Mail servers. MWI itself does not support multiple customers.
- Multi-Tenant Service
Meridian Mail Phase 8 supports Multi-Tenant Service. Tenants from the same customer can use one or more Message Center servers. Tenants from different customers cannot use the same Meridian Mail. The customer controls tenant access to MWI using the sets that belong to the tenant.
- Network Message Services, Meridian Mail
Although the Facility message for MWI uses a different TCAP format from that used for NMS, the X11 Release 16 functions continue to operate. X11 Release 19 Meridian 1 converts the message format when exchanging MWI Facility messages with earlier Releases.
- DCH Error Monitoring
The X11 Release 19 DCH Error Monitoring monitors ISDN messages on a per feature basis. If the MWI RCAP for the D-channel is added or deleted in LD 17, DCH Error Monitoring will work properly for the NMS feature when the D-channel message monitoring is disabled and then enabled.
- Trunk Optimization Before Answer
There is no Trunk Optimization when the call is redirected to DMS or answered by Meridian Mail. This applies to applications such as Auto Attendant.
- AML Link Recovery
AML Link Recovery redirects the calls to the ACD queue of the ACD Night Forward DN when the AML link goes down. This recovery is also available for calls in the Meridian Mail ACD queue on Meridian 1 when the AML link goes down.

Feature packaging

The package requirements for MWI are different for each node, as described in the following tables.

The originating node has Message Center users. Table 37 shows the originating node package requirements.

Table 37
Package requirements for an MWI originating node

Package	Name of package	Mnemonic
219	Message Waiting Indication (if connected to DMS/BCS36 for interworking)	MWI
10	End-to-End Signaling	EES
40*	Basic ACD	BACD
45*	ACD package A	ACDA
46	Message Center	MWC
145	ISDN Signaling	ISDN
146	ISDN Primary Rate Access	PRI
or 147	ISDN Signaling Link (for Meridian 1 to Meridian 1)	ISL
148	ISDN Network Service	NTWK
175	Network Message Service	NMS
* This package is required only if ACD DN is used as the Message Center DN.		

The host node holds the Message Center. Table 38 shows the host node package requirements.

Table 38
Package requirements for an MWI host node

Package	Name of package	Mnemonic
	X11 Release 19 or above	
219	Message Waiting Indication (if connected to DMS/BCS36 for interworking)	MWI
10	End-to-End Signaling	EES
35	Integrated Message Service	IMS
40	Basic ACD	BACD
45	ACD package A	ACDA
46	Message Center	MWC
77	Command Status Link	CSL
145	ISDN Signaling	ISDN
146	ISDN Primary Rate Access	PRI
or 147	ISDN Signaling Link (for Meridian 1 to Meridian 1)	ISL
148	ISDN Network Service	NTWK
175	Network Message Service	NMS

The tandem node does not have Message Center users. Table 39 shows the tandem node package requirements

Table 39
Package requirements for an MWI tandem node

Package	Name of Package	Mnemonic
219	Message Waiting Indication (if connected to DMS/BCS36 for interworking)	MWI
145	ISDN Signaling	ISDN
146 or 147	ISDN Primary Rate Access ISDN Signaling Link (for Meridian 1 to Meridian 1)	PRI
148	ISDN Network Service	

Feature operation

MWI performs two basic operations: 1) activate the MWI at the caller's set; 2) deactivate the MWI at the subscriber's telephone. Table 40 shows the sequence that activates the MWI. Table 41 shows the sequence that deactivates the MWI.

Table 40
When MWI is activated

Sequence	Caller	Receiver	Message Center
1	Calls receiver	Receiver not available	
2	Leaves a message		Records caller's message
3	Hangs up		Activates MWI on receiver's set

Table 41
When MWI is deactivated

Sequence	Receiver	Message Center
1	Notifies MWI	
2	Calls the Message Center	Plays messages
3	Clears the mailbox	
4	Hangs up	Deactivates MWI on receiver's set

Some of the failures that can occur and how MWI handles them are:

- AML/CSL link failure
If the AML/CSL link fails between Meridian Mail and its host Meridian 1, calls are redirected to the ACD Night Call Forward DN for the ACD queue involved.
- D-channel failure
If the D-channel is out of service, the Facility message is lost. For Meridian Mail, an MWI CSL message is sent to the Meridian Mail server.

Inconsistent data base

If the Facility message that toggles MWI cannot be delivered to the caller's set because the DN is invalid or MWI is not supported in the remote switch, an MWI CSL message is sent to Meridian Mail.

Network ACD (NACD)

Description

Network ACD provides call centers with multiple locations with the ability to solve the problem of fluctuating call traffic. The Network ACD feature uses the Integrated Services Digital Network (ISDN) to efficiently and quickly route calls to available agents within a network of Meridian 1 systems. Using the timed overflow routing tables associated with ACD, customers can specify ACD-DNs within the Meridian 1 network to which calls should be directed.

NACD is cost effective because calls remain in queue at their original source queue while waiting for an agent to become available in either the source queue or one of the target queues located throughout the network. Only when an agent becomes available in a remote queue (a different Meridian 1) will the call actually be sent over an ISDN voice channel. Prior to the call being sent, communication between locations is handled over the ISDN D-channel.

Management report information on package C reflects the changes introduced by the NACD feature.

ACD-MAX Release 3 supports the NACD feature. Three new reports and three additional management displays are provided to allow customers to determine call traffic pattern and service levels throughout the network. Centralization of this information is provided with the introduction of the Network Administration Centre. Network Administration Centre requires an ACD-MAX in each network location and provides reports and displays on the entire NACD environment.

At the present time NACD supports only a UDP numbering plan (CDP will be supported at a later date). It is important to note that the system regular users can operate using one dialing plan (i.e. CDP or UDP) while the NACD operates using another numbering scheme (i.e. UDP with location codes).

Interactions and Limitations

Option 11 systems only support Meridian ACD-MAX.

NACD is not supported on Package D, PDP 11 - Generic 9000 software.

Package Requirements

This option requires the following feature packages:

Feature	Package Number
BRTE	14
DDSP	19
BQUE	28
NCOS	32
NSIG	37
BACD	40
NARS	58
CDP	59
FCBQ	61
PBXI	75
TOF	111
ISDN	145
PRI2 or ISL	154 or 157
NTWK	148
EOVF	178

Network and Executive Distinctive Ringing

Description

The Network and Executive Distinctive Ringing feature allows distinctive ringing to function across an ISDN. Incoming calls to a node may tandem to another node by way of ISDN TIE trunks. If the incoming trunk route has been marked as distinctive, the terminating node rings the terminating set with a distinctive ringing cadence (network distinctive ringing).

This feature also allows certain telephone sets to distinctively ring any other sets in the ISDN (executive distinctive ringing).

Network wide distinctive ringing

The FTC table contains a choice of four possible Distinctive Ringing cadences. These four cadences are for network distinctive ringing only. A message is sent to the terminating node to indicate which of the four cadences is to be used. (If no indication is sent, ringing occurs as it would in a standalone case.)

Executive distinctive ringing

If a set is defined as an 'executive set', any party called by that set is rung distinctively. Selection of one of the four available tones and cadences is accomplished by defining class of service for the set.

The FTC table producing the distinctive tone and cadence depends on whether the call is strictly local (standalone, on the local node), or whether the call is a network call. For the network call, the customer option ROR/RTR on the terminating node determines which FTC table to use. For a local call, the customer FTC table is used.

Interactions and Limitations

All the limitations of standalone distinctive ringing also apply to network distinctive ringing. Feature interactions are the same as those for the standalone case, where trunks are treated as sets.

When there are five routes marked as distinctive, and an incoming call tandems to another node connected by a single TIE route, the terminating node can provide unique distinctive ringing for only four of the five. (While at the originating node, it can provide five unique rings, since each route can use a different FTC table.)

Index information is ignored if the terminating node does not have the distinctive ringing package equipped. In this case, normal ringing is given.

Since Executive Distinctive Ringing and Network Distinctive Ringing share the same four tones and cadences, selecting one of these for Executive Distinctive Ringing leaves only three unique rings. (All four rings can still be assigned to Network Distinctive Ringing, but if this is done then a trunk call rings exactly like an executive call.

Package Requirements

Network Distinctive Ringing is not packaged, but requires that the FTC, ISDN, ISDN International and Distinctive Ringing packages be equipped.

Executive Distinctive Ringing requires the same packages to operate, but is also packaged itself, as EDRG (package # 185).

All nodes involved in a connection must have the networking packages. The originating and terminating nodes must have the appropriate DRNG or EDRG packages equipped. If one of the two lacks the appropriate package, then this feature will not operate across the network.

Network Alternate Route Selection

Description

This package allows for efficient alternate routing of calls across the network as well as providing a uniform dialing plan in an ISDN environment.

Package Requirements

This option requires the following feature packages:

Feature	Package Number
BRTE	14
BQUE	28
NCOS	32
FCBQ	61

Network Application Protocol Enhancement

Description

This enhancement of the ISDN Network Application Protocol (ISDN NAP) feature for Meridian Mail services provides improved fault-detection responses. When a NAP link fails, a predefined time period is allowed to pass before the link is declared 'out of service'. Restoration of the link within this recovery time period results in uninterrupted service.

Signaling from the caller's set is ignored during the link failure. However, calls which were already established remain established during the recovery period. Should the link be declared 'out of service', only those established calls which use trunks without disconnect supervision are not maintained. (Such calls are re-queued to the front of the waiting queue.)

When a link is in the recovery period, calls ringing to an agent are held in the ringing state, until either the recovery period ends or the link is reestablished. Incoming calls requiring Meridian Mail services are placed in the ACD queue, linked to an agent (if possible) and given ringback tone.

When a link is declared 'out of service', calls which are ringing at an agent are re-queued. Re-queued calls and incoming calls on the 'out of service' link receive night-service treatment.

Normal operations resume as soon as the link is restored.

Network Attendant Service

Network Attendant Service (NAS) provides stand-alone attendant capabilities throughout a network. Any node in the network may have its attendant services located at any other node in the network, part-time or full-time.

Functionality

The activation of Network Attendant Service is controlled by the NAS key on the Attendant Console. When a call is presented to an attendant and the NAS key is not activated, the call receives normal attendant service. If the NAS key is activated, the call receives alternate treatment, based on one of the NAS conditions described below.

Network Attendant Service schedule

The NAS schedule allows a prioritized list of attendant alternatives (attendant locations) to be defined for up to 31 half-hour time periods. Each period can have up to four alternate attendants defined against it. During one of these time periods, a request for attendant service will use one of the alternatives, according to the specified order - the first attendant alternate is attempted, and if all attendants are in Position Busy, the following alternate in the sequence is attempted, and so on.

If none of the attendant alternatives are available, an attempt is made to give local attendant service, which may lead to the application of local NIGHT service.

Night Service

Night Service treatment is defined individually for each node.

With the introduction of NAS routing, identifying whether or not a system is in Night Service becomes more complex. Table 42 shows when Night Service treatment is in effect. When all the conditions in a vertical column are met, the system is considered to be in Night Service.

Table 42
Determination of Night Service

Local Attendants exist	No	No	No	Yes	Yes	Yes
Local Night active				Yes	Yes	Yes
NAS key active	No	Yes	Yes	No	Yes	Yes
No alternatives in current NAS schedule		Yes			Yes	
All alternatives in current NAS schedule tried			Yes			Yes

Status of local attendants

When NAS routing is active and there is an available attendant at the local switch, the call terminates at the local attendant, if one is available. If a local attendant is not available, the call may either be placed in the local attendant queue (if the Call Waiting threshold has not been exceeded and the console Call Waiting lamp is not flashing), or may receive local NIGHT service.

Alternate routing to attendant positions

Calls can be routed from one attendant to another attendant location in the network. Each node can be programmed with the attendant DNs for up to seven other switches. Up to four of these seven attendant DNs can be used as alternatives for any one time period. Each of the four attendant alternatives for each time period can have Drop Back Busy and/or Off-hook Queuing options configured.

If a call is to be routed to an alternate attendant at another node and all trunks to the other node are busy, the call is queued for an available route to the alternate (if the alternate has the Off-hook Queuing feature). The call remains queued for a specified time, which, upon expiring, causes the call to be directed to the next alternate attendant.

The operation of NAS routing, as controlled by the activation of the NAS key, is summarized in Table 43.

Table 43
NAS Routing Possibilities

	NAS Key	Home Attendant	Night Key	CW Lamp	Trunks to Remote	Action
1	Dark	All Busy	Dark	X	X	Queue to Home Attendant
2	Dark	Not Busy	Dark	Dark	X	Select Home Attendant
3	Dark	X	Lit	X	X	Route to Home Night
4	Lit	Not Busy	Dark	Dark	X	Select Home Attendant
5	Lit	All Busy	Dark	No Flash	X	Queue to Home Attendant
6	Lit	All Busy	Dark	Flash	Not Busy	Route to Remote Attendant
7	Lit	All Busy	Dark	Flash	All Busy	Queue to Home Attendant
8	Lit	X	Lit	X	Not Busy	Route to Remote Attendant
9	Lit	X	Lit	X	All Busy	Queue to route

Drop Back Busy

When a call reaches a Remote node, no attendant is available, and NAS routing or Night Service at that switch is about to be performed (if the Drop-back busy option was set at the originating switch and sent with the call), the call is treated by the originating switch as though no attendant is available at the Remote switch. Another alternative is selected from the originating switch.

Note: Drop Back Busy (DBB) and Remote Virtual Queuing (RVQ) are both packaged under the Originating Routing Control/Remote Virtual Queuing (ORC-RVQ) package 192. If RVQ and DBB are both configured, DBB will take precedence over RVQ. If the user wishes to activate RVQ, DBB must be disabled for a route entry in response to the IDBB prompt. If the user wishes to activate DBB, it must be enabled by entering DBI or DBA.

Attendant Off-hook Queuing

During peak periods, a call may be prevented from reaching a remote attendant if all possible routes to that attendant are busy. If these routes have off-hook queuing, the call will queue on these routes. If activated in the NAS programming and on the route list for the call, queuing occurs before the system attempts other alternatives. The alternatives are tried after the Off-hook Queuing (OHQ) timer expires.

Network Attendant Service capabilities

The following sections describe the network capabilities that are offered as part of NAS.

Break-in

The Break-in feature allows an attendant to enter an established call anywhere in the network and to offer another call or important message to one of the parties involved in the call. Break-in is provided as a key (BKI) function. The BKI key can be pressed before dialing the DN of the required station (pre-dial) or after (post-dial).

When the BKI key is pressed, the following situations may exist:

- **Break-in request recognized** — this is a temporary processing state, that occurs when an attendant tries to break-in to a party at another node, and lasts until the signaling protocol for the break-in attempt is completed.
- **Break-in ignored** — this state occurs if the required party has disconnected at the point of break-in, is idle without the Make Set Busy or Do Not Disturb features active, or is idle with the Call Forward feature active. In all cases, the party is rung, and the attendant can extend the call.
- **Break-in allowed** — the attendant can break into the required party, consult with the required party, and extend another call to the party if (a) the source is an external call, (b) Camp-on is available on the required party, and (c) the established call is a basic call.
- **Break-in denied** — break-in is not allowed if any of the above conditions are not met.

- **Break-in temporarily denied** — the attendant temporarily cannot break-in to the required party, but can try again later. This situation occurs if:
 - the required party is dialing, involved in a consultation call, or is on hold
 - the required party's set is ringing or receiving an indication tone
 - the required party's set is connected to a paging, dictation, recorded announcement, or integrated voice messaging trunk
 - the party to be extended is a local party or trunk with warning tone denied Class of Service
 - network blocking to a remote node has been encountered
- **Break-in consult only** — the attendant can break into the required party, and can consult with the party but cannot extend another call. This situation occurs if:
 - the attendant originates the call
 - the source is an internal call
 - the source is an external call and neither Camp-on nor Call Waiting is possible on the required party
 - the attendant attempts a pre-dial break-in and the required party is busy with Call Forward active.

Call extension

Restrictions that apply to attendants at local nodes also apply to calls extended across the network to remote node attendants.

Attendant control prevents the calling set from disconnecting until the attendant releases. Once the attendant releases or extends and releases the call, attendant control is relinquished.

Timed reminder recalls

Timed reminder recalls are attendant-extended calls that return to the attendant when their timers expire. The timers apply to slow answer calls, camped on calls, and call waiting calls.

When the attendant extends the call to a party at another node, the trunks between the attendant and the source and the attendant and the destination are not taken down after the Release key is pressed. This is true even if tromboning has occurred. This allows the attendant node to monitor the outgoing trunk for an answer signal. If no answer signal occurs within the specified time, the call is presented again to an attendant (or to the original attendant, if Recall to Same Attendant is active on the attendant node).

Anti-tromboning is only invoked after the destination has answered and the attendant has released from the call.

Incoming Call Indication

The Incoming Call Indicators (ICI) will operate in the same manner as a stand-alone Meridian 1 for incoming calls from stations or trunks.

New ICI types are introduced for NAS routed calls for each trunk type. The indicators can be the following:

- a dial 0 call (ordinary or fully restricted)
- a recall request
- a line lockout intercept
- Call Redirection (Call Forward No Answer, Call Forward Busy)
- an interpositional call
- a Listed Directory Number (LDN)
- a Message Center call
- NAS trunk types (type of trunk on which the call came into the originating switch. The choices are: NDID, NCO, NTIE, NFEX, and NWAT)

Attendant Display

Three features which pass information useful for attendant display are:

- Calling Line Identification (CLID)
- Network Call Party Name Display
- Network Call Redirection

For incoming calls, the CLID received (if available), is displayed instead of the Trunk Route Access Code and Trunk Member Number. If the call comes from the private ISDN network and there is a name associated with the calling telephone, the name is displayed on consoles with alphanumeric displays.

Reason for redirection codes are displayed if the call has been redirected by a feature such as Call Forward All Calls.

Camp-on and Call Waiting

Both Camp-on and Call Waiting allow calls extended to a busy set a further opportunity to reach that set. If the call is extended across the network to a busy set, Camp-on or Call Waiting may be attempted in the same manner as in the stand-alone case.

Tenant Service

If Multi-Tenant (TENS) software package 86 is equipped, the table number in the NAS overlay programming can be referenced to an individual Console Presentation Group. Each tenant can have its own individual NAS routing programmed.

Centralized Attendant Service vs. Network Attendant Service

NAS has some similarities to a previous non-ISDN application called Centralized Attendant Service (CAS). There are significant differences as well.

Note: NAS and CAS cannot be configured on the same Meridian 1.

Centralized Attendant Service

- Allows attendants to be centralized at one Main switch connected to Remote switches with Release Link Trunks (RLTs).

- When a remote switch has the CAS feature key activated, incoming calls to the Remote switch are redirected to the Main switch where an attendant answers the call.
- The call comes into the Main switch on an RLT. The attendant can extend the call back to a telephone at the Remote switch. The call is routed on the same RLT trunk which was used when the call came into the Main switch.
- Once the call is ringing at the telephone, the RLT trunk is released.
- If the telephone goes unanswered, the call recalls to the attendant at the Main switch. A Release Link Trunk is used for the recall.
- The CAS feature is activated using a key on a proprietary telephone or console at the Remote switch.
- When users at different switches call each other or the attendant places a call to another switch, normal TIE trunks are used.

Network Attendant Service

- NAS allows several switches to have attendants who answer calls which have been redirected by the NAS feature. The switches are connected with Primary Rate Interface (PRI) or ISDN Signaling Link (ISL) connections.
- NAS routing allows a call destined for the local attendant to be diverted to a Remote attendant. This routing is governed by a NAS schedule, a NAS key, or a permanent NAS Active setting which can be programmed.
- NAS calls can result in tromboned trunk connections between the Remote switch and the Attendant switch. These connections remain until the ringing telephone answers the call. This can result in two tromboned TIE trunks or channels being held up while a telephone is ringing.
- If anti-tromboning has been programmed, the tromboning is corrected after the telephone is answered.
- If the telephone goes unanswered, the call recalls to the attendant at the Main switch. A PRI channel or ISL trunk is used for the recall.
- A NAS key may be optionally programmed on an Attendant Console.
- NAS calls and normal user calls all use the same PRI/ISL trunks.

There are three additional differences between NAS and CAS:

- NAS routing is used when calls waiting in the local attendant queue exceed one of two thresholds. CAS routes all calls to the Main CAS switch.
- NAS chooses an attendant from up to four alternatives per time period. With CAS, each Remote site has only one Main switch to which to route calls.
- NAS routing can also be based on the status of trunks to other nodes and Night Service indicator status. These factors do not affect call routing with CAS.

Operating parameters

NAS and the Centralized Answering Position (CAP) capability are mutually exclusive. A Meridian 1 system may not have an actual attendant console. Instead, the system may be configured to use a Centralized Answering Position (using a Meridian 2616 digital set.) The CAP Directory Number (DN) is the customer Night DN. Since no attendant is configured, the customer is viewed to be in Night Service and any calls for the attendant are directed to the CAP.

Note: An attendant answering position must be an attendant console, and not a Meridian digital set.

NAS and Centralized Attendant Service packages are mutually exclusive and may not be packaged together.

NAS and Attendant Overflow Position packages are mutually exclusive and may not be packaged together.

Trunk Group Busy applies only to trunk groups on the node where the attendant resides.

The Uninterrupted Line Connection feature continues to work on a stand-alone switch basis; however, Network-wide Break-in, based on warning tone Class of Service, cannot be guaranteed. Only one Attendant at a time is allowed to break into a connection. Break-in is allowed for telephones only, not trunks or other attendants. Attendant Control is not provided if the connection is a result of a Network Break-in.

Unsupported features

NAS does not support:

- Supervisory Console
- Call Park
- Charge Account
- Do-Not-Disturb and Group Do-Not-Disturb.

The following functions and features are not supported across the network:

- Barge-in
- Busy Verify
- Emergency transfer.

Feature interactions

Attendant Interposition Call

An attendant is not able to call a specific attendant on another node by dialing the attendant DN followed by the attendant number. The attendant dials the Network Alternate Route Selection (NARS), Coordinated Dialing Plan (CDP), or Listed Directory Number (LDN) the same way a telephone dials to reach the attendants at another node.

Busy Verify

Pre-dial Break-in provides equivalent functionality but has several advantages over Busy Verify. Busy Verify does not operate on a network-wide basis.

Call Forward, Break-In and Hunt Internal or External Network Wide

When a call is transferred, a new facility message is sent to the transferred party's node to transport the terminal indicator parameter or the access trunk information parameter.

DPNSS1 Route Optimisation/MCDN Trunk Anti-Tromboning Interworking

If tromboning trunks are removed on the MCDN side of a RO/TAT Interworking gateway scenario by the Network Attendant Service feature (since NAS has presence over TAT), the RO/TAT Interworking functionality is not invoked. The result is that, if NAS is equipped, attendant-extended calls that are in a tromboning state are optimised on the MCDN side, but DPNSS1 trunks are not optimised on the DPNSS1 side of the RO/TAT Interworking gateway scenario.

EuroISDN Master Mode

NAS signaling is not supported on a EuroISDN Master Mode connectivity. However, NAS will interwork with an incoming call from the EuroISDN Master Mode (routing and call handling).

INIT ACD Queue Call Restore

Call information associated with Network Attendant Service is lost after system initialization and call restoration.

ISDN QSIG Basic Call

ISDN QSIG Basic Call interacts with Network Attendant Services (NAS) as if the call is going to a route without NAS being equipped.

Listed Directory Numbers, Network-wide

This feature enables LDNs to be recognized network-wide when NAS is used. Up to six LDNs can be defined per Meridian 1. The same LDNs must be configured in multiple nodes. The LDN which users dial can be programmed as a CDP number, for example, and input in the NAS overlay as the ID of remote attendants.

Recall to Same Attendant

This feature operates on a network-wide basis for the following call types:

- Slow Answer Recall
- Camp-on Recall, and
- Call Waiting Recall.

The operation of this feature is affected by the programming for the option in the Customer Data Block of the system where the attendant answering the call resides.

Feature packaging

The following packages are required for NAS:

- Network Attendant Services (NAS) package 159;
- Advanced ISDN Network Services (NTWK) package 148;
- Network Automatic Routing System (NARS) package 58 **or** Coordinated Dialing Plan (CDP) package 59;
- Network Class of Service (NCOS) package 32;
- Basic Routing (BRTE) package 14;
- Attendant Break-In/Trunk Offer (BKI) package 127;
- Basic Queuing (BQUE) package 28;
- Flexible Call Back Queuing (FCBQ) package 61;
- Originating Routing Control/Remote Virtual Queuing (ORC-RVQ) package 192 for Drop Back Busy (please see note under “Drop Back Busy” description.);
- Integrated Services Digital Network (ISDN) package 145;
- ISDN Supplementary Features (ISDNS) package 161; and
- 2.0 Mbps Primary Rate Interface (PRI2) package 154 **or** ISDN Signaling Link (ISL) package 147.

Example of NAS implementation

Figure 57 illustrates a simple application of the NAS functionality in a private ISDN PRI network connecting three Meridian 1 systems. Although this example uses NARS, the same functionality can be accomplished using CDP.

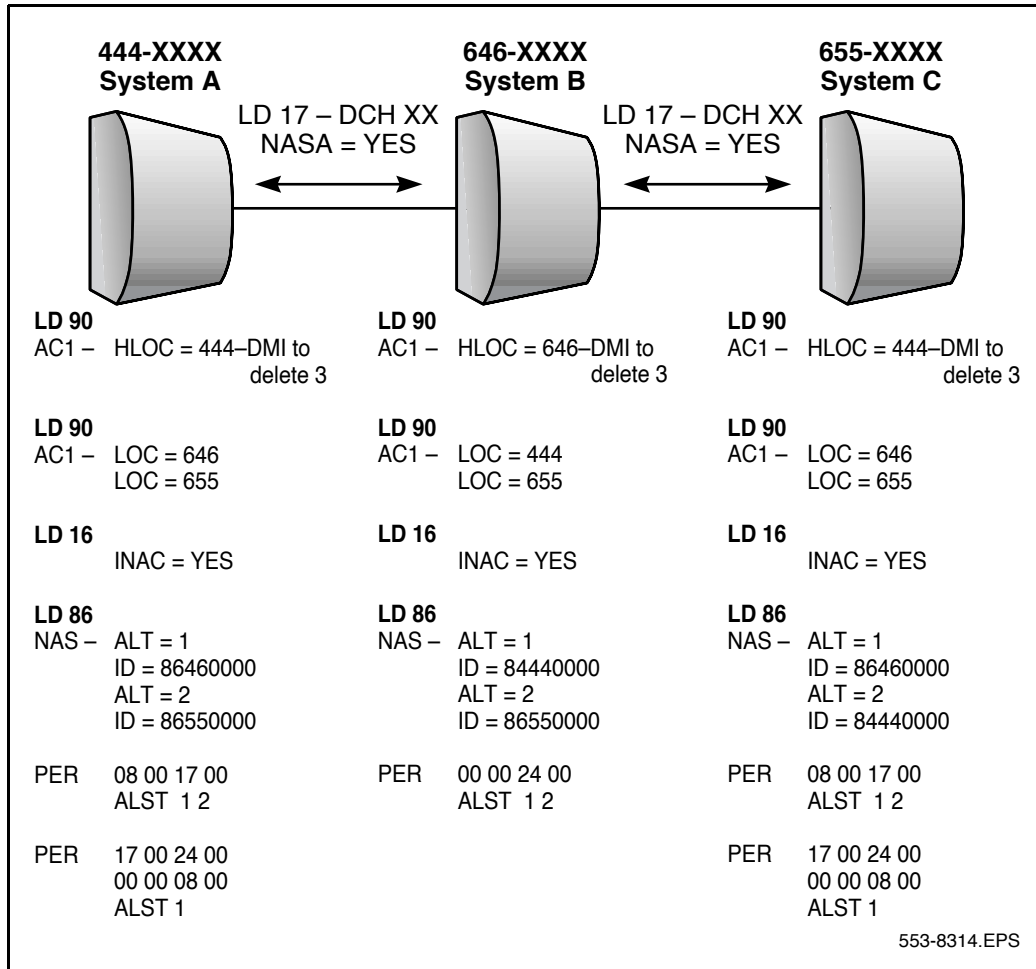
Each system has the translations in place for a HLOC and the two LOC codes for the other two Meridian 1 locations. This programming was done in the NARS database. The route lists required in order to process these types of calls have been programmed also.

The INAC implementation was part of the basic ISDN/Networking programming.

The NAS programming involves the configuration of the D-channel for NAS functionality, the ID of the other attendants to be used, a list of the sequence in which these attendants are to be scanned for availability and the time periods for each list.

System B, in this example, is the location where attendants should be available 24 hours a day. During the day, the other two locations are programmed to use System B as their first choice backup and then as a second choice System A uses System C and System C uses System A. At night, Systems A and C use System B as the only backup. If it is unavailable, calls will be given Night Service.

Figure 57
A simple application of NAS functionality in a private ISDN PRI network



Feature operation

There are no operating procedures specified for this feature.

Network Call Party Name Display and Network Name Delivery (NND)

Description

Network Call Party Name Display (NCPND) offers a network-wide visual display of the Calling Party and Called Party Name on the telephone sets with Alpha-Numeric Display.

Across the ISDN DTI/PRI network, the NCPND feature displays the following:

- the calling party's name on the terminating set
- the called party's name on the originating set

With X11 Release 16 and earlier, the NCPND feature may only be used for private network applications that support the Meridian 1 to Meridian 1 connection (SL-1 and SL-100). The name and number display for the called party is visible for the duration of the call. The name display is retained for the calling party, however, the number will be replaced by the trunk route and member if the call is placed on hold and then later retrieved.

With X11 Release 18 and later, Network Name Display Interworking allows caller and calling party name display in Meridian 1 connections to DMS-100, DMS-250 and Meridian 1 Options 11-81.

The following telephones support the NCPND feature:

- M2250 Attendant Console
- Meridian Modular Telephones
 - M3000 Touchphone
 - M2317
 - M2008 with Display
 - M2616 with Display
 - M2016S with Display
 - M2216

Please consult the telephone User Guides to determine the exact display for each particular set.

Network Name Delivery (NND) is the method used to send the names and numbers across the network. There are two NND methods available: ND1 and ND2.

Operating parameters

Network Name Delivery method 1 (ND1) is available in X11 Release 14 and later. The following list describes ND1 operating parameters.

- Meridian 1 to Meridian 1 connections only are supported.
- For X11 Release 14, ND1 supports name display for redirected calls by Call Forward All Calls.
- With X11 Release 16, ND1 supports the name display for call redirection by Call Forward No Answer, Hunt and Call Pickup if the redirection is done across the network. See "Network Call Redirection" in this chapter for configuration.
- If the called party is at a remote switch, the called party name is displayed on the caller's display only after the call is answered unless there was a redirection. (See Network Call Redirection in this chapter for a discussion of call redirection with X11 Release 16 and later hardware.)
- A CPND enhancement allows the display of the redirecting name on the terminating telephone instead of the calling name by a service change option only if the first redirecting party is on the terminating switch.
- Prior to X11 Release 16, ND1 supports the name display for call transfer only for a local transfer over PRI. In that case, the name is displayed on the terminating telephone.
- The maximum number of characters in a displayed name is 24.

Network Name Delivery method 2 (ND2) is available in X11 Release 18 and later. ND2 provides network-wide name display in compliance with the Meridian Customer Defined Networking (MCDN) protocol.

The following list describes the ND2 operating parameters.

- ND2 supports direct PRI connections to Meridian 1 (with X11 Release 18 and later), SL-100, and DMS (with BSC32 and later software).

- For Meridian 1 to Meridian 1, the maximum number of characters in a displayed name is 24. When connecting to a Central Office, 15 characters only are supported. Names exceeding this length are truncated.
- ND2 supports name delivery for Call Pickup, Call Transfer, Hunt and Call Forward All Calls/No Answer/Busy.
- A CPND enhancement allows the display of the redirecting name on the terminating telephone instead of the calling name by a service change option only if the first redirecting party is on the terminating switch.
- In all cases when the name is available, the called party name is displayed on the caller's display during the ringing phase. This is an enhancement over ND1 which displays names on connect.
- ND2 enhancements are not supported on connections to Meridian 1 using ND1. ND1 is applicable for X11 Release 14 through 16. ND2 is recommended for X11 Release 18 and up. The far ends must match.

Interactions and Limitations

The NCPND feature may only be used for private network applications that support the Meridian 1 to Meridian 1 connection. The name and number display for the called party is visible for the duration of the call.

Package Requirements

This option requires the following feature packages:

Feature	Package Number
DDSP	19
CPND	95
ISDN	145
PRI2 or ISL	154 or 147

Note: In addition, the appropriate digital set options (Telset package 88, 89 and 91 or 170) must be equipped to support the M3000, M2317 and Meridian Modular telephones.

Network Call Redirection (NCRD)

Description

Network Call Redirection allows calls to be redirected across the ISDN network as a result of Hunting, Call Forward All Calls and Call Forward No Answer (these features, with the exception of call forward all calls, used to be limited to the called sets PBX).

CLID and CPND

The feature provides an update of information for both the calling and the called set's display for CLID and Network Call Party Name Display (NCPND) when the call is redirected. The feature enables the display of the calling party to show the connected number when the called number is redirected to another number in the ISDN network. If Network Call Party Name Display (NCPND) is optioned, the calling party's set also displays the connected party's name. If a call is forwarded more than once, the calling party's set displays only the first re-direct number and name with the connected name and number.

Redirection Reason

This feature also enables the terminating set to display the CLID from across the DTI/PRI network as well as the forwarded set's number. If NCPND is optioned, the calling party's name and number as well as the originally called party number is displayed at the terminating set. In addition, the calling and terminating party sets will display the redirection reason (CFWD, CFNA, HUNT, PKUP, XFER).

Network Call Redirection also allows the customer to either block calls from being redirected over the network, or limit the number of redirections from 1-5 times per call.

Network Call Transfer and Attendant Extended Calls

Description

Network Call Transfer and Attendant Extended Calls display the calling party name and number to the "Transferred to"/"extended to" party across the network. Also the calling party's display is updated to show the connected party's name (if NCPND is optioned) and number.

Network Call Transfer does not provide the ESN capability that verifies if double trunking is being utilized; i.e., Network Transfer now allows calls to be blindly transferred across the ISDN network.

Network Hunt

Description

The Network Hunt feature extends the current capabilities of Hunt to network wide functionality using a Meridian 1 Network connected through either ISDN Primary Rate Interface (PRI), or ISDN Signaling Link (ISL). Also, the displays for the originating and terminating sets of a call that is redirected are updated to correctly present information to the connected parties and the reason for redirection (Hunt).

For example, in an ISDN Network with three Meridian 1 SL-1s, when a user on site "A" calls a user on site "B" but is forwarded to site "C", site "A" and site "C" set displays are updated. Site "A's" set shows the originally dialed number, the connected number (site "C" number), and the reason for redirection, i.e., Busy. Site "C" set shows; the calling number and name, the originally called number (site "B" number), and the reason for redirection.

Network Individual Do Not Disturb

Description

This feature extends several functionalities of the Individual Do Not Disturb feature to operate within a network environment. A DN in the Do Not Disturb mode is free to make calls, but appears busy to incoming calls. Incoming DID calls to a DN in the Do Not Disturb mode are intercepted to the local attendant, or the network attendant servicing the call. All other calls receive customer-defined intercept treatment defined in LD 15 (i.e., busy treatment, call routed to the attendant, or call routed to recorded announcement).

An attendant dialing a busy DN receives overflow (the DN may be on the same or different node as the attendant). The attendant may then verify whether or not the DN is in Do Not Disturb mode by pressing the IND DND key on the console; if the associated lamp remains steadily lit, it indicates that the DN is in Do Not Disturb mode. The attendant may then temporarily override Do Not Disturb by activating the Break-in feature.

Interactions

Call Forward All Calls, Hunting

Call Forward All Calls, and then Hunting, take precedence over DNDI treatment.

Make Set Busy

The DNDI intercept treatment takes precedence over Make Set Busy indication.

Group 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.

Package Requirements

This option requires the following feature packages:

Feature	Package Number
DNDI	9
NAS	159
BKI	127

Network Intercom (Hot Type D and Hot Type I Enhancements)

Hot Line enables a designated telephone to place calls to a predetermined destination that may be internal or external to the Meridian 1. The call does not require attendant intervention. 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 System Speed Call (SCC) using LD 18. Once a Hot Line call enters the ringing state, it is the same as a normal call.

Prior to Release 21, there were two types of Hot Line keys (DN-based Hot Type D, and Speed Call List-based Hot Type L). Release 21 introduces another type of Hot Line key – Hot Type I, while also providing improvements to the existing Hot Type D. These two improvements function in both standalone and network environments.

Hot Type D provides the ability for Meridian Digital Telephones to have two-way calls on specially designated keys (not on the DN keys) with other Meridian Digital Telephones connected to Meridian 1 PBXs across a Meridian Customer Defined Network (MCDN) Integrated Services Digital Network (ISDN). A Hot Type D call can now terminate in three different modes: Voice, Ringing, and Non-ringing. With the Voice mode, speech path is automatically connected after a short ring. With Ringing and Non-ringing modes, the call must be manually answered by the called party; the difference between the two modes is that for Non-ringing no audible tone is given (the Hot Line key flashes to indicate the call).

Hot Type I

An option is available with Hot Type I to provide a No Answer Indication informing the called party that a Hot Line call was made during the called party's absence. If an Hot Line call cannot be completed on the Hot Line key, the calling party is informed via the set's display, and the call is completed over the network as a normally dialed call that attempts to terminate on the destination Prime DN.

Hot Type D Enhancement

Hot Type D allows more than one set having the same target DN defined for the Hot Type D key. This enhancement includes Voice, Ringing, and Non-ringing termination modes, and the capability to leave a No Answer Indication in some situations (i.e., the Hot Line key winks). A call terminating on an enhanced Hot Type D key operates the same as if terminating on a Hot Type I key, if the originating DN is the same as the target DN defined for the key. If it is not the same, but another Hot Line key exists on the set that has a target DN that matches the originating DN, a No Answer Indication is left on that key.

Operating parameters

Hot Type I calls are not allowed on analog (500/2500 type) telephones.

Hot Line keys cannot be defined for M3000 sets.

A Hot Line key should not be defined on a station without a Prime DN and likewise should not be defined on the primary key. If this is not done, the improved functionality will not work and the call is treated as a non-Hot Line call.

The network DN for Hot Type I and the No Answer Indication component of Hot Type D must be either a Coordinated Dialing Plan or a Universal Dialing Plan number that must terminate on a Prime DN of a Meridian Digital Telephone; otherwise, the call is competed as a non-Hot Line call.

The network-wide application of Hot Type I is only applicable to nodes in a Primary Rate Interface (PRI), ISDN Signaling Link (ISL), Virtual Network Services (VNS), and Basic Rate Interface network. The originating and terminating nodes of the Network Intercom (Hot Type D and Hot Type I Enhancements) feature must be Meridian 1 switches running a minimum of Release 21 software. Tandem nodes in the private network must be Meridian 1 (minimum Release 21 software) or another MCDN node able to transfer Remote Operations Service Element (ROSE) facilities.

If the termination mode is voice and the called party is idle and the handsfree voice call (HZA in LD 15) is active, there is no indication to the software that the called party really answered the call. If any other key is pressed, the No Answer Indication is not left.

Hot Line keys must be defined with the same dialing plan.

The Hot Type D enhancements can also function across R2MFC or Digital Private Network Signaling System #1 (DPNSS1) trunks provided that the calling party's DN is provided.

Feature interactions

Attendant Blocking of DN

A Hot Type I key cannot be blocked by the attendant, because it has no DN.

Pressing a Hot Type D key that is attendant blocked establishes the call on the source side of the attendant.

Auto Hold

If a user who originated a Hot Type I call receives or makes another call on another DN, pressing that DN puts the established call on hold. If a user presses the Hot Type I key while a call is established on it, the call is placed on hold. If the Hot Type I key is pressed while a call is established on another DN, the established call is put on hold. If a station with automatic hold allowed Class of Service receives an Hot Line call, the user of that station can put the active call on hold by pressing the Hot Type I key or by making or answering another call on another key.

Automatic Call Distribution (ACD)

Hot Type I calls cannot terminate on an ACD DN. A call attempting to terminate on an ACD DN receives overflow tone. Hot Line calls involving ACD sets must use the Hot Type D option.

Busy Forward Status (BFS)

In a Secretarial filtering scenario, the secretary's BFS lamp also will reflect that the boss' set is busy if the boss is on a Hot Type I call.

Call Forward

Hot Type I calls respect or override all kinds of Call Forward features (Busy, No Answer, All Calls, Internal, etc.) according to per-set definitions. If Call Forward is respected, the call becomes a normally dialed call and the originator will receive the appropriate indication on their display.

Call Join

Hot Type I calls can be moved to the Conference key with the Call Join feature.

Call Park

Hot Type I calls cannot be parked.

Call Party Name Display

Hot Type I calls display names the same as a normal call.

Hot Type I calls that become a normal call indicate on the originating station's display that the call is no longer a Hot Line call.

Call Pickup

Hot Type I calls cannot be picked up. An attempt to pick up a Hot Type I call results in an overflow tone.

Call Transfer

Hot Type I calls may be transferred to another Hot Line key or to a normal DN key; likewise calls on a normal DN key may be transferred to a Hot Line key.

Display Key

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.

Do Not Disturb (DND)

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

Flexible Feature Code (FFC) Boss Secretarial Filtering

Hot Type I calls overrides this feature (i.e., 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.

Last Number Redial

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

Make Set Busy

Hot Type I calls terminating on a station in the Make Set Busy mode override Make Set Busy.

Multiple Appearance Redirection Prime (MARP)

If more than one set is allocated the same prime DN, the Hot Type I call will terminate on the set designated as the Multiple Appearance Redirection Prime (MARP). If the MARP DN is not the prime DN on the set, or if the set designated as the MARP DN is not a Meridian Digital Telephone set, the first Meridian Digital Telephone set with the prime DN will be used. If none of these conditions are met, the call will terminate as a non-Hot Line call and the calling party will be notified on the display.

Hot Type D calls can have voice termination only on a MARP Terminal Number (TN), or if there is no MARP TN, then on the first TN in the TN list. A No Answer Indication for Hot Type D can only be left on the MARP TN, or if there is no MARP TN, then on the first TN in the TN list.

Override

An internal Hot Type I call never returns busy, unless the call became a non-Hot Line call due to the Hot Line key being busy. In this case, the call behaves like a normally dialed call, and override can be used upon receipt of a busy signal.

Ring Again

Hot Line calls terminating on a busy key become normal calls. Hence, they may use the Ring Again feature under normal circumstances.

Ring Again – No Answer

If Ring Again No Answer is activated for a Hot Type I call, it is activated as if the call had been dialed normally.

Ringing Change Key

The ringing/non-ringing mode of an enhanced Hot Type D or of a Hot Type I key is not changeable by using the Ringing Change Key feature.

Vacant Number Routing

Hot Types I keys and enhanced Hot Type D keys support vacant number routing.

Feature packaging

The Network Intercom (Hot Type D and Hot Type I Enhancements) feature is included in Enhanced Hot Line (HOT) package 70.

For Hot Type I in an ISDN network the following packages are required:

- Advanced ISDN Network Services (NTWK) package 148
- Integrated Services Digital Network (ISDN) package 145
- and at least one of 1.5 Mbps Primary Rate Interface (PRA) package 146, Integrated Services Digital Network Signaling Link (ISL) package 147, 2.0 Mbps Primary Rate Interface (PRI2) package 154, Virtual Network Services (VNS) package 183, and ISDN BRI Trunk Access (BRIT) package 233.

DNPSSS1 connectivity for Hot Type D requires:

- Integrated Digital Access (IDA) package 122, and
- Digital Private Signaling System 1 (DPNSS) package 123.

R2MFC connectivity for Hot Type D requires:

- Multifrequency Compelled Signaling (MFC) package 128.

Network Message Services - Message Centre

Network Message Services - Message Centre extends the Message Centre function to permit remote locations in a network to be served by a single message centre.

Software messages are transported among Meridian 1 PBXs over ISDN Primary Rate Interface or ISDN Signaling Link. These permit a caller to a station on one Meridian 1, to leave a message with a Message Centre located on another Meridian 1 in the network. All existing Message Centre (digital and ACD) sets are supported.

Network Message Services - Message Centre was developed to be transparent to end-users. Callers using sets without displays to a busy or unanswered station are not aware that their call is being answered by a message centre at another PBX location. The message centre sends an indication of a message waiting to the user's station using the same mechanisms as are in place today. Users can call and retrieve their messages by simply pressing the Message Waiting Key on their telephone sets.

Package Requirements

This option requires the following feature packages:

Feature	Package Number
ISDN	145
PRI2 or ISL	154 or 147
NTWK	148
NMS	175

Note: If an ACD type Message Centre is to be used, ACDA package #45 is required. The combination of packages # 46 (MWC), #40 (BACD), # 45 (ACDB) and # 42 (ACDC) provides ACD features and reports for the ACD type Message Centre. If an ACD type message centre is used, then ACD-A, Option 45 is required. NMS Option 175 is required at all sites.

Network Message Services - Meridian Mail

Description

Network Message Services - Meridian Mail extends the Network Message Services capability by supporting the use of a single Meridian Mail voice processing system for multiple network sites. Network Message Services, offers Meridian 1 ISDN network users complete call coverage regardless of their location in the customer's ISDN network. Users at any location in the network, may access the Meridian Mail system as easily as if it were at their own locations.

Option 11 systems may utilize this feature; however, only as an originating site or a tandem node, not as the location which actually possesses the Meridian Mail hardware (i.e. the terminating node).

Access to Meridian Mail features is unchanged, regardless of whether Meridian Mail is being accessed by a user at the same location as Meridian Mail or by a user at a different location from the Meridian Mail system.

For example, with Network Message Services, an unanswered call to a user is forwarded across the ISDN network to Meridian Mail, which plays the user's mailbox greeting and takes the message. Meridian Mail then directs the Meridian 1/SL-1 serving the user to check the status of the user's message waiting light and turn the light on if necessary (message waiting indication for users of analog (500/2500 type) telephones is supported as well). The user accesses Meridian Mail and retrieves messages as usual.

Meridian Mail uses its unique link with the Meridian 1/SL-1 to match calling and called number information to user mailbox information as appropriate. Customers can centralize administration of Meridian Mail resources while still offering voice messaging to users throughout the network. Network Message Services enables customers to offer voice messaging to smaller network locations that might not be able to justify the purchase of a dedicated Meridian Mail system. All users have complete feature functionality including message waiting indication and personal verification across the network.

Network Message Services complements other Meridian Mail options such as Meridian Mail networking, allowing customers maximum flexibility in providing voice messaging services to network users. Furthermore, shared and dedicated Meridian Mail systems can be linked via Meridian Mail networking, providing complete voice messaging access for all users on the network.

Network Message Services extends the value of ISDN Primary Rate Interface and ISDN Signaling Link networks by using the intelligence in these networks to enable shared access to Meridian Mail resources, better using these resources and offering additional services to network users.

Note: Password Suppression is not supported across a Meridian 1 network.

Package Requirements

Requirements for NMS - Meridian Mail vary depending on the switch location in the network. The Meridian 1/SL-1 that originates the call to the Meridian Mail location is the “originating node”. The Meridian 1/SL-1 where the call terminates (i.e. where the Meridian Mail is physically located) is the “terminating node”. An intermediate node that lies between the originating and terminating nodes is a “tandem node”.

Note: An Option 11 may only be configured as either the originating or tandem node, not the terminating node.

Prerequisites:

Originating node prerequisites are as follows:

Feature	Package Number
EES	10
BACD	40
ACDA	45
MWC	46
ISDN	145
NTWK	148
NMS	175

Tandem node prerequisites are as follows:

Feature	Package Number
EES	10
ISDN	145
PRI2 or ISL	154 or 147
NTWK	148

Meridian Mail prerequisites are as follows:

Feature	Package Number
IMS	35
BACD	40
ACDA	45
MWC	46
CSL	77
CDRQ	83
APL	109

Note: Meridian Mail must be equipped with Release 7 software and the Network Message Services software option package. The Meridian Mail system should also be engineered to support the total number of users expected to be served throughout the network. Generic X11 Release 16 is a prerequisite.

Network Ring Again

Description

Network Ring Again (NRAG) provides ring-again capability within a customer's private network equipped with PRI/ISL. In other words, NRAG allows ring again to be activated between an Option 11 in location A and another Meridian 1/SL-1 in location B (see Figure 58).

This extension of Meridian 1 Ring Again operates on telephones called over the ISDN PRI/ISL network.

This feature allows a caller (station "A") encountering a busy destination (station "B") anywhere in the ISDN network to be notified when the busy destination ("B") becomes available. After notification, when user "A" accepts this recall, a normal call setup is then placed to destination "B" without user "A" re-dialing the called number.

Network Ring Again (NRAG) can be originated by a station in the Make Set Busy mode, and it is also allowed to be activated against a station in the Make Set Busy Mode.

NRAG can operate in concurrence with the Ring Again on No Answer feature, described later in this document.

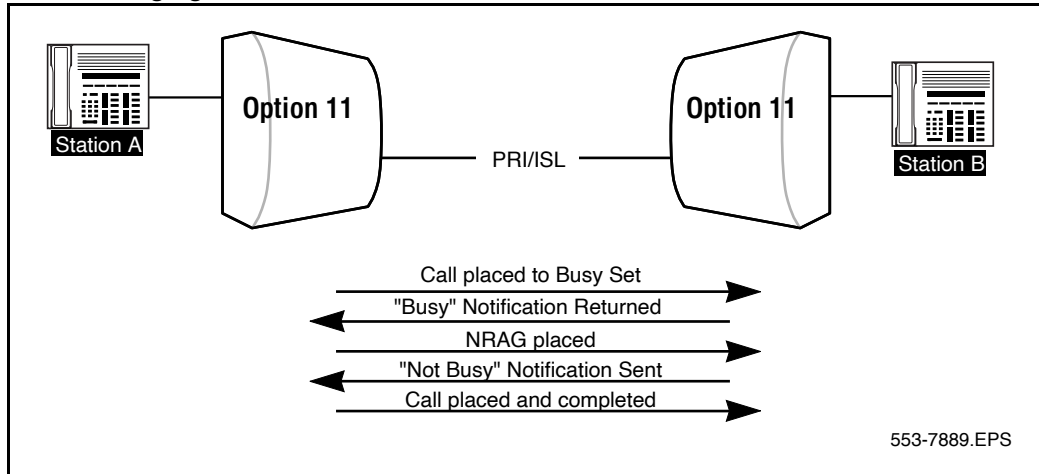
The NRAG feature operates in "free notification" mode. This means the terminating switch determines when the called party becomes idle and notifies the originating switch.

When more than one caller activates NRAG on a terminating station, the calls are queued on a first-come, first-served basis. When the called station becomes idle, only the first caller in the queue is signaled. The second caller in the queue is signaled only after the Queue Advance Timer expires.

Once the NRAG feature has been activated, the status of the called party is observed until:

- the line is idle and a new call set-up can be attempted
- or until NRAG is deactivated by a timeout or manually disconnected by the caller

Figure 58
Network Ring Again



Interactions and Limitations

NRAG is supported in UDP and CDP networks. Network Ring Again is only supported if the CLID uses the prime DN. For Meridian 1/SL-1 Options 21-71, X11 Releases 15 and later support up to a 10 digit CDP with the DNXP package equipped. Option 11 systems support up to a 10-digit CDP.

NRAG does not operate on redirected calls. As well, NRAG cannot be activated on an Automatic Call Distribution (ACD) termination because ACD users are serviced by the ACD queue.

Package Requirements

This option requires the following feature packages:

Feature	Package Number
ISDN	145
PRI2 or ISL	154 or 147
NTWK	148

Network Tenant Service

Description

This feature provides the capability of Network Attendant Service (NAS) on a tenant basis, with network wide tenant to tenant and tenant to trunk blocking.

Attendant routing

NTS allows Console Presentation Groups (CPGs) and related attendants to be defined at any node in the ISDN.

- Calls can be directed to another node based upon:
- a time of day schedule defined at the CPG level
- overflow conditions
- status of local attendants in the CPG group
- status of trunks to other attendant locations
- status of the NAS key on a customer basis
- night service indicator of the CPG group

Each attendant console group (ACG) can have its own NAS routing table defined. Calls not associated with a particular console group are presented to ACG 0.

Tenant to tenant blocking

It is possible to define tables which allow or deny calls between tenant groups in the same customer. Normal intercept treatment is given to a station attempting to dial into a denied access tenant.

Tenant to tenant blocking is maintained across the network. Access is allowed or denied according to information defined on the node which currently processes the call.

It is possible with network-wide tenant-to-tenant blocking to define one-way restrictions (a cannot call B, but B can call A), even though this is not possible in standalone cases. If nodes do not have the same tenant to tenant restrictions defined, one-way blocking can result.

Timed reminder recalls

Timed Reminder Recalls work in the same way as in NAS, but on a tenant basis. The attendant is chosen based on the tenant number of the target party.

Interactions and Limitations

To ensure correct call routing, all nodes in the ISDN must have the Network Tenant Service feature equipped.

Each ACG can have a NAS routing table, or several ACGs may share the same NAS routing table. However, the NAS key is defined only on a customer basis—it is not a CPG-level parameter. When one attendant console activates the NAS key, regardless of ACG, the entire system is in NAS service.

The Multi-Location Business Group (MBG) number is the grouping agent used to associate users belonging to a customer group or ‘sub-group’. Network Tenant Service (NTS) uses MBG numbers to define the relationship between the Multi-Location Business Group Subgroup (MBGS) number and the tenant number on a particular node. The tenant number related to an outgoing call is translated into the corresponding MBGS number and then transmitted through the network. At the terminating node, the MBGS number is translated back into a meaningful tenant number.

Package Requirements

This option requires the following feature packages:

Feature	Package Number
BRTE	14
NCOS	32
NARS	58
TENS	86
ISDN	145
PRI2	154
NAS	159

Note: Any packages needed to support ISDN PRI are needed for Network Tenant Service. Package AOP (#56) must not be equipped.

Network Time Synchronization

Description

The Network Time Synchronization feature allows nodes in an ISDN to query one another for time checks. Slave nodes query a Master node for its clock, and the Master can in turn query a backup.

A time difference (delta) is provided for every node, to allow for time zone differences. In this way, local time is maintained for local reference, and centralized (global) time is maintained for those features which need it (for instance, centralized CDR).

On Slave nodes, requests for time checks are sent automatically either in the background routines, with the daily routines, or every time a time change is made, and on every sysload and initialization.

On the Master node, requests are sent to a backup node upon initialization. The Master does not synchronize Slave nodes during these backup periods.

Interactions and Limitations

This feature has the following limitations:

- When the Master and Backup nodes both request synchronization at the same time, the real time is considered to be the one on the Backup node, so long as that node is not initializing. If the Master is sysloading and the Backup is only initializing, the Backup also carries the real time. In all other cases, the Master continues to carry the real time.
- Should both the Master and the Backup node sysload at the same time, Network Time Synchronization is halted until the Master node has reset its clock.

Each node must be defined as either a Master, a Slave or a standalone node (meaning this feature is not used). A customer must also be selected for the feature (one customer on each node must be chosen, to allow calls to be made between nodes for the time checks). Local access codes with the virtual DN, the Master (or Slave) routing digits, the time difference between the local and the Master node, and the request mode must all be defined. (The time delta and the requesting modes are optional.) This is done through overlay program 2.

Network Wide Listed Directory Numbers

Listed Directory Numbers (LDNs) can be defined as Incoming Call Identification (ICI) keys on an Attendant Console, making it possible to have different presentations when different DN's are dialed. Previously, it was only possible to define four LDNs on a Meridian 1. This feature makes it possible to define six LDNs on a Meridian 1.

Prior to Release 20, when an LDN call was routed from one node to another, the call was presented as per trunk type (e.g., NDID, NTIE, NCO, NFEX, or NWAT). The call was presented on LDN key zero if none of these ICIs was configured. With this feature, if the dialed DN is an LDN and an LDN key exists corresponding to the dialed LDN, the call will be presented on that ICI LDN key.

This feature also enables LDNs to be recognized network wide when Network Attendant Service (NAS) is used. The same LDNs must be configured in multiple nodes. Network LDN is defined on a customer basis.

Operating parameters

The network part of this feature works in a Meridian Customer Defined Network (MCDN) environment with NAS configured.

The LDNs to be used network wide cannot be used in conjunction with Distant Steering Codes.

No new hardware is required for this feature.

Feature interactions

Call Forward No Answer

With this feature, the LDN ICI has a higher priority than CFNA ICI. When a call is forwarded to an LDN via Flexible DN, the call will be presented on the LDN ICI.

Departmental Listed Directory Number

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

Console Presentation Group

This feature provides two more LDNs per Console Presentation Group.

Console Operation/Console Presentation

Console Operation makes it possible for each console to select which ICI call types will be presented to the console. Network wide LDN does not work with the Console Presentation feature because it is not supported by NAS. Console Operation can, however, be configured with two new LDNs.

Network Message Center

With this feature, the LDN ICI has a higher priority than MWC ICI. When a call is forwarded to an LDN over the network to a message center, the call will be presented on the LDN ICI.

Network Attendant Service

The way the network LDN calls are presented in a NAS environment is changed by this feature. The presentation on the NDID, NTIE, NCO, NFEX or NWAT, and the LDN0 key is changed to the correct LDN key, if it exists. Otherwise, it will be presented as it previously was on the NDID or LDN0 key.

Console Operation/Queue Thermometer

The queue thermometer indicates how many calls are in the queue for a certain ICI key. An ICI key can correspond to more than one ICI type. Even though the ICI type of a call may be different with or without this feature active, it will not interact with queue thermometer operations.

Centralized Attendant Service

Centralized Attendant Service (CAS) is mutually exclusive to the NAS package. As the network wide LDN feature requires NAS for its networking functions, the network part of this feature will not work with CAS, but the two extra LDNs can be used locally.

Feature packaging

Since Network Wide LDN requires Network Attendant Service routing, the following existing software packages must be provisioned: Network Attendant Service (NAS) package 159; Network Alternate Route Selection (NARS) package 58; Network Class of Service (NCOS) package 32; Basic Routing (BRTE) package 14; and applicable ISDN options depending upon customer requirements.

To use the attendant queue thermometer, Console Operations (COOP) package 169 must be provisioned.

For Departmental LDN to be configured with six LDNS, Departmental LDN (DLDN) package 76 must be provisioned.

NI-2/QSIG Compliance Update

The NI-2/QSIG Compliance Update feature makes the X11 Release 24.2x NI-2 and QSIG interfaces compliant with the latest standards documents.

Compliance with the standards has evolved as follows:

- ETSI QSIG basic call functionality now complies with the ETS 300-172, fourth edition (1997) standards document. Initially, support for the ISDN ETSI QSIG interface and the basic call capability was introduced in X11 Release 20. This version of the interface was based on the ETS 300-172, first edition (1990) document.
- ISO QSIG basic call functionality now complies with the North American standards document, ISO/IEC 115172 second edition. Initially, support for the ISDN ISO QSIG interface and the basic call capability was introduced in X11 Release 20. This version of the interface was based on the ISO/IEC 115172 first edition document.
- National ISDN-2 (NI-2) basic call functionality now complies with the latest North American standards document, Bellcore TR-NWT-001268.

The NI-2/QSIG Compliance Update feature changes the way the Meridian 1 handles a recoverable error. Instead of sending incorrect call state information, the Meridian 1 now sends a correct call state in the STATUS message.

The mandatory parts of the STATUS message are: CAUSE (indicates type of error) and CALL STATE. Whenever the Meridian 1 encounters a recoverable error it sends a STATUS message. Recoverable errors can be cases such as:

- the IE received is non-mandatory and unrecognized
- the IE is non-mandatory and has invalid contents

In cases such as these, the basic call can still be processed, without affecting basic services, if the call state sent does not hinder this process. All the protocol must do is report the errors.

Before the protocol compliancy was updated, the Meridian 1 sent the call state information as it existed before receiving the incoming call message with the recoverable error. This was not appropriate for other-vendor switches which dropped the call when the incorrect call state information was received. Now, the Meridian 1 sends the call state information, adjusted after the incoming call message was received. The adjusted call state information allows the other vendor switch to proceed with the basic call.

Operating parameters

A hardware upgrade is not required for this feature.

Fatal errors and errors in non-call-associated messages (such as maintenance messages) are not affected by this feature. These errors continue to use the existing error reporting mechanism.

This feature applies to both Basic Rate Interface and Primary Rate Interface connections.

Feature interactions

There are no feature interactions specified for this feature.

Feature packaging

The following packages are required for the ISDN PRI QSIG interface:

- ISDN Signaling (ISDN) package 145
- Primary Rate Interface (PRA) package 146
- 2 Mbps PRI (PRI2) package 154
- International PRA (IPRA) package 202
- Multi-Purpose Serial Data Link (MSDL) package 222
- QSIG Interface (QSIG) package 263

The following packages are required for the ISL QSIG interface:

- ISDN Signaling (ISDN) package 145
- Primary Rate Interface (PRA) package 146
- ISDN Signaling Link (ISL) package 147
- 2 Mbps PRI (PRI2) package 154

- Multi-Purpose Serial Data Link (MSDL) package 222 (for ISL on PRI only)
- QSIG Interface (QSIG) package 263

The following packages are required for the ISDN BRIT QSIG interface:

- ISDN Signaling (ISDN) package 145
- ISDN Basic Rate Interface (BRI) package 216
- Multi-Purpose Serial Data Link (MSDL) package 222
- ISDN BRI Trunk Access (BRIT) package 233
- QSIG Interface (QSIG) package 263

The following packages are required for the PRI NI-2 interface:

- ISDN Signaling (ISDN) package 145
- Primary Rate Interface (PRA) package 146
- Multi-Purpose Serial Data Link (MSDL) package 222
- National ISDN-2 Interface (NI-2) package 291

Feature operation

There are no operating procedures specified for this feature.

Non-Associated Signaling Channels (nB+D)

Description

On an Option 11 system, the D-channel has the capability of supporting multiple PRIs (up to 8) in an nB+D environment with a minimum of 30 to a maximum of 240 B-channels. The 240 B-channel restriction is derived from the calculation of 8 (PRIs) x 30 (B-channels per PRI). This enhancement allows a PRI facility to consist of 30 B-channels, while the D-channel controlling these B-channels is being carried by, and shared with another PRI link.

Package Requirements

This option requires the following feature packages:

Feature	Package Number
ISDN	145
PRI2 or ISL	154 or 147

NPI and TON in CDR Tickets

The Numbering Plan Identification (NPI) and Type Of Number (TON) in Call Detail Recording (CDR) Tickets feature allows NPI and TON information to be optionally displayed on the third line of CDR tickets. NPI and TON are associated with Calling Line Identification (CLID) information, and are useful for billing incoming calls to the originating party.

The NPI and TON are only displayed for calls on an incoming ISDN trunk. Also, the New Format CDR (FCDR) prompt must be set to NEW and the Calling Line Identification (CLID) prompt must be set to YES in Overlay 17. Figure 59 illustrates the format of the third line of a CDR ticket.

Figure 59
Third line format for CDR ticket

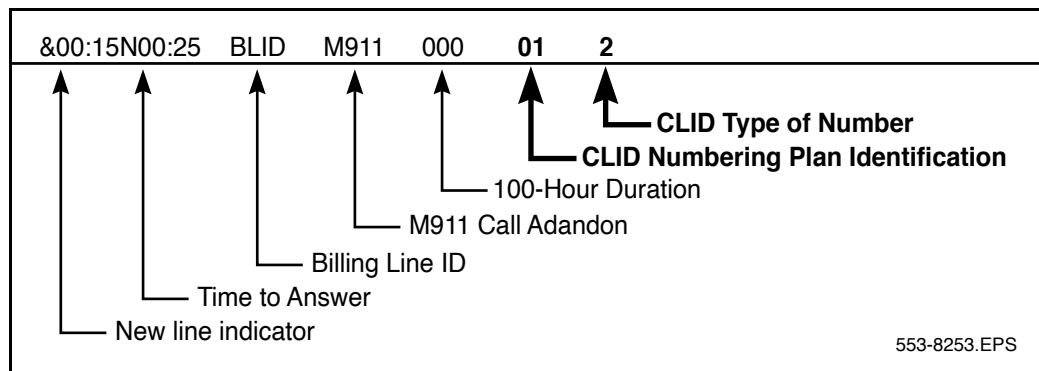


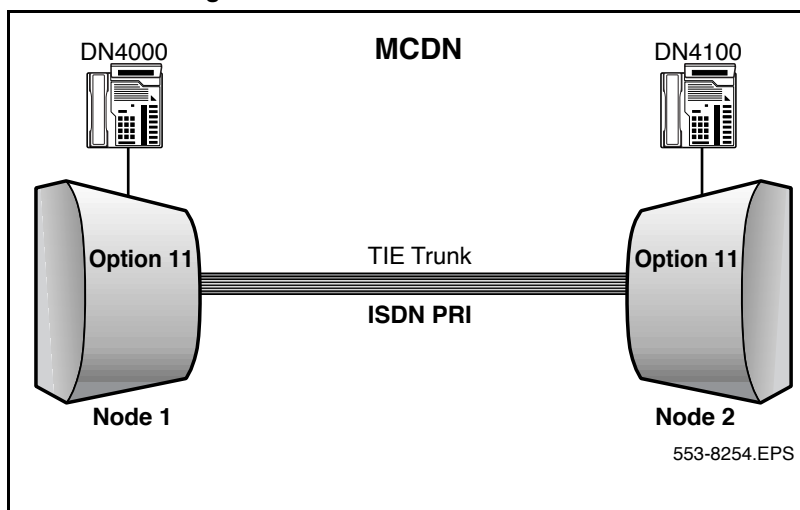
Table 44 describes the contents of the third line of CDR tickets.

Table 44
Third line contents of CDR tickets

Line	Position	Field	Field Definition
3	1	blank	
3	2	&	New line indicator
3	3-7	TTA	Time To Answer (Total ringing time)
3	8	REDIR/B	Time To Answer (Redirection Indicator)/Busy Tone Identifier
3	9-13	TWT	Time To Answer (Total Waiting Time)
3	14	blank	
3	15-30	BLID...xxx	Billing Line ID
3	31	blank	
3	32-38	ABANDON	M911 Call Abandon Tag
3	39	blank	
3	40-42	000	100-Hour Duration
3	43	blank	
3	44-45	NPI	CLID Numbering Plan Identification
3	46	blank	
3	47	TON	CLID Type Of Number
3	48	blank	

Figure 60 illustrates a scenario in which DN 4000 (on Node 1) places a call to DN 4100 (on Node 2) over a Meridian Customer Defined Network (MCDN), using a Coordinated Dialing Plan (CDP) Distant Steering Code (DSC). The call arrives at Node 2 on Route 201 Member 4. A CDR N ticket is produced when the call is disconnected.

Figure 60
Scenario involving a call over a Meridian Customer Defined Network



The CDR N ticket produced in the above scenario has the following format:

N 001 02 T201004 DN4100 02/10 18:07:15:00:02:24.0 & 4000XXXXXXXXXXXX &00:15N00:25 BLIDXXXXXXXXXXXX 000 09 6

On line 3 of the ticket, the NPI value of “09” represents a private numbering plan. The TON value of “6” represents an Electronic Switched Network (ESN) Customer Dialing Plan (CDP). Please refer to Tables 45 and 46 for the NPI and TON information for an MCDN incoming trunk.

When an incoming call arrives on the Meridian 1, NPI and TON are sent in the calling party Information Element (IE) and are mapped into internal values. The correspondence between the Meridian 1 values and the values given in the specifications are described in Tables 45 to 54.

Tables 45 to 54 show the information printed in the CDR ticket, depending on the incoming trunk protocol. As shown in these tables, not all combinations of NPI and TON exist.

In the TON tables, only ISDN/Telephony numbering plan (Rec. E.164/E.163) and private numbering plans are detailed. For all other supported NPI values, TON has the value of “unknown number”.

Referring to the scenario in Figure 60, Tables 45 and 46 show the NPI and TON information for an MCDN incoming trunk.

Table 45
NPI information printed in the CDR ticket for an MCDN incoming trunk

NPI code in CDR	Corresponding value of NPI in specification
00	000 - unknown numbering plan
01	0001 - ISDN/Telephony numbering plan (Rec. E.164)
02	not used
03	not used
04	not used
08	not used
09	1001 - private numbering plan

Table 46
TON information printed in the CDR ticket for an MCDN incoming trunk

TON code in CDR	Corresponding value of TON in specification	
	NPI = ISDN/Telephony numbering plan (Rec. E.164)	NPI = private numbering plan
0	0000 - unknown number	0000 - unknown number
1	0001 - international number	not used
2	0010 - national number	not used
3	not used	0011 - ESN SPN
4	0100 - local number	not used
5	not used	0101 - ESN LOC
6	not used	0110 - ESN CDP

Tables 47 and 48 show the NPI and TON information for a EuroISDN incoming trunk.

Table 47
NPI information printed in the CDR ticket for a EuroISDN incoming trunk

NPI code in CDR	Corresponding value of NPI in specification
00	0000 - unknown
01	0001 - ISDN/Telephony numbering plan (Rec.E.164/E.163)
02	not used
03	0011 - data numbering plan (Rec.X.121)
04	0100 - telex numbering plan (Rec.F.69)
08	1000 - national standard numbering plan
09	1001 - private numbering plan

Table 48
TON information printed in the CDR ticket for a EuroISDN incoming trunk

TON code in CDR	Corresponding value of TON in specification	
	NPI = ISDN/Telephony numbering plan (Rec. E.164)	NPI = private numbering plan
0	000 - unknown or 110 - abbreviated number	000 - unknown or 110 - abbreviated number or 001 - level 2 regional number
1	001 - international number	cannot be mapped
2	010 - national number	010 - level 1 regional number
3	011 - network specific number	011 - network specific number
4	100 - subscriber number	100 - subscriber number
5	not used	not used
6	cannot be mapped	cannot be mapped

Tables 49 and 50 show the NPI and TON information for a QSIG incoming trunk.

Table 49
NPI information printed in the CDR ticket for a QSIG incoming trunk

NPI code in CDR	Corresponding value of NPI in specification
00	0000 - unknown
01	0001 - ISDN/Telephony numbering plan (Rec.E.164/E.163)
02	not used
03	0011 - data numbering plan (Rec.X.121)
04	0100 - telex numbering plan (Rec. F.69)
08	1000 - national standard numbering plan
09	1001 - private numbering plan

Note: QSIG refers to ISO QSIG and ETSI QSIG.

Table 50
TON information printed in the CDR ticket for a QSIG incoming trunk

TON code in CDR	Corresponding value of TON in specification	
	NPI = ISDN/Telephony numbering plan (Rec. E.164)	NPI = private numbering plan
0	000 - unknown or 110 - abbreviated number	000 - unknown or 110 - abbreviated number or 001 - level 2 regional number or 101 - level3 regional number
1	001 - international number	cannot be mapped
2	010 - national number	010 - level 1 regional number
3	011 - network specific number	011 - PTN specific number
4	100 - subscriber number	100 - local number
5	not used	cannot be mapped
6	cannot be mapped	cannot be mapped

Note: QSIG refers to ISO QSIG and ETSI QSIG.

Tables 51 and 52 show the NPI and TON information for a non-UIPE and non-MCDN incoming trunk.

Table 51
NPI information printed in the CDR ticket for a non-UIPE and non-MCDN incoming trunk

NPI code in CDR	Corresponding value of NPI in specification
00	0000 - unknown numbering plan
01	0001 - Rec. E.164
02	0010 - Rec. E.163
03	0011 - Rec. X.121
04	0100 - Telex numbering plan
08	1000 - national numbering plan
09	1001 - private numbering plan

Note: Non-UIPE refers to the 1TR6, AXE-10 for Australia and Sweden, Swissnet 2, Numeris VN4, SYS-12, and D70 connectivities.

Table 52
TON information printed in the CDR ticket for a non-UIPE and non-MCDN incoming trunk

TON code in CDR	Corresponding value of TON in specification	
	NPI = ISDN/Telephony numbering plan (Rec. E.164)	NPI = private numbering plan
0	0000 - unknown number ¹	0000 - unknown number ¹
1	0001 - international number ²	not used
2	0010 - national number ²	not used
3	not used	0011 - network specific number ²
4	0100 - subscriber number ²	not used
5	not used	not used
6	not used	0110 - abbreviated number ²

1. For SYS-12, AXE-10 for Australia and Sweden, Swissnet, Numeris VN4, and D70 interfaces, all received values are mapped into unknown code.

2. For all interfaces not mentioned in 1.

Tables 53 and 54 show the NPI and TON information for an NI-2 incoming trunk.

Table 53
NPI information printed in the CDR ticket for an NI-2 incoming trunk

NPI code in CDR	Corresponding value of NPI in specification
00	0000 - unknown numbering plan
01	0001 - ISDN/Telephony numbering plan (Rec. E.164)
02	unused
03	unused
04	unused
08	unused
09	1001 - private numbering plan

Table 54
TON information printed in the CDR ticket for an NI-2 incoming trunk

TON code in CDR	Corresponding value of TON in specification	
	NPI = ISDN/Telephony numbering plan (Rec. E.164)	NPI = private numbering plan
0	not used	not used
1	001 - international number	not used
2	010 - national number	not used
3	not used	not used
4	100 - local number	100 - subscriber number
5	not used	not used
6	not used	not used

Operating parameters

The NPI and TON in CDR Tickets feature applies only for incoming ISDN trunk calls. NPI and TON information depends on the incoming trunk protocol.

The NPI and TON fields are left blank for internal calls, outgoing trunks, incoming non-ISDN trunks, or if the CLID prompt is set to NO.

When the FCDR prompt is set to OLD, the NPI and TON fields do not exist, regardless of how the CLID prompt is defined.

NPI and TON information is available with the following incoming interfaces: EuroISDN, QSIG (ISO and ETSI), MCDN, non-UIPE and non-MCDN, and NI2.

NPI and TON information is included in all types of CDR records that contain CLID information.

NPI and TON information is lost when system initialization occurs and the call is then reconstructed. In this case, if the call involves an incoming ISDN trunk and if the NPI and TON in CDR Tickets feature is configured, the NPI field contains two zeros (00) and the TON field contains one zero (0), regardless of the NPI and TON sent at call setup.

Feature interactions

The NPI and TON in CDR Tickets feature does not have any specific interactions with other features.

Feature packaging

The NPI and TON in CDR Tickets feature requires the following packages:

- Call Detail Recording (CDR) package 4
- Call Detail Recording on Teletype Machine (CTY) package 5
- New Format Call Detail Recording (FCDR) package 234
- Calling Line Identification in Call Detail Recording (CCDR) package 118

Feature operation

No specific operating procedures are required to use this feature.

Network Signaling on Virtual Network Services

The Network Signaling on Virtual Network Services (NSIG on VNS) feature enhances the functionality of Virtual Network Services by supporting Network Signaling on the VNS D-Channel (VNS_DCH).

With VNS, the Public Switched Telephone Network (PSTN) trunk serves as a B-Channel. However, NSIG is not allowed over PSTN trunks; therefore, it is necessary to send NSIG over the D-Channel in order to maintain the Network Class of Service (NCOS) information.

NCOS information is transported via Network Signaling over routes, depending on the value of the Signaling arrangement (SIGO) prompt in the Route Data Block. When SIGO = STD, no NCOS information is sent on the Bearer Channel (B-Channel).

The value of the Virtual Signaling (VSIG) prompt in Overlay 17 determines whether or not NSIG is supported on the VNS D-Channel. When VSIG = NO, NSIG is not sent over the VNS D-Channel. When VSIG = YES, NSIG is sent over the VNS D-Channel, and the standard ESN5 protocol is used.

VSIG = YES and SIGO is set to STD

In this situation, NSIG is configured on the VNS D-Channel. Two nodes are connected by a VNS D-Channel, and the bearers are TIE trunks connected to the Central Office (CO). When Set A initiates a call to Set B, using the VNS D-Channel, the NCOS of Set A travels over the VNS D-Channel using ESN5 protocol and is present at the terminating node for further access restrictions. No Network Signaling is on the bearer in this case.

VSIG = YES and SIGO is not set to STD

In this situation, NSIG is configured on the VNS D-Channel and on the bearer. When Set A initiates a call to Set B, the NCOS of Set A travels over the VNS D-Channel and on the bearer.

VSIG = NO and SIGO is not set to STD

In this situation, NSIG is not configured on the VNS D-Channel. When Set A initiates a call to Set B, no NCOS information travels over the VNS D-Channel, but will travel on the bearer.

Scenarios involving Satellite (SAT) Routes

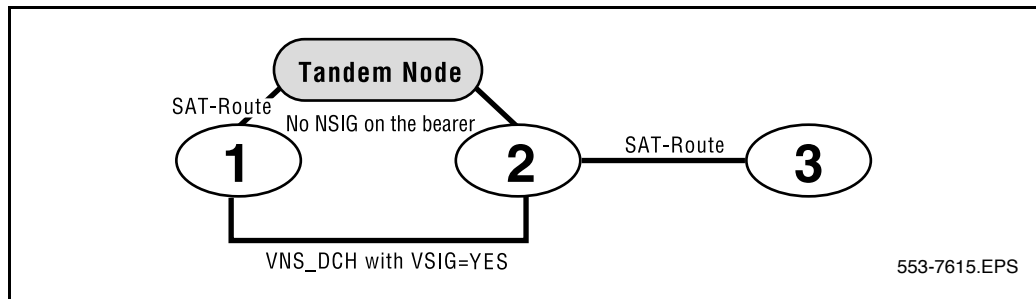
When Satellite Routes are involved, NSIG on VNS behaves as shown in the following scenarios.

VSIG = NO and SIGO is not set to STD

Referring to Figure 61, a call from Node 1 to Node 3 passes a Satellite Route. The route between Node 2 and Node 3 is also a Satellite Route; therefore, the call is blocked at Node 2.

Figure 61

A scenario involving two SAT-Routes where VSIG = NO and SIGO is not set to STD

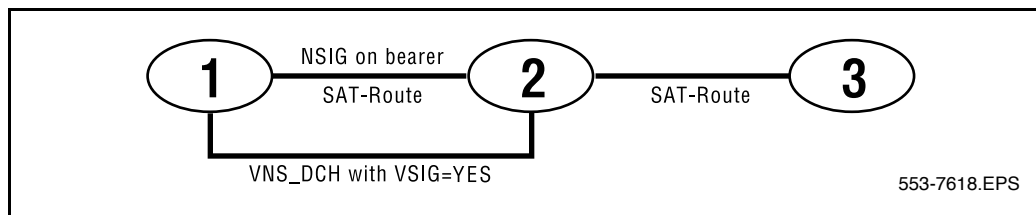


VSIG = YES and SIGO is not set to STD

Referring to Figure 62, a call from Node 1 to Node 3 passes a Satellite Route. The route between Node 2 and Node 3 is also a Satellite Route; therefore, the call is blocked at Node 2. The Satellite information is transported over the bearer.

Figure 62

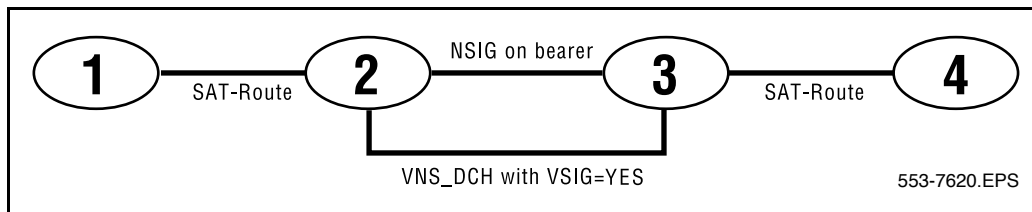
A scenario involving two SAT-Routes where VSIG = YES and SIGO is not set to STD



Referring to Figure 63, a call from Node 1 to Node 4 passes a Satellite Route. The Satellite Count is updated at Node 2, and the Satellite information is sent to Node 3. The route between Node 3 and Node 4 is also a Satellite Route; therefore, the call is blocked at Node 3.

Figure 63

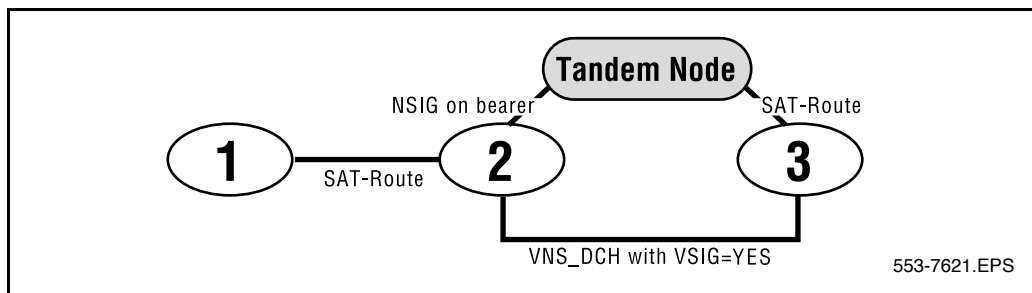
A scenario involving two SAT-Routes where VSIG = YES and SIGO is not set to STD



Referring to Figure 64, a call from Node 1 to Node 3 is initiated, and the call is blocked at the Tandem Node. In this scenario, the Satellite information is sent from Node 2 to the Tandem Node. The route between the Tandem Node and Node 3 is also a Satellite Route; therefore, the call is blocked on the bearer side of the Tandem Node.

Figure 64

A scenario involving two SAT-Routes where VSIG = YES and SIGO is not set to STD

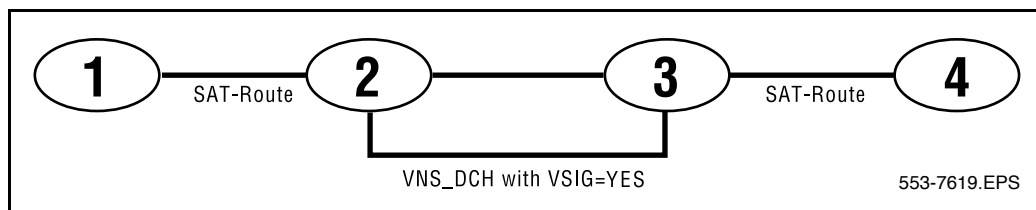


VSIG = YES and SIGO is set to STD

Referring to Figure 65, a call from Node 1 to Node 4 passes a Satellite Route, and the Satellite Count is updated at Node 2. The Satellite information travels over the VNS D-Channel to Node 3. The route between Node 3 and Node 4 is also a Satellite Route; therefore, the call is blocked at Node 3.

Figure 65

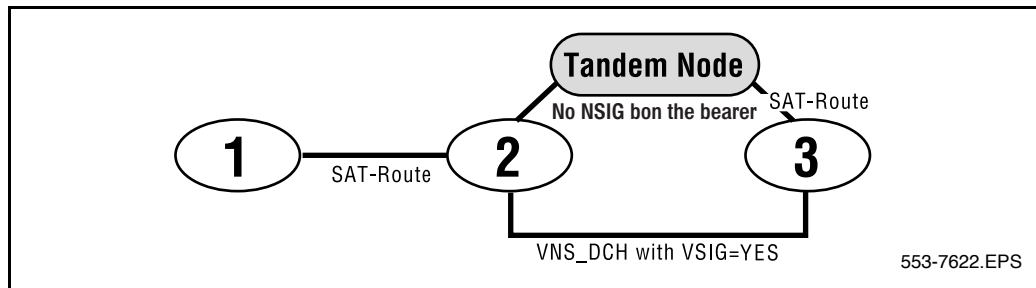
A scenario involving two SAT-Routes where VSIG = YES and SIGO is set to STD



Referring to Figure 66, if NSIG is not configured on the bearer side, the Satellite information is not sent over the bearer. A call from Node 1 to Node 3 passes even if there is a second Satellite Route involved in the call. At Node 3, the only Satellite information received is that which is received over the VNS D-Channel.

Figure 66

A scenario involving two SAT-Routes where VSIG = YES and SIGO is set to STD



Operating parameters

This feature does not support Call-Back Queuing.

When the NSIG on VNS feature is disabled, the previous functionality exists. In this case, no signaling digits are sent over the VNS D-Channel. However, if configured, NSIG is transported over the bearer. With the NSIG on VNS feature configured, the bearer is transported over the VNS D-Channel, and the signaling digits are present on the termination node.

A non-symmetric configuration is supported for bearer trunks.

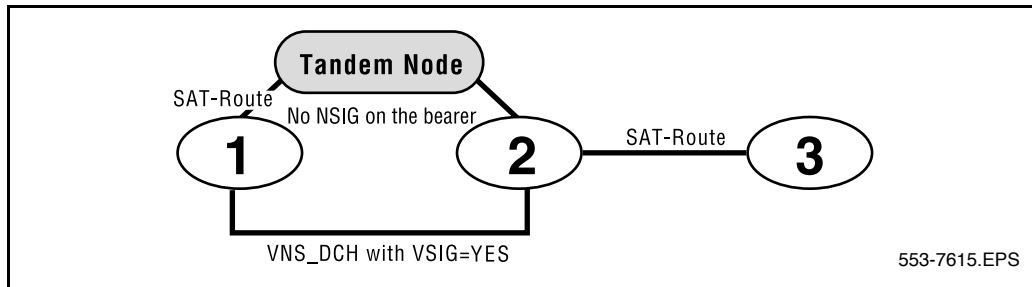
The VNS D-Channel only supports the NSIG ESN5-protocol. It does not support EN19 and ETN protocols.

For signaling, VNS requires a D-Channel link. The following hardware is also required:

- Clock Controller (QPC775)
- Network card (QPC414)
- PRI2 pack (NT8D72)
- DCH card (NT6D11)

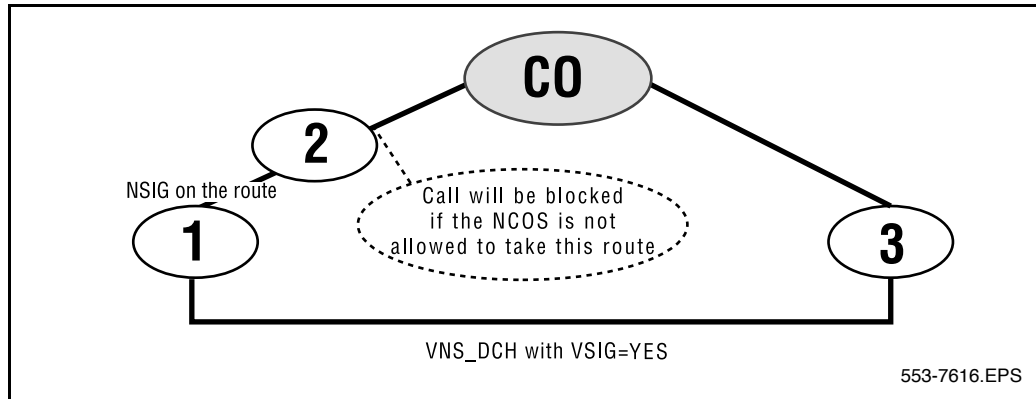
Referring to Figure 67, when NSIG is configured only on the VNS D-Channel, the Satellite Count is not updated if a Satellite Route is passed on the bearer side. A call from Node 1 to Node 3 passes even when there are two Satellite Routes involved in the call. When no NSIG information is sent from the Tandem Node to Node 2, the Satellite information is lost.

Figure 67
SAT-count is not updated when NSIG is not configured on the bearer (VSIG = YES and SIGO is set to STD)



Referring to Figure 68, in a case where the bearer passes a Tandem Node (Node 2) with NSIG on the route, the call is blocked at Node 2 when the NCOS is insufficient.

Figure 68
A scenario where the call is blocked, as NCOS is insufficient (VSIG = YES and SIGO is not set to STD)



Feature interactions

NSIG on VNS has no specific interactions with existing features.

Feature packaging

The NSIG on VNS feature requires the following packages:

- Virtual Network Services (VNS) package 183
- Network Alternate Route Selection (NARS) package 58
- Network Class of Service (NCOS) package 32
- Basic Routing (BRTE) package 14
- ISDN Signaling (ISDN) package 145
- ISDN Signaling Link (ISL) package 147
- Advanced Network Services (NTWK) package 148

Feature operation

No specific operating procedures are required to use this feature.

Option 11 Downloadable D-Channel

The Downloadable D-channel card brings MSDL D-channel application to the Option 11 system.

The Downloadable D-channel card (NTBK51) is a daughter board that mounts on the NTBK50 2 Mbit Primary Rate Interface card. The NTBK50/51 can reside in any slot 1-9 of the main cabinet. A single Serial Data Interface (SDI) I/O address (1-15) is used by the DDCH.

Limitations

- Only one selected software application (MSDL or D-channel) can be active on the DDCH card.
- Only one ISDN D-channel related protocol mode is supported. This feature does not provide Serial Data Link (SDI) or Enhanced Serial data Link (ESDI).
- The DDCH is a daughterboard that attaches to the NTBK50 2.0 Mb PRI motherboard. The maximum number of card slots available for the 2.0 Mb PRI card (and therefore the DDCH card) depends on the system configuration. The maximum is nine (from slot 1 to 9).
- The card type of the backup D-channel must be configured the same as the primary D-channel. That is, if the card type (in software) of the primary D-channel is MSDL, the backup D-channel must also be MSDL. Similarly, if the primary D-channel is set to DCHI in software, the backup D-channel must also be DCHI.
- The Downloadable D-channel card does not support ISDN Signaling Link (ISL) or Virtual Network Services (VNS) because it does not have an external port connector. The NTAK02 SDI/DCH card must be used to provide ISL or VNS services.

Feature interactions

2 Mbit Primary Rate Interface

The 2 Mbit Primary Rate Access feature and the connectivities supported in Release 18 will continue to work with both the new and old D-channel interface cards.

1.5/2.0 Mbits ISDN Gateway Feature

The International 1.5/2.0 Mbits ISDN Gateway feature will continue to work with the existing D-channel interface and the new Downloadable D-channel interface.

Release 20 Features

The following features are dependent on this feature to provide support for Option 11: ECMA QSIG PRI; PRI for New Zealand NEAX-61; and EURO ISDN.

For more information about these features, please see the individual features descriptions contained in this document.

Feature packaging

The following packages must be provisioned in the Option 11: Basic System Features (BASIC) package 0; Integrated Services Digital Network (ISDN) package 145; 2 Mbit Primary Rate Access (PRI2) package 154; Meridian 1 Extended Peripheral Equipment (XPE) package 203; and Multipurpose Serial Data Link (MSDL) package 222.

Radio Paging Improvement

A Radio Paging system is a communications tool used to contact mobile parties by means of radio signals. A caller can use their telephone set to page a mobile party who has a mobile portable receiving device.

Prior to Release 20, the Radio Paging (RPA) feature supported attendant recall in standalone operation only. The RPA recalls to the local attendant on the node where the RPA system is directly connected. This product improvement enables RPA to recall the attendant who originated the Radio Paging call only; the attendant may be located anywhere within an ISDN Meridian Customer Defined Network (MCDN) configured with Network Attendant Services (NAS).

The improvement also enables the upgrade the attendant's display with paged name to display paged name instead of answering name on the paging party when answered, and to make network Radio Paging show the same display information as in the standalone operation.

Operating parameters

The originating tandem and paging nodes must be Meridian 1 switches running a minimum of Release 20 software.

For Pre-selection Paging, if the paged DN following the RPAX FFC is not local to the paging node, the Call Party Name Display (CPND) name for this DN cannot be obtained to be displayed on the calling party's terminal. If the paged DN is local on the paging node and has CPND defined, the CPND can be retrieved and sent to the calling party for display purposes. For Post-selection Paging, the CPND of the paged DN will be displayed even if the DN is not local to the paging node.

If a network call comes in to a set on the paging node and is redirected to paging by CFNA, the calling name cannot be retrieved and updated on the answering set when the paging call is answered. This happens only if the set on the paging node has CPND defined. If the set does not have CPND defined, the calling name can updated on the answering party.

The following hardware is required for Radio Paging operation: Radio Paging System equipment meeting European Selective Paging Manufacturers' Association (ESPA) requirements; NT trunk cards (QPC296/QPC287/QPC551/QPC71/QPC237/NTD9742A/NT5K19AA) or XFEM packs (NT5K83/NT5K72/NT5K50/NT5K19).

The following hardware is required for MCDN ISDN: PRA – DCHI and PRI/PRI2 circuit packs; ISL – DCHI circuit packs and TIE trunks; and VNS – DCHI circuit packs and any trunks.

Feature interactions

Call Detail Recording Enhancement

With this Release 20 enhancement to CDR, an “S” record is printed when an attendant extends an outgoing trunk call to a destination party. When the extended outgoing trunk call or the destination party Releases to disconnect, an “E” record is printed. Prior to this enhancement, no CDR record was printed until the extended outgoing trunk call or the destination party Released, when an “N” record was printed. This CDR enhancement also applies with network Radio Paging.

When an attendant makes an outgoing call (established on the source side) and then extends the call to remote radio paging on another node by using a normal trunk (e.g. Trunk Pxxx), an “S” record is printed when the attendant Releases to extend the call to network RPA.

If the outgoing trunk call Releases before the paged call is answered, the “E” record will show the normal trunk ID (TrunkPxxx).

If the paged call has been answered when the outgoing trunk call Releases, the “E” record will show the paged DN instead of Trunk Pxxx.

For more information, please refer to the CDR Enhancement feature description contained in this document.

Display of Calling Party Denied

If the DNPd package is ON (packaged under the International Supplementary features package), additional classes of service can be assigned to sets that determine whether or not their DN and CPND information will be displayed on other sets. No CPND or DN information is displayed on sets involved in a network RPA call that have name display denied or digit display denied class of service.

Slow Answer Recall Modification (SLAM)

With the Slow Answer Recall Modification feature enabled, when the attendant answers a recall the destination party is disconnected. This also applies to Radio Paging.

When the attendant answers a paging recall, the call is removed from the meet-me queue and the recall cannot be answered by the paging party by using RPA Answer. The paging party is put on the source side of the attendant; there is nothing connected on the destination side. The attendant cannot extend the call to paging by pressing the Release key. Pressing the Release key will disconnect the paging party from the source side and the attendant will become idle.

The attendant can extend the call to Radio Paging again by either: dialing the RPAX FFC + the DN (preselection); or dialing the DN, and while the DN is ringing or busy pressing the RPAG key (post-selection).

Network Attendant Services (NAS)

NAS configuration is a requirement for the Network Radio Paging (NRPA) Recall to Same Attendant (RTSA) feature. Without NAS, NRPA RTSA is not active, and existing operation will be followed.

With NAS configured, if a RPA recall to the attendant on the originating node is not allowed, the recall will be presented on the paging node. Existing operation prior to this development is performed. There is no new interaction introduced with NAS features.

Feature packaging

Radio Paging (RPA) package 187 must be provisioned to activate this feature.

To gain access to RPA, Flexible Feature Codes (FFC) package 139 must be provisioned.

For the Radio Paging network recall operation, Network Attendant Service (NAS) package 159 must be provisioned.

For Remote Radio Paging, Coordinated Dialing Plan (CDP) package 59 is required to define RPA FFCs as Distant Steering Codes (DSCs) or Trunk Steering Codes (TSCs).

To display characters instead of the Radio Paging Flexible Feature Code, Calling Party Name Display (CPND) package 95 is required.

Integrated Services Digital Network (ISDN) package 145, and its dependencies, are required for operation in an MCDN ISDN network.

Radio Paging Product Improvement Continuation

A Radio Paging System (RPS) is a communications tool used to contact mobile parties by means of radio signals. With this system, a set can page a mobile party that is equipped with a radio paging device. The Radio Paging Product Improvement Continuation enhances the performance of the Radio Paging feature by providing the following:

- an increase in the number of digits sent to and displayed on a Radio Paging device
- the ability to activate/deactivate Pretranslation for Radio Paging calls
- five internal/external call treatments to a pager installed in the paging rack

Pager Display

With the existing Radio Paging functionality, when Calling Line Identification (CLID) information is sent to a paging device, a maximum of seven digits are displayed on the pager.

With the Radio Paging Product Improvement Continuation, however, up to 16 digits can be displayed on a pager. Therefore, it is possible for the entire CLID information to be displayed. In order to specify the number of digits (0-16) to be sent to the Radio Paging System, the Transmit Caller's DN (TRDN) prompt must be defined in Overlay 58.

Pretranslation

Pretranslation allows the creation of a flexible dialing plan by using Speed Call lists as Pretranslation Tables. With the Radio Paging Product Improvement Continuation, Pretranslation is activated/deactivated for Radio Paging calls by defining the Pretranslation (PRET) prompt in Overlay 58. This activation/deactivation takes place regardless of whether or not Pretranslation is allowed at a customer level.

Pagers installed in the paging rack

With existing Radio Paging functionality, the treatment of external calls forwarded to pagers in the paging rack is defined by the Recall if busy from Radio Paging (RCAL) prompt in Overlay 58. If RCAL is set to NO, the caller receives a busy tone. If RCAL is set to YES, the call is routed to the attendant. When an internal call is forwarded to a pager in the paging rack, the caller receives a busy tone.

With this Product Improvement Continuation, the user chooses what happens to internal/external calls forwarded to a pager in the paging rack. The treatment of these calls is defined by the Treatment for Internal Calls (INTR) and Treatment for External Calls (EXTR) prompts in Overlay 58. The INTR and EXTR prompts replace the RCAL prompt.

CAUTION

With conversion to X11 Release 23 and later software, the treatment for external calls to a pager in the paging rack is **not** converted automatically. Therefore, the EXTR prompt must be defined. If EXTR is not defined, when an external call is forwarded to a pager in the paging rack, the call receives the default treatment for external calls (busy tone).

The Radio Paging Product Improvement Continuation offers the following five possibilities for the treatment of calls to pagers in the paging rack:

- The caller receives a busy tone.
- The call is routed to an attendant.
- The caller receives a special tone (SRC1-SRC8) or an announcement (with RAN equipment) delivered from the Tone and Digit Switch (TDS) card.
- The caller receives an announcement from a RAN machine.
- The call is routed to Meridian Mail.

Busy Tone

When INTR or EXTR is set to BUSY, the caller receives a busy tone.

Routed to an Attendant

When INTR or EXTR is set to ATT, the call is routed to an attendant.

Special Tone or Announcement

When INTR or EXTR is set to SRC1-SRC8, the caller receives a special tone, programmed in Overlay 56, or an announcement. After an announcement is provided to the caller, the call is disconnected. Recorded Announcement (RAN) equipment is required to provide this announcement.

Announcement from RAN

When INTR or EXTR is set to RAN, the caller receives an announcement from a RAN machine and is then disconnected or routed to an attendant after the message is heard. Post RAN treatment is defined by the RAN post announcement treatment (POST) prompt in Overlay 16.

For this enhancement to function, a RAN route must be specified by defining the Route number that provides the Recorded Announcement (RANR) prompt in Overlay 58. The RAN route must be specified prior to defining the RANR prompt.

Meridian Mail

When INTR or EXTR is set to MAIL, the call is routed to Meridian Mail. In this case, the caller receives an announcement stating that the call is being rerouted to Meridian Mail. With this enhancement, all Meridian Mail functions are available.

For this enhancement to function, the Meridian Mail Directory Number (MMDN) prompt must be defined in Overlay 58. Prior to defining the MMDN prompt, however, the Voice Automatic Call Distribution (ACD) messages queue must be defined in Overlay 23. The maximum input for Voice ACD is four digits or seven digits if the Directory Number Expansion (Seven Digit) (DNXP) package 150 is equipped.

Operating parameters

The Radio Paging Product Improvement Continuation is applicable on a stand-alone Meridian 1 switch with a Radio Paging system or in an Integrated Services Digital Network (ISDN) Meridian Customer Defined Network (MCDN) with a centralized Radio Paging System.

A maximum of 16 digits can be sent to Radio Paging equipment, as only 16 digits can be stored in the Calling Line Identification (CLID) field.

As per existing Radio Paging functionality, if the calling number is not available, the Route Access Code of the incoming trunk is displayed on the Radio Paging device.

If the calling number is shorter than the specified value defined at the TRDN prompt, the missing digits are replaced by zeros on the pager's display. With the existing functionality, a shorter calling number is also displayed on a pager in this manner.

If the calling number is greater than the specified value defined at the TRDN prompt, the most significant digits are displayed. The unnecessary digits are deleted.

The treatment of calls to a pager in the paging rack is only applicable if the Radio Paging device conforms to the standards of the European Selective Paging Manufacturer's Association (ESPA).

When the Recorded Paging Announcement (PANN) prompt is set to YES in Overlay 58, each redirected call to the paging equipment receives a recorded announcement stating that the called party is being paged. This announcement is provided even if the pager is in the paging rack.

When a pager is in the paging rack and PANN is set to YES, the caller receives an announcement stating that the pager is in the paging rack. After this announcement, the treatment, as a result of the INTR and EXTR prompts, is performed.

When the INTR or EXTR prompts are set to RAN and all Recorded Announcement (RAN) trunks are busy, the caller receives normal ringback tone. As soon as a RAN trunk becomes available, the caller hears a recorded announcement. This is as per existing RAN functionality.

Meridian Mail must be located on the same node as the paging device, in order for calls to a pager in the paging rack to be re-routed to Meridian Mail. If Meridian Mail and the paging device are not located on the same node, an error message appears at the overlay level.

When INTR or EXTR is set to Mail and the maximum number of calls to the Meridian Mail DN exceeds the limit that was set at the MAXP prompt in Overlay 23, the caller receives normal ringback tone. As soon as the number of calls is less than or equal to the MAXP value, the caller receives the recorded announcement or the defined Meridian Mail function. This is as per existing Meridian Mail functionality.

Feature interactions

Radio Paging Product Improvement Continuation has no specific interactions with existing features.

Feature packaging

The Radio Paging Product Improvement Continuation requires the following packages:

- Radio Paging (RPA) package 187, which requires the following package to access Radio Paging:
 - Flexible Feature Codes (FFC) package 139
- Pretranslation (PXLT) package 92

The following packages are required for Meridian Mail:

- Make Set Busy (MSB) package 17
- Basic Automatic Call Distribution (BACD) package 40
- Automatic Call Distribution Package A (ACDA) package 45
- Command Status Link (CSL) package 77
- Command and Status Link with Alpha Signaling (CSLA) package 85
- Integrated Message System (IMS) package 35
- Message Waiting Center (MWC) package 46
- End-to-End Signaling (EES) package 10
- Directory Number Expansion (Seven Digit) (DNXP) package 150 for the Meridian Mail DN (MMDN) to contain a maximum of seven digits

The following package is required for Recorded Announcement (RAN):

- Recorded Announcement (RAN) package 7

Feature operation

No specific operating procedures are required to use this feature.

Recall With Priority During Night Service

Description

With this feature, incoming calls which would normally be routed to attendants are instead directed to predefined Night Service DN's.

This feature is activated on a customer basis from the attendant console, when the customer is put into night service. Local night service is provided when night service is activated and the NAS key is dark (or where no NAS key is defined and NAS is turned off through administration). Where the NAS key is lit (NAS in effect) and night service is activated, a call may be presented to a remote night station under Drop Back Busy.

For further information on this feature, refer to 553-2301-105, Appendix 2.

Remote Virtual Queuing

Description

Remote Virtual Queuing (RVQ) allows you to perform queuing type functions on a trunk (B-Channel) in your ISDN network. This feature is similar to the ESN features Coordinated Call-Back Queuing (CCBQ), and Call-Back Queuing to a Conventional Main (CBQCM). Prior to operating RVQ, you must have a working knowledge of the ESN features. Refer to *Electronic Switched Network description* (309-3001-100) for a complete description of these features.

Remote Virtual Queuing (RVQ) is supported on a private ISDN PRI/ISL network only.

When an outgoing network path is unavailable, Remote Virtual Queuing (RVQ) provides call back queuing. RVQ scans ahead on the ISDN trunk to be sure the entire path from origination to termination is available. When a path becomes idle, you are notified and can ring the path again to terminate the call.

Operating parameters

In addition to the function offered by CCBQ and CBQCM, RVQ provides the following:

- There is no limit to the number of nodes in the ISDN network that RVQ searches.
- The originating node has control of the call.

RVQ is supported in a private ISDN PRI/ISL network only. Any trunk other than an ISDN PRI or ISL trunk is considered a Non-RVQ Compatible (NRC) trunk. NRC trunks include analog trunks, T1 trunks, or public trunks. NRC also includes trunks connecting to a non-Meridian 1, non-ISDN, or Pre-Release 18 node.

The originating, tandem, and terminating nodes must run X11 Release 18 or later software, and include all the required packages (refer to Feature packaging in this module).

Meridian Digital Telephones , analog (500/2500 type) telephones, and SL-1 sets can activate and receive RVQ calls. Data terminals are also supported. Attendants cannot activate RVQ.

RVQ is activated only by telephones with Ring Again allowed.

AUTHCODE is supported on the first tandem node only.

The RVQ retry timer is set for each NCOS. Setting a specific NCOS with a low retry timer searches for more connections more often. The lower the timer, the greater the chance to connect the call.

Trunk Access codes are not supported.

ACD DN's are supported as a terminating DN.

Attendants are supported as terminating stations.

The maximum total amount of time RVQ searches for available paths is 30 minutes. The countdown begins as soon as the originator activates the Ring Again key.

Only one RVQ Ring Again attempt at a time is allowed. If Ring Again is pressed again, after activating it for RVQ, the most recent number dialed is the one attempted.

RVQ does not check the terminating telephone's status. The dialed DN may be invalid or busy. Once the path is available, the RVQ call can reach the dialed DN. It is possible the dialed telephone is busy, and Network Ring Again may be used if enabled. The tone indicating the trunks are busy is a fast busy tone. A busy DN is indicated by a standard busy tone.

RVQ searches throughout Meridian 1 networks only. Off-net trunks paths cannot be checked beyond the first trunk.

RVQ is a virtual queuing feature. Each RVQ call is independent of another. It does not operate by first in first out policy. The first caller to initiate RVQ is not necessarily the first person connected.

It may take up to 30 seconds for notification to reach the originator. The network path is reserved while RVQ notifies the caller. In a private network, the path is reserved from the originating node to the terminating node. Otherwise the path is reserved up to the first NRC trunk.

RVQ supports Uniform Dialing Plan (UDP), and Coordinated Dialing Plan (CDP). For NRC trunks, the E.164 public numbering plan is supported.

The following lists the trunk types RVQ supports.

- Private network ISDN trunks
 - TIE
 - COT
 - Direct Outgoing Dial (DOD)
 - WATS
 - ISA
 - FEX

- NRC trunks
 - COT
 - Direct Outgoing Dial (DOD)
 - WATS
 - FEX
 - ISA
 - TIE

RVQ supports Meridian 1 machines only, in a private ISDN PRI/ISL network. If tandeming to a non-Meridian 1 machine, it is treated as a Non-RVQ Compatible (NRC) trunk.

RVQ is canceled if the originating node system initializes.

If the originating node performs a cold start, RVQ is canceled.

RVQ from a Conventional Main (RVQCM) requires a special configuration.

With RVQ, callers cannot activate Ring Again to refuse expensive routes after the Expensive Route Warning Tone (ERWT) is given.

When using RVQ from a Conventional Main (RVQCM), the originating node seizes the same TIE trunk group that was used to initiate RVQCM for the callback. Thus, these trunk groups must be two-way (incoming/outgoing) and configured for far end disconnect.

Conventional mains must provide answer supervision on TIE trunks connected to the originating node. The system must also permit transmission or repetition of telephone dial pulses for RVQCM operation. This feature cannot be used with systems that operate in senderized mode. Operation may require adjustment of the interdigit timeout on systems that employ simulated cut-through operation.

Multiple callback queues are allowed per trunk group for the Conventional main by dialing any digits (up to 7) based on the availability of Meridian 1 call registers.

When utilizing RVQCM, do not call forward the calling telephone when awaiting callback. If the telephone is forwarded, it is possible that the TIE trunk will not be Released at the end of the call.

Feature interaction

Drop Back Busy

Drop Back Busy (DBB) and Remote Virtual Queuing (RVQ) are both packaged under the Originating Routing Control/Remote Virtual Queuing (ORC-RVQ) package 192. If RVQ and DBB are both configured, DBB will take precedence over RVQ. If the user wishes to activate RVQ, DBB must be disabled for a route entry in response to the IDBB prompt. If the user wishes to activate DBB, it must be enabled by entering DBI or DBA. Refer to the “Implementation” section for more information.

Coordinated Call-Back Queuing

CCBQ and RVQ can both be enabled on a single machine.

Call-Back Queueing to a Conventional Main

CBQCM and RVQ can both be enabled on a single machine.

Direct Inward System Access

DISA DN's are not supported by RVQ.

Make Set Busy

RVQ can be originated by, and terminated on an ACD DN in Make Set Busy mode.

Network ACD

If a target agent is available on a remote node, RVQ cannot be activated if the busy trunk is found on its way to the target node.

Network Call Redirection

RVQ does not guarantee connection if the terminating telephone call forwards to a telephone on another node. When a terminating telephone call forwards to another node, that node is a redirected node. RVQ cannot search between terminating and redirected nodes.

Network Ring Again

Network Ring Again (NRAG) is activated only when the terminating telephone or console is busy. If both the network path and the terminating telephone are busy, RVQ is activated first. When the path is available, but the telephone is busy, then NRAG can be activated.

Off Hook Queueing

RVQ and Off Hook Queueing (OHQ) are compatible in a system. If OHQ is configured, it is implemented by leaving the handset off hook. RVQ is implemented by pressing the Ring Again key. Choosing one or the other method activates the specified feature.

Feature operation

RVQ supports private ISDN PRI/ISL networks, non-RVQ compatible (NRC) blocked trunks, and conventional mains (RVQCM).

RVQ on a private network. RVQ is supported for Meridian 1 machines, running X11 Release 18 on an ISDN PRI/ISL network only. When a blocked network path is discovered, RVQ searches through the network to ensure the entire path will be clear when the blocked section is cleared. When all trunks along the path are available, RVQ notifies you.

When a trunk is busy, RVQ follows this checklist before invoking RVQ.

- The first tandem node must be a Meridian 1 node, running X11 Release 18 or later software.
- The originating node's Network Class of Service (NCOS) must allow Call Back Queueing (CBQ).
- At least one route in the initial set of the route list must allow CBQ.
- The system has CBQ allowed at the Network Control Block.

When the checks are complete, activate RVQ by pressing the Ring Again key. As soon as the key is pressed, the retry timer begins. When the retry timer expires, RVQ checks the path again. The total amount of time RVQ will spend on a search is 30 minutes.

The retry counter controls the number of times the initial set is checked before RVQ searches the extended set. The extended set search includes all the trunks in both the initial and extended sets.

Tandem nodes must recognize the RVQ messages and pass them on to another tandem or terminating node.

RVQ with NRC trunks. RVQ searches the path until it reaches the first Non-RVQ Compatible (NRC) trunk. RVQ notifies the originator when that trunk is available, but cannot search beyond it. It is possible that the path may be blocked beyond the NRC trunk, and the call cannot go through.

Any trunk other than an ISDN PRI or ISL trunk is considered an NRC trunk. NRC trunks include analog, T1, or public trunks, as well as trunks connecting to a non-Meridian 1, non-ISDN, or non-X11 Release 18 node.

When an NRC trunk is busy, RVQ follows this checklist before implementing RVQ.

- First tandem node must be a Meridian 1 node, with X11 Release 18 or later.
- Originating node's NCOS must allow Call Back Queuing (CBQ).
- At least one route in the initial set of the route list must allow CBQ.
- The system has CBQ allowed at the Network Control Block.

When the checks are complete, activate RVQ by pressing the Ring Again key. As soon as the key is pressed, the retry timer begins. When the retry timer expires, RVQ checks the path again. The total amount of time RVQ will spend on a search is 30 minutes.

The retry counter controls the number of times the initial set is checked before RVQ searches the extended set. The extended set search includes all the trunks in both the initial and extended sets. Only the first NRC trunk is checked. RVQ cannot search beyond that trunk, even if the subsequent trunks are ISDN PRI/ISL.

RVQ on a conventional main (RVQCM). A conventional main call is one that comes from a non-Meridian 1 system into the private network through a TIE trunk to a Meridian 1 node. When the conventional main call enters the network, the default NCOS is assigned to the TIE trunk to determine RVQCM eligibility. Use Authcode last to upgrade the NCOS if necessary.

When a conventional main call is blocked, RVQ checks the following at the first Meridian 1 node to ensure RVQCM can be activated. The Meridian 1 node is considered the Originating node, and controls the call.

- The tandem and terminating nodes must be Meridian 1 nodes, running X11 Release 18 or later software.
- The originating node's Network Class of Service (NCOS) must allow Call Back Queuing (CBQ).
- At least one route in the initial set of the route list must allow CBQ.
- The system has CBQ allowed at the Network Control Block.
- The incoming TIE trunk allows CBQ.

RVQCM is offered to the conventional main caller with a special offer tone (3 beeps). To accept RVQ, enter the calling number. After the same special confirmation tone (3 beeps) is heard, hang up. The originating node begins the retry timer and search process. When the retry timer expires, RVQ checks the path again. The maximum total amount of time RVQ will spend on a search is 30 minutes. When the path is free, the originating node calls the caller back (at the number entered). The phone rings, the same special callback tone (3 beeps) is heard, and the call is connected. RVQCM cannot be canceled from the conventional main once started.

If the calling telephone is busy, or does not answer the callback, the RVQ callback is placed in a 5 minute suspense state. After 5 minutes have passed, RVQ attempts the callback again. If the calling telephone is still busy, or does not answer, RVQ is canceled.

The retry counter controls the number of times the initial set is checked before RVQ searches the extended set. The extended set search includes all the trunks in both the initial and extended sets.

Note: The calling number is accepted only if the database has been configured to accept the given number of digits. If not, or if the caller does not hang up, overflow tone is given, and the call is disconnected. RVQCM does not verify that the dialed DN is valid or free.

Once started, RVQ cannot be canceled from the convention main. Operating parameters relating to Call Back Queuing for Conventional Mains (CBQCM) also apply to RVQCM as follows. Refer to the *ESN description* (309-3001-100) for complete discussions of CBQCM.

Users at Conventional Mains cannot activate Ring Again to refuse expensive routes after the Expensive Route Warning Tone (ERWT) is given.

The node seizes the same tie trunk group that was used to initiate RVQCM for the callback. These trunk groups must be two way (incoming/outgoing).

Conventional Mains must provide answer supervision on tie trunks connected to the node. These switches must also permit transmission, or repetition of telephone dial pulses for RVQCM operation. This feature cannot be used with systems that operate in senderized mode. Operation may require adjustment of the interdigit timeout on systems that employ simulated cut-through operation.

Multiple callback queues are allowed per trunk group for the Conventional Main by dialing up to 7 digits (any digits are allowed) based on the availability of Meridian 1 call registers.

CAUTION

Conventional Mains must not allow RVQCM callback calls to be modified by call transfer or call forward to an outside line. Call modifications like this can result in the tie trunk not being Released at the end of the call.

Operating RVQ from a digital telephone. The following lists the steps required to activate RVQ from any Meridian digital, Meridian Modular, or SL-1 telephone.

- 1 Place a call to a DN on another node.
- 2 Hear overflow tone indicating a busy trunk along the call path.
- 3 Press the Ring Again (RGA) key. The RGA lamp lights, and the tone stops. If the lamp remains dark, RVQ may not apply.
- 4 Go on-hook.

- 5 The RGA lamp fast flashes, and a 1 second buzz indicates the path is available. If you are on another call, you hear the Ring Again tone.
- 6 Go off-hook (or press a DN key), and press the RGA key to connect the call. If the RVQ notice is not answered within 16 seconds, the RVQ notice stops, and the request is canceled.

If the terminating telephone is busy, you hear regular busy tone. You may activate Network Ring Again (NRAG) at this point.

Cancel the RVQ request at any time by pressing the RGA key. The associated lamp goes dark.

Operating RVQ from an analog (500/2500 type) telephone. The following lists the steps required to activate RVQ from any analog (500/2500 type) telephone.

- 1 Place call to a DN on another node.
- 2 Hear overflow tone indicating a busy trunk along the call path.
- 3 Switchhook flash, and hear special tone (3 beeps).
- 4 Dial the Ring Again SPRE or FFC, and hear confirmation tone.
If RVQ is not enabled, intercept treatment is given.
- 5 Go on-hook.
The RVQ search takes place only when the telephone is idle. RVQ searches only while you are off-hook. However, the timer runs while you are on-hook, or off-hook.
- 6 Special ringing cadence indicates the path is available.
- 7 Go off-hook to connect the call.
If the RVQ notice is not answered within 16 seconds, the RVQ notice stops, and the request is canceled.

Cancel the RVQ request at any time by going off hook and entering the Ring Again deactivation SPRE or FFC.

Operating RVQCM from conventional main telephone. The following lists the steps required to activate RVQCM from a conventional main telephone.

- 1 Place call to a DN on a Meridian 1 node.
- 2 Hear special dial tone (3 beeps), or RAN if configured, indicating a busy trunk along the call path.
- 3 Dial the calling DN, and hear the same special tone (3 beeps).
- 4 Go on-hook.
- 5 The phone rings when the originating node calls back.
- 6 Go off-hook, hear the same special tone (3 beeps). Stay off-hook, and the call is connected. If the RVQ notice is not answered within the time set in LD 87 (CBTL prompt), the RVQ notice stops, and the request is canceled.

RVQ cannot be canceled from a conventional main telephone, but may timeout after 20 seconds

If the calling telephone is busy, or does not answer the callback, the RVQ callback is placed in a 5 minute suspense state. After 5 minutes have passed, RVQ attempts the callback again. If the calling telephone is still busy, or does not answer, RVQ is canceled.

Package Requirements

- Originating Routing Control/Remote Virtual Queuing (ORC-RVQ) package 192, requiring:
 - Network Signaling (NSIG) package 37
 - Network Call Back Queuing (MCBQ) package 38
- ISDN Supplementary (ISDNS) package 161
- Main Network Queuing package 38
- Flexible Call Back Queuing package 61
- Digital Trunk Interface package 75
- ISDN signalling package 145

- ISDN PRI package 146 or ISL package 147
- Advanced Network Services package 148 (for NRAG capability)

Ring Again on No Answer

Description

This feature allows a caller to an idle but unanswered station to activate a form of Ring Again. When the called station goes off-hook and then on-hook, the activating station is rung back in the same way that traditional “Ring Again on Busy” operates.

This feature can operate across an ISDN using Network Ring Again.

Slow Answer Recall Enhancement

This enhancement to the Slow Answer Recall feature changes how the recall is treated once presented to the attendant console. If an incoming call extended by the attendant to a set is not answered after a preprogrammed time period, it is recalled to the attendant console. The call type may be indicated by an Incoming Call Indicator (ICI) key programmed to flash for recalls. The target set will continue to ring after the call is presented to the attendant. The target set can answer the call before the attendant does, in which case the call is cleared from the attendant console and the incoming call and target set will be connected.

If the attendant answers the recall before the target set, a speech connection is established between the calling party on the source (SRC) side of the console. The target set continues to ring while still being connected to the destination (DEST) side of the console. This feature only affects the operation after the attendant has answered the recall.

When a Slow Answer Recall occurs the call is placed in the attendant queue and appears on the console. The target set will continue to ring while the recall is queued and presented on the console but unanswered. When the attendant answers the recall, by pressing the appropriate Loop key or the Recall ICI key, the target set will be disconnected as soon as the attendant console answers the Slow Answer Recall.

In a ISDN environment, the feature works in a similar way regardless of the location of the called party (on the same node as the attendant or on a remote

node), and whether Network Attendant Service (NAS) routing is involved in the call.

Call Waiting Recalls and Camp-on Recalls

This enhancement adds Call Waiting Recall and Camp-on Recall functionality to Slow Answer Recall, as explained in the following sections.

Call Waiting Recall

If an incoming call extended by the attendant or a set (equipped with the Multiple Party Operations feature) to a busy station (equipped with Call Waiting) is not answered within a customer-defined period of time, it is recalled to the attendant. The recall is presented to the attendant or placed in the attendant queue. Until the attendant answers the call, it remains waiting on the target set, and can still be answered. If the attendant answers the recall by pressing the appropriate Loop key or the Recall key, the target set is disconnected and can no longer answer the call – the target set will have to be redialed to extend the call.

Camp-on Recall

An incoming call is extended by the attendant or a set (equipped with the Station Camp-on feature) to a busy station that is not equipped with Call Waiting. The attendant or set camps on the call to the target set. If the call is not answered within a customer-defined period of time, it is recalled to the attendant. The call is presented to the attendant or placed in the attendant queue. Until the attendant answers the call, the call remains camped-on to the target set, and can still be answered. If the attendant answers the recall by pressing the appropriate Loop key or the Recall key, the target set is disconnected and can no longer answer the call; the target set will have to be redialed to extend the call.

Within a network environment, the Call Waiting Recall and Camp-on Waiting Recall enhancements have to be configured at a node. Both the Call Waiting Recall and Camp-on Waiting Recall enhancements operate in the same way as in the standalone case. The location of the calling and called party and the attendant have no affect on the call processing.

Network Attendant Service (NAS) is not required, but it may be applied at a node. In this case, NAS takes precedence over the Call Waiting Recall and Camp-on Waiting Recall enhancements, in that the target set is disconnected

from the call due to time-out and not to the attendant pressing the Loop key or Recall key.

Operating parameters

The same as for Slow Answer Recall.

Feature interactions

The Call Waiting Recall and Camp-on Waiting Recall enhancements take precedence over Attendant Recall Splitting (ATS), Secrecy (SYA), Enhanced Secrecy (EHS), and Multiple Party Operations.

The Call Waiting Recall and Camp-on Waiting Recall enhancements are compatible with Station Camp-on (STCA).

A forced Camp-on override recall occurs to the attendant. If the Call Waiting Recall and Camp-on Waiting Recall enhancements are equipped, the destination is automatically disconnected when the attendant answers. If the Call Waiting Recall and Camp-on Waiting Recall enhancements are not equipped, and the attendant answers the recall at the same time that the destination answers, a conference is established between the attendant, source, and destination.

Feature packaging

International Supplementary Features (SUPP) package 131.

Station Activity Records

When a set is configured with Class of Service Call Detail Monitoring Allowed (CDMA) for all incoming and outgoing calls, Station Activity Records are produced. The format of Station Activity Records is identical to other Call Detail Recording (CDR) records, but they have a new type identifier (D). Existing CDR records are not affected by this new functionality.

Operating parameters

None.

Feature interactions

Call Redirection

A Station Activity Record is only produced for a set designated as CDMA that is involved in a call with a trunk. A Station Activity Record is not generated for any set which does not answer the call, regardless of whether it has Class of Service CDMA or CDMD. Any other CDR records generated during call redirection are not affected.

Call Transfer

A Station Activity Record is generated when a set with Class of Service CDMA transfers a trunk call to another set. CDR “X” record generation is not affected by this development. The set to which the call is transferred also produces a Station Activity Record if it has Class of Service CDMA and answers the call. When the second “D” record is produced (by the set to which the call is transferred), the digits field of the “D” record shows the digits dialed by the transferring set.

Conference Calls

For a set with Class of Service CDMA involved in a call with a trunk, a Station Activity Record is produced only when that set conferences in the first party. Conferencing of all subsequent parties does not generate a “D” record. An additional “D” record is produced when the last conferee with Class of Service CDMA connected to the trunk goes on hook. This does not affect any other CDR record generation during a conference.

Internal CDR

Internal CDR records are produced according to the Class of Service ICDA/ICDD of a set. The Station Activity Record enhancement does not affect the ICDR record generation.

Feature packaging

Station Activity Records are packaged under the Station Activity Records (SCDR) package 251.

The following two software packages are required for Station Activity Records: Call Detail Recording (CDR) package 4; and Call Detail Recording – TTY (CTY) package 5.

T309 Timer**Description**

The T309 timer provides a 90-second wait interval after a D-channel failure before the B-channels are cleared. When the D-channel is re-established before T309 expires, a STATUS ENquiry message is sent by the network side. The user side responds with a STATUS message to report the status of the queried channel. If the call state matches, the call is preserved; otherwise, a Release message is sent to the user side, clearing the call. If the user side does not respond with the STATUS message before the T322 timer expires, the call is cleared.

If T309 expires before the D-channel is re-established, indicating that there is a serious problem with the link, Meridian 1 clears all calls internally and sends a global RESTART when the D-channel is re-established.

For a Meridian 1 to Meridian 1 interface, the T309 timer is not used. The Meridian 1 keeps active calls established as long as possible and lets B-channel users decide when a call is to be dropped. The Meridian 1 specified as network side sends the STATUS ENquiry message on all active calls when the D-channel is re-established.

Prior to X11 Release 18, the Meridian 1 does not support the T309 and sending the STATUS ENquiry message for North American interfaces. All active calls are cleared with a global RESTART whenever a D-channel is re-established.

The compatibility checking by means of the STATUS ENquiry message might leave some channels with mismatched states, which are cleared only when trying to reuse the channel.

T309 is dependent on both network and user sides. If a side does not have a Timer T309, it sends a global RESTART when D-channel is re-established.

AT&T 4ESS and 5ESS do not clear transient calls when the D-channel fails. If T309 expires, the calls are not cleared until the D-channel is re-established or the call is cleared by the far end.

Integrated Services Access—Integrated Services Access (ISA) tracks the number of active calls. Prior to X11 Release 18, the counter for active calls is reset when the D-channel comes back. With the implementation of T309, active calls are preserved if the ninety seconds does not expire; the counter for active calls is not reset since active calls need to be tracked during the D-channel failure and reset.

Feature packaging

The T309 and STATUS ENquiry message are included in basic ISDN PRI/ISL functionality. No additional packages are required.

Taiwan R1 Modified Signaling

The Taiwan R1 Modified Signaling feature supports Taiwan R1 Modified Signaling (TWR1 MS) over Taiwan R1 Direct Inward Dial (DID) or Direct Outward Dial (DOD) DTI trunks. With TWR1 MS, the capability to carry Calling Line Identification (CLID) across a Meridian 1 network is supported for local and national calls. Please note that, in a networking environment, the CLID is supported only for TWR1 calls that are tandemed over MCDN, ISDN, or QSIG trunks. A discussion on how CLID is processed for TWR1 MS is found in the section “CLID processing” on page 600.

The Taiwan R1 Modified Signaling feature may operate in a standalone or network environment. In a standalone environment, the Meridian 1 connects directly to a Taiwan Public Exchange over Taiwan R1 Modified Signaling trunks.

In a networking environment, the tandem Meridian 1 connects to a Taiwan Public Exchange over Taiwan R1 or Taiwan ISDN trunks, and to one or more other Meridian 1 Private Exchange(s) over MCDN, QSIG, Dual Tone Multi Frequency (DTMF), or Dial Pulse (DP) trunks. Connectivity to the Private Exchange over Taiwan R1 Modified Signaling, Digital Private Networking Signaling System No. 1 (DPNSS1), or Level 2 Multi Frequency Compelled Signaling (R2 MFC) trunks is not supported.

Types of supported TWR1 protocols

The Taiwan R1 Modified Signaling feature supports the following call signaling protocols, for national and international calls:

- incoming local or national call protocol

Note: Incoming international calls are accepted and treated by the Meridian 1 using the national call protocol.

- outgoing local or national call protocol
- outgoing international call protocol

The section “How the Meridian 1 handles the supported types of TWR1 protocols” on page 603 describes the call processing that takes place by the Meridian 1 to handle the supported TWR1 protocols. But first, we discuss how the system processes CLID information for the Taiwan R1 Modified Signaling feature, for incoming and outgoing calls.

CLID processing

Incoming CLID

An incoming TWR1 route on the Meridian 1 may be configured with or without the CLID option. If a Central Office expects to receive the 210 - 220ms CLID request wink signal from the Answer signal, then the TW_INC_CLID prompt in LD16 may be set to YES for the incoming route. If the prompt is set to YES, once the called number is received from the Central Office, the Meridian 1 will send out a CLID request wink to request CLID digits. The called number received is not processed until after the CLID information has been received. When the TW_INC_CLID prompt is set to YES, if CLID digits are not received within a specified time, the call is completed with CLID failure treatment, as follows:

- A series of seven ‘-’ (dashes) is shown on the displays of sets that have displays.
- The CLID field in the CDR ticket for the call is left blank.
- No CLID is made available to the Application Module Link (AML), Application Processor Link (APL), and the High Speed Link (HSL).

For a Central Office that does not expect to receive the CLID request wink signal, the TW_INC_CLID prompt must be set to NO for the incoming route. CLID failure treatment is given to such calls because no CLID is received. In this case, the called number received is processed as soon as it is received.

The CLID information received is made available to existing features that use CLID such as an answering digital terminal's display and such applications as Meridian Link, Customer Controlled Routing, Meridian Mail, Meridian MAX, and ACD MAX using the Application Module Link (AML), Application Processor Link (APL), and High Speed Link (HSL).

For a Taiwan R1 initiated tandem call to an outgoing MCDN, QSIG, or Taiwan ISDN trunk, the CLID that is received with the incoming call may be sent out over the MCDN, QSIG, or Taiwan ISDN trunk to identify the caller. If the outgoing trunk is either a DTMF or DP trunk, no CLID is sent over the outgoing trunk.

Outgoing CLID

The following section discusses how the Meridian 1 handles outgoing CLID information, for a station (set or attendant) and a trunk.

Station (set or attendant console) initiated call

On a standalone PBX, whenever a call is made from a set or an attendant console, the CLID is generated to identify the caller. If the response to the CLID prompt for a customer in LD 15 is YES, the existing customer CLID database is used. This function allows the system administrator to control the CLID contents using the CLID entry configuration in LD 15. If the response to the CLID prompt is NO, there is no CLID composed for calls originated by sets belonging to that customer.

If the response to the existing CLID prompt for a customer in LD 15 is YES, a CLID table can then be defined for the customer. The size of the CLID table is service changeable, ranging from 0 to a maximum of 4000 entries, with a default of 256 entries. Each entry in the table is a CLID entry. Each CLID entry contains the codes for:

- Home National Code (HNTN). This is the national number, which is the area code for Taiwan.
- Home Local Code (HLCL). This is the home local code, which is the office code for Taiwan.
- Direct Inward Dialing DN (DIDN). If DIDN is set to YES, then HLCL is suffixed with the DN of the active key. If DIDN is set to NO, then HLCL is not suffixed with anything. If DIDN is set to SRCH (search), then starting from key 0, all keys are searched until the key for which DIDN is set to YES is found. Then the DN of that key is used.

A CLID entry can be assigned to a DN of an analog (500/2500) set or an ISDN BRI set, and to a DN key for a Meridian 1 proprietary set. For a station set initiated Taiwan call, the CLID is composed based on the CLID entry that the originating DN uses. If the user does not select a CLID entry number, CLID entry 0 is used as the default. During set administration, there is no cross-checking of the CLID entry associated with the set against the CLID entry in the customer data block. If a CLID entry is removed from the CLID table and the CLID entry is still assigned to a DN key, CLID is not generated and CLID failure indication is sent out.

For an attendant console initiated Taiwan call, the CLID composition is based on CLID entry 0. If CLID entry 0 is not defined, CLID is not generated and CLID failure indication is sent out.

Once the CLID is composed, it is sent to the far-end. If the originating terminal has requested privacy as per the Calling Party Privacy (CPP) or Display Party Denied (DPD) features, then CLID is sent out with the CLIR indicator. Otherwise the CLIP indicator is sent.

Trunk initiated tandem call

For an MCDN, ISDN, or QSIG trunk initiated tandem call, the Meridian 1 does not compose any CLID for the incoming trunk. If CLID is available, and presentation is allowed, the CLID that is passed in by the incoming trunk may be directly passed on to the TWR1 trunk to identify the caller. The CLIP indicator is sent out in this case. If CLID is available, but the caller has requested privacy, CLID is sent out with the CLIR indicator.

If, no CLID is available, as is the case for a DTMF or DP trunk initiated tandem call, the CLID failure indicator is sent to the outgoing trunk.

The possibility of an invalid CLID composition

The Taiwan R1 Modified Signaling feature builds the CLID from the CLID entry in LD15. It is up to the craftsperson to ensure that the right digits are entered. There is no validation of digits in the CLID entries of the CLID table. Therefore, it is possible that the CLID built from the CLID entries may be an invalid number or not match the number of the dialing station.

TWR1 Calling Party Privacy CLID

The following sections discuss how the Meridian 1 handles incoming CLID information with CLIR indicator, and how and when the Meridian 1 sends out CLID information with CLIR indicator.

Incoming CLID digits with CLIR indicator

If an incoming Taiwan R1 call with a CLIR indicator is received by the Meridian 1 in a standalone scenario, seven '-' (dashes) will be displayed on the answering set or ACD agent set, and CLID digits are not sent to such applications as Meridian Mail, Meridian Link, Meridian CCR, and Meridian MAX. However, the CLID digits are sent to CDR.

If the Meridian 1 is a network PBX, the call with the CLIR indicator will be tandemmed to an outgoing MCDN, QSIG, or Taiwan ISDN trunk with the CLID Restriction indicator. If the call is tandemmed to an outgoing DTMF or DP trunk, no CLID is sent.

Outgoing CLID digits with CLIR indicator

For outgoing calls from a standalone Meridian 1 PBX, if the caller requests privacy as per the existing Calling Party Privacy (CPP) and Display Party Denied (DPD) features, CLID is sent out with the CLIR indicator.

If the Meridian 1 is a network PBX, if the incoming MCDN, QSIG, or Taiwan ISDN call contains the “CLID Presentation Restricted” indicator, the call is tandemmed to Taiwan R1 trunks with the CLIR indicator. In case of tandem calls from a DTMF or DP trunk, the CLID failure indicator is sent.

How the Meridian 1 handles the supported types of TWR1 protocols**Incoming local and national call processing**

The Central Office seizes a TWR1 trunk, and sends a called number to the Meridian 1. If the TW_INC_CLID is set to NO, then the call terminates as soon as the called number is received. The call is given the following CLID failure treatment:

- A series of seven ‘-’ (dashes) is shown on the displays of sets that have displays.
- The CLID field in the CDR ticket for the call is left blank.
- No CLID is made available to the Application Module Link (AML), Application Processor Link (APL), and the High Speed Link (HSL).

If the TW_INC_CLID is set to YES, on receiving the called number, the Meridian 1 sends a request wink signal to the Central Office, to request the calling number and CLID information. The CO sends the calling number in MF tones. The calling number includes the Information digit “I”, where “I” may be one of the following digit values:

- 0 or 1, which signifies that CLID is available with Calling Line Identification Presentation (CLIP) allowed. This is the normal CLID delivery.
- 2, which signifies that CLID is not available. The call receives the CLID call failure treatment described above.

- 3, which signifies that CLID is available with Calling Line Identification Restriction (CLIR). The call receives the following treatment:
 - A series of seven ‘-’ (dashes) is shown on the displays of sets that have displays.
 - The CLID field in the CDR ticket for the call is appropriately filled.
 - No CLID is made available to the Application Module Link (AML), Application Processor Link (APL), and the High Speed Link (HSL).

Outgoing local and national call processing

For outgoing calls, Electronic Switched Network (ESN) and Trunk Access Code (ACOD) dialing are supported. To handle the inconsistent dialing patterns for outgoing calls, the Flexible Numbering (FNP) package 160 has to be equipped on the Meridian 1 PBX. For ESN dialing, the call is terminated as soon as the Flexible Length (FLEN) value is reached for the dialed Special Number (SPN). For ACOD dialing, the call is terminated after a time of five seconds elapses from when a last digit, or ‘#’ is immediately dialed.

When a called number is composed using ESN dialing, the Meridian 1 collects the dialed digits, as determined by the Flexible Numbering Plan configuration, and seizes an outgoing TWR1 trunk to the far-end Central Office. The Meridian 1 sends the called number digit string to the CO, and awaits for the calling number and CLID information request wink signal. If this signal is not received within a specified period of time, as defined by the Taiwan Time Out timer¹ (TTO), the calling number is not sent.

1. The TTO is a programmable waiting timer for the CLID request wink signal on outgoing trunks. It can be defined in LD 16 to be in a range of 128ms to 7seconds, with a default value of 128ms. The TTO has been implemented for the following reasons. For users who don't expect CLID request wink from CO, the called party would hear the ring back tone from the terminating end only after seven seconds, which is too long. For users who do expect CLID request wink from the CO, the programmable timer provides a flexibility to configure the proper timers according to the different situations of different COs.

The default value of TTO is 128ms, which is defined for the users who don't expect CLID request wink from the CO. However, for the users who do expect CLID request wink from the CO, they should configure the TTO timer to a suitable value. The value may differ depending on different COs, because different CO may have different delay when they send the wink signal. Users may change the value through testing.

On receiving the CLID request wink signal from the Central Office, the Meridian 1 sends the calling number in MF tones. The calling number includes the Information digit “I”, where “I” may be one of the following digit values:

- 0, which signifies that CLID is available with Calling Line Identification Presentation (CLIP) allowed. A value of ‘0’ is sent to the far-end CO if the originating set has not requested privacy as per the Calling Party Privacy (CPP) or Display Party Denied (DPD) features.
- 2, which signifies that CLID is not available.

Note: For outgoing calls, as long as a CLID has been configured for the originating set, CLID information will always be sent. It is up to the far-end CO to handle the CLID accordingly.

- 3, which signifies that CLID is available with Calling Line Identification Restriction (CLIR). The CLIR indicator is sent to the far-end CO to request calling party privacy, if one of the following conditions exists:
 - For a call originating from a set or attendant, the originator requests calling party privacy as defined by the Calling Party Privacy (CPP), Display Party Denied (DPD), Calling Party Privacy Override (CPPO), or BRI set privacy feature.
 - For tandem calls over MCDN, ISDN, or QSIG trunks, the tandem call contains a Presentation Restriction indicator.

Outgoing international call processing

Taiwan R1 Modified Signaling uses two stage outpulsing of the called party number for outgoing international calls. The first stage includes sending the international call access code (002) to the CO, and the second stage includes sending the country code and the called number to the CO. The second stage of outpulsing is started by the Meridian 1 when the “second stage outpulsing request” wink is received from the CO.

Note: There is no CLID sent for outgoing international calls.

Both ESN dialing and ACOD dialing are supported for making TWR1 outgoing international calls. If using ESN dialing, it is the responsibility of the craftsperson to configure a Special Number (SPN) of “002”, which is the Taiwan international call access code. The SPN 002 is configured using LD 90. Also in LD 90, an appropriate Flexible Length (FLEN) of at least 16 digits should be defined, and the Inhibit Time Out Handler (ITOH) should be set to a value of “NO”.

Taiwan numbering plans

This section describes the called party and calling party numbering plan format used by the Taiwan Public Switched Telephone Network (PSTN) system. The numbering plan is variable, having different digit string lengths for large cities and small cities.

For outgoing calls, Electronic Switched Network (ESN) and Trunk Access Code (ACOD) dialing are supported. To handle the inconsistent dialing patterns for outgoing calls, the Flexible Numbering (FNP) package 160 has to be equipped on the Meridian 1 PBX, if ESN dialing is used. With FNP equipped, the call is terminated as soon as soon as the Flexible Length (FLEN) value is reached for the dialed Special Number (SPN). For ACOD dialing, the call is terminated after a time of five seconds elapses from when a last digit is dialed, or if ‘#’ is dialed immediately.

Note: Large cities in Taiwan will soon be upgrading to the new dialing plan, indicated by “Large cities (future).”

Table 55 represent the dialing plans used by the Taiwan PSTN. The sections that follow then show the numbering plans, according to incoming and outgoing call formats.

Table 55
Taiwan PSTN numbering plan formats

	Area Code	Office Code	Subscriber Number
Large cities (future)	A	+BCDD'	+XXXX
Large cities (current)	A	+BCD	+XXXX
Small cities	AB	+CD	+XXXX
The letters represent the variable digits allowed in each position.			
— The first digit of the area code can be 2-8.			
— The first digit of the office code can be 2-9.			
— All other letters imply that the digits can be 0-9.			
— The service code numbers can be three digits, such as 10X, 11X, and 16X.			

Incoming call formats - Local/National and International¹ calls

This section shows the numbering plan formats for the supported incoming call protocols, namely Local/National calls. Please note that the Meridian 1 treats incoming International calls using the same protocol as for incoming Local/National calls. For this reason, there is reference to International calls in the tables that follow.

Incoming National and International Called Party Number format

Table 56 shows the format for a called party number of a Local/National or International incoming call.

Table 56
Called party number format for Local/National and International incoming calls

Call type	Called Area	Called Party Number format expected by the Meridian 1
Local/National, International*	Large city (future)	A+BCDD'+XXXX or BCDD'+XXXX
	Large city (current)	A+BCD+XXXX or BCD+XXXX
	Small city	AB+CD+XXXX or CD+XXXX
* For international calls, the Meridian 1 uses the same protocol as for national calls.		

Note: There is no CLID sent for International incoming calls.

1. For incoming International calls, the Meridian 1 treats them using the same incoming call protocol as for incoming National calls.

Incoming Local/National Calling Party Number format

Table 57 shows the Calling Party composition for a Local/National incoming call.

Note: For International incoming calls, no CLID is available from the far end, as per the Taiwan R1 Modified Signaling protocol.

Table 57
Calling party composition for Local/National incoming calls

Call type	Calling Area	Calling Party Number format expected by the Meridian 1
Local/National	Large city (future)	A+BCDD'+XXXX
	Large city (current)	A+BCD+XXXX
	Small city	AB+CD+XXXX
	Global System for Mobile communication (GSM)	93X+XXXXXX
	Advanced Mobile Phone Service (AMPS)	90+XXXX

Outgoing call formats - Local/National, and International calls

This section shows the numbering plan formats for the supported outgoing call protocols, namely Local/National, and International calls.

Outgoing Local/National, International Called Party Number format

Table 58 shows the format for a called party number of a Local/National and International outgoing call.

Table 58
Called party format for Local/National and International outgoing calls

Call type	Called Area	Called Party Number format sent from the Meridian 1
Service call	All	1XX
Local	Large city (future)	BCDD'+XXXX
	Large city (current)	BCD+XXXX
	Small city	CD+XXXX
National	Large city (future)	A+BCDD'+XXXX
	Large city (current)	A+BCD+XXXX
	Small city	AB+CD+XXXX
	Global System for Mobile communication (GSM)	93X+XXXXXX
	Advanced Mobile Phone Service (AMPS)	90+XXXX
	Pager	AB+XXXXXX or ABB'+XXXXXX
International	All	002+Country Code+Area Code+Subscriber Code

Outgoing Local/National Calling Party Number format

Table 59 shows the Calling Party composition for a Local and National outgoing call.

Note: There is no CLID sent for outgoing International calls.

Table 59
Calling Party composition for Local/National outgoing calls

Call type	Calling Area	Calling Party Number format sent from the Meridian 1
Local or National	Large city (future)	A+BCDD'+XXXX or BCDD'+XXXX
	Large city (current)	A+BCD+XXXX or BCD+XXXX
	Small city	AB+CD+XXXX or CD+XXXX

Operating parameters

The Taiwan R1 trunk is a one-way trunk, being either outgoing only or incoming only (as configured in LD 16 in response to the ICOG prompt).

Taiwan R1 trunks use Multi Frequency (MF) signaling only. Dual Tone Multi Frequency (DTMF) signaling and Dial Pulse (DP) signaling are not supported.

Taiwan R1 trunks support Loop Dial Repeating Signaling (LDR) only.

Only DID/DOD and DTI Taiwan R1 trunks are supported for this feature.

Both ESN and ACOD dialing are supported for making TWR1 outgoing international calls. If using ESN dialing is used for making outgoing international calls, it is the responsibility of the craftsperson to configure a Special Number (SPN) of “002”, which is the Taiwan international call access code, in LD 90.

The Taiwan R1 Modified Signaling feature builds the CLID based on the CLID entry defined in LD 15. Since there is no system validation of the CLID digits entered by the craftsperson for the CLID table, it is possible that the CLID that is built from the CLID entries may be invalid. It is the responsibility of the craftsperson to ensure integrity.

The CLID digit string received is not validated by the system software. For a standalone PBX, the CLID digits are displayed on the answering terminal exactly as they are received, without any manipulation. The same information is made available to other applications via the AML, APL and HSL. For a network PBX, the CLID digits are passed on to the outgoing MCDN, QSIG, Taiwan R1 or Taiwan ISDN trunk.

All timers, other than the Disconnect Supervision Timer (DSI), the Seizure Supervision timer (SST) and the Taiwan Time Out timer (TTO), are hard-coded and not programmable. The DSI and SST are existing timers that are programmable via LD 16. The TTO is a new timer that has been introduced to provide flexibility in handling delays pertaining to outgoing trunk calls. TTO is also programmable using LD 16.

Hardware requirements

For Meridian 1 Option 11C, the 1.5 Mbit NTAK09xx DTI/PRI card is required, operating in DTI mode.

Note: xx = The card vintage is to be determined.

This card sends the timed wink (modified wink start) required for the Taiwan R1 Modified Signaling operation.

Feature interactions

Automatic Call Distribution

For incoming calls over Taiwan R1 trunks to an ACD set, the CLID information of a calling party is displayed on the ACD set. If CLID is not available, or if Calling Line Identification Restriction (CLIR) applies, seven “-” (dashes) are displayed on the ACD set instead.

BRI sets

ISDN BRI sets, and their associated privacy functionality, are supported on the Taiwan R1 Modified Signaling feature. When the Caller Number Presentation Denied capability is configured on an ISDN BRI set that originates a call over a Taiwan R1 trunk, the Calling Line Identification Restriction (CLIR) indicator in the TWR1 protocol is sent out with the CLID.

Call Hold

When an incoming call over a Taiwan R1 trunk is taken off hold on a set, the CLID display remains on that set.

Call Forward (all types), Hunt (including Network Hunt)

Consider the scenario where Calling Line Identification Restriction (CLIR) applies to incoming calls over Taiwan R1 trunks. These calls are forwarded to an MCDN, QSIG, or Taiwan ISDN trunk as a result of treatment by all types of Call Forward, and by Hunt (including Network Hunt).

In this scenario, the Privacy Indicator of the call is sent to the outgoing trunk to inhibit the display of the calling party number.

In the case where an incoming call with CLIR originates from an MCDN, QSIG, or Taiwan ISDN trunk, and is forwarded or hunted to a Taiwan R1 trunk, the CLID is sent with the CLIR indicator.

If an incoming trunk call is forwarded or hunted to a Taiwan R1 trunk, and Calling Party Privacy has been activated for the forwarding DN, the CLIR indicator is sent over the Taiwan R1 trunk to inhibit the display of the number of the forwarding set.

Call Transfer

In the case where an incoming call with CLIR originates from an MCDN, QSIG, or Taiwan ISDN trunk, and is transferred over a Taiwan R1 trunk, the outgoing CLID is sent with the CLIR indicator.

In the case where a call comes in over a Taiwan R1 trunk, and the CLIR indicator is received with the CLID, and the call is transferred, then the Presentation Restricted indicator is set for the outgoing call to inhibit the display of the number of the transferring set.

If an incoming call over a Dual Tone Multi Frequency (DTMF) or Dial Pulse (DP) trunk is transferred to a Taiwan R1 trunk, or if an incoming call over a Taiwan R1 trunk is transferred to a DTMF or DP trunk, no CLID is sent over the outgoing trunk.

Call Pickup/Call Pickup Network Wide

If a call with the Calling Line ID Presentation (CLIP) indicator comes in over a Taiwan R1 trunk, and is picked up locally or remotely, the CLID of the incoming call is displayed on the set performing the call pickup.

If the incoming call would have had the Calling Line Identification Restriction (CLIR) indicator instead of CLIP, then seven “-” (dashes) would have been displayed on the set performing the call pickup.

Calling Party Privacy

If the Calling Party Privacy Flexible Feature Code is dialed, or if Calling Party Number and Name Blocking Allowed (CLBA) Class of Service is activated, for a call going out over a Taiwan R1 trunk, then the CLIR indicator is sent with the CLID over the outgoing trunk. Both of these actions block the CLID of the originating set from being displayed on the terminating set.

Calling Party Privacy Override

For an outgoing call over a Taiwan R1 trunk, if the Calling Party Privacy Override Flexible Feature Code is dialed, the Calling Line ID Presentation (CLIP) indicator in the TWR1 protocol is sent out over the Taiwan R1 trunk.

Custom Local Area Signaling Services sets

If the CLID delivery on a Custom Local Area Signaling Services (CLASS) set is restricted, the information that is displayed depends on the type of vendor CLASS set being used.

Conference

The Conference feature is supported over Taiwan R1 trunks. If an incoming or originating local call is conferenced over an outgoing Taiwan R1 trunk, the CLID is passed over the Taiwan R1 trunk.

If an incoming call over a Dual Tone Multi Frequency (DTMF) or Dial Pulse (DP) trunk is conferenced over an outgoing Taiwan R1 trunk, no CLID is sent over the outgoing Taiwan R1 trunk. The same applies if an incoming Taiwan R1 call is conferenced over an outgoing DTMF or DP trunk.

If the set originating the conference has requested Calling Party Privacy by dialing the Flexible Feature Code, or if Calling Party Number and Name blocking (CLBA) Class of Service is activated for the set, then the CLIR indicator is sent with the CLID over the outgoing Taiwan R1 trunk.

Display Calling Party Denied

If a local set with a Class of Service of Display on other Set Denied (DDGD) originates a call over an outgoing Taiwan R1 trunk, then the Presentation Indicator field for the call is set to Presentation Restricted.

For incoming Taiwan R1 calls, there is no interaction with the Display Calling Party Denied feature.

Emergency Services Access

ESA calls may be made over Taiwan R1 trunks. The rules that apply for the CLID composition for ESA calls are in effect.

Flexible Numbering Plan

If ESN dialing is used, the Flexible Numbering Plan is required to accommodate Taiwan's variable numbering plan.

Incoming Digit Conversion

The incoming digits received from the Central Office via Taiwan R1 trunks can be converted to unrelated extension numbers within the Meridian 1.

Incoming Trunk Programmable CLID

For calls coming in over Taiwan R1 trunks to a network PBX that has Incoming Trunk Programmable CLID configured, then the billing number and not the incoming CLID is passed over the outgoing Taiwan R1 trunk connected to the Central Office.

Network Message Services

An call incoming over a Taiwan R1 trunk to a Message Center attendant will not display the CLID if the Calling Line Identification Restriction (CLIR) is sent with the CLID.

ACD MAX

The Taiwan R1 Modified Signaling feature provides the proper CLID information to the ACD MAX operation, via the High Speed Link (HSL) messages. If the Calling Line Identification Restriction (CLIR) is sent with the CLID, the CLID information is not included in the HSL messages sent to ACD MAX.

Application Module Base (AM Base)

The Taiwan R1 Modified Signaling feature provides the proper CLID information to the Application Module Base (AM Base), via the Application Module Link (AML) messages. If the Calling Line Identification Restriction (CLIR) is sent with the CLID, the CLID information is not included in the AML messages sent to the AM Base.

Customer Controlled Routing

The Taiwan R1 Modified Signaling feature provides the proper CLID information to the Customer Controlled Routing (CCR) operation, via the Application Module Link (AML) messages. If the Calling Line Identification Restriction (CLIR) is sent with the CLID, the CLID information is not included in the AML messages sent to CCR.

Call Detail Recording

The Taiwan R1 Modified Signaling feature provides the proper CLID information to the CDR operation. If the Calling Line Identification Restriction (CLIR) is sent with the CLID, the CLID information is still included in the CDR ticket.

Meridian IVR

The Taiwan R1 Modified Signaling feature provides the proper CLID information to the Meridian IVR operation, via the Application Module Link (AML) messages. If the Calling Line Identification Restriction (CLIR) is sent with the CLID, the CLID information is not included in the AML messages sent to Meridian IVR.

Meridian Link

The Taiwan R1 Modified Signaling feature provides the proper CLID information to the Meridian Link operation, via the Application Module Link (AML) messages. If the Calling Line Identification Restriction (CLIR) is sent with the CLID, the CLID information is not included in the AML messages sent to Meridian Link.

Meridian Mail

The Taiwan R1 Modified Signaling feature provides the proper CLID information to the Meridian Link operation, via the Application Module Link (AML) messages.

If the Calling Line Identification Restriction (CLIR) is sent with the CLID for an incoming call over a Taiwan R1 trunk to a Meridian Mail set, the CLID information is not sent to Meridian Mail to be recorded. When the called party retrieves the message, no CLID information will be played. The Call Sender feature cannot be initiated by the called party, since the CLID was not recorded.

Meridian MAX

The Taiwan R1 Modified Signaling feature provides the proper CLID information to the Meridian MAX operation, via the Application Module Link (AML) messages. If the Calling Line Identification Restriction (CLIR) is sent with the CLID, the CLID information is not included in the AML messages sent to Meridian MAX.

Meridian VISIT

The Taiwan R1 Modified Signaling feature provides the proper CLID information to the Meridian VISIT Messenger, via the Application Module Link (AML) messages. If the Calling Line Identification Restriction (CLIR) is sent with the CLID, the CLID information is not included in the AML messages sent to the Meridian Messenger.

Feature packaging

The following packages are required for the Taiwan R1 Modified Signaling feature:

- Automatic Number Identification (ANI) package 12
- Taiwan R1 (TWR1) package 347
- If ESN dialing is used, the Flexible Numbering Plan (FNP) package 160, to support Taiwan's variable numbering plan

Feature operation

No operating procedures are required to use this feature.

Total Redirection Count

Description

This feature allows a limit to be defined on the number of redirections within a private network allowed to a call, before the call receives Intercept treatment.

Both the limit on the redirection for a call and the type of Intercept treatment that the call receives are customer-defined in LD 15. This applies to on-node and off-node redirections, and to all types of redirections. When the total redirection count exceeds the defined limit, the call receives the customer-defined intercept treatment. This treatment includes receiving busy indication, overflow indication, or recorded announcement, receiving one of eight special tones, or being routed to the attendant. If the call is routed to the attendant, it is presented on the ICI Intercept key and the reason for redirection is given on the console display. The attendant may then use Break-in to connect to the desired station (if the desired station is established on a call).

The maximum value that may be given to the TRCNT limit is seven. The Total Redirection Count (TRCNT) is kept active until the call is established or directed to the attendant. The TRCNT takes precedence over higher count limits placed on redirected calls, while lower count limits take precedence over the TRCNT.

It is possible to define a different TRCNT limit at each node. For this reason, it is possible for a node to receive a redirected call from another node that exceeds its TRCNT limit. In this case, the TRCNT count for the call is set to the TRCNT limit defined for the node. At least one attempt is made to terminate the call before intercept treatment is given.

For off-node operation, the TRCNT count overrides the Redirection Count (RCNT) count in the ISDN field in the SETUP message. This implies that the count transmitted to a node is either interpreted as TRCNT or Call Redirection Threshold (RCNT), depending on the configuration at the receiving node.

For off-node calls, this feature applies only to Meridian 1 systems using Meridian Customer Defined Networking (MCDN) signaling over ISL/ISDN TIE links. Network Attendant Service is required to route a call to an attendant at another node. Intercept to the attendant does not count as a redirection attempt.

The following ISDN call restrictions apply:

- Tandem Threshold, which is the limit placed on the number of tandem nodes allowed in a network connection.
- The Public Service Telephone Network Threshold, which is the limit placed on the number of PSTNs allowed in a network connection.
- The Call Redirection Threshold, which is a limit on the number of times that a call can be redirected off-node. If the Total Redirection Count (TRCNT) Limit is set a value greater than zero, then the ISDN field in the SETUP message transports the TRCNT information rather than the Redirection Count (RCNT) information.
- The M μ /A Law Conversion Threshold, which is a limit on the number of M μ /A Law Conversions allowed in a network connection.
- Satellite Delay Threshold, which is a limit on the number of satellite delays allowed in a network connection.
- Disconnect Supervision Threshold, which limits to one the number of unsupervised trunks allowed in a network connection.

Feature interactions

Hunt, Call Forward Busy, Call Forward All Calls, Call Forward No Answer, and Second-level Call Forward No Answer

Hunt, Call Forward Busy, Call Forward All Calls, Call Forward No Answer, and Second-level Call Forward No Answer redirections are limited to the value defined in the TRCNT limit (if greater than 0). If this limit is exceeded, intercept treatment is given.

Group Hunt

Group Hunt takes precedence over the TRCNT feature, in that the TRCNT limit is not applied to a Group Hunt call.

Intercept treatment

Intercept treatment is not given if a call is a Network ACD call, if a call is a

Central Office trunk in Night Service (specific treatment is given rather than customer-defined intercept treatment), or if the call is a data call (overflow tone is automatically given).

Call Forward No Answer and Transferred

If a call has attempted Call Forward No Answer and was extended by the attendant, the call will not be intercepted when the TRCNT Limit has been exceeded. The call will continue to ring the set until recalled to the attendant.

Package Requirements

This feature requires the following feature packages:

Feature	Package Number
ISDN	145
NAS	159
CPND	95
BKI	127

Trunk Anti-Tromboning

The purpose of the Trunk Anti-Tromboning (TAT) feature is to optimize tromboned ISDN PRI trunks, ISDN Signaling Link (ISL), Virtual Network Services (VNS) trunks, or analog trunks within an ISDN PRI network. Anti-tromboning will be performed only after the third party answers, and provided that the tromboned PRI/ISL/VNS or analog trunks are associated with the same primary D-channel (with or without a backup D-channel) and the trunks are associated with the same customer. Anti-tromboning will not be performed if the tromboning trunks belong to different customers on the same node (even though they are associated with the same primary D-channel.)

***Note:* TAT also applies to a call entering a Meridian 1 network over a Central Office trunk. TAT eliminates the tromboning of private network trunks that may occur if the call is redirected or modified.**

The term tromboning is used to define a situation whereby two PRI/ISL/VNS trunks, associated with the same D-channel, are being used in parallel. Basically, one PRI/ISL/VNS or analog trunk (B-channel) is established to handle an incoming call from a calling set at the originating node to a called set at the terminating node. A second PRI/ISL/VNS trunk or analog (B-channel) is then established to handle the loop-back of the same call that is redirected from the called station back to a different set at the originating node.

The loop back may result from a call being treated by Network Message Services - Meridian Mail, Network Call Redirection (such as Network Call Forward All Calls, Network Call Forward No Answer, Network Call Forward Busy, and Network Hunting), or by Call Modification (such as a call transfer.) These tromboned trunks are redundant. The Trunk Anti-Tromboning (TAT) feature eliminates these redundant trunks after the call is answered. Refer to “Example of an anti-tromboning operation” on page 623 for a basic example of how TAT is applied.

TAT may function with a Meridian 1 interworking with another Meridian 1, or Meridian 1 interworking with a DMS¹. In the first case, each Meridian 1 must be equipped with TAT. In the latter case, the DMS must be equipped with the MCDN Release Link Trunk (RLT) feature. The trunks may be PRI or VNS.

TAT and Trunk Optimization Before Answer (TRO-BA) may be configured on the same system. In this case, TRO-BA would attempt to optimize the trunks before a call is connected, and TAT would then attempt to optimize the trunks after the call is connected. Please refer to the “Trunk Anti-Tromboning and Trunk Optimization - Before Answer both equipped on the same Meridian 1” on page 635 for more information and examples.

Example of an anti-tromboning operation

The following example may be used as a basic demonstration of TAT at work within a simple Meridian 1 to Meridian 1 interworking (other interworking scenarios are described in the section Anti-tromboning scenarios beginning on page 626).

In Figure 69, the following situation exists:

- Station B at the terminating node receives an incoming call over a PRI trunk from Station A at the originating node; or, an incoming CO call is tandemed from the originating node to the terminating node;
- Station B answers the call, and activates Call Transfer to Station C residing at the originating node. A second PRI trunk is established to handle the new call;
- Station B completes the call transfer, leaving Station A connected to Station C (which is in ringing state) using the two PRI trunks. The PRI trunks remain tromboned until Station C answers the call.

1. Throughout this chapter, a DMS switch should be interpreted as being either an SL-100, DMS-250, or DMS-100 equipped with RLT.

Figure 69
Before Trunk Anti-Tromboning operation

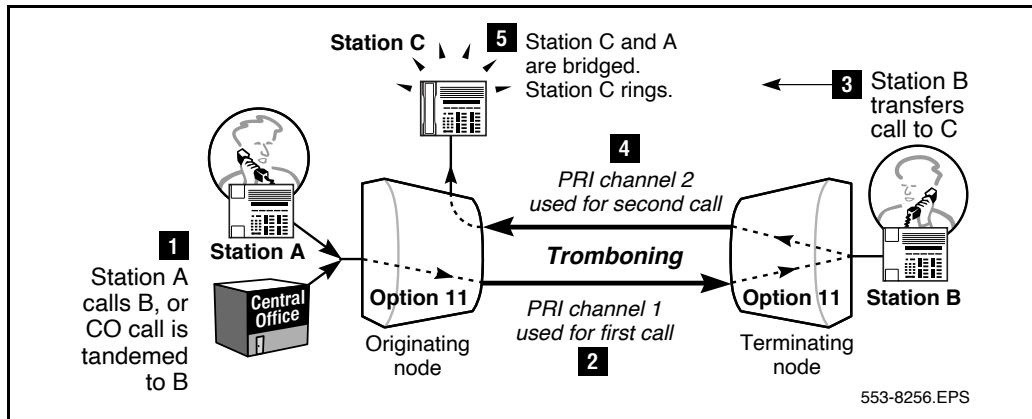
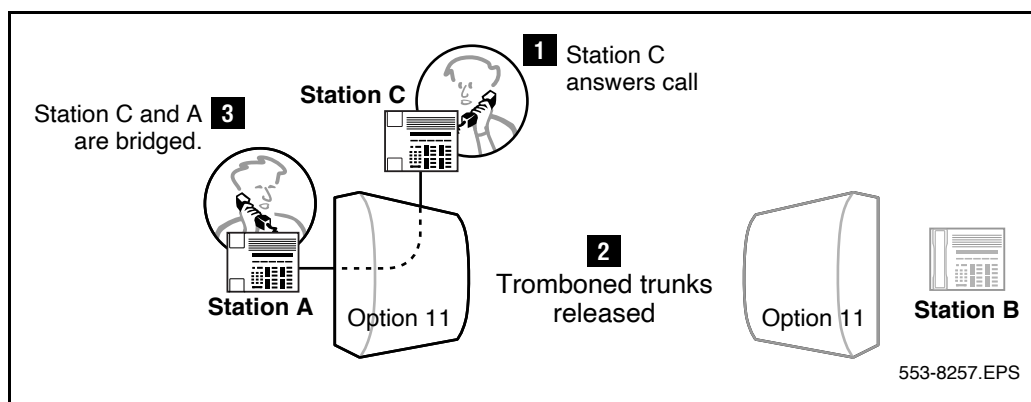


Figure 70 illustrates how the application of TAT eliminates the redundant ISDN PRI B-channels forming the loop-back. After Station C answers the call, Station A and Station C are bridged at the originating node and the redundant trunks are released, provided the trunks involved are associated with the same primary D-channel, and the same customer. This operation eliminates the use of extra network trunking facilities for the duration of the call.

Figure 70
After Trunk Anti-Tromboning operation



Note: If the TAT feature is not supported by either the originating or terminating node, or if the originating node is unable to bridge the call between Station A and Station C, then Station A and Station C remain connected via the tromboned trunks, for the duration of the call. Also, TAT would not release the tromboning trunks if Station A and Station C are both attendants.

Anti-tromboning scenarios

This section provides examples of various anti-tromboning scenarios, as applied with the Meridian 1 interworking with another Meridian 1, and a Meridian 1 interworking with a DMS equipped with RLT.

Please note that the following legend is used when showing connectivity in the illustrations.

Legend

———— PRI B-channel, or
ISL or VNS trunk

M1 = Meridian 1 switch equipped with TAT.

DMS = SL-100, DMS-250, or DMS-100 equipped with RLT.

Note: In the examples that follow, the Trunk Optimization Before Answer (TRO-BA) feature is not equipped.

Anti-tromboning operation for Network Call Redirection (case 1), Meridian 1 interworking with DMS

In Figure 71, the following case is represented for TAT as applied to a Network Call Redirection scenario (in the example, a case of a call forward is used), with two Meridian 1 switches interworking with a DMS.

- Station A located at an originating switch (Public Central Office or PBX) DMS node, makes an internodal call through a DMS and a tandem Meridian 1 node to Station B located at a terminating Meridian 1 node (first call);
- Station B, located at the terminating Meridian 1 node, is call forwarded through the tandem Meridian 1 node and the DMS, to Station C located at the originating switch (redirected call);
- Station C answers;
- Station A connects to Station C.

Figure 71
Anti-tromboning for Network Call Redirection (case 1), M1/DMS

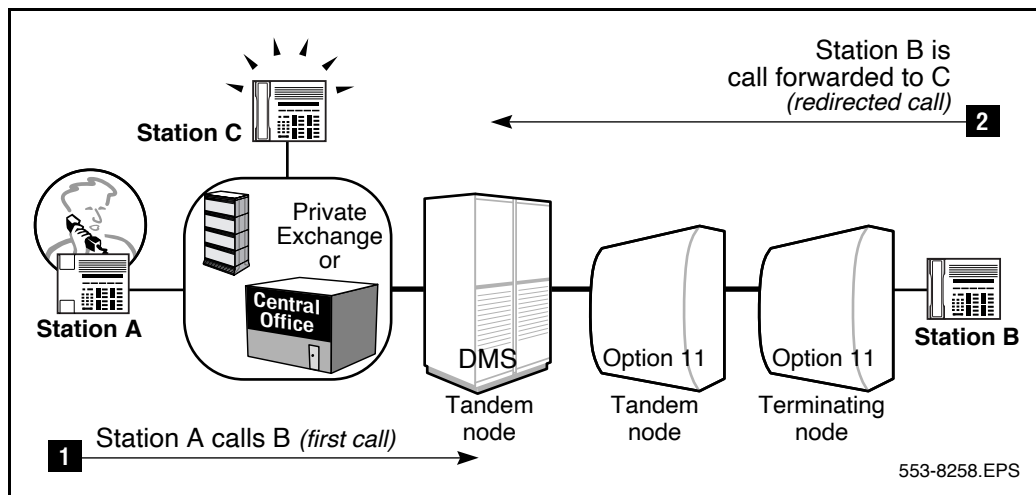
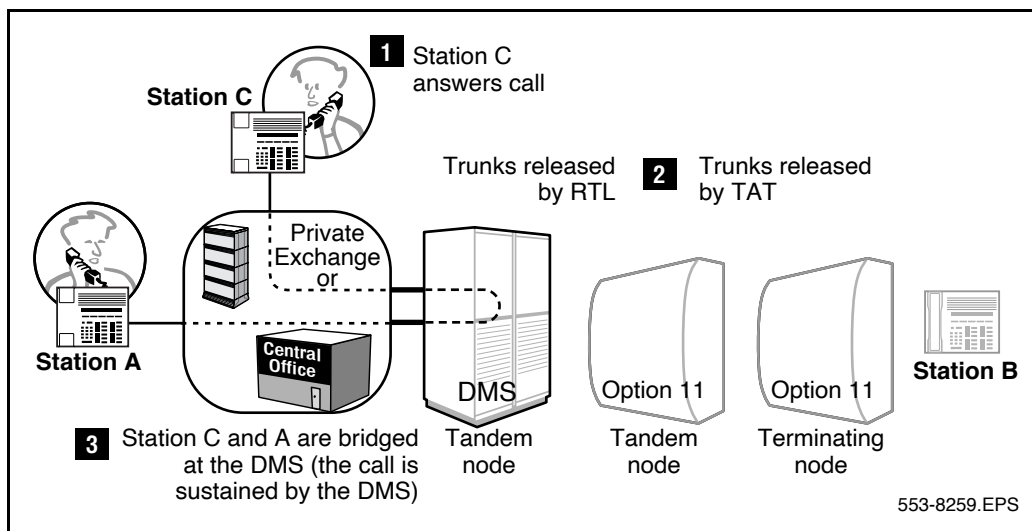


Figure 72 shows the results of anti-tromboning after Station C answers the call.

- The call is bridged at the DMS tandem node, between A and C;
- The trunks between the terminating Meridian 1 and the tandem Meridian 1 are released by TAT;
- The trunks between the tandem Meridian 1 and the DMS are released by the RLT feature.

Figure 72
Results of anti-tromboning for Network Call Redirection (case 1), M1/DMS



Anti-tromboning operation for Network Call Redirection (case 2), Meridian 1 interworking with DMS

In Figure 73, another example is presented where TAT optimizes redundant trunks due to a call forward. In this case, the trunks between a terminating Meridian 1 node and tandem Meridian 1 node are optimized. The originating node is a DMS. Station C is a centralized attendant or a Meridian Mail position.

- Station A located at an originating switch (Public Central Office or Private Exchange), makes an internodal call through a tandem DMS and a tandem Meridian 1 node to Station B located at a terminating Meridian 1 node (first call);
- Station B, located at the terminating Meridian 1 node, is call forwarded to Station C located at the tandem Meridian 1 node (redirected call). Station C is a centralized attendant or Meridian Mail position;
- Station C answers;
- Station A connects to Station C.

Figure 73
Anti-tromboning for Network Call Redirection (case 2), M1/DMS

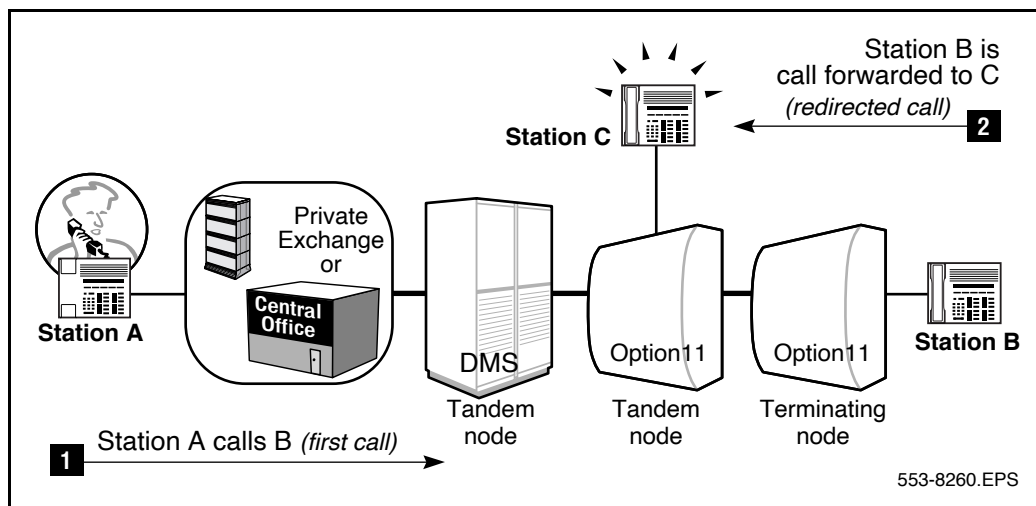
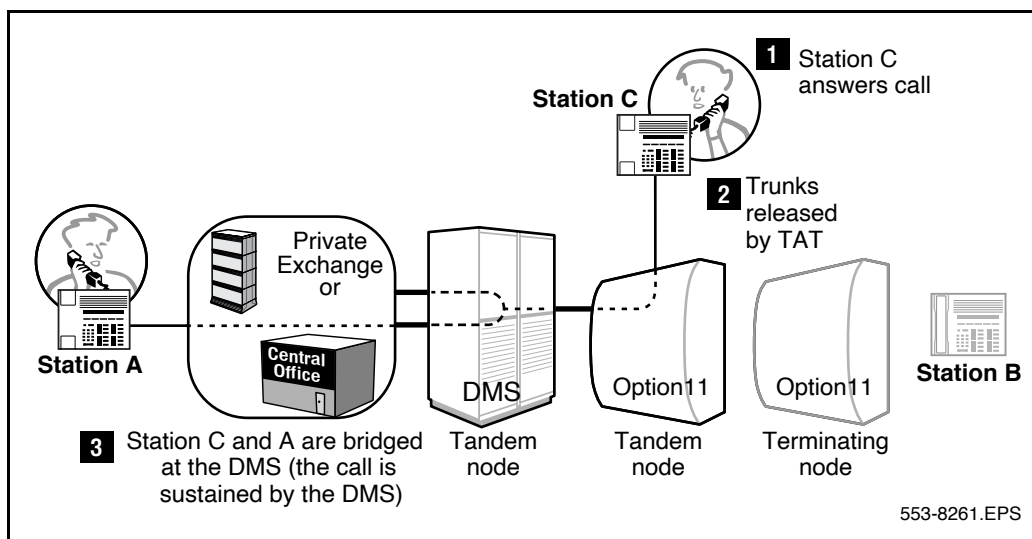


Figure 74 shows the results of anti-tromboning after Station C answers the call.

- The trunks between the terminating Meridian 1 and the tandem Meridian 1 are released by TAT;
- Station C and Station A are connected.

Figure 74
Results of anti-tromboning for Network Call Redirection (case 2), M1/DMS



Anti-tromboning operation for Call Modification, DMS interworking with Meridian 1

In Figure 75, the following case is represented for TAT as applied to a Call Modification scenario, with DMS interworking with Meridian 1 (in the example, a case of a call transfer is used. Please note that the same effect would take place if Station B conferences in Station C and then drops out, leaving Station A and Station C connected.)

- Station A, located at an originating Meridian 1 node, makes an internodal call through a tandem DMS node to Station B located at a terminating Meridian 1 node (first call);
- Station B, located at a terminating Meridian 1 node, answers the call and initiates a call transfer, through the tandem DMS node, to Station C located at the originating Meridian 1 node (second call);
- Station C answers;
- Station B completes the call transfer.

Figure 75
Anti-tromboning for Call Modification, DMS/M1

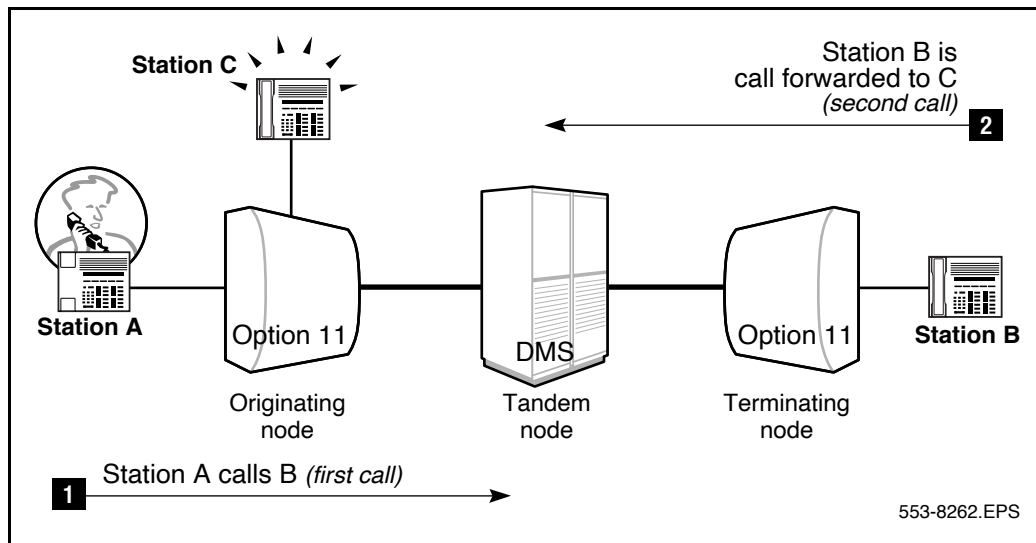
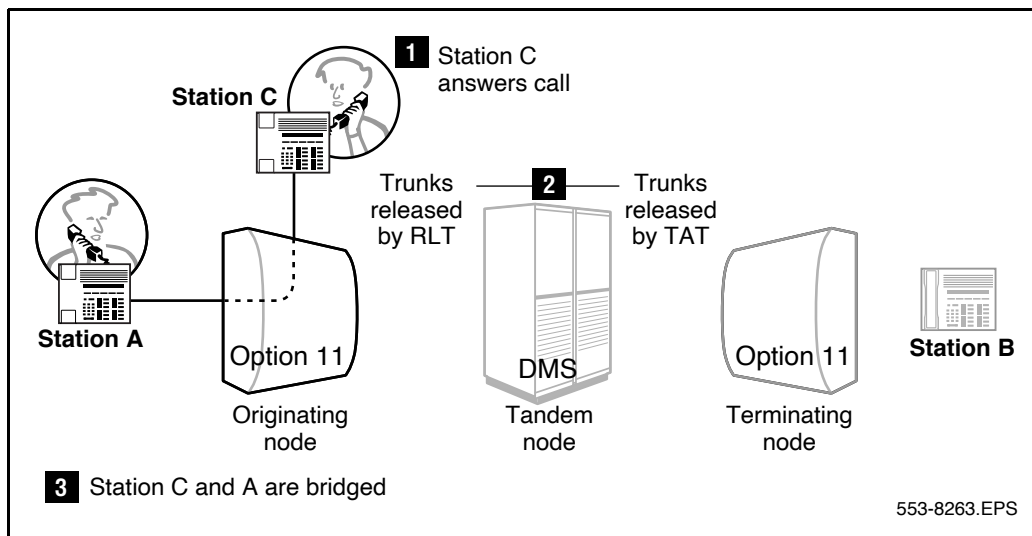


Figure 76 shows the results of anti-tromboning, after Station C answers the call.

- The call is bridged at the Meridian 1 originating node, between A and C;
- The trunks between the Meridian 1 terminating node and the DMS, and between the DMS and the Meridian 1 originating node and, are released by the RLT feature.

Figure 76
Results of anti-tromboning for network call transfer, DMS/M1



Anti-tromboning operation for Call Modification, Meridian 1 interworking with Meridian 1

In Figure 77, the following case is represented for TAT as applied to a Call Modification scenario, with Meridian 1 interworking with Meridian 1 (in the example, a case of a call transfer is used.)

- Station A, located at an originating Meridian 1 node, makes an internodal call through a tandem Meridian 1 node to Station B located at a terminating Meridian 1 node (first call);
- Station B, located at a terminating Meridian 1 node, answers the call and initiates a call transfer, through the tandem Meridian 1 node, to Station C located at the originating Meridian 1 node (second call);
- Station C answers;
- Station B completes call transfer.

Figure 77
Anti-tromboning for network call transfer, M1/M1

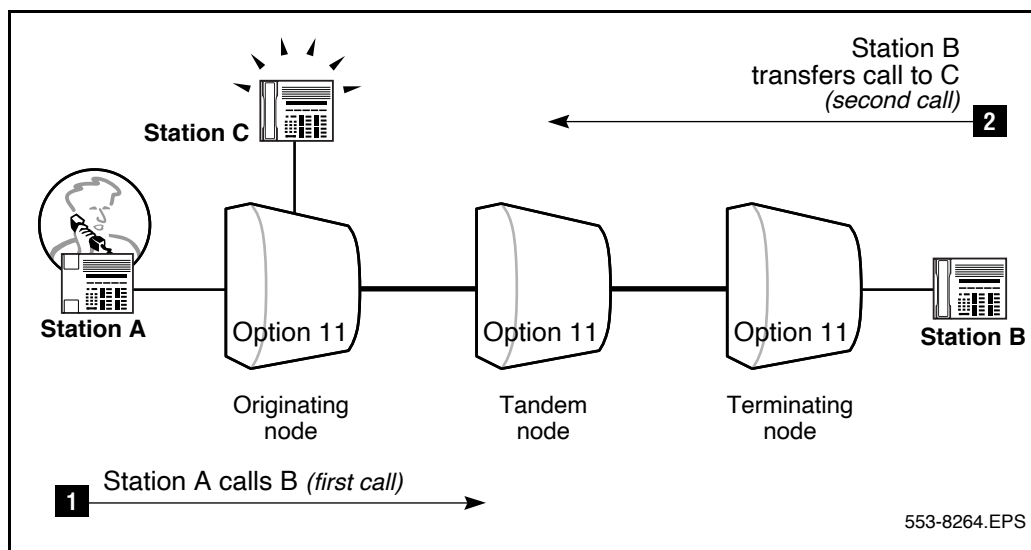
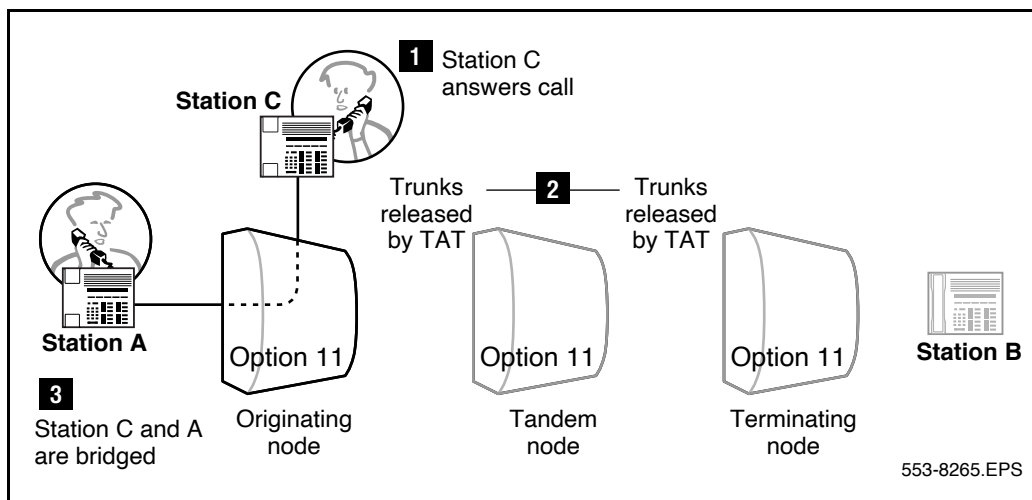


Figure 78 shows the results of anti-tromboning after Station C answers the call.

- The call is bridged at the Meridian 1 originating node, between Station A and Station C;
- The tromboning trunks between the Meridian 1 terminating node and the Meridian 1 tandem node, and between the Meridian 1 tandem node and Meridian 1 originating node, are released by TAT.

Figure 78
Results of anti-tromboning for network call transfer, M1/M1



Trunk Anti-Tromboning and Trunk Optimization - Before Answer both equipped on the same Meridian 1

TAT and TRO-BA (including Virtual TRO-BA) may be configured together on the same Meridian 1 switch.

While TAT and TRO-BA both provide trunk optimization capabilities, there are major differences pertaining to how each feature optimizes trunks.

- If TAT and TRO-BA are both configured on the same Meridian 1 switch, TRO-BA takes precedence over TAT (refer to “TRO-BA takes precedence over TAT” on page 636.) TRO-BA attempts to optimize a route before a call is answered, and then TAT attempts to optimize trunks after the call has been answered. The TAT feature enhances this capability by eliminating redundant trunks between Meridian 1 and switches equipped with the MCDN Release Link Trunk (RLT) feature.
- TAT and TRO-BA both eliminate redundant trunks for calls that are redirected by Network Call Forward All Calls, Network Call Forward No Answer, Network Call Forward Busy, or Network Hunting.
- TRO-BA will not optimize tromboned trunks resulting from a Call Transfer, while TAT will.
- TRO-BA will not optimize tromboned trunks resulting from an incoming trunk call, while TAT will.
- In the case of a call triangulation scenario, TRO-BA will attempt to eliminate tromboned trunks, while TAT will not. However, in this scenario, tromboning may actually result after TRO-BA has performed optimization. In this case, TAT will attempt to optimize the trunks. Refer to “TRO-BA and TAT functionality in a call triangulation scenario” on page 643.

Note: For more detailed information on the TRO-BA feature, refer to the ‘Trunk Optimization-Before Answer’ feature description module in this document.

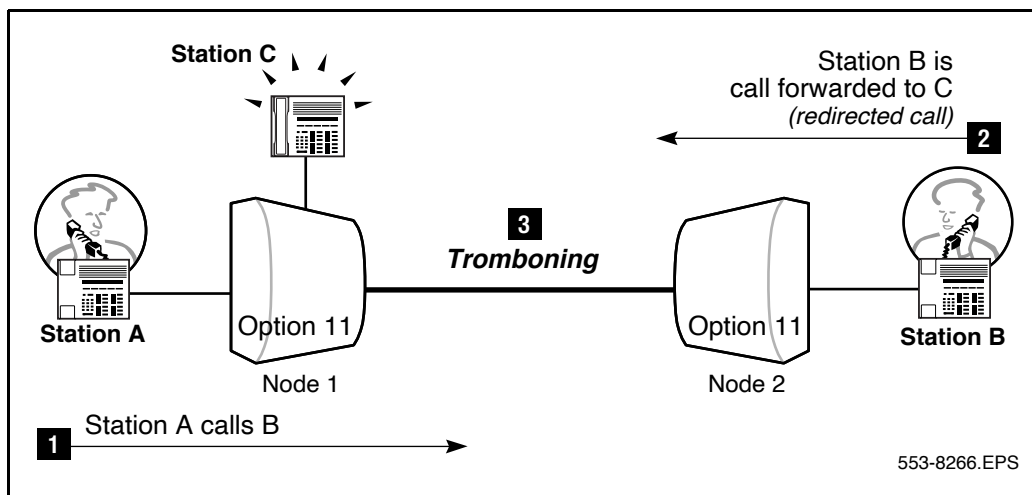
TRO-BA takes precedence over TAT

The following provides a basic example of how TRO-BA takes precedence over TAT in a call forward scenario.

Referring to Figure 79:

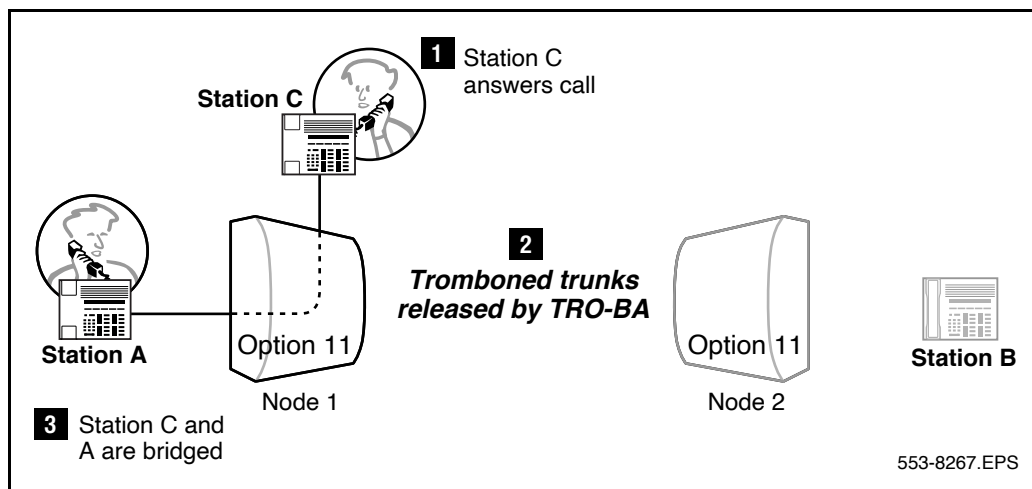
- Station A on Node 1 calls Station B on Node 2, over an ISDN PRI/ISL/VNS trunk;
- Station B on Node 2 is call forwarded all calls to Station C on Node 1, over another PRI/ISL/VNS trunk (redirected call.) This means that tromboning exists between Node 1 and Node 2;

Figure 79
TRO-BA/TAT operating together, before optimization in a call forward scenario.



Before Station C answers, TRO-BA releases the tromboned trunks between Node 1 and Node 2, and bridges Station A and Station C locally. Refer to Figure 80. Since there no longer exist any tromboned trunks, TAT is not invoked.

Figure 80
After optimization has been applied by TRO-BA.

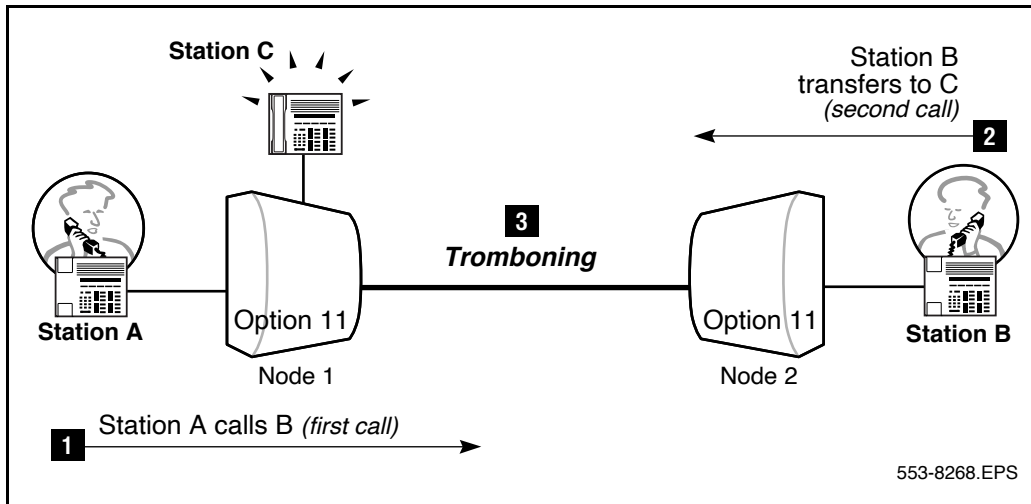


TRO-BA does not optimize trunks due to a call transfer - anti-tromboning is completed after the call is connected

As mentioned, TRO-BA will not release tromboned trunks that result from a call transfer. In this case, TAT will optimize the trunks, after the call is answered. Referring to Figure 81:

- Station A on Node 1 calls Station B on Node 2, over a PRI/ISL/VNS trunk (first call);
- Station B initiates call transfer to Station C on Node 1, over another PRI/ISL/VNS trunk, and completes the call transfer while Station C is still ringing. This means that a case of tromboning exists between Node 1 and Node 2 (second call);

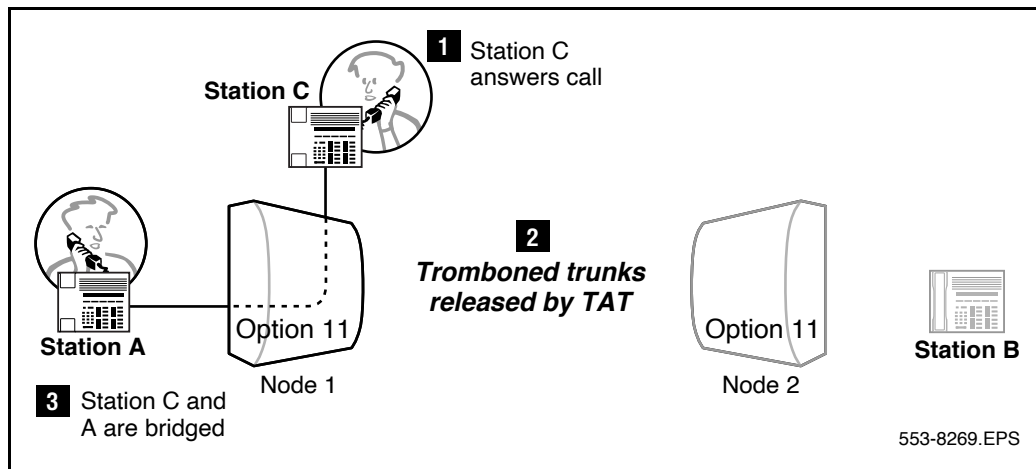
Figure 81
TRO-BA/TAT operating together, before optimization in a call transfer scenario.



Since TRO-BA does not release tromboned PRI trunks resulting from a call transfer, the trunks between Node 1 and Node 2 remain tromboned until Station C answers the call transfer.

While the call is still in ringing state, TRO-BA does not release the tromboned trunks. After Station C answers, TAT releases the tromboned trunks between Node 1 and Node 2, and bridges Station C and Station A locally. Refer to Figure 82.

Figure 82
After TAT applied to tromboned trunks due to a call transfer.



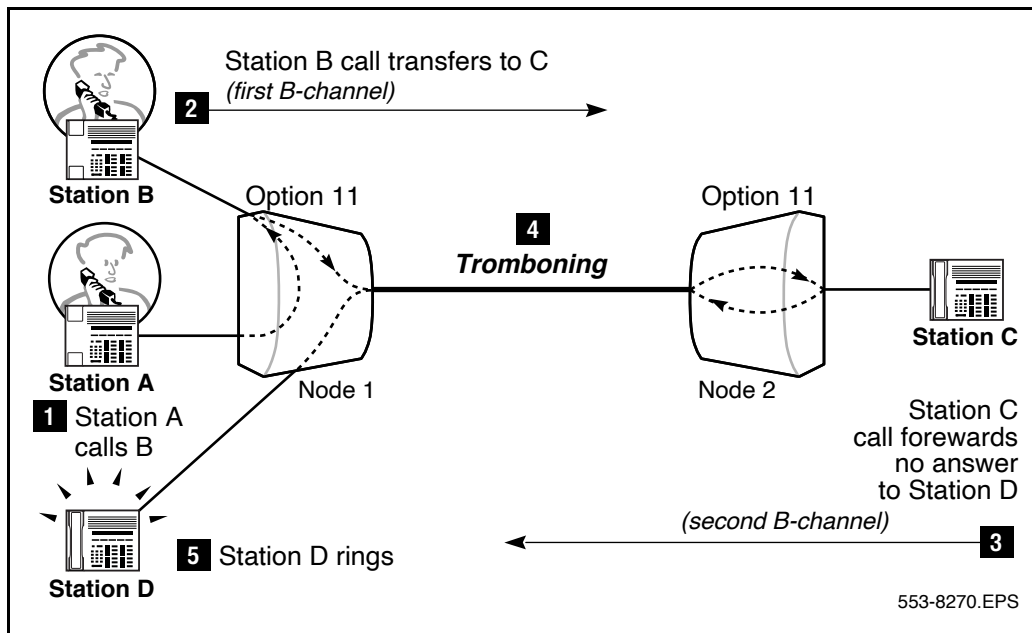
Trunk optimization with call redirection due to call transfer

Another case of TRO-BA and TAT interaction in the case of a call redirection initiated on the transfer key is as follows. Referring to Figure 83:

- Station A on Node 1 calls Station B, also on Node 1;
- Station B initiates a call transfer to Station C on Node 2, over a PRI/ISL/VNS trunk (first B-channel);
- Station C call forwards no answer to Station D on Node 1, over another PRI/ISL/VNS trunk (second B-channel, due to call redirection.) Therefore, tromboning of trunks exists between Node 1 and Node 2;

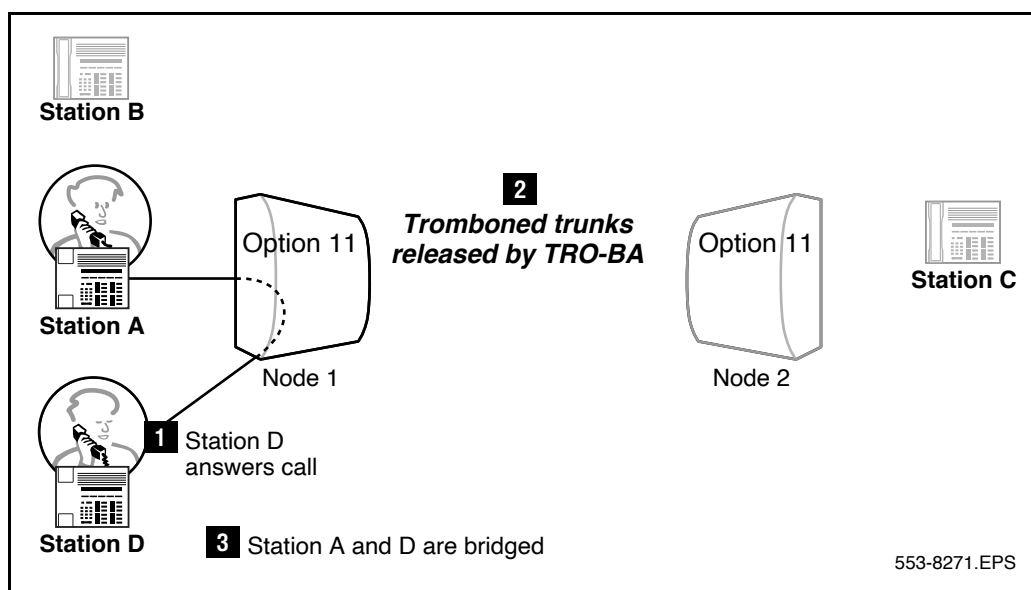
Figure 83

TRO-BA/TAT operating together, before optimization in a call transfer scenario.



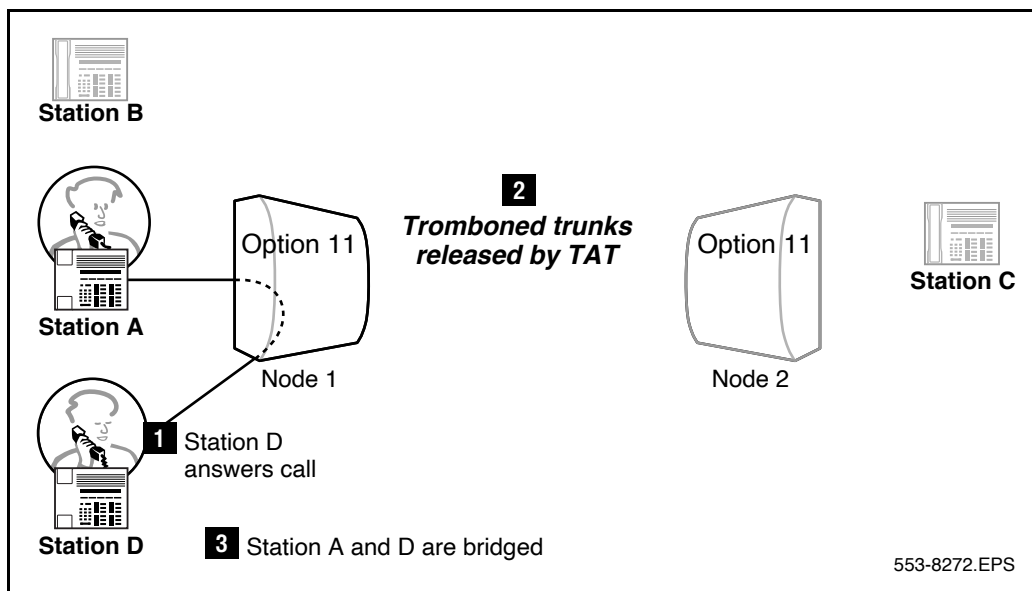
- Station B completes the call transfer while Station C is ringing. After the call is presented to Station D, TRO-BA optimizes the trunks, and bridges Station A and Station D locally. Refer to Figure 84.

Figure 84
Tromboned trunks released by TRO-BA.



- If Station B does not complete the call transfer while Station C is ringing, but waits until Station D answers, TRO-BA will not perform optimization.
- After Station D answers, TAT applies anti-tromboning to release the trunks between Node 1 and Node 2, and bridges Station A and Station D locally. Refer to Figure 85.

Figure 85
Tromboned trunks released by TAT



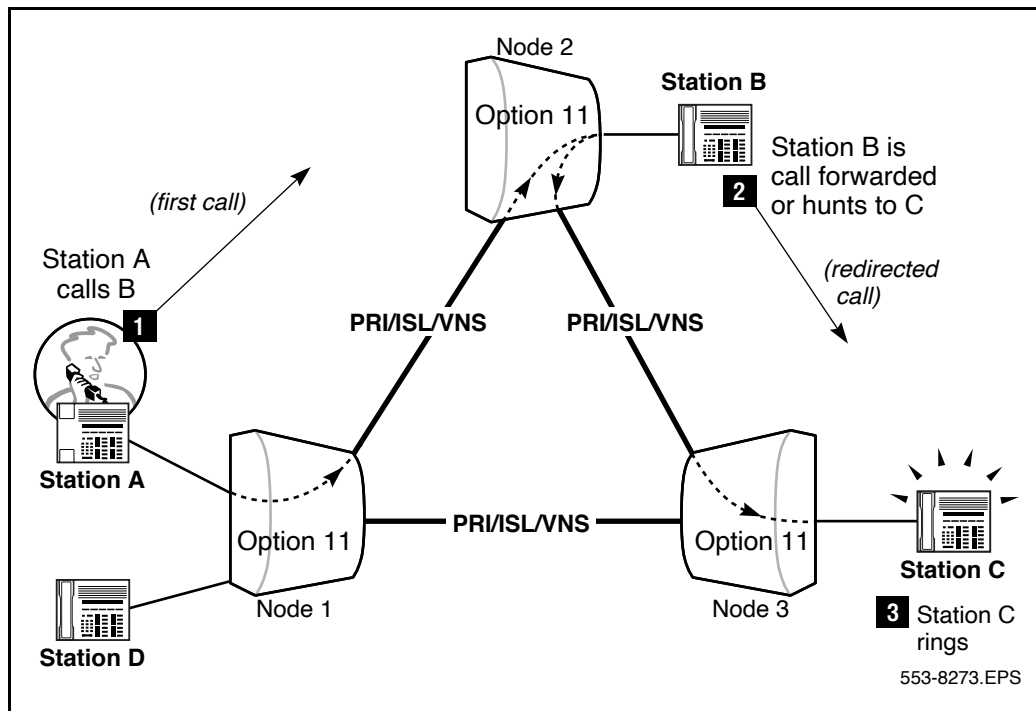
TRO-BA and TAT functionality in a call triangulation scenario

In the case of a call triangulation scenario, TRO-BA will attempt to eliminate tromboned trunks. TAT cannot perform this operation because a second D-channel is involved.

Referring to Figure 86:

- Station A on Node 1 calls Station B on Node 2 over a PRI/ISL/VNS trunk (first call);
- Station B on Node 2 is call forwarded or hunts to Station C on Node 3, over another PRI/ISL/VNS trunk (redirected call.) This means that two trunks are being used to handle one call;

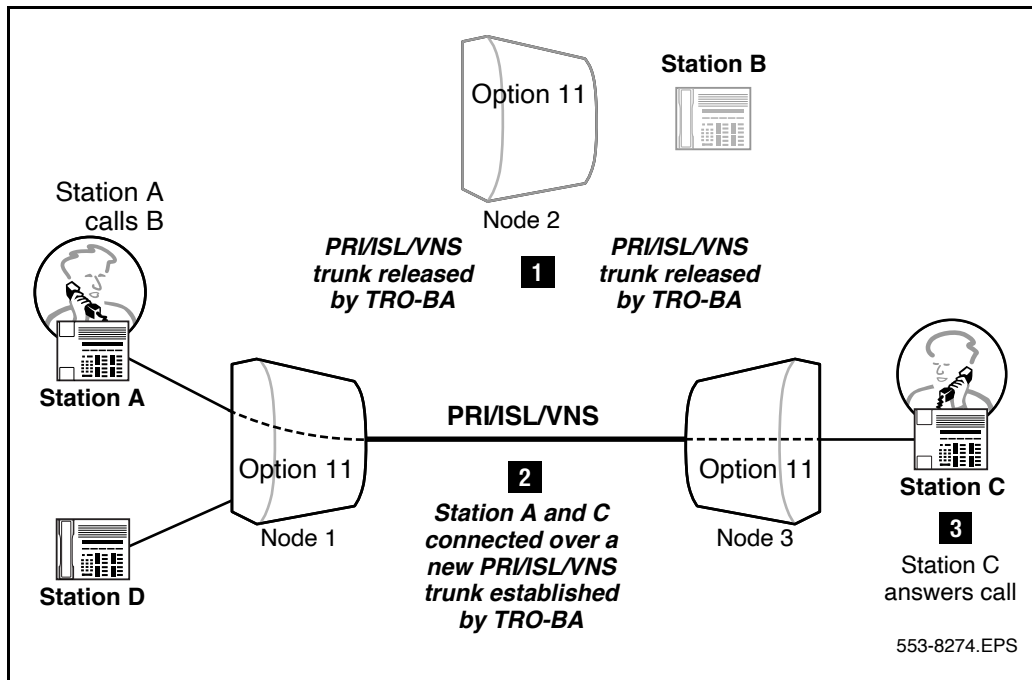
Figure 86
Call path before TRO-BA has been applied.



Before Station C answers, TRO-BA releases the trunks between Node 1 and Node 2, and between Node 2 and Node 3, and establishes a new PRI/ISL/VNS trunk connection between Node 1 and Node 3. Refer to Figure 87.

Note: The TAT feature cannot provide this functionality, because multiple D-channels are used.

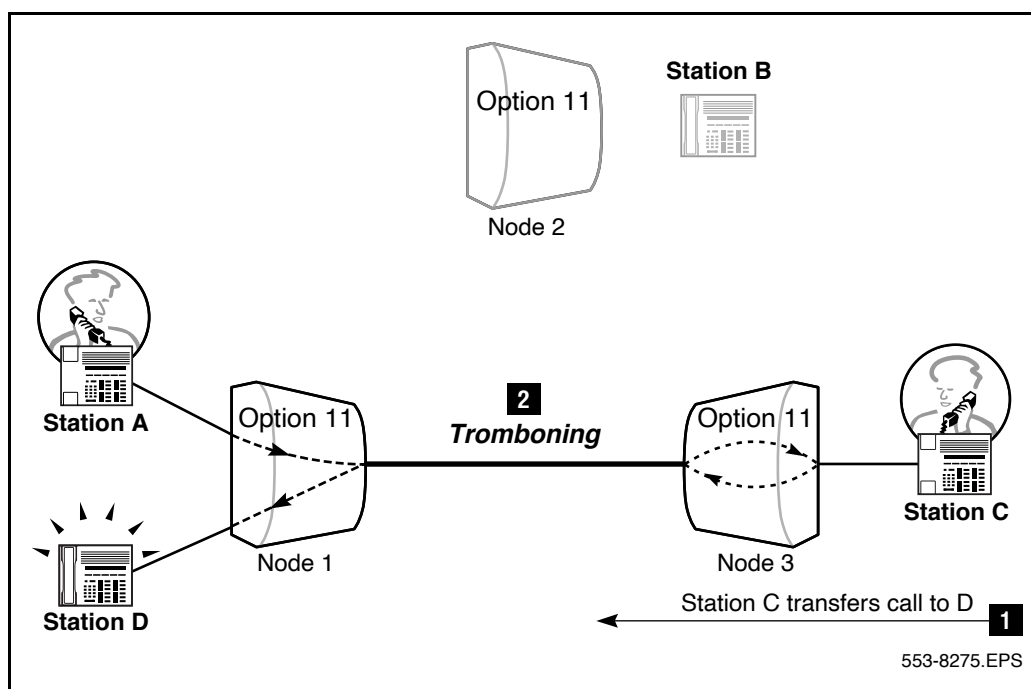
Figure 87
Call path after TRO-BA has been applied.



Tromboning may actually result after TRO-BA has performed optimization. In this case, TAT will attempt to optimize the trunks. An explanation whereby this scenario may occur follows.

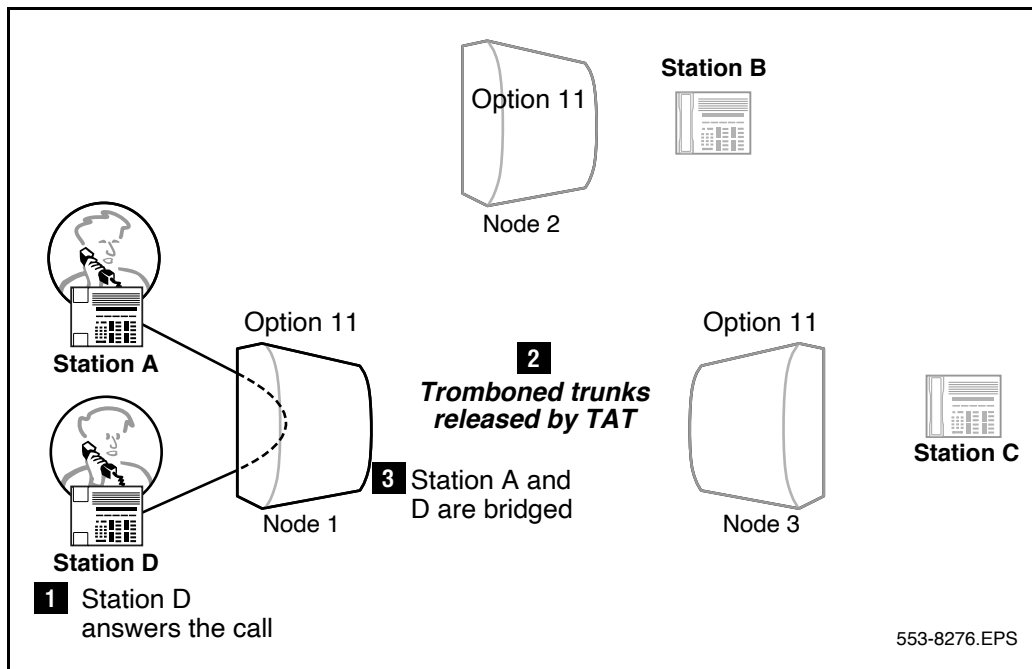
- Station C answers the call, and initiates a call transfer to Station D on Node 1, over a separate PRI/ISL/VNS trunk. This results in tromboning between Node 1 and Node 3. Refer to Figure 88.

Figure 88
Tromboning of trunks after TRO-BA has been applied.



- After Station D answers the call, and Station C completes the call transfer, TAT releases the tromboned trunks between Node 3 and Node 1, leaving Station D and Station A bridged locally. Refer to Figure 89.

Figure 89
Tromboned trunks released by TAT



Operating parameters

If the NTAK02BB SDI/DCH circuit card is used on systems with a software vintage earlier than Generic X11 Release 21.40, it must be ensured that the power is removed from the card before it is used for the operation of the Trunk Anti-Tromboning feature. This will normally take place during software upgrade; otherwise, the card must be removed and the power must be reset.

To support the Trunk Anti-Tromboning feature within a dedicated ISL/VNS configuration, a NTAK02 (minimum vintage **BB**) SDI/DCH circuit card must be used.

The Trunk Anti-Tromboning feature will only be supported on the NTBK51BA D-Channel Daughterboard, for D-channel handling with PRA.

Anti-tromboning will be performed only after the third party answers, and provided that the tromboning PRI/ISL/VNS or analog trunks are associated with the same primary D-channel (with or without a backup D-channel) and the trunks are associated with the same customer. Anti-tromboning will not be performed if the tromboning trunks belong to different customers on the same node (even though they are associated with the same primary D-channel.)

Anti-tromboning will not be performed for a tromboned call between two attendants on the same node.

There are two types of protocols used for TAT operations, depending on the interface type. One is for the Meridian 1 to Meridian 1 interface, and the other is for Meridian 1 to a DMS equipped with the Meridian Customer Defined Network Release Link Trunk (RLT) feature.

If non-ISDN PRI trunks are involved in a call transfer call, ISDN signaling messages cannot be sent and anti-tromboning will not be performed.

Anti-tromboning will not be performed for tromboned trunks associated with a call originating on a set, and routed back to the same set.

The Trunk Anti-Tromboning feature will only be supported on the Multi-purpose Serial Data Link (MSDL) card, for D-channel handling.

Anti-tromboning will not be performed for ISDN Basic Rate Interface (BRI) trunks, except if the trunks are Virtual Network Services bearer trunks.

TAT can cause a momentary interruption in data transmission during optimization. When TAT operations are performed at multiple tandem nodes, this effect is cumulative (in the milliseconds range.) Therefore, the impact of this loss is dependent on the terminals on both ends of the transmission, and may be recovered through re-transmission.

Feature interactions

Note: The feature interactions described in this section assume that only TAT (and not TRO-BA) is configured on a Meridian 1 system.

If TRO-BA is also configured with TAT on the same system, TRO-BA will take precedence over TAT, where it can.

Attendant

If an attendant has activated Busy Verify or Barge-in at the time that a message to invoke TAT is received, the anti-tromboning operation will be aborted.

Automatic Call Distribution (ACD)

The Trunk Anti-Tromboning feature will perform anti-tromboning operations to eliminate the PRI trunks associated with the same D-channel due to the following ACD operations: Enhanced Network Call Forward; Network ACD; Interflow Options; and Enhanced Interflow.

If an ACD agent is being observed at the time that a message to invoke TAT is received, the anti-tromboning operation will be aborted.

If an incoming PRI call that is in the ACD queue is answered by a Recorded Announcement (RAN), then the anti-tromboning operation will be performed only after an ACD agent answers the call.

Call Park Network Wide

The Trunk Anti-Tromboning feature is invoked if programmed at all interim Private Branch Exchanges (PBXs) in the call.

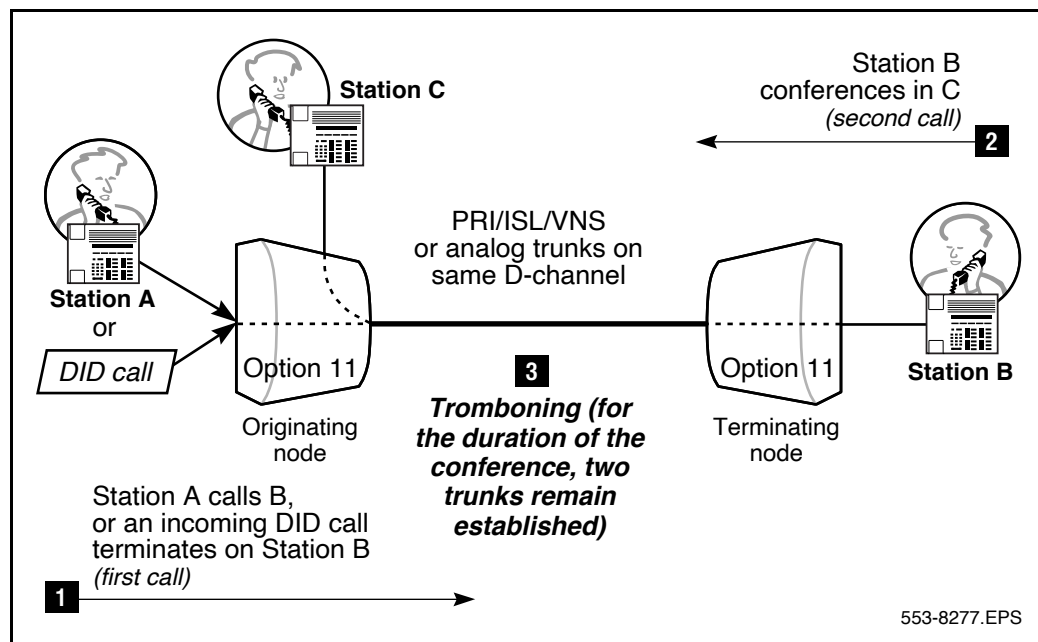
Conference

If the Conference feature is activated, the Trunk Anti-Tromboning feature will perform the anti-tromboning operations only when there are two parties remaining in the call, and the two parties are using PRI trunks associated with the same D-channel.

Referring to Figure 90:

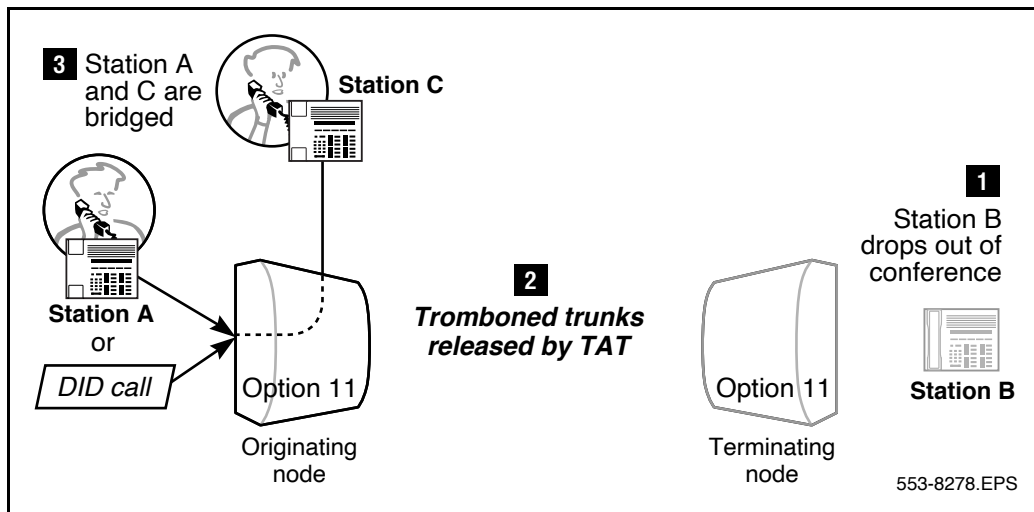
- Station A, or an incoming DID trunk call, at the originating node, is connected to Station B at the terminating node over a PRI/ISL/VNS or analog trunk (first call);
- Station B conferences in Station C, also located at the originating node, over a second PRI/ISL/VNS or analog trunk that is associated with the same D-channel as the first (second call);
- During the extent of the three-party conference, the two trunks are tromboned.

Figure 90
Conference call before TAT



When Station B drops out of the conference, anti-tromboning is invoked (TAT will not be invoked as long as the conference call situation exists.) The two tromboned trunks are released, and Station A, or the DID trunk, and Station C are bridged. This is illustrated by Figure 91.

Figure 91
Conference call after TAT



Digital Private Signaling System (DPNSS1) Route Optimization

There is no interworking between TAT and DPNSS1 Route Optimization at the MCDN/DPNSS1 gateway.

End to End Signaling

If End to End Signaling is present when TAT is being invoked, then anti-tromboning will be aborted.

External Recorded Announcement.

If an attendant originates a call which, through call modification or call redirection creates tromboned trunks, and eventually terminates on a RANX equipment, TAT will not optimize the trunks. That is, TAT will not release tromboning PRI/ISL/VNS trunks resulting from an attendant initiating an outgoing trunk call.

INIT ACD Queue Call Restore

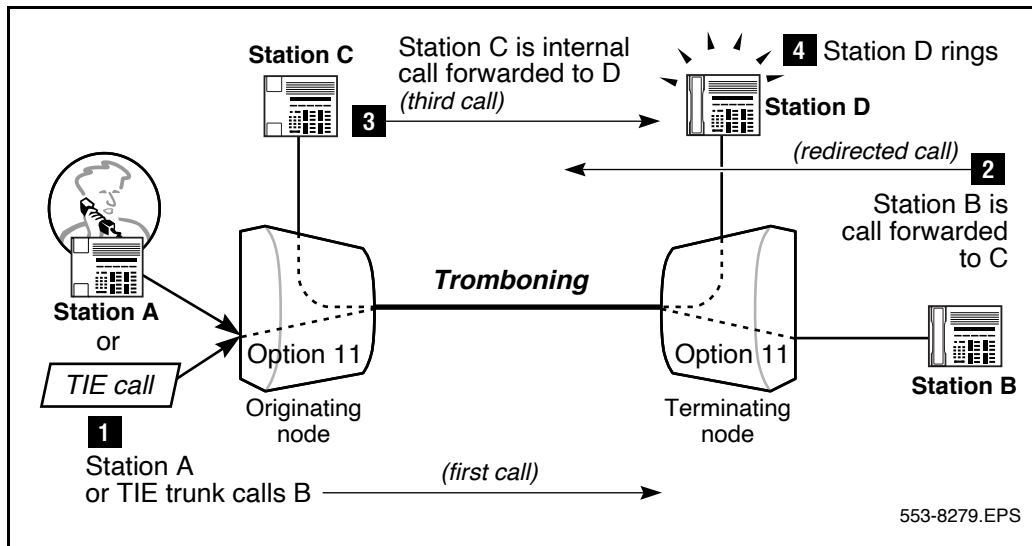
Trunk Anti-Tromboning is not supported on a call restored by INIT ACD Queue Call Restore.

Internal Call Forward (Meridian 1 to Meridian 1)

TAT will not affect the operation of Internal Call Forward between two Meridian 1 switches, while applying anti-tromboning to trunks. Referring to Figure 92:

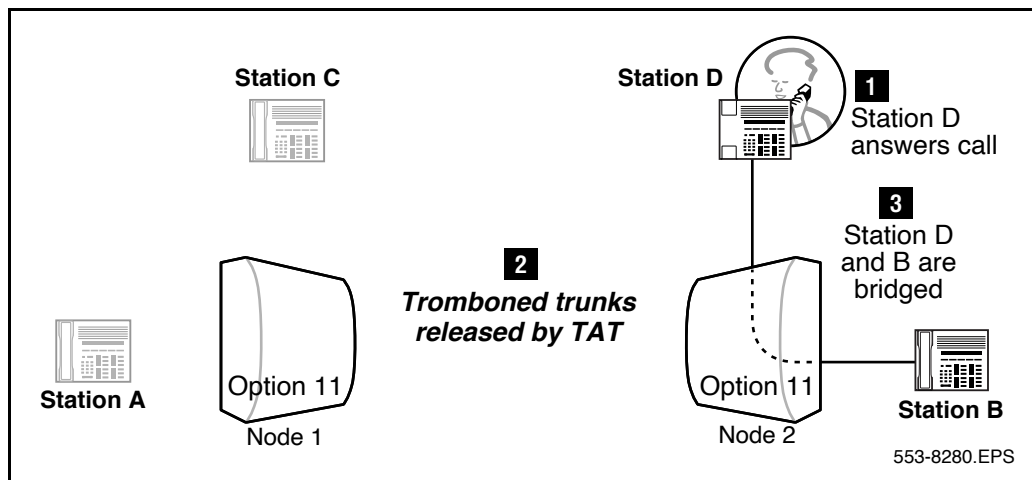
- Station A, or an incoming TIE trunk, on Node 1 calls Station B at Node 2. The call terminates on Station B at Node 2, over a PRI/ISL trunk (first call);
- Station B on Node 2 is call forwarded to Station C on Node 1 (redirected call). The trunks are tromboned.
- If the call is determined to be internal, and Station C is Internal Call Forwarded to Station D, the call is automatically forwarded to Station D on Node 2 over a third PRI/ISL trunk.

Figure 92
Internal Call Forward before TAT, Node 2 is a Meridian 1.



After Station D answers the call, TAT releases the tromboned trunks between Node 1 and Node 2. Refer to Figure 93.

Figure 93
Internal Call Forward after TAT, Node 2 is a Meridian 1

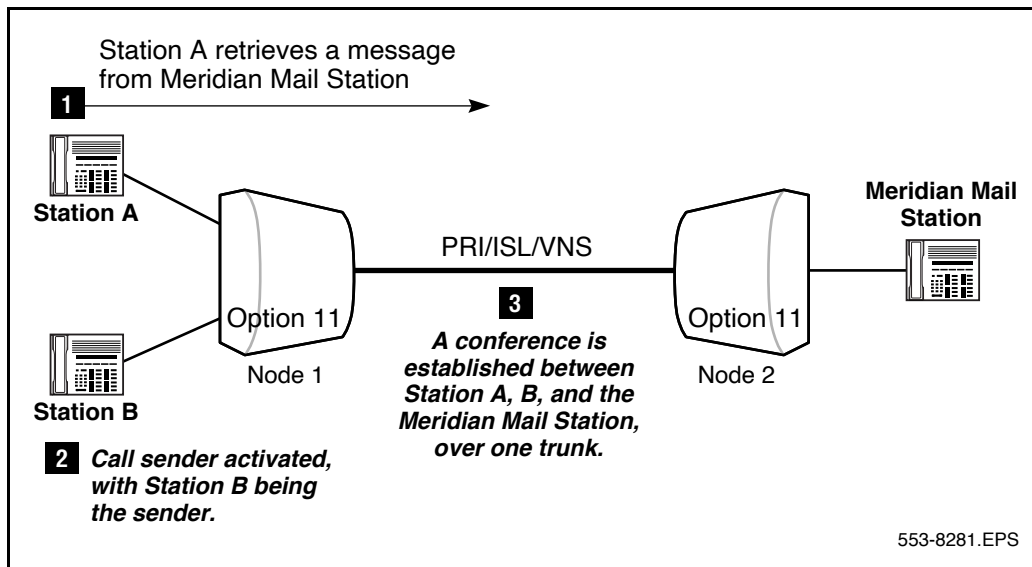


Meridian Mail

The TAT feature will release tromboned PRI/ISL/VNS trunks arising from the application of the Auto Attendant, Thru-Dialing, and Operator Revert capabilities of Meridian Mail. If Network Message Services is activated, the associated Call Sender capability does not create an additional trunk when it is activated. Therefore, TAT is not applied. Referring to Figure 94:

- Station A, on Node 1, retrieves a message from a Meridian Mail station located on Node 2;
- Call Sender is activated, with Station B, located on Node 1, being the sender;
- Station A directly conferences in Station B. A consultation conference between Station A, Station B, and the Meridian Mail station is then established over one PRI/ISL/VNS trunk.

Figure 94
Call Sender operation, with Network Message Services active



Network Attendant Service (NAS)

If both TAT and NAS are equipped in a Meridian 1 network, the NAS feature takes precedence over TAT, if NAS is equipped end-to-end. There is no interworking between NAS and TAT. Refer to Figure 95 for the following calling scenarios.

Scenario 1

Station C at Node 2 calls the attendant at the remote node (Node 3), over a PRI trunk. The attendant extends the call to Station D, also at Node 2, over a separate PRI trunk. These trunks are considered to be tromboned.

Case 1 - TAT and NAS are equipped on all nodes

When Station D answers, The NAS functionality drops the tromboned trunks, and Station C and Station D are bridged.

Case 2 - NAS is not equipped

If NAS is not equipped, TAT performs anti-tromboning operations, removing the tromboned trunks and bridging Station C and Station D.

Scenario 2

Station A at Node 1 calls the attendant at Node 3, and the attendant extends the call to Station B on Node 1.

Case 1 - TAT and NAS are equipped on all nodes

If NAS and TAT are both equipped between Node 1 and Node 2, and between Node 2 and Node 3, then NAS takes precedence over TAT to optimize the tromboned trunks between Node 1 and Node 2, and Node 2 and Node 3.

Case 2 - NAS is not equipped

If only TAT is equipped between Node 1 and Node 2, and between Node 2 and Node 3, then TAT will release the tromboned trunks between Node 1 and Node 2, and Node 2 and Node 3.

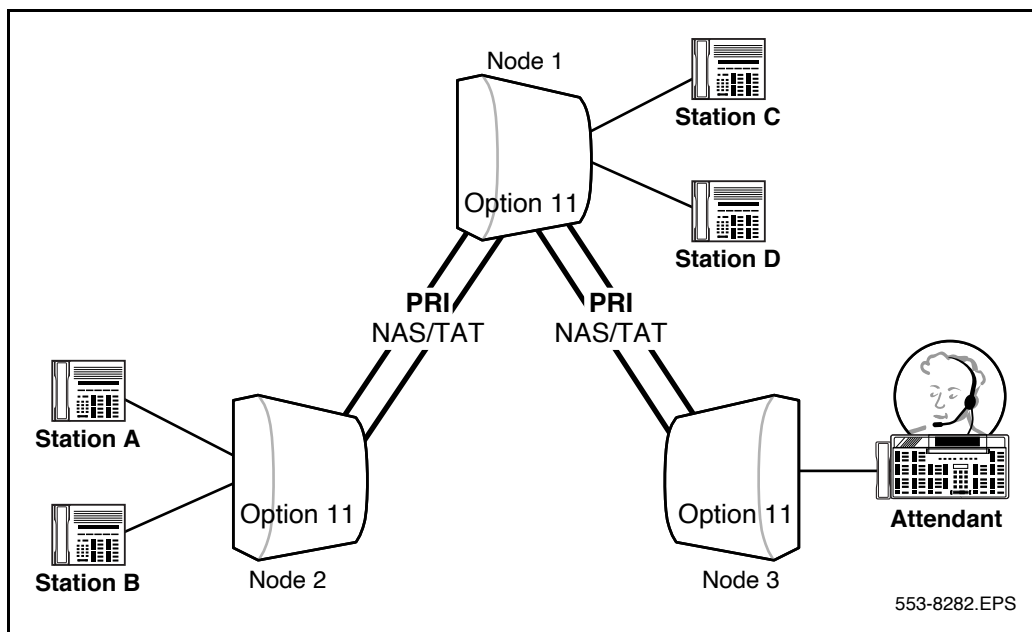
Case 3 - NAS is equipped between Node 1 and Node 2, and TAT is equipped between Node 2 and Node 3

If only NAS is equipped between Node 1 and Node 2, and only TAT is equipped between Node 2 and Node 3, then TAT releases the tromboned trunks between Node 2 and Node 3, and the call is bridged at Node 2. NAS does not optimize the tromboned trunks between Node 1 and Node 2.

Case 4 - TAT is equipped between Node 1 and Node 2, and NAS is equipped between Node 2 and Node 3

If only TAT is equipped between Node 1 and Node 2, and only NAS is equipped between Node 2 and Node 3, no anti-tromboning will be performed, because NAS is not equipped end to end.

Figure 95
NAS/TAT interworking



Network Call Pickup

The TAT feature will optimize tromboned trunks arising from the operation of the Network Call Pickup feature. For example, in Figure 96, Station A on Node 1 makes a call to Station B on Node 2 over a PRI/ISL/VNS or analog trunk. The call is picked up by station C on Node 1, over a separate PRI/ISL/VNS or analog trunk. After the call has been picked up, TAT will apply anti-tromboning to release the two tromboned PRI trunks. Station A and Station C are connected locally. Refer to Figure 97.

Figure 96
Network Call Pickup before TAT

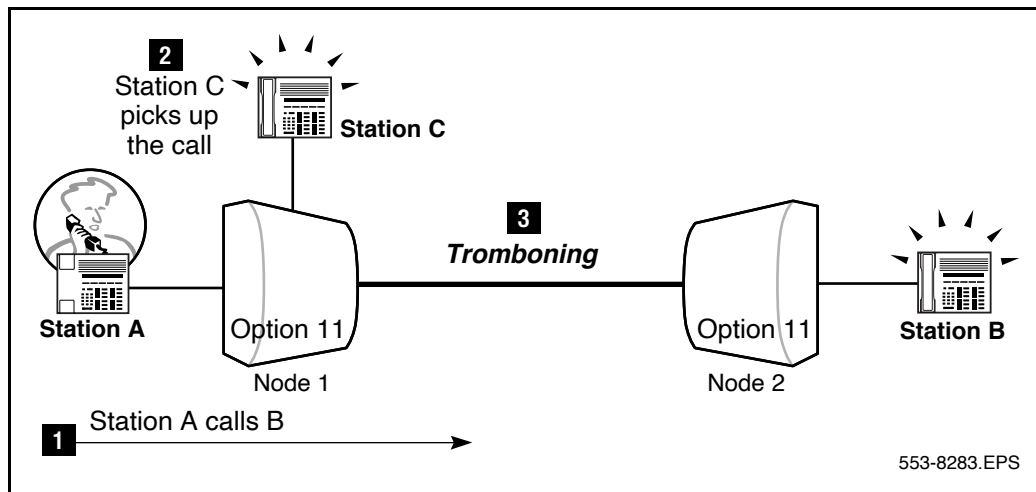
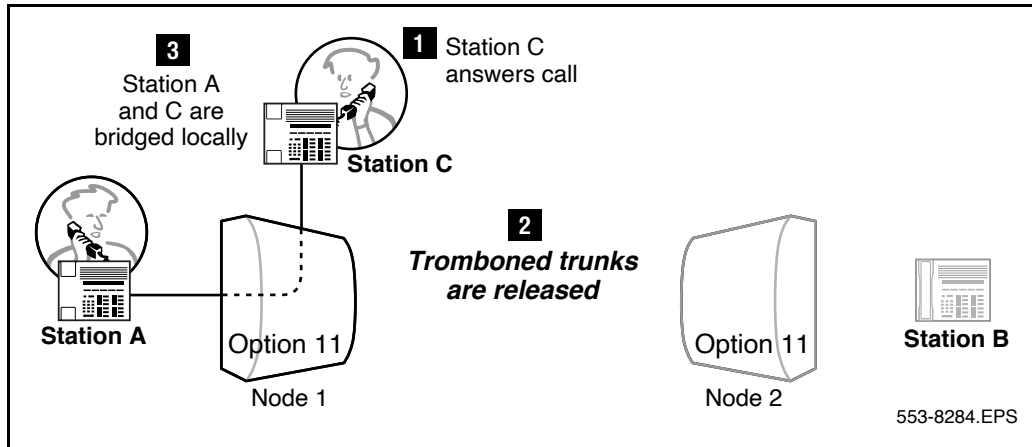


Figure 97
Network Call Pickup after TAT



Network Call Redirection

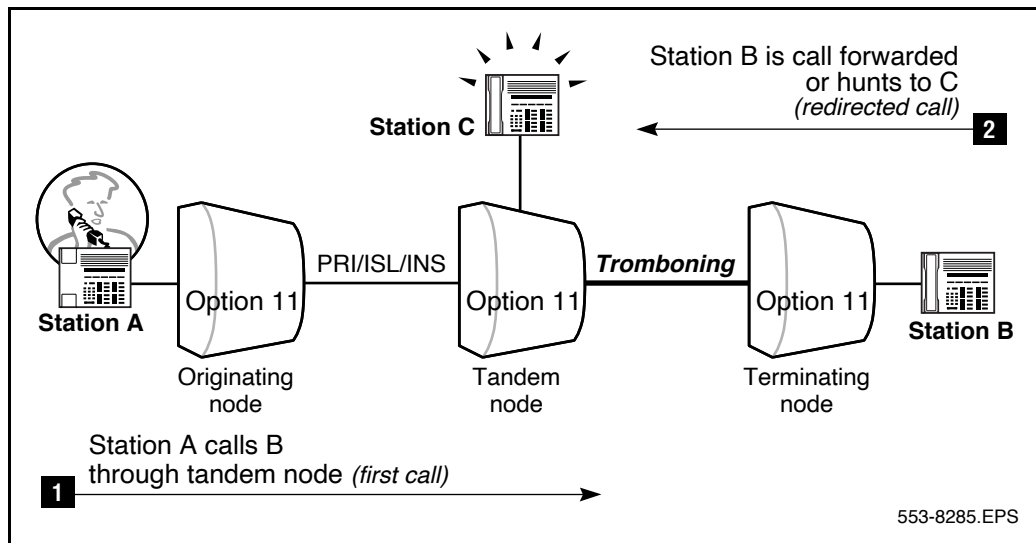
The Trunk Anti-Tromboning feature eliminates the tromboning of PRI/ISL/VNS or analog trunks resulting from the operation of any of the following Network Call Redirection features:

- Network Call Forward Unconditional
- Network Call Forward No Answer
- Network Call Forward Busy
- Network Call Forward by Call Type
- Network Hunt
- Internal Call Forward

Figure 98 illustrates the following scenario:

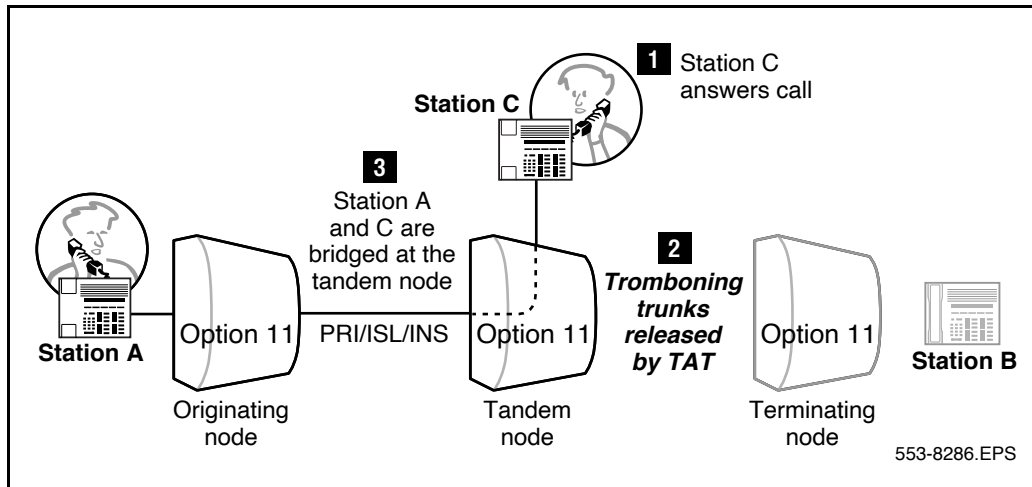
- Station A, on the originating node, calls Station B, on the terminating node, across a tandem node over a PRI/ISL/VNS trunk (first call);
- Station B is call forwarded or hunts over another PRI/ISL/VNS trunk, to Station C located on a tandem node (redirected call.) This causes tromboning between the terminating node and the tandem node.

Figure 98
Network Call Redirection before TAT



After Station C answers, anti-tromboning is invoked. This causes the tromboned trunks to be released. Station A and Station C are connected at the tandem node. The results of this anti-tromboning are shown in Figure 99.

Figure 99
Network Call Redirection after TAT



Radio Paging System

If an attendant originates a call which, through call modification or call redirection creates tromboned trunks, and eventually terminates on a Radio Paging equipment, TAT will not optimize the trunks. That is, TAT will not optimize tromboning PRI/ISL/VNS trunks resulting from an attendant initiating an outgoing trunk call.

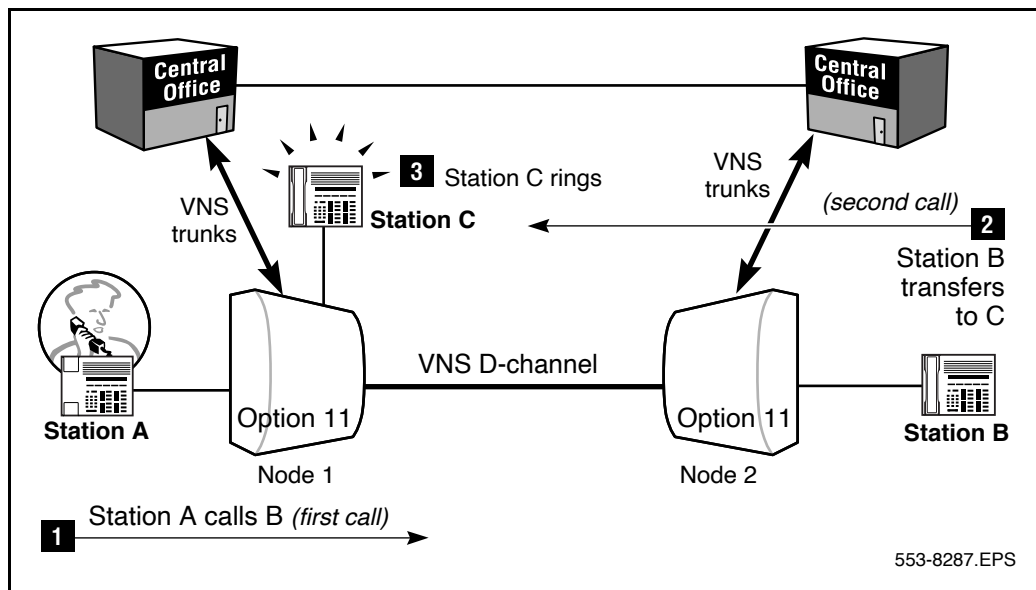
Trunk Route Optimization - Before Answer

See “Trunk Anti-Tromboning and Trunk Optimization - Before Answer both equipped on the same Meridian 1” on page 635.

Virtual Network Services (VNS)

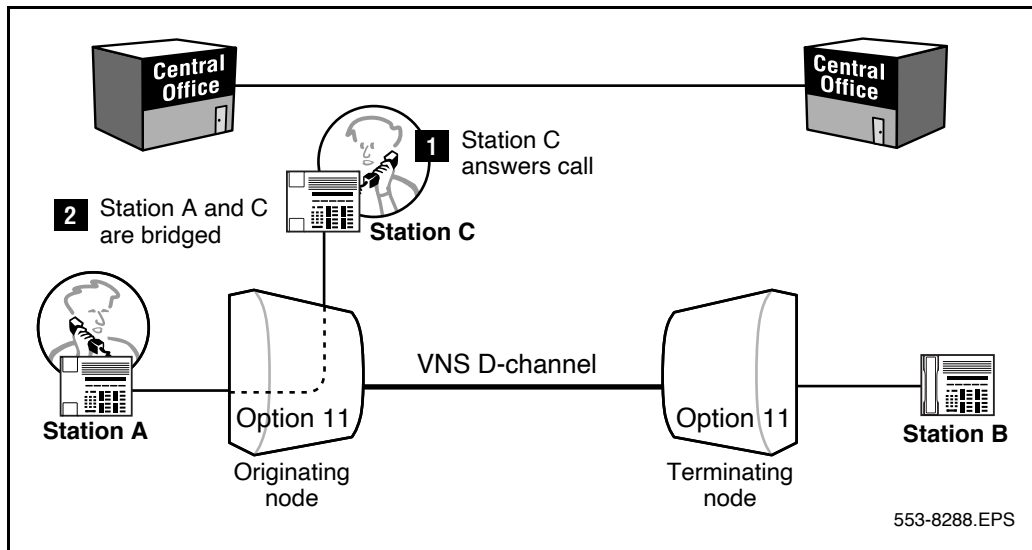
The Trunk Anti-Tromboning feature will perform anti-tromboning operations to eliminate tromboned trunks (physical B-channels) associated with the same VNS D-channel. For example, in Figure 100, Station A at the originating node calls Station B at the terminating node, over a VNS D-channel trunk. Station B transfers the call to Station C, also located at the originating node. The physical B-channel trunks associated with the D-channels are considered to be tromboned.

Figure 100
VNS before TAT



After Station C answers, the tromboned trunks are released, and Station A and Station C are bridged. Refer to Figure 101.

Figure 101
VNS after TAT.



Feature packaging

A new software package has been introduced for the Trunk Anti-Tromboning feature. This is TATO, package 312. The following packages are required as dependencies:

- Multi-purpose Serial Data Link (MSDL) package 222;
- Integrated Services Digital Network (ISDN) package 145; and
- Integrated Service Digital Network Signaling Link (ISL) package 147; or
- 2.0 Mbit Primary Rate Interface (PRI2) package 154; and, optionally
- Advanced ISDN Network Services (NTWK) package 148.

Feature operation

The Trunk Anti-Tromboning feature is transparent to the user, provided all datafills and hardware are assigned and equipped properly.

Trunk Optimization (Meridian 1 to Meridian 1 only)

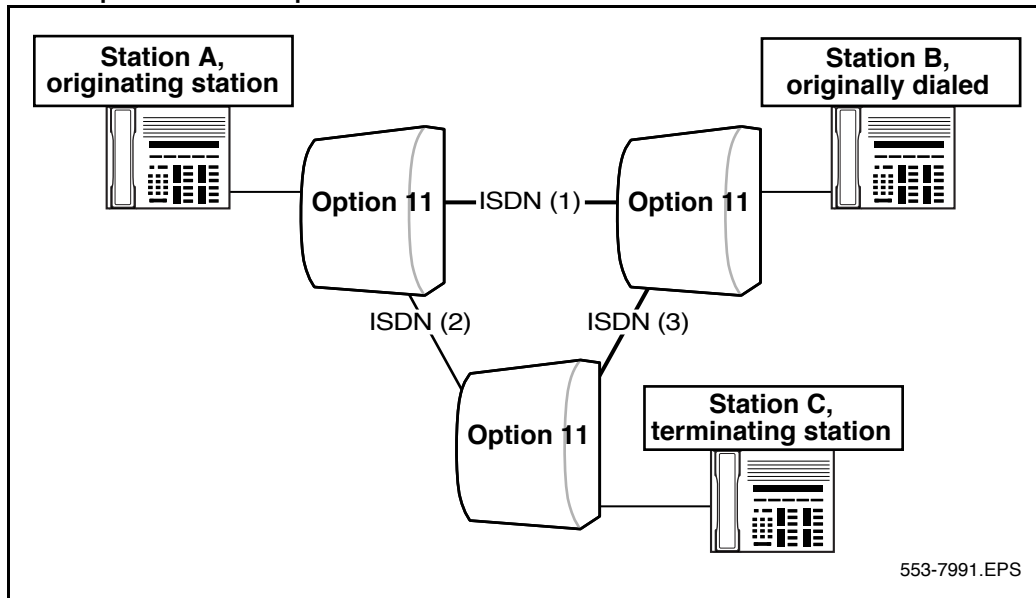
Description

Trunk Optimization provides better utilization of the available networking facilities. For calls originated from a set on the private Meridian 1/SL-1 ISDN, the Trunk Optimization feature provides the Meridian 1 with the ability to automatically optimize trunks before answer for redirected calls (Call Forward All Calls, Call Forward No Answer, Hunting, Busy Conditions) to avoid unnecessary usage of additional PRI or ISL channels/trunks.

TRO does not apply to DID calls or calls originated by a non-ISDN TIE trunk.

Figure 102 illustrates an example of Trunk Optimization. If a call is placed from originating station A to a remote station B over route (1), and the call is forwarded to station C over route (3), the system can reconnect calls on the most economical route available--in this case, physically directed over route (2), allowing routes (1) and (3) to be used for additional calls.

Figure 102
Trunk Optimization example



It is important to point out the mechanism used to achieve trunk optimization. In the example on the following page the redirecting node will send a message to the originating node informing it of the redirection number and the redirection reason. If the originating node supports trunk optimization and has a first route choice member available to the destination PBX to terminate the call, it will send a message back accepting Trunk Optimization (first choice member is part of ESN route list entries in overlay 86). The redirecting node will send a message to Release the connection, the originating node will confirm the disconnect and will establish a new non redirected connection to the redirection number. This process may repeat itself for every redirection hop that a call takes (number of hops is software programmable).

However, if the originating node does not support Trunk Optimization, the call will then proceed without optimization on the current route. Trunk Optimization is programmable on a route basis using prompt TRO.

TRO does not apply to DID calls or calls originated by a non-ISDN TIE trunk. TRO is transparent to users for NCRD.

If the users have alpha-numeric display telephone sets;

- a** the "calling party" will see:
 - the number originally dialed,
 - the connected to number, and
 - the reason for redirection
- b** the "called party" will see:
 - the originating number,
 - the originally called number, and
 - the reason for redirection

Package Requirements

This option requires the following feature packages:

Feature	Package Number
ISDN	145
ISL	147
NTWK	148
PRI2	154

Note: The above scenarios require that an ISDN Primary Rate Interface (PRI) or ISDN Signaling Link (ISL) network is in place between the Meridian 1 switches.

Trunk to Trunk Connection

The Trunk to Trunk Connection feature introduces the following capabilities: transfer on ringing of external trunk across the network, transfer of one supervised outgoing external trunk to another, conference of external trunks and outgoing trunk to trunk charging. These capabilities are available on an analog (500/2500 type) set, Meridian 1 proprietary set or an Attendant Console.

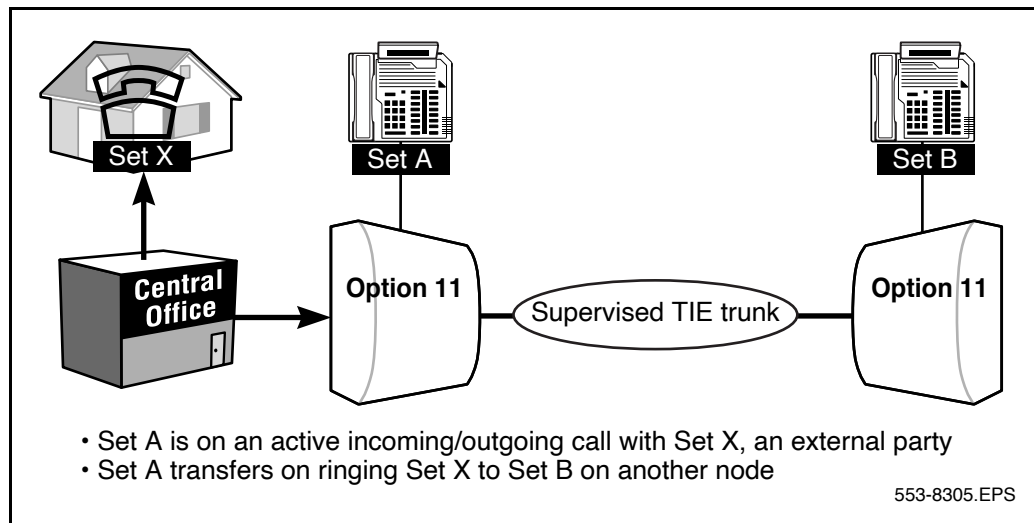
Transfer on Ringing of External Trunk over Network

Allows the transfer on ringing of an established external trunk call over a supervised analog network TIE trunk. If the called party does not answer within a specified time, the call will slow answer recall to the attendant on the transferring node. This capability ensures that available network resources are not occupied indefinitely.

Transfer of External Trunks

Allows the transfer of one outgoing external trunk to another trunk provided both calls are answered and both trunks have answer supervision.

Figure 103
Transfer on ringing of external call



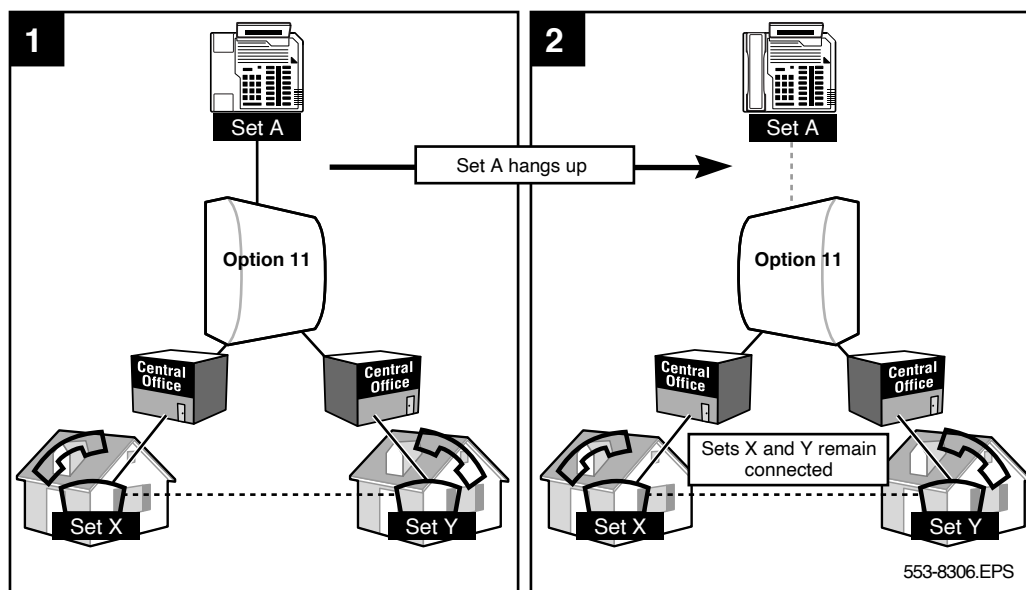
As illustrated in Figure 103, Set A is on an incoming/outgoing call with Set X, an external trunk. Set A initiates a call transfer of Set X to Set B. With the Trunk to Trunk Connection feature, Set A can transfer on ringing without waiting for Set B to answer. If Set B does not answer the transferred call, the external trunk slow answer recalls to the attendant on the transferring node.

Conference of External Trunks

Allows external trunks to remain established in a conference call in circumstances when all external trunks involved in the call offer disconnect supervision.

Figure 104 illustrates the Conference of External Trunk capabilities of this feature. Set A is on an established conference call with two or more external trunks, Set X and Set Y. When Set A disconnects during the conference, Set X and Set Y continue in the established call.

Figure 104
Conference of external trunks



Outgoing Trunk to Trunk Charging

Ensures that outstanding charging information, relevant to both outgoing calls, is contained in relevant Call Detail Recording records.

Operating Parameters

Slow answer recall occurs when an external trunk is transferred on ringing across an answer supervised network TIE trunk to a set that does not answer. However, the resulting recall will be to an attendant on the transferring node and not to the original set which transferred the call.

When transferring one outgoing trunk to another, it is required that the two external calls involved are both answered prior to completing the transfer. Both external trunks involved must have both answer and disconnect supervision.

When the last internal party disconnects from a conference call, involving 2 or more external trunks, all external trunks must have disconnect supervision for the call to remain established. If any one of the remaining external trunks does not have disconnect supervision, all external trunks will be dropped.

No change is made to existing VNS operation.

Feature interactions

Busy Tone Detection for Japan

Busy Tone Detection for Japan does not impact Trunk to Trunk Connection. However, which ever occurs first prevails.

Call Transfer

To transfer an external trunk on ringing across a supervised analog network TIE trunk, the external trunk and internal TIE line must have both answer and disconnect supervision, and the external call must be established. To transfer one outgoing external trunk to another, both external trunks must have answer and disconnect supervision, and both external calls must be established.

Centralized Answering Position

The Option 11C system may not have an actual attendant console. Instead, the Option 11C will use Centralized Answering Position (CAP). The CAP Directory Number (DN) is the customer Night DN. Since no attendant is configured, the customer is viewed to be in Night Service and any calls for the attendant are directed to the CAP. Slow Answer Recall may be presented to a CAP when no attendant console is configured for the customer.

Conference

Trunk to Trunk Connection allows external trunks to remain established in a call, provided that all external trunks involved have disconnect supervision. With respect to charging costs associated with a conference call, once the last set involved in the conference call disconnects, a search is made of all remaining trunks in the call to determine which call is established in the call for the longest period of time. This trunk is the chargeable Terminal Number (TN). This process is repeated to find the next chargeable TN.

Multi-Party Operations - Ringing No Answer

In a standalone environment, the RGNA prompt in the Customer Data Block will be used when an external trunk is transferred on ringing and the called party does not answer. In a network environment, the RTIM timer value in the Customer Data Block will be used for slow answer recall.

Message Registration

The last party releasing the call collects the total value of outstanding Periodic Pulse Metering (PPM) generated on outgoing trunks. If the last party is an internal set, the outstanding PPM is stored against the meter of the set. If the last party is an internal TIE trunk, the outstanding PPM is stored against the meter associated with the internal TIE trunk access code. If the last party is an outgoing external trunk, the outstanding PPM is stored against the meter associated with the external trunk access code.

Night Service

If an attendant is placed in Night Service, calls to the attendant are directed to a station with the Night DN. Recalls are not directed to the Night DN. Recalls are put in the attendant call waiting queue when in Night Service.

Night Service Enhancement

Recalls made while the attendant is in Night Service are routed to the Night DN, if the original call is an external call. In such a case, the destination party is disconnected, the internal network trunk is released and the original extended call is presented to the Night DN. If the original call is internal, recalls are put in the attendant call waiting queue when in Night Service.

Trunk Barring

Trunk Group Access Restriction

Trunk Barring and Trunk Group Access Restriction takes precedence over the Trunk to Trunk Connection feature.

Feature packaging

Trunk to Trunk Connection is included in base X11 system software.

Note: DID to TIE (DTOT) for Japan package 176 must be restricted to enable this feature.

Feature operation

There are no operating procedures specified for this feature.

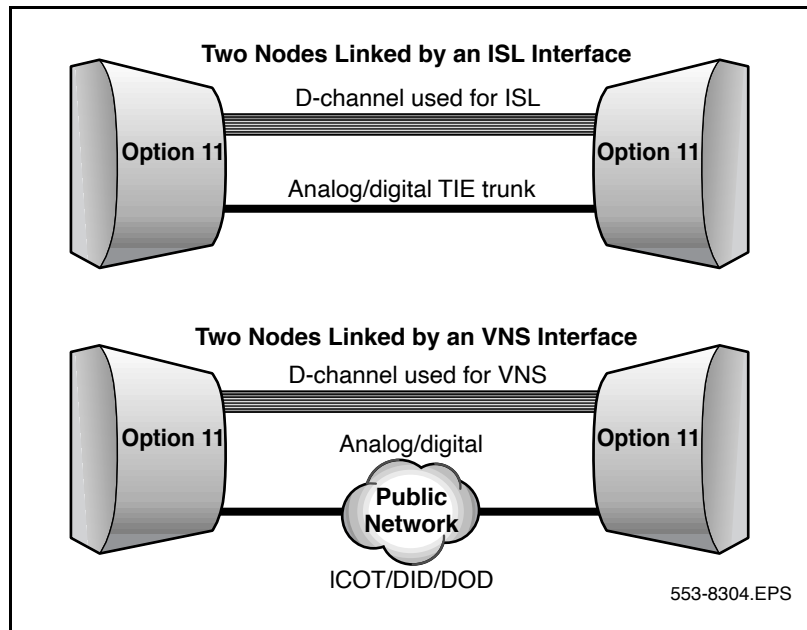
Virtual Network Services

The Virtual Network Services (VNS) feature offers a means to use Meridian 1 Integrated Services Digital Network (ISDN) networking features when two Meridian 1 systems are linked by a D-channel and an available Public Switched Telephone Network (PSTN) trunk exists. The PSTN trunks in this configuration serve as the B-channel for the call's duration.

VNS is an enhancement of the ISDN Signaling Link (ISL) interface. The enhancement allows the voice and the signaling of a call to take different physical paths. In the case of an ISL call, the D-channel is used for signaling, whereas the voice/data uses a TIE trunk (Meridian 1 to Meridian 1 connection only). The VNS feature extends the number of trunk types supported for the voice path by including the possibility of using different public trunks (e.g., COT, DID, DOD).

Figure 105 depicts the difference between two nodes linked by an ISL interface and two nodes linked using VNS.

Figure 105
VNS Interface/ISL Interface Comparison



Similar to the case of an ISL interface, the D-channel used for VNS can be shared with the D-channel of a Primary Rate Interface (PRI) that links two Meridian 1 switches, or it can use a D-channel that communicates with the far end through a pair of dedicated line modems, dial-up modems, or Meridian 1 data adapters.

When the D-channel used by VNS is on a PRI, the D-channel mode is defined as `USR = SHAV`, in LD 17, meaning that VNS, ISL and ISDN PRI calls can share this D-channel. In the other situation, the D-channel is defined as (`USR = VNS` in LD 17) which allows ISL, and VNS calls to use this D-channel for signaling.

The following types of trunks and connectivities can be used for VNS calls:

- PRI2 trunks with:
AXE10, NUMERIS, SwissNet 2, 1TR6, and Meridian Customer Defined Network (MCDN) connectivity.
- Analog trunks with:
disconnect supervision.
- 2.0 Mbit Digital Trunk Interface (DTI2) trunks with:
Multifrequency Compelled Signaling (MFC), Digitone Signaling (DTN), or Dial Pulse Signaling (DIP) connectivity.
- EuroISDN and QSIG trunks with:
PRI and Basic Rate Interface (BRI).
- Digital Private Network Signalling System No.1 (DPNSS1) or Digital Access Signalling System No.2 (DASS2) trunks.

All of these trunks are being referenced throughout this document as the VNS Bearers, or simply the Bearer trunks.

Although the VNS feature can use Central Office Trunks (COT), Direct Outward Dialing (DOD), and Direct Inward Dialing (DID) trunks to link two Meridian 1 nodes, VNS calls are presented, handled, and interact with other features as though private ISDN TIE trunks are used for that purpose. The exceptions that apply are stated in the “Feature interactions” section.

Note that TIE trunks can also be used with VNS. When this is the case, the connection does not necessarily need to be a direct Meridian 1 to Meridian 1 connection (e.g., when QSIG is used).

Speechpath availability

When extending a call to a Remote node, the attendant relies on the tones and lamp states to determine if it is possible to extend a call, camp-on a busy extension, break-in to a conversation, etc.

When using a VNS Bearer trunk, the tones can be those of the PSTN (especially for “intelligent” trunks such as ISDN connectivity). For example, when extending a call to a busy set over an ISDN PSTN Bearer, the attendant hears busy tone (from the PSTN) instead of silence.

If Attendant Console operation has to remain the same with VNS, an answer signal has to be sent right away to the PSTN for attendant extended calls, whatever the state of the terminating party. This allows the attendant to hear

the tones coming from the terminating Meridian 1 node, instead of tones from the PSTN.

The prompt VRAT exists for the incoming Bearer route (LD 16) to allow or disallow automatic answering of attendant extended calls. The Network Attendant Service (NAS) feature must be configured to provide this functionality.

When a route is configured with VRAT = YES, tones are similar to those provided on a private Meridian 1 ISDN PRI network; however, some unanswered calls may be charged. If a route is configured with the default value (VRAT = NO), tones heard when extending a call from the attendant might be different than the ones provided over a private Meridian 1 ISDN network. Users are not charged for unanswered calls.

PSTN clearing of unanswered calls

In some countries, a PSTN call left unanswered is disconnected after a few minutes, in order to reduce the usage of equipment that has not been paid for. This creates a major difference between private networks using TIE lines and private networks using VNS. For example, with VNS a call left in a camp-on state would be cleared by the public network after several minutes.

The VNS Set Speechpath (VSS) timer of the incoming Bearer route (LD 16) can be configured to avoid this situation. It defines the interval after which an answer signal is sent from the terminating Meridian 1 node to the Public Exchange/Central Office.

The configuration of the VSS timer has an impact on the cost of the calls, because the public network starts charging upon the reception of the answer signal.

Numbering Plan

A requirement of VNS is that customers have to configure VNS DN (VDNs) as part of their numbering plan. The VNS VDN expansion feature sets a new limit on the number of VDNs at 4000. VDNs are configured on Overlay 79.

The VDNs are defined in sets of blocks. Each block may contain any number of contiguous VDNs, as long as the total number does not exceed 4000.

Within a block, it is possible to define a range of VDNs as well as individual VDNs. For example, a customer may define the following VDNs (in this example, the customer is defining a total of 164 VDNs.)

7200-7299
7320-7325
7355
7400-7455
7676

A customer may remove a block of contiguous VDNs, as long as there is no VNS call using a VDN in the block. To remove a block of contiguous VDNs, the block has to be first disabled in Overlay 79 (by entering DIS against the REQ prompt), to prevent new VNS calls from using any of the VDNs in the block. Once the block has been disabled, the system administrator has to wait until all the VNS calls, using any of the VDNs in the block, are cleared.

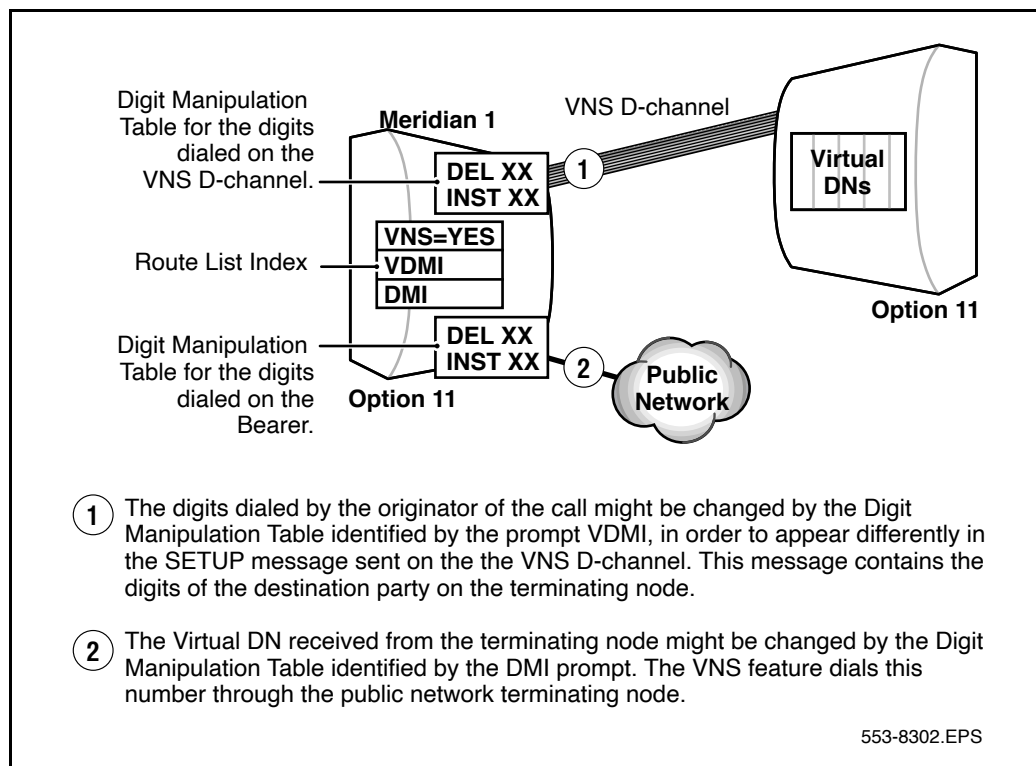
The system administrator can determine the number of VDNs used by VNS calls, by printing VNS information using the PRT command in response to the REQ prompt in Overlay 79 (note that the block of VDNs must be first disabled.) Once all of the VDNs are free (unused), the block of VDNs may be removed by using the OUT command.

An Electronic Switched Network (ESN) call is identified as being VNS once a valid entry configured when the prompt VNS = YES has been found in a Route List Block (LD 86) used for the digits dialed. A SETUP message is sent to the terminating node on the VNS D-channel. Manipulation of the digits included in the SETUP message is performed, if needed, using the digits dialed and according to the content of the Digit Manipulation Table referenced with the prompt VDMI.

On reception of the SETUP request, the terminating node sends one of its Virtual DNs back to the originating node. The terminating node expects to receive a call for that Directory Number (VDN) from one of its trunks linked to the public network. The originating node, upon reception of the VDN,

seizes a trunk and dials a number consisting of the VNS Virtual DN affected by the manipulations defined in the Digit Manipulation Table referenced by the prompt DMI. These manipulations are performed in order to dial the Virtual DN through the public network to the destination node.

Figure 106
Digit Manipulation



Failure of VNS D-channel

If the D-channel is not operable, the VNS route list entry is chosen, and then this entry is bypassed. In a network VNS environment, calls will remain queued at the source node.

If the VNS D-channel fails when a call is established, the call remains established. This is, however, without the ability to use the networking features normally supported by the VNS feature.

Failure of Bearer

Failure of the VNS Bearer trunk results in the VNS call being cleared.

Operating parameters

No new hardware is required for the VNS feature.

VNS requires a contiguous block of existing unused DN's (virtual DN's) at both the originating and the terminating nodes, which are a subset of the DID range of the customer. VNS virtual DN's can be up to seven digits long. A maximum of 100 virtual DN's can be assigned per customer.

VNS D-channel

VNS requires an established D-channel between the originating and terminating nodes for signaling. No intermediate nodes are permitted.

A VNS D-channel belongs to only one customer and cannot be shared by other customers for VNS calls.

VNS does not support Backup D-channels for VNS signaling.

D-channels provided by Basic Rate Interface (BRI) are not supported as VNS D-channels.

VNS reverts back to conventional signaling if the VNS D-channel fails.

VNS Bearer Trunks

Disconnect supervision is mandatory for all trunks, while answer supervision is not. In cases when it is not available, VNS will assume that the Bearer is answered as soon as the "end-of-dialing" timer has expired.

The following trunks are not supported as VNS Bearer trunks:

- Autoterminate trunks;
- Analog Private Networking Signalling System (APNSS); and
- Basic Rate Interface trunks (used for NUMERIS, 1TR6 or MCDN).

The following signaling systems are not supported for VNS:

- Japan D70; and
- Multifrequency Extended Signaling for Socotel (MFE).

Any Incoming Digit Conversion on the incoming VNS Bearer trunk causes a VNS call to fail.

Data calls are supported only if the Bearer supports data calls.

Maximum number of VNS calls

Different elements control the maximum number of simultaneous VNS calls that can be processed by a Meridian 1 system. The number depends on the resources allocated for the VNS feature itself, the resources to be used for the voice path (Bearer trunks) and the resources available for VNS signaling over the D-channel.

For each customer

It is not possible to perform more VNS calls (incoming and outgoing) than the number of Virtual DNs defined for the customer in LD 79.

The number of VNS calls is limited by the number of available Bearer trunks. When all Bearers defined to be usable with VNS are busy, no other VNS calls can be made.

For each D-channel of the customer

The maximum number of VNS calls allowed to use a D-channel is limited by the value defined to the D-channel by the VNSM prompt in LD 17.

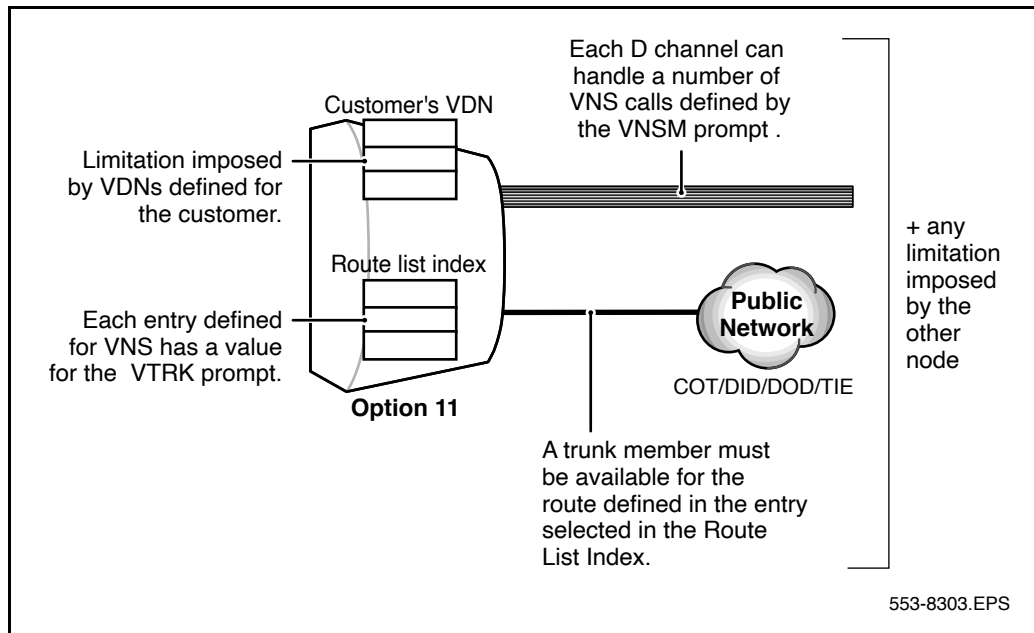
For each entry in a Route List Block

The maximum number of VNS calls that can be performed on a route listed in an entry of a Route List Block in LD 86 is limited by the value entered in response to the VTRK prompt.

On a network level

In the case of a non-symmetrical configuration between originating and terminating VNS nodes, the call limitation imposed by the more restrictive node applies.

Figure 107
VNS Call Limitations



Feature interactions

The following features are supported over VNS routes:

- Attendant Barge-In
- Attendant Break-In
- Attendant Recall with Splitting
- Basic Authorization Code
- Basic Rate Interface/Primary Rate Interface Interworking
- Calling Party Number
- Call Party Name Display
- Call Selection (ICI + loop)

- Console Digit Display
- Control of Trunk Group Access
- Digit Display
- Display of Calling Party Denied
- Electronic Lock Network Wide/Electronic Lock on Private Lines
- Network Attendant Service
- Network Call Redirection
- Network Class of Service
- Network Flexible Feature Codes
- Network Message Center
- Network Message Service to Meridian Mail
- Network Ring Again
- Network Ring Again No Answer
- Originator Routing Calls
- Radio Paging
- Trunk Optimization Before Answer

In addition, VNS has interactions with the following features:

Access Restriction

When a user has trunk access barred (i.e., via Trunk Group Access Restriction (TGAR), New Flexible Call Restriction (NFCR), or Toll Denied Class of Service), the user will still be able to make a VNS call.

If a customer wants users to be toll restricted even for VNS calls, the Minimum Facility Restriction Level (FRL) in the Route List Block (i.e., LD 86) can be used.

Analog Private Networking Signalling System (APNSS)

APNSS trunks cannot function as VNS Bearer trunks.

Attendant Call

When an attendant extends a call over a VNS network, it is not possible to release the call before the status of the destination is known. This applies regardless of Network Attendant Service (NAS) configuration.

Autoterminate

Autotermination on VNS Bearer trunks is not supported, because a VNS Virtual DN is expected on a VNS Bearer exactly the way it was passed back on the VNS D-channel.

Call Detail Recording (CDR)

Call Detail Recording is supported on VNS routes with the exception of the 911 CDR Improvement and Meridian 911 ANI in CDR features. The CDR output is controlled by the Bearer trunk (i.e., CDR output defines the details of the call made on the Bearer, and not the VNS D-channel). The Calling Line Identity will be output in the CDR record for incoming calls.

Call Pickup Network- Wide

The Call Pickup Network Wide feature will not work in conjunction with the Virtual Network Services feature.

Calling Line Identification

Calling Line Identification is supported on VNS routes. All display information is taken from messages on the VNS D-channel. The VNS Bearer does not drive the display of such information.

Charge Display at End of Call

In the case of VNS calls, no charge information is displayed.

Data Calls

Data calls are supported on DPNSS1 or DASS2 VNS Bearer trunks if the DPNSS1 or DASS2 VNS Bearer trunks are configured to support data calls. Similarly, data calls are supported on DPNSS1 or DASS2 Bearer trunks in VNS to DPNSS1/DASS2 gateways, if the DPNSS1 or DASS2 VNS Bearer trunks are configured to support data calls.

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

Virtual Network Services via CIS DTI2 is not supported.

Direct Inward Dialing (DID) Treatment

Since VNS is handling calls as though they were made using ISDN TIE trunks even when VNS DID Bearer trunks are used, the VNS feature does not adhere to the special treatment given to regular DID trunks.

Distinctive Ringing

Except in the case of Network Distinctive Ringing, the Distinctive Ringing feature is not supported by VNS. An incoming call using VNS on a Bearer trunk defined with the prompt DRNG = YES will ignore this value and will perform the same treatment as though the value was DRNG = NO (i.e., no distinctive ringing provided).

DPNSS1 Attendant Call Offer

DPNSS1 Attendant Call Offer is not supported over VNS Bearer trunks (DPNSS1 Attendant Call Offer allows an attendant-extended call, routed over a DPNSS1 trunk, to be camped-on to a remote busy extension.) Standard ISDN Camp-on may be provided instead, if NAS is configured over the VNS Bearer trunks.

DPNSS1 Attendant Timed Reminder Recall and Attendant Three-Party Service

DPNSS1 Attendant Timed Reminder Recall and Attendant Three-Party Service are not supported over VNS Bearer trunks. If NAS is configured over the VNS Bearer trunks, NAS call extension and Attendant Recall will be offered instead.

DPNSS1 Call Back When Free and Call Back When Next Used

DPNSS1 Call Back When Free and Call Back When Next Used are not supported over VNS Bearer trunks. Network Ring Again or Network Ring Again on No Answer may be provided instead, if Network Ring Again or Network Ring Again on No Answer are configured over the VNS Bearer trunks.

DPNSS1 Diversion

DPNSS1 Diversion is not supported over VNS Bearer trunks. Network Call Redirection and Trunk Route Optimization can be provided instead, if configured over the VNS D-Channel.

DPNSS1 Extension Three-Party Service

DPNSS1 Extension Three-Party Service is not supported over VNS Bearer trunks. Network Call Redirection and Trunk Route Optimization can be provided instead, if configured over the VNS D-Channels.

DPNSS1 Loop Avoidance

DPNSS1 Loop Avoidance is not supported over VNS Bearer trunks (DPNSS1 Loop Avoidance prevents a call from being looped through a DPNSS1 network by placing a limit on the number of channels that a call can use.) The ISDN Call Connection Limitation is provided, if it is configured over the VNS D-Channel.

DPNSS1 Route Optimization

DPNSS1 Route Optimization is not supported over VNS Bearer trunks.

DPNSS1 Route Optimization/ISDN Trunk Anti-Tromboning Interworking

ISDN Trunk Anti-Tromboning may be applied to the VNS part of the call, if configured on the VNS D-Channel.

DPNSS1 Route Optimisation/MCDN Trunk Anti-Tromboning Interworking

The Route Optimisation/Trunk Anti-Tromboning Interworking feature is not supported over VNS trunks, since VNS uses only MCDN signaling. DPNSS1 is not supported.

DPNSS1 Step Back On Congestion

DPNSS1 Step Back On Congestion handles high traffic situations when congestion is encountered by DPNSS1 trunks. The following scenarios apply for interworking with VNS.

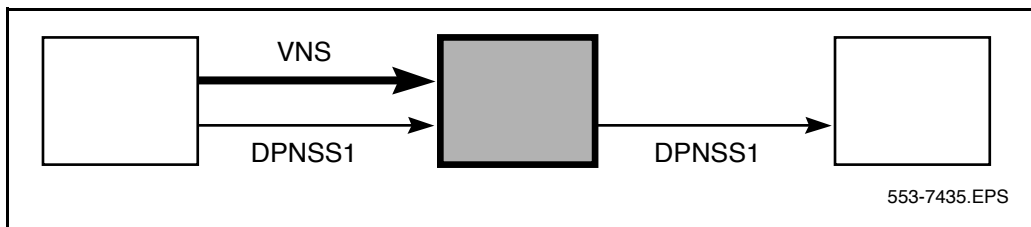
Homogeneous Networks

DPNSS1 Step Back On Congestion is supported over VNS Bearer trunks, if all the transit nodes within the DPNSS1 network used for VNS are configured accordingly:

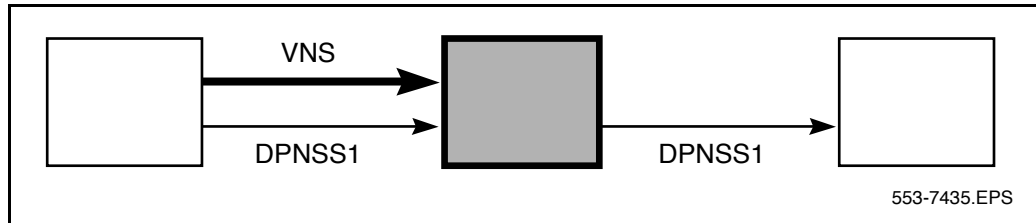
- In LD 86, if the SBOC (Step Back On Congestion) prompt is set to NRR (No Reroute) or RRO (Reroute Originator), then it would be sufficient that the VNS originating node be configured with either RRO (Reroute Originator) or RRA (Reroute All).
- In LD 86, if the SBOC (Step Back On Congestion) prompt is set to RRA (Reroute All) for a transit node, then the different alternative routes at this node must be configured with VNS and must be configured as VNS Bearers.

Hybrid Networks

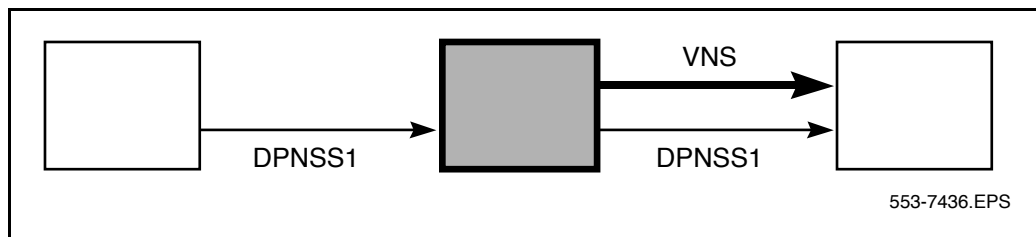
MCDN/VNS with DPNSS1 node



- If a congestion is encountered inside the VNS portion of the path, the node behaves as an MCDN/MCDN tandem. The ISDN Drop Back Busy (IDBB) and ISDN Off-Hook Queuing (IOHQ) are transmitted, so that they may be applied further along the VNS portion of the path, or at the tandem node.
- If a congestion is encountered within the DPNSS1 network, the VNS portion of the call is cleared and the disconnection is propagated back to the originating side of the MCDN path. Neither Drop Back Busy nor Off-Hook Queuing is activated at the tandem node, even if IDBB or IOHQ are activated.

VNS with DPNSS1/DPNSS1 node

This scenario is considered as an MCDN/DPNSS1 gateway. The functionality is the same as for the Step Back on Congestion feature, as documented in the *DPNSS1 Product Information Guide* (553-3921-100.)

DPNSS1/VNS with DPNSS1 node

- If a congestion is encountered inside the VNS portion of the path, the VNS portion of the call is cleared and the disconnection is propagated back to the originating DPNSS1 side. The Step Back on Congestion feature is invoked, if it is configured.
- If a congestion is encountered the within the DPNSS1 portion of the path, with the DPNSS1 trunk being used as a VNS Bearer, the VNS portion of the call is cleared and a normal disconnection is propagated back to the originating DPNSS1 side. The Step Back on Congestion feature is not invoked, even if it is configured.

DPNSS1 Executive Intrusion

DPNSS1 Extension Three-Party Service is not supported over VNS Bearer trunks. Attendant Break-in may be provided instead, if NAS is configured over the VNS Bearer trunks.

Stand-alone Meridian Mail

Stand-alone Meridian Mail is not supported over VNS Bearer trunks. A mailbox user may access Meridian Mail, if the ISDN Network Message Services is configured.

DPNSS1 Enhancements for ISDN Interworking

The Generic X11 Release 20B software introduced enhancements to allow DPNSS1 to interwork with QSIG and EuroISDN. At an ISDN gateway, ISDN information may be carried into some DPNSS1 messages, if DPNSS_189I package 284 is equipped.

DPNSS1/DASS2 to ISDN PRI Gateway

A VNS call over a DPNSS1 or DASS2 Bearer trunk of a DPNSS1/DASS2 to ISDN PRI Gateway acts as the ISDN leg of the Gateway.

Electronic Switched Network (ESN)

The following ESN features are supported on VNS routes:

- ESN Basic Automatic Route Selection
- ESN Coordinated Dialing Plan Routing Enhancement
- ESN Coordinated Dialing Plan
- ESN (999 Loc.)
- ESN Flexible Numbering Plan
- ESN Free Calling Area Screening
- ESN Incoming Trunk Group Exclusion
- ESN Network Authorization Code
- ESN Network Automatic Route Selection
- ESN Network Routing Controls (Time-of-day Scheduling)
- ESN Network Speed Call

- ESN Tone Detection, and
- ESN Off-Hook Queuing.

EuroISDN Continuation

A EuroISDN link can be used as a B-Channel for the Virtual Network Services feature.

EuroISDN Master Mode

Virtual Network Services is supported on a EuroISDN Master Mode connectivity, meaning that a EuroISDN Master Mode trunk can be used as a VNS bearer trunk.

Incoming Digit Conversion

Since VNS Virtual DNs must be received on the incoming VNS Bearer (DID) trunks, as they are defined at the terminating node, the VNS feature is not supported for routes that perform digit conversion on the digits used for VNS Virtual DNs.

ISDN Call Connection Limitation (ICCL) Checks

The VNS trunk from the external network looks exactly like an ISDN TIE trunk and accordingly is subject to ICCL checks and call restrictions applied thereon.

ISDN PRI/BRI Meridian 1 to Asia Pacific Connectivity

It is not possible to configure an Asia Pacific D-channel as a VNS D-channel. However, the voice connection through the Public Exchange of a VNS call may use a PRI/BRI COT or DID as a virtual TIE trunk.

ISDN QSIG Basic Call

A QSIG link can be used as a B-channel for the Virtual Network Service (VNS) over a private network. All VNS services are supported as normal. QSIG is only used as a speech bearer.

Intercept Treatment

Intercept treatment applied to VNS calls is configured as for TIE trunks.

Network ACD

Network ACD (NACD) is supported by VNS. When a call queued at a source node receives notification that an agent is free and reserved at the target node, a VNS call is made to the target node. At that time, the call is removed from

the queue at the source node. The call will continue to receive ringback (or Recorded Announcement (RAN) if configured) while it is waiting for the VNS call to be completed. When an alerting message is received from the target node, the ringback or RAN is removed and the speechpath with the Bearer is connected. The user will thus hear ringback tone (given either by the PSTN or the Meridian 1 at the target node, depending on the Bearer type), until the agent answers the call.

If for some reason the ALERTING message is not received, the user is requeued (at the top of the queue at the source node). Since ringback or RAN were not removed in the first place, the user is unaware that an attempt to reach a remote agent has been made and has failed.

Since PSTN Bearer trunk call establishment may take longer than PRI2 TIE trunk call establishment, it is necessary to allow more time for the call to arrive at the agent's set than provided by the regular NACD feature. The Agent Reservation Timer should be configured to accommodate a longer interval.

If an agent is reserved at the target node, but the call fails to reach that node (e.g., all trunks are busy), the agent will stay reserved until NACD Target Reservation Timer (RAGT) expires (the target node has no indication that the call has encountered a busy condition). This is more critical when NACD is used with VNS, because (RAGT) must be set to a greater value.

NACD allows up to 20 remote target nodes to be defined in the routing tables. Since the maximum number of VNS Virtual DNs per customer is 100, it is not realistic to configure 20 target nodes using VNS routing: this would only allow five Virtual DNs per target node and thus a maximum of five simultaneous calls.

Network Call Party Name Display

Network Call Party Name Display is supported on VNS routes. All display information is taken from messages on the VNS D-channel. The VNS Bearer does not drive the display of such information.

Network Distinctive Ringing

For an incoming DID call (without VNS) that tandems to another node using VNS, the feature will be supported as it would have been if an ISDN TIE trunk had been used for the VNS call.

Overlap Sending and Overlap Receiving

Call establishment over the VNS D-channel will not use overlap sending/receiving even if configured. Enbloc sending will always be used on the VNS D-channel. Overlap sending/receiving will be supported on the Bearer trunks if that capability already exists.

Periodic Pulse Metering

Periodic Pulse Metering is supported on the VNS Bearer trunks only.

R2 Multifrequency Compelled Signaling to DPNSS1 Gateway

If the call on the DPNSS1 (or R2MFC) trunk is tandeming to the R2MFC or (DPNSS1) trunk on a Virtual Network Services (VNS) call, the R2MFC to DPNSS1 Gateway feature does not apply. If a DPNSS1/R2MFC tandem is encountered during the routing of a VNS call, the R2MFC to DPNSS1 Gateway feature applies. The following figure illustrates how the R2MFC-DPNSS1 gateway may apply to a VNS call.

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

Virtual Network Services is not supported on CIS trunks.

Trunk Barring

Three cases apply for Trunk Barring:

- 1 When the second trunk involved in the call is used by VNS, no trunk barring is applied regardless of the configuration of the first trunk. The call is always allowed to get through.

Note: This implementation completely overrides the Trunk Barring feature.
- 2 When the first trunk involved in the call uses VNS, and the second one is not used by VNS, trunk barring is performed according to the content of the default Access Restriction Table (ART) for the TIE trunk.
- 3 When neither the first trunk nor the second trunk is involved in a VNS call, the usual rules for Trunk Barring apply.

Vacant Number Routing

Calls rerouted by the Vacant Number Routing feature are not allowed to have digit manipulation performed on the DN dialed. Thus, for non VNS calls, no Digit Manipulation Index (DMI) can be associated in the Route List Index (RLI) used by Vacant Number Routing. For VNS calls (in LD 86, the entries on this RLI are defined with the prompt VNS = YES), it is not allowed to associate any manipulation on the digits included in the messages exchanged over the VNS D-channel. In these cases, the prompt VMDI is not displayed, but the prompt DMI is still displayed.

Advice of Charge for Euro ISDN

For a Virtual Network Service (VNS) simple call connectivity, received charging information is stored against the Message Registration meter of the calling party's set.

On an outgoing trunk call on VNS connectivity, charging information is processed based on the Trunk to Trunk Connection feature.

Feature packaging

The Virtual Network Services (VNS) package is 183.

The following software packages are also required for VNS:

- Network Alternate Route Selection (NARS) package 58, which is dependent on Basic Routing (BRTE) package 14 and Network Class of Service (NCOS) package 32;
- Integrated Services Digital Network (ISDN) package 145;
- Integrated Services Digital Network Supplementary (ISDNS) package 161;
- ISDN Signaling Link (ISL) package 147;
- Advanced Network Services (NTWK) package 148; and
- PRI2 package 154 is required if VNS uses the D-channel of a PRI2 trunk for its signaling (USR = SHAV).

Feature operation

There are no operating procedures specified for this feature.

Virtual Network Services Virtual Directory Number Expansion

Virtual Network Services (VNS) provides ISDN features to customers when no ISDN Primary Rate Interface (PRI) or ISDN Signaling Link (ISL) Bearer Channels are available between two Meridian 1 switches (Please refer to the VNS feature description module in this NTP for detailed information on VNS).

A requirement of VNS is that customers have to configure VNS DN (VDN) as part of their numbering plan. The VNS VDN expansion feature sets a new limit on the number of VDNs at 4000. VDNs are configured on Overlay 79.

The VDNs are defined in sets of blocks. Each block may contain any number of contiguous VDNs, as long as the total number does not exceed 4000. Within a block, it is possible to define a range of VDNs as well as individual VDNs. For example, a customer may define the following VDNs (In this example, the customer is defining a total of 164 VDNs.)

```
7200-7299
7320-7325
7355
7400-7455
7676
```

A customer may remove a block of contiguous VDNs, as long as there is no VNS call using a VDN in the block. To remove a block of contiguous VDNs, the block has to be first disabled in Overlay 79 (by entering DIS against the REQ prompt), to prevent new VNS calls from using any of the VDNs in the block. Once the block has been disabled, the system administrator has to wait until all the VNS calls, using any of the VDNs in the block, are cleared.

After disabling the block, the system administrator can determine the number of VDNs used by VNS calls by printing VNS information (by using the PRT command in response to REQ in Overlay 79). Once all of the VDNs are free, the block may be removed by using the OUT command.

Operating parameters

There are no operating parameters specified for this feature.

Feature interactions

There are no feature interactions specified for this feature.

Feature packaging

The VNS VDN expansion feature requires the following packages:

- Virtual Network Services (VNS) package 183;

The following software packages are also required:

- Network Alternate Route Selection (NARS) package 58, dependent on:
 - Basic Routing (BRTE) package 14, and
 - Network Class of Service (NCOS) package 32;
- Integrated Services Digital Network (ISDN) package 145;
- Integrated Services Digital Network Supplementary (ISDNS) package 161;
- ISDN Signaling Link (ISL) package 147;
- Advanced Network Services (NTWK) package 148; and
- 2.0 Mbit PRI (PRI2) package 154 is required if VNS uses the D-Channel of a PRI or PRI2 trunk for its signaling (USR = SHAV).

Feature operation

There are no operating procedures specified for this feature.

X08/X11 Gateway

Description

This feature allows the SL-1 using X11 software with Supplementary features to act as a gateway between other SL-1s, running either X08 or X11 software, and various central-office (CO) trunks. This allows nodes using X08 and X11 generic software to be mixed in a single private network.

The Gateway feature allows bridging, or tandeming, between ISDN routes and either R2/MFC-signalled routes or L1-signalled routes.

Meridian 1 Feature Packages

The following table lists all Meridian 1 software feature packages sequentially by number, along with mnemonics, names and dependencies.

Feature Packages

Number	Mnemonic	Name	Dependencies
0	BASIC	Basic System Features	1
1	OPTF	Optional Features	0
2	CUST	Multi-Customer	
4	CDR	Call Detail Recording	5
5	CTY	Call Detail Recording, Teletype Terminal	4
6	CLNK	Call Detail Recording, Data Link	4, 5
7	RAN	Recorded Announcement	
8	TAD	Time and Date	
9	DNDI	Do Not Disturb Individual	16
10	EES	End to End Signaling	
11	INTR	Intercept Treatment	
12	ANI	Automatic Number Identification	13
13	ANIR	Automatic Number Identification, Route Selection	12
14	BRTE	Basic Routing	32
15	RPE1.5	1.5-Mbit Remote Peripheral Equipment	not with 165 (RPE2)

Number	Mnemonic	Name	Dependencies
16	DNDG	Do Not Disturb Group	9
17	MSB	Make Set Busy	
18	SS25	Special Service for 2500 Sets	73
19	DDSP	Digit Display	
20	ODAS	Office Data Administration System	
21	DI	Dial Intercom	
22	DISA	Direct Inward System Access	
23	CHG	Charge Account	52,24
24	CAB	Charge Account/Authorization Code Base	23,52
25	BAUT	Basic Authorization Code	23,24,52
26	CASM	Centralized Attendant Service, Main	not with 56 (AOP),159 (NAS)
27	CASR	Centralized Attendant Service, Remote	not with 56 (AOP),159 (NAS)
28	BQUE	Basic Queuing	
29	NTRF	Network Traffic	must have Network packages
32	NCOS	Network Class of Service	
33	CPRK	Call Park	
34	SSC	System Speed Call	32
35	IMS	Integrated Message Services	7,17,40,45,46,109
36	ROA	Recorded Overflow Announcement	7
37	NSIG	Network Signaling	32
38	MCBQ	Network Call Back Queuing	14,28,32,37.57/58/59,61
39	NSC	Network Speed Call	14,28,32,34, 57/58/59,61
40	BACD	Basic Automatic Call Distribution	
41	ACDB	Automatic Call Distribution, Package B	7,17,40,44,45,111
42	ACDC	Automatic Call Distribution, Package C	7,17,40,41,44,45,111
43	LMAN	Automatic Call Distribution, Load Management Reports	7,17,40,41,42,44,45,111
44	MUS	Music	7

Number	Mnemonic	Name	Dependencies
45	ACDA	Automatic Call Distribution, Package A	7,17,40
46	MWC	Message Waiting Center	40
47	AAB	Automatic Answerback	
48	GRP	Group Call	
49	NFCR	New Flexible Code Restriction	32
50	ACDD	Automatic Call Distribution, Package D	7,17,40,41,42,43,44,45,51,111
51	LNK	Automatic Call Distribution, Auxiliary Processor Link	7,17,40,41,42,43,44,45,50,111
52	FCA	Forced Charge Account	23,24
53	SR	Set Relocation	
54	AA	Attendant Administration	
55	HIST	History File	
56	AOP	Attendant Overflow Position	not with 26,27,159
57	BARS	Basic Alternate Route Selection	14,28,32,58/59,61
58	NARS	Network Alternate Route Selection	14,28,32,57/59,61
59	CDP	Coordinated Dialing Plan	14,28,32,57/58,61
60	PQUE	Network Priority Queuing	14,28,32,57/58/59,61
61	FCBQ	Flexible Call Back Queuing	14,28,32,57/58/59
62	OHQ	Off Hook Queuing	14,28,32,57/58/59,61
63	NAUT	Network Authorization Code	14,23,24,25,28,32, 52, 57/58/59,61
64	SNR	Stored Number Redial	
65	TDET	Tone Detector	
67	NXFR	Network Transfer	32,37
70	HOT	Enhanced Hotline	
71	DHLD	Deluxe Hold	
72	LSEL	Automatic Line Selection	
73	SS5	500 Set Dial Access to Features	18
74	DRNG	Distinctive Ringing	
75	PBXI	1.5-Mbit Digital Trunk Interface	

Number	Mnemonic	Name	Dependencies
76	DLDN	Departmental Listed Directory Number	
77	CSL	Command Status Link	
79	OOD	Optional Outpulsing Delay	
80	SCI	Station Category Indication	
81	CCOS	Controlled Class of Service	
82	RSDB	Resident Debug	
83	CDRQ	Call Detail Recording Queue Record	4,5
84	ATM	Automatic Trunk Maintenance	65
86	TENS	Multi-Tenant Service	
87	FTDS	Fast Tone and Digit Switch	
88	DSET	M2000 Digital Sets	
89	TSET	M3000 Digital Sets	88
90	LNR	Last Number Redial	
91	DLT2	M2317 Digital Sets	88
92	PXLT	Pretranslation	
93	SUPV	Supervisory Console	
95	CPND	Calling Party Name Display	88/89/91/140/170
96	SLST	Meridian SL-1 ST Package	only on ST/Option 21
97	JPN	Japan Central Office Trunks	
98	DNIS	Dialed Number Identification System	7,17,40,45
99	BGD	Background Terminal	
100	RMS	Room Status	81,99
101	MR	PPM/Message Registration	131
102	AWU	Automatic Wake Up	7,81,99
103	PMSI	Property Management System Interface	81,99,100/101/102 (if 102 then 7 is required)
104	OPAO	Outpulsing, asterisk (*) pound (#)	
107	MCT	Malicious Call Trace	4,5
108	ICDR	Internal CDR	4,5
109	APL	Auxiliary Processor Link	7,17,40,45

Number	Mnemonic	Name	Dependencies
110	TVS	Trunk Verification from Station	
111	TOF	Automatic Call Distribution, Time Overflow Queuing	7,17,40,41,44,45
113	IDC	Incoming Digit Conversion	49,32
114	AUXS	Automatic Call Distribution Package D, Auxiliary Security	7,17,40,41,42,43,44,45, 50,51,111
115	DCP	Direct Call Pick Up	
116	PAGT	Automatic Call Distribution, Priority Agent	7,17,40,45
117	CBC	Call By Call Service	145,154/147
118	CCDR	Calling Line ID in CDR	4,5,145,154/147
119	EMUS	Enhanced Music	7,44
120	PLDN	Group Hunt/DN Access to SCL	32,34,81, 131,139
121	SCMP	Station Camp-On	131
122	IDA	Integrated Digital Access	129
123	DPNSS	Digital Private Network Signaling System 1	122,148, 129, 14, 28, 32, 57/58/59, 61, 145, 154/147
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126	OPCB	Operator Call Back (China #1)	131,107/127/128 9 (if 107 then 4,5 are required)
127	BKI	Attendant Break-In/Trunk Offer	
128	MFC	Multifrequency Compelled Signaling	
129	DTI2	2 Mb Digital Trunk Interface	
131	SUPP	Supplementary Features	
132	TBAR	Trunk Barring	
133	ENS	Enhanced Night Service	
135	MFE	Multifrequency Signaling for Socotel	
136	JDMI	Japan Digital Multiplex Interface	
137	LSCM	Local Steering Code Modifications	
138	DTD	Dial Tone Detector	

Number	Mnemonic	Name	Dependencies
139	FFC	Flexible Feature Codes	81
140	DCON	M2250 Attendant Console	84
141	MPO	Multi-Party Operations	
143	ICP	Intercept Computer Interface	7,17,35,39,40,45,46,81, 109, 131,139
144	ABCD	16-Button Digitone/Multifrequency Telephone	
145	ISDN	Integrated Services Digital Network	
146	PRA	1.5 Primary Rate Access (ISDN)	75,145
147	ISL	ISDN Signaling Link	145
148	NTWK	Advanced ISDN Network Services	14,28,32,57/58/59,61,145,154 /147
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154	PRI2	2 Mb Primary Rate Interface	145
155	ACNT	Automatic Call Distribution, Activity Code	7,17,40,41,42,43,44,45,5051, 111,114
157	THF	Trunk Hook Flash (Centrex)	
159	NAS	Network Attendant Service	14,28,32,57/58/59,65, 61,145,147/154, not with 26,27,56
160	FNPN	Flexible Numbering Plan	14,28,32,57/58,/59,61
161	ISDNS	ISDN Supplementary Features	14,28,32,57/58/59,61,159 145, 147/154
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163	MINT	Message Intercept	125,74
164	LAPW	Limited Access to Overlays	
165	RPE2	2 Mb Remote Peripheral Equipment	not with 15 (RPE)
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170	ARIE	Aries Digital Sets	88
171	JTDS	Japan Tone and Digit Switch	74

Number	Mnemonic	Name	Dependencies
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174	AAA	Attendant Alternative Answer	not with 134 (AFNA)
175	NMS	Network Message Services	14,28,32,40,46,57/58/59, 61,145,148,154/147
176	DTOT	DID to Tie	Japan only
178	EOVF	ACD Enhanced Overflow	7,17,40,41,44,45,111
179	HVS	Meridian Hospitality Voice Services	7,10,17,35,40,45,46,77, 109,180
180	DKS	Digit Key Signaling	7,10,17,35,40,45,46,77, 109
181	SACP	Semi-Automatic Camp-On	
182	TFM	Trunk Failure Monitor	
183	VNS	Virtual Network Services	2,14,28,32,57/58/59,61,, 145147/154,148,159,161
184	OVLP	Overlap Signaling	(M1 to M1 and 1TR6CO)
185	EDRG	Executive Distinctive Ringing	14,28,32,57/58/59,61,74, 125,145,147/154,159,161
186	POVR	Priority Override/Forced Camp-On	139,141,81
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188	L1MF	X08 to X11 Gateway	14,28,32,128,145,154/147 161,57/58/59,61,159
190	UK	UK program	203,204,205
191	SECL	Series Call	
192	ORC	Originator Routing Control (Dropback Busy)	14,28,32,37,38,57/59,58,61, 62,145,147/154,159,161
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209	MLM	Meridian Link module	7,8,10,17,19,35,40,45,46,77,109,153,180
222	MSDL	Multipurpose Serial Data Link	
231	DNW	DPNSS1 Network Services	
252	KD3	Spanish KD3	129, 126, 131, BARS or NARS
259	CDRX	CDR Enhancement (Phase 1)	4, 5, 234
261	EURO	Euro ISDN	19, 145, 154, 184, 202
263	QSIG	ISDN QSIG Basic Call	19, 145, 154, 184

Meridian 1

Option 11C

2.0 Mb DTI/PRI Administration and
Maintenance guide
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Option 11C

2.0 Mb DTI/PRI Administration and Maintenance guide

Book two of two

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Chapter 3 — DPNSS1/DASS2 features description and implementation

Overview

Intelligent private digital networks predominantly use the DPNSS1 protocol that allows PBXs from different manufacturers to interwork. In a private network that utilizes analog trunks, the signaling system that allows ISDN interaction is APNSS. Both APNSS and DPNSS1 support the same user features. In the UK, ISDN access to the Public Network is provided using the DASS2 protocol.

This section provides detailed information on features that are available for the current release of DPNSS1 on Option 11.

For some supplementary features, subsections have been created that offer a functional description, detail any specific requirements or limitations, explain interactions with other features, list software package requirements, and outline implementation procedures. These will be detailed at the end of this chapter.

DPNSS1/DASS2 feature descriptions

Add On Conference

This service allows the add on of more parties to a 3 party conference set up with the Three Party Service feature.

Bearer Service Selection

During call set up, the A party can inform the network that further services will be required later in this call, e.g. change from voice to data.

Call Back Messaging

When a call cannot be established, the A party can leave an indication on the B parties instrument.

Call Back When Free

The A party can initiate a call back order when either the B party is busy or trunks in the network are congested.

Call Back When Next Used

If no answer is received from the B party in a call, this feature can be initiated. When B is next used, the procedure will be the same as the Call Back When Free feature.

Call Distribution

When a call is made to the Destination Address for this service, the call is routed to one controlling PBX in the network. Stations in any PBX in the network can form an answer group and the call will then be passed to an extension allocated to this group or queued and monitored.

Call Diversion

Calls may be diverted to anywhere in the Network under these situations:

- Immediate
- On busy
- On no reply
- Follow Me

Call Park

The originating party on making a call to a busy extension may request to be parked on the called party. The called party may receive a tone and act as required.

Call Waiting

An extension can select whether a call waiting indication is given under the call offer situation.

Centralized Operator

Operators in a network can be in one or more nodes providing that an operator group is not spread between nodes.

Charge Reporting

Details of call charging can be sent through the network to the originator.

Controlled Diversion

When a call is made to a diverted extension, the A party can select the action to take:

- proceed with the diversion
- clear the call
- bypass immediate diversion if the called extension is free
- invoke another supplementary feature to a busy extension
- continue to ring called extension if diversion no reply is invoked

Data Call

A simple data call between two compatible terminals running in the same mode and at the same speed.

Do Not Disturb

Calls to an extension that have invoked do not disturb will receive busy unless the A party has requested override.

Executive Intrusion

The A party can intrude on a busy B party under control of intrusion and protection levels of all parties involved in call. Levels are 0 to 3 and the A party intrusion level must be higher than the protection levels.

Extension Status

The status of an extension can be requested without actually making a call. This would normally be used by the Operator.

Forced Release

Following intrusion to a busy extension a three party conference is set up, and the initiator of the intrusion may force release the unwanted party in the conference.

Hold

Either party in a call can place the other on hold to access other features. Both parties can not use the hold feature simultaneously.

Network Address Extension

This is the addition of up to 40 decimal digits to the networking routing digits which may be needed by an auxiliary machine or other switching device.

Night Service

By control of each Operator group in the network, calls may route to their own night service position or another operator. The specified night service point can be in any node.

Non Specified Information

Signaling information that is not specified in DPNSS1 for manufacture specific use. The header of the data string for this feature contains a system manufacturers code.

Priority Breakdown

On calling a busy extension or meeting network congestion, the A party can request clearing of an existing call in order to establish its own. Breakdown and protection levels (0-15) control this feature.

Register And Cancel Diversion

The initialization or cancellation of diversion features can be controlled from a remote PBX.

Register Do Not Disturb

The initialization and cancellation of do not disturb can be controlled from a remote PBX.

Remote Alarm Reporting

PABX alarm information can be sent around the network,. normally to a central location.

Series Call

A call can be marked as a series call, making it return to the assistance point on clearing by the B party. This will continue until either the series marking is canceled or the A party clears.

Simple Telephony Call

A voice call made between two extensions across the Network without operator assistance or any other feature invoked.

Swap

The change of a call type from either:

- voice to data
- data to voice
- data to data with change of operating characteristics e.g. mode or speed.

Text Message

A short message can be sent across the network for display on a telephone instrument. This is done purely by signaling without any call set up.

Three Party Takeover

If a call is in the three party situation but not in conference, the non controlling party make take control then continue any three party operation.

Time Synchronization

This allows the real time and date to be synchronized throughout the network.

Traffic Channel Maintenance

This gives the ability to busy a channel and perform tests on it. The channel may be looped back to itself or connected to another channel and tests may be manual or automatic.

Service Independent Strings

Service independent strings are strings of data sent over ISDN links indicating activity in progress. These strings contain data relating to miscellaneous information that may be required during the set up and use of other features e.g. class of service, queue information, passwords. The following group of user features use service independent strings:

Auto Answer

Indicates that answer has been generated by the called terminal equipment.

Busy Information

Indicates which features may be used after calling a busy extension:

- call offer
- intrusion
- call back
- call back messaging

Class Of Service

The A parties class of service controlling:

- Route restriction
- call barring
- Access to facilities

DASS2 Service Indicator Code

A DASS2 service indicator code may be passed within a DPNSS1 message. Normally for calls to or from DASS2.

Device Identity

Identifies a destination address if Originating or called line identities do not achieve this.

Direct Route Select

This indicates that the route specified in the call set up must be used and not any alternative:

- selection of higher grade circuits
- avoidance of routes that may be suspect
- avoidance of more expensive routes
- test calls

External Rerouting Disabled

Calls encountering congestion cannot be rerouted via another signaling system or to a different address using the DPNSS1 network.

Handsfree - Activated

This detail can be sent at any time in the call to either party that the other is in or has gone into the hands free state.

Handsfree - Deactivated

As point 13 but sent on deactivation of the feature.

Hunt Group Forwarded

If a call is made to a Hunt group that is split over the network, this information tells the receiving PBX that the call was originally for a Hunt group:

- to avoid nesting
- to take specific action for diversion or call waiting
- to give indication to called party

Intercom Call

If a intercom call is made between PBX systems, special action may be needed:

- calling signal
- calling display
- call logging
- diversion features

Internal Rerouting Disabled

When encountering congestion, the call cannot be rerouted via the DPNSS1 network.

NAE - Identity Incomplete/Complete

If the Network Address Extension is being used the identity complete message is normally sufficient, but if the NAE is greater than 45 octets then it is split into 2 messages. The first will contain the incomplete message and the second the complete.

Password

This string is used to check passwords when needed within the network.

Queue Information

Used when calling a busy extension to indicate:

- the number of calls currently queued
- the position of this call in the queue
- the number of positions capable of answering the queued calls

Queuing Priority

On calls to busy extensions when it is needed to take special action, priority levels (0-255) can be used to control this.

Satellite Barred

This indicates that a call may not be routed via a satellite:

- if the call has already made one satellite hop
- if the call should not use satellites at all

Supplementary Features Inhibited

This indicates that the use of any supplementary features should be inhibited.

Text Display

To send additional text information relating to features:

- more than basic CLI e.g. name
- reason for call failure

Trunk Identity

Information on the incoming trunk call:

- PBX number
- trunk route number
- trunk route member number

DPNSS1 supplementary services

This section provides detailed information on the supplementary services, or features, that are available for the current release of DPNSS1 on Meridian 1.

For each supplementary service, subsections have been created that offer a functional description, detail any specific requirements or limitations, explain interactions with other features, list software package requirements, and outline implementation procedures.

DPNSS1 Attendant Call Offer

The DPNSS1 Attendant Call Offer feature allows an attendant-extended calls routed over DPNSS1 to be camped-on to a remote busy extension. The DPNSS1 signaling used to implement the feature complies with section 14 of BTNR 188. This Call Offer functionality is provided over a DPNSS1 network or over a DPNSS1 to ISDN gateway.

After being offered the camp-on, the destination party may accept or reject it. During the camp-on offer, the destination party receives camp-on tone, heard over the conversation. The destination party accepts the call offer by clearing the established call (the offered call may not be accepted by simply placing the established call on hold). The destination party rejects the call offer by not answering it.

If the busy party goes on hook, allowing the offered call to ring the set, the recall timer for the call is reset to the value programmed for ringing calls. If the call remains unanswered when this timer expires, the offered call is recalled to the attendant queue. If the call is accepted, the originating party receives ringback until the destination party goes off hook to answer the call.

If the call is rejected, the camp-on is recalled to the attendant after the camp-on timer times out (timing for camp-on recall begins as soon as the attendant presses the Release key to extend the camp-on to the destination party.) The destination party may still answer the camp-on as long as the call is still on the attendant console (that is, while the attendant is talking to the source.) The attendant may clear the camp-on by releasing the destination.

Operating parameters

- Timed Reminder Recall feature for DPNSS1 must be equipped.
- Call Offer over DPNSS1 applies only to attendants with an established call on the source side. An attendant can then camp-on a call only if the destination station is on an established call.
- An inquiry call cannot camp-on to a busy station without attendant intervention — the camp-on attempt is rejected (an inquiry call exists when two stations are established in a simple connection, and one station offers a call transfer to another station. The set making the call transfer places the other established station on hold before making the call transfer.) If the busy station has Call Waiting Allowed Class of Service, the call offer will be presented to the busy station as a call waiting call.
- Only one call at a time may be camped-on to a busy destination station.
- Camping-on is possible to a set in a ringback or dialing state.
- Calls cannot be camped-on to a busy destination station with Call Waiting Allowed Class of Service, or that is second degree busy.
- During Night Service, any camped-on call is canceled and recalled to the night DN or re-routed to an attendant at another node if NAS is configured and active.
- If mixed ISDN/DPNSS1 route lists are programmed at a gateway node, an incoming call over an ISDN route that uses an outgoing DPNSS1 route for a first call without call offer, will use the same DPNSS1 route for a call offer.

Feature interactions

- The destination receives camp-on tone if the destination set has Warning Tone Allowed Class of Service, and Camp-on Tone Allowed has been configured for the customer in overlay 15 (the Customer Data Block.)
- Semi-Automatic Camp-on does not function over DPNSS1.
- DPNSS1 Executive Intrusion Conversion is not supported on the Meridian 1.

- The Flexible Orbit Prevention Timer *should be set* to a value of “0” in overlay 15. A non-zero value may cause problems for DPNSS1 calls encountering call forwarding, since two consecutive calls would be initiated from the originating station to the terminating station — refer to the feature interaction description for DPNSS1 Diversion which follows.
- In the case of DPNSS1 Diversion, if a call encounters a station with Call Forward active, then a new call is initiated from the originating node to the call forward extension. The following situations are considered involving attendant-extended calls:
 - An attendant extends a DPNSS1 call originating from a Meridian 1, and the call does not contain a request for call offer. If the destination has Divert Busy or Divert Immediate active, then a new call is initiated as a simple call. If the call contains a call offer request, a new call is initiated using the new address and containing a Call Offer String.
 - An attendant extends a DPNSS1 call terminating at a Meridian 1, and the call does not contain a request for call offer. If the destination has Call Forward All Calls active, then the call is routed to the call forward extension. This also applies if the destination is busy, and active with Call Forward Busy
 - If the call contains a request for call offer and encounters a destination with Call Forward All Calls active, then the call is routed to the call forward extension. If the routing involves a DPNSS1 trunk, then the call will contain a Call Offer Supplementary Information String. If the call encounters a destination with Call Forward Busy active, the call is camped-on to the destination, if camp-on is allowed. If camp-on is *not* allowed, the call is routed to the call forward busy extension. If the routing involves a DPNSS1 trunk, then the call will contain a Call Offer Supplementary Information String.

- If an attendant-extended DPNSS1 call encounters Diversion Busy or Diversion Immediate at a Meridian 1 gateway node, then a new call is initiated from the gateway node to the diverted destination. If a request for call offer was contained in the original DPNSS1 call, it will also be contained in the new call. If a request for call offer was *not* contained in the original DPNSS1 call, and if the new destination is busy and camp-on is allowed on it, then the call offer request will be included in the new call.
- The displayed information normally provided by the Attendant First-Second Degree Busy Indication that indicates that a station on a far node is in first or second degree busy status, is *not* provided in an ISDN/DPNSS1 gateway scenario.
- The Slow Answer Recall Modification feature, upon recall to the attendant from the original camp-on destination, requires the attendant to dial the extension again to be able to extend and camp-on the call again.
- Integrated Services Access (ISA) Call Types are not supported over DPNSS1.
- DPNSS1 operation and features are not supported with Meridian Link.
- Where the Secrecy feature is concerned, DPNSS1 calls, while at the source of the attendant, are considered as internal calls, unless the destination is a trunk.

Feature packaging

- DPNSS1 Network Services (DNWK) package 231

Dependencies:

- Digital Private Network Signaling System No.1 (DPN) package 123;
- Digital Access Signaling System No.2 (DAS) package 124;
- Integrated Digital Access (IDA) package 122, dependent on Digital Trunk Interface (PBXI) package 75, 2 Mbit Digital Trunk Interface (DTI2) package 129;
- Network Attendant Service (NAS) package 159;
- 2 Mbit Primary Rate Interface (PRI2) package 154, dependent on Integrated Services Digital Network (ISDN) package 145, Digital Trunk Interface (PBXI) package 75

Feature implementation

- None.

DPNSS1 Attendant Timed Reminder Recall and Attendant Three Party Service

This feature implements the portion of the DPNSS1 Three Party Service which relates to attendant console operation. It also extends the operation of the existing DPNSS1 Timed Reminder Recall feature to calls extended from Meridian 1 operator consoles over DPNSS1 links.

Timed Reminder Recall allows a call that is extended by an attendant over a DPNSS1 trunk, to be recalled to the attendant if not answered within a customer-defined period of time. The source and destination parties remain active in the call.

Three Party Service allows the source and destination set displays to be updated, after the extended call has been answered at the destination set.

The recall timing is implemented at the Meridian 1 attendant node in the form of the slow answer recall timer and the camp-on timer. When the attendant extends a call to a destination extension in the ringing or call waiting state, by pressing the RLS key, the slow answer timer is started. If the extended call is camped on to a busy destination set, the camp-on timer is started.

If the idle set answers the call extension or the busy set answers the call waiting, the slow answer timer is stopped and the attendant node becomes a standard transit node. Messaging is immediately sent to update, by the Three Party Service feature, the displays on the source and destination sets.

If the busy set becomes free to answer the camp-on, the camp-on timer is stopped, the set receives ringing and the slow answer timer is started. If the call is answered, the slow answer timer is stopped and the attendant node becomes a standard transit node. The displays on the source and destination sets are updated.

If the slow answer recall timer expires, the source is recalled to the attendant, with the destination party still ringing or busy (in the case of call waiting). The attendant may extend the call once more by pressing the Release key. This causes the slow answer timer to start again. If the camp-on timer expires, the source is recalled to the attendant. The attendant may extend the camp-on once more by pressing the Release key. This causes the camp-on timer to start again.

During Night service, a DPNSS Timed Reminder recall will recall to the Night DN only if the Source call is extended; otherwise it will be placed on the attendant queue. If NAS is available, the call will be routed to the first available attendant found over NAS routing.

The attendant is also able to operate the Hold key before extending the Source call (by using the RIs key. This allows the attendant to break back into the Source call before the destination answers or before waiting for the DPNSS Recall Timer to expire. If the attendant does not operate the Hold key, they will not be able to retrieve the extended call, but rather must wait until the recall timer expires (assuming that the destination does not answer).

It may be that the attendant node is a gateway node. If a call on the source comes in over an ISDN trunk and is extended to the destination over a DPNSS1 trunk, then recall timing is done by the DPNSS1 Timed Reminder Recall feature. After the attendant extends the call and the destination answers, the recall timer is stopped. The controlling (attendant) node sends signaling to the destination node, to update the display of the destination set. The display on the source set is updated only if Network Call Redirection feature is equipped. If the recall timer expires before the destination answers the call, the source is recalled to the attendant with no messaging being sent over the ISDN link.

If the source comes in over a DPNSS1 link and is extended to the destination over an ISDN link, then the recall timing is done by the Network Attendant Service (NAS) feature. After the attendant extends the call and the destination answers, the recall timer is stopped. The controlling (attendant) node sends signaling to the source node, to update the display of the source set. The display on the destination set is not updated by NAS; it is updated only if Network Call Redirection feature is equipped, and the call has been extended after the destination party has answered. If the recall timer expires before the destination answers the call, the source is recalled to the attendant with no messaging being sent over the DPNSS1 link. If the attendant transfers the call after it has been answered by the destination, the display of the source set is updated by Three Party Service signaling. The display of the destination set is updated if Network Call Redirection is equipped.

Operating parameters

- The slow answer timer is used for calls extended to a set in the call waiting state (since the DPNSS1 Call Waiting supplementary service has not yet been implemented).

Feature interactions

- If a call is extended over a DPNSS1 link to an ACD set, the controlling node will time for a slow answer recall while the call is in the ACD queue at the destination node.
- Recall timing is not performed for attendant-to-attendant calls.
- For calls extended over a DPNSS1 link, the slow answer recall timer is used instead of the call waiting recall timer, since the Meridian 1 does not distinguish between a call extended to a set that is idle or in call waiting state.
- To prevent recall timing from being done at the originating node if the timing is done at the controlling node, the EEM messages TRFD and RECON are not sent from the controlling node to other parties involved in the call (this inhibits the DPNSS1 Call Redirection feature).
- The DPNSS1 Loop Avoidance string (LA) may be added to the Initial Service Request Message (ISRM) of an inquiry call.
- The Attendant Three Party Service at a controlling Meridian 1 node initiates the signaling sequence that causes the DPNSS1 Route Optimization feature, equipped on the originating node, to optimize the route between the originating and destination parties.
- A Recall ICI key that has been defined in the customer data block lights up when DPNSS1 timed reminder recall occurs.
- If the Night Service Improvements package is equipped, recalls to the attendant during night service are routed to the night DN. This applies to calls extended over ISDN and DPNSS1 trunks.
- If, while extending a call, the attendant presses the HOLD key before pressing the RLS key, the call is placed on permanent hold on the loop key. The call is subject to the Timed Recall feature.

- Pressing the Recall key on a set has no effect, if the set is connected to the attendant over a DPNSS1 trunk and is being timed for slow answer recall or camp-on recall.
- If the Recall to the Same Attendant feature is configured for an attendant, slow answer recalls and camp-on recalls occur to the same attendant.
- If the Secrecy feature is activated by the attendant, the source side of a call being extended by the attendant is excluded if the destination is a trunk. As applied to a DPNSS1 trunk, the Calling Line Category received from the trunk must be DEC, DASS2, PSTN, or MF5.
- If the Slow Answer Modification feature is activated by the attendant, the destination party is dropped when the recall occurs to the attendant.
- The Semi-Automatic Camp-on feature does not apply to calls extended over a DPNSS1 trunk.
- If a call is routed during night service by the Network Attendant Service feature to a set over a DPNSS1 link, the Night Forward No Answer feature will not apply as the information indicating that the call has been answered cannot be sent.
- DPNSS1 does not support either the Group Hunt or Group Hunt Queuing features.
- DPNSS1 operation will not support the following:
 - ISDN Basic Rate Interface (BRI) setsor
 - ISDN BRI Numeris Terminals.
- DNIS information (number and name) for redirected calls will be retained and available to the called party if the redirection terminates on the original node where the DNIS information is available.

Feature packaging

- DPNSS1 Network Services (DNWK) package 231

Dependencies:

- Digital Private Network Signaling System No. 1 (DPN) package 123;
- Integrated Digital Access (IDA) package 122, dependent on Digital Trunk Interface (PBXI) package 75, 2 Mbit Digital Trunk Interface (DTI2) package 129;
- International Supplementary Features (SUPP) package 131
- Basic Alternate Route Selection (BARS) package 57, dependent on Basic Routing (BRTE) package 14 and Network Class of Service (NCOS) package 32
- Co-ordinated Dialing Plan (CDP) package 59, dependent on Dependent on Basic Routing (BRTE) package 14 and Network Class of Service (NCOS) package 32
- Pretranslation (PXL) package 92
- Flexible Numbering Plan (FNP) package 160
- Network Attendant Service (NAS) package 159, for a mixed ISDN/DPNSS1 environment

Feature implementation

- **LD15-** In the Customer Data Block, define the parameters for recall timers and the attendant DN. Define a Night DN for the customer.
 - In response to the ATDN prompt, enter the attendant DN. Recalls occur to this DN, upon expiration of the recall timer.
 - In response to the RTIM prompt, enter a value for the slow answer recall timer (0-(30)-378), camp-on recall timer (0-(30)-510), and call waiting recall timer (note that for calls extended over a DPNSS1 link, the slow answer recall timer is used instead of the call waiting recall timer, since the Meridian 1 does not distinguish between a call extended to a set that is idle or in call waiting state. The slow answer value is used in both cases).

DPNSS1 Diversion

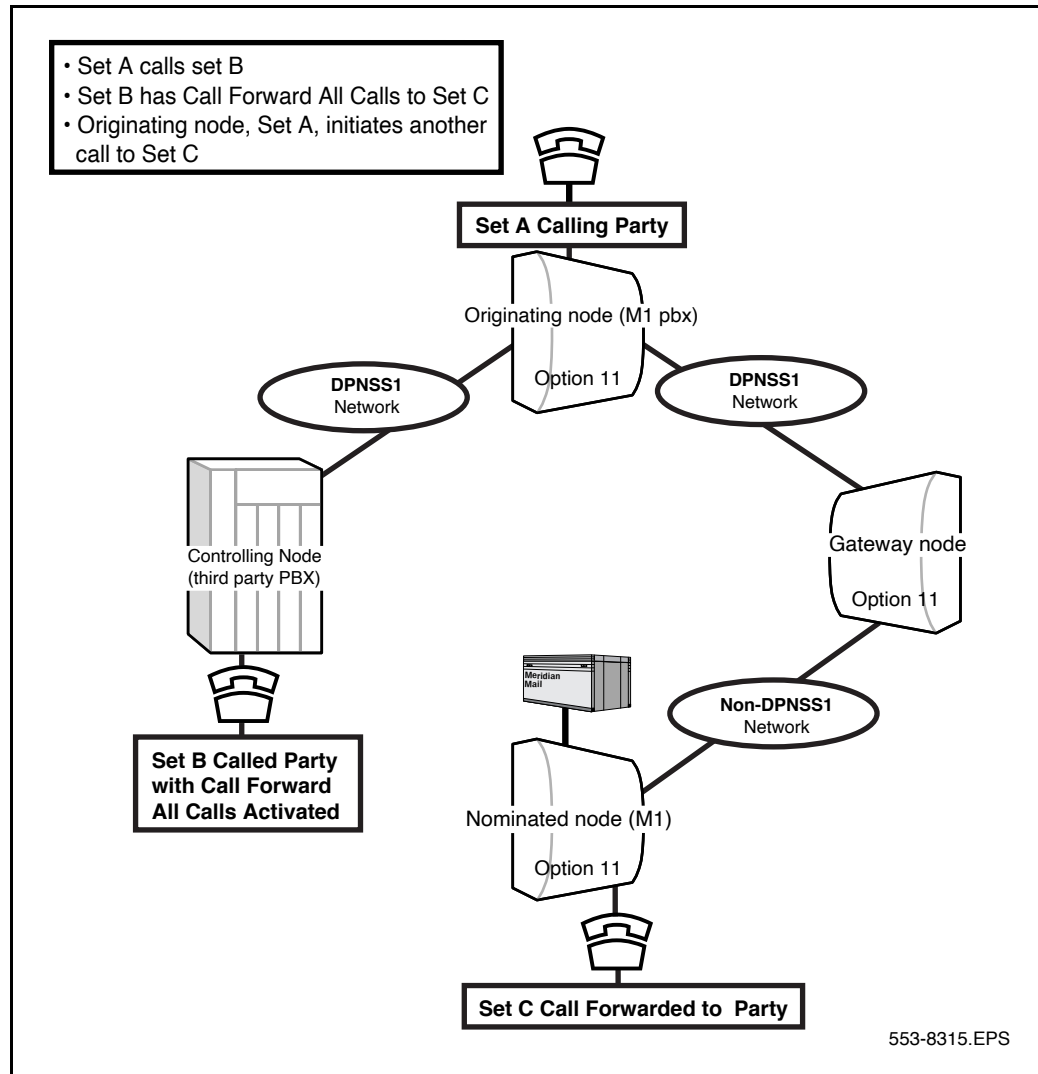
DPNSS1 Diversion is a British Telecom Network Requirement (BTNR) supplementary service that provides full DPNSS1 Diversion signaling on DPNSS1 links, when one of the redirection features listed below is invoked:

- Call Forward All Calls;
- Call Forward No Answer;
- Call Forward by Call Type;
- Call Forward Busy;
- Hunting/Group Hunting;
- Intercept Computer Call Forward All Calls;
- Call Forward Internal Calls;
- Meridian Customer Defined Network Call Redirection; and
- Call Party Name Display.

When a set activates a redirection feature such as Call Forward All Calls, DPNSS1 signaling informs the call originating node that the call is being forwarded to another set. If the forwarded party is located on another node, the call originating node is requested to initiate a new call. When the forwarded to party is reached by the call originator via DPNSS1, the forwarded to party is notified that the incoming call has been forwarded.

As illustrated in Figure 108, Call Diversion functions on Meridian 1 nodes that are linked to third party Private Branch Exchanges (PBXs) within a DPNSS1 network. Meridian 1 gateway nodes provide links to other Meridian 1 nodes by Meridian Customer Defined Network (MCDN) and to Meridian 1 and third party PBX through DPNSS1. In this figure, the Nominated party or forwarded to party, is reached through full DPNSS1 or combined DPNSS1 or non-DPNSS1 (such as MCDN) routes.

Figure 108- DPNSS1 Diversion Environment.



The following capabilities are provided as part of the DPNSS1 Diversion: Diversion Validation, Diversion Cancellation, Diversion Follow-Me, Diversion By-Pass, Diversion Immediate, Diversion On Busy and Diversion On No Reply. These capabilities are described as follows.

Diversion Validation

DPNSS1 Diversion operates on Meridian 1 nodes that are linked to third party PBXs, within a full DPNSS1 environment. Meridian 1 gateway nodes are linked with other Meridian 1 nodes through a Meridian 1 Customer Defined Network (MCDN) and the other Meridian 1, or to other third party PBXs via DPNSS1 networks.

Diversion Cancellation

Diversion Cancellation allows the forwarded to party to remotely deactivate call diversion initiated by the forwarding party. Meridian 1 DN's cannot originate Diversion Cancellation requests however, Meridian 1 PBXs can process Diversion Cancellation requests but cannot initiate requests.

The sequence for Diversion Cancellation is the following. Set A has activated Call Forward All Calls (CFAC) to Set B. Set B, the forwarded to party, requests either Diversion Immediate or Diversion-All Cancellation to Set A. Upon receipt to the cancellation request, Set A's node determines that Set B is currently CFAC activate to Set A's DN. If the DN is confirmed, then the CFAC feature is deactivated and Set B is notified that the cancellation request is successful.

If Diversion Cancellation request encounters any gateway, the gateway responds with a "Service Unavailable" notification.

Diversion Follow-Me

Diversion Follow-Me allows the forwarding party to remotely request and change a new forwarded to party. As an example, Set A has activated Call Forward All Calls to Set B, in a full DPNSS1 environment. Set A then decides to change Set B, the forwarded to party, to Set C. When Diversion Follow-Me is activated, Set A's node uses Diversion Validation to confirm that the new forwarded DN is valid.

If a Diversion Follow-Me request encounters any gateway, the gateway responds with a “Service Unavailable” notification. A Follow-Me request is always rejected when routed through a gateway.

Meridian 1 PBXs can process Diversion Follow-Me requests but cannot initiate any requests.

Diversion By-Pass

Diversion By-Pass allows the calling party to ignore the diversion assigned by the party that activated call redirection. Meridian 1 DNPs cannot originate Diversion By-Pass requests, but can process requests.

Diversion Immediate

With Diversion Immediate, the calling party, Set A, dials Set B that has activated Call Forward All Calls (CFAC) to Set C. Upon receipt of the call, Set B’s node instructs Set A’s node to Divert-Immediate to Set C.

When instructed to divert, Set A’s node clears the old call and initiates a new call to Set C. Set A’s display is updated with diversion information, when the call is established with Set C.

Diversion On Busy

The sequence for Diversion On Busy via Separate Channel is similar to Diversion-Immediate. The differences occur with message contents and the reason for diversion, if Call Party Name Display is activated.

If Diversion on Busy is triggered by the Hunt feature, it is also triggered by Call Forward By Call Type applied to Hunt. A node determines an internal call on the Calling Line Category (CLC) received with the incoming call.

For Call Forward Busy, the following two cases exist. If the forwarded set has enabled Message Waiting Forward Busy, the call may be directed to the FDN or Message Waiting key. In this case, the Diversion On Busy signaling applies. However, if the forwarded set is not equipped with Message Waiting Forward Busy, the call is always routed to the Attendant.

Diversion On No Reply

Call Diversion on No Reply ensures that a Call Forward No Answer (CFNA) call is not disconnected until the new diversion call is successful.

The sequence for Diversion On No Reply functionality is the following. Set A, the calling party, dials Set B. Set B rings and has Call Forward No Answer activated to Set C. When requested by Set B's node to Divert the call on No Reply, Set A's node initiates a new call to Set C. When Set C answers the diverted call, the original call between Set A and Set C is disconnected.

Operating parameters

Interworking with MCDN Trunk Route Optimization (TRO) is supported.

On Meridian 1 nodes, M3000 Meridian 1 proprietary sets are not supported when using DPNSS1 signaling.

The Nominated party's, the forwarded to set, display is updated in full DPNSS1 or mixed DPNSS1/MCDN routes. The Nominated party can be Meridian Mail.

Access forwarding to Meridian Mail via a full DPNSS1 environment operates from X11 Release 20 and later Meridian 1 nodes. Access forwarding to Meridian Mail via full MCDN/DPNSS1 gateway operates, if all the gateways between the calling party and the Meridian Mail host node are X11 Release 22 Meridian 1 nodes.

The Message Waiting Indication (MWI) key of a Meridian 1 DN is never lit if reached from a Meridian 1 Meridian Mail node via DPNSS1. No Message Waiting Indication signaling is implemented on DPNSS1 between Meridian 1 nodes. No MCDN/DPNSS1 functionalities exist for Meridian Mail Message Waiting Indication capabilities.

In a mixed Uniform and Coordinated Dialing Plan environment, it is recommended to always use the Uniform Dialing Plan (UDP) format for forwarded data.

Feature interactions

Attendant Forward No Answer

If an incoming call is handled for Network Attendant Services routing towards DPNSS1, no diversion signaling is sent back to the calling party.

Call Forward All Types

The Call Forward All Types features on unanswered calls are activated in the following order: Call Forward All Calls, Message Waiting, Call Forward No Answer, Slow Answer Recall. For busy sets the order is: Call Forward All Calls, Hunting, Calling Waiting/Camp On, Message Waiting Busy Forward, Call Forward Busy.

Group Hunting

Only simple DPNSS1 calls support Group Hunting. All DPNSS1 supplementary services do not support Group Hunting.

Meridian Mail

Following DPNSS1 Diversion, the new call can reach Meridian Mail via full MCDN, full DPNSS1 and mixed MCDN/DPNSS1 links. Following MCDN Call Forward towards DPNSS1, the new call can reach Meridian Mail via full MCDN, full DPNSS1 and mixed MCDN/DPNSS1 links.

Night Service

If a diverted call encounters an attendant in night service, the call receives Night Service Diversion if available.

Phantom Directory Numbers Phantom Terminal Numbers

If an incoming call to a Phantom TN contains a DIVERSION BY-PASS REQUEST, Call Forward All Calls applies.

Route Optimization

If a Route Optimization call setup encounters any redirection features, these features are ignored. The condition for a diverted call to have Route Optimization after connection is the same as a simple DPNSS1 call. Route Optimization starts if the diverted call is routed through a non-first choice route or when a call transfer involving the diverted call is completed.

User Selectable Call Redirection

The User Selectable Call Redirection feature triggers Diversion Validation. If the numbering plan is DPNSS1 then diversion occurs. Numbering plan routes are checked to determine if redirection DN's are through DPNSS1 on a first choice route basis. If the number plan is not a DN through DPNSS1, then User Selectable Call Redirection works as usual.

Feature packaging

DPNSS1 Diversion requires DPNSS1 Network Services (DNWK) package 231.

Basic DPNSS1 networking requires:

- Integrated Digital Access (IDA) package 122;
- Digital Private Network Signaling System 1 (DPNSS) package 123;
- 2.0 Mbps Primary Rate Interface (PRI2) package 154.

DPNSS1/MCDN Gateway requires:

- International Supplementary Features (SUPP) package 131;
- Integrated Services Digital Network (ISDN) package 145;
- ISDN Advanced Network Services (NTWK_SRVC) package 148;
- Network Attendant Services (NAS) package 159;
- ISDN Call Connection Limitations (ICCL) package 161 for gateway with loop avoidance.

Feature implementation

For DPNSS1 Diversion to occur redirection features must be configured.

Feature operation

Activating Call Forward All Calls over DPNSS1

Set A invokes Call Forward All Calls (CFAC) to Set B, the forwarded to party over a DPNSS1 network. In a non-DPNSS1 network environment, then the Call Forward All Calls is normal operation.

- 1** If the dialing plan reaches Set B via DPNSS1, a VALIDATION REQUEST is sent to Set B. The CFAC key remains flashing.
- 2** Upon receipt of the request, Set B's node responds to the validity of Set B's DN. If the DN is valid, the CFAC feature is activated. The CFAC key is lit. However, if the DN is not valid, Set A hears an overflow tone, and the CFAC key remains flashing.
- 3** If the forwarded DN is local or reached through a non-DPNSS1 network, the usual CFAC activation process applies. No DPNSS1 messaging occurs.

DPNSS1 Executive Intrusion

Digital Private Network Signaling System 1 (DPNSS1) Executive Intrusion enables an originating party to break-in to an established call under certain circumstances. The Meridian 1 only allows this feature to be activated from Attendant Consoles; however, it will accept an Executive Intrusion activation request from a regular set on a third-party PBX.

For the purposes of this feature description the term “requested” party will be used to describe the person on the established call who the originating party desires to talk with, and the “unrequested” party will mean the person on other end of the call. On an Option 11, Executive Intrusion is only activated if the attendant places the call to the requested party over a DPNSS1 link. If the attendant and the requested party are located on the same node, the current Attendant Break-in feature is activated. Executive Intrusion is activated by using the existing Break-In key on an Attendant Console.

When an attendant presses the Break-In (BKI) key to invoke Executive Intrusion, the node where the requested party resides checks the Intrusion Capability Level (ICL) of the Attendant Console against the Intrusion Protection Levels (IPLs) of the parties involved in the call. If the ICL is higher than the IPLs, Executive Intrusion is allowed and a conference is set up between the attendant, requested, and unrequested parties.

Operating parameters

The Option 11 implementation of Executive Intrusion can be used on an Option 11 in any environment where DPNSS1 connectivity is involved.

Executive Intrusion with prior validation is not supported.

Withdrawal from Intrusion is not supported.

Executive Intrusion has the same limitations as Post-Dial Attendant Break-In as follows:

- Only one Break-In/Executive Intrusion key is allowed per Attendant Console
- An Executive Intrusion connection cannot be put on hold
- Only one attendant at a time is allowed to intrude for a given connection, and
- Executive Intrusion is permitted only if the requested party is a BCS or PBX set and has Warning Tone Allowed (WTA) Class of Service.

In a full DPNSS1 environment, Executive Intrusion adds the following limitations:

- Executive Intrusion is permitted only if the unrequested party is a BCS or PBX set having Warning Tone Allowed (WTA) Class of Service,
- Executive Intrusion is not permitted if the requested or unrequested party is involved in a conference.

In a DPNSS1/Meridian Customer Defined Network (MCDN) gateway between the originating party and the requested party, Executive Intrusion is not permitted if Call Offer has been activated at the terminating node by the same attendant. Call Offer takes precedence over Executive Intrusion in a DPNSS1/MCDN gateway.

In a DPNSS1/MCDN or MCDN/DPNSS1 gateway between the originating party and the requested party, only Executive Intrusion activation requests from attendants are supported. Executive Intrusion from sets on a third party PBX are ignored in an DPNSS1 Initial Service Request Message (ISRM) and rejected in a DPNSS1 End-to-End Message (EEM).

At the gateway node, if mixed MCDN/DPNSS1 route lists are programmed, an incoming MCDN call using an outgoing DPNSS1 route for the first call attempt (without Executive Intrusion) will also use a DPNSS1 route for the Executive Intrusion request.

On the Option 11 the following hardware is required:

- DPNSS1 XPRI2 – NTAK799.

Feature interactions

Interactions with other DPNSS1 Supplementary Services

DPNSS1 Diversion

In the following scenario an Executive Intrusion request is made on a diverted call (Immediate). Set B has diversion immediate active to set C (set B is on a third-party PBX). Set C may be on the same node or on another node. C is busy on an call with another set. The attendant calls B. The answer to the Initial Service Request Message (ISRM) is a Number Acknowledge Message (NAM) with a Destination Address of C. The attendant position then sends a regular ISRM to C. Since C is busy, the attendant receives Clear Request Message (CRM) in response to the ISRM. The attendant presses the BKI key. In this case, an Executive Intrusion ISRM is sent to C, and C is considered the requested party.

In the following scenario an Executive Intrusion request is made on a diverted call (Busy). Set B has diversion on busy active to set C. Set C may be on the same node or on another node. Both B and C are busy on calls with other sets. The attendant sends a regular ISRM to C. Since C is busy, the attendant position receives a Clear Request Message in response. The attendant presses the BKI key. In this case, an Executive Intrusion ISRM is sent to C, and C is considered the requested party.

DPNSS1 Route Optimization

If the requested party is involved in a Route Optimization process when it receives an Executive Intrusion request, the request is rejected. Conversely, the originating, requested and unrequested parties will be able to send a Route Optimization request only after the Executive Intrusion conference reverts to a simple call. Finally, if an Executive Intrusion request is received after a Route Optimization Request Supplementary Information String has been sent but Route Optimization has not actually commenced, the Route Optimization process is aborted and the Executive Intrusion may proceed.

DPNSS1/Uniform Dialing Plan Interworking

DPNSS1/Uniform Dialing Plan Interworking does not affect Executive Intrusion operation, except with regard to displays. The Executive Intrusion states normally displayed are Coordinated Dialing Plan Calling Line IDs and Originating Line IDs. If a Uniform Dialing Plan is active in the network, displays will change to Uniform Dialing Plan Calling Line IDs and Originating Line IDs.

Executive Intrusion denied for the Wanted Node during DPNSS1 Three-party Service

Executive Intrusion will not be allowed if either the requested or unrequested party is involved in an enquiry call. In addition, Executive Intrusion will be denied if the requested party or the unrequested party is the controlling or the added-on party of a three-party conference call. The third party (the one held during the enquiry call before the conference is completed) is not subject to this restriction.

Step Back on Congestion

If Step Back on Congestion (SBOC) is active, an ISRM containing an Executive Intrusion request will undergo the SBOC routing process as per any other call.

Other Interactions

Attendant Blocking of DN

If an Executive Intrusion attempt is made for an Attendant Blocking of DN call, the Executive Intrusion attempt is denied.

Attendant Conference

If an Executive Intrusion conference is established on the Destination side, pressing the Attendant conference key is ignored.

Attendant Secrecy

Enhanced Secrecy

If attendant secrecy is not active when the attendant attempts Executive Intrusion, the source is automatically excluded. If Enhanced Secrecy is equipped, source exclusion includes the removal of the Enhanced Secrecy warning tone when Executive Intrusion is activated.

Automatic Call Distribution (ACD)

Once the requested party has established the call with an ACD agent, the attendant is able to intrude into the call. However, if the requested party is in an ACD queue, Executive Intrusion is denied.

Break-In

Break-In to Enquiry Calls

Break-In with Secrecy

Break-In Indication – Prevention

Executive Intrusion and Break-In are mutually exclusive. Pressing the BKI key will activate Break-In or Executive Intrusion. In addition, intrusion is not allowed into a Break-In conference.

Call Park

Attempts to intrude into a parked call receive Executive Intrusion Denied treatment.

Call Waiting

Executive Intrusion is permitted (consult-only state) into a requested party having call waiting.

Conference

Enquiry Calls

Executive Intrusion is denied if the requested party is established in a local conference, or if the requested party is involved in an enquiry call. These restrictions may apply to the unrequested party depending on the connection being used between the requested and unrequested parties.

Data Calls

Executive Intrusion cannot be applied to data calls.

Hold or Permanent Hold

Executive Intrusion is denied if the requested party is put on hold by another station at the same node. This restriction also applies to the unrequested party if the unrequested party is located at the same node as the requested party (standalone) or if the requested party and the unrequested party are linked via DPNSS1.

Hunting

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.

Integrated Services Digital Network (ISDN) Basic Rate Interface (BRI) Extension

Activation of Executive Intrusion for an ISDN BRI extension is not possible. Attempts to intrude on ISDN BRI extensions (either the requested or unrequested party) will fail.

Intercept Computer (Dial from Directory)

Executive Intrusion can be activated by dialing an extension DN from the Intercept Computer Terminal, and then pressing the BKI key on the Attendant Console.

Line Lockout

Executive Intrusion is not allowed for any set that is in Line Lockout state.

Make Set Busy Do Not Disturb

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

Multiple Appearance DN

If the attendant tries to extend a call to a DN which appears on more than one set, this DN can either be:

- Multiple-Call Arrangement with Ringing (MCR): when a call terminates on this DN, all idle stations on which the DN appears are rung. The call is established only with the station which has answered first. All others are idle.
- Multiple-Call Arrangement with No Ringing (MCN): the only difference between MCN and MCR is that the called stations are not rung (only their DN keys flash).

- Single-Call Arrangement with Ringing (SCR): when a call terminates on this DN, all idle stations on which the DN appears are rung. The call is established only with the station which has answered first. All others are busy.
- Single-Call Arrangement with No Ringing (SCN): the only difference between SCN and SCR is that the called stations are not rung (only their DN keys flash).

Switchhook Flash

If a PBX (500/2500) set is part of an Executive Intrusion conference, any Switchhook Flash is ignored.

Feature packaging

DPNSS1 Executive Intrusion is included in Enhanced DPNSS1 Services (DPNSS_ES) package 288.

For configuration of Attendant Consoles the following package is required:

- Attendant Break-In/Trunk Offer (BKI) package 127.

For basic DPNSS1 network functionalities the following packages are required:

- Integrated Digital Access (IDA) package 122
- Digital Private Signaling System 1 (DPNSS) package 123
- International Supplementary Features (SUPP) package 131
- Integrated Services Digital Network (ISDN) package 145, and
- 2.0 Mbps Primary Rate Interface (PRI2) package 154.

The following package is required to provide DPNSS1 Loop Avoidance, Three-Party Service, Call Offer, Step Back on Congestion, and Route Optimization:

- DPNSS1 Network Services (DNWK) package 231.

The following packages are required to provide DPNSS1/MCDN Gateway functionality:

- Advanced ISDN Network Services (NTWK) package 148

- Network Attendant Services (NAS) package 159, and
- ISDN Supplementary Features (ISDNS) package 161 (required to support MCDN/DPNSS1 gateway with Loop Avoidance).

Feature implementation

Setup of the DPNSS1 Executive Intrusion is similar to that of the Attendant Break-in feature. It includes:

- Allowing warning tones for sets and trunks.
- Configuring Priority Levels for Forced Camp-on and Priority Override (PLEVs) for sets and routes. PLEVs for routes determine IPLs when applicable.
- PLEV/IPL mapping for sets.
- Configuring the BKI key on the Attendant Console.

LD 10 – Allow warning tone for PBX (500/2500) sets:

Prompt	Response	Description
REQ	CHG	Change.
TYPE	500	Telephone type.
TN	c u	Terminal Number.
CLS	WTA	Class of Service. Warning tone allowed (WTA) must be set for Executive Intrusion.

LD 11 – Allow warning tone for BCS (digital) sets:

Prompt	Response	Description
REQ	CHG	Change.
TYPE	aaaa	Telephone type, where aaaa = SL-1, 2006, 2008, 2009, 2016, 2018, 2112, 2216, or 2616.
TN	c u	Terminal Number.
CLS	WTA	Class of Service. Warning tone allowed (WTA) must be set for Executive Intrusion.

LD 14 – Allow warning tone for trunks to permit Executive Intrusion:

Prompt	Response	Description
REQ	CHG	Change.
TYPE	aaa	Trunk type, where aaa = ADM, AID, ATVN, AWR, CAA, CAM, COT, CSA, DIC, DID, FEX, ISA, MDM, MUS, PAG, RAN, RCD, RLM, RLR, TIE, or WAT.
TN	c u	Terminal Number.
CLS	WTA	Class of Service. Warning tone allowed (WTA) must be set for Executive Intrusion.

LD 10 – Define PLEV for PBX (500/2500) sets:

Prompt	Response	Description
REQ	CHG	Change.
TYPE	500	Telephone type.
TN	c u	Terminal Number.
PLEV	n, (2)	Give PLEV n to the route (0 ≤ n ≤ 7).

LD 11 – Define PLEV for BCS (digital) sets:

Prompt	Response	Description
REQ	CHG	Change.
TYPE	aaaa	Telephone type, where aaaa = SL-1, 2006, 2008, 2009, 2016, 2018, 2112, 2216, or 2616.
TN	c u	Terminal Number.
PLEV	n, (2)	Give PLEV n to the route (0 ≤ n ≤ 7).

LD 16 – Define PLEV for routes:

Prompt	Response	Description
REQ	CHG	Change.
TYPE	RDB	Route Data Block.
PLEV	n, (2)	Give PLEV n to the route (0 ≤ n ≤ 7).

The Meridian 1 ICL/IPL implementation uses the existing PLEV scale. PLEVs are defined in LDs 10 and 11 for sets, and LD 16 for routes. Make the IPL/PLEV mapping consistent with Priority Override/Forced Camp-On (POVR) operation in case both features exist.

The mapping is as follows:

— ICL/IPL for Attendants:

Since attendants do have any POVR priority, there is no PLEV – ICL and no PLEV – IPL mapping for attendants. When ICL information must be sent through DPNSS1, ICL = 3 (maximum capability level) is assumed for the attendant. When an Executive Intrusion request is received from another node, IPL = 3 (maximum protection level: non intrudable) is assumed for the attendant.

— ICL for sets:

Since sets cannot originate Executive Intrusion requests on the Meridian 1, there is no PLEV – ICL mapping for sets.

— IPL for sets (or routes):

Make the PLEV – IPL mapping consistent with the meaning of PLEV for the POVR feature.

The mapping is as follows:

Table 60
PLEV/IPL mapping for sets

PLEV of Set A	Meaning of POVR	IPL set A is considered to have	Meaning for Executive Intrusion
0	POVR not active: cannot override, cannot be overridden.	3	Total protection: cannot be intruded.
1	Cannot override, can be overridden by PLEVs 1-7.	0	Minimum protection: can be intruded by ICLs 1-3.
2	Can override PLEVs 1-2, can be overridden by PLEVs 2-7.	1	Intermediate protection: can be intruded by ICLs 2-3.
3	Can override PLEVs 1-3, can be overridden by PLEVs 3-7.	2	Intermediate protection: can be intruded by ICL 3.
4	Can override PLEVs 1-4, can be overridden by PLEVs 4-7.	3	Maximum protection: cannot be intruded.
5	Can override PLEVs 1-5, can be overridden by PLEVs 5-7.		
6	Can override PLEVs 1-6, can be overridden by PLEVs 6-7.		
7	Can override PLEV 7, can be overridden by PLEV 7.		

As a consequence, the effect on an incoming EI request (ICL included) on a set with a PLEV configured is the following:

Table 61
Effect of ICLs on different PLEVs for an EI request

ICL in the incoming EI request	PLEVs for which Executive Intrusion is allowed
1	1
2	1-2
3	1-3

LD 12 – Define a BKI/Intrusion key on the Attendant Console:

Prompt	Response	Description
REQ	CHG	Change.
TYPE	1250 or 2250	Attendant Console type.
TN	c u	Terminal Number for the Option 11.
KEY	xx BKI	Define key xx as the BKI key.

Feature operation

From Attendant Consoles the Executive Intrusion feature operates in a similar manner to that of the existing Attendant Break-In feature as follows:

- 1** The attendant dials the destination DN.
- 2** The attendant receives busy tone.
- 3** The attendant presses the Break-In (BKI) key on the console.
- 4** If the ICL on the Attendant Console is higher than the IPLs of both the requested and the unrequested parties, a conference is established between all three parties.
- 5** After the unrequested party disconnects, the attendant can extend the incoming call to another DN if desired.

DPNSS1 Extension Three Party Service

The DPNSS1 Three-party Service feature allows a controlling party to place an established party on hold and make an inquiry call to a third party. The controlling party may then transfer the held party to the inquired-to party, or form a three-party conference. The three parties may be located anywhere across a Meridian 1 DPNSS1 network.

The controlling party may use a 500/2500 set, M1000 series set, or Meridian feature telephone. On a 500/2500 set or M1000 series set, an inquiry call may be initiated by pressing the Recall key or performing a switch-hook flash. On a Meridian feature phone, an inquiry call may be initiated by pressing the Transfer or Conference key.

After a call transfer, this feature provides messaging that allows DPNSS1 Route Optimization supplementary service to be invoked, in order to optimize the routing of the call through the DPNSS1 network. Also, user set displays are updated, and applicable DPNSS1 access restrictions are applied. These include:

- restrictions configured as part of the Trunk Barring feature;
- Public Switch Telephone Network call barring, configured at the telephone set level;
- system restrictions dependent on trunk types (not configurable).

The following access restrictions are not supported:

- Tenant Service restrictions;
- Network Class of Service restrictions.

This feature handles various types of misoperation when the controlling party attempts to transfer from the held party to the inquired-to party. If the user of a Meridian feature phone presses the Transfer key a second time after having already pressed it once to transfer the call, the action is ignored.

For a 500/2500 set, misoperation may occur if the controlling party attempts to transfer after performing an unsuccessful inquiry call to the third party. The inquiry call may have failed due to the controlling party dialing an incomplete number, the inquiry call still being in the set-up stage, or the inquiry call encountering busy tone, overflow tone, or recorded announcement. The held call, if external, is intercepted to the attendant rather than dropped (the held call is dropped if it is an internal call). However, in the cases of overflow and recorded announcement, if the inquiry call remains connected to the overflow tone or recorded announcement until time out occurs, then the held party is dropped.

Misoperation from a 500/2500 set is also prevented in cases where an inquiry call to the third party is successful, but a transfer connection between the held party and inquired-to party is prevented due to trunk-to-trunk access restriction. If the controlling set hangs up, then the inquiry call is disconnected. If the held call is external, it is recalled to the controlling party. If the held call is internal, it is disconnected.

In cases where a call transfer from a held party to an inquired-to party is successful when it should not have been allowed, the call is forced to disconnect.

Operating parameters

- None.

Feature interactions

- Within a mixed ISDN/DPNSS1 environment, all nodes with ISDN links must be equipped with the Network Attendant Service feature.
- As part of Multi-party Operation, a control digit (0-9, or an # or *) must be dialed to toggle, disconnect, or conference.
- If three-party conference is provided as part of Multi-party Operation, and if MPO is configured as 'disconnect during consultation connection', then a held party cannot be transferred directly from an inquiry — the controlling party must first form a three-party conference, and then hang up, in order to connect the held party to the inquired-to party as a simple call.

- If six-party conference is configured, up to six parties may be conference-in. If the controlling party hangs up during the conference, the conference is disconnected *if all remaining parties are trunks*. If at least one of the remaining parties is local, then the conference remains established. If the conference reaches a state where there is connection between a set on the controlling node and two other trunks, and the set disconnects, then there becomes a simple call connection between the two trunks. In a simple call connection, if one of the remaining parties is external, it becomes the originating party and the other becomes the terminating party. If both parties are external, this implies that at least one of the parties is a set. The set then becomes the originating party, and the other party becomes the terminating party.
- The held party may be transferred to the inquired-to party over a an ISDN/DPNSS1 tandem. DPNSS1 calls may be held by the controlling party in the normal way.
- A call transferred to a party that has answered may be route optimized, upon completion of the transfer. A call transferred to a ringing set may be optimized upon answer. A held call at the originating or terminating node may be optimized upon being established in a simple call.
- Transfer after inquiry has priority over route optimization. If a node receives end-to-end messaging indicating ‘transferred’ after sending end-to-end messaging containing ‘route optimization’, the request for route optimization is aborted.
- Call Forward No Answer may override the DPNSS1 access restrictions placed on transfer after inquiry in the following scenario. An inquiry call is made to a set with Call Forward No Answer active. If a transfer is attempted while the inquired-to set is ringing, messaging is not sent to the controlling node. The DPNSS1 access restrictions are checked between the held party and the forwarding party, and not between the held party and forwarded-to party. If the forwarded-to party answers before the transfer is attempted, or if the call is successfully transferred before it is call-forwarded, then the DPNSS1 access restrictions are properly checked.

- Call Join allows a user of an M1000 series or 2000 series set, to conference into an active call, a party waiting on a secondary DN or the Call Waiting key. The call is then treated as a conference. If the controlling set disconnects during the conference, and if transfer is allowed, the remaining parties remain connected. Notification of the transfer is sent via end-to-end messaging.
- If an inquiry call is made to a busy set with Call Waiting active, the call is placed in call waiting to the inquired-to set. The controlling party, while receiving ringing, may transfer the call.
- If Multi-party Operation is equipped at the controlling node, the controlling party may toggle between the held party and inquired-to party, after the inquired-to party has answered the inquiry call from the controlling party.
- Multi-party Operation is a stand-alone feature, and does not support network-wide misoperation. It does allow local misoperation treatment to be configured for call transfer, for external and internal calls. The options that may be configured are ATN (route to attendant), DAR (disconnect after re-ring cycle of 1-15), AAR (route to attendant after re-ring cycle), OVF (overflow tone), DIS (disconnect), or STD (standard operation, which is disconnect for internal and route to attendant for external).
- DPNSS1 does not support either the Group Hunt or Group Hunt Queuing features.
- DPNSS1 operation will not support the following:
 - ISDN Basic Rate Interface (BRI) setsor
 - ISDN BRI Numeris Terminals.

Feature packaging

— DPNSS1 Network Services (DNWK) package 231

Dependencies:

- Digital Private Network Signaling System No. 1 (DPN) package 123;
- Integrated Digital Access (IDA) package 122, dependent on Digital Trunk Interface (PBXI) package 75, 2 Mbit Digital Trunk Interface (DTI2) package 129;
- International Supplementary Features (SUPP) package 131
- Basic Alternate Route Selection (BARS) package 57, dependent on Basic Routing (BRTE) package 14 and Network Class of Service (NCOS) package 32
- Co-ordinated Dialing Plan (CDP) package 59, dependent on Dependent on Basic Routing (BRTE) package 14 and Network Class of Service (NCOS) package 32
- Pretranslation (PXLT) package 92
- Flexible Numbering Plan (FNP) package 160
- Network Attendant Service (NAS) package 159, for a mixed ISDN/DPNSS1 environment
- If the toggle capacity is required, the Multi-Party Operations (MPO) package 142 for basic functionality, with the Flexible Tones and Cadences (FTC) package 125 and International Supplementary Features (SUPP) package 131 for enhanced functionality
- Incoming DID Digit Conversion (IDC) package 113, dependent on New Flexible Code Restriction (NFCR) package 49 and Network Class of Service (NCOS) package 32

Feature implementation

- **LD10-** Configure the transfer/conference capabilities.
 - *If Multi-Party Operation is not equipped*, in response to the CLS prompt, enter XFA/(XFD) to allow/(deny) transfer, or enter XFA/(XFD) to allow/(deny) three-party conference, or enter XFA or C6A/(C6D) to allow/(deny) six-party conference.
 - *If Multi-Party Operation is equipped*, in response to the CLS prompt, enter TSA/(XFD) to allow/(deny) three-party conference, or enter TSA or C6A/(C6D) to allow/(deny) six-party conference.
- **LD11-** In response to the KEY prompt, enter xx TRN for transfer, xx A03 for three-party conference, or xx A06 for six-party conference (where xx is the defined value).
- **LD95-** For the Calling Party Name Display data, configure the transfer indication mnemonic displayed on the telephone sets.
 - In response to the XFER prompt, enter the four-digit transfer indication mnemonic.

- **LD15-** Configure the Multi-party Operation attributes.
 - In response to the MPO prompt, enter YES.
 - In response to the CCDO prompt, enter YES/NO to allow/deny transfer after inquiry.
 - In response to the RALL prompt, enter YES/NO to allow/deny mandatory recall.
 - In response to the CDTO prompt, enter a value from 2-(14) to define the control digit for time out.
 - In response to the IFLS prompt, enter YES/NO to ignore/allow a switch-hook flash operation.
 - In response to the MHL D prompt, enter YES/NO to indicate whether or not manual hold is required.
 - In response to the PCDS prompt, enter YES/NO to allow/deny the entering of control digits.
 - In response to the CNFD prompt, enter a value from 0-(1)-9, or # or *, to define the control digit for conference.
 - In response to the TGLD prompt, enter a value from 0-(2)-9, or # or *, to define the control digit for toggle.
 - In response to the DISD prompt, enter a value from 0-(3)-9, or # or *, to define the control digit for disconnect.
 - In response to the FMPO prompt, enter YES to define flexible misoperation treatments for Multi-party Operation.
 - In response to the AOCS prompt, enter XXXX (treatment for internal calls) and YYYY (treatment for external calls), where XXXX and YYYY may be ATN (route to attendant), DAR (disconnect after re-ring cycle of 1-15), AAR (route to attendant after re-ring cycle), OVF (overflow tone), DIS (disconnect), or STD (standard operation, which is disconnect for internal and route to attendant for external).

DPNSS1 Loop Avoidance

The DPNSS1 Loop Avoidance feature prevents a DPNSS1 call from being looped through a network, due to errors in configuration, by placing a limit on the number of channels that a call may use.

A Loop Avoidance (LA) Supplementary Information String (SIS) has been added in all outgoing Initial Service Request Messages (ISRM)s, for each call at the originating PBX. The SIS contains a parameter that sets the limit on the number of DPNSS1 transit nodes that a call may use, as defined in the customer data block (overlay 15). The maximum value to which this limit may be defined is 25.

At each Meridian 1 transit node, the parameter of the Loop Avoidance Supplementary Information String is decreased by one, and a check is done to see if the limit has been reached. If the limit has not been reached, an Initial Service Request Message is sent along the outward channel to route the call onward. If the limit has been reached, the call is cleared back to the originating node and the originating exchange receives a Clear Request Message (CRM) message. The request message contains a specific clearing cause of Network Termination and a Loop Avoidance Supplementary Service.

The call is treated as if clearing has occurred due to congestion. If configured, alternative routing using Step Back on Congestion is attempted at the an originating only, if all of the available routes for the call have not been used. If alternative routing using Step Back on Congestion is not available, the treatment that the call receives depends on the originating party.

If the originating party is a non-ISDN trunk, the originating party receives congestion treatment as customer-defined in the customer data block (overlay 15). This may be busy or overflow tone. If the call was routed due to Network Alternate Route Selection (NARS), NARS call blocking intercept treatment is given (either overflow, busy, recorded announcement, or route to attendant). If the originating party is an ISDN trunk, the originating party receives congestion treatment as customer-defined in overlay 15 (busy or overflow).

If the originating party is a local set, treatment depends on the customer-defined congestion treatment (either busy or overflow) or NARS call blocking intercept treatment (either overflow, busy, recorded announcement, or route to attendant). If the originating party is a local attendant, busy indication is given. At this point, the DPNSS1 Attendant Camp-on feature may not be used.

If a Meridian 1 transit node receives an Initial Service Request Message that does not contain a Loop Avoidance Supplementary Information String, before the ISRM is sent over a new channel, a Loop Avoidance Supplementary Information String is added to the ISRM. The Loop Avoidance parameter is set to the pre-defined Loop Avoidance limit (as programmed against the TNDM prompt in the Customer Data Block, overlay 15) less one, to account for the incoming DPNSS1 channel.

The Loop Avoidance Supplementary Information String is ignored at a terminating Meridian 1 node that is not a gateway. If a terminating Meridian 1 node is a DPNSS1 to ISDN gateway, then the call is cleared back if the loop avoidance limit has been reached. If the ISDN Call Connection Limitation (ICCL) feature is equipped, the Loop Avoidance Limit is used to create the ICCL Tandem Threshold Limit in the outgoing SETUP message.

Operating parameters

- The intercept treatment for Network Alternate Route Selection calls that are blocked, configured in overlay 15 in response to the INTR prompt, should be the same as that for calls receiving Loop Avoidance call-back treatment, configured in overlay 15 in response to the CONG prompt.

Feature interactions

- Calls extended by the attendant across a DPNSS1 trunk contain a Loop Avoidance String, with the value of the loop avoidance parameter being customer-defined in overlay 15.

- A Loop Avoidance Supplementary Information String is included in an Initial Service Request Message requesting the following:
 - Camp-on/Call offer
 - Route optimization call set-up
 - DPNSS1 Call Back When Next Used
 - DPNSS1 Call Back When Free
 - DPNSS1 Redirection
 - DPNSS1 Three Party Service inquiry call
- The DPNSS1 Attendant Camp-on feature may not be used following call failure due to loop avoidance.
- DPNSS1 Call Back When Free cannot be used from an originating set receiving overflow as a loop avoidance clear-back treatment.
- After originating a DPNSS1 call, a Meridian 1 will attempt a new call if a Divert Immediate or Busy Instruction is received in a Number Acknowledgment Message (NAM). If the originating item is ISDN containing a Tandem Count value, this value is used to determine the Loop Avoidance Limit of the new DPNSS1 call; otherwise, the Tandem Count value defined in the Customer Data Block, overlay 15, is used.
- The Loop Avoidance Limit configured at an originating DPNSS1 Meridian 1 overrides the Step Back On Congestion configuration at a transit PBX.
- When a call transfer occurs over DPNSS1 links, the held and inquiry segments of the call must not individually exceed the Loop Avoidance parameter limit for the DPNSS1 channels that are used. On completion of the call transfer, the limit may be exceeded.
- Remote Virtual Queuing is not allowed on an ISDN call cleared back due to Tandem Threshold Exceeded.
- If an incoming DPNSS1 or ISDN call is call forwarded all calls on busy over a DPNSS1 or ISDN trunk, the Loop Avoidance Limit of the incoming call is used for the forwarded call.

- If the ISDN Call Connection Limitation (ICCL) feature is equipped, when an ISDN call reaches the terminating Meridian 1 node, it returns the Tandem Threshold count in the ALERT message to the originating node. If an ISDN call encounters a DPNSS1 gateway while being channeled, the complete tandem count is not known since DPNSS1 does not pass this information back to the originating node. Therefore, the ICCL Tandem Threshold count in the ALERT message passed from the DPNSS1 gateway to the originating node is incorrect (the actual value returned is that received at the gateway node, increased by one).

For the outgoing portion of a call, the gateway will use the received value of the Loop Avoidance Supplementary Information String or Tandem Count to adjust the Tandem Count or Loop Avoidance Limit information.

- DPNSS1 operation will not support the following:
 - ISDN Basic Rate Interface (BRI) setsor
 - ISDN BRI Numeris Terminals.
- When an incoming DPNSS1 call to a local station hunts across a DPNSS1 trunk, the Loop Avoidance Limit will be used for the outgoing call to avoid the possibility of a call looping continuously because of the Call Hunt feature.

Feature packaging

- DPNSS1 Network Services (DNWK) package 231

Dependencies:

- Digital Private Network Signaling System No. 1 (DPN) package 123;
- Integrated Digital Access (IDA) package 122, dependent on Digital Trunk Interface (PBXI) package 75, 2 Mbit Digital Trunk Interface (DTI2) package 129;
- International Supplementary Features (SUPP) package 131
- Basic Alternate Route Selection (BARS) package 57, dependent on Basic Routing (BRTE) package 14 and Network Class of Service (NCOS) package 32
- Network Alternate Route Selection (NARS) package 58
- Co-ordinated Dialing Plan (CDP) package 59, dependent on Dependent on Basic Routing (BRTE) package 14 and Network Class of Service (NCOS) package 32
- Pretranslation (PXLT) package 92
- Network Attendant Service (NAS) package 159;
- Flexible Numbering Plan (FNP) package 160
- For the ISDN Call Connection Limitation (ICCL) feature, the following packages are required:
 - ISDN Supplementary features (ISDN SUPP) package 161
 - Network Attendant Services (NAS) package 159
 - Integrated Services Digital Network (ISDN) package 145
 - ISDN Primary Rate Access (PRA) package 146 or ISDN Integrated Services Link (ISL) package 147

Feature implementation

- **LD15-** In the Customer Data Block, define the Loop Avoidance Limit for DPNSS1 calls or the Tandem Threshold Limit for ISDN calls.
 - If the ISDN and ICCL packages are equipped, enter YES to the ISDN prompt. This causes the TNDM prompt to be displayed. In response to the TNDM prompt, enter a value from 0-(15)-31 to define the Tandem Threshold Limit for ISDN calls.
 - If the ISDN and ICCL packages are not equipped, the ISDN prompt does not appear. The TNDM prompt is displayed automatically. In response to the TNDM prompt, enter a value 0-(15)-25 to define the Loop Avoidance Limit for DPNSS1 calls.

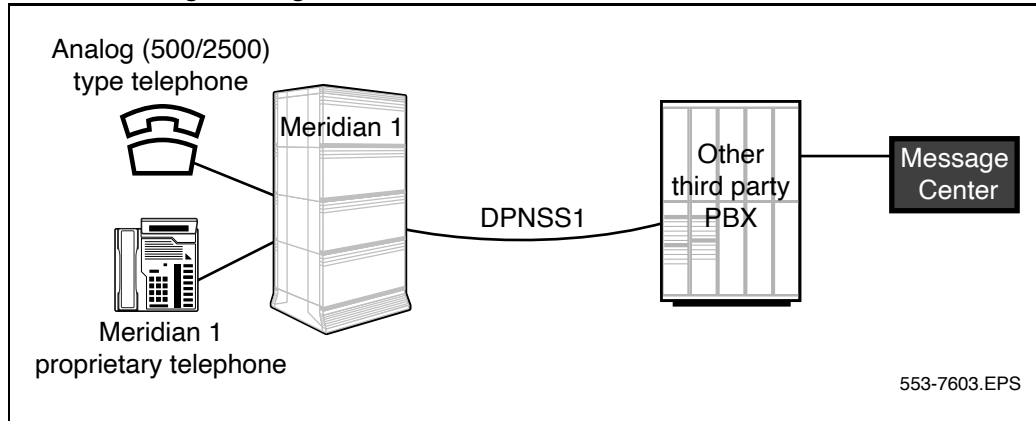
DPNSS1 Message Waiting Indication

With the DPNSS1 Message Waiting Indication (DMWI) feature, Meridian users can subscribe to a third party voice message system across a Digital Private Network Signalling System No. 1 (DPNSS1) network.

When provisioned, this feature provides a means to pass Message Waiting Indication across a private DPNSS1 network with Meridian 1 and other third party PBXs. This feature allows the Meridian 1 to recognize DPNSS1 Non-Specified Information (NSI) from a third party voice message node. This recognition capability allows a voice message system located on another node to notify or cancel Message Waiting Indication for Meridian 1 users.

Figure 109 illustrates DPNSS1 Message Waiting Indication.

Figure 109
DPNSS1 Message Waiting Indication



DPNSS1 Message Waiting Indication interworks with the DPNSS1 Call Diversion feature. The DPNSS1 Call Diversion Feature automatically routes an incoming trunk or an internal call to a third party voice message node if the call is not answered on the Meridian 1 node. When a calling party leaves a message for the called party, the voice message node sends a message waiting notification to the controlling node. When the called party retrieves the voice message, a message waiting cancellation is sent by the host node to the controlling node where the Meridian 1 user is located.

For telephones equipped with a visual message waiting device such as an LCD or LED, message notification is provided by lighting the device and message cancellation by switching off the device. Otherwise, the indication and cancellation is provided by an audible indication when the called party goes off-hook.

Operating parameters

DPNSS1 Call Diversion feature is a prerequisite for the DPNSS1 Message Waiting Indication feature. With DPNSS1 Call Diversion, one of the following redirection features must be configured: Call Forward All Calls, Call Forward No Answer, Call Forward by Call Type, Call Forward Busy, Hunting/Group Hunting, ICP Forward All Calls or Internal Call Forward.

This feature is only supported on analog (500/2500 type) sets and Meridian 1 proprietary sets.

DPNSS1 Message Waiting Indication is supported across Analog Private Network Signaling System (APNSS).

The size of a parameter for a Message Waiting Indication non-specified information (NSI) string is limited to 80 characters. The size of all parameters for a Message Waiting Indication non-specified information (NSI) string is limited to 126 characters. String size limitations do not include octothorpe (#) or asterisk (*) delimiters.

The DPNSS1 Message Waiting Indication does not check the presence and validity of a suffix following a non-specified information identifier.

This feature supports Coordinated Dialing Plan (CDP) and Uniform Dialing Plan (UDP).

The total limit of configured Message Waiting Indication (MWI) Non-Specified Information (NSI) tables must not exceed 512. The size of an MWI NSI consists of adding up the table's number of parameters, the total number of characters for the table's parameters and the number 7. Any creation or change that causes this limit to be exceeded, results in the output of an error message (SCH0097).

Any number of DPNSS1 trunks can be involved in the path between the Voice Messaging System and the Meridian 1. However, if a non-DPNSS1 trunk is involved in this path, then the DPNSS1 Message Waiting Indication feature is not supported. If this occurs, the Message Waiting Indication NSI is not passed at the gateway node.

The calling party, the originating node, and the called party, the controlling node, can be the same. When this occurs, the local call is forwarded to a Voice Messaging System across DPNSS1.

Feature interactions

Network Messaging Service

With the DPNSS1 Message Waiting Indication feature, no gateway functionality between Integrated Services Digital Network (ISDN) and DPNSS exists for Meridian Mail access or message waiting capabilities.

Feature packaging

The DPNSS1 Message Waiting Indication requires DPNSS Message Waiting Indication (DMWI) package 325.

All Meridian 1 nodes require the following packages:

- Integrated Digital Access (IDA) package 122
- Digital Private Network Signaling System (DPNSS) package 123

The Meridian 1 originating node (i.e. node with calling party) and controlling node (i.e. node with Message Center users) require DPNSS1 Network Services (DNWK) package 231 for the DPNSS1 Call Diversion feature.

Meridian 1 controlling nodes require the following packages:

- End-to-End Signaling (EES) package 10
- Message Waiting Center (MWC) package 46

For an audible Message Waiting Indication on analog (500/2500 type) sets, Flexible Tones and Cadences (FTC) package 125 is required.

For a Message Waiting announcement, Message Intercept (MINT) package 163 is required.

Feature implementation

Note: Prior to configuring DPNSS1 Message Waiting Indication, the DPNSS1 Call Diversion feature must be configured and one of the following redirection features must also be activated: Call Forward All Calls, Call Forward No Answer, Call Forward by Call Type, Call Forward Busy, Hunting/Group Hunting, ICP Call Forward All Calls and Internal Call Forward. To configure, refer to the DPNSS1 Call Diversion feature in this guide.

LD 15 – Add, change or delete a Message Waiting Indication NSI table.

Prompt	Response	Description
REQ	NEW	Add new data.
	CHG	Change existing data
TYPE	NET	Networking data block.
CUST	xx	Customer number.
...		
DMWM	YES	Enable output of error messages. (NO) = disables output of error messages (default).
MWNS	YES	Recognize Message Waiting Indication NSI string. (NO)= Do not recognize Message Indication NSI string (default).
- REQ	(NEW)	Create new NSI table (default). OUT = Delete Message Waiting Indication table.

- MFID	a	Enter the Manufacturer Identifier of the Message Waiting Indication NSI table to add, change, or delete, where a = any alpha character or <CR>. Note: SCH9996 message will appear if the command CHG is entered and no Message Waiting Indication NSI tables corresponds to the alpha character entered. When this occurs, the MFID prompt is re-prompted. Note: SCH0097 will appear if the NEW or CHG commands are entered and if the number of MWI NSI tables for the customer exceeds the limit (512). Note: If the prompt XALL is entered, then all existing Message Waiting Indication NSI tables are deleted.
- - NOTI	YES NO	YES = NSI string for Message Waiting Notification. If NO or <CR> is entered on NEW command then the SCH0274 message is output. If NO or <CR> is entered on CHG command then CANC is prompted.
- - MSSC	a	Note: Manufacturer specific service character for MW notification where a = any alphanumeric character is accepted for an SIS parameter. If <CR> is entered on NEW command then the SCH0274 message appears and MSSC is re-prompted. If <CR> is entered on CHG command then PRMT prompt appears. Note: If a = a character that is not an alphanumeric character, then SCH008 appears and MSSC is reprompted.
- - - PRMT	aaa	NSI parameter(s) for Message Waiting Notification, where aaa = any alphanumeric sequence is accepted for a SIS parameter to a maximum of 126 characters. PRMT appears until <CR> is entered. Note: If aaa includes a character that is not an alphanumeric character, then SCH008 appears and PRMT is reprompted.
- - CANC	YES NO	YES = NSI string for Message Waiting Cancellation. If NO or <CR> is entered on NEW command then the SCH0274 message appears and CANC is re-prompted. If NO or <CR> is entered on CHG command then the MFID prompt appears.

- - - MSSC	a	Manufacturer specific service character for Message Waiting cancellation where a = any alphanumeric character is accepted for an SIS parameter.
- - - PRMT	aaa	NSI parameter(s) for Message Waiting Cancellation where aaa = any alphanumeric sequence is accepted for an SIS parameter to a maximum of 126 characters. PRMT appears until <CR> is entered. Note: When REQ = CHG, both cancellation and notification, once <CR> is entered at the PRMT prompt, the only parameters kept are the ones that have just been entered. Any existing parameters not re-entered are removed from the MWNS.

LD 10 – Allow Message Waiting Class of Service.

Prompt	Response	Description
REQ:	NEW CHG	Add new data. Change existing data.
TYPE:	500	Type of telephone set.
TN	c u	Terminal Number.
...		
CLS	MINA	Message Interrupt Allowed MIND = Message Interrupt Denied (default).
CLS	MWA	Message Waiting Allowed. MWD = Message Waiting Denied (default).

Note: To receive an announcement as a message waiting indication, analog (500/2500 type) sets must configure the Message Intercept feature and activate Flexible Tones and Cadences (FTC) in LD 56.

LD 56 – Message Intercept and Flexible Tones and Cadences.

Prompt	Response	Description
REQ	NEW, CHG	Add new data. Change existing data.
TYPE	FTC	Flexible Tones and Cadences data block.
TABL	0 - 31	Flexible Tones and Cadences Table number.
...		
MINT	YES	Allow tones or announcements. NO = Deny tones or announcements (default).
- MWAN	0 - 255 0 - 255	Message Waiting.

Note: If the Message Intercept feature is not equipped, a Message Waiting dial tone is provided on a set basis if this tone has been defined in Tones and Cadences data block in Overlay 56. Or, Call Forward Message Waiting tone is provided if Call Forward Message Waiting has been defined in Overlay 56 and the set has Call Forward Active.

LD 11 – Allow Message Waiting Class of Service.

Prompt	Response	Description
REQ:	NEW CHG	Add new data. Change existing data.
TYPE:	xxxx	Telephone type, where: xxxx = SL1, 2006, 2008, 2009, 2016, 2018, 2112, 2216, 2317, 2616, or 3000.
TN	c u	For Option 11C.
...		
CLS	MWA	Message Waiting Allowed. (MWD) = Message Waiting Denied (default). If CLS = MWA and no Message Waiting Key (MWK) is defined, then broken dial tone is provided for message waiting notification.

Feature operation

No specific operating procedures are required to use this feature.

DPNSS1 Night Service

The Digital Private Networking Signaling System No.1 (DPNSS1) Night Service feature introduces the “Diversion via a Different Channel” capability of the DPNSS1 Night Service Supplementary Service. That is, it allows a Meridian 1 to treat a third-party PBX’s request to divert a call queued to an attendant that is in Night Service mode, back to the local attendant queue of the originating DPNSS1 node.

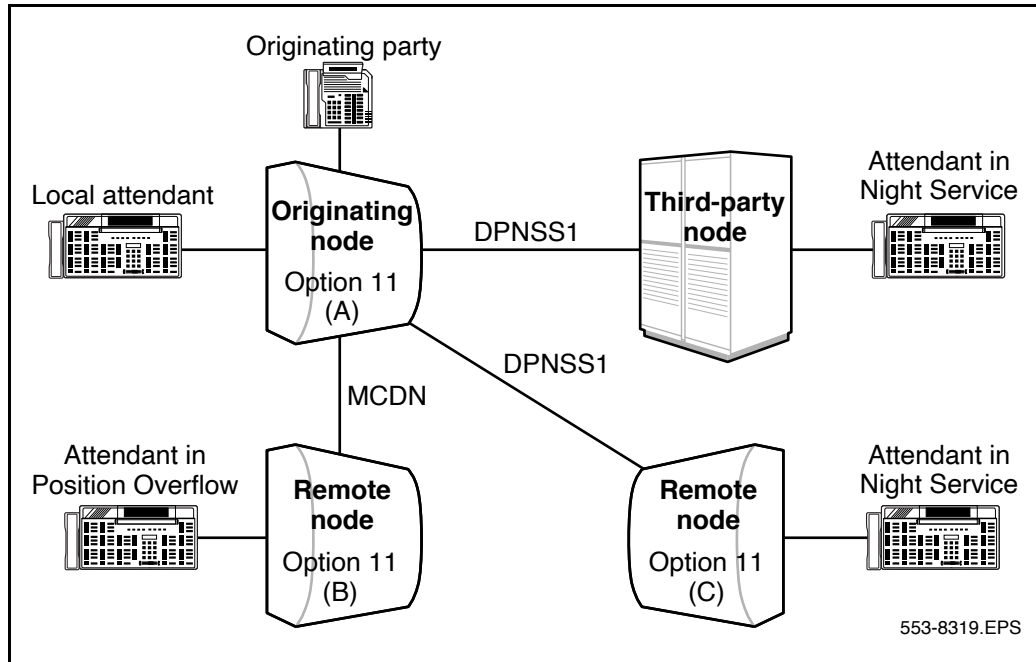
The following example illustrates a DPNSS1 Night Service call processing scenario. Also refer to Figure 110.

A DPNSS1 call from the originating Meridian 1 node (M1A) terminates to the attendant on a third-party PBX. The attendant is in Night Service. The third-party PBX signals the Meridian 1 to initiate Night Service Diversion. The call is then diverted back to the originating node, where a new call is initiated to the queue of the local attendant.

Note: This diversion is the functionality that has been introduced by the DPNSS1 Night Service feature. The call processing which follows is part of the standard Network Attendant Service (NAS) functionality.

At this point, the call is treated as a standard call to the local attendant. If the local attendant is also in Night Service, Network Attendant Service (NAS) routing is applied. The call is routed to a remote attendant (on M1B.) Since this attendant is in Position Overflow, it cannot take the call and clears it. The next alternative in the NAS routing table is tried, which is for the originating Meridian 1 to route the call to the remote attendant (M1C). Here, the attendant is also in Night Service and clears the call. Eventually, the Night DN is tried successfully. The new call from the originating Meridian 1 to the NIGHT DN is kept, and the old call to the third-party PBX is released.

Figure 110- Example of DPNSS1 Night Service Diversion



Operating parameters

None applicable.

Feature interactions

DPNSS1 Redirection

A redirected call may undergo Night Service Diversion, if a new call is attempted to an attendant on a third-party PBX that initiates Night Service Diversion.

DPNSS1 Route Optimisation

Route Optimisation is applied if a non-optimum path has been taken by a call answered by either the third-party PBX on which the target operator is located, the local attendant, remote attendant, or the Night DN.

DPNSS1 Step Back on Congestion

If a call to the remote attendant encounters congestion, Step Back on Congestion is initiated and attempted at any node.

DPNSS1 Extension Three Party Service

An enquiry call reaching an attendant in Night Service will undergo Night Service diversion, if available.

Diversion

A diverted call reaching an attendant in Night Service will undergo Night Service diversion, if available.

Attendant Incoming Call Indicators

When a Night Service call is diverted to an attendant, the Incoming Call Indicator is the number of the incoming route (this is the same as for a NAS MCDN call routed to an attendant.)

Call Waiting

If a call is diverted to a third-party operator Night DN that is busy, Call Waiting may be activated (if equipped). The call to the third-party operator PBX is released.

Feature packaging

The following software packages are required for the DPNSS1 Night Service feature:

For basic DPNSS1 network functionality:

- Integrated Digital Access (IDA) package 122;
- Digital Private Networking Signaling System No.1 (DPNSS) package 123;
- 2.0 Mbps Primary Rate Interface (PRI2) package 154.

For enhanced functionality:

- International Supplementary Features (SUPP) package 131;
- DPNSS1 Networking Services (DNWK) package 231.

For Network Attendant Service interworking:

- Integrated Services Digital Network (ISDN) package 145;
- Advanced ISDN Network Services (NTWK) package 148;
- Network Attendant Service (NAS) package 159;
- ISDN International Features (ISDN INTL SUP) package 166 (to support the MCDN/DPNSS1 gateway with Loop Avoidance.)

Feature implementation

The following steps are required to configure the DPNSS1 Night Service feature:

LD 15 - Configure the local attendant DN.

Prompt	Response	Description
REQ	CHG	Change the existing data.
TYPE	ATT	Attendant consoles data.
CUST	0-31	Customer Number.
- OPT	aaa	Options
- ATDN	(0)-xxxx(xxx)	Four-digit Attendant Directory Number (up to seven digits with the Directory Number Expansion (DNXP) package 150.)
...		

LD 15 - Configure the Night Service DN.

Prompt	Response	Description
REQ	CHG	Change the existing data.
TYPE	NIT	Night Service data.
CUST	0-31	Customer Number.
- NIT1	xxxx	First Night Service DN.
...		

LD 86 - Define the Remote Attendant data.

Prompt	Response	Description
REQ	CHG	Change the existing data.
CUST	0-31	Customer Number.
TBL	(0)-63	NAS routing table. 0 is the customer routing table; it is also associated with Attendant Console Group 0.
ALT	1-7	Attendant Alternative number.
ID	x...x	Digits (up to 16) dialed to reach a remote attendant.
TODS	1-31	Schedule period to be changed.

DPNSS1 Redirection

The DPNSS1 Redirection feature allows a DPNSS1 call that is extended by an attendant and not answered after a defined period of time, to be recalled to an attendant. This attendant may be the attendant that originally extended the call, or another attendant on the same or different node within the network.

Note: The DPNSS1 Redirection feature is required for DPNSS1 networks using a Centralized Operator Service, if the network nodes on which operator consoles are located use DPNSS1 Redirection to provide timed operator recall functionality. If operator consoles are located on a Meridian 1 PBX, timed operator recall is provided by the DPNSS1 Timed Recall feature described in this section.

When an attendant extends a call to a destination, and the destination does not answer before the attendant releases the call, information is passed to the originating DPNSS1 node to initiate recall timing. If the information indicates that the destination is free, then the slow answer recall timer is started. If the information indicates that the destination is busy, then the camp-on recall timer is started. For camp-on timing, if the destination party becomes free before the camp-on timer expires, then the destination party receives ringing. The camp-on timer is canceled, and the slow answer recall timer is started.

If the destination answers the call extension before the recall timer expires, the recall timer is canceled and the source and destination are connected. If the recall timer expires before the call extension is answered, a new call is initiated to the local attendant. If the local attendant is not available, Network Attendant Service (NAS) routes the call to another node. If the call reaches a state of attendant receiving buzzing, attendant receiving ringing, or queued to attendant, then the originating party is connected to the new call and the original call is dropped.

If a new call cannot be established, a Clear Request Message (CRM) is sent to the originating node and the original call remains connected. If the original call, while in call waiting or camp-on, is answered by the destination party before the new call has attained ringing state is, a Call Connected Message (CCM) is sent to the originating node. The new call is cleared and the original call remains connected.

Operating parameters

- The DPNSS1 Three Party Service must be equipped in order for the Redirection feature to function, since the Redirection feature uses the Three Party Service messaging to perform recall timing.
- Special care must be taken when configuring NAS routing for call redirection. If NAS routing is to be used to make the redirected call, a Location Code (LOC) or Distant Steering Code (DSC) must be used and entered in response to the ID prompt in overlay 86. The digits entered for the ID prompt in must allow the call to be routed immediately, without any timing. It is strongly suggested that separate DSCs be used for programming the NAS alternatives.

Flexible Numbering must not be used for the configuration of the NAS alternatives. The prompt FLEN should be given a value of “0” in overlay 87 for the DSC and in overlay 90 for the LOC.

Since there is no system verification during configuration, it is up to the craftsperson to ensure proper programming. If these guidelines are not followed, when the new call is attempted, it will be dropped and the old call retained.

Feature interactions

- The Redirection feature does not apply to calls passing through an DPNSS1/ISDN gateway. If a call comes in from an originating node over an ISDN trunk, passes through a Meridian 1 gateway PBX, is routed to an attendant over a DPNSS1 trunk, and is then extended to a set over a DPNSS1 trunk, then the Redirection feature may only initiate recall timing at the ISDN/DPNSS1 boundary.
- The destination party must be within the DPNSS1 or DPNSS1/ISDN network in order for recall timing to be activated at the originating node.
- If the destination party to which call waiting or camp-on is applied is on a non-Meridian 1 node, or on a node that does not treat call waiting as does a Meridian 1 node, then it may not be possible to distinguish a call waiting call from a camp-on call. In this case, the call is timed as it were a camp-on call.
- If an attendant at one node is established in a call to an attendant at another node, this feature does not apply if the second attendant transfers the call.

- If Attendant Forward No Answer is active, a call that has been redirected to an attendant may be passed from one console to another, if the call has been presented but not yet answered. The previous console is placed in night service. If a call is called is passed to the last console which is service, the call is passed from this console to the night DN.
- If a call is extended from an attendant node that relies on the originating node for recall timing using the Redirection feature, to a ringing set on a Meridian 1 node with Call Forward No Answer (CFNA) active, the recall timing takes precedence over the CFNA timing. When the call is extended to the set, the recall timer is started at the originating node. When the set begins to ring, the CFNA timer is started. If the CFNA timeout is less than the recall timer timeout, then the call is forwarded to the CFNA DN. The CFNA DN is rung until the recall timer expires, at which time the CFNA DN stops ringing and the call is routed to the attendant. If the CFNA timeout is greater than the recall timer timeout, then, when the recall timer expires, the set ceases to ring and the call is routed to the attendant rather than to the CFNA DN.
- Redirection timing is not done at a Meridian 1 DPNSS1 originating node for DPNSS1 calls transferred from sets.
- DPNSS1 Loop Avoidance string (LA) added for a normal call is also added to the redirected call.
- If a redirected call encounters congestion, the DPNSS1 Step Back on Congestion feature, if active, may cause the call to step back. Another call may be redirected using an alternate, non-congested route.
- During system initialization, calls not yet established are dropped.
- After the Redirection recall timer expires, recalls to the attendant leave only the source active, with the destination being dropped. Therefore, there is no splitting with the Redirection feature.

- The Slow Answer Modification feature may be used in a mixed network environment consisting of attendant nodes that do their own recall timing, and nodes using the Redirection feature for recall timing. This application would result in a more consistent console operation within the network. Where recall timing is done by the attendant node, when a recall occurs to the attendant, the Slow Answer Modification feature causes the destination to be dropped when the attendant answers the recall. Where recall timing is done by the Redirection feature, when a recall occurs to the attendant, the source remains active while the destination is dropped.
- After the Redirection recall timer expires, a call extended by an attendant may or may not recall to the attendant that originally extended the call, since the original call is dropped and a new call is originated.
- DPNSS1 does not support either the Group Hunt or Group Hunt Queuing features.
- DPNSS1 operation and features are not supported with ISDN Basic Rate Interface (BRI) or BRI trunks.

Feature packaging

- DPNSS1 Network Services (DNWK) package 231

Dependencies:

- Digital Private Network Signaling System No. 1 (DPN) package 123;
- Integrated Digital Access (IDA) package 122, dependent on Digital Trunk Interface (PBXI) package 75, 2 Mbit Digital Trunk Interface (DTI2) package 129;
- International Supplementary Features (SUPP) package 131
- Basic Alternate Route Selection (BARS) package 57, dependent on Basic Routing (BRTE) package 14 and Network Class of Service (NCOS) package 32
- Co-ordinated Dialing Plan (CDP) package 59, dependent on Dependent on Basic Routing (BRTE) package 14 and Network Class of Service (NCOS) package 32
- Pretranslation (PXL) package 92
- Network Attendant Service (NAS) package 159
- Flexible Numbering Plan (FNP) package 160

Feature implementation

- **LD15-** In the Customer Data Block, define the parameters for recall timers and the attendant DN.
 - In response to the ATDN prompt, enter the attendant DN. Recalls occur to this DN, upon expiration of the recall timer.
 - In response to the RTIM prompt, enter a value for the slow answer recall timer (0-(30)-378), camp-on recall timer (0-(30)-510), and call waiting recall timer (0-(30)-510). Note that for recalls timed at the local node, no distinction is made between call waiting calls and slow answer recalls. The slow answer value is used in both cases.

DPNSS1 Route Optimization

The DPNSS1 Route Optimization feature has been developed to optimize trunk usage within a DPNSS1 network, by replacing non-optimum call paths through a DPNSS1 private network with optimum paths. An optimum path is the path that uses only the first choice routes to link two PBXs across the network. The first choice is determined by the programming of the network numbering and routing at each PBX. This optimization applies to established simple voice calls which were routed during set-up, or transferred or attendant-extended to another party.

Route optimization is initiated by the originating PBX, after recognizing that a DPNSS1 call may have been set up over a non-optimum path due to alternative routing or call modification. If the call is ringing, the originating PBX waits for an answer signal before initiating optimization. If the call has been transferred, on answer, or attendant-extended to a another party, then the transfer or extension signaling sequence initiates the optimization.

The originating PBX sends a Route Optimization Request message, which contains a Call Reference Number (CRN) field, to the terminating PBX. The CRN is used as a destination address to route the call back to the originating PBX and uniquely identify the call being optimized (the Originating Line Identity sent in the Initial Service Request Message is used for this purpose). The set-up message for the backward call contains a field that identifies the call set-up as route optimization. This causes the call, throughout its path, to be restricted to only first choice routes.

If the route optimization request call set-up successfully gets back to the originating PBX, a conference is established at the originating node between the originating party, the original path still carrying the speech, and the silent new path. A message of acknowledgment is returned to the terminating PBX on the new path. Upon receiving this acknowledgment, the terminating PBX replaces the old path with the new (optimized) path, and sends a connect indication across the new path to the originating PBX. The old path is silenced. Upon receiving the connect indication, the originating PBX terminates the conference, connects the originating party to the optimized path, and clears the original path.

If the route optimization request call set-up fails, the originating PBX receives a notification message that the route optimization request was not successful. The originating PBX may then attempt route optimization again, at 60 second intervals. During this interval, the Meridian 1 may initiate route optimization requests for other DPNSS1 calls.

A customer may define the following route optimization options in overlay 15, the Customer Data Block:

- NRO (no route optimization). Route optimization is inhibited for all calls (this option would typically be used on PBXs having high levels of call traffic).
- ROA (route optimization for alternatively routed calls). Route optimization is initiated for calls which have undergone alternative routing. A call is considered to be alternatively routed if it originated over a route which was not the first choice route, or if alternative routing indication is sent in the Routing Information (RTI) of a Network Indication Message (NIM).
- ROX (route optimization for transferred calls). Route optimization is initiated for transferred or attended-extended calls.
- RAX (route optimization for transferred or alternatively routed calls). Route optimization is initiated for calls which have been alternatively routed, such as by Step Back on Congestion, or for transferred or attended-extended calls.

Operating parameters

- While a PBX may response to simultaneous requests for route optimization, only one call at a time may be optimized from any PBX (this is to prevent ambiguity as to which call is being optimized if a route optimization request was simultaneously made for two or more calls on the same DN of a multiple appearance DN).
- Care must be taken when configuring the incoming and outgoing digit manipulation for the Meridian 1, so that when the insert (INST) digits followed by the Call Reference Number (CRN) are dialed at the terminating PBX, then the call is routed back to the originating PBX.

- Some special configuration needs have to be considered for the optimization of incoming trunk calls. If the Coordinated Dialing Plan (CDP) uses Local Steering Codes (LSCs), then the prompt LSC has to be configured in overlay 15. If the CDP uses only Distant Steering Codes (DSCs) as part of the DNs, then a Trunk Steering Code (TSC) has to be configured at each network node, for each network non-DPNSS1 trunk in the network.
- For a Coordinated Dialing Plan (CDP) configuration, each steering code (Distant or Trunk) has to be defined, in overlay 87, with a Flexible Numbering Plan (FLEN) prompt other than 0 in order to have route optimization working.
- Route optimization may be applied on a private line, which may cause the private line being removed from a call and replaced by another trunk. This may likely occur when a call is being transferred. It is recommended that a network is not configured to have calls alternatively routed to private lines or alternatively routed after using private lines.
- When defining a numbering plan, the insert (INST) digits followed by the Call Reference Number (CRN) should exactly represent the digits to be dialed to reach the DN represented by the Originating Line Identity (OLI) in the Route Optimization Request message.

Feature interactions

- Route optimization is only supported on DPNSS1/APNSS trunks. If a call from a non DPNSS1/APNSS trunk comes in to a set within a DPNSS1 network, the call takes the optimum path (if route optimized) from the non DPNSS1/APNSS trunk to the set.
- Access restrictions placed on sets give them pretranslation, which prevents the sets from dialing certain numbers (a different DN is substituted for the dialed DN). When implementing route optimization, access restriction must not be set up to substitute a dialed DN with another DN that would prevent optimization. The terminating PBX must be allowed to originate a call to the originating PBX.
- Break-in is not allowed during route optimization, and route optimization is not allowed during a break-in. After break-in has ended for a call, route optimization may be applied to the call if it is eligible.

- Call Detail Recording (CDR) records are not printed at the originating or terminating PBX, during route optimization. CDR records are printed at tandem nodes when the non-optimum path is released. The CDR records contain the same information as if the call had occurred on the new path at the time that the original trunks were seized. The cost of the call (that is, the Periodic Pulse Metering information) that has been optimized is the sum of the cost before route optimization plus the cost after optimization. The originator of the original call is shown as the originator of the new call, at the originating PBX. The terminator of the call is shown as the terminator of the new call, at the terminating PBX. At transit PBXs, normal information is printed, showing original tandem connections being released as if for calls being cleared at the time of route optimization, and new tandem connections being released as if for calls being originated at the time of route optimization.

If an optimized call does not use any trunks, that is, the originating party and terminating party are on the same PBX, then CDR records show the call as being cleared as normal.

- A call that has been call-forwarded may be optimized upon being answered only if it has undergone alternative routing. If the forwarded call was not alternatively routed, it may have used a non-optimum path.
- A call that has been picked up or that has undergone hunting may be optimized upon being answered only if it has undergone alternative routing.
- A Ring Again new call may be optimized only if it has undergone alternative routing.
- A call transferred to another party may be optimized only after the call transfer has been completed. A call transferred to a ringing set may be optimized only after being answered.
- A call that has been rerouted due to Step Back on Congestion may be optimized after it is answered.
- During a group hunt, a call to a Pilot DN which has been defined as a trunk access code may be optimized upon being answered only if it has undergone alternative routing.
- A call which is camped-on or call-waiting to a set may not be optimized until the call is answered on the set

- Route optimization may be applied to a call that is being overridden only after it becomes a simple call.
- DPNSS1 calls in the ringing state are optimized immediately upon being answered. Transferred calls, on answer, are optimized as soon as the call transfer has been completed.
- Route optimization cannot be applied to the following calls:
 - data calls;
 - conference calls (however, route optimization may be applied when the conference call reverts to a normal two-party connection);
 - calls on hold;
 - attendant-originated calls.
- Single channel working is not supported on Meridian 1.
- If the conference tone is not switched off on the conference card, the parties involved in the call may hear conference tone during the optimization sequence.
- During a route optimization attempt, the originating PBX and terminating PBX do not initiate signaling for any other DPNSS1 supplementary service for the call.
- During a route optimization attempt, any key operation from a set involved in the call is ignored, except the release or onhook function. If a set not involved in a call is configured in a single call multiple appearance DN arrangement with a set involved in a route optimization attempt, then any key operation that interferes with the route optimization attempt is ignored. Therefore, the set is inhibited from joining the call during the route optimization attempt.
- After system initialization, conference calls are lost. Thus, route optimization may cause some established calls over non-optimum paths to be lost. Also, after system initialization, all route optimization requests are dropped at the PBX where the initialization has occurred. If the requesting party is not on this PBX, the requesting party is not informed that the request has been dropped.

- Pretranslation may be used with route optimization. The stored Call Reference Number (CRN) and the insert (INST) digits are pretranslated by the Initial Service Request Message (ISRM) before being sent, as if being pretranslated after been dialed by terminating party. Similarly, the Destination Address (DA) digits at the terminating PBX are pretranslated as if being dialed by the called party.
- Incoming Digit Conversion is not applied to the INST and CNR digits sent in the Route Optimization call set-up message. This interaction is intended to prevent the CNR digits from being corrupted by Incoming Digit Conversion.
- It is possible to configure Trunk Barring (TBAR) to prevent trunk-to-trunk connections on a local node. If a trunk call has tromboned over the network to another local trunk, the call will not be optimized if the TBAR configuration restricts the local connection.
- DPNSS operation and features are not supported over ISDN Basic Rate Interface (BRI) or BRI trunks.

Feature packaging

- DPNSS1 Network Services (DNWK) package 231

Dependencies:

- Digital Private Network Signaling System No. 1 (DPN) package 123;
- Integrated Digital Access (IDA) package 122, dependent on Digital Trunk Interface (PBXI) package 75, 2 Mbit Digital Trunk Interface (DTI2) package 129;
- International Supplementary Features (SUPP) package 131
- Basic Alternate Route Selection (BARS) package 57, dependent on Basic Routing (BRTE) package 14 and Network Class of Service (NCOS) package 32
- Co-ordinated Dialing Plan (CDP) package 59, dependent on Dependent on Basic Routing (BRTE) package 14 and Network Class of Service (NCOS) package 32
- Pretranslation (PXLT) package 92
- Flexible Numbering Plan (FNP) package 160

Feature implementation

- **LD15-** In the Customer Data Block, configure the route optimization options.
- In response to the ROPT prompt, enter (NRO)/ROA/ROX/RAX, where NRO is used to inhibit route optimization, ROA is used to initiate route optimization for alternatively routed calls, ROX is used to initiate route optimization for calls which have been transferred or attendant-extended, and RAX is used to initiate route optimization for alternatively routed calls, or for calls which have been transferred or attendant-extended.

DPNSS1 Route Optimisation/MCDN Trunk Anti-Tromboning Interworking

The Digital Private Networking Signalling System No.1 (DPNSS1) Route Optimisation (RO)/Meridian Customer Defined Networking (MCDN) Trunk Anti-Tromboning (TAT) Interworking feature provides RO and TAT interworking at DPNSS1/MCDN gateway nodes.

Note: For detailed information on the DPNSS1 Route Optimisation feature, please refer to the *DPNSS1 Product Information Guide* and the *DPNSS1 Administration Guide*. For detailed information on the Trunk Anti-Tromboning feature, please refer to the *International ISDN PRI features description and administration* (for 2.0 Mbit applications), or the *ISDN Primary Rate Interface description and administration* (for 1.5 Mbit applications.)

RO/TAT interworking scenarios

RO/TAT interworking within a DPNSS1 to MCDN gateway

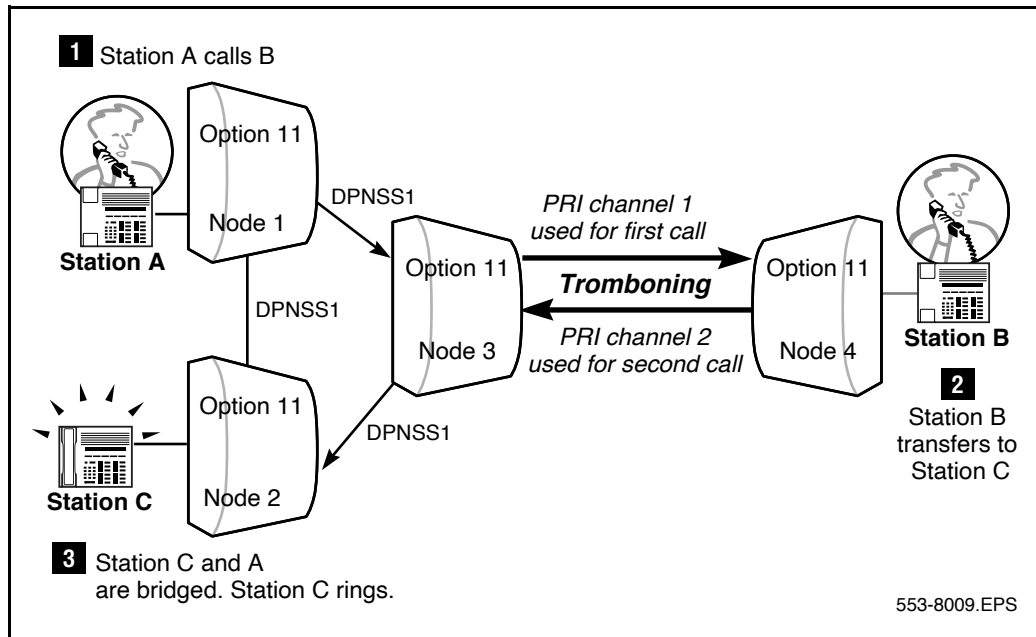
The following example presents a case where RO/TAT interworking occurs within a DPNSS1 to MCDN gateway.

Note: In this example, we have used the case where a call has been redirected due to Network Call Transfer. The same functionality would apply if the call had been redirected by Network Call Forward No Answer, and Network Hunting, or modified by Network Call Transfer or Attendant Call Transfer.

Referring to Figure 111, Station A, located at Node 1 on the DPNSS1 side of the DPNSS1/MCDN gateway, calls Station B located at Node 4 on the MCDN side of the gateway. It is to be assumed that the optimum DPNSS1 route has been selected at the originating node (the case where a non-optimum route is selected is discussed in the note following Figure 112.) Station B activates Network Call Transfer to Station C, located at Node 2 on the DPNSS1 side of the gateway.

Upon activation, the existing call is put on hold and a new call is originated to Station C. Station C Answers. Station B completes the call transfer, leaving A connected to C using two DPNSS1 trunks and two PRI trunks.

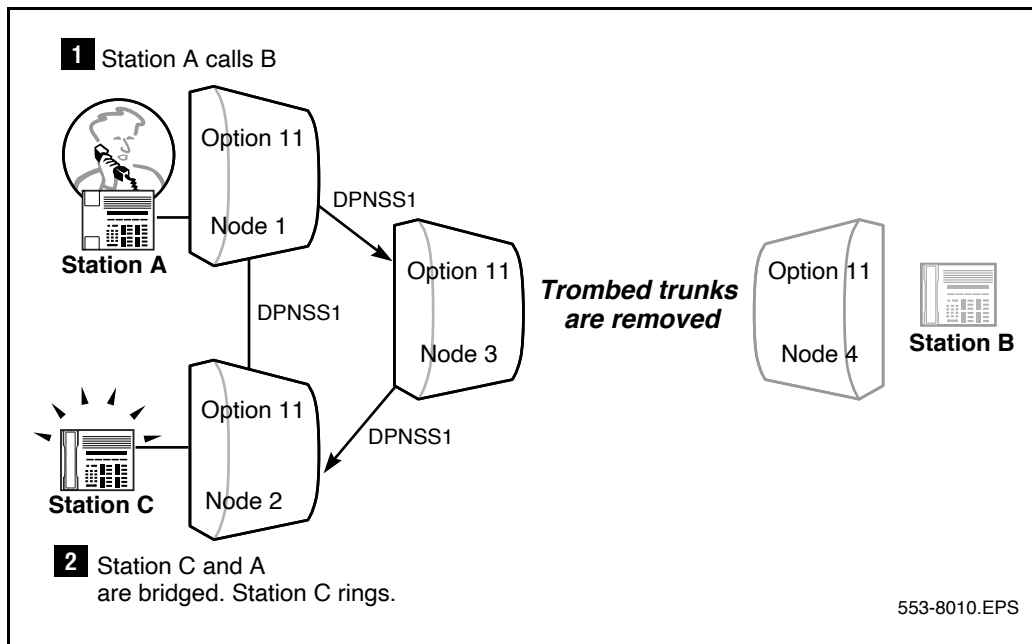
Figure 111
DPNSS1/MCDN scenario with Network Call Transfer, before RO/TAT optimisation



Note: The Network Call Transfer/Three Party Service gateway is not supported at the gateway Node 3. Therefore, RO is not initiated at Node 1, and the non-optimised DPNSS1 trunks remain connected.

On the MCDN side, TAT is initiated at Node 4. The call between A and C is bridged, and the redundant PRI trunks are removed between Node 4 and Node 3. For the meantime, the non-optimised DPNSS1 trunks remain connected, as shown in Figure 112.

Figure 112
DPNSS1/MCDN RO/TAT Interworking scenario, after TAT has been applied

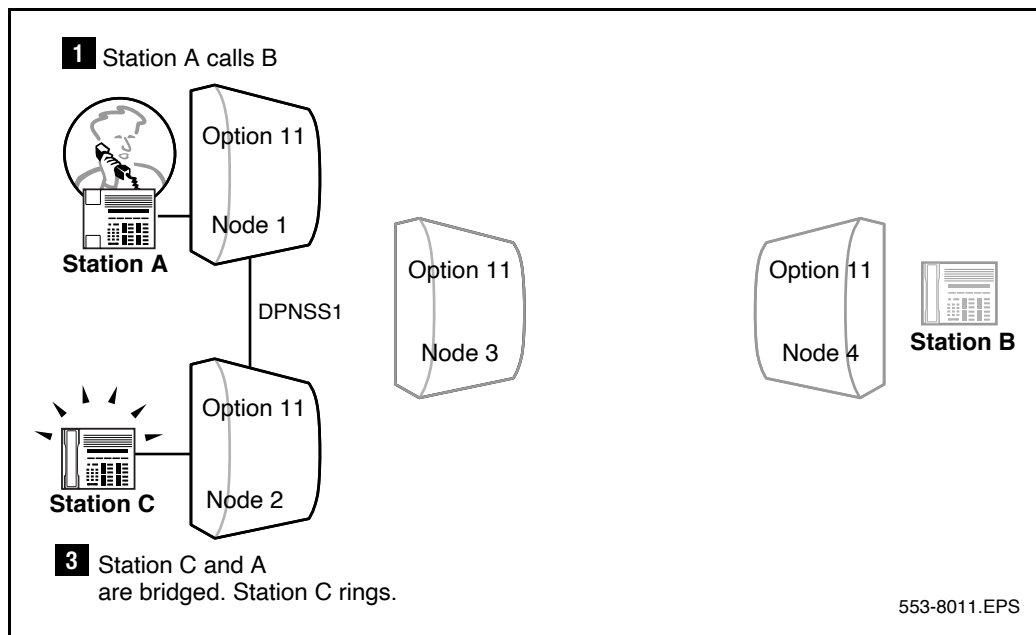


When TAT is completed on the MCDN side, The RO/TAT Interworking feature initiates RO on the DPNSS1 side by simulating a transfer at the gateway Node 3. The Three Party Service feature initiates signaling to update displays. Then, RO is initiated at Node 1, the originating node. The DPNSS1 trunks are dropped between Node 3 and 2 and Node 3 and Node 1, with Station A and Station C being connected over one DPNSS1 trunk. This is shown in Figure 113.

Note: If a non-optimum route is used at the originating node or at any transit node, Route Optimisation may start from Node 1 (the normal RO operation for the first call optimisation) or Node 3 (the normal RO operation for the second call optimisation), before TAT is completed. If TAT invocation is received on Node 3 while RO is being applied between Node 1 and Node 3 or Node 3 and Node 2, the completion of TAT is delayed until RO is totally finished.

Upon the completion of TAT on Node 3, a call transfer operation is simulated, and a new RO operation is initiated to remove any potential triangulation of routes.

Figure 113
DPNSS1/MCDN RO/TAT Interworking scenario, after RO has been applied



Note: If Station A is an attendant, TAT takes place on the MCDN side of the gateway but RO cannot take place on the DPNSS1 side. This is a RO limitation.

RO/TAT interworking within a DPNSS1 to MCDN gateway

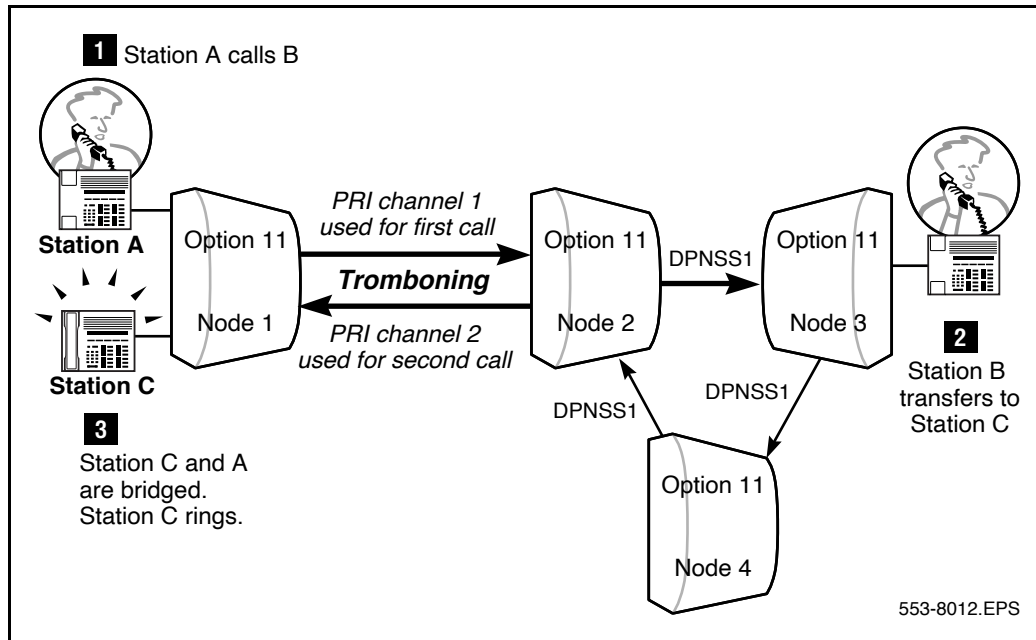
The following example presents a case where RO/TAT interworking occurs within an MCDN to DPNSS1 gateway. Here, too, we are using the case of a call being transferred (using the DPNSS1 Three Party Service feature) across the gateway.

Referring to Figure 114, Station A, located at Node 1 on the MCDN side of the MCDN/DPNSS1 gateway, calls Station B located at Node 3 on the DPNSS1 side of the MCDN/DPNSS1 gateway. Station B transfers the call (using the Three Party Service feature) to Station C, also located at Node 1 on the MCDN side of the gateway.

Upon activation, the existing call is put on hold and a new call is originated to Station C.

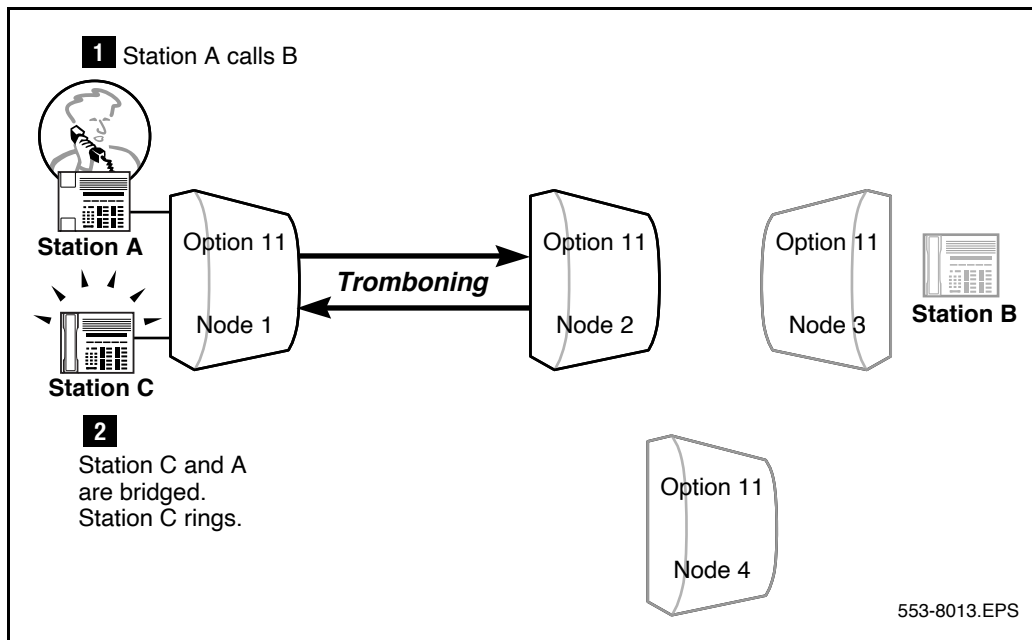
Station C Answers. Station B completes the call transfer, leaving A connected to C using three DPNSS1 trunks (in the example, the call is routed through Node 4) trunks and two PRI trunks.

Figure 114
MCDN/DPNSS1 RO/TAT Interworking scenario, before RO has been applied



Once Three Party Service messaging has taken place, Node 2 initiates RO. The initial DPNSS1 routes are cleared. Node 2 becomes a MCDN/MCDN transit node, and the two tromboning PRI routes between Node 2 and Node 1 remain, as shown in Figure 115.

Figure 115
MCDN/DPNSS1 RO/TAT Interworking scenario, after RO has been applied

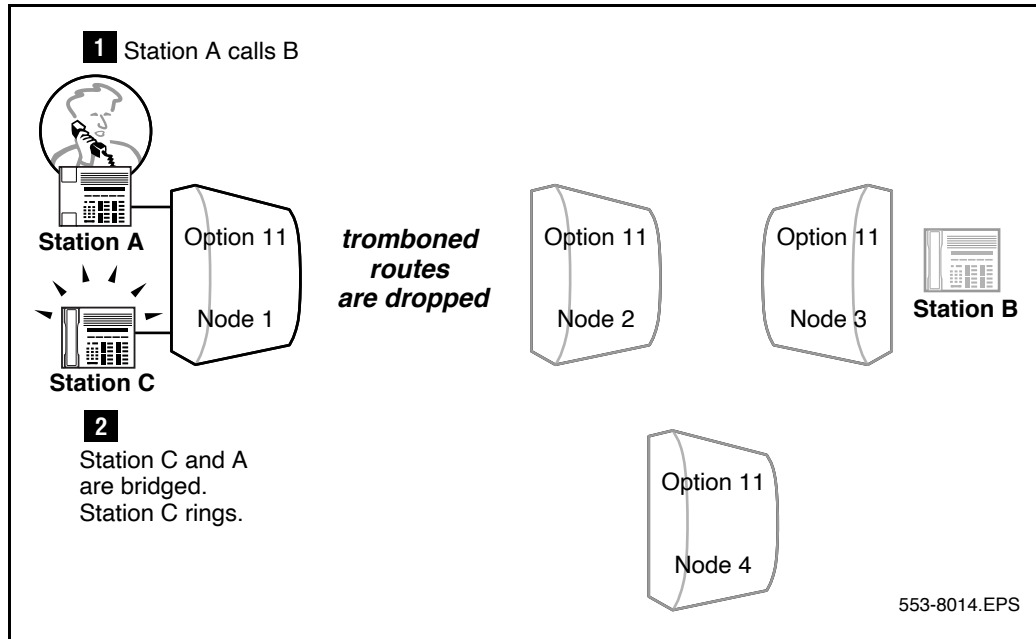


As soon as RO is completed, the RO/TAT initiates TAT at gateway Node 2. After TAT has been completed at Node 1, Node 2 simulates a transfer message to both Station A and Station C. This allows the Network Call Redirection feature to update the displays.

Note: If the originating and terminating nodes are one and the same, and if this node is not a tandem node, as is the case for Node 1 in our example, the displays are updated without the notification from the Network Call Redirection feature.

TAT is then completed. The redundant routes are cleared, and Station A and Station C are bridged, as shown in Figure 116.

Figure 116
MCDN/DPNSS1 RO/TAT Interworking scenario, after TAT has been applied



Note 4: If Station A is an attendant, and the Network Attendant Service feature is configured, Station B cannot transfer to Station C, and no optimisation can take place. If NAS is not configured, Station B may transfer to Station C, and optimisation takes place as described in this example.

Note 5: In the case of call diversion on the DPNSS1 side (Diversion Immediate, Diversion on Busy, and Diversion on No Reply), there is no interaction with the RO/TAT Interworking feature (the interaction occurs between the Diversion and TAT features.) In the case of tromboning on the DPNSS1 side, the Diversion feature clears the DPNSS1 tromboning trunks before Station C answers the call. When C answers, TAT is applied transparently.

Note 6: Node 1 cannot be a DMS switch for the RO/TAT Interworking feature to operate.

RO/TAT interworking within multiple MCDN/DPNSS1 gateways

A RO/TAT Interworking is supported within a multiple gateway scenario, as illustrated by the following example. Referring to Figure 117, Station A on the originating node call Station B across the multiple gateway scenario over PRI and DPNSS1 trunks, as shown below. Station B then transfers to Station C, over different PRI/DPNSS1 trunks. When Station C has completed the call transfer, and Station C answers, TAT is first activated at the far end node, removing the two end PRI trunks. The RO/TAT Interworking feature then activates RO on the DPNSS1 portion of the gateway, removing the DPNSS1 trunks. Then, TAT is activated to remove the last two PRI trunks at the near end of the gateway, leaving Station C and Station A bridged, as shown in Figure 118.

Figure 117
RO/TAT Interworking within multiple DPNSS1/MCDN gateways, before RO/TAT

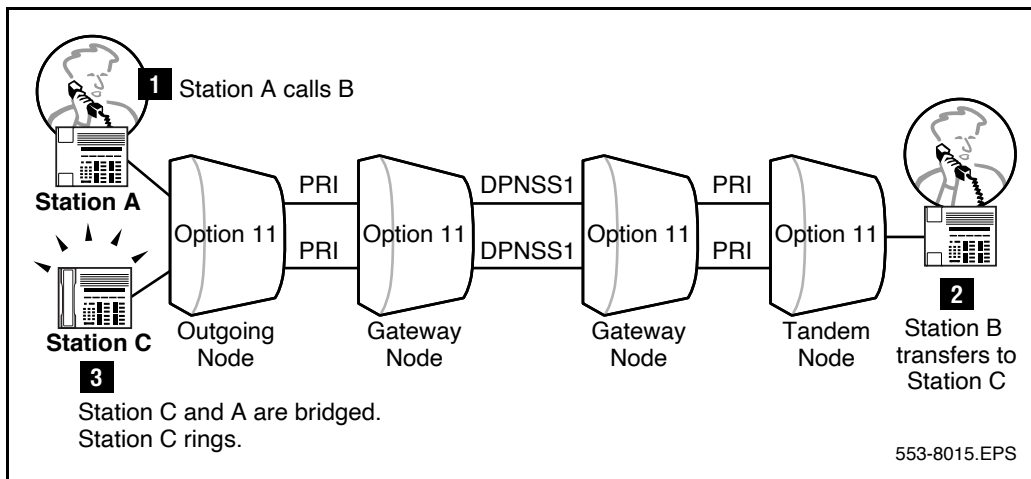
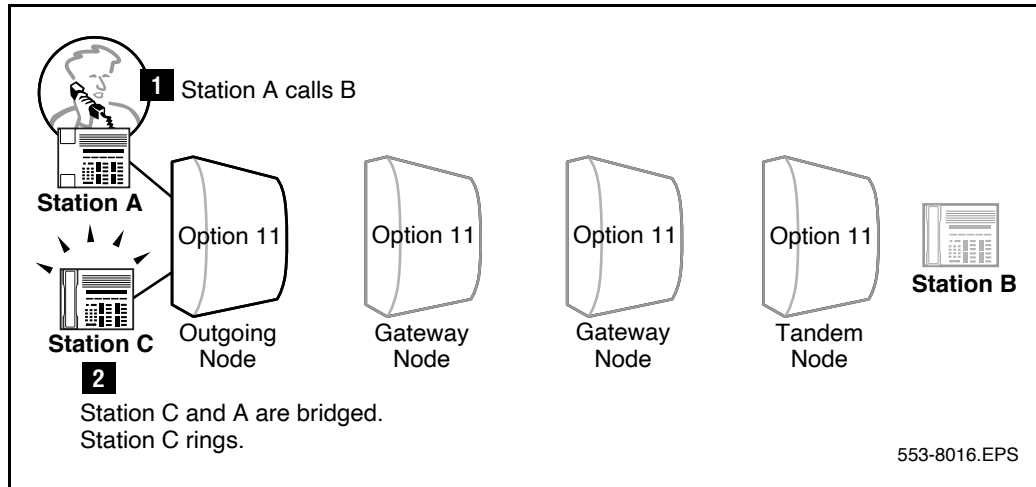


Figure 118
RO/TAT Interworking within multiple DPNSS1/MCDN gateways, after RO/TAT



Abnormal RO/TAT interworking scenarios

The following are possible scenarios whereby the RO/TAT Interworking feature may function abnormally.

- RO fails or is not configured, and TAT is configured.

In the case of a DPNSS1/MCDN gateway, TAT optimises the PRI trunks on the MCDN side, but the DPNSS1 trunks are not optimised on the DPNSS1 side.

In the case of an MCDN/DPNSS1 gateway, RO is not activated and the DPNSS1 side is not optimised. Since the DPNSS1 trunks remain, TAT is not invoked at the gateway node, even though it is equipped. Therefore, if RO is not activated, the RO/TAT Interworking functionality is not invoked.

- TAT fails or is not configured, and RO is configured.

In the case of an MCDN/DPNSS1 gateway, RO optimises the DPNSS1 trunks on the DPNSS1 side, but the MCDN trunks are not optimised on the MCDN side.

In the case of a DPNSS1/MCDN gateway, TAT is not activated on the MCDN side and the tromboning PRI trunks remain. Since the PRI trunks remain, RO is not invoked at the gateway node, even though it is equipped, and DPNSS1 trunks are not optimised on the DPNSS1 side. Therefore, if TAT is not activated, the RO/TAT Interworking functionality is not invoked.

Operating parameters

Although Trunk Anti-Tromboning functions between a Meridian 1 switch and a DMS switch, no TAT messaging is initiated to a DMS switch after Route Optimisation is activated on the DPNSS1 side of an ISDN MCDN/DPNSS1 gateway.

As explained in “Abnormal RO/TAT interworking scenarios” on page 787, both RO and TAT must be activated in order for the RO/TAT Interworking functionality to operate.

The RO/TAT Interworking functionality is only activated after call connection.

RO/TAT Interworking functionality is not applied if the originating party of the first call or the terminating party of the second call is on a conference call.

RO/TAT Interworking functionality is not applied if the originating party of the first call is an attendant.

RO/TAT Interworking functionality is not applied to data calls.

Route Optimisation may be applied to any portion of a DPNSS1 network, as long as both the originating node and terminating nodes are equipped with the RO feature. This is because optimisation is performed by initializing a new call between the originating node and terminating node. However, for the same to apply to Trunk Anti-Tromboning within an MCDN network, every exchange along the network must be equipped with the TAT feature. This is because TAT releases trunks step by step.

Multiple hops across a gateway are supported separately by RO and TAT.

Feature interactions

Multiple Hops

Multiple hops are supported within every RO/TAT Interworking gateway scenario, since they are supported separately by RO and TAT.

Network Attendant Service

If tromboning trunks are removed on the MCDN side of a RO/TAT Interworking gateway scenario by the Network Attendant Service feature (since NAS has precedence over TAT), the RO/TAT Interworking functionality is not invoked. The result is that, if NAS is equipped, attendant-extended calls that are in a tromboning state are optimised on the MCDN side, but DPNSS1 trunks are not optimised on the DPNSS1 side of the RO/TAT Interworking gateway scenario.

Network Call Pickup

If tromboning trunks are removed on the MCDN side of a gateway scenario by the Network Call Pickup feature (since Network Call Pickup has precedence over TAT), TAT is invoked since the Network Call Pickup action is considered as a call forward action. RO/TAT functionality is invoked upon completion of the TAT operation.

Network Call Redirection

If Network Call Redirection is not configured in an DPNSS1/MCDN gateway, the displays are updated normally, since the RO/TAT Interworking feature is not affected.

If Network Call Redirection is not configured in an MCDN/DPNSS1 gateway, the displays are not updated on the bridged sets on the MCDN side. However, if the bridged sets are on the same node, the displays are updated even though NCRD is not configured.

Three Party Service

DPNSS1 Three Party Service is required for every RO/TAT Interworking scenario.

Trunk Route Optimization before Answer

There is no interaction between the Trunk Route Optimization before Answer feature and the RO/TAT Interworking feature, since Trunk Route Optimization before Answer is activated before call completion, and the RO/TAT Interworking functionality is only activated after call connection.

Virtual Network Services

The RO/TAT Interworking feature is not supported over VNS trunks, since VNS uses only MCDN signaling (DPNSS1 is not supported.)

Feature packaging

For the software packages required to support the DPNSS1 Route Optimisation/MCDN Trunk Anti-Tromboning Interworking feature, consult the following publications:

- For DPNSS1 network functionality:
 - *DPNSS1 Product Information Guide* 553-3921-100;
- For MCDN Network Attendant Service interworking:
 - *International ISDN PRI features description and administration* 553-2901-301

Feature administration

No new steps are required to configure the DPNSS1 Route Optimisation/MCDN Trunk Anti-Tromboning Interworking feature. However, the following basic configuration must be done:

- Configure MCDN Trunk Anti-Tromboning at the far-end switch, in LD 17 (TAT is configured on a D-Channel basis, and not on a route basis). Refer to the *International ISDN PRI features description and administration* 553-2901-301
- Configure DPNSS1 Route Optimisation options, in LD 15. Refer to *DPNSS1 Administration Guide*.
- Configure call display transfer indication for DPNSS1 Three-Party Service, in LD 95. Refer to *DPNSS1 Administration Guide*.
- Optionally (to update terminal displays), configure Network Call Redirection, using LD 15, LD 16, LD 95, LD 10, and LD 11. Refer to the *International ISDN PRI features description and administration* 553-2901-301

Feature Operation

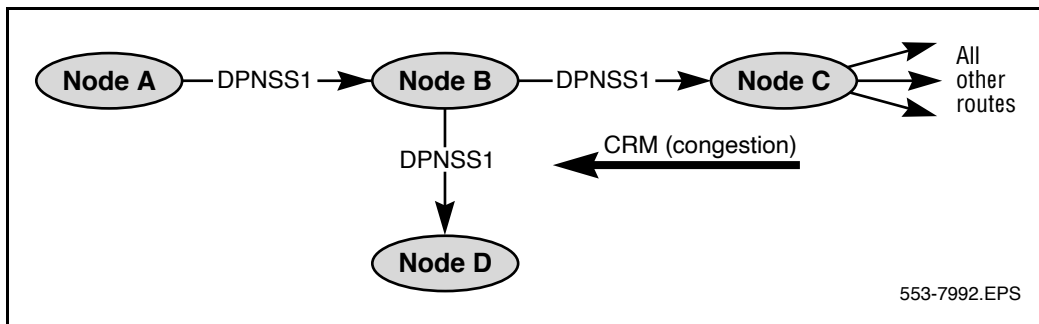
There are no operating procedures specified for this feature.

DPNSS1 Step Back on Congestion

This feature has been developed to handle high traffic situations, when DPNSS1 calls may encounter congestion. If a call over a DPNSS1 network is blocked due to congestion, a Clear Request Message (CRM) is sent back to the preceding node. A transit node or a DPNSS1/ISDN gateway node may receive a CRM with a clearing cause of congestion. An originating node may receive either a CRM containing a clearing cause of congestion, or a CRM containing a clearing cause of Network Termination and a Loop Avoidance Supplementary string. Depending on the SBOC option configured in overlay 86, the Electronic Switched Network overlay, the call may be passed back or re-routed using the next free alternative route.

If a CRM with a clearing cause of congestion is received at a transit node, the call may be passed back or re-routed. If a CRM of congestion is received at a DPNSS1/ISDN gateway node, the ISDN Drop Back Busy options, included in the SETUP message according to the Route List Block, are checked to determine whether the call is to be dropped back. If not, the DPNSS1 Step Back on Congestion feature is invoked. If an originating node receives either a CRM containing a clearing cause of congestion, or a CRM containing a clearing cause of Network Termination and a Loop Avoidance Supplementary string. The call may be routed using the next free alternative route, or receive call blocking treatment if no re-routing is configured or if no alternative route is available.

Transit node operation



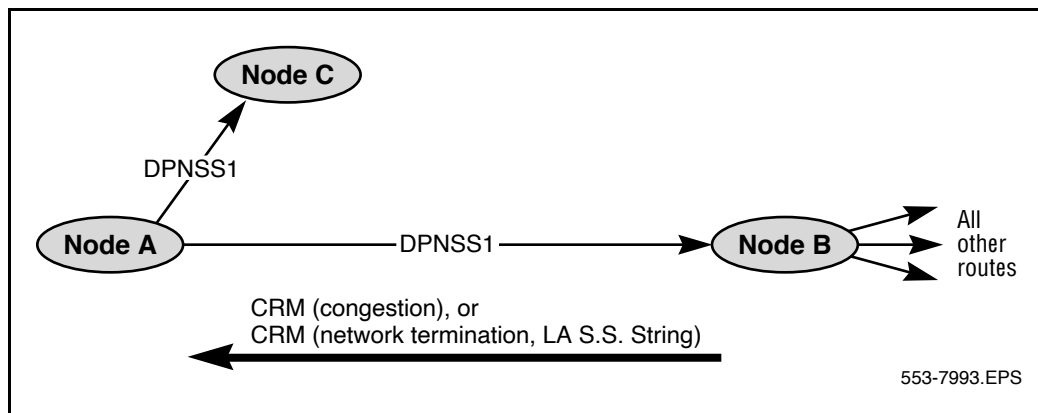
An attempt is being made to establish a call through a DPNSS1 network, from originating node A to terminating node C, via transit node B. All the trunks at node C are busy, so that a CRM with a clearing cause of congestion is sent to the preceding node (node B). At transit node B, alternative 1 is to re-route to node C, and alternative 2 is to re-route to node D.

The SBOC option for node B is checked in overlay 86 to determine the treatment. If SBOC = RRA, the next free alternative is tried. If the Class of Service and Network Class of Service access checks are passed, the call is re-routed to the next free alternative, which is node D. A Network Indication Message indicating alternative routing is sent to the preceding node (node A). If there had been no free alternatives, a CRM of congestion would have been sent back to node A.

If SBOC = NRR or RRO, a CRM of congestion is passed back from transit node B to the preceding node A.

Note: If the call is a route optimization attempt, there is no attempt made to re-route it — a CRM of congestion is passed back to the preceding node.

Originating node operation



An attempt is being made to establish a call through a DPNSS1 network, from originating node A to terminating node B. All the trunks at node B are busy, so that a CRM with a clearing cause of congestion, or a CRM containing a clearing cause of Network Termination and a Loop Avoidance Supplementary string, is sent to the preceding node (node A).

The SBOC option for node A is checked in overlay 86 to determine the treatment. If SBOC = RRA or RRO, the next free alternative (node C) is tried.

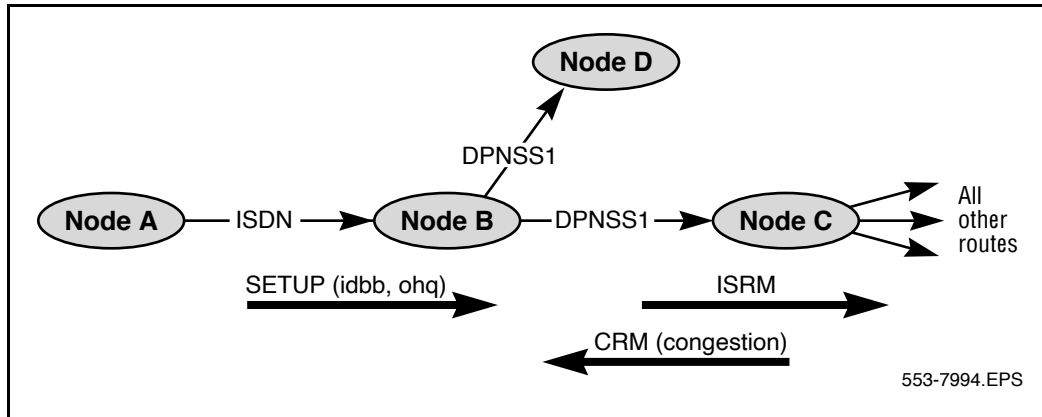
If SBOC = NRR, or if no alternatives are available, the network blocking treatment, as defined by prompt NBLK in overlay 15, is applied to the call at node A. Note that, for local extensions, if the dialing has not been completed, the provision of busy tone treatment (if defined) is delayed so that digits may be dialed for other features such as Ring Again.

Operating parameters

- This feature uses the ESN Coordinated Dialing Plan, or Network Alternate Route Selection (NARS) or Basic Alternate Route Selection (BARS) to re-route a congested call. Re-routing is not attempted if a trunk access code was used to originate the call.

Feature interactions

- If a call that has undergone digit manipulation encounters congestion, digit manipulation is re-applied using the originally dialed digits before re-routing is attempted.
- A call that is blocked due to the DPNSS1 Loop Avoidance feature may be re-routed at the originating node, but not at a transit node.
- DPNSS1 route optimized calls that encounter congestion are not re-routed, since route optimization only uses first choice routes.
- The intercept treatment applied due to network blocking is customer-defined in overlay 15.
- DPNSS1/ISDN gateway interworking may be illustrated as follows:

ISDN-to-DPNSS1 gateway node operation

An attempt is being made to establish a call through an ISDN-to-DPNSS1 gateway, from originating node A to terminating node C, via gateway node B. When a gateway node (node B) receives a SETUP message, the IDBB and OHQ options are stored. The IDBB option is used to determine which route sets can be used for ISDN Drop Back Busy, and the OHQ option is used to decide if Off Hook Queuing is to be applied at the congested node.

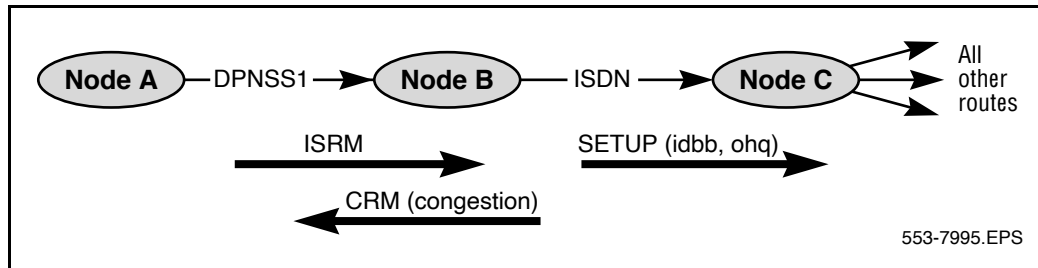
If all the trunks at node C are busy, a CRM with a clearing cause of congestion is sent to the gateway node (node B). At node B, the decision is made whether to apply Drop Back Busy or Off Hook Queuing. If OHQ = NO, the call is dropped back to the originating node — the SBOC option is not checked.

If OHQ = YES, treatment is applied according to the SBOC option. If SBOC = RRA, an attempt is made to find a free alternative route, as defined by the IDBB option (if IDBB = DBI, then only I-SET routes may be used to route the call; if IDBB = DBA, then all E-SET routes may be used to route the call). If a free alternative route is found, the call is routed to node D. If no free alternative route is found, the call is dropped back to node A. If SBOC = RRO or NRR, node B drops the call back to node A by sending a DISCONNECT message with a cause of normal clear.

The following illustrations depict the DPNSS1 Step Back on Congestion functionality as applied to a call trying to be established through a

DPNSS1 network at the originating node, at a transit node, at an ISDN-to-DPNSS1 gateway, and at a DPNSS1-to-ISDN gateway.

DPNSS1-to-ISDN gateway node operation



An attempt is being made to establish a call through an DPNSS1-to-ISDN gateway, from originating node A to terminating node C, via gateway node B. In order to provide a consistent interworking between ISDN Drop Back Busy and DPNSS1 Step Back on Congestion, the options for IDBB and OHQ are included at gateway node B in the SETUP message from the outgoing route list block. If all the trunks at node C are busy, and drop back occurs, a DISCONNECT message with a cause of normal clear is sent to gateway node B, where it is recognized as a drop back attempt. The DISCONNECT message is mapped to a CRM with a reason of congestion, so that Step Back on Congestion is invoked in the DPNSS1 segment of the path.

— DPNSS1 operation will not support the following:

- ISDN Basic Rate Interface (BRI) sets

or

- ISDN BRI Numeris Terminals.

Feature packaging

- DPNSS1 Network Services (DNWK) package 231

Dependencies:

- Integrated Digital Access (IDA) package 122
- Digital Private Network Signaling System No. 1 (DPN) package 123
- International Supplementary Features (SUPP) package 131
- Basic Alternate Route Selection (BARS) package 57, dependent on Basic Routing (BRTE) package 14 and Network Class of Service (NCOS) package 32
- Co-ordinated Dialing Plan (CDP) package 59, dependent on Basic Routing (BRTE) package 14 and Network Class of Service (NCOS) package 32
- Pretranslation (PXLT) package 92
- Flexible Numbering Plan (FNP) package 160
- At gateway and ISDN nodes, the Network Attendant Service (NAS) package 159
- For the ISDN Drop Back Busy feature, the Originating Routing Control/Remote Virtual Queuing (ORC-RVQ) package 192, the Network Alternate Route Selection (NARS) package 58, the Network Class of Service (NCOS) package 32, the Basic Routing (BRTE) package 14, the Flexible Call Back Queuing (FCBQ) package 61, and Integrated Services Digital Access (ISDN) package 145.
- For ISDN signaling links, the Primary Rate Access (PRA) package 146

Feature implementation

- **LD86-** In the ESN overlay, define the step back or re-routing options for calls encountering congestion.
 - In response to the SBOC prompt, enter NRR for no re-routing of calls, enter RRO to re-route calls if at an originating node, or to step back if at a transit node, or enter RRA to re-route calls if at all nodes.
 - In response to the IDBB prompt, enter (DBA)/DBI to define the alternative route sets for ISDN Drop Back Busy. DBA = all E-SET routes may be used to route calls, DBI = only I-SET routes may be used to route calls.
- **LD15-** In the Customer Data Block overlay, define the network blocking treatment for calls stepped back to the originating node.
 - In response to the NBLK prompt, enter four of the following treatments: OVF, ATN, RAN, BSY, SRC1.SRC8. The default is OVF, OVF, OVF, ATN.

DPNSS1/DASS2 Uniform Dialing Plan Interworking

The Digital Private Network Signaling System (DPNSS1)/Digital Access Signaling System (DASS2) Uniform Dialing Plan (UDP) Interworking feature enables DPNSS1 to use Uniform Dialing Plan numbering. The feature allows DPNSS1 calls to be routed from a switch in one geographical location to another switch in any other geographical location in a cost-effective and easy-to-use manner via the Network Alternate Route Selection (NARS) and Basic Alternate Route Selection (BARS) features.

The following NARS/BARS functions are supported by this development:

- On-network routing over DPNSS1 using a standardized dialing format (Access Code (AC) - Location Code (LOC) - destination DN).
- Off-network routing and break-outs to supported public network interfaces.
- Incoming DASS2 calls routed through the UDP DPNSS1 with NARS if the received digits are in the format AC - LOC or AC-SPN Special Number (SPN) - X...X, or if an Incoming Digit Conversion (IDC) table is applied to generate a number in such a format. In this case, all NARS functionalities supported on DPNSS1 apply to the incoming DASS2 call.

- Least cost route selection by arranging the routes based on relative cost.
- Route control allowing or restricting access to routes based on their restriction level, the time of day, or the dialed sequence.
- Simple calls using UDP numbering across the DPNSS1 Meridian Customer Defined Network (MCDN) gateway, operating with either enbloc or overlap sending and receiving.
- The following DPNSS1 Supplementary Services are supported on UDP DPNSS1 and across the DPNSS1 MCDN gateway:
 - Call Back When Free
 - Call Back When Next Used
 - Executive Intrusion
 - Loop Avoidance
- The following DPNSS1 Supplementary Services are supported on UDP DPNSS1
 - Three-Party
 - Call Offer
 - Redirection
 - Step Back on Congestion
 - Route Optimisation

With DPNSS1, no parameters are exchanged between the switches to indicate whether a Coordinated Dialing Plan (CDP) or UDP number is being sent. To bypass this situation, the MCDN Insert Access Code (INAC) prompt is extended to DPNSS1 on a per route basis to indicate whether the route is dedicated for the reception of UDP numbers, or non-UDP numbers and UDP numbers with a NARS access code.

Operating parameters

The BARS feature by itself (i.e., without NARS) is not supported.

If INAC = YES, an incoming DPNSS1 route only supports UDP numbers with or without the NARS Access Code. If INAC = NO, both non-UDP and UDP numbers are supported providing UDP numbers are appended with the local NARS Access Code.

All HLOCs, LOCs and SPNs NARS codes for a customer must be leftwise unique.

At an originating node, the Traveling Class of Service (TCOS) is not transmitted over DPNSS1. At a receiving node, the Network Class of Service's Facility Restriction Level of the incoming trunk group is used for any further check on access restrictions.

Any number made up of a Route Access Code followed by a DN, received on a route configured with the INAC prompt set to YES, is blocked.

CBWF/CBWN, Loop Avoidance, Three-Party, Call Offer, Redirection, Step Back on Congestion, Route Optimisation, and Executive Intrusion are the only supported Supplementary Services on a DPNSS1 UDP.

The following hardware is required (minimum vintage):

- DASS2/DPNSS1 D-channel Interface Handler
 - Standard Mode (0-15 D-channels) – NT5K35AA
 - Expanded Mode (0-159 D-channels) – NT5K75AA
- 2 Mbps Primary Rate Interface Card – NT8D72AA
- PRI card for MCDN/DPNSS1 – NT6D11AE
- Network Interface Card – QPC414
- Clock Controller Card – QPC775D
- D-channel Handler (for the Option 11).

Feature interactions

Access Restrictions

The connection between the network user (extension or trunk) and the DPNSS1 UDP trunk can be barred based on the Class of Service Restrictions of the parties involved. The connection between the network user (extension or trunk) and the DPNSS1 trunk can also be barred based on the Trunk Group Access Restrictions feature. It is possible to bar the connection between originator and terminator through a DPNSS1 UDP trunk based on the DPNSS1 signaling information.

The Code Restriction sub-feature is not supported.

Attendant Alternate Answering

If an incoming DPNSS1 UDP call presented to an idle loop key of an attendant is not answered within a predefined period of time, the call can be rerouted to the Attendant Alternate DN.

Attendant Interpositional Transfer

This feature is supported in a UDP DPNSS1 network. An attendant can call or transfer a call to another attendant in a multiple-console group, even when the destination Attendant Console is busy.

Attendant Overflow Position

This feature is supported on a UDP DPNSS1 network. If an incoming DPNSS1 UDP call is queued to the attendant, and if the call is not answered within a predefined period of time, the call can be redirected to the Attendant Overflow DN.

Automatic Call Distribution (ACD)

This feature is supported; however, when a call is answered by an agent through a DPNSS1 UDP route, the display on the originator's set is not updated when the ACD agent answers the call.

Call Detail Recording

This feature is supported in a DPNSS1 UDP network. The following items should be noted:

- If an expensive route (EXP prompt in the Route List Index block) is used to route the call, and if the calling party is allowed the expensive route warning tone (RWTA prompt in Network Control block), the Digit Type Identifier field is “E” in the call record output; otherwise it is “A”.
- If both NARS and BARS packages are equipped in a DPNSS1 UDP network, the Digits field in the call record follows the BARS format. If the Outpulsed Digits feature is used, and OPD = NO, the Route Access Code (ACOD) + the digits dialed after the NARS/BARS Access Code are displayed in the call output record. If the Outpulsed Digits feature is used, and OPD = YES, the Route Access Code + the outpulsed digits are displayed in the call output record.

Call Forward

Calls can be forwarded to and from a DPNSS1 UDP network.

Call Party Name Display

This feature is supported in a DPNSS1 UDP network. Names can be associated with the access codes of the DPNSS1 UDP routes defined in LD 95.

Call Pickup

This feature is supported in a DPNSS1 UDP network.

Custom Call Routing

This feature is not supported in a DPNSS1 UDP network.

Digit Display

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

Direct Inward Dialing

This feature is supported in a DPNSS1 UDP network. A connection between a DASS2 DID trunk and a DPNSS1 UDP trunk can be barred using the DITI option in the Customer Data Block.

Direct Inward System Access

This feature is not supported.

DPNSS1 Gateway

The supplementary services supported in a DPNSS1 UDP network are Call Back When Free/Call Back When Next Used, Loop Avoidance, Three-Party, Call Offer, Redirection, and Executive Intrusion.

Electronic Switched Network (ESN)

The DPNSS1/DASS2 Uniform Dialing Plan Interworking feature is a form of ESN routing. The following list describes which ESN functionalities are applicable to a DPNSS1 UDP network.

Alternative Routing for DID/DOD

This feature is supported in a DPNSS1 UDP network.

Basic Alternate Route Selection

The BARS feature alone is not supported.

Coordinated Dialing Plan

Non-UDP and UDP numbers, if they are appended with the NARS access code, received on a route configured with INAC set to NO are now able to terminate, transit, or be sent across the gateway if they are valid.

ESN Signaling (Network Signaling)

The Network Signaling feature is incompatible with DPNSS1 routes.

Eleven Digit Translation

This feature is supported in a DPNSS1 UDP network. Numbers received on an incoming UDP DPNSS1 route can be translated with the existing NARS translator up to 11 digits for route selection.

Flexible Call Back Queuing

This feature is incompatible with DPNSS1 routes.

Flexible ESN 0 Routing

This feature is not supported in a DPNSS1 UDP network.

Flexible Numbering Plan

This feature is supported in a DPNSS1 UDP network.

Free Calling Area Screening

Free Special Number Screening

These features are not supported in a DPNSS1 UDP network.

Incoming Trunk Group Exclusion

This feature is supported in a DPNSS1 UDP network.

Multiple DID Office Code Screening

This feature is not supported in a DPNSS1 UDP network.

NARS Traffic Measurement (Network Traffic Measurement)

This feature is supported in a DPNSS1 UDP network in the following areas:

- Routing Traffic Measurements – provides data related to route list utilization, and
- NCOS Measurements – provides data about the quality of service for a defined NCOS group.

Network Authorization Code

This feature is supported in a DPNSS1 UDP network.

Network Control

This feature is supported in a DPNSS1 UDP network.

Network Routing Control

This feature is supported in a DPNSS1 UDP network.

Network Speed Call

This feature is supported in a DPNSS1 UDP network.

Network Call Transfer

This feature is not applicable to a DPNSS1 UDP network, as it only applies to analog trunks.

Network Queuing (Call Back Queuing)

This feature is incompatible with DPNSS1 routes.

Off-Hook Queuing

This feature is not supported in a DPNSS1 UDP network.

Off-Network Number Recognition

This feature is supported in a DPNSS1 UDP network.

Priority Queuing

This feature is incompatible with DPNSS1 routes.

Satellite Link Control

This feature is not supported in a DPNSS1 UDP network.

Special Common Carrier Access

This feature is not supported in a DPNSS1 UDP network.

Tone Detection

This feature is not supported in a DPNSS1 UDP network.

1+ Dialing

This feature is not supported in a DPNSS1 UDP network.

Group Hunting

Only basic DPNSS1 UDP calls are supported with group hunting.
Interactions between DPNSS1 Supplementary Services and Group Hunting are not supported.

Incoming DID Digit Conversion (IDC)

An IDC table can be used to convert digits received on a DASS2 DID trunk into a digit string having the UDP format. This allows a DASS2 DID call to access the DPNSS1 UDP network.

Intercept

The NARS blocking treatments that can be defined through the Intercept feature are applicable to a DPNSS1 UDP network.

Intercept Computer

This feature is not supported in a DPNSS1 UDP network.

Interchangeable NPA/NXX

This feature is not supported in a DPNSS1 UDP network.

Meridian Link

This feature is not supported in a DPNSS1-UDP network.

Meridian Mail

This feature is not supported in a DPNSS1 network.

Meridian Mail, Standalone

This feature is supported in a DPNSS1 UDP network.

Network Message Services

This feature is not supported in a DPNSS1 UDP network.

New Flexible Code Restrictions (NFCR)

Toll-denied users (CLS = TLD) may be subject to NFCR if they make a NARS call across the DPNSS1 UDP network. This feature is supported in a DPNSS1 UDP network.

Overlap Signaling

This feature is supported in a DPNSS1 UDP network.

Pretranslation

This feature is supported in a DPNSS1 UDP network.

Recorded Announcement for Calls Diverted to External Trunks

This feature is not supported in a DPNSS1 UDP network.

Route Optimisation

This DPNSS1 feature is supported.

R2MFC to DPNSS1 Gateway

This gateway is supported with UDP numbers at the same level as it is supported with CDP numbers.

Special Dial Tone after Dialed Numbers

This feature is supported in a DPNSS1 UDP network.

Feature packaging

The DPNSS1/DASS2 Uniform Dialing Plan Interworking feature is part of basic X11 system software. The following package is required:

- Network Alternate Route Selection (NARS) package 58.

Basic Alternate Route Selection (BARS) package 57 may also be equipped.

For on-net and off-net routing capabilities, the following package is required:

- Flexible Numbering Plan (FNP) package 160.

If more ESN functions are desired the following packages are required:

- Network Traffic Measurement (NTRF) package 29
- Network Authorization Code (NAUT) package 63
- Basic Authorization Code (BAUT) package 25
- System Speed Call (SSC) package 34
- Network Speed Call (NSCL) package 39, and
- Directory Number Expansion (DNXP) package 150.

If a Group Dialing Plan is to be used, the following package is required:

- Coordinated Dialing Plan (CDP) package 59.

If displays are required, the following package must be equipped:

- Digit Display (DDSP) package 19.

The following packages are required for DASS2/DPNSS1:

- Integrated Digital Access (IDA) package 122
- Digital Private Network Signaling System 1 (DPNSS) package 123
- Digital Access Signaling System 2 (DASS2) package 124, and
- International Supplementary Features (SUPP) package 131.
- 2.0 Mbps Primary Rate Interface (PRI2) package 154 is a prerequisite.

Feature implementation

LD 87 – Define the NCOS groups to which the users will belong.

Prompt	Response	Description
REQ	NEW	New.
CUST	0-31	Customer Number.
FEAT	NTCL	Network Control Block.
NCOS	0-31	Network Class of Service.
FRL	0-7	Facility Restriction Level.
RWTA	(NO), YES	Expensive Route Warning Tone.

LD 86 – Define the NARS feature parameters (all prompts of the ESN Block are applicable to a DPNSS1 UDP network).

Prompt	Response	Description
REQ	NEW	New.
CUST	0-31	Customer Number.
FEAT	ESN	Electronic Switched Network Block.
...		
AC1	xx	NARS Access Code 1 (1-4 digits if Flexible Numbering Plan (FNP) package 160 is equipped; otherwise 1-2 digits).
AC2	xx	NARS Access Code 2 (1-4 digits if Flexible Numbering Plan (FNP) package 160 is equipped; otherwise 1-2 digits).

LD 86 – Define the Digit Manipulation tables.

Prompt	Response	Description
REQ	NEW	New.
CUST	0-31	Customer Number.
FEAT	DGT	Digit Manipulation Table.
DMI	x...x	Digit Manipulation Table Index.

LD 86 – Define the Route List blocks.

Prompt	Response	Description
REQ	NEW	New.
CUST	0-31	Customer Number.
FEAT	RLB	Route List Block.
RLI	x...x	Route List Index number.
ENTR	0-63	Entry number.
- ROUT	0-511	Route number associated with the index.
...		
- OHQ	NO	On-Hook Queuing is not supported in a DPNSS1 UDP network (ROUT is a DPNSS1 route).
- CBQ	NO	Call Back Queuing is not supported in a DPNSS1 UDP network (ROUT is a DPNSS1 route).

LD 90 – Define the NARS LOC translation table.

Prompt	Response	Description
REQ	NEW	New.
CUST	0-31	Customer Number.
FEAT	NET	Network Translation Table.
TRAN	AC1	NARS Access Code 1 (1-4 digits if Flexible Numbering Plan (FNP) package 160 is equipped; otherwise 1-2 digits).
TYPE	LOC	Location Code.
LOC	x...x	Location Code (3-7 digits).
FLEN	(0)-10	Flexible number of digits for Location Code (prompted if Flexible Numbering Plan (FNP) package 160 is equipped).
RLI	x...x	Route Line Index.

LD 90 – Define the NARS HLOC translation table.

Prompt	Response	Description
REQ	NEW	New.
CUST	0-31	Customer Number.
FEAT	NET	Network Translation Table.
TRAN	AC1	NARS Access Code 1 (1-4 digits if Flexible Numbering Plan (FNP) package 160 is equipped; otherwise 1-2 digits).
TYPE	HLOC	Home Location Code.
LOC	x...x	Home Location Code (3-7 digits).
DMI	x...x	Digit Manipulation Table Index.

LD 16 – Select the DPNSS1 UDP routes.

Prompt	Response	Description
REQ	CHG	Change.
TYPE	RDB	Route Data Block.
...		
TKTP	IDA	Integrated Digital Access.
SIGL	DPN	DPNSS1 signaling on this route.
...		
RCLS	...	
- INAC	YES	DPNSS1 route supporting only UDP numbers.
- SPN	(YES), NO	Insert first the LOC's Access Code to search for a valid UDP number.

LD 15 – Configure the HLOC in the Customer Data Block to use the UDP digit format on DPNSS1 UDP routes.

Prompt	Response	Description
REQ	CHG	Change.
TYPE	CDB NET_DATA	Customer Data Block. Release 21 gate opener.
...		
HLOC	100-3199	Home Location Code (the HLOC entered here should be the same as that defined in the NET block).

LD 90 – Define SPN numbers.

Prompt	Response	Description
REQ	NEW, CHG, OUT	Create, change, or remove data.
CUST	0-31	Customer Number.
FEAT	NET	Network Translation Tables.
TRAN	AC1, AC2, SUM	Access Code 1, 2, or summary tables.
TYPE	SPN	Special Number translation code.
SPN	xxxx xxxx x..	Special Number translation.
RLI	0-319	Route List Index.
SDRR	aaa	Type of supplemental restriction or recognition.
DENY	x...x	A number to be denied within the SPN. The maximum number of digits allowed is 10 minus n, where n is the number of digits entered for the prompt SPN. Repeat to deny other numbers.
ARRN	xxxxx	Alternate Routing Remote Number.
ARLI	0-319, <CR>	Only output if ARRN is output.
LDID	x...x	Local DID number recognized with the NPA, NXX, or SPN.
DMI	1-255	Digit Manipulation Table Index.
LDDD	x...x	Local DDD number recognized within the NPA, NXX, or SPN.
DID	x...x	Remote DID number recognized within the NPA, NXX, or SPN.
DDD	x...x	Remote DDD number recognized within the NPA, NXX, or SPN.
ITED	x...x	Incoming trunk group exclusion codes for NPA, NXX, or SPN.
ITEI		Incoming trunk group exclusion index.

LD 16 – Configure a Conventional Main for Off-net recognition.

Prompt	Response	Description
REQ	NEW, CHG	New, or change.
TYPE	RDB	Route Data Block.
CUST	0-31	Customer Number.
TKTP	TIE	TIE trunk.
CNVT	(NO), YES	Route to conventional switch (prompted if the response to TKTP is TIE).
DDMI	(0)-255	Digit Manipulation Index (prompted if the response to CNVT is YES).
ATDN	xxxx	Attendant DN of Conventional Main (prompted if the response to CNVT is YES).

Feature operation

No specific operating instructions are required to use this feature; however, the following validation algorithm is used by the system.

Validation Algorithm

Upon reception of a number on a DPNSS1 UDP route configured with INAC = YES, the following validation algorithm applies if SPN is set to YES in the Route Data Block:

- 1 The SPN's NARS Access Code is appended to the received number and a valid NARS code is searched for.
- 2 If no valid NARS code is found, the appended SPN's NARS Access Code is stripped off and a valid SPN or LOC NARS code is searched for.
- 3 If no NARS code is found and if the LOC's NARS Access Code differs from the SPN's NARS Access Code, the LOC's NARS Access Code is appended to the received number and a valid LOC code is searched for.
- 4 If no valid LOC code is found, the appended LOC's NARS Access Code is striped off and the received number is considered invalid, and the call is released.

If SPN was set to NO in the Route Data Block a similar validation algorithm applies, except that the LOC's NARS Access Code is appended first in front of the received number.

Attendant Through Dialing Networkwide

Attendant Through Dialing Networkwide extends the functionality of through dialing via an attendant to any Integrated Services Digital Network (ISDN) or DASS2 outgoing trunk. This feature allows an attendant to seize an outgoing Integrated Services Digital Network (ISDN) or DASS2 trunk for a calling party located on the same or another node.

Refer to “Chapter 2 - ISDN Feature descriptions”, for a description of the Attendant Through Dialing Networkwide feature.

ISDN to DPNSS1/DASS2 gateway

The ISDN to DPNSS1/DASS2 Gateway provides interworking between DPNSS1/DASS2 and the following ISDN interfaces:

- ISDN BRI (line and trunk, Net-3 compliant);
- EuroISDN (ETS 300-102);
- ISDN QSIG (ETS 300-172);
- MCDN (NT proprietary PBX-PBX ISDN protocol).

The following features and capabilities are supported over the ISDN to DPNSS1 gateway:

- Basic call service (3.1 kHz speech, 64K restricted/unrestricted data);
- Overlap sending and receiving;
- 64 kbps bearer capability;
- Data rate adaptation (in accordance with ETS 300-102, and in compliance with BTNR 188 Section 7);
- Mapping of BTNR 189-I supplementary information service strings within DPNSS1 messages.

Operating parameters

The feature has the following limitation:

- PSTN incoming trunks are not to be allowed access to PSTN outgoing trunks

Feature packaging

- DPNSS1 Network Services (DNWK) package 231
- UIGW package 283
- Integrated Digital Access (IDA) package 122
- DPNSS1 package 123
- DASS2 package 124
- PRI2 package 154

— DPNSS189I package 284

Feature implementation

None.

Standalone Meridian Mail

The Standalone Meridian Mail feature provides a Meridian Mail system interface to a third-party vendor's network by way of a DPNSS1 interface to the Meridian Mail system. Users on the third-party exchange can be alerted of the message waiting, and can access the messages that have been left from a remote telephone.

This feature will depend on the features supported via DPNSS1 on your PBX. The Standalone Meridian Mail feature allows access to and from a DPNSS1 interface into and out of Meridian Mail. Prior to Release 20, Meridian Mail could not interface to a remote PBX via DPNSS1, but with this feature it is now possible to send and receive messages from a remote DPNSS1 node. Calls to busy sets, sets with call forwarding activated, or when there is no answer, can be routed to Meridian Mail across a DPNSS1 link to provide voice mail service. All features present on Meridian Mail can then be used as required.

Standalone Meridian Mail introduces the following enhancements:

Automatic Login to Personal Mailbox from User on Third-party PBX

This will enable Meridian Mail users to login from their telephones to the Meridian Mail system offered on the Meridian 1 by simply entering a “#” from the set.

Busy Notification

When Meridian Mail is accessed from the third-party PBX because the user is on another call, Standalone Meridian Mail informs the caller of this by using a voice message.

Call Answer

Call answer enables incoming calls to the third-party PBX which have been diverted to the Standalone Meridian Mail to access the correct mailbox.

Custom System Greeting

This feature is provided to inform callers to the third-party PBX of the organization they have called, if they are calling in on a DID trunk. This announcement precedes the regular voice greeting.

Dual Personal Greeting

For callers terminating on the mail system, this feature provides the option of delivering different messages to internal or external users.

Remote Message Notification

A message waiting lamp can be lit on the remote PBX when a message is left on Meridian Mail for that extension. Currently, this feature is only supported on the Plessey iSDX system.

Operating parameters

Incoming user control to the Standalone mail must be in-band DTMF as post End-of-Dialing digits cannot be supported via DPNSS1 out-of-band signaling. For the Meridian Mail user, in-band DTMF tones are required from the originating party, wherever they originate.

The third-party exchange must be able to allow diversion to a remote switch to allow messages to be left.

Call Sender (a Meridian Mail feature) is not supported with the Standalone Meridian Mail feature.

No gateway functionality between ISDN and DPNSS1 for Meridian Mail access or message waiting capability is implemented with this feature. Access to Meridian Mail for any mailbox user on a network must be via a single signaling system, either Q.931 for a Meridian 1, or DPNSS1 for other PBXs.

No gateway functionality between ISDN and DPNSS1 for DPNSS1 originator diversion is implemented with this feature. Specifically, if a call originates via a DPNSS1 trunk and is redirected via Q.931, no notification of this diversion is made to the originator. The reverse also applies.

The following hardware is required: Meridian Mail module and standard attachments; third-party PBXs with DPNSS1 as needed; Meridian 1 system with DPNSS1 and Meridian Mail hardware, and Q.931 networking trunks.

Operating parameters

None.

Feature interactions

None.

Feature packaging

The new Standalone Meridian Mail (SAMM) package 262 has been introduced for this feature.

Digital Private Network Signaling System 1 (DPNSS1) package 123, and Integrated Digital Access (IDA) package 122 are required for DPNSS1 interworking.

The following packages are required for Meridian Mail: Make Set Busy (MSB) package 17; Integrated Message Services (IMS) package 35; Automatic Call Distribution Package B (ACDB) package 40; Automatic Call Distribution Package A (ACDA) package 45; Message Waiting Center (MWS) package 46; Command Status Link (CSL) package 77; and Auxiliary Processor Link (APL) package 109.

Network Message Services (NMS) package 175 and Advanced ISDN Network Services (NTWK) package 148 are required for remote Meridian Mail operation.

Feature implementation

LD 74 – This overlay is modified to include a DPNSS1 interface specifier.

Prompt	Response	Description
REQ	NEW, CHG	New, or change.
TYPE	DDSL	DPNSS1 signaling link.
...		
MWIF	STD, ISDM	ISDM should be used if the third party PBX is an iSDX switch supporting message waiting with DPNSS1.

LD 15 – The SAMM prompt must be set to YES to enable Standalone Meridian Mail.

Prompt	Response	Description
REQ	NEW, CHG	New, or change.
TYPE	CDB IMS	Customer data block. Release 21 gate opener.
...		
IMS	YES	Integrated Message System.
SAMM	YES, NO	Allow or disallow Standalone Meridian Mail.

LD 10 – Configuration of Standalone Mail Message Servers in this overlay is only necessary when using links to iSDX nodes. The SAMM prompt must be set to YES in LD 15 to enable all call processing functions for Standalone Meridian Mail.

Prompt	Response	Description
REQ	NEW	New.
TYPE	500	Analog PBX set.
...		
CLS	SMSA, SMSD	Allow or disallow Standalone Meridian Mail Server.

Feature operation

This feature enables incoming calls arriving at a third party vendor's PBX to be diverted to Meridian Mail via DPNSS1. The identity of the calling and called parties is given in the Calling Line Category (CLC) of the DPNSS1 message sent by the third party vendor's PBX. The Standalone Meridian Mail feature uses this information to access the correct mailbox on the Meridian Mail system. Telephone set operation is the same for Standalone Meridian Mail as it is for Meridian Mail.

R2 Multifrequency Compelled Signaling to DPNSS1 Gateway

Prior to Release 20, there are a number of countries in Europe, Central America, and South America that require interworking to Other Equipment Manufacturer's PBXs in multivendor networking environments using Digital Private Network Signaling System One (DPNSS1). In these countries, the Central Office protocol is R2 Multifrequency Compelled Signaling (R2MFC) Direct Inward Dialing (DID) and in some cases R2MFC Direct Outward Dialing (DOD). In order for the Meridian 1 to operate in these environments, interworking of DPNSS1 and R2MFC trunks must be provided.

The R2MFC to DPNSS1 Gateway feature provides an interface between R2MFC DID/DOD trunks and DPNSS1 trunks, and can also provide Calling Number Identification (CNI) support for incoming calls.

In addition, the R2MFC to DPNSS1 Gateway feature introduces the following enhancements to the R2MFC incoming CNI request functionalities:

- The ability to request CNI for an incoming R2MFC call is possible immediately after a predetermined number of digits are received. The allowable range for this option is 0 to 7.
- The ability to request CNI for an incoming R2MFC call is possible immediately after an Electronic Switched Network (ESN) code is dialed. The ESN codes recognized for this purpose are Distant Steering Codes (DSC), Trunk Steering Codes (TSC), and NARS/BARS Access Codes (AC1, and AC2).

By using these CNI request options, CNI information will be available before the incoming R2MFC call is routed. This is necessary to provide CNI support for R2MFC DID to DPNSS1 gateway calls, but also provides an alternative for supporting CNI requests for incoming R2MFC calls in general. These options are applicable for incoming R2MFC DID/TIE calls.

Operating parameters

Meridian 1 DPNSS1 networks currently support only Coordinated Dialing Plan (CDP), Special Numbers (SPNs), and Basic Automatic Route Selection (BARS) (for outgoing calls), therefore the R2MFC to DPNSS1 feature does not support Universal Dialing Plans (UDPs).

For R2MFC DID calls routing to DPNSS1 trunks, the option Accept CNI (ACNI) is provided in the DPNSS1 route data block to identify if CNI information should be passed at the gateway. If the ACNI option is set to YES, the far end PBX must accept the Originating Line Identity (OLI) string for Called/Calling Line Category (CLC) Public Switched Telephone Network (PSTN) calls in the Initial Service Request Message (ISRM).

Two additional options are provided for requesting CNI before the R2MFC DID call is even routed as previously described. Using one of these options is the only mechanism that will provide CNI for an R2MFC-DPNSS1 gateway call (unless the call is the result of call redirection). If the interfacing Central Office cannot support such options, both of these options have to be disabled and therefore no CNI will be available for the R2MFC-DPNSS1 gateway call.

CNI is for R2MFC trunks tandeming to DPNSS1 trunks only. For calls originating from DPNSS1 trunks, and tandeming to an outgoing R2MFC trunk, the CNI information in the DPNSS1 call is not used. Existing methods of generating the CNI locally at the gateway node are used.

Interworking of R2MFC TIE trunks and DPNSS1 trunks is not supported for this feature.

Interworking between MFE DID/DOD trunks and DPNSS1 trunks is not supported by this feature. Interworking between MFE KD3 DID/DOD trunks and DPNSS1 trunks is also not supported by this feature.

External Operator Features and Toll Call Identification (from China Number 1 signaling) are not supported by this feature.

No new hardware is required for this feature.

Feature interactions

DPNSS1 Basic Call

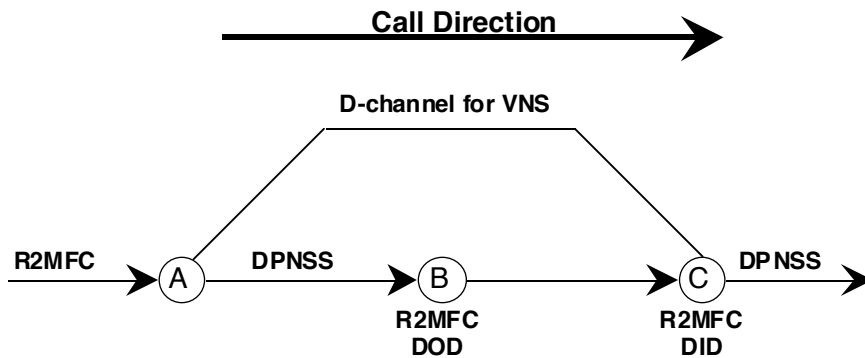
The R2MFC Gateway feature introduces a change in the content sent in the Initial Service Request Message (ISRM) when the originator of a DPNSS1 call is an R2MFC DID trunk. If CNI information has been obtained from the incoming trunk, the CNI digits are sent as an OLI string in the ISRM. In that case, the Trunk Identity (TID) string is not sent. If no CNI information is available from the originating trunk, the TID string is sent. The transport of the CNI digits as an OLI is controlled by the Accept CNI (ACNI) option on the outgoing DPNSS1 route.

At the terminating node of the DPNSS1 call, an OLI string instead of a TID string may now be received for calls that originate from trunks which are not using ISDN or Integrated Digital Access (IDA) signaling. The information available for call display is now different. If the terminating node cannot handle receiving an OLI from such trunk calls, the ACNI option should be set to NO on the outgoing DPNSS1 route at that gateway.

The Step Back on Congestion (SBOC) option programmed for an outgoing DPNSS1 route is ignored for R2MFC-DPNSS1 gateway calls. Specifically, if an R2MFC DID to DPNSS1 gateway call receives a Clear Request Message (CRM) due to congestion, the call is not rerouted (i.e., does not search for an idle trunk based on the next entry in the Route List Block), regardless of whether or not the SBOC option is programmed. Instead, the call is treated as a congested call and intercept is provided if necessary. If the SBOC option is allowed, there is a potential problem in the gateway signaling because the next outgoing route may not be a DPNSS1 route.

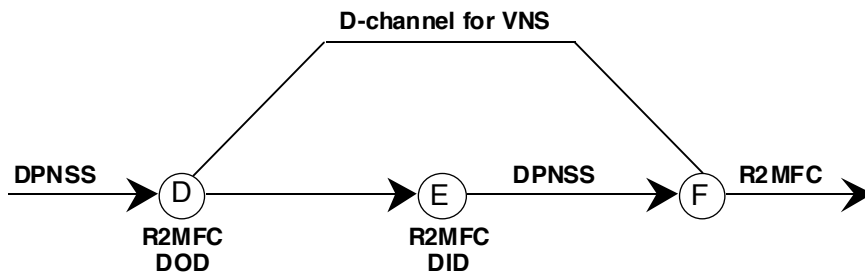
Virtual Network Services (VNS)

If the call on the DPNSS1 (or R2MFC) trunk is tandeming to the R2MFC or (DPNSS1) trunk on a VNS call, the R2MFC to DPNSS1 Gateway feature does not apply. If a DPNSS1/R2MFC tandem is encountered during the routing of a VNS call, the R2MFC to DPNSS1 Gateway feature applies. The following figure illustrates how the R2MFC-DPNSS1 gateway may apply to a VNS call.

Applicability of R2MFC-DPNSS1 Gateway to VNS Calls

For node A, the call is considered as an R2MFC to VNS call.
 For node C, the call is considered as a VNS to DPNSS call.
 In either case, the R2MFC-DPNSS gateway does not apply.

For node B, the call is considered to be a direct DPNSS to R2MFC connection. The R2MFC-DPNSS gateway applies.



For node D, the call is considered as a DPNSS to VNS call.
 For node F, the call is considered as a VNS to R2MFC call.
 In either case, the R2MFC-DPNSS gateway does not apply.

For node E, the call is considered to be a direct R2MFC to DPNSS connection. The R2MFC-DPNSS gateway applies.

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CDR Calling Line ID for DPNSS1

At the terminating PBX, the OLI string for the R2MFC DID originated call may contain the CNI information. The Release 20 CDR Enhancement feature automatically prints the contents of the OLI string in the CLID field of the CDR. Hence, the CNI information will be made available on the CDR (i.e., it will be printed in the CLID field).

Feature packaging

No new software option package has been introduced with this feature; however, the following packages are required at the gateway Meridian 1 to provide the basic DPNSS1 and R2MFC signaling functionalities: Integrated Digital Access (IDA) package 122; Digital Private Network Signaling System 1 (DPNSS1) package 123; and Multifrequency Compelled Signaling (MFC) package 128.

For network numbering the following packages are recommended: Coordinated Dialing Plan (CDP) package 59; Basic Automatic Route Selection (BARS) package 57; Pretranslation (PXLT) package 92; Incoming Digit Conversion (IDC) package 113; and Flexible Numbering Plan (FNP) package 160.

The CNI request enhancements are packaged under the existing Multifrequency Compelled Signaling (MFC) package 128.

Feature implementation

LD 16 – Two new prompts (NCNI and CNIE) are introduced in the route data block for defining CNI request options for the R2MFC route.

Prompt	Response	Description
REQ	NEW, CHG	New, or change.
TYPE	RDG	Route data block.
...		
TKTP	DID	Direct Inward Dialing trunk data block.
...		
MFC	YES	
...		
ALRM	...	

Prompt	Response	Description
NCNI	(O)-7	Request CNI after the defined number of digits are received. If NCNI = 0, the CNI request does not depend on the number of digits received. If NCNI is defined to be greater than the number of digits required for routing the call, the CNI will not be requested but the call will be routed.
CNIE	(NO), YES	Request CNI after an ESN code is dialed. This R2MFC route requests/(does not request) CNI after an ESN code is dialed. The ESN code could be a Distant Steering Code (DSC), a Trunk Steering Code (TSC), the NARS Access Code 1 (AC1) or the NARS Access Code 2 (AC2). NCNI and CNIE are prompted when the following occurs: the MFC package is equipped TKTP = DID or TIE MFC = R2MF, and the MFC signaling table is defined on the route. If NCNI > 0 and CNIE = YES, CNI is requested when either one of the conditions is first met.
CNIT	(NO), YES	Request CNI only if the dialed station has Malicious Call Trace Allowed (MCTA) Class of Service.

LD 16 – A new prompt (ACNI) is introduced in the route data block for defining whether the DPNSS1 route supports R2MFC CNI.

Prompt	Response	Description
REQ	NEW, CHG	New, or change.
TYPE	RDB	Route data block.
...		
TKTP	IDA	Integrated Digital Access.
...		
TTBL	...	
ACNI	(NO), YES	Accept Call Number Identification (CNI). This DPNSS1 route allows/(does not allow) R2MFC CNI sent as an Originating Line Identity (OLI) string in the initial service request message (ISRM). This is prompted when the following occurs: the IDA package is equipped the MFC package is equipped TKTP = IDA, and SIGL = DPN/APNS.

Feature operation

Not applicable.

Virtual Network Services in the UK with DASS2/DPNSS1 Bearers

Virtual Network Services (VNS) provides ISDN features to customers when no ISDN Primary Rate Interface (PRI) or ISDN Signalling Link (ISL) Bearer Channels are available between two Meridian 1 switches (please refer to the VNS feature description module in this NTP for detailed information on VNS.)

The Virtual Network Services with DASS2/DPNSS1 Bearers feature introduces VNS in the UK using Digital Private Network Signalling System No.1 (DPNSS1) or Digital Access Signalling System No.2 (DASS2) trunks as VNS Bearer trunks.

Operating parameters

All of the operating parameters that pertain to the Basic VNS feature also apply to the Virtual Network Services with DASS2/DPNSS1 Bearers feature (please refer to the VNS feature description module in this NTP.) The following parameters also apply.

Analog Private Networking Signalling System (APNSS) trunks cannot function as VNS Bearer trunks.

No DPNSS1 Supplementary Service is provided when DPNSS1 trunks are used as a VNS Bearer trunk. ISDN features are provided instead. If any of the DPNSS1 Supplementary Service features requires a DPNSS1 route, it cannot use a VNS route.

If ESN is configured, a route list entry with both VNS and DPNSS1 is not chosen.

For DPNSS1/VNS gateway nodes in mixed DASS2/DPNSS1 and VNS networks, the gateway nodes are subject to the same feature support and limitations as the standard DPNSS1/ISDN gateway without VNS. If there is no DPNSS1/ISDN gateway, the feature will be stopped at the DPNSS1/VNS node.

Feature interactions

Analog Private Networking Signalling System (APNSS)

APNSS trunks cannot function as VNS Bearer trunks.

Data calls

Data calls are supported on DPNSS1 or DASS2 VNS Bearer trunks if the DPNSS1 or DASS2 VNS Bearer trunks are configured to support data calls. Similarly, data calls are supported on DPNSS1 or DASS2 Bearer trunks in VNS to DPNSS1/DASS2 gateways, if the DPNSS1 or DASS2 VNS Bearer trunks are configured to support data calls.

DPNSS1 Attendant Call Offer

DPNSS1 Attendant Call Offer is not supported over VNS Bearer trunks (DPNSS1 Attendant Call Offer allows an attendant-extended call, routed over a DPNSS1 trunk, to be camped-on to a remote busy extension.) Standard ISDN Camp-on may be provided instead, if NAS is configured over the VNS Bearer trunks.

DPNSS1 Attendant Timed Reminder Recall and Attendant Three-Party Service

DPNSS1 Attendant Timed Reminder Recall and Attendant Three-Party Service are not supported over VNS Bearer trunks. If NAS is configured over the VNS Bearer trunks, NAS call extension and Attendant Recall will be offered instead.

DPNSS1 Call Back When Free and Call Back When Next Used

DPNSS1 Call Back When Free and Call Back When Next Used are not supported over VNS Bearer trunks. Network Ring Again or Network Ring Again on No Answer may be provided instead, if Network Ring Again or Network Ring Again on No Answer are configured over the VNS Bearer trunks.

DPNSS1 Diversion

DPNSS1 Diversion is not supported over VNS Bearer trunks. Network Call Redirection and Trunk Route Optimization can be provided instead, if configured over the VNS D-Channel.

DPNSS1 Extension Three-Party Service

DPNSS1 Extension Three-Party Service is not supported over VNS Bearer trunks. Network Call Redirection and Trunk Route Optimization can be provided instead, if configured over the VNS D-Channels.

DPNSS1 Loop Avoidance

DPNSS1 Loop Avoidance is not supported over VNS Bearer trunks (DPNSS1 Loop Avoidance prevents a call from being looped through a DPNSS1 network by placing a limit on the number of channels that a call can use.) The ISDN Call Connection Limitation is provided, if it is configured over the VNS D-Channel.

DPNSS1 Route Optimization

DPNSS1 Route Optimization is not supported over VNS Bearer trunks.

DPNSS1 Route Optimization/ISDN Trunk Anti-Tromboning Interworking

ISDN Trunk Anti-Tromboning may be applied to the VNS part of the call, if configured on the VNS D-Channel.

DPNSS1 Step Back On Congestion

DPNSS1 Step Back On Congestion handles high traffic situations when congestion is encountered by DPNSS1 trunks. The following scenarios apply for interworking with VNS.

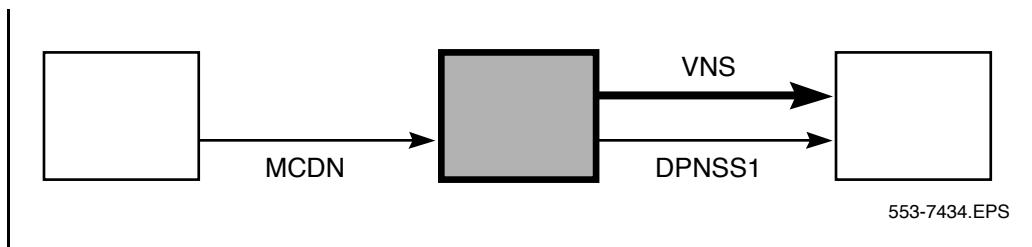
Homogeneous Networks

DPNSS1 Step Back On Congestion is supported over VNS Bearer trunks, if all the transit nodes within the DPNSS1 network used for VNS are configured accordingly:

- In LD 86, if the SBOC (Step Back On Congestion) prompt is set to NRR (No Reroute) or RRO (Reroute Originator), then it would be sufficient that the VNS originating node be configured with either RRO (Reroute Originator) or RRA (Reroute All).
- In LD 86, if the SBOC (Step Back On Congestion) prompt is set to RRA (Reroute All) for a transit node, then the different alternative routes at this node must be configured with VNS and must be configured as VNS Bearers.

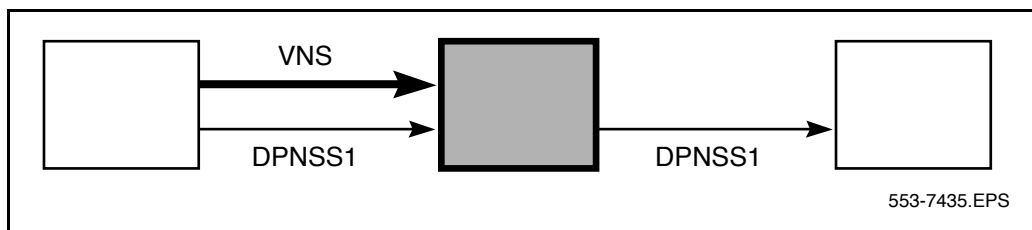
Hybrid Networks

MCDN/VNS with DPNSS1 node

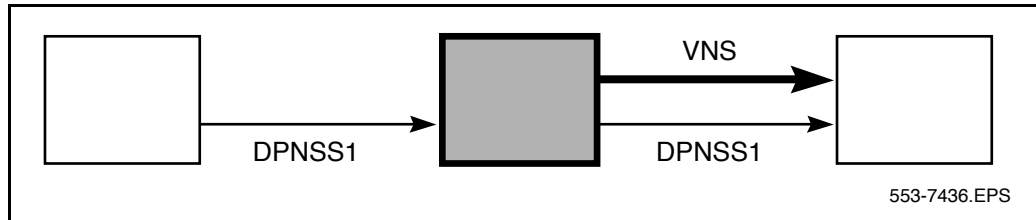


- If a congestion is encountered inside the VNS portion of the path, the node behaves as an MCDN/MCDN tandem. The ISDN Drop Back Busy (IDBB) and ISDN Off-Hook Queuing (IOHQ) are transmitted, so that they may be applied further along the VNS portion of the path, or at the tandem node.
- If a congestion is encountered within the DPNSS1 network, the VNS portion of the call is cleared and the disconnection is propagated back to the originating side of the MCDN path. Neither Drop Back Busy nor Off-Hook Queuing is activated at the tandem node, even if IDBB or IOHQ are activated.

VNS with DPNSS1/DPNSS1 node



This scenario is considered as an MCDN/DPNSS1 gateway. The functionality is the same as for the Step Back on Congestion feature, as documented in DPNSS1 Product Information Guide 553-3921-100.

DPNSS1/VNS with DPNSS1 node

- If a congestion is encountered inside the VNS portion of the path, the VNS portion of the call is cleared and the disconnection is propagated back to the originating DPNSS1 side. The Step Back on Congestion feature is invoked, if it is configured.
- If a congestion is encountered the within the DPNSS1 portion of the path, with the DPNSS1 trunk being used as a VNS Bearer, the VNS portion of the call is cleared and a normal disconnection is propagated back to the originating DPNSS1 side. The Step Back on Congestion feature is not invoked, even if it is configured.

DPNSS1 Executive Intrusion

DPNSS1 Extension Three-Party Service is not supported over VNS Bearer trunks. Attendant Break-in may be provided instead, if NAS is configured over the VNS Bearer trunks.

Standalone Meridian Mail

Standalone Meridian Mail is not supported over VNS Bearer trunks. A mailbox user may access Meridian Mail, if the ISDN Network Message Services is configured.

DPNSS1 Enhancements for ISDN Interworking

The Generic X11 Release 20B software introduced enhancements to allow DPNSS1 to interwork with QSIG and EuroISDN. At an ISDN gateway, ISDN information may carried into some DPNSS1 messages, if DPNSS_189I package 284 is equipped.

DPNSS1/DASS2 to ISDN PRI Gateway

A VNS call over a DPNSS1 or DASS2 Bearer trunk of an DPNSS1/DASS2 to ISDN PRI Gateway acts as the ISDN leg of the Gateway.

Feature packaging

For total feature functionality, the following packages are required:

- Virtual Network Services (VNS) package 183;
- Network Alternative Route Selection (NARS) package 58;
- Network Class of Service (NCOS) package 32;
- Basic Routing (BRTE) package 14;
- Integrated Services Digital Networking (ISDN) package 145;
- ISDN Signaling Link (ISL) package 147;
- Advanced Network Services (NTWK) package 148;
- Integrated Digital Access (IDA) package 122;
- 2 MBit Primary Rate Interface (PRI2) package 154;
- Digital Private Network Signaling System No.1 (DPNSS) package 123, for routes using DPNSS1 signaling; and
- Digital Access Signaling System No.2 (DASS2) package 124, for routes using DASS2 signaling.

For ISDN to DPNSS1/DASS2 gateway:

- International Supplementary Features (SUPP) package 131; and
- Network Attendant Service (NAS) package 159.

For the Step Back on Congestion Supplementary Service feature:

- DPNSS1 Network Services (DNWK) package 231.

The following packages may also be used:

- Universal ISDN Gateway (UIGW) package 283; and
- ISDN SIS (BTNR-I on DPNSS1), (DPNSS1_189I) package 284.

Feature administration

LD 17- Configure the VNS D-Channel to be associated with the VNS route.

The D-Channel should be associated with each node and customer, that is, both ends of the D-Channel link should be configured.

Prompt	Response	Description
REQ	CHG, END	Change data, or exit the Overlay.
TYPE	ADAN	Action Device and Number.
ADAN	CHG DCH 0-15	Change the D-Channel.
USR	VNS SHAV	VNS=Dedicated D-Channel SHAV=Shared D-Channel.
VNSM	0-300	The maximum number of VNS channels supported by the D-Channel. Note: This is the potential VNS capability for the D-Channel, and is not associated with any other restriction placed on the VNS capability, such as the number of VNS Virtual DNs.
VNSC	xx	At least one D-Channel must be configured with USR=VNS or USR=SHAV.
VNSP	0-32700	Private Network Identifier (PNI) of the far-end customer.
VCNA	(NO) YES	Network Call Party Name Display is (not) available over the D-Channel.
VCRD	(NO) YES	Network Call Redirection is (not) available over the D-Channel.
VTRO	(NO) YES	Trunk Route Optimization Before Answer is (not) available over the D-Channel.

LD 96 - Enable the D-Channel that has been configured in LD 17.

Prompt	Response	Description
...	ENL DCH 0-15	Enable the D-Channel.

LD 79 - Define the VNS DNs for both nodes/customers to be associated with the D-Channel configured in LD 17. You may add a new individual VDN to an existing VNS VDN block, or create a new VNS VDN block.

Prompt	Response	Description
REQ	NEW	Add an individual VDN to an Create a new VNS data block.
TYPE	VNS	Virtual Network Services.
CUST	xx	Customer number. At least one D-Channel must be configured with USR=VNS or USR=SHAV and having VNS=Customer number.
VNDN	xxxxxxx 1-4000 xxxxxxx <CR>	Individual VDN to be added. 1-4000=number of contiguous VDN to be added, xxxxxxx=first VDN to be added. You may add another single VDN by entering <CR> (VDN is prompted until <CR> is entered.) Note: For the above entries, the VDNs must be pat of the customer's numbering plan.

LD 16 – Set up the VNS Bearer Trunk.

The Bearer trunk should be associated with each node and customer, that is, both ends of the Bearer link should be configured.

Prompt	Response	Description
REQ	NEW, CHG	Add, or change data.
CUST	xx	Customer number. At least one D-Channel must be configured with USR=VNS or USR=SHAV, in LD 17.
ROUT	0-511	The number of the route to be associated with the VNS Bearer Channel.
CNTL	YES	Change controls or timers.
TIMR	VSS (0)	0=Do not answer the Bearer channel until the terminating party answers.
	1	1=Answer the Bearer channel immediately on arrival.
	2-1023	2-1023=Answer the Bearer Channel after specified seconds (rounded down to multiple of two seconds) if the terminating party has not already answered.
TIMR	VGD 0-(6)-31	Enter the guard timer on the associated VNS DN (the time allowed for the Bearer trunk call to disconnect, in seconds)
VRAT	(NO) YES	(Do not) immediately answer the attendant extended VNS call on the incoming Bearer trunk.

LD 86 – Configure the VNS trunk route.

Prompt	Response	Description
REQ	NEW, CHG	Add, or change.
CUST	xx	Customer number.
FEAT	RLB	Route list data block feature.
RLI	0-MXRL	The Route List Index to be associated with the VNS Bearer Channel.
ENTR	0-63	The entry within the Route List Index to be associated with the VNS Bearer Channel.
ROUT	0-511	The number of the Route to be associated with the VNS Bearer Channel.
VNS	YES	Virtual Network Services.
- VDCH	0-15	The D-Channel used for VNS call. At least one D-Channel must be configured with USR=VNS or USR=SHAV, in LD 17.
- VDMI	(0) 1-31 1-255 0-999	VNS Digit Manipulation Table to be used on the VNS D-Channel. None With CDP. With NARS/BARS. With Flexible Numbering Plan.
-VTRK	1-(20)-100	Number of VNS trunks allowed on the VNS route.
...		
DMI	(0) 1-31 1-255 0-999	VNS Digit Manipulation Table to be used on the VNS Bearer. None. With CDP. With NARS/BARS. With Flexible Numbering Plan.

Feature operation

There are no operating procedures specified for this feature.

DASS2 feature descriptions

Call Charge Information

Call cost information is sent to the originating PBX at the end of a call.

Calling/Called Line Identity

Identity of the calling and called parties being sent over DASS2 trunks within limitations of any connected or interconnecting circuits.

Category 1 Call

A voice or data call that must be on a 64 Kbit digital channel. This call may use the swap feature and may not be intruded upon.

Category 2 and Telephony Calls

With a category 2 call it is desirable to use a digital channel but not essential and swap may be used if digital. A digital channel is not essential for a telephony call, swap may not be used and some supplementary features may be barred.

Closed User Group

A group that are only allowed to communicate with each other but allowing for some members to make or receive calls from outside the group.

Maintenance Procedure

This provides just a status information interchange.

Network Address Extension

DASS2 allows additional routing information of up to 6 digits to pass through transparently when required by connected PBX systems.

User to User Signalling and Tie Line Signalling

Customers can send signals over the link during a call to provide services. One or more channels of a DASS2 link may be allocated as a permanent circuit to another PBX to provide a tie line.

Feature operation

The use of NT hardware does not change the operation of the existing DPNSS1/DASS2 feature.

Data Calls

A Service Indicator Code with the following characteristics is sent if a call is originated from a data set:

Type of Data:	Data:
Data Rate	64 kbit/sec
Data Type, Asynchronous	1 stop bit
Data Mode	Full Duplex
Details of Data	Asynchronous
Data Format	8 data bits
Flow Control	TA does not have capability

DPNSS1/DASS2 Limitations

The DPNSS1/DASS2 Protocols have the following limitations:

- BARS/NARS is not supported for incoming DPNSS1 calls
- DASS2 Swap Supplementary Service is not supported
- The Meridian 1 supports most Supplementary Services as a transit node. However, Step Back On Congestion, Priority Breakdown and Loop Avoidance are not supported. In these cases the Meridian 1 appears transparent, passing the Supplementary Service Request through to the next DPNSS1 node.

DPNSS1/DASS2 Package Requirements

The following options comprise the DASS2/DPNSS1 Protocols software set:

Feature	Option
ISDN	145
PRI2	154
IDA	122
DNWK	231
DPNSS1	123
DASS2	124

DASS2/DPNSS1 Packaging Notes

- A Meridian 1 with routes using Level 3 DPNSS1 signaling requires the IDA and DPNSS1 packages.
- A Meridian 1 with routes using Level 3 DASS2 signaling requires the IDA and DASS2 packages.
- In a network which uses DPNSS1 and DASS2, the DASS2 package is required only at those Meridian 1s which are serving as DASS2 gateways.
- A semi-permanent connection across the ISDN requires the DPNSS1 package rather than the DASS2 package.
- Interworking of Network Ring Again across DPNSS1 and Q.931 requires the NRAG package.

Chapter 4 — Network Clocking

Overview

Need for synchronization

When digital signals are being transported over a communication link, the receiving end must operate at the same frequency (data rate) as the originating end to prevent loss of information. This is referred to as link synchronization. If both ends of a communication link are not in synchronization, data bit slips occur and therefore a loss of data results. In general, accurate timing is very important, but more importantly synchronized timing is a must for reliable data transfer.

When only two Meridian 1 switches are interconnected, synchronization can be achieved by operating the two systems in a master/slave mode whereby one system derives its timing from the other. However, in a network of digital systems, slips can be better prevented by forcing all digital systems to use a common reference clock (see Figure 119).

Synchronization methods

There are two common methods of maintaining timing coordination between switching systems.

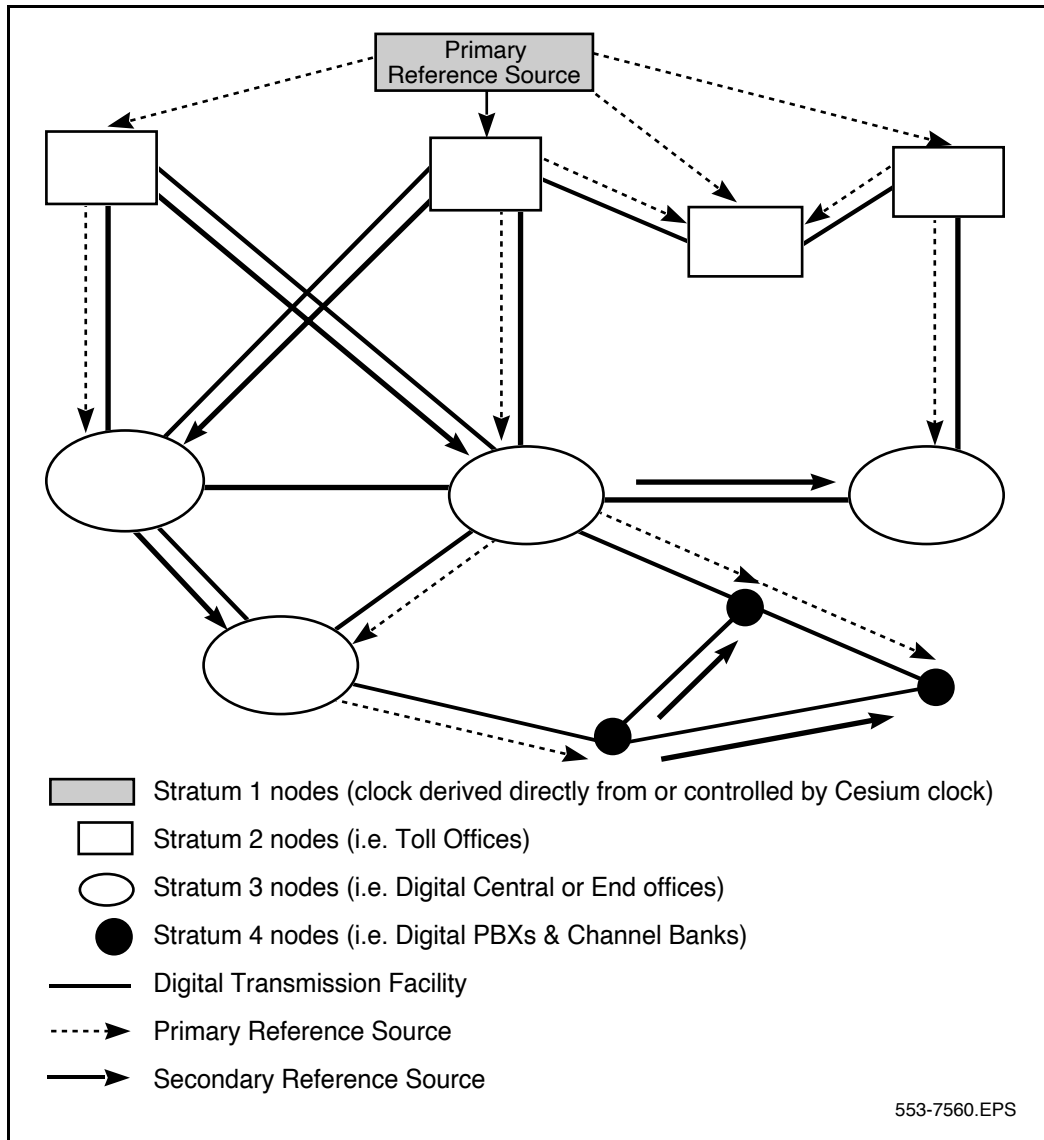
- a Plesiosynchronous operation:** nodal clocks run independently (free run) at the same nominal frequency. There are frequency differences between clocks resulting in frame slips (see section “Frame Slips”). The magnitude of frame slips are directly proportional to the frequency difference. Slips are inevitable but can be minimized by using very stable clocks and elastic stores or buffers. These buffers are capable of absorbing a certain number of data bits to compensate for slight variances in clock frequencies.
- b Mesosynchronous operation:** nodal clocks are continuously and automatically locked to an external reference clock. With this method, frame slips can be eliminated if elastic stores are large enough to compensate for transmission variances.
Mesosynchronous operation is virtually slip free.

Whenever possible the Meridian 1 PBX uses the Mesosynchronous mode of operation by using the Clock Controller circuit cards to lock onto an external reference source (i.e. the Central Office, another Meridian 1 PBX, etc.). The above statement is true unless the Meridian 1 is used as a Master in an independent/private network (no digital links to a higher Node Category).

Hierarchical synchronization

Figure 119 provides a general view of a Digital Network Synchronization including the four stratum levels Node Categories of clocks (Stratum 1 being the highest--most accurate to Stratum 4 being the lowest). Meridian 1 clocking meets Node Category E--Stratum 4 requirements. Also shown are ways of providing a Secondary Clock Source while preventing timing loops.

Figure 119
Hierarchical Synchronization



Stratum Levels

In a digital network, nodes are synchronized using a priority master/slave method. Digital nodes are ranked in Stratum levels 1 to 5. Each node is synchronized to the highest ranking node in its neighborhood with which it has a direct link.

	Stratum 2	Stratum 3	Stratum 4
Accuracy	$\pm 1.6 \times 10^{-8}$ Hz	$\pm 4.6 \times 10^{-6}$ Hz	$\pm 3.2 \times 10^{-5}$ Hz
Holdover	1×10^{-10} per day	≤ 255 frame slips in 1st 24 hours	Not Required
Hardware Duplication	Required	Required (Note 1)	Not Required
MTIE During Rearrangement	MTIE ≤ 1 msec Phase Change Slope: ≤ 81 ns in any 1.326 msec	MTIE ≤ 1 msec Phase Change Slope: ≤ 81 ns in any 1.326 msec	No Requirement (Note 2)
Pull-in Range	3.2×10^{-8} Hz	9.2×10^{-6} Hz	6.4×10^{-5} Hz
Dedicated Timing Required	Required	Required	Not required

Note 1: Non-duplicated clock hardware that meets all other stratum 3 requirements is referred to as stratum 3ND.

Note 2: Stratum 4 clock hardware that meets MTIE requirements during rearrangements is referred to as 4E.

Frame Slip

Digital signals must have accurate clock synchronization for data to be interleaved into or extracted from the appropriate timeslot during multiplexing and demultiplexing operations. A Frame Slip is defined (for 2 Mb links) as the repetition of, or deletion of the 256 data bits of a CEPT frame due to a sufficiently large discrepancy in the read and write rates at the buffer (clocks aren't operating at exactly the same speed).

When data bits are written into (added to) a buffer at a slightly *higher* rate than that at which they are being read (emptied), sooner or later the buffer overflows. This is a slip-frame deletion.

In the opposite situation, when data bits are written (added) into a buffer at slightly *lower* rate than that at which they are being read (emptied), eventually the buffer runs dry or underflows. This is also a slip-frame repetition.

A 2 Mb PRI contains a buffer large enough to contain 2 full frames ($256 \times 2 = 512$ bits), and is normally kept half full (1 frame). See the following table for the impact of one slip on various types of data.

All of the degradations shown in the following table can be controlled or avoided by proper clock (network) synchronization.

Performance Impact of one Slip.

Service	Potential Impact
Encrypted Text	Encryption key must be resent.
Video	Freeze frame for several seconds. Loud pop on audio.
Digital Data	Deletion or repetition of Data. Possible Misframe.
Facsimile	Deletion of 4-8 scan lines. Drop Call.
Voice Band Data	Transmission Errors for 0.01 to 2 s. Drop Call.
Voice	Possible Click

Guidelines

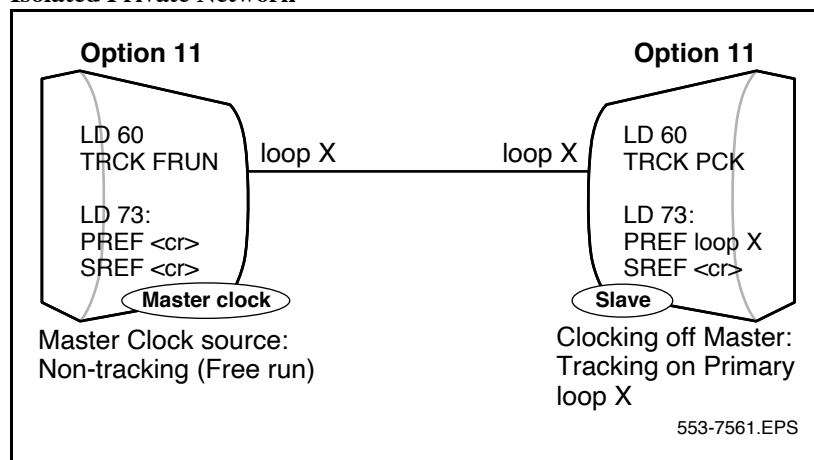
Some key points to keep in mind when designing Network Synchronization:

- Where possible, the Master Clock Source should always be from a Node Category/Stratum with higher clock accuracy — i.e. PBX connected to the C.O.; the CO is the Master and the PBX is the Slave.
- The source should not be in free-run itself (providing its own clock) unless it is operating in a fully independent network where the source acts as a Master (see Plesiosynchronous operation).
- When connecting two PBXs together (no CO connections), the most reliable PBX should be the Master. Reliability here refers to Dual CPU/Dual Clock, battery back-up or stratum level of the clock controller.

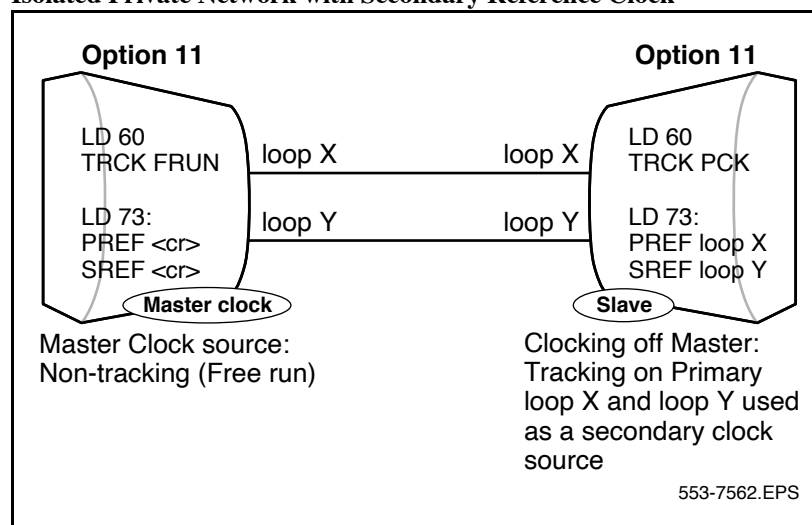
- Avoid timing loops. A timing loop occurs when a clock using as its reference frequency a signal that it itself traceable to the output of that clock. The formation of such a closed timing loop leads to frequency instability and is not permitted. Timing loops are sometimes unavoidable on the secondary clock reference source.
- Ensure all CO/PBX links used as clock references have a traceable path back to the same stratum 1 clock source.

While it is beyond the scope of this guide to provide detailed Network Synchronization, the following examples illustrate some of the basic concepts to achieve stable clocking.

Example 1
Isolated Private Network

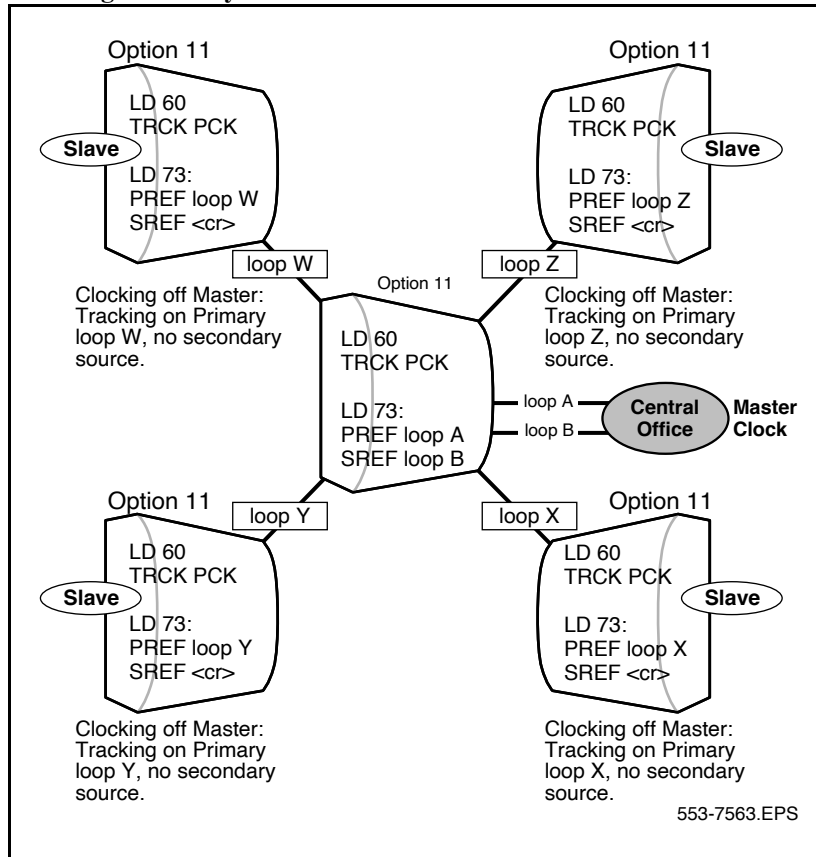


In this example, there is no digital connection to the Central Office.

Example 2**Isolated Private Network with Secondary Reference Clock**

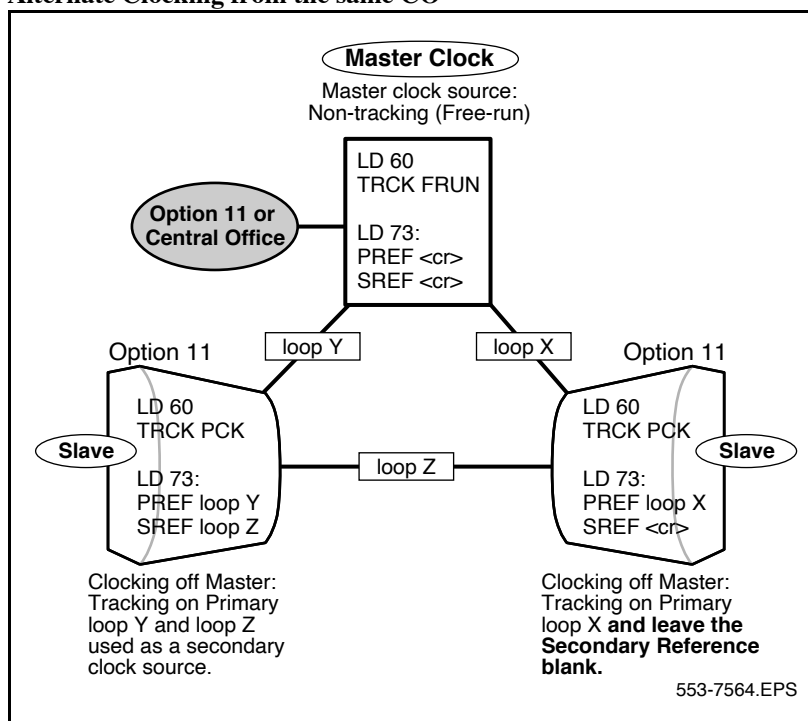
In this example, there is no digital connection to the Central Office. For tie lines between PBXs facilitated by a central office, clocking is derived from the PBX, not the CO. When a second Digital loop is available, it can be used as a Secondary Clock source in case the Primary Source fails.

Example 3
Clocking Hierarchy referenced to a Public Network Master Clock



This is an example of a “STAR” arrangement— one Hub PBX is linked to the Central Office and all other PBXs are connected as slaves. When a second Digital loop from the Meridian 1 which forms the hub of this network becomes available, it can be used as a Secondary Clock Source in case the Primary Source fails.

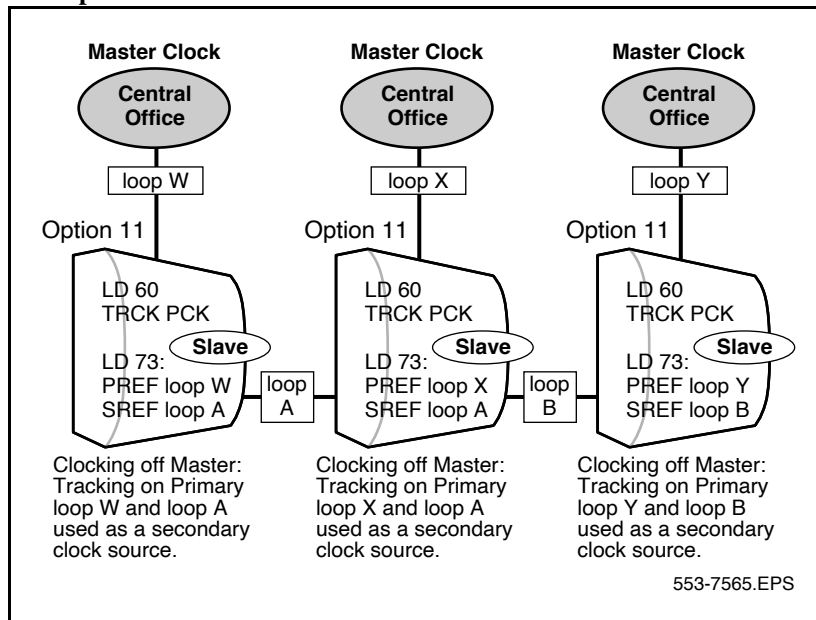
Example 4 Alternate Clocking from the same CO



In this case, a digital connection to the Central Office may exist (i.e. Loops X and Y). When a second Digital loop from the CO or Master M-1 becomes available, it can be used as a Secondary Clock Source in case the Primary Source fails.

To avoid timing loops, in example 4-4 the most reliable slave system should not have a Secondary Clock Source (SREF= <cr>). In this example, this is illustrated by the node which supports loops X and Z.

Example 5

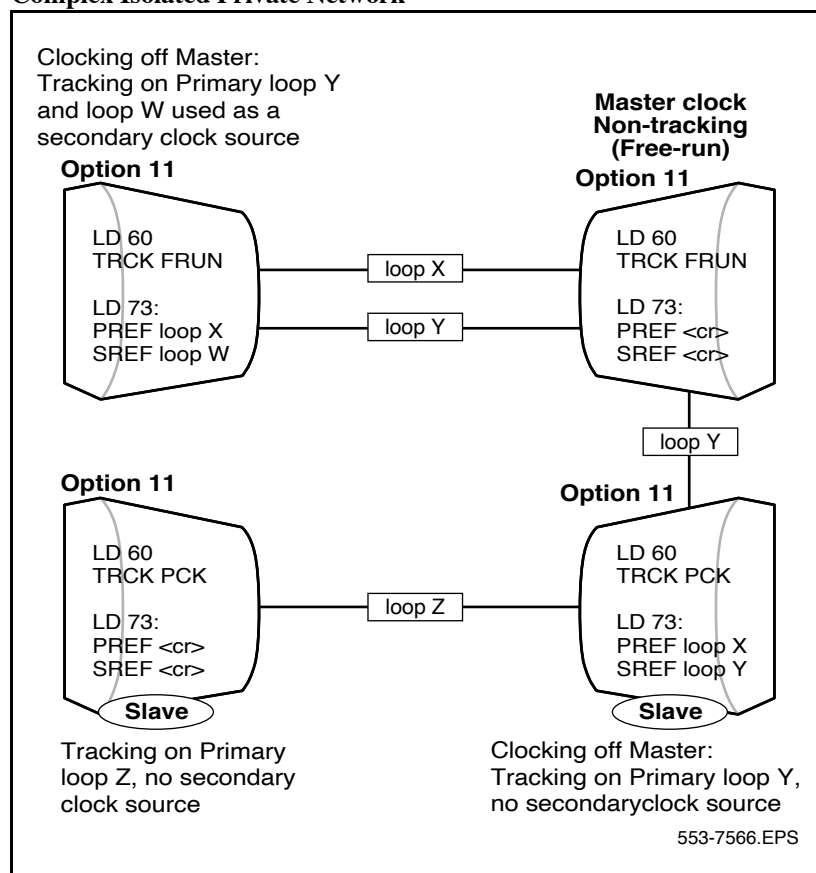


Here, digital connections to the Central Office do exist. When a second Digital loop from the CO becomes available, it can be used as a Secondary Clock Source in case the Primary Source fails.

Slaves can track on each other as a secondary source since the chances of both links to the Central Offices going down at the same time are minimal (very unlikely).

All central offices must have a path back to the same stratum 1 source.

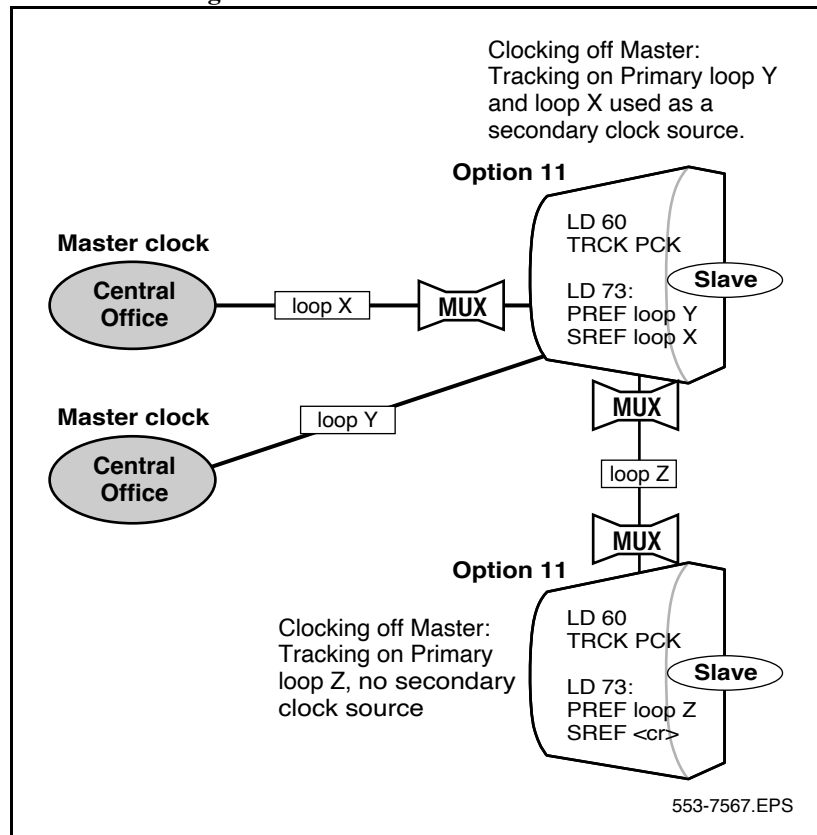
Example 6 Complex Isolated Private Network



Digital connections to the Central Office do not exist in this example. If it does, the PBX connected to it will track off the CO and will in turn be used as a clock source to other nodes.

When a second Digital loop from the Master Meridian 1/SL-1 becomes available, it can be used as a Secondary Clock Source in case the Primary Source fails.

Example 7
Network Clocking with MUX



The direct connection to the CO (without a MUX) should be used as a primary clock reference since there is the least amount of hardware involved. The MUX must pass the clock and not generate its own clock; in other words, it must also be a slave (not Free Run). Synchronized clocking is required.

Digital Trunks

Synchronization, Option 11

Digital trunk synchronization is provided via a Clock Controller on the digital trunk pack.

There are two modes of operation:

Tracking Mode

In tracking mode, the Primary Rate Interface (PRI) or Digital Trunk Interface (DTI) loop supplies an external clock reference to the on-board clock controller. Two PRI or DTI packs can operate in tracking mode, with one defined as the primary reference source for clock synchronization, the other defined as a secondary reference source. The secondary reference acts as a back-up to the primary reference.

Free Run (Non-Tracking Mode)

The clock synchronization for a PRI loop may operate in free-run mode if:

- the loop is not defined as the primary or secondary clock reference
- the primary and secondary references are disabled
- the primary and secondary references are in a local alarm state

For Option 11 systems, a single cable supports either PRI or DTI;

PRI/DTI — LTU 20 ft NTBK05CA

Option 11 clock controllers

Digital trunking requires synchronized clocking so that a shift in one clock source will result in an equivalent shift of the same size and direction in all parts of the network.

The Option 11 system supports a single clock controller (CC) located on either:

- the NTAK10 2 Mb DTI
- the NTAK79 2 Mb PRI
- the NTBK50 2 Mb PRI
- the NTBK22 MISP (BRI applications).

This clock controller can operate in one of two modes: tracking, or non-tracking (also known as free-run).

The Clock Controller circuitry synchronizes the Option 11 to an external reference clock and generates and distributes the clock to the system. This enables the Option 11 to function either as a slave to an external clock or as a clocking master.

Note: When configuring ISL over analogue trunks, clock controllers are not required.

Tracking mode

In tracking mode, a reference clock is supplied to the clock controller. The clock controller uses this reference to adjust the system clock so that the two are of the same frequency. The CC is capable of tracking to a primary or secondary reference supplied on the DTI/PRI/BRI circuit card, or to an external reference clock.

When tracking using a reference clock derived from a DTI/PRI/BRI source, an optional secondary clock source can be defined. The primary clock source is derived from the clock controller's host circuit card. The secondary source may be defined as any other DTI/PRI/BRI installed in the Option 11 system. The secondary reference acts as a back-up to the primary reference.

There are two stages to clock controller tracking:

- tracking a reference, and
- locked onto a reference.

When tracking a reference, the clock controller uses an algorithm to match its frequency to the frequency of the incoming clock. When the frequencies are very near to being matched, the clock controller is locked onto the reference. The clock controller will make small adjustments to its own frequency until both the incoming and system frequencies correspond.

If the incoming clock reference is stable, the internal clock controller tracks it, locks onto it, and matches frequencies exactly. Occasionally, however, environmental circumstances cause the external or internal clocks to drift. When this happens, the internal clock controller briefly enters the tracking stage.

If the incoming reference is unstable, the internal clock controller is continuously in the tracking stage. This condition does not present a problem, rather, it shows that the clock controller is continually attempting to lock onto the signal. If slips are occurring, however, it means that there is a problem with the clock controller or the incoming line.

Free-run (non-tracking)

In Free-Run (Non-tracking) mode, the clock controller does not synchronize on any source, it provides its own internal clock to the system. This mode can be used when the Option 11 is used as a master clock source for other systems in the network. Free-run mode is undesirable if the Option 11 is intended to be a slave. It can occur, however, when both the primary and secondary clock sources are lost due to hardware faults. It can also be turned on manually using software commands.

Chapter 5 — 2 Mb DTI Implementation

Overview

This chapter provides the information required to install 2 Mb DTI on Meridian 1 Option 11 systems, including:

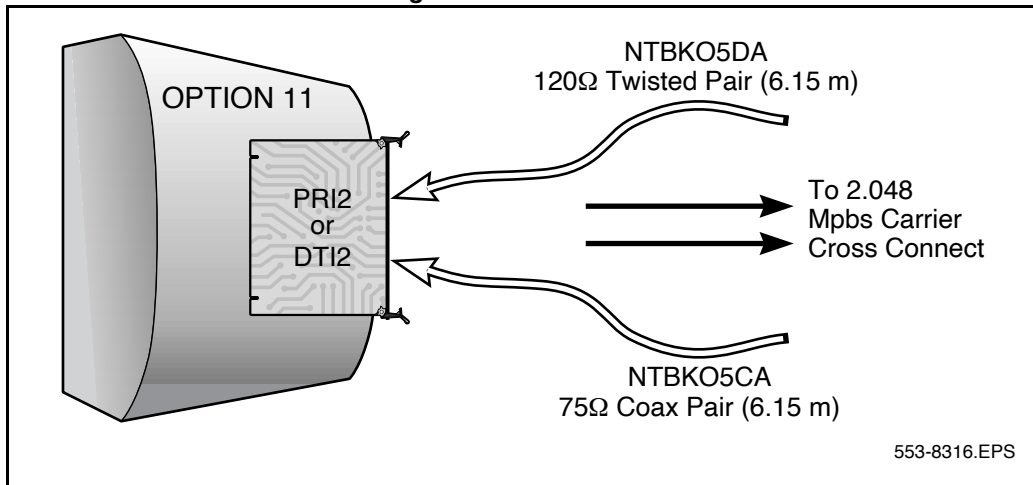
- hardware installation
- software programming

Hardware Requirements

Hardware requirements for 2 Mb DTI are as follows:

- 2 Mb DTI Circuit card - NTAK10
(a Clock Controller is incorporated into the circuit card)
- CEPT Cable - NTBK05DA (120 $\frac{3}{4}$ twisted pair—6.15 m length)
or
CEPT Cable - NTBK05CA (75 $\frac{3}{4}$ coaxial—6.15 m length)

Figure 120
2 Mb DTI Cabling



NTAK10 2 Mb DTI card

The 2 Mb Digital Trunk Interface card provides the physical interface for the digital E-1 carrier on the Option 11 system. The card includes an on-board clock controller and is installed in slots 1 through 9 in the main cabinet. The NTAK10 2 Mb DTI card is also used for ISL shared mode applications.

For information on the NTAK10 faceplate LEDs, refer to the chapter in this guide titled “DTI *maintenance*”. For more information on the operation of the on-board clock controller, refer to the chapter in this guide called “Network clocking”.

Installing DTI hardware

The NTAK10 circuit card can be installed in any available card slot 1 - 9 in the main cabinet.

Inspecting the NTAK10 circuit card

Inspect the circuit card before installing it in the main cabinet:

- Locate the NTAK10 2 Mb circuit card and carefully remove it from its packaging.
- Inspect the circuit card for any visible damage which may have occurred during shipping.

Setting the switches

The NTAK10 incorporates four surface mounted dip switches. The following tables provided information on the various settings and related functions of these switches.

Note: The ON position for all the switches is towards the bottom of the card. This is indicated by a white dot printed on the board adjacent to the bottom left corner of each individual switch.

Set the switches found on the circuit card as per the requirements of your specific installation:

Switch S1—Clock Controller Configuration

This switch enables/disables the on-board Clock Controller.

Switch	Off (Up)	On (Down)
S1-1	Spare	Spare
S1-2	Clock Controller Enabled	Clock Controller Disabled

Switch S2—Carrier Impedance Configuration

This switch sets the carrier impedance to either 120 Ω or 75 Ω . Twisted pair cable is usually associated with 120 Ω . Coaxial cable is usually associated with the 75 Ω setting.

Switch	Off (Up)	On (Down)
S2-1	120 Ω	75 Ω
S2-2	75 Ω	120 Ω

Switch S3—Mode of Operation

This switch selects the mode in which the NTAK10 operates. The NTAK10 supports firmware that allows it to operate in the standard CEPT format mode, or the modified CEPT format used in France.

Switch	Off (Up)	On (Down)
S3-1	Non-French Firmware	French Firmware
S3-2	Spare	Spare

Switch S4—Carrier Shield Grounding

This switch allows for the selective shield grounding of the Tx and/or Rx pairs of the carrier cable. Closing the switch (down position) applies Frame Ground (FGND) to the coaxial carrier cable shield, creating a 75 Ω unbalanced configuration. The Tx and Rx pairs are referenced with respect to the 2 Mb DTI card (i.e. Rx is carrier received from the far end device).

Switch	Off (Up)	On (Down)
S4-1	Receive Shield Unconnected	Frame Ground on Receive Shield
S4-2	Transmit Shield Unconnected	Frame Ground on Transmit Shield

Note: The usual method is to ground the outer conductor of the receive coax signal.

Inserting the NTAK10 into the main cabinet

Slide the circuit card into any unused slot (1 - 9) in the main cabinet. Secure the circuit card in the cabinet by locking the lock latch assemblies.

Connecting the cables

- In the cabling area, located directly below the card cage, remove the retaining bar used to secure the MDF cables. Connect the NTBK05DA/CA interface cable to the 50-pin Amphenol connector below the card slot holding the NTAK10 2 Mb DTI circuit card. Re-install the retaining bar to secure the cable(s) in place.
- Terminate the NTBK05DA/CA carrier cable as required.

NTBK05DA pinouts

The pinouts for the NTBK05DA cable are as follows:

From: 50 pin MDF connector	To: 9 pin connector	Colour	Signal
pin 23	pin 6	Black	R0
pin 48	pin 7	White	T0
pin 50	pin 9	Bare	R0/T0 FGND
pin 24	pin 2	Black	R1
pin 49	pin 3	Red	T1
pin 25	pin 5	Bare	R1/T1 FGND

NTBK05CA pinouts

The pinouts for the NTBK05CA cable are as follows:

From: 50 pin MDF connector	To: Transmit coax connector	To: Receive coax connector	To: 50 pin MDF connector
pin 23	Inner conductor	—	—
pin 48	outer conductor	—	—
pin 24	—	Inner conductor	—
pin 49	—	outer conductor	—
pin 21	—	—	pin 49
pin 46	—	—	pin 48

DTI software programming

The following information describes the process required to program the 2 Mb DTI feature in the Meridian 1 Option 11 software.

The information is presented in the sequence in which it must be programmed. As an example, 2 Mb DTI loops must be configured before defining any error detection thresholds, loop timers, pad information or ABCD signaling bit information.

Procedure summary

Step	Overlay	Action
1	LD 17 Configuration Record	Add a 2 Mb DTI loop.
2	LD 73 Digital Data Block	Define the 2 Mb DTI ABCD signaling bit tables
3	LD 73 Digital Data Block	Define the 2 Mb DTI pad tables
4	LD 73 Digital Data Block	Define the 2 Mb DTI loop timers
5	LD 73 Digital Data Block	Define the 2 Mb DTI system timers
6	LD 16 Route Data Block	Define service routes.
7	LD 14 Trunk Data Block	Define the associated list of service trunks.

The prompts and responses are explained in the text that follows. Responses in parentheses are default values throughout the procedure.

Step 1: Add a 2 Mb DTI loop. Use Overlay 17.

The allowable range of base-cabinet slot numbers for the DTI circuit card is 1-9. The CPU/CONF circuit card is provisioned in slot 0, and the TDS card is provisioned in slot 1, effectively leaving slots 2-9 available to these interface card.

Prompt	Response	Description
REQ	CHG	
TYPE	CFN	configuration data block
•		
PARM	YES	
•		
PCML	(MU) A	System PCM law. Default is MU law
•		
•		
CEQU	YES	
•		
•		
DTI2	//	Enter a loop number for 2 Mb DTI.

Step 2: Define the 2 Mb DTI ABCD signaling bit tables. Use Overlay 73 to allow the implementation and administration of the 2 Mb DTI software and hardware.

The “abcd” response represents the following trunk supervisory signals

- Steady signals: bits 0, 1 or U (do-not-care). If c or d are not input they default to 0 and 1 respectively.
- Pulsed signals: P (pulsing), X (not pulsed) or U⁴ (do-not-care).

The following prompts are given when FEAT = ABCD:

Prompt	Response	Description
REQ	NEW,CHG, OUT, PRT, END	
TYPE	DTI2	2 Mb DTI
FEAT	ABCD	Request the digital signaling category. Refer to NTP 553-2911-200 for default abcd table with suggested values.
SICA	2-16	Signaling category
	1	If REQ = PRT then 1 must be input to print default table
	<CR>	If REQ = PRT all signaling tables are printed
TNLS	YES, (NO)	List of trunk TNs using requested SICA tables will (will not) be printed following the table
DFLT	(1)-16	Default signaling category to be used for default values JDMI defaults to 16 with <CR> ³ .

Incoming/Outgoing Calls:

IDLE(S)	abcd	(send) idle signal bits
IDLE(R)	abcd	(receive) idle signal bits
FALT(S)	abcd	(send) bits. 2 Mb DTI out-of-service
	N	if FALT (send) signal not required
FALT(R)	abcd	(receive) bits. 2 Mb DTI out-of-service
	N	if FALT (receive) signal not required

Incoming Calls:

SEZ(R)	abcd	seize signal (send or receive) for voice or data calls from or to a non-SL-1.

Prompt	Response	Description
SEZD(R)	abcd	seize signal (send or receive) for data calls between SL-1s.
	N	if SEZD(R) signal not required
SEZV(R)	abcd	seize signal (send or receive) for voice calls.
	N	if SEZV(R) signals not required
P CALL(R)	abcd	(receive) signal sent during seize by an incoming CO trunk.
TIME	ON, OFF	length of pulse time on, and time off. (default 2 seconds on, 8 seconds off)
SEZA(S)	abcd	seize signal acknowledgment (send).
	N	if SEZA(S) signal not required
P WNKS(S)	abcd	wink start (corresponds to a pulsed seize acknowledgment). Prompted when SEZA(S) not required.
	N	if WNKS(S) signal not required
TIME	10-(220)-630	time for WNKS(S) signal in milliseconds
P DIGT(R)	abcd	(receive) decadic pulses
	N	if DIGT(R) not required
NRCV(S)	abcd	number received signal (send).
	N	if NRCV(S) signal not required
P EOSF(S)	abcd	end of selection free (send)
	N	if EOSF(S) not required
TIME	(100)-150	time for EOSF(S) in milliseconds
P EOSB(S)	abcd	end of selection busy (send)
	N	if EOSB(S) not required
TIME	(100)-150	time for EOSB(S) in milliseconds
P OPCA(R)	abcd	operator calling time (receive) signal
	N	if OPCA(R) not required
TIME	64-(128)-192	time of OPCA(R) pulse in milliseconds
REPT	(1)-5	number of OPCA(R) pulses.
CONN(S)	abcd	connect send

Prompt	Response	Description
CONN(R)	abcd	connect receive
P RRC(S)	abcd	register recall (send) signal. Activated by Malicious Call Trace.
	N	if RRC(S) not required
TIME	10-(100)-150	time of RRC(S) signal in milliseconds
P BURS(S)	abcd	bring up receiver (send). Uses switch-hook flash timer for timer.
	N	if BURS(S) not required
P BURS(R)	abcd	bring up receiver (receive). Uses switch-hook flash timer for timer.
	N	if BURS(R) not required
TIME	64-(128)-192	length of BURS(R) pulse in milliseconds
P CAS(S)	abcd	CAS Flash. Same timing as analogue trunks. Note: Prompted for JDMI only. Operational only if CASM package equipped.
	N	if CAS(S) not required
CLRB(S)	abcd	clearback (send) signal.
	N	if CLRB(S) not required (IDLE signal is used).
P RCTL(S)	abcd	Release control (send) signal.
	N	if RCTL(S) not required Note: Prompted only when CLRB is unused or is defined the same as IDLE.
TIME	100-(150)-300	time value is stored in multiples of 10 milliseconds.
R RCOD(S)	abcd	Release Control Originating party Disconnect. This signal is another pulsed SL-1 signal sent on incoming trunks when the originating party disconnects first.
	N	if RCOD(S) not required
TIME	150	timer value in milliseconds is fixed
P OPRS(R)	abcd	operator (receive) manual recall signal

Prompt	Response	Description
	N	if OPRS(R) not required
TIME	xxx yyy	minimum and maximum time range for OPRS(R) in milliseconds. xxx = 8-(48)-2040 yyy = xxx-(128)-2040
P NXFR(S)	abcd	network transfer signal (send) pulse. Pulse time not variable.
	N	if NXFR(S) not required
P ESNW(S)	abcd	ESN wink signal (send) pulse. Pulse time not variable.
	N	if ESNW(S) not required
P CAS(S)	abcd	Centralized Attendant signal (send) pulse. Pulse time not variable. Note: Prompted for 2 Mb DTI only.
	N	if CAS(S) not required
CLRF(R)	abcd	Clear forward (receive)
	N	if CLRF(R) not required
SOSI	abcd (N)	special operator signal defined undefined. Prompted when OPRC = N

Prompt	Response	Description
Outgoing Calls:		
SEZ(S)	abcd	seize voice or data from or to a non SL-1 switch
SEZD(S)	abcd	seize data (send) signal. Only for SL-1 to SL-1 applications
	N	if SEZD(S) not required
SEZV(S)	abcd	seize voice (send) signal. Only recommended for SL-1 to SL-1 applications
TIME	xxx yyy	minimum and maximum time range for OPRS(R) in milliseconds. xxx = 8-(48)-2040 yyy = xxx-(128)-2040
	N	if SEZV(S) not required
SEZA(R)	abcd	seize acknowledgment (receive) signal
	N	if SEZA(R) not required
P WNKS(R)	abcd	wink start pulsed seize acknowledgment (receive) signal
	N	if P WNKS(R) not required
TIME	20-(140)-500 , 20-(290)-500	minimum and maximum length of WNKS(R) pulse in milliseconds
	P EOS(R)	end of selection (receive) signal
	N	if EOS(R) not required
TIME	(64)-320 64-(256)-320	length of EOS(R) pulse stored in multiples of 8 milliseconds.
CONN(S)	abcd	connect send
CONN(R)	abcd	connect receive
P OPRC(R)	abcd	operator recall signal for special services. minimum three pulses of 160 milliseconds each.
	N	if OPRC(R) not required
P BURS(S)	abcd	bring up receiver (send) for L1 networking

Prompt	Response	Description
	N	if BURS(S) not required
P BURS(R)	abcd	bring up receiver (receive) for L1 networking
	N	if BURS(R) not required
TIME	64-(128)-192	length of BURS(R) pulse in milliseconds.
	N	if SEZV(S) not required
P CAS(R)	abcd	CAS Flash. Same timing as analogue trunks. Prompted for JDMI only. Operational only if CASR package equipped.
	N	if CAS(R) not required
CLRB(R)	abcd	clear back
	N	if CLRB(R) not required, when IDLE would be used
P RCTL(R)	abcd	Release control. Prompted only when CLRB is unused or is defined the same as IDLE.
	N	if RCTL(R) not required
TIME	96-(128)-320 96-(256)-320	time value stored in multiples of 8 milliseconds
P NXFR(R)	abcd	network transfer
	N	if not required
P ESNW(R)	abcd	ESN wink signal
	N	if ESNW(R) not required
P CAS(R)	abcd	centralized attendant service signal. 2 Mb DTI only
	N	if CAS(R) not required
CLRF(S)	abcd	Clear forward (send)
	N	if CLRF(S) not required

Prompt	Response	Description
TIME	(0)	800 milliseconds Note: Prompted when the abcd bits entered in response to the CLFR(S) prompt are different from the abcd bits of the IDLE signal
SOSO	abcd,(N)	special operator signal defined (undefined). Note: Prompted when OPRC = N

Step 3: Define the 2 Mb DTI pad tables, using overlay 73. The following prompts in Overlay 73 require a response.

Prompt	Response	Description
REQ	NEW,CHG, OUT, PRT, END	
TYPE	DTI2	2 Mb DTI
FEAT	PAD	Request the digital pad feature.
PDCA	1-16	PAD Category table. If one channel is using the specified table, then the command is aborted. Table 1 cannot be modified or deleted.

Prompt	Response	Description
TNLS	YES (NO)	TN List. This is for the print command only. A YES response means that a list of the trunk TNs using the requested PAD category tables will be printed after the table.
DFLT	(1)-16	For NEW only. The table is used for default values.
The following prompts define the pad levels. The receiving pad code is <i>r</i> and the transmission pad code is <i>t</i>. These entries have the range 0-15. The pad values (in decibels) relating to these codes are shown after this table.		
ONP	<i>r t</i>	On-premises extension
OPX	<i>r t</i>	Off-premises extension
DTT	<i>r t</i>	Digital TIE trunks
SDTT	<i>r t</i>	digital Satellite TIE trunks
NTC	<i>r t</i>	Nontransmission compensated
TRC	<i>r t</i>	Transmission compensated
DCO	<i>r t</i>	digital COT, FEX, WAT, and DID trunks
VNL	<i>r t</i>	VIA NET LOSS
DTO	<i>r t</i>	2 Mb DTI digital TOLL office trunks
ACO	<i>r t</i>	Analogue CO or WATS trunks
AFX	<i>r t</i>	Analogue FEX trunks
ADD	<i>r t</i>	Analogue DID trunks
SATT	<i>r t</i>	analogue satellite TIE trunks

Prompt	Response	Description
ATO	r t	analogue TOLL office trunks
DTI2	r t	2 Mb DTI trunk (prompted only if the 1.5/2 Mb Gateway feature is equipped and TYPE=DTI2) Author's Note—p7/dave/gateway
XUT	r t	analogue CO trunk (prompted only if the 1.5/2 Mb Gateway feature is equipped and TYPE=DTI2) Author's Note—p7/dave/gateway
XEM	r t	analogue TIE trunk (prompted only if the 1.5/2 Mb Gateway feature is equipped and TYPE=DTI2) Author's Note—p7/dave/gateway

The following pads are available for the 2 Mb DTI card. Their respective codes are also given. Positive dB represents loss and negative dB represents gain.

code	0	1	2	3	4	5	6	7
value (dB)	0.0	+1.0	+2.0	+3.0	+4.0	+5.0	+6.0	+7.0
code	8	9	10	11	12	13	14	15
value (dB)	+8.0	+9.0	+10.0	+11.0	+12.0	+13.0	+14.0	-1
code	16	17	18	19	20	21	22	23
value (dB)	-2	-3	-4	-5	-6	-7	-8	-9
code	24	25	26					
value (dB)	-10	idle	+0.6					

Step 4: Define the 2 Mb DTI loop timers, using overlay 73. The following prompts in Overlay 73 require a response.

Prompt	Response	Description
REQ	NEW,CHG, OUT, PRT, END	
TYPE	DTI2	2 Mb DTI
FEAT	LPTI	Set the timers used for a 2 Mb DTI
LOOP	I	2 Mb DTI loop number.
P DIGT(S)	abcd	(send) digit pulse timing from TDS. Bits P, X or U.
	N	if DIGT(S) signal not required
P METR(R)	abcd	metering (receive). Bits P, X or U. Only two P bits allowed. COT and DID trunks only. Note: PPM package must be enabled.
	N	if METR(R) signal not required
EDGE	0 1	PPM bit counted when changed from 1 to 0. PPM bit counted when changed from 0 to 1.
TIME	40-(240)-480	maximum time METR signal can be on in milliseconds.
SASU	0-(1920)-8064	seize acknowledge supervision period in milliseconds. 2 Mb DTI default = 1920

Prompt	Response	Description
MFAO	YES, NO <CR>	DTI card will or will not set bit 3 of timeslot 0 if loss of Multiframe Alignment Signal (MFAS) occurs. 2 Mb DTI default = NO no change is required
SZNI	YES, (NO)	PSTN incoming seizure during lockout of MFAS and far-end fault states allowed (denied)
MFF	CRC (AFF)	Multiframe format - Cyclic Redundancy Check (CRC 4) Alternative Frame Format Note: Prompted for Option 11 only. For non Option 11 only AFF is supported.

The following prompts and associated responses define the grade of service timers for the DTI card. Group I problems are treated individually. They are bipolar violations, bit error rate (frame alignment) slips and CRC-4 errors.

Group II problems are treated as a group. They are bit 3 of TS0 (far end out of service), bit 6 of TS16 (far end lost multiframe alignment), AIS (alarm indication signal), loss of frame alignment and loss of multiframe alignment.

Responses are interpreted as follows:

- mt = maintenance threshold time.
- ct = new call suppression (hardware service removal) threshold time.
- ot = out of service threshold time.
- dt = no new data calls suppression threshold time.

Each of the response times are expressed as follows:

- nnnY = time in milliseconds where nnnn = 20-5000 (input to nearest 20 milliseconds.)
- nnnS = time in seconds where nnn = 1-240

- nnnM = time in minutes where nnn = 1-240
- nnH = time in hours where nn = 1-24

Prompt	Response	Description
CRC	NC mt dt ct ot	Cyclic redundancy check error counts NC=Error count values are in the range 1-255 mt = Maintenance threshold time (MNT) dt = No new data calls threshold time (NNDC) ct = No new calls threshold time (NNC) ot = Out of service threshold time (OOS). Note: The following requirements must be met for input mt = Maintenance threshold must be equal to or greater than dt. dt = Data calls threshold must be equal to or greater than ct. ct = Calls threshold must be equal to or greater than ot.
BPV	NB mt dt ct ot	Bipolar violation error counts NB = Error count values are in the range 1-255 mt = Maintenance threshold time (MNT) dt = No new data calls threshold time (NNDC) ct = No new calls threshold time (NNC) ot = Out of service threshold time (OOS). Note: The following requirements must be met for this input: mt = >dt = >ct = >ot.
FAP	NF mt dt ct ot	frame alignment problem thresholds NF = Error count values are in the range 1-255 mt = Maintenance threshold time (MNT) dt = No new data calls threshold time (NNDC) ct = No new calls threshold time (NNC) ot = Out of service threshold time Note: The following requirements must be met for this input: mt = >dt = >ct = >ot.

Prompt	Response	Description
SLP	NS mt dt ct ot	Maintenance threshold slip count NS = Error count values are in the range 1-255 mt = Maintenance threshold time (MNT) dt = No new data calls threshold time (NNDC) ct = No new calls threshold time (NNC) ot = Out of service threshold time (OOS).
GP2	T2 mt dt ct ot	Group 2 error thresholds. This is the maximum amount of time that can occur before software checks the associated thresholds of 120 to 32,640 msec and rounds it to the closest multiple of 128 msec. T2 error count values are in the range 1-(20)-255. NC = Error count values are in the range 1-255 mt = Maintenance threshold time (MNT) dt = No new data calls threshold time (NNDC) ct = No new calls threshold time (NNC) ot = Out of service threshold time (OOS). Note: The following requirements must be met for this input: mt = >dt = >ct = >ot.
FRFW	YES,(NO)	This 2 Mb DTI loop is (is not) equipped with special Firmware for France.

Step 5: Define the 2 Mb DTI system timers, using overlay 73. The following prompts in Overlay 73 require a response.

Prompt	Response	Description
REQ	NEW,CHG, OUT, PRT, END	
TYPE	DTI2	2 Mb DTI
FEAT	SYTI	Change the switch timers and counters for 2 Mb DTI. There is only one such block per switch.
MAND	0-(15)-1440	maintenance guard time in minutes. Note: For 2 Mb DTI only.
NCSD	0-(15)-1440 1S-59S	new call suppression guard time in minutes and seconds. Note: For 2 Mb DTI only.
OSGD	0-(15)-1440	out of service guard time in minutes. Note: For 2 Mb DTI only.
OOSC	0-(5)-127	out of service occurrences since midnight (DTI disabled). Note: For 2 Mb DTI only.
PERS	0-(100)-254	persistence timer in milliseconds for far end problems. Note: For 2 Mb DTI only.
DBNC	(10)-32	debounce timer in milliseconds Note: For 2 Mb DTI Option 11 only
CLKN	1-9	Clock controller slot number
	<CR>	no change for defaults
	X	to remove clock controller data
PREF CK0	c	primary reference for clock controller zero. The response is the loop from which the clock controller will be deriving its primary clock pulses. NOT PROMPTED FOR Option 11
	<CR>	primary reference remains at current setting.

Prompt	Response	Description
	X	primary reference reverts to the free-run mode.
SREF CK0	c	secondary reference for clock controller zero. The response is the loop from which the clock controller will be deriving its secondary clock pulses.
	<CR>	secondary reference remains at current setting.
	X	secondary reference reverts to the free-run mode.
	C	Card number for the secondary reference clock for clock controller 0 for Option 11 only.
	<CR>	primary reference remains at current setting.
	X	primary reference reverts to the free-run mode.
SREF CK1	c	secondary reference for clock controller one. The response is the loop from which the clock controller will be deriving its secondary clock pulses. NOT PROMPTED FOR Option 11
	<CR>	secondary reference remains at current setting.
	X	secondary reference reverts to the free-run mode.
		Note: The clock controller prompts will only appear for clocks which are valid for the machine type being configured. The prompts will only appear if the system is in a valid state for the definition of the 2 Mb DTI clock controller data (i.e., the DTI clock references must be unused or in a free-run mode).
CCGD	0-(15)-1440	clock controller free run guard time in minutes.

Prompt	Response	Description
CCAR	0-(15)	clock controller audit rate. The time, in minutes, between normal CC audits. Only programmable on units equipped with 2-Mb DTI. Note: Before programming clock controller references, the QPC775 clock controller card(s) must be plugged in, and the switches on the system's QPC441 3 Port Extender appropriately set. Unless this is done, the PREF and SREF prompts are not given.

Step 6: Define service routes. Use Overlay 16 to configure service routes or set the NRAG duration timer.

Prompt	Response	Description
REQ	NEW,CHG, OUT	
TYPE	RDB	route data block
CUST	0-31 0-511	customer number all system options other than 71, XN or NT system options 71, XN and NT
ROUT	0-30, 32-127	route number
TKTP	TIE COT DID	service routes allowed with ISDN — Tie trunk route (the only service trunk type permitted for SL-1 to SL-1 connections) — Central office trunk — Direct Inward Dial trunk
RCLS	(EXT) INT	class marked route as Internal or External
DTRK	YES (NO)	digital trunk route — digital — analogue
DGTP	DTI2	Select a digital trunk type of 2 Mb DTI
•		
•		
NCNA	YES, (NO)	Network Call Name is (is not) allowed
NCRD	YES, (NO)	Network Call Redirection. Allows network call redirection messages to be sent (or blocks messages if NCRD =no).

Prompt	Response	Description
PTYP	(ATT) AST AOT (DTT) DCT DST	Port type at far end: Analogue tie trunk routes: — analogue Tie trunk — analogue satellite SL-1 Tie trunk or ESN satellite Meridian SL-1 Tie trunk — analogue Tie trunk, used instead of ATT whenever the Meridian SL-1 has one or more digital satellite trunk routes (DST) to any digital satellite SL-1 which includes OPX sets Digital tie trunk routes: — digital Tie trunk — combination digital Tie trunk — digital satellite SL-1 Tie trunk
AUTO	YES,(NO)	auto-terminate must be NO if response to DSEL is VOD
ICOG	IAO ICT OGT	incoming and outgoing trunk incoming trunk outgoing trunk
SRCH	(LIN) RRB	linear search, or round-robin search—use for outgoing trunks
ACOD	xxxx	trunk route access code
TARG	1-15	trunk access restriction group for routes
OABS	0-9	outgoing digit(s) to be absorbed
INST	(0)-999	digits to be inserted
CNTL	(NO),YES	changes to controls or timers
NEDC		near end disconnect control
	ETH	either end control
	ORG	originating end control. Default for TIE, ATVN, DID, and CCSA trunk types
FEDC		far end disconnect control Note: Loop start trunks may be assigned either ETH or ORG.
	ETH	either end

Prompt	Response	Description
	FEC	far end
	JNT	joint
	(ORG)	originating end
DLTN	YES, (NO)	dial tone on originating calls
TIMER	(30)-240	Network Ring Again duration timer—time is in minutes Note: Package 148, Advanced ISDN Features, is required.

Step 7: Define the associated list of service trunks. Use Overlay 14.

Prompt	Response	Description
REQ	NEW,CHG, MOV,OUT	
TYPE	TIE COT DID	— Tie trunk route (only type allowed with SL-1 to SL-1 ISDN ISL or 2 Mb DTI) — Central Office trunk data block — Direct Inward Dial trunk data block
CHID		
TN	l c	terminal number address—use loop number created in Overlay 17 l = 0-9 —2 Mb DTI loop number c = 1-30 —2 Mb DTI channel
TOTN	l c	new loop and channel. Loop must be a 2 Mb DTI loop. l = 0-9 —2 Mb DTI loop number c = 1-30 —2 Mb DTI channel
PDCA	(1) - 16	Pad Category Number. This prompt is given if the loop is a 2 Mb DTI loop, or if the loop is a 2 Mb DTI loop and the 1.5/2 Mb Gateway feature is equipped. For 2 to 16, the pad category used must already be defined in Overlay 73.

PCML	MU, A	System PCM law. Prompted if the loop is 2 Mb DTI, or if the loop is a 2 Mb DTI loop and the 2 Mb Gateway feature is equipped. The default was entered in the PCML prompt in Overlay 17.
CUST	0-31	customer number
NCOS	0-3 0-7 0-15	network class-of-service group number: CDP BARS/NFCR NARS
RTMB	0-127 1-254	route and member numbers
NITE	xxxx	night-service directory number
TGAR	(0)-15	trunk group access restriction
CLS		class-of-service restrictions Note: If DIP option is selected, a new SLCA table must be defined in LD 73.

Chapter 6 — 2 Mb PRI Implementation

Overview

This chapter provides the information required to install 2 Mb PRI on a Meridian 1 Option 11 system:

- hardware installation
- software programming instructions

ISDN feature implementation is found in the chapter called "ISDN Feature Implementation".

Note: This chapter does not provide installation instructions for DPNSS1 or DASS2. This information is found in the chapter called "DPNSS1/DASS2 Implementation".

This chapter builds on the ESN programming that is assumed to already be in place. Installation technicians should possess a basic understanding of NARS and CDP. (See Appendix B for additional information on ESN). This chapter covers the most common type of Northern Telecom PRI installations.

- 30B+D PRI installation between Meridian 1 PBXs.
- 30B+D PRI installation between Meridian 1 PBX and a public exchange/central office.

Hardware requirements

Circuit cards

To implement 2.0 Mb PRI on the Meridian 1 Option 11, an NTAK79 or an NTBK50 PRI card plus associated daughterboards is required.

Table 62
2.0 Mb PRI hardware requirements

Circuit card	Description
NTBK50	2.0 Mb PRI circuit card Supports the NTAK20 clock controller daughterboard and a D-channel handler interface daughterboard (NTAK93 DCHI or NTBK51 DDCH).
NTAK79	2.0 Mb PRI circuit card Supports an on-board clock controller and an on-board D-channel handler interface.
NTAK20	Clock-controller daughterboard. Option 11 supports only one active clock controller per system.
NTAK93	Connects to the NTBK50 PRI card D-Channel-handler interface (DCHI) daughterboard Connects to the NTBK50 PRI card.
NTBK51	Downloadable D-Channel daughterboard (DDCH). Connects to the NTBK50 PRI card.

Cables

The following cables are required for PRI connections:

- CEPT Cable - NTBK05DA (120 $\frac{3}{4}$ twisted pair—6.15 m length)
- or
- CEPT Cable - NTBK05CA (75 $\frac{3}{4}$ coaxial—6.15 m length)

Hardware description

2.0 Mb PRI cards

Two PRI cards are available on the Option 11 system:

- NTAK79 2.0 Mb PRI card
- NTBK50 2.0 Mb PRI card

The difference between the two PRI cards is the ability of the NTBK50, when equipped with the NTBK51 D-channel daughterboard, to download software onto the card. This feature is based on the MSDL platform and essentially replaces the D-channel circuit on the NTAK79 PRI card. (The NTAK79 PRI card does not support the NTBK51 Downloadable D-channel handler daughterboard.)

A second difference between the NTAK79 and NTBK50 2.0 Mb PRI cards is that the NTAK79 has an on-board clock controller while the NTBK50 supports the NTAK20 clock controller daughterboard.

Note: If the NTAK93 D-channel daughterboard is attached to the NTBK50 PRI card instead of the NTBK51 Downloadable D-channel daughterboard, the NTBK50 PRI card functions in the same manner as the NTAK79 PRI card.

NTAK79 2 Mb PRI circuit card

The 2 Mb Primary Rate Interface card provides the physical interface for the digital E-1 carrier on the Option 11 system. The card includes an on-board clock controller and on-board D-channel handler. It is installed in slots 1 through 9 in the main cabinet.

For information on the NTAK79 faceplate LEDs, refer to the chapter in this guide called "PRI maintenance". For more information on the operation of the on-board clock controller, refer to the chapter in this guide called "Network clocking"

NTBK50 2 Mb PRI circuit card

The 2 Mb Primary Rate Interface card provides the physical interface for the digital E-1 carrier on the Option 11 system. The card is installed in slots 1 through 9 in the main cabinet and supports clock controller and D-channel handler daughterboards:

- NTAK20 clock controller daughterboard
- NTAK93 D-channel handler daughterboard, or NTBK51 Downloadable D-channel daughterboard.

If the NTAK93 D-channel daughterboard is attached, the NTBK50 PRI card functions in the same manner as the NTAK79 PRI card. If the NTBK51 D-channel daughterboard is attached, software is downloaded to the card instead of residing in a D-channel circuit.

For information on the NTBK50 faceplate LEDs, refer to the chapter in this guide called "PRI maintenance". For more information on the operation of the clock controller, refer to the chapter in this guide called "Network clocking".

NTAK20 Clock Controller (CC) daughterboard

The NTAK20 Clock Controller daughterboard is used with the NTBK50 2.0 Mb PRI card. (The NTAK79 PRI card has an *on-board* clock controller).

Digital Trunking requires synchronized clocking so that a shift in one clock source will result in an equivalent shift of the same size and direction in all parts of the network. On Option 11 systems, synchronization is accomplished with the NTAK20 clock controller circuit card.

The Clock Controller circuitry synchronizes the Option 11 to an external reference clock, and generates and distributes the clock to the system. Option 11 can function either as a slave to an external clock or as a clocking master. The NTAK20AB version of the clock controller meets AT&T Stratum 3 and Bell Canada Node Category D specifications. The NTAK20BB version meets CCITT stratum 4 specifications. See the chapter called *Network clocking* for details.

Shelf slot assignment

On non-ECM system cabinets, the NTAK20 may be placed in slots 1-9. On cabinets NTAK11Dx and NTAK11Fx, the active NTAK20 must be placed in slots 1-3 (slots 4-10 may not be used.)

NTAK93 D-Channel Handler Interface (DCHI) daughterboard

The NTAK93 DCHI daughterboard interfaces with the Meridian 1 Option 11 Central Processing Unit (CPU) and mounts on the NTBK50 PRI card for PRI (but not ISL) applications. The equivalent circuit is contained on-board the NTAK79 2.0 Mb PRI card.

The DCHI is responsible for performing the Q.921 layer 2 protocol information. It transfers layer 3 signaling information between two adjacent network switches.

The NTAK93 DCH daughterboard, when installed on the NTBK50 circuit card, is addressed in the same slot as the NTBK50. The NTAK93 can use SDI I/O addresses 1 to 15 and port 1.

A minimum of one NTAK93 is required for each PRI link. If more than one PRI link is connected to the same end location, a single DCHI circuit card can support up to a maximum of 8 PRI connections for the Option 11 system. This allows for the support of 240 B-Channels or PRI trunks.

NTBK51 Downloadable D-Channel daughterboard

The NTBK51 DDCH daughterboard interfaces with the Meridian 1 Option 11 Central Processing Unit (CPU) and mounts on the NTBK50 2.0 Mb PRI circuit card for PRI D-Channel applications. The DDCH is equivalent to the MSDL card used on the larger Meridian 1 systems, but it only supports D-channel applications (no SDI or ESDI).

The NTB51 DDCH daughterboard, when installed on the NTB50 circuit card, is addressed in the same slot as the NTB50 .

A minimum of one NTB51 is required for each PRI link. If more than one PRI link is connected to the same end location, a single DDCH circuit card can support up to a maximum of 8 PRI connections for the Option 11 system. This allows for the support of 240 B-Channels or PRI trunks.

Installing PRI hardware: NTAK79 PRI card

The steps required to install PRI are as follows:

- 1 Inspect the PRI card
- 2 Set the switches on the PRI card
- 3 Insert the PRI card in the main cabinet (slots 1-9 only)
- 4 Connect the cables

Each of these steps is described in the pages that follow. The PRI hardware installation procedure is the same regardless of the type of system at the far end (i.e. another Meridian 1, AXE-10, SYS-12, etc.).

Inspecting the NTAK79 circuit card

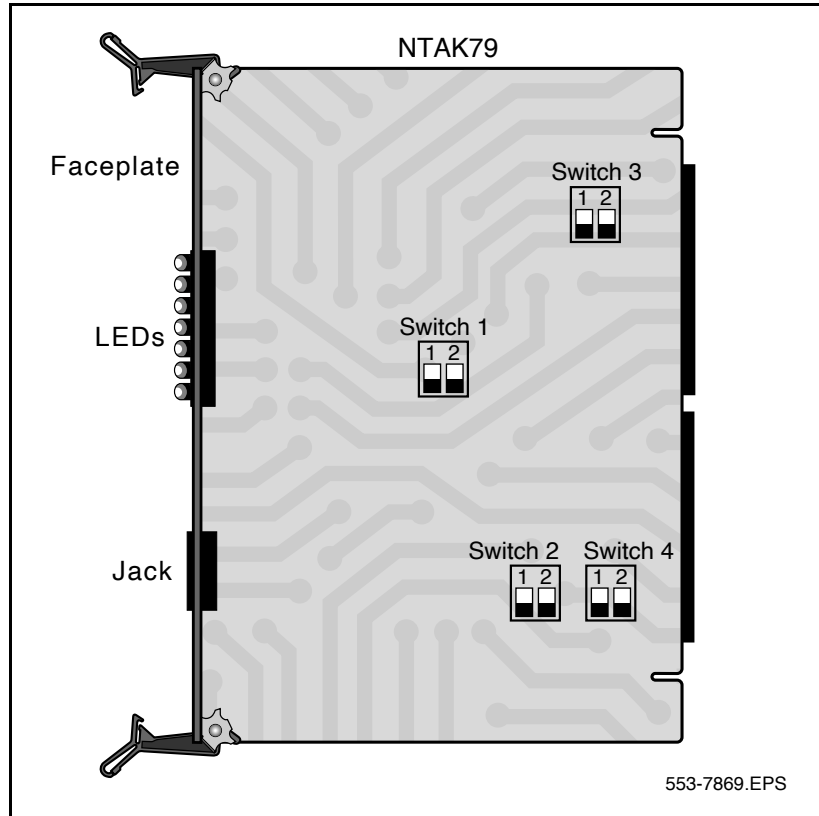
- Locate the NTAK79 2 Mb circuit card and carefully remove it from its packaging.
- Inspect the circuit card for any visible damage which may have occurred during shipping.

Setting the switches on the NTAK79

The NTAK79 incorporates four on-board dip switches. The tables that follow provide information on the various settings and related functions of these switches.

Note: The ON position for all the switches is towards the bottom of the card. This is indicated by a white dot printed on the board adjacent to the bottom left corner of each individual switch.

NTAK79 with Switch locations



Set the switches found on the circuit card as per the requirements of your specific installation:

Switch SW1—DCHI Configuration

This switch enables/disables the on-board DCHI and sets the operating mode of the DCHI.

For the U.K., use DPNSS1 mode. For all other countries, use Q.931 mode.

Switch	Down (On)	Up (Off)
SW 1-1	enable DCHI	disable DCHI
SW 1-2	DPNSS1/DASS2	Q.931

Switch SW2—Carrier Impedance Configuration

This switch sets the carrier impedance to either 120 Ω or 75 Ω . Twisted pair cable is usually associated with 120 Ω . Coaxial cable is usually associated with the 75 Ω setting.

Cable Type	SW 2-1	SW 2-2
75 Ω	Up (Off)	Down (On)
120 Ω	Down (On)	Up (Off)

Switch SW3—Clock Controller Configuration

This switch enables/disables (H/W) the on-board Clock Controller. SW 3-2 should be disabled if the on-board clock controller is not in use.

Switch	Down (On)	Up (Off)	Note
SW 3-1	—	—	Spare
SW 3-2	Disabled	Enabled	

Switch SW4—Carrier Shield Grounding

This switch allows for the selective grounding of the Tx and/or Rx pairs of the carrier cable. Closing the switch (down position) applies Frame Ground (FGND) to the coaxial carrier cable shield, creating a 75 $\frac{3}{4}$ unbalanced configuration. This applies only to the NTB05CA cable.

Switch	Down (On)	Up (Off)
SW 4-1	Rx—FGND	Rx—OPEN
SW 4-2	Tx—FGND	Tx—OPEN

Note: The usual method is to ground the outer conductor of the receive coax signal.

Inserting the NTAK79 into the main cabinet

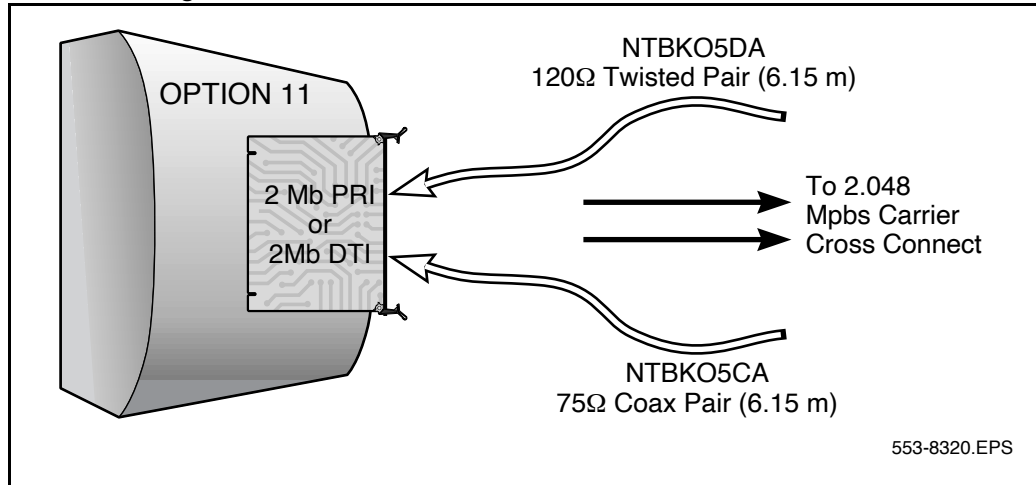
Slide the circuit card into any unused card slot 1 through 9 in the main cabinet. Secure the circuit card in the cabinet by locking the lock latch assemblies. The card number associated with a 2 Mb PRI card is based on the slot in which the card is installed.

Connecting the cables

Follow the instructions below to connect cables to the NTAK79 PRI card.

- In the cabling area, located directly below the card cage, remove the retaining bar used to secure the MDF cables. Connect the NTB05DA/CA interface cable to the 50 pin Amphenol connector below the slot in which the NTAK79 is installed. Re-install the retaining bar to secure the cable(s) in place.
- Terminate the NTB05DA/CA carrier cable as required.

Figure 121
NTAK79 Cabling



NTBK05DA pinouts

The pinouts for the NTBK05DA cable are as follows:

From: 50 pin MDF connector	To: 9 pin connector	Colour	Signal
pin 23	pin 6	Black	R0
pin 48	pin 7	White	T0
pin 50	pin 9	Bare	R0/T0 FGND
pin 24	pin 2	Black	R1
pin 49	pin 3	Red	T1
pin 25	pin 5	Bare	R1/T1 FGND

NTBK05CA pinouts

The pinouts for the NTBK05CA cable are as follows:

From: 50 pin MDF connector	To: Transmit coax connector	To: Receive coax connector	To: 50 pin MDF connector
pin 23	Inner conductor	—	—
pin 48	outer conductor	—	—
pin 24	—	Inner conductor	—
pin 49	—	outer conductor	—

Installing PRI hardware: NTBK50 PRI card

The NTBK50 serves as a motherboard to the NTAK20 clock controller, and either the NTBK51 Downloadable D-Channel handler or the NTAK93 D-channel handler.

The steps required to install PRI are as follows:

- 1 Inspect the PRI card and daughterboards
- 2 Set the switches on the PRI card
- 3 Mount the daughterboard(s) on the PRI card
- 4 Insert the PRI card in the main cabinet
- 5 Connect the cables

Each of these steps is described in the pages that follow. The PRI hardware installation procedure is the same regardless of the type of system at the far end (i.e. another Meridian 1, AXE-10, SYS-12, etc.).

Inspecting the NTBK50 circuit card

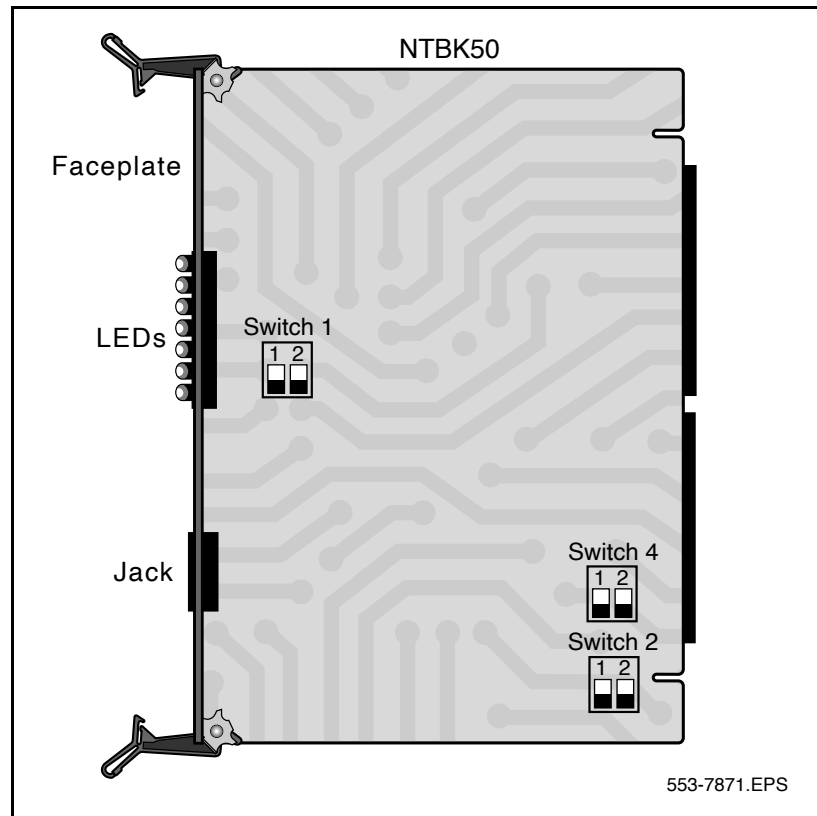
- Locate the NTBK50 2 Mb circuit card plus associated daughterboard(s) and carefully remove them from their packaging.
- Inspect the circuit cards for any visible damage which may have occurred during shipping.

Setting the switches on the NTBK50

The NTBK50 incorporates three on-board dip switches. The following tables provided information on the various settings and related functions of these switches.

Note: The ON position for all the switches is towards the bottom of the card. This is indicated by a white dot printed on the board adjacent to the bottom left corner of each individual switch.

NTBK50 with switch locations



Set the switches found on the circuit card as per the requirements of your specific installation:

Switch SW1—DCHI Configuration (NTAK93 only)

This switch enables/disables the DCHI and sets the operating mode of the DCHI. It is only used if an NTAK93 D-channel handler daughterboard is being used. It has no effect when using the NTBK51 DDCH daughterboard.

For the U.K., use DPNSS1 mode. For all other countries, use Q.931 mode.

Switch	Down (On)	Up (Off)
SW 1-1	—	—
SW 1-2	DPNSS1/DASS2	Q.931

Switch SW2—Carrier Impedance Configuration

This switch sets the carrier impedance to either 120 $\frac{3}{4}$ or 75 $\frac{3}{4}$. Twisted pair cable is usually associated with 120 $\frac{3}{4}$. Coaxial cable is usually associated with the 75 $\frac{3}{4}$ setting.

Cable Type	SW 2-1
75 $\frac{3}{4}$	Down (On)
120 $\frac{3}{4}$	Up (Off)

Switch SW4—Carrier Shield Grounding

This switch allows for the selective grounding of the Tx and/or Rx pairs of the carrier cable. Closing the switch (down position) applies Frame Ground (FGND) to the coaxial carrier cable shield, creating a 75 $\frac{3}{4}$ unbalanced configuration. This applies only to the NTBK05CA cable.

Switch	Down (On)	Up (Off)
SW 4-1	Rx—FGND	Rx—OPEN
SW 4-2	Tx—FGND	Tx—OPEN

Note: The usual method is to ground the outer conductor of the receive coax signal.

Mounting the daughterboards on the NTBK50

Use the following procedure to mount and remove the NTAK20 CC and the NTBK51 DDCH or NTAK93 DCHI daughterboards onto the NTBK50 PRI.

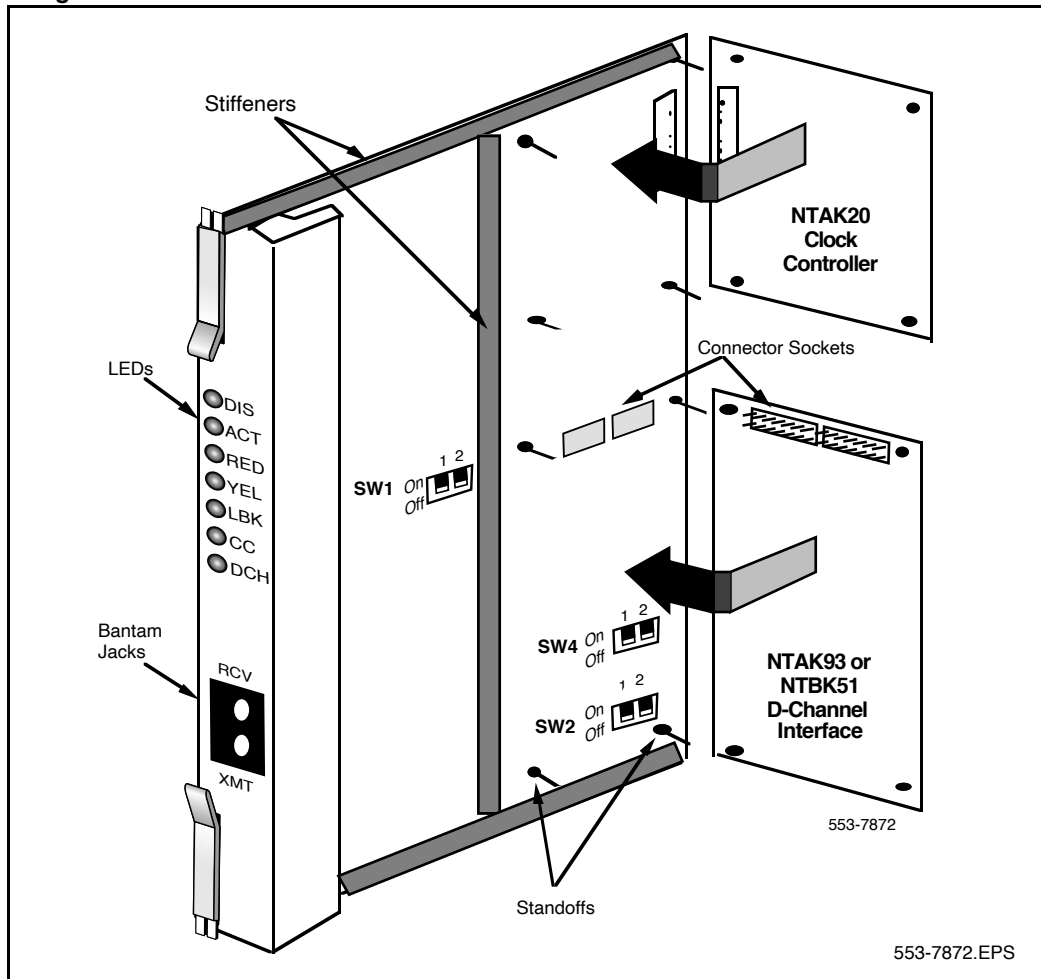
Install the NTAK93 or NTBK51 daughterboard before the NTAK20 daughterboard. Work on a flat surface when mounting or removing daughterboards.

- 1** Visually inspect the connector pins on the underside of the daughterboard. Any pins that are bent should be re-aligned prior to mounting.
- 2** Place the NTBK50 down flat on an anti-static pad.
- 3** From an overhead viewpoint, with the daughterboard parallel above the NTBK50 and the connector pins aligned over the connector sockets, line up the mounting holes on the daughterboard (see the figure below) with the tops of the standoffs on the NTBK50.
- 4** Slowly lower the daughterboard towards the NTBK50, keeping the standoffs in line with all four holes, until the holes are resting on the tops of the four standoffs.

If more than a very slight amount of pressure is required at this point, the connector pins may not be aligned with the connector socket. If so, lift the daughterboard off the NTBK50 and return to step 2.

- 5** Gently apply pressure along the edge of the board where the connector is located until the standoffs at the two corners adjacent to the connector snap into a locked position. Then press down on the two corners on the opposite side until they also are locked into place.

Figure 122
Daughterboard installation



Removing the daughterboards from the NTBK50

Use these guidelines to remove the NTAK20 and NTBK51 or NTAK93 from the NTBK50 PRI card. Because of the physical layout of the mother and daughterboards, the NTAK20 should be removed before the NTAK93 or NTBK51.

- 1** Starting at the two corners opposite the connector, gently lift each corner out of the locking groove of the standoff.
- 2** At the two corners adjacent to the connector, gently lift the entire side until the mounting holes are clear of the locking groove of the standoff.
- 3** To remove the connector pins, grasp the edge of the board adjacent to the connector and lift gently.

If more than one NTBK50 card is installed, the additional cards may not carry daughterboards, depending on the system configuration. At least one NTAK20 (per system) is always required, however.

Inserting the NTBK50 into the main cabinet

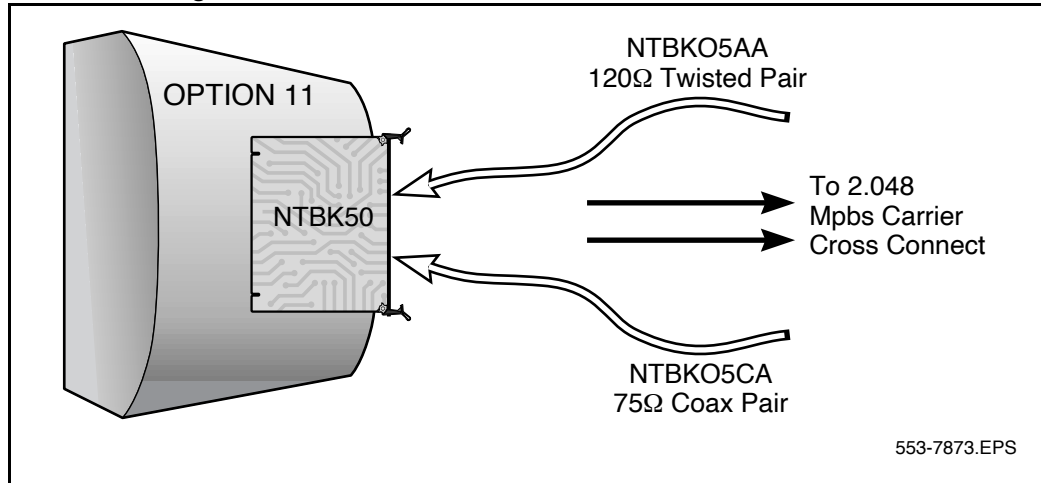
Slide the circuit card into any unused card slot 1 through 9 in the main cabinet. Secure the circuit card in the cabinet by locking the lock latch assemblies. The card number associated with a 2 Mb PRI card is based on the slot in which the card is installed.

Connecting the cables

Follow the instructions below to connect cables to the NTBK50 PRI card.

- In the cabling area, located directly below the card cage, remove the retaining bar used to secure the MDF cables. Connect the NTBK05DA/CA interface cable to the 50 pin Amphenol connector below the slot in which the NTBK50 is installed. Re-install the retaining bar to secure the cable(s) in place.
- Terminate the NTBK05DA/CA carrier cable as required.

Figure 123
2 Mb PRI Cabling



NTBK05DA pinouts

The pinouts for the NTBKO5DA cable are as follows:

From: 50 pin MDF connector	To: 9 pin connector	Colour	Signal
pin 23	pin 6	Black	R0
pin 48	pin 7	White	T0
pin 50	pin 9	Bare	R0/T0 FGND
pin 24	pin 2	Black	R1
pin 49	pin 3	Red	T1
pin 25	pin 5	Bare	R1/T1 FGND

NTBK05CA pinouts

The pinouts for the NTBK05CA cable are as follows:

From: 50 pin MDF connector	To: Transmit coax connector	To: Receive coax connector	To: 50 pin MDF connector
pin 23	Inner conductor	—	—
pin 48	outer conductor	—	—
pin 24	—	Inner conductor	—
pin 49	—	outer conductor	—

PRI software programming

Use the following procedure to configure PRI loops, the DCHI/DDCH interface, the DCH link and ISDN trunk route and trunks (B-channels). No feature applications other than Basic Call Service are programmed.

Prompts which do not show a response can be left at default. For more information on any of these prompts, refer to Option 11 *Software guides*.

Overlay 17 is used to create, change or remove a logical D-Channel which is associated with:

- a D-channel handler interface (DCHI):
 - NTAK79 PRI card
 - NTAK02 SDI/DCH card
 - NTBK50 PRI card with the NTAK93 daughterboard
- a downloadable D-channel handler (DDCH) on an NTBK50 PRI card with the NTBK51 daughterboard.

Limitations

PRI loops must be configured before defining DCH links or PRI applications. The following restriction apply when configuring a D-Channel:

- The card slot number used for the DCHI/DDCH must be between 1-9.
- The port number must be set to 1.
- A primary D-Channel can not act as the backup D-Channel of another primary D-Channel.
- A primary D-Channel with a backup D-Channel can not be taken out. The craftperson must first take out the backup D-Channel.
- The D-Channel must be disabled if any DCH parameters are changed.
- Card type of the backup D-channel must be the same as the card type of the primary D-channel. (Example: an NTAK02 DCH can not act as a backup to an NTAK79 DCH. The backup must be another NTAK02).
- The MOV DCH command is not supported.

Procedure summary

Step	Overlay	Action
1	LD 17 (CFN)	Add a PRI loop.
2	LD 17 (CFN)	Add a DDCH/DCHI card.
3	LD 15 (CDB)	Define a PRI customer.
4	LD 16 (RDB)	Define a PRI service route.
5	LD 14 (TDB)	Define service channels (B channels) and PRI trunks.
6	LD 73 (PRI2)	Define clock controller parameters.
7	LD 73 (PRI2)	Define PRI parameters and thresholds. Note: In most cases, the settings should be left at their default values. However, if changes to these values are necessary, use this step to make them.
8	LD 73 (PRI2)	Change trunk pad category values.

1 Add a PRI loop. Use Overlay 17.

Prompt	Response	Description
REQ	CHG	
TYPE	CFN	configuration data block
CEQU	YES	changes to common equipment
PRI2	XXX	where XXX is the PRI2 digital card number (1-9)

2 Add a DCHI or DDCH. Use Overlay 17.

Prompt	Response	Description
REQ	CHG	
TYPE	CFN	configuration data block
ADAN	NEW DCH 1-15 CHG DCH 1-15 OUT DCH 1-15	Add a primary D-Channel port number. Any unused SDI port number Change a primary D-Channel Out the primary D-Channel

Prompt	Response	Description
CTYP	DCHI/MSDL	DCHI for D-Channel configuration on the NTAK79, NTAK02, NTAK93 D-channel. MSDL for Downloadable D-Channel configuration (NTBK51 D-channel). The MSDL base and D-Channel application software are downloadable into the DDCH card.
CDNO	1-9	Card slot number to be used as the primary DDCH/DCHI
PORT	1	PORT must be set to "1"
DES	<CR>	Back-up DCHI port number, if required
DPNS	<CR>	Default is no
USR	PRI	D-channel is for ISDN PRI only. Note: 2.0 Mb only supports PRI or SHA user mode.
IFC	SL1	Interface type is Meridian 1 - Meridian 1
DCHL	(1-9)	PRI card number which will be carrying the D-channel. Must match entry made for the "CDNO" associated with the "DCHI" prompt above
PRI2	<CR>	Additional PRI Loops controlled by this DCHI. Remember one DCHI can control up to 16 PRI loops going to the same destination. For the Option 11 system, the maximum limit is eight loops.
OTBF	1-(16)-127	number of output request buffers. Note: for a single PRI link, leave this prompt at default (16). Add 5 output request buffers per additional link.
SIDE	NET (USR)	Prompted only if IFC is set to SLI. Default is set to slave. NET = network, the controlling switch USR= slave to the controller Note: In X11 release 17 and earlier, enter MAS for NET and SLAV for USR.

Prompt	Response	Description
RLS	XX	X11 software release of far-end. This is the current software release of the far end. If the far end has an incompatible release of software, it prevents the sending of application messages, i.e. for Network Ring Again.
RCAP	MSL	MSDL RCAP capability only applies to SL-1 interface and on release 18 or above.
OVL R	<CR>	Allow or disallow overlap receiving on a D-channel. Default is NO.
LAPD	YES,(NO)	change LAPD parameters. Enter carriage return if timers are to be left at default. The following timers are prompted only if LAPD is set to YES. (They can all be left at default during initial set-up.)
T23	1-(20)-31	interface guard timer checks how long the interface takes to respond. In units of 0.5 seconds (default 20 = 10 seconds).
T200	2-(3)-40	retransmission timer in units of 0.5 seconds (default 3 = 1.5 seconds).
N200	1-(3)-8	maximum number of retransmissions
N201	4(260)	maximum number of octets in information field
K	1-(7)-32	maximum number of outstanding unacknowledged frames (NAKS)

3 Define a PRI customer. Use Overlay 15.

Prompt	Response	Description
REQ	NEW,CHG	
TYPE	CDB NET_DATA	customer data block Release 21 gate opener
CUST	0-31	customer number
LDN	XXXX	enter the customer's Listed Directory Number
AC2		Access Code 2. Enter call types (type of number) that use access code 2. Multiple responses are permitted. This prompt only appears on NARS equipped systems. If a call type is not entered here, it is automatically defaulted to access code 1.

Prompt	Response	Description
	NPA	E.164 National
	NXX	E.164 Subscriber
	INTL	International
	SPN	Special Number
	LOC	Location Code
ISDN	YES	customer is equipped with ISDN
PNI	1-32700	customer private network identifier. This number must be unique to this customer in the private network. e.g. it is used as part of the setup message for feature operation such as Network Ring Again, Network ACD....Note that if set to zero (0), NRAG and NACD will not work.
HNPA (PFX1)	NPA	telephone area code for this Meridian 1. Sent as part of setup message as calling line identification.
HNXX (PFX2)	NXX	telephone local exchange code for this Meridian 1. Sent as part of setup message as calling line identification.
HLOC	XXX	home location code (NARS)
LSC	1-9999	one to four digit Local Steering Code established in the Coordinated Dialing Plan (CDP). The LSC prompt is required for Calling Line I.D. and Network ACD.

4 Define a PRI service route. Use Overlay 16.

Prompt	Response	Description
REQ	NEW, CHG	
TYPE	RDB	route data block
TKTP	TIE	TIE trunk only, allowed between MSL-1.
DTRK	YES	digital trunk route
DGPT	PRI2	
ISDN	YES	ISDN option
MODE	PRI	route used for PRI only

Prompt	Response	Description
PNI	1-32700	customer private network identifier-must be the same as the CDB PNI at far end.
IFC	SL1	interface type: Meridian 1 to Meridian 1
CHTY	BCH	signaling type- prompted if DTRK is YES. D-channel signaling for B-channels.
CTYP	<CR>	Call Type. Enter the call type to be associated with the outgoing route for direct dialing using the trunk access code (instead of NARS access code) See the "X11 Software Guide - Including Supplementary Features" for a listing of possible responses.
INAC	YES	Insert Access Code. Permits the NARS AC1 or AC2 access code to be re-inserted automatically on an incoming ESN call. This prompt only appears on a TIE route and must be set to "YES" in order for features such as Network ACD to function. On an existing ESN network, setting this prompt to "YES" may also require modifying the Digit Manipulation Index (DMI) associated with this route at the far end (so that the Access Code doesn't get re-inserted twice). The INSERT prompt (INST) is bypassed if INAC is set to YES.

5 Define service channels and PRI trunks. Use Overlay 14.

Prompt	Response	Description
REQ	NEW,CHG	Note: when assigning several members at once use the multiple create command NEW XX.
TYPE	TIE	TIE trunk only, allowed between MSL-1.
TN	c ch	enter the PRI trunk card (c) and channel number (ch). c = 1-9, ch = 1-30
RTMB	RR MM	route (created in step 5) (RR) and member number (MM).
INC		
TKID		

6 Define clock controller parameters. Use Overlay 73.

Free-run is used when this Meridian PBX serves as the clock source master for the private network. This is only prompted if the CC has been physically connected to the system. See the “*Network clocking*” chapter for more details on clocking.

Prompt	Response	Description
REQ	CHG	
TYPE	PRI2	2 Mb PRI
FEAT	SYTI	System timers
CLKN	1-9	Card slot number of the PRI circuit card which will have the active clock controller.
PREF	1-9	Primary reference source for clock controller. Enter the PRI card number of the PRI card which will have an active clock controller. This is the PRI which the clock controller will use as it's primary source to synchronize (to track) the system network clock. A carriage return <CR> here signifies the system will operate in free-run (non-tracking).
SREF	1-9	Secondary reference source for clock controller—prompted only if primary source is not free-run. Enter the card number of the PRI card that is to be used as the secondary clock reference. This is the PRI link that the clock controller will use as it's secondary source to synchronize (to track) the network clock. It is only used when unable to track on the primary source (i.e. too many slips).
CCAR	0-(15)	Clock Controller Audit Rate. Enter the time (in minutes) between normal CC audits.

7 Define PRI parameters and thresholds. Use Overlay 73.

Prompt	Response	Description
REQ	CHG	
TYPE	PRI2	2 Mb PRI
FEAT	LPTI	
LOOP	X	X is the slot number of the 2 Mb PRI card.
MFF	AFF, (CRC)	Alternate mode or CRC multi-frame mode
ALRM	(REG), ALT	Default or alternate alarms selected
G10S		
SLP	mc mt oc ot	Slip error count, <i>mc</i> —Maintenance threshold slip count, 1-(5)-255 <i>mt</i> —Maintenance threshold time, default 24 hours, see Note after Step 3 <i>oc</i> —Out-of-service threshold slip count, 1-(30)-255 <i>ot</i> —Out of service threshold time, default 1 hour, see Note after Step 3
BPV	n1 n2	Bipolar violation error count, 1-(128)-255 for n1, 1-(122)-255 for n2. n1 is multiplied by 16 to obtain the actual count, giving an actual count range of 16-4080.
CRC	n1 n2	Cyclic redundancy check error count 1-(201)-255 for n1, 1-(97)-255 for n2. n1 is multiplied by 4 to obtain the actual count, giving an actual count range of 4-1020.
FAP	n1 n2	Frame alignment problem error count 1-(28)-255 for n1, (1)-255 for n2..
RATS	1-(10)-15	Number of seconds firmware has to check BPV/CRC/FAP for excessive error rate.

Prompt	Response	Description
GP2	T2 mt dt ct ot	Group 2 error thresholds. This is the maximum amount of time that can occur before software checks the associated thresholds of 120 to 32,640 msec and rounds it to the closest multiple of 128 msec. T2 = Error count values are in the range 1-(20)-255 mt = Maintenance threshold time (MNT)(default =100S) dt = No new data calls threshold time (NNDC)(default =12S) ct = No new calls threshold time (NNC)(default =12S) ot = Out of service threshold time (OOS)(default =4S). Note : The following requirements must be met for input mt = >dt = >ct = >ot. Note : Threshold times must be one of
MNG1	nnnM	Maintenance Guard time Group 1 default = 15M
NCG1	nnnM	No New Calls Guard time Group 1 default = 15M
OSG1	nnnM	Out Of Service Guard time Group 1 default = 15M
MNG2	nnnS	Maintenance Guard time Group 2 default = 15S
NCG2	nnnS	No New Calls Guard time Group 2 default = 15S
OSG2	nnnS	Out Of Service Guard time Group 2 default = 15S
PERS	ttt	Persistence Timer for Group II problems. Enter 0 - 256 msec in increments of 2 msec. Default is 50 (=100ms).
CLRS	ttt	Clearance Timer for Group II problems. Enter 0 - 256 msec in increments of 2 msec. Default is 50 (=100ms).

Prompt	Response	Description
OOSC	nnn	Out of Service Counter. Range for <i>nnn</i> of 0-255 with a default of 5.

8 Use Overlay 73 to change trunk pad category values

Prompt	Response	Description
REQ	CHG	
TYPE	PRI2	2 Mb PRI
FEAT	PADS	
PDCA	#	PAD table-0 is default and is hard coded
The following prompts define the pad levels. The receiving pad code is <i>r</i> and the transmission pad code is <i>t</i>. These entries have the range 0-15. The pad values (in decibels) relating to these codes are shown after this table.		
ONP	<i>r t</i>	On-premises extension
DSET	<i>r t</i>	Meridian Digital set (prompted only if the 2 Mb Gateway feature is equipped) Author's Note—p7/dave/gateway
OPX	<i>r t</i>	Off-premises extension
DTT	<i>r t</i>	Digital TIE trunks
SDTT	<i>r t</i>	digital Satellite TIE trunks
NTC	<i>r t</i>	Nontransmission compensated
TRC	<i>r t</i>	Transmission compensated
DCO	<i>r t</i>	digital COT, FEX, WAT, and DID trunks
VNL	<i>r t</i>	VIA NET LOSS
DTO	<i>r t</i>	2 Mb PRI2 digital TOLL office trunks
ACO	<i>r t</i>	Analogue CO or WATS trunks
AFX	<i>r t</i>	Analogue FEX trunks
ADD	<i>r t</i>	Analogue DID trunks
SATT	<i>r t</i>	analogue satellite TIE trunks
ATO	<i>r t</i>	analogue TOLL office trunks
PRI2	<i>r t</i>	2 Mb 2 Mb PRI trunk (prompted only if the 1.5/2 Mb Gateway feature is equipped and TYPE=2 Mb PRI) Author's Note—p7/dave/gateway

Prompt	Response	Description
XUT	r t	analogue CO trunk (prompted only if the 1.5/2 Mb Gateway feature is equipped and TYPE=PRI2) Author's Note—p7/dave/gateway
XEM	r t	analogue TIE trunk (prompted only if the 1.5/2 Mb Gateway feature is equipped and TYPE=PRI2) Author's Note—p7/dave/gateway

The following are the pads available to 2 Mb PRI. Positive dB represents loss and negative dB represents gain.

code	0	1	2	3	4	5	6	7
value (dB)	0.0	+1.0	+2.0	+3.0	+4.0	+5.0	+6.0	+7.0
code	8	9	10	11	12	13	14	15
value (dB)	+8.0	+9.0	+10.0	+11.0	+12.0	+13.0	+14.0	-1
code	16	17	18	19	20	21	22	23
value (dB)	-2	-3	-4	-5	-6	-7	-8	-9
code	24	25	26					
value (dB)	-10	idle	+0.6					

PRI software programming: Option 11 to Central Office (ISA)

Introduction

Connection to a Central Office can be accomplished using three methods:

- 1 **Dedicated channels** (similar to Digital Trunking-DTI) but with Calling Line Identification--CLID, speed of call setup and out-of-band signaling.
- 2 **Call by Call (ISA)** service selection **without** maximum or minimum settings for each trunk type. This arrangement would ultimately provide blockage to some routes.
- 3 **Call by Call (ISA)** service selection **with** maximum and minimum settings for each trunk type. This may also be referred to as **Dynamically allocating** trunk types on a call-by-call basis.

Note that a combination of options is also possible on the same link. An example is dynamically allocating some channels, while providing a number of dedicated channels.

Options 2 and 3 are explained in detail in this section.

Purpose

Accessing an ISA member

Integrated Service Access (Call by Call or Dynamic Channel Assignment) simply provides a path for the various services using it. Therefore the ISA MEMBER IS NEVER ACCESSED DIRECTLY. Instead, Service Routes are defined (COT, DID, WATS, FX and TIE) and these are accessed through the normal fashion (see example that follows).

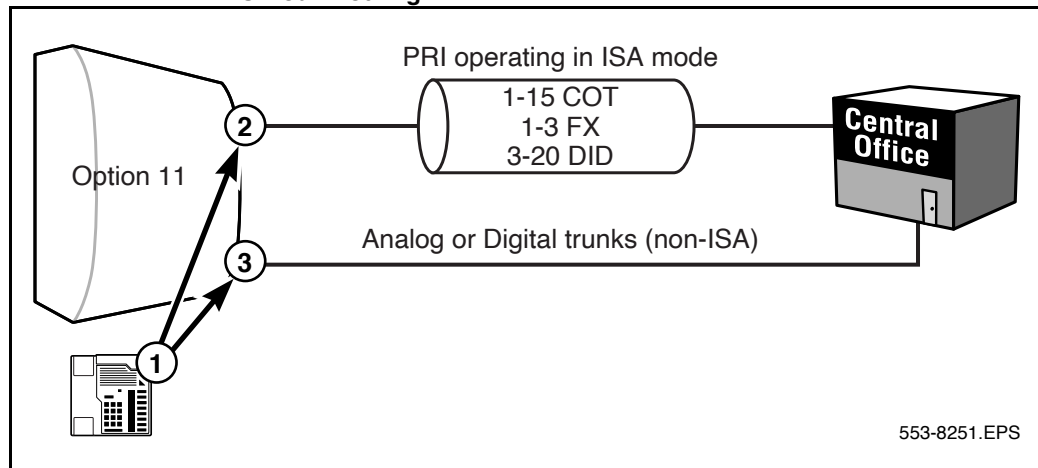
Outgoing Calls: Since ISDN requires some form of ESN (CDP or/and NARS), then the Route List Tables (ESN LD 86) include the Service route(s). Therefore, the Meridian 1 software performs the digit translation and sends the call out on an ISA route associated with the service route selected. This feature is transparent.

Incoming Calls: Similar translation is performed by the Central Office and an ISA channel is selected based on the service required. This feature is transparent.

Limitations

The caller is prevented from dialing the access code of an ISA route directly. It is also possible to step to a standard trunk route (NON-ISA) when all ISA channels are busy. This is done by using the STEP prompt (for direct access) or as part of the route list table (ESN-LD 86).

Figure 124
ISA Call Routing



- 1 Callers usually access the PRI Service Route by dialing the usual ESN number (NARS or CDP).
- 2 The Service Route does not have any trunk members of its own and therefore selects an ISA "B" channel.
- 3 In the event that all ISA "B" channels are busy or the Service Route MAX setting has been reached (i.e. 15 outgoing COT calls), stepping to a non-ISA trunk member (if so equipped) is possible.

Dependencies

Basic PRI configuration must be performed before Integrated Services Access (ISA) is defined (Steps 1, 2, 3, 4 and 8 under programming the PRI-ISA which follows). An ISA route and associated ISA trunks are then defined, followed by service routes. Service routes (COT, DID, FX, WATS, TIE) DO NOT HAVE TRUNK MEMBERS but utilize the ISA trunk members. Therefore the ISA trunks are shared by the various service routes associated with them on a call-by-call basis.

ISA Prompt Assignments

The following few pages identify and describe some of the most important prompts which are used to control the Dynamic Channel Assignment mechanism.

- a MIN and MAX
- b SID
- c NSF

MIN and MAX

You can control the number of simultaneous calls of a certain type (from a certain service route) through an ISA route by defining the Maximum and Minimum settings found in the service routes. You can define a maximum number of calls allowed per service route. In addition, you can define a minimum number of "B" channels that are ready for use by a service route (in other words channels which are reserved by a service route). The MIN and MAX prompts appear when the NSF prompt is set to YES in the ISA route.

The MIN and MAX prompts in LD 16 limit the minimum and maximum simultaneous calls associated with each service route. The rule of thumb for setting these two threshold is as follows:

MIN--The sum of all MIN values of the service routes associated with an ISA route MUST never exceed the number of "B" channels (ISA trunks) defined for the ISA route.

Sum of all the MIN values = or < total "B" channels.

MAX--The MAX settings for any service route:

MAX for service route "X" = or < Total "B" channels - (sum of all the MINs except this service route).

If the service route exceeds its maximum number of simultaneous calls allowed, or all the ISA "B" channels are busy, then you can specify an overflow route in LD 16 (STEP prompt) or LD 86 Route List Table Entry. The route number for the step prompt is not permitted to be an ISA route number or a service route number.

Service Identifier (SID)

In addition to the call type (route type), each call is identified by a service identifier (SID) which is used by the Meridian 1 and the CO for routing. The SID prompt is available when the NSF prompt is set to YES. These service identifier must be set up in coordination with the CO on a per route basis. The function of the SID is simply to identify the Service Route being used; since the Route number as we know it (LD 16) as programmed in the Meridian 1 does not mean anything to the CO

Note that what we call SID on the Meridian 1, is referred to by the DMS table as "FACNUM".

Network Service Facility (NSF)

NSF=NO

Disabling the Network Service Facility provides the Call-by-Call Pool of trunks without the MIN and MAX settings (therefore no control as to the number of "B" channels used for any one service). Additionally, when you assign NSF to NO in LD 16, a route number for COTR and TIER should also be assigned. The COTR can be a route number for an incoming CO or incoming DID route. It is used for public calls. For incoming public calls, if COTR is a CO route, it is routed to the attendant. If it is a DID route, it translates the 3 or 4 digits. For private calls (TIER), the called number is translated based on the programming for the Tie service route in overlay 16 (i.e. INAC) and ESN digit manipulation table if appropriate.

NSF=YES

This is the preferred method which provides control over the allocation of ISA "B" channels for the various Service Routes. When the NSF prompt is YES, then a route for COTR and WATR should be assigned. The COTR can be a route number for an incoming CO or incoming DID route same as above. The WATR is the route used for WATS service. The TIER prompt does not appear and TIE calls are handled based on their SID. Additionally, when NSF is set to YES, the MIN and MAX prompts must be programmed as explained earlier. When the MAX value is reached in a service route, the All Trunks Busy counter is incremented. (The ATB count is not incremented when NSF is NO). The ATB value is provided during the printing of traffic reports. The ATB counter of an ISA route is also incremented when the last "B" channel of the ISA is busied. This occurs regardless of whether the NSF prompt is YES or NO.

Configuring Option 11 to Central Office (ISA)

Applicability

Use the following procedure to configure ISA between Option 11 and a CO.

Prompts that do not show a response can be left at default. For more information on any of these prompts, refer to Option 11 *Software guides*.

Limitations

PRI loops must be configured before defining DCH links or PRI applications.

Procedure summary

Step	Overlay	Action
1	LD 17 (CFN)	Add a PRI loop.
2	LD 17 (CFN)	Add a DCHI/DDCH card.
3	LD 15 (CDB)	Define a PRI customer.
4	LD 16 (RDB)	Define a PRI service route.
5	LD 14 (TDB)	Define service channels (B channels) and PRI trunks.
6	LD 73 (PRI2)	Define clock controller parameters.
7	LD 73 (PRI2)	Define PRI parameters and thresholds.
8	LD 73 (PRI2)	Change trunk pad category values. In most cases, the settings should be left at their default values. However, if changes to these values are necessary, use this step to make them.

1 Add a PRI loop. Use Overlay 17.

Prompt	Response	Description
REQ	CHG	
TYPE	CFN	configuration data block
CEQU	YES	changes to common equipment
PRI2	XXX	where XXX is the DTI/PRI digital card number (1-9)
MODE	PRA	select Primary Rate Interface mode

2 Add a DCHI or DDCH. Use Overlay 17.

Prompt	Response	Description
REQ	CHG	
TYPE	CFN	Configuration data block
ADAN	NEW DCH 1-15 CHG DCH 1-15 Out DCH 1-15	Add a primary D-Channel port number. Any unused SDI port number Change a primary D-Channel Out the primary D-Channel
CTYP	DCHI/MSDL	DCHI for existing D-Channel configuration. (NTAK93 D-channel) MSDL for Downloadable D-Channel configuration (NTBK51 D-channel). The MSDL base and D-Channel application software will be downloadable into the DDCH card.
CDNO	1-9	Card slot number to be used as the primary DDCH/DCHI
PORT	1	PORT must be set to "1"
DES	<CR>	Back-up DCHI port number, if required
DPNS	NO	Default is no
USR	PRI	PRI: D-channel is for ISDN PRI only

Prompt	Response	Description
IFC	(D100) S100 SL1 ESS4 ESS5 AXEA SS12 SWIS D250 1TR6 AXES NUME EURO ISIG ESIG TCNZ	Interface type for the route DMS-100 SL-100 Meridian SL-1 AT&T ESS#4 AT&T ESS#5 Australian AXE-10 Norwegian SYS-12 SwissNet DMS-250 1 TR 6 Swedish AXE-10 Numeris France EuroISDN interface Interface ID for ISO QSIG Interface ID for ETSI QSIG Telecom New Zealand (NEAX-61)
DCHL	1-9	PRI card number which will be carrying the D-channel. Must match entry made for the "CDNO" associated with the "DCHI" prompt above
PRI2	<CR>	Additional PRI Loops controlled by this DCHI. Remember one DCHI can control up to 16 PRI loops going to the same destination. For the Option 11 system, the maximum limit is eight loops.
OTBF	1-(16)-127	number of output request buffers. Note: for a single PRI link, leave this prompt at default (16). Add 5 output request buffers per additional link.
SIDE	NET (USR)	Prompted only if IFC is set to SLI. Default is set to slave. NET = network, the controlling switch USR= slave to the controller Note: In X11 release 17 and earlier, enter MAS for NET and SLAV for USR.

Prompt	Response	Description
RLS	20	X11 software release of far-end. This is the current software release of the far end. If the far end has an incompatible release of software, it prevents the sending of application messages, i.e. for Network Ring Again.
RCAP	MSL	MSDL RCAP capability only applies to SL-1 interface and on release 18 or above.
OVL R	<CR>	Allow or disallow overlap receiving on a D-channel. Default is NO.
LAPD	YES,(NO)	change LAPD parameters. Enter carriage return if timers are to be left at default value. The following timers are prompted only if LAPD is set to YES. The following can all be left at default during initial set-up.
T23	1-(20)-31	interface guard timer checks how long the interface takes to respond. In units of 0.5 seconds (default 20 = 10 seconds).
T200	2-(3)-40	retransmission timer in units of 0.5 seconds (default 3 = 1.5 seconds).
N200	1-(3)-8	maximum number of retransmissions
N201	4(260)	maximum number of octets in information field
K	1-(7)-32	maximum number of outstanding unacknowledged frames (NAKS)

3 Define a PRI customer. Use Overlay 15.

Prompt	Response	Description
REQ	NEW,CHG	
TYPE	CDB NET_DATA	customer data block Release 21 gate opener
CUST	0-31	customer number
LDN	XXXX	enter the customer's Listed Directory Number
ISDN	YES	customer is equipped with ISDN
PNI	1-32700	customer private network identifier. This number MUST be unique to this customer in the private network. e.g. it is used to as part of the setup message for feature operation such as Network Ring Again, Network ACD...
HNPA	NPA	telephone area code for this Meridian 1. Sent as part of setup message as calling line identification.
HNXX	NXX	telephone local exchange code for this Meridian 1. Sent as part of setup message as calling line identification.
HLOC	XXX	home location code (NARS)
LSC	1-9999	one to four digit Local Steering Code established in the Coordinated Dialing Plan (CDP). The LSC prompt is required for Calling Line I.D. and Network ACD.
AC2		Access Code 2. Enter call types (type of number) that use access code 2. Multiple responses are permitted. This prompt only appears on NARS equipped systems. If a call type is not entered here, it is automatically defaulted to access code 1.
	NPA	E.164 National
	NXX	E.164 Subscriber
	INTL	International
	SPN	Special Number
	LOC	Location Code

4 Configure the ISA route. Use Overlay 16

Prompt	Response	Description
REQ	NEW, CHG	
TYPE	RDB	route data block
TKTP	ISA	create an ISA trunk route
DTRK	YES	digital trunk route
DGPT	PRI2	
ISDN	YES	ISDN option
MODE	PRI	route used for PRI only
PNI	1-32700	customer private network identifier-must be the same as the CDB PNI at the far end.
IFC	(D100)	interface to DMS-100
	S100	interface to SL-100
	SL1	Meridian SL-1
	ESS4	interface to AT&T ESS#4
	ESS5	interface to AT&T ESS#5
	AXEA	interface to Australian AXE-10
	SS12	interface to Norwegian SYS-12
	SWIS	SwissNet
	D250	interface to DMS-250
	1TR6	interface to 1 TR 6
	AXES	interface to Swedish AXE-10
	NUME	interface to Numeris France
	EURO	EuroISDN interface
	ISIG	Interface ID for ISO QSIG
	ESIG	Interface ID for ETSI QSIG
	TCNZ	Telecom New Zealand (NEAX-61)
NSF	YES,(NO)	Network Service facility.
COTR	0-511	incoming or combination DID/CO route number--prompted if NSF is set to YES or NO.
TIER	0-511	incoming or combination TIE route number--prompted if NSF is NO.

Prompt	Response	Description
WATR	0-511	incoming or combination WATS route number--prompted if NSF is YES.

5 Configure the ISA trunks. Use Overlay 14

Prompt	Response	Description
REQ	NEW,CHG	
TYPE	ISA	ISA trunk type
TN	c ch	enter the DTI/PRI trunk card (c) and channel number (ch). c = 1-9, ch = 1-30
RTMB	RR MM	ISA route (created in step 5) (RR) and member number (MM).

6 Define clock controller parameters. Use Overlay 73.

Free-run is used when this Meridian PBX serves as the clock source master for the private network. This is only prompted if the CC has been physically connected to the system. See the “*Network clocking*” chapter for more details on clocking.

Prompt	Response	Description
REQ	CHG	
TYPE	PRI2	2 Mb PRI
FEAT	SYTI	System timers
CLKN	1-9	Card slot number of the PRI circuit card which will have the active clock controller.
PREF	1-9	Primary reference source for clock controller. Enter the PRI card number of the PRI card which will have an active clock controller. This is the PRI which the clock controller will use as it's primary source to synchronize (to track) the system network clock. A carriage return <CR> here signifies the system will operate in free-run (non-tracking).

Prompt	Response	Description
SREF	1-9	Secondary reference source for clock controller—prompted only if primary source is not free-run. Enter the card number of the PRI card that is to be used as the secondary clock reference. This is the PRI link that the clock controller will use as it's secondary source to synchronize (to track) the network clock. It is only used when unable to track on the primary source (i.e. too many slips).
CCAR	0-(15)	Clock Controller Audit Rate. Enter the time (in minutes) between normal CC audits.

7 Define PRI parameters and thresholds. Use Overlay 73.

Prompt	Response	Description
REQ	CHG	
TYPE	PRI2	2 Mb PRI
FEAT	LPTI	
LOOP	X	X is the slot number of the 2 Mb PRI card.
MFF	AFF, (CRC)	Alternate mode or CRC multi-frame mode
ALRM	(REG), ALT	Default or alternate alarms selected
G10S		
SLP	mc mt oc ot	Slip error count, <i>mc</i> —Maintenance threshold slip count, 1-(5)-255 <i>mt</i> —Maintenance threshold time, default 24 hours, see Note after Step 3 <i>oc</i> —Out-of-service threshold slip count, 1-(30)-255 <i>ot</i> —Out of service threshold time, default 1 hour, see Note after Step 3

Prompt	Response	Description
BPV	n1 n2	Bipolar violation error count, 1-(128)-255 for n1, 1-(122)-255 for n2. n1 is multiplied by 16 to obtain the actual count, giving an actual count range of 16-4080.
CRC	n1 n2	Cyclic redundancy check error count 1-(201)-255 for n1, 1-(97)-255 for n2. n1 is multiplied by 4 to obtain the actual count, giving an actual count range of 4-1020.
FAP	n1 n2	Frame alignment problem error count 1-(28)-255 for n1, (1)-255 for n2..
RATS	1-(10)-15	Number of secs. firmware has to check BPV/CRC/FAP for excessive error rate.
GP2	T2 mt dt ct ot	Group 2 error thresholds. This is the maximum amount of time that can occur before software checks the associated thresholds of 120 to 32,640 msec and rounds it to the closest multiple of 128 msec. T2 = Error count values are in the range 1-(20)-255 mt = Maintenance threshold time (MNT)(default =100S) dt = No new data calls threshold time (NNDC)(default =12S) ct = No new calls threshold time (NNC)(default =12S) ot = Out of service threshold time (OOS)(default =4S). Note : The following requirements must be met for input: mt = >dt = >ct = >ot. Note : Threshold times must be one of the following nnnnT, nnnS, nnnM, or nnH, where nnnn is an integer and T,S,M, or H show the
MNG1	nnnM	Maintenance Guard time Group 1 default = 15M
NCG1	nnnM	No New Calls Guard time Group 1 default = 15M
OSG1	nnnM	Out Of Service Guard time Group 1 default = 15M

Prompt	Response	Description
MNG2	nnnS	Maintenance Guard time Group 2 default = 15S
NCG2	nnnS	No New Calls Guard time Group 2 default = 15S
OSG2	nnnS	Out Of Service Guard time Group 2 default = 15S
PERS	ttt	Persistence Timer for Group II problems. Enter 0 - 256 msec in increments of 2 msec. Default is 50 (=100ms).
CLRS	ttt	Clearance Timer for Group II problems. Enter 0 - 256 msec in increments of 2 msec. Default is 50 (=100ms).
OOSC	nnn	Out of Service Counter. Range for <i>nnn</i> of 0-255 with a default of 5.

8. Use Overlay 73 to change trunk pad category values

Prompt	Response	Description
REQ	CHG	
TYPE	PRI2	2 Mb PRI
FEAT	PADS	
PDCA	#	PAD table-0 is default and is hard coded
The following prompts define the pad levels. The receiving pad code is <i>r</i> and the transmission pad code is <i>t</i>. These entries have the range 0-15. The pad values (in decibels) relating to these codes are shown after this table.		
ONP	r t	On-premises extension
DSET	r t	Meridian Digital set (prompted only if the 2 Mb Gateway feature is equipped) Author's Note—p7/dave/gateway
OPX	r t	Off-premises extension
DTT	r t	Digital TIE trunks

Prompt	Response	Description
SDTT	r t	digital Satellite TIE trunks
NTC	r t	Nontransmission compensated
TRC	r t	Transmission compensated
DCO	r t	digital COT, FEX, WAT, and DID trunks
VNL	r t	VIA NET LOSS
DTO	r t	2 Mb PRI2 digital TOLL office trunks
ACO	r t	Analogue CO or WATS trunks
AFX	r t	Analogue FEX trunks
ADD	r t	Analogue DID trunks
SATT	r t	analogue satellite TIE trunks
ATO	r t	analogue TOLL office trunks
PRI2	r t	2 Mb 2 Mb PRI trunk (prompted only if the 1.5/2 Mb Gateway feature is equipped and TYPE=2 Mb PRI) Author's Note—p7/dave/gateway
XUT	r t	analogue CO trunk (prompted only if the 1.5/2 Mb Gateway feature is equipped and TYPE=PRI2) Author's Note—p7/dave/gateway
XEM	r t	analogue TIE trunk (prompted only if the 1.5/2 Mb Gateway feature is equipped and TYPE=PRI2) Author's Note—p7/dave/gateway

The following are the pads available to 2 Mb PRI. Positive dB represents loss and negative dB represents gain.

code	0	1	2	3	4	5	6	7
value (dB)	0.0	+1.0	+2.0	+3.0	+4.0	+5.0	+6.0	+7.0
code	8	9	10	11	12	13	14	15
value (dB)	+8.0	+9.0	+10.0	+11.0	+12.0	+13.0	+14.0	-1
code	16	17	18	19	20	21	22	23
value (dB)	-2	-3	-4	-5	-6	-7	-8	-9
code	24	25	26					
value (dB)	-10	idle	+0.6					

Chapter 7 — DTI Maintenance

Overview

From a maintenance perspective, Option 11 DTI operation consists of the following major aspects:

- hardware and software states
- near-end and far-end status
- link and/or span integrity
- clocking status
- frame alignment

Option 11 DTI operation is monitored and reported on through maintenance messages, out-of-service alarms, and circuit card faceplate LEDs.

Option 11 maintenance provides several tools, either manual or automatic, for maintaining effective DTI operation. These tools are service change and maintenance commands that are accessible through the software overlays and resident diagnostic routines.

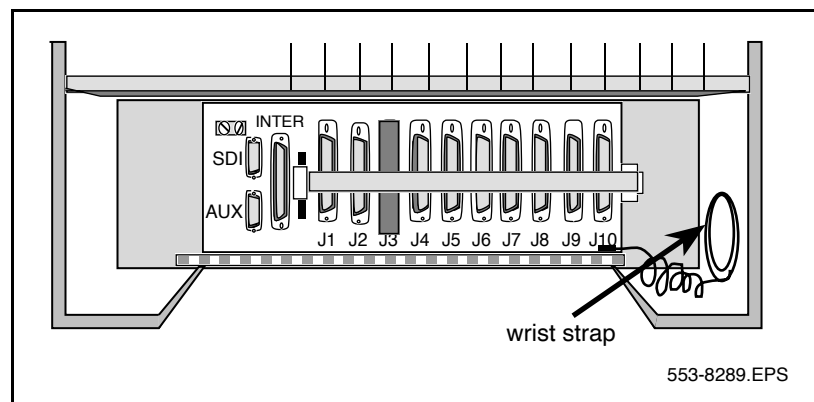
Replacing the NTAK10 2 Mb DTI card

To replace a DTI circuit pack, follow the procedure below:

- 1 Software disable the NTAK10 2 Mb DTI card(s) using overlay 60 (command DISL N, where N is the 2 Mb DTI card number).
- 2 Software disable the clock controller using overlay 60 (DIS CC 0).
- 3 Hold the NTAK10 by the lock latches, unlock the latches and slide the card out of the cabinet. Once out of the slot you may remove any of the daughterboards.

To avoid damage to the circuit cards from electrostatic discharge, wear the wrist strap connected to the inside of your Option 11 cabinet whenever handling the circuit cards. Figure 125 shows the location of the wrist strap in relation to the cabinet.

Figure 125
Wrist strap connection to the Option 11 cabinet



- 4 Slide the new card into the cabinet and lock the latches.
- 5 Software enable all NTAK10 2 Mb DTI cards using overlay 60 (command ENLL N, where N is the 2 Mb DTI card number).
- 6 Software enable the clock controller using overlay 60 (ENL CC 0).
- 7 Enable clock tracking on primary digital loop by issuing the following command TRCK PCK.

Monitoring Option 11 DTI operation

Maintenance messages

2 Mb DTI status and error conditions are reported in the following types of messages. Additional information on DTI messages can be found in the Option 11 Software Guide.

Table 63
Maintenance messages

Message	Meaning
DTA	Digital Trunk Alarms (Resident Monitor)
DTI	Digital Trunk Interface

2 Mb DTI Error messages

The Digital Trunk Interface diagnostic program (LD60) is used to maintain the NTAK10 2 Mb DTI Interface Card.

Comprehensive lists of the self-test failure codes for the NTAK10 2 Mb DTI follow the self-test description, later in this chapter.

Grade-of-Service messages

Grade-of-Service messages (DTA XXX) provide near and far end switch status. A summary of these status messages is as follows:

Status	2 Mb DTI Capability
Acceptable	Both incoming and outgoing calls allowed
Maintenance	Both incoming and outgoing calls allowed
No new data calls	No new outgoing data calls
No new calls	No new outgoing data or voice calls
Out of Service	2 Mb DTI is disabled

2 Mb DTI Alarms

There are two groups of alarm indicators monitored by the 2 Mb DTI. Within these two alarm groups there are several individual alarm types.

Group I alarms are event driven and include indicators that decrement a counter whenever an error is detected. Grade-of-service is changed based on how quickly the threshold of the counter is exceeded. The threshold count is determined in LD73 and downloaded to the 2 Mb DTI. Group 1 alarms are:

- 1 Bipolar Violations (BPV)—Near End Alarm
- 2 Slips (SLP)—Near End Alarm
- 3 Frame Alignment Problems (FAP)—Near End Alarm
- 4 Cyclic Redundancy Check (CRC-4)—Near End Alarm

Group II alarm indicators may be either continuous or spurious. Grade-of-service is changed based on the duration of the alarm within a defined period of time. A minimum persistence time is defined in LD73 and downloaded to the 2 Mb DTI.

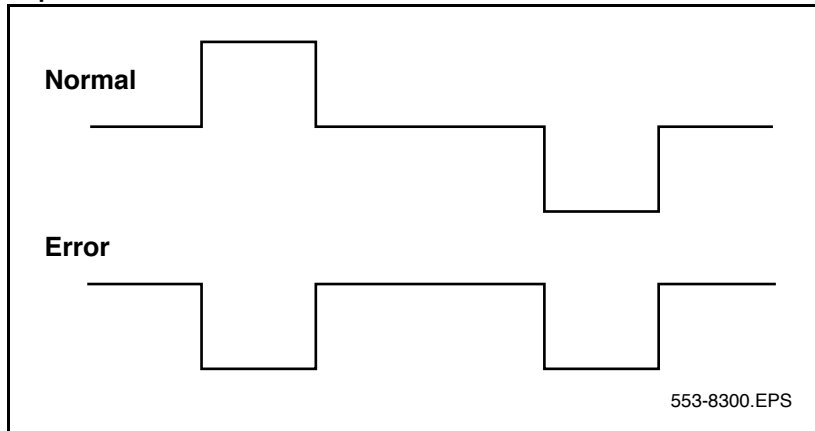
Group II alarm types are listed from high to low priority. A Group II alarm type may not be reported when a greater type is detected.

- 1 Alarm Indication Signal (AIS)—Far End Alarm
- 2 Loss of Frame Alignment Signal (LFAS)—Near End Alarm
- 3 Loss of Multiframe Alignment Signal (LMAS)—Near End Alarm
- 4 Loss of CRC-4 Multiframe Alignment Signal (CFAS)—Near End Alarm
- 5 Remote Alarm Indication (RAI, B3)—Far End Alarm
- 6 Remote Yellow Alarm (B6)—Far End Alarm
- 7 Alarm Indication Signal, 64 Kilobit (AIS 64)—Far End Alarm

Group I

Bipolar violation (BPV) In a bipolar pulse stream, pulses alternate in polarity. A bipolar violation has occurred if, after transmission, two pulses of the same polarity are received in succession (this could be caused by an electrical disturbance such as noise).

Figure 126
Bipolar violations



Cyclic Redundancy Check (CRC-4) When the 2 Mb DTI card is running in CRC-4 mode, the CRC-4 word contains a checksum of all data in the multiframe. The receiving side uses the checksum to verify the data.

The primary difference between BPV and CRC is that bipolar violation tracking indicates errors in the local span, while CRC indicates errors on an end-to-end span. For example, on a satellite link, BPV only detects errors in the span between the Meridian 1 and the satellite connection. Since CRC traverses the entire span, it indicates an end-to-end bit error rate.

Slip (SLP) Digital signals must have accurate clock synchronization for data to be interleaved into or extracted from the appropriate timeslot during multiplexing and demultiplexing operations. A Frame slip is defined as the repetition of, or deletion of 256 bits (one frame) due to a discrepancy in the read and write rates at the buffer (clocks aren't operating at EXACTLY the same speed)

When data bits are written into (added to) a buffer at a slightly *higher* rate than that at which they are being read (emptied), sooner or later the buffer overflows. This is a slip-frame deletion.

In the opposite situation, when data bits are written (added) into a buffer at slightly *lower* rate than that at which they are being read (emptied), eventually the buffer runs dry or underflows. This is also a slip-frame repetition.

All of the degradations shown in Table 64 can be controlled or avoided by proper clock (network) synchronization.

Table 64
Performance Impact of one Slip.

Service	Potential Impact
Encrypted Text	Encryption key must be resent.
Video	Freeze frame for several seconds. Loud pop on audio.
Digital Data	Deletion or repetition of Data. Possible Misframe.
Facsimile	Deletion of 4-8 scan lines. Drop Call.
Voice Band Data	Transmission Errors for 0.01 to 2 s. Drop Call.
Voice	Possible Click

Clock synchronization can be either tracking, on the primary or secondary reference clock, or free run (non-tracking). In LD 73 (prompts PREF and SREF), the 2 Mb DTI which supports the active clock controller is defined as the primary clock reference. Another 2 Mb DTI may be defined as the secondary clock reference. The clock controller synchronizes from the primary or secondary's incoming pulse stream. The clock controller in turn supplies clocking to all the other 2 Mb DTI cards.

2 Mb DTI hardware detects frame slips in tracking and free run modes. For tracking mode, running the midnight routines prints the number of overflows and clears the counter. For free run mode, running the midnight routines prints the number of slips and clears the counters.

Frame Alignment Problem (FAP) A Frame Alignment Problem is counted when a bit error occurs within the framing pattern.

Group II

Loss of Frame Alignment Signal (LFAS) A Loss of Frame Alignment condition can be defined as: frame alignment is lost on three consecutive frame alignment errors. This condition may occur as a result of the far end of the E1 span going completely out of service.(due to a power interruption, for example) or any other reason resulting in losing the incoming pulse stream. The B3 alarm is sent to the far end after the persistence time expires.

Loss of frame alignment thresholds 2 Mb DTI hardware detects out of frame conditions. Running the midnight routines prints the number of occurrences when frame alignment was lost and clears the counters.

Alarm Indication Signal (AIS) AIS is defined as receiving all ones without framing. The detection algorithm for AIS is two or less zeros per two frames. This allows detection of AIS in the presence of a $10E-3$ error rate, without mistaking framed all 1s for AIS.

When the AIS is detected, the prompt maintenance alarm indication associated with LFAS and excessive bit error rates should be inhibited. B3 alarm is sent to the far end after persistence time expires

Alarm Indication Signal 64 Kilobit (AIS64) There are also specific AIS requirements for CNET (France). The first requirement relates to the transmission of AIS in TS16. In that case, all 1 should be transmitted in the event abcd signaling cannot be supported.

With respect to the reception of AIS in TS16, AIS 64 Kbits must be detected when multiframe alignment has been lost and the binary content of TS16 is 95% 1, counting on 256 or 512 bits.

When AIS in TS16 is detected, then bit 6 of outgoing TS16 frame 0 should be set. When the fault disappears, it should then be cleared. Setting and clearing bit 6 of TRS 16 frame 0 must follow changes in status of the fault by less than 100 msec.

Other requirements when AIS 64 Kbit is detected include setting an LED, printing a TTY message and the updating of present status and history files. When a fault is detected, further changes in other error types should continue to be reported.

Loss of Multiframe Alignment Signal (LMAS) This condition is sometimes called a multiframe yellow alarm. An LMAS occurs when two consecutive multiframes contain errors in their Multiframe Alignment Signal, or when all TS16 bits are zero for at least one multiframe.

B6 alarm is sent to the far end after persistence time expires. If the option is enables, B3 alarm is also sent.

Loss of CRC-4 Multiframe Alignment Signal This condition is declared when the CRC-4 multiframe search fails to recognize two valid multiframe alignment words within 8 msec. This category is only relevant however, when CRC-4 option is selected.

B6 alarm is sent to the far end after persistence time expires. If the option is enables, B3 alarm is also sent.

Remote Alarm Indication (RAI) Bit 3 of TS0 in non-FAS frames is set to “1”. This alarm is sometimes called a yellow alarm. The far end is receiving AIS.

Remote Yellow Alarm (B6) Bit 6 of TS16 in Frame 0 is “1”. This alarm occurs when the far end has detected a loss of MFA.

NTAK10 Faceplate LEDs

The NTAK10 2 Mb DTI circuit card has a total of six LEDs on its face. Five of the LEDs are directly associated with the operation of the NTAK10 2 Mb DTI circuit card. The remaining LED is associated with the on-board clock controller. Following is a table of the LEDs found on the NTAK10 2 Mb DTI circuit card and the meaning of each:

LED	State	Definition
DIS	On (Red)	The NTAK10 2 Mb DTI circuit card is disabled
	Off	The NTAK10 2 Mb DTI is not in a disabled state
OOS	On (Yellow)	The NTAK10 2 Mb DTI circuit card is in an out of service state.
	Off	The NTAK10 is not in an out of service state.
NEA	On (Yellow)	A near end alarm state has been detected
	Off	No near end alarm
FEA	On (Yellow)	A far end alarm state has been detected
	Off	No far end alarm
LBK	On (Yellow)	NTAK10 2 Mb DTI is in loop-back mode
	Off	NTAK10 2 Mb DTI is not in loop-back mode
CC	On (Red)	The clock controller is switched on and disabled
	On (Green)	The clock controller is switched on and is either locked to a reference or is in free run mode
	Flashing (Green)	The clock controller is switched on and locking onto the primary reference
	Off	The clock controller is switched off

Using Option 11 DTI maintenance tools

The tables below provide DTI and clock controller maintenance commands

WARNING

You must disable the clock controller before unseating circuit cards, otherwise the system will initialize and momentarily interrupt call processing.

DTI commands

Below is a quick reference list of important 2 Mb DTI commands in LD 60.

Command	Action
DLBK L	disable "loopback mode" for 2 Mb DTI
RLBK L	enable "loopback mode" for 2 Mb DTI
DISI L	disable 2 Mb DTI when idle
DISL L	force disable 2 Mb DTI
ENLL L	enable 2 Mb DTI
LCNT (L)	list alarm counters
RCNT (L)	reset alarm counters
SLFT (L)	do 2 Mb DTI self-test
STAT (L)	list 2 Mb DTI status

Clock Controller commands

Below is a quick reference list of clock controller commands in LD 60.

Command	Action
DIS CC 0	disable clock controller N
ENL CC 0	enable clock controller N
SSCK 0	status of clock controller N
TRCK XXX	set clock controller tracking. XXX can be:
PCK	track primary clock reference source
SCLK	track secondary clock reference source
FRUN	free run mode

2 Mb DTI tests

Self test/Local loopback

The NTAK10 self-tests when requested in LD 60. This procedure checks the sanity of the on board processors, operation of memory and peripheral hardware as well as per-channel and per-loop loopback.

Before this test is run, the loop must be disabled as follows:

- 1 Disable the NTAK10 using LD 60:

LD 60

DISL L CH

- 2 Run the self-test using LD 60:

LD 60

SLFT L (for the entire loop)

SLFT L CH (for a specific channel)

Local loopback may also be performed on a per-channel basis without having to disable the entire loop. In this case, only the tested channel must be disabled. The procedure for this test is as follows:

- 1 Disable the 2 Mb DTI channel using LD 60:

LD 60

DISL L CH

- 2 Run the self-test using LD 60:

LD 60

SLFT L CH (for a specific channel)

For self testing individual channels, follow the same procedure as above, but use the following commands:

DSCH C CH

SLFT C CH (specific channel)

Table 65
Self-test failure codes for NTAK10 2 Mb DTI

DTI009 loop ch	DTI/PRI loop or channel failed hardware self test. For DTI009 L M E, the output data is L = loop M = N for NI microprocessor M = C for CI microprocessor E = error code for debug purposes.
DTI009C ch	(For System Option 11 only) DTI Card C or channel ch of Card C failed hardware self-test.
	Error codes for NI microprocessor (M=N): 00 = NI self test has finished. 01 = Undefined messout received 02 = Problem with group 2 error handling (invalid level) 03 = NI to CI FIFO full (128 messages lost)
	04 = CI-1 Micro failed to initialize on power-up 05 = NI group 1 error handling - undefined condition found 06 = Bad MESSOUT number 6 encountered. 07 = NI Messout queue is full.

	08 = NI Messin queue full. 09 = NI priority Messin queue is full. 10 = Bad MESSOUT number 10 encountered. 11 = TN = 0 read from regular queue. 12 = TN = 0 read from priority queue. 14 = Bad TN associated with MESSOUT number 14 15 = Bad TN associated with MESSOUT number 15 50 = External RAM in range 880h-8EFH failed (MESSIN queue) 51 = Internal RAM test failed. 52 = Pad RAM test failed. 53 = External RAM test failed.
DTI009C ch	Error codes for NI microprocessor (M=N) continued: 54 = 8253 or DALLAS timer/counter test failed. 55 = Slip counter test failed. 56 = Loopback of TS16 frame 0 failed. 57 = Loopback of non fas TSO failed. 58 = Echo test to CI-1 micro failed. 60 = A07 device failed 61 = Motorola DUART failed 62 = Multiframe loopback test failed 255 = Loss of NI FIFO synchronization (Stop byte = 0 not found). DTI009 error codes for CI microprocessor (M=C): 03 = A complete message was not received from NI micro
	128 = Message received by CI-1 through FIFO requested an un-defined task. 129 = Request for a timed two-state pulse was received, with the TN of TS0 or 16 130 = An attempt was made to set the flag to invoke the pulse timer for TS 0. 131 = A request for a task defined under Messout 30 has been received with the TN of TS 0. 132 = Attempt was made to enable outpulsing TS0 or TS16.

	133 = A Messout 31 has been received for TS 0 or TS 16 with the pulse hold time not = 0. 134 = An attempt has been made to set the bit to invoke the pulse timer for TS 0 or TS 16. 135 = A request for outpulsing was received, but outpulsing data was not downloaded. 137 = A request for PPM counting was received, but the was not downloaded. 138 = A Messout 30 was received requesting a task to be performed for DTI TS 16 which is not allowed.
DTI009C ch	DTI009 error codes for CI microprocessor (M=C): 139 = The 8031 on CHIP RAM failed self test. 140 = The TS 16 signaling RAM and/or the TS 16 pick up buffer failed self test. 141 = The CI-1 micro external RAM failed self test. 142 = Attempt was made to set/clear the flag used to invoke PPM pulse timing for DTI timeslot 0 or 16. 143 = CI-2 micro responded to echo request message but response was in error. 144 = CI-2 micro failed to respond to request echo message. 145 = The request for self test received did not have the TN of TS0 146 = The TN of MESSOUT 26 received was not that or TS 0.
	147 = The TN of MESSOUT 28 received was not that of TS 0. 148 = The TN of MESSOUT 29 received was not that of TS 0. 149 = Upon enabling the DTI card, the CI-1 was unable to write Frame 0, TS 16 with '0B'. 150 = MESSOUT 26 was received with the PPM counting bit (abcd) all equal to zero. 151 = MESSOUT 28 was received with the outpulsing bit (abcd) equal to zero. 152 = CI-2 failed to respond to the CI-1 watch dog message. 153 = The CI-2 failed to respond to five consecutive watchdog messages and is assumed to be out of service. 154 = Messout received requesting the lower nibble of MFAS pattern to be written with something other than '0000'.

DTI009C ch	155 = MESSOUT received requesting '0000' to be written into an 'abcd' state. 156 = MESSOUT received with a TN outside the range shelf 0, card 8-1 unit 3-0. DTI009 error codes for CI microprocessor (M=C): 157 = CI-1, NI FIFO overflowed, and has been cleared. 128 messages were lost. 255 = TS16-DS30X FIFO overflowed, and has been cleared. 128 messages were lost. 147 = The TN of MESSOUT 28 received was not that of TS 0. 148 = The TN of MESSOUT 29 received was not that of TS 0. 149 = Upon enabling the DTI card, the CI-1 was unable to write Frame 0, TS 16 with '0B'. 150 = MESSOUT 26 was received with the PPM counting bit (abcd) all equal to zero. 151 = MESSOUT 28 was received with the outputting bit (abcd) equal to zero. 152 = CI-2 failed to respond to the CI-1 watch dog message. 153 = The CI-2 failed to respond to five consecutive watchdog messages and is assumed to be out of service. 154 = Messout received requesting the lower nibble of MFAS pattern to be written with something other than '0000'. 155 = MESSOUT received requesting '0000' to be written into an 'abcd' state. 156 = MESSOUT received with a TN outside the range shelf 0, card 8-1 unit 3-0.
DTI009C ch	DTI009 error codes for CI microprocessor (M=C): 157 = CI-1, NI FIFO overflowed, and has been cleared. 128 messages were lost. 255 = TS16-DS30X FIFO overflowed, and has been cleared. 128 messages were lost.

Near End Trunk loopback test

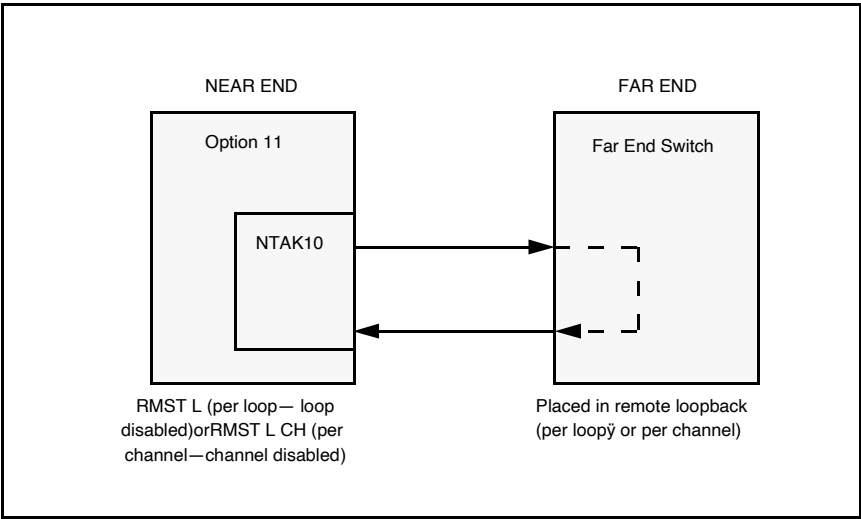
The near end trunk loopback test checks the integrity of the 2 Mb link from the Option 11 to the far end. Either a per-loop or per channel test may be run, with the far end device placed in the respective mode of remote loopback. The procedure for a near-end per-loop loopback test is as follows:

Procedure	Overlay	Command
Disable the 2 Mb DTI card	LD 60	DISL L
Have a technician at the far end place the far end device in remote per-loop loopback.		
Run the loopback test	LD60	RMST L

The procedure for a near-end per-channel loopback test is as follows:

Procedure	Overlay	Command
Disable the 2 Mb DTI channel	LD 60	DSCH L CH
Have a technician at the far end place the channel of the far end device in remote loopback.		
Run the loopback test	LD60	RMST L CH

Figure 127
Near End Trunk Loopback



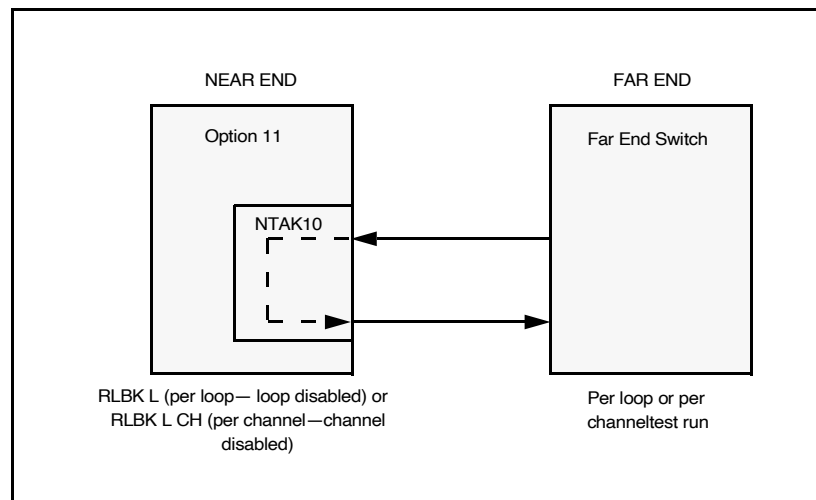
Far End Trunk loopback test

The far end trunk loopback test checks the integrity of the 2 Mb link from the far end to the carrier interface of the 2 Mb DTI (it does not test the 2 Mb DTI card). Either a per-loop or per channel test may be run, with the near end device placed in the respective mode of remote loopback. The procedures for **per-loop loopback tests** follow:

Procedure: near-end	Overlay	Command
Disable the 2 Mb DTI card	LD 60	DISL L
Place the 2 Mb DTI card in remote per-loop loopback	LD60	RLBK L
Have a technician at the far end run a per-loop loopback test		

Procedure: far-end	Overlay	Command
Disable the 2 Mb DTI channel	LD 60	DSCH L CH
Place the 2 Mb DTI channel in remote loopback	LD60	RLBK L CH
Have a technician at the far end run a loopback test on the channel		

Figure 128
Far End Trunk Loopback



Chapter 8 — PRI Maintenance

Overview

This chapter provides an overview of the maintenance tools available for 2.0 Mb PRI or ISL features:

- commands used to maintain PRI, DDCH, DCHI and clock controller hardware
- tests for 2 Mb PRI operation
- error detection for 2 Mb PRI, including the various thresholds found in the 2 Mb PRI loop timers in LD 73.

Enabling the 2 Mb PRI after installation

To enable an NTAK79 or NTBK50 PRI circuit pack, follow the appropriate procedure below.

Enabling the NTAK79 PRI

- 1 Software enable all 2 Mb PRI cards using overlay 60 (command ENLL N). The DCHI will enable automatically from the enable PRI command.
- 2 Software enable the clock controller using overlay 60 (command ENL CC 0).
- 3 Enable clock tracking on primary digital loop by issuing the following command: TRCK PCK.

Within about 30 seconds the D-channel layer 3 should be established.

- 4 You can request the current status of the D-channel by issuing the command STAT DCH. The system should respond DCH N EST in LD 96 (meaning that the D-channel is established and operational).

Enabling the NTB50 PRI

- 1 If using the NTB51 DDCH daughterboard, enable the DDCH in LD 96 by issuing the command ENL MSDL X.
- 2 Software enable all 2 Mb PRI cards using overlay 60 (command ENLL N). The DCHI will enable automatically from the enable PRI command.
- 3 Software enable the clock controller using overlay 60 (command ENL CC 0).
- 4 Enable clock tracking on primary digital loop by issuing the following command: TRCK PCK.

Within about 30 seconds the D-channel layer 3 should be established.
- 5 You can request the current status of the D-channel by issuing the command STAT DCH. The system should respond DCH N EST in LD 96 (meaning that the D-channel is established and operational).

Disabling the 2 Mb PRI before removal

To disable a PRI circuit pack, follow the appropriate procedure below.

Disabling the NTAK79 PRI

- 1 Software disable the DCHI using overlay 96, command DIS DCH N, where N is the D-channel device number.
- 2 Software disable the clock controller using overlay 60 (DIS CC 0).
- 3 Software disable the PRI card using overlay 60 (command DISL N, where N is the PRI card number).

Disabling the NTB50 PRI

- 1 Software disable the DCHI using overlay 96, command DIS DCH N, where N is the D-channel device number.
- 2 If using the NTB51 DDCH daughterboard, enter the DIS MSDL N command in overlay 96.
- 3 Software disable the clock controller using overlay 60 (DIS CC 0).

- 4 Software disable the PRI card using overlay 60 (command DISL N, where N is the PRI card number).

Monitoring Option 11 PRI operation

Maintenance messages

Service messages report on near and far end switch status. Both service and service acknowledge messages are supported on PRI B-channels and ISL channels. These messages are used for backup D channel and D channel sanity polling. The status may be in-service and out-of-service.

Service and service acknowledge messages for B-channels and ISL channels are supported between:

- Meridian 1 to Meridian 1: ISL and PRI
- Meridian 1 to CO: PRI only

The status of these messages is reported by the service and service acknowledge messages for B-channels and ISL channels:

- in-service
- maintenance
- out-of-service

Near end and far end subcategories are defined for each maintenance status. See Table 66 for possible combinations of near end and far end status, and the channel capability for each status. When the status of the near end and far end does not match, the more severe maintenance status takes effect over the less severe maintenance status.

Table 66
Maintenance message status

Near end status	Far end status	B channel capability
In-service	In-service	both incoming and outgoing calls allowed
In-service	Maintenance	only incoming calls allowed
In-service	Out-of-service	not allowed to use
Maintenance	N/A	not allowed to use
Out-of-service	N/A	not allowed to use

Service message function

Service messages are used to monitor the following:

- D-channel establishment
- D-channel sanity polling
- B-channel or ISL status change
- Channel status audit.

D-channel establishment

When the D-channel establishes, the B-channel status is supported by sending service messages for each B-channel controlled by a D-channel. This allows the far end to synchronize its channel states. These services messages are sent when the D-channel is brought up automatically by the system or manually by using LD 96.

D-channel sanity polling

If a D-channel has been idle for 30 seconds, a service message is sent to poll the sanity of the link. The service message is sent regardless of whether the near end is configured as master or slave.

B-channel status change

Whenever there is a status change for a B-channel or an ISL channel, the new status is reported to the far end in a service message. Status change can occur through service change or maintenance operations, such as the addition or deletion of a channel in LD 14, or disabling of the associated loop, shelf, card or unit in LD 30, LD 32, LD 36, LD 41 or LD 60.

Channel status audit

LD 30 is enhanced to allow channel status audit to be initiated. The channels associated with each D-channel are examined and their status is reported to the far end by service messages.

Service message commands

You activate the service messages in LD 96 on a per D-channel basis. The commands are:

- ENL SERV N: turns on the support of service and service acknowledge messages for DCH link N. The command should only be executed when the specified D-channel is in the disabled state.

- DIS SERV N: turns off the support of service and service acknowledge messages for DCH link N.
- STAT SERV (N): displays the current service and service acknowledge message SERV setting for individual DDCH N or for DCHIs.

When configuring these messages, the SERV command should only be enabled if both switches are equipped with a minimum of X11 release 15 software.

Two new statuses are added for maintenance messages, FE MbSY = Far end maintenance and FE DSBL = Far end disabled.

2 Mb PRI error detection

The Primary Rate Interface categorizes errors and alarm conditions into two main groups, Group I and Group II errors.

Group I Errors

These are real-time calculated error-rate thresholds. Group I errors can include:

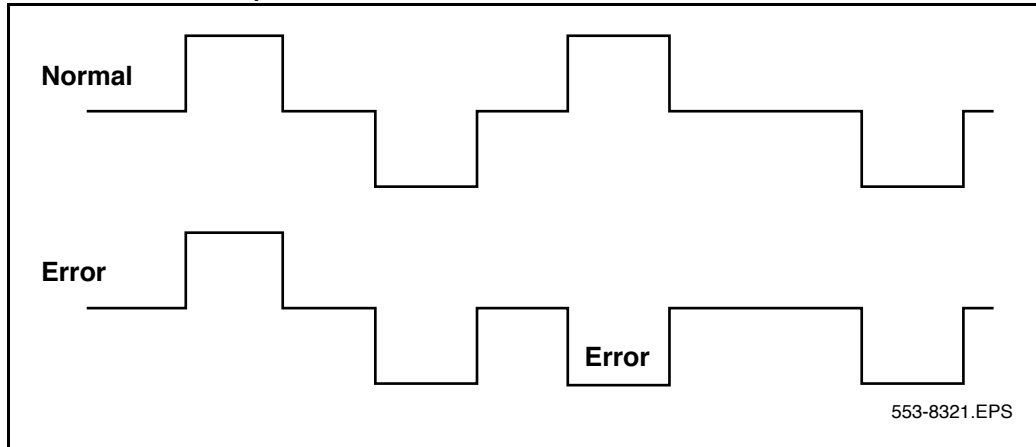
- Bipolar Violations (BPV)
- Frame Bit Errors (FBER)
- CRC-4 Word Errors (CRC)
- Controlled Frame Slips

Bipolar Violations

In a alternate mark inversion (AMI) bipolar pulse stream, pulses alternate in polarity. A bipolar violation has occurred when two pulses of the same polarity are received in succession, as shown in Figure 129. This might be caused by an electrical disturbance such as noise, a low signal level or a faulty repeater. Bipolar violations give a good indication of the integrity of the local span from the Meridian 1 to the first repeater.

The PRI card, following the CEPT PCM-30, uses a zero code suppression technique known as HDB3 which introduces intentional bipolar violations onto the carrier. This is accounted for by the receiver and disregarded as bipolar violation errors.

Figure 129
Bipolar violations



Frame Bit Errors

The NTAK79/NTBK50 PRI CEPT PCM-30 stream is structured into a frame of 32 8-bit timeslots. To maintain alignment of this frame structure, alternate timeslot zeros have a specific frame alignment pattern (X0011011).

A frame bit error is acknowledged when any of the bits in the received pattern are in error. This error rate provides an indication of the integrity of the complete end-to-end span.

CRC-4 Word Errors

When operating in the CRC-4 multiframe format (CRC), frames are bundled into groups of 16. This format includes a specific multiframe alignment pattern in bit 0 of every even frame's timeslot 0 (001011XX).

Each odd frame timeslot 0 includes a CRC-4 cyclic redundancy checksum of the previous 8 frames (sub-multiframe). This transmitted checksum is compared against a calculated checksum at the receiver. If the two checksums do not match, a CRC-4 Word error is indicated.

Group I error rates are processed on the PRI card based on software downloadable parameters N1 and N2 (for each BPV, CRC and FBER). The values for N1 are scaled on the card as follows:

$$\text{— BPV} = \text{N1} \cdot 16$$

- $CRC = N1 \cdot 4$
- $FBER = N1 \cdot 1$

N2 values are not scaled. These parameters may be established in LD 73 on a per card basis, or they can be left at their default values.

Parameters N1 and N2 provide for the following error rate thresholds reported to the system:

Severely Errored Second: when the number of occurrences of the error exceeds the value of N1 in the previous second. This error is reported and counted (LCNT), but no action is taken by the system.

Unavailable Condition: this is reported when 10 severely errored seconds are received in 10 consecutive seconds. This is equivalent to an error rate worse than 10^{-3} with the default value. When this condition is reached, the 2 Mb PRI is put into an out-of-service condition until the Group I OOS guard timer expires and the error condition has ceased or improved.

No New Call Condition: when the number of occurrences of the error exceeds the value of $(10 \times N2)$ in the previous minute. This corresponds to an error rate of 10^{-3} to 10^{-5} using the default N2 value. When this condition is reported, the system will record the error (LCNT) and place the PRI card into a no new call condition, with all idle channels set to MbSY. The card automatically returns to normal state when the condition improves after the Group I NNC guard timer expires.

Maintenance Condition: this condition implements when the number of occurrences of the error exceeds the value of N2 in the previous minute. This corresponds to an error rate between 10^{-5} and 10^{-6} based on the N2 default values. When this condition is reported, the system will record the error (LCNT) and place the 2 Mb PRI card into a maintenance alarm state. This state has no effect on call processing but simply alerts you of the line degradation. The card automatically returns to normal state when the condition improves after the Group I MAINT guard timer expires.

There is no error report for the error rates below 10^{-6} ($10 \times N2$), as such rates are considered satisfactory.

Programming Group I Thresholds

To set your Group I thresholds, use the following commands in LD 73:

Prompt	Response	Description
REQ	CHG	
TYPE	PRI2	
FEAT	LPT1	
.		
BPV	N1 N2	default: 128 122
CRC	N1 N2	default: 201 97
FBER	N1 N2	default: 28 1
.		
OOS1	1 - 60 M	default: 15
NNC1	1 - 60 M	default: 15
MNT1	1 - 60 M	default: 15

Frame Slips

Digital signals must have accurate clock synchronization for data to be interleaved into or extracted from the appropriate timeslot during multiplexing and demultiplexing operations. A Frame Slip is defined (for 2 Mb links) as the repetition of, or deletion of the 256 data bits of a CEPT frame due to a sufficiently large discrepancy in the read and write rates at the buffer (clocks aren't operating at EXACTLY the same speed).

When data bits are written into (added to) a buffer at a slightly *higher* rate than that at which they are being read (emptied), sooner or later the buffer overflows. This is a slip-frame deletion.

In the opposite situation, when data bits are written (added) into a buffer at slightly *lower* rate than that at which they are being read (emptied), eventually the buffer runs dry or underflows. This is also a slip-frame repetition.

A 2 Mb PRI contains a buffer large enough to contain 2 full frames (256 x 2 = 512 bits), and is normally kept half full (1 frame). See the following table for the impact of one slip on various types of data.

All of the degradations shown in the following table can be controlled or avoided by proper clock (network) synchronization.

Performance Impact of one Slip.

Service	Potential Impact
Encrypted Text	Encryption key must be resent.
Video	Freeze frame for several seconds. Loud pop on audio.
Digital Data	Deletion or repetition of Data. Possible Misframe.
Facsimile	Deletion of 4-8 scan lines. Drop Call.
Voice Band Data	Transmission Errors for 0.01 to 2 s. Drop Call.
Voice	Possible Click

Clock synchronization can be either tracking, on the primary or secondary reference clock, or free run (non-tracking). In LD 73 (prompts PREF and SREF), the 2 Mb PRI which supports the clock controller is defined as the primary clock reference. Another 2 Mb PRI (or DTI) may be defined as the secondary clock reference. The clock controller synchronizes from the primary or secondary's references incoming bit stream. The clock controller in turn supplies a synchronized reference for the rest of the system, including all 2 Mb PRIs and DTIs.

The 2.0 Mb PRI card detects and reports frame slips (repetitions and deletions) to the Meridian 1. The count of slips is recorded (LCNT), printed out, and cleared at each midnight routine.

Frame slips have two alarm thresholds as defined in LD 73. They are as follows:

Slip - Out of Service Threshold

When this threshold (variable count versus variable time) is reached, the PRI card is placed in an out-of-service state. The card automatically returns to a normal state if the slip rate improves after the Group I OOS guard timer expires.

Slip - Maintenance Threshold

When this threshold is reached, the PRI card is placed in an maintenance state that has no impact on call processing. The maintenance state is simply an indication that a degraded condition exists. The card automatically returns to a normal state if the slip rate improves after the Group I MAINT guard timer expires.

Defining Slip Thresholds

To define your slip thresholds, use the following commands in LD 73:

Prompt	Response	Description
REQ	CHG	
TYPE	PRI2	
FEAT	LPTI	
.		
SLIP	5 24H (This entry establishes a maintenance threshold of 5 slips in 24 hours)	20 1H (This entry establishes an OOS threshold of 20H [32] slips in 1 hour)

Times range from 1M to 24H (1 minute to 24 hours). Counts are 1 to 255.

Group II Errors

Group II errors are event-based alarms that can be separated into Red (local) and Yellow (far end) alarms.

Yellow (Far End) Alarms

A Yellow alarm on the 2.0 Mb PRI card indicates that the card is receiving an alarm indication from the far end.

This type of alarm may be received in two ways:

- **Remote Alarm Indication (RAI)**—Bit 3 of non-FAS frame set
or,
- **Alarm Indication Signal (AIS)**—All 1s on the carrier.

Since the 2 Mb PRI is receiving a Yellow alarm signal, that indicates that there is a carrier connection, but the far end is not ready. It is possible, however, that the carrier connection is one way only (2 Mb PRI receiving).

When the 2 Mb PRI receives a Yellow alarm signal, all channels are placed in a maintenance busy mode (MBSY). Group II Yellow alarms are recorded (LCNT) on each occurrence, but the card is not placed into an out-of-service state.

Red (Local) Alarms

A Red alarm on the PRI card indicates that the card is having problems synchronizing with the incoming bit stream.

This type of alarm may be caused by the following:

- **Loss of Signal (LOS)**
- **Loss of Frame Alignment (LFAS)**
- **Loss of CRC-4 Multiframe Alignment (LMAS)**

When the 2 Mb PRI is in a Red alarm state, RAI is transmitted to the far end.

For both Group II Red and Yellow alarm states, action is taken after the condition has persisted for a downloaded persistence (PERS) threshold. The parameter is set in LD 73 in multiples of 2 msec, with $2 \times 50 = 100$ msec as the default.

For Group II Red alarm conditions, there are four definable time thresholds that can be set in LD 73 on a per-loop basis. The first parameter sets an aggregate time in multiples of 128 msec (default 20=2.5 sec). The next four parameters provide time thresholds for MAINT, NNDC, NNC and OOS.

For example, if the total cumulative time that an error has been present reaches the aggregate time (2.5 sec default) in less than the time set in OOS, the card is put into an out-of-service state. If it took more than the preset time for the error condition to register, the card could be put into NNDC, NNC or MAINT states. The card remains in the alarm condition until the error has improved and after the Group II guard timers expire.

Defining Group II Error Thresholds

You are able to define Group II error thresholds in LD 73, as follows:

Prompt	Response	Description			
TYPE	2 Mb PRI				
FEAT	LPTI				
.					
GP2	20 (Aggregate count default [20x128ms= 2.5sec])	100s (Maintenance threshold)	12s (NNDC threshold)	12s (NNC threshold)	4s (OOS threshold)
.					
OOS2	1-(15)-255s	(Group II guard timer)			
NNC2	1-(15)-255s	(Group II guard timer)			
MNT2	1-(15)-255s	(Group II guard timer)			
PERS	50	(in multiples of 2ms)			
CLRS	50	(in multiples of 2ms)			
OOSC	1-(5)-255	(Out of service count limit) After this number is reached, the card does not auto-enable.			

Only the highest priority Group II alarm condition is active at a time. The order of priority (from highest to lowest) is: LOS, AIS, LFAS, LMAS, RAI.

NTAK79 Faceplate LEDs

The NTAK79 circuit card has a total of seven faceplate LEDs. Five of the LEDs are directly associated with the operation of the Primary Rate interface (PRI). The remaining two LEDs are associated with the on-board Clock Controller and the on-board D-channel interface (DCHI).

NTAK79 faceplate LEDs

LED	State	Definition

OOS	On (Red)	The NTAK79 2 Mb PRI circuit card is either disabled or out-of-service.
	Off	The NTAK79 2 Mb PRI is not in a disabled state.
ACT	On (Green)	The NTAK79 2 Mb PRI circuit card is in an active state.
	Off	The NTAK79 2 Mb PRI is in a disabled state. The OOS LED is red.
RED	On (Red)	A red alarm state has been detected. This represents a local alarm state of Loss of Carrier (LOS), Loss of Frame (LFAS) or Loss of CRC Multiframe (LMAS).
	Off	No red (local) alarm.
YEL	On (Yellow)	A yellow alarm state has been detected. This represents a remote alarm indication from the far end. The alarm may be either Alarm Indication (AIS) or Remote Alarm (RAI).
	Off	No yellow (remote) alarm.
LBK	On (Green)	NTAK79 2 Mb PRI is in loop-back mode.
	Off	NTAK79 2 Mb PRI is not in loop-back mode

NTAK79 faceplate LEDs (continued)

LED	State	Definition
CC	On (Red)	The clock controller is switched on and software disabled
	On (Green)	The clock controller is enabled and is either locked to a reference or is in free run mode
	Flashing (Green)	The clock controller is enabled and is locking onto a reference.
	Off	The clock controller is switched off (by switch SW3).
DCH	On (Red)	DCHI is switched on and disabled
	On (Green)	DCHI is switched on and enabled, but not necessarily established.
	Off	DCHI is switched off (by switch SW1).

NTBK50 Faceplate LEDs

The NTBK50 circuit card has a total of seven faceplate LEDs. Five of the LEDs are directly associated with the operation of the Downloadable D-Channel handler and D-Channel interface. The remaining two LEDs are associated with the Clock Controller.

NTBK50 faceplate LEDs

LED	State	Definition
OOS	On (Red)	The NTBK50 2 Mb PRI circuit card is either disabled or out-of-service. Also, the state of the card after power-up, completion of self test, and exiting remote loopback.
	Off	NTBK50 is not in a disabled state.
ACT	On (Green)	NTBK50 PRI circuit card is in an active state.
	Off	NTBK50 2 Mb PRI is in a disabled state. The OOS LED is red.
RED	On (Red)	A red alarm state has been detected. This represents a local alarm state of Loss of Carrier (LOS), Loss of Frame (LFAS) or Loss of CRC Multiframe (LMAS).
	Off	No red (local) alarm.
YEL	On (Yellow)	A yellow alarm state has been detected. This represents a remote alarm indication from the far end. The alarm may be either Alarm Indication (AIS) or Remote Alarm (RAI).
	Off	No yellow (remote) alarm.
LBK	On (Green)	NTBK50 2 Mb PRI is in loop-back mode.
	Off	NTBK50 2 Mb PRI is not in loop-back mode

NTBK50 faceplate LEDs (continued)

LED	State	Definition
CC	On (Red)	The clock controller is software disabled
	On (Green)	The clock controller is enabled and is either locked to a reference or is in free run mode
	Flashing (Green)	NTAK20 is equipped and is attempting to lock (tracking mode) to a reference. If the LED flashes continuously over an extended period of time, check the CC STAT in LD60. If the CC is tracking this may be an acceptable state. Check for slips and related clock controller error conditions. If none exist, then this state is acceptable, and the flashing is identifying jitter on the reference.
	Off	The clock controller is not equipped.
DCH	On (Red)	DCH is disabled.
	On (Green)	DCH is enabled, but not necessarily established.
	Off	DCH is not equipped.

Using Option 11 PRI maintenance tools

2 Mb PRI commands (LD 60)

Command	Action
DISI L	disable 2 Mb PRI when idle
DISL L	force disable PRI
ENLL L	enable PRI
LCNT (L)	list alarm counters
RCNT (L)	reset alarm counters and clear alarms
SLFT (L)	do 2 Mb PRI self-test (2 Mb PRI must be disabled first)
STAT (L)	list 2 Mb PRI status
RLBK L (C)	enable remote loopback
DLBK L (C)	disable remote loopback
RMST L (C)	Perform remote loopback

PRI status and error conditions are reported in the following types of messages. These messages can be found in the X11 Software Guide - Including Supplementary Features.

DCHI commands

D-channel commands are found in LD 96. The following is a quick reference list of D-channel commands:

Command	Action
DIS DCHI N	disable DCHI port N
ENL DCHI N	enable DCHI port N
EST DCH N	establish D-channel N
PLOG DCHI N	print D-channel statistics log N
RLS DCH N	release D-channel N
SDCH DCH N	release a D-channel and switch D-channels
RST DCH N	reset D-channel N
STAT DCH (N)	print D-channel status (link status)
TEST 100/101	DCH tests
STAT SERV	Print the current service and service acknowledge message for DCHI N
ENL SERV N	enable service messages for DCHI N
DIS SERV N	disable service messages for DCHI N

D-channel status and error conditions are reported as DCH messages. These messages can be found in the Option 11 *Software guide*.

DDCH commands

Downloadable D-channel commands are found in LD 96. The following is a quick reference list of D-channel commands with minor modification:

Command	Action
DIS MSDL X (ALL)	disable DCHI card X
ENL MSDL X (FDL, ALL)	enable DCHI card X, with or without Force Download
RST MSDL X	Reset MSDL card X
STAT MSDL X (X (full))	Get MSDL status X, or a "FULL STATUS"
SLFT MSDL X	Execute a self test on MSDL card X
DIS LLB X	Disable local loop back on MSDL DCH X
DIS RLB X	Disable remote loop back on MSDL DCH X
DIS TEST X	Disable Test mode on MSDL DCH X
ENL LLB X	Enable local loop on MSDL DCH X
ENL RLB X	Enable remote loop on MSDL DCH X
ENL TEST X	Enable Test mode on MSDL DCH X
PCON DCH X	Print configuration parameters on MSDL DCH X
PMES DCH X	Print incoming layer 3 messages on MSDL DCH X
PTRF DCH X	Print traffic report on MSDL DCH X
TEST LLB X	Start local loop back test on MSDL DCH X
TEST RLB X	Start remote loop back test on MSDL DCH X

Note: "X" represents the D-Channel device number

D-channel status and error conditions are reported as DCH messages. These messages can be found in the Input/Output guide.

Clock Controller commands

Clock Controller commands are accessed using LD 60. The following is a quick reference list of clock controller commands:

Command	Action
DIS CC 0	disable clock controller
ENL CC 0	enable clock controller
SSCK 0	status of clock controller
TRCK XXX	set clock controller tracking where XXX can be: PCK—track primary clock reference source SCLK—track secondary clock reference source FRUN—free run mode

NTAK79/NTBK50 Power on self test

When power is applied to the NTAK79/NTBK50 2 Mb PRI circuit card, the card performs a power-on self-test. The self-tests verify the operation of most of the on-board hardware.

If all the self-tests pass, the upper 5 LEDs blink simultaneously three times. If any of the self-tests fail, the LEDs do not blink. Only the OOS LED illuminates. The corresponding error code is then printed on the TTY.

Self-test error codes

The following table lists the self test failure codes for the NTAK79/NTBK50 2 Mb PRI. These codes could be returned on card power-up in the form “DTA105 L X” (where X is the failure code), or during a self test procedure in LD 60 as “DTI009 L X” (where X is the error code).

Failure codes 1-14 are hardware failures on the NTAK79/NTBK50 card. Codes 15-16 may be due to carrier span problems or lack of loopback activation at the far end.

Failure Code	Associated Error
0	Self test passed
1	Self test general failure
2	LCAs failed to program correctly
3	8031 code checksum failure
4	8031 internal RAM failure
5	8031 external RAM failure
6	PAD RAM failure
7	AO7 signaling interface failure
8	UART (card LAN) failure
9	CEPT transceiver failure
10	Line interface failure
11	Receiver framing failure
12	Transmit/receive (inter)national bit failure
13	Yellow (remote) alarm failure
14	PCM path integrity failure
15	Loop remote loopback failure
16	Channel remote loopback failure

2 Mb PRI self test

This self test can be run manually on a per-loop or per channel basis using overlay 60. The DCHI/DDCH and 2 Mb PRI must be disabled before performing the self-test.

Self-test (entire loop)

- 1 Disable the DCHI/DDCH using:

LD 96

DIS DCH N

- 2 Disable the 2 Mb PRI card and run the self-test using:

LD 60

DISL L

SLFT L (entire loop)

Self-test (specific channel)

DSCH L CH (disable an idle channel)

SLFT L CH (specific channel)

2 Mb PRI automatic loop test

This procedure is not recommended since it causes yellow at the far-end and there is a possibility of some of the B-channels not being re-enabled once the test is completed. Therefore leave the ATLP command set to 0.

The automatic loop test checks the same functions as the self test. Unlike the self test, it can be run automatically as part of the midnight routines.

Procedure (2 Mb PRI automatic loop test):

- 1 With ATLP command set to one.
- 2 If all 30 channels are idle at midnight, the Meridian 1 software disables the card and performs a self test on all channels. This causes a yellow alarm to be generated to the far end.

- 3 If any of the 30 channels are busy at midnight, the software disables one idle channel, chosen at random, and checks it while the card is enabled.

With the ATLP command set to zero, only one channel is tested. The channel tested is randomly selected by software; it cannot be specified.

To perform the automatic loop test as part of midnight routines, use

LD 60

ATLP 1 or 0

Link diagnostic and remote loop back tests

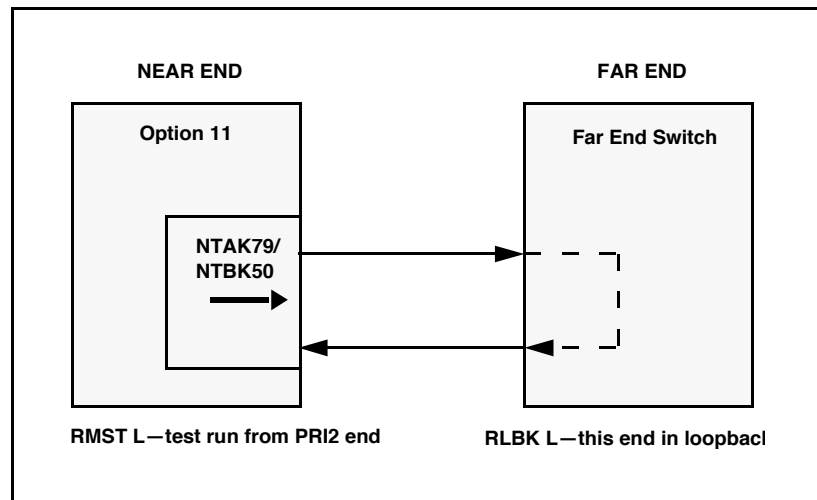
The remote loop-back and the link diagnostic test are performed manually on a per channel or per-loop (or card) basis.

Link diagnostic test

The link diagnostic test, also called the far end loop-back test, does not test the Meridian 1 2 Mb PRI. It puts the 2 Mb PRI in loop-back mode so a remote loop-back test can be performed on equipment at the far end. (See Figure 130.)

The 2 Mb PRI channel or loop (card) tested must be disabled.

Figure 130
2 Mb PRI Link Diagnostic (Far End Loopback) Test



Running the link diagnostic test

To run the link diagnostic test on the Meridian 1, use the following procedure:

- 1** Call a technician at the far end. Ask for loopback mode at that facility.
- 2** When loop-back mode at the far end is confirmed:

- Disable the DCHI/DDCH using:

LD 96

DIS DCH N

- Disable the 2 Mb PRI card and run loop-back test using:

LD 60

DISL L

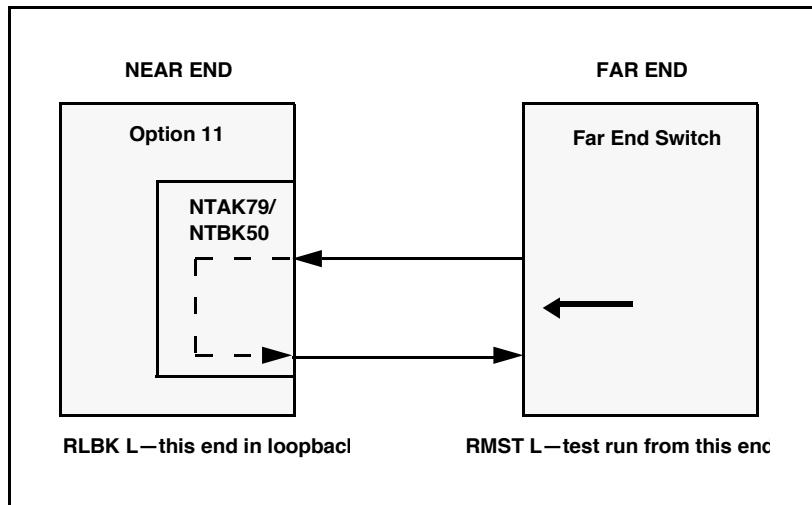
RMST L or RMST L C

Remote loop-back test

The remote loop-back test also called the near end loop-back test, checks the integrity of the 2 Mb PRI from the Meridian 1 system to the far end. The far end must be in loop-back mode before this test can be performed. (See Figure 131.)

The 2 Mb PRI channel or loop (card) tested must be disabled.

Figure 131
2 Mb PRI Remote Loopback Test



Coordinating the remote loopback tests

When a technician at the far end asks for loop-back mode on the Meridian 1:

Disable the DCHI/DDCH using:

LD 96

DIS DCH N

Disable the 2 Mb PRI card and activate loopback mode using:

LD 60

DISL L

RLBK L or RLBK L C

Chapter 9 — DPNSS1/DASS2 Implementation

Overview

This chapter provides installation instructions for DPNSS1 or DASS2. Hardware installation for DPNSS1/DASS2 is the same as for 2 Mb PRI installation found in Chapter 7. Make sure switch SW1 on the 2 Mb PRI card is set for DPNSS1 mode of operation.

DPNSS1/DASS2 Software programming

Use the following Overlay sequence to configure DPNSS1/DASS2

Step	Overlay	Action
1	17	DDCS loop & D-channel
2	74	Manipulating data blocks
3	15	Customer data block
4	16	Route data block
5	14	Trunk data block
6	73	Define clock controller parameters.
7	75	Bring loop into service

Step 1: Use LD 17 to configure the DPNSS1/DASS2 signaling link (DDSL) and PRI card number (DDCS).

Prompt	Response	Comment
req	chg	
type	cfn	
ADAN		
USR		
IFC		
DPNS	YES	Default is NO
DCHI	0-15	D-channel Port number This prompt is only given if DPNS is YES.
CDNO	1-9	DCHI card number
PORT	1	ESDI port number. On the 2 Mb PRI card, port number is always 1.
...		
parm	yes	Only required when making changes to buffers
...		
DTIB	35-1000	Size of IDA trunk input buffers for system (determined according to traffic)
DTOB	4-100	Size of IDA trunk output buffers per DTSL (determined according to traffic)
...		
cequ	yes	Change to CE parameters
...		
DDCS	1-9	Loop number for NT DPNSS1/DASS2 hardware. Enter multiples separated with a space.

The system must be initialized to invoke changes made to DTIB and DTOB. The previous prompts already exist. The sequence outlined in the previous table is used to configure the DDCS loop used either for either DPNSS1 or DASS2.

Step 2: Use overlay 74 to set protocols for the DPNSS1/DASS2 link.

Prompt	Response	Comment
REQ	NEW	
TYPE	DDSL	DPNSS1 link
DDSL	0-9	NT hardware
SIGL	DA	DPNSS1/DASS2 signaling
DDCS	1-9	PRI card number used for the DPNSS1/DASS2 link
PRIV	(YES)/NO	Enter YES for a private link (i.e. DPNSS1 to another PBX). Enter NO for a link to the public exchange (DASS2).
SIDE	(AET)/BNT	A or B end of DPNSS1 link. DASS2 defaults to B.
FLOW CNTL	YES/(NO)	Change flow control parameters. Prompted for DASS2 links only. Note: Must be set to NO if user to user or Tie line signaling are not used.
BRST PARM	0/4/8/16/32	Enter the flow control burst parameter set on the public network. Prompted only if FLOW CNTL =YES. See Note 2 below for more information.
REPL PARM	1/2/4/8	Enter the flow control replenishment parameter set on the public network. This is only prompted if FLOW CNTL =YES.

Prompt	Response	Comment
CNTL	YES/(NO)	Enter YES to receive the next two prompts.
ALRM	TBF PP MM CC FAE PP MM CC HER PP MM CC TSF PP MM CC AIS PP MM CC LOI PP MM CC DAI PP MM CC	Enter the desired persistence time (PP), monitor time (MM), and repeat count threshold (CC) for one of the seven types of alarms. The alarm condition thresholds are shown below.

Alarm Mnemonic	PP	MM	CC
TBF	0-15 secs (5)	0-24 hrs (0)	0-15 (1)
FAE	0-15 secs (2)	0-24 hrs (1)	0-15 (4)
HER	0-15 mins (1)	0-24 hrs (1)	0-15 (10)
TSF	0-15 secs (0)	0-24 hrs (0)	0-15 (0)
AIS	0-15 mins (1)	0-24 hrs (1)	0-15 (4)
LOI	0-15 secs (0)	0-24 hrs (0)	0-15 (0)
DAI	1-15 mins (1)	0-24 hrs (1)	0-15 (5)
CNTR	CRT, TMT, SCT 0- 255	Only prompted if CNTL=YES. Enter the desired threshold for one of the three counters in the range 0-254. If 255 is entered, the threshold is set to infinity. The defaults are: CRT (channel reset threshold) 120 TMT (test message threshold) 50 SCT (stop count threshold) 20	

A DPNSS1 link must have one end set to AET and the other end set to BNT.

Flow control is used to control the number of messages that are sent across a DASS2 link when using user to user or Tie line signaling features.

N = the number of messages that can be sent

X = the value of BRST PARM

Y = the value of REPL PARM

On call set up, $N = X$.

On every message sent, $N = N - 1$

Every 10 seconds, $N = N + Y$

N can never be greater than X

X must be equal to or greater than Y

If $N = 0$, no messages can be sent.

Step 3: Use LD 15 to define a DPNSS1/DASS2 customer.

Prompt	Response	Comment
REQ	NEW,CHG	
TYPE	CDB NET_DATA	Customer Data Block Release 21 gate opener
CUST	0-31	Customer number
LSC	dddd	For DPNSS1, enter the location number that identifies this PBX.
TIDM	YES/(NO)	Used for DPNSS1/DASS2 only. Enter YES if the Trunk Group reference number of a Trunk Identity is meaningful (as part of the CDP DN). Enter NO if the PBX reference number is to be displayed without the Trunk Group Reference Number.
DASC	dddd	Enter the access code that is to be placed on displays before OLI's and TLIs received from the DASS2 trunk. This may be any one to four digits. Entering the attendant's DN will remove an existing value. The value also defaults to nothing if <CR> is entered.

Step 4: Use LD 16 to initialize the routes being used.

Prompt	Response	Comment
REQ	NEW, CHG	
TYPE	RDB	Route Data Block
CUST	0-31	Customer number
ROUT	XXX	Route number
TKTP	IDA	Trunk type = IDA (Integrated Digital Access)
SIGL	DPN DAS	Level 3 signaling: enter DPN for DPNSS1 signaling, or DAS for DASS2 signaling.
...		
ICOG	IAO ICT OGT	Incoming and Outgoing trunk Incoming trunk Outgoing trunk
ACOD	XXXX	Access code for direct access to the route. The ACOD has to be set, but after the set-up of a network numbering scheme it is only used for test access.

Step 5: Use LD 14 to initialize the channels within the routes.

Prompt	Response	Comment
REQ	NEW, CHG	
TYPE	RDC VDC	Real Digital Channel Virtual Digital Channel (DPNSS1 only)
TN	Ill ch	Loop and channel number for first trunk For DNPSS/DASS2, loop is 1-9 and channel is 1-15 or 17-31.
DDSL	0-15	DASS2/DPNSS1 D-channel number. Only prompted for NT hardware.
SIGL	DPN DAS	DPNSS1 channel DASS2 channel
CUST	0-31	Customer number
RTMB	XX YY	Route and member number
INC	(YES)/NO	If YES, channel numbers will be associated with members starting at the TN, both channel and member numbers increasing. If NO, member numbers decrease as channel numbers increase.
PRI0	(XHP) YLP	High priority on channel seizure Low priority on channel seizure

Note: On DPNSS1 links only, the high/low priority must be different at each end. Virtual channels for DPNSS1 use the same TNs as real channels.

Step 6: Use LD 73 to set clock references for the loop.

Prompt	Response	Description
REQ	CHG	
TYPE	PRI2	2 Mb PRI
FEAT	SYTI	Change the switch timers and counters for PRI2. There is only one such block per switch.
CLKN	1-9	Card slot number of the NTAK79 circuit card which supports the active Clock Controller
PREF	1-9	Primary reference source for clock controller. Enter the PRI card number of the NTAK79 PRI card which supports the active Clock Controller This is the PRI E1 span which the clock controller will use as it's primary source to synchronize (to track) the system network clock. A carriage return <CR> here signifies the system will operate in free-run (non-tracking). PREF should only be removed if the system is a master for clock synchronization.
LPTI		
SYTI		
— Continued —		

Free-run is used when this Meridian PBX serves as the clock source master for the private network. See the “*Network clocking*” chapter in this guide for more details on clocking.

Step 6: (continued)

Prompt	Response	Description
SREF	1-9	Secondary reference source for clock controller-- prompted only if primary source is not free-run. Enter the PRI card number of the DTI/PRI card which will be used as the secondary clock reference. This is the PRI which the clock controller will use as it's secondary source to synchronize (to track) the system network clock. Only used when unable to track on the primary source.
CCGD	0-(15)-1440	CC Free Run Guard Time.
CCAR	0-(15)	Clock Controller Audit Rate. Enter the time in minutes between normal CC audits.
EFCS	(NO), YES	Enable Fast Clock Switching Note: EREF option in LD 60 must be chosen to enable this prompt.

Threshold times may be entered in milliseconds (X=T), seconds (X=S), minutes (X=M) or hours (X=H). For milliseconds, nn is multiples of 20 ms, in the range 1 to 5000 (effective time range of 20 to 5000 ms). For seconds, nn is in increments of seconds, in the range 1 to 240. For minutes, nn is in increments of minutes, in the range 1 to 240. For hours, nn is in hour increments, in the range 1 to 24. This is the same as for the existing ISDN PRI feature.

Step 7: Use LD 75 to bring the link into service:

- 1** Enable the DDCS loop (ENL DDCS cc)
- 2** Enable the DCHI link (ENL DDSL n). This puts the DCHI into the ENBL IDLE state.
- 3** Start the DCHI link (STRT n). This makes the DCHI go from ENBL STARTING to ENBL ACTIVE.

Both ends of the link should be started within 5 minutes of each other.

Chapter 10 — DPNSS1/DASS2 Maintenance

Overview

This chapter provides maintenance information for DPNSS1/DASS2.

D-channel service states

The D-channel handler interface (DCHI) has five software service states:

- 1 Disabled Not Responding** - in this state the DCHI is disabled and does not respond to a read/write test. All channels are disabled.
- 2 Disabled Responding** - in this state the DCHI only responds to an enable message.
- 3 Enable Idle** - in this state the DCHI responds to messages sent to it but all channels are disabled. This is a transitory state during the enable sequence.
- 4 Enabled Starting** - in this state the DCHI responds to messages sent to it. The DCHI is in the process of being started and some channels are in the 'Enabled Idle' state while others are in the 'Disabled' state.
- 5 Enabled Active** - in this state the DCHI responds to all messages sent to it and all channels which are configured should be 'Enabled Idle' or in one of the busy states.

B-channel service states

Each of the channels can be in one of eleven states:

- 1 **Unequipped** - the channel is not configured in software
- 2 **Disabled** - the channel is in the disabled state
- 3 **Enabled Idle** - the channel is available for use
- 4 **Enabled Software Busy** - the channel is being used for speech transmission.
- 5 **Enabled Maintenance Busy** - the channel is maintenance busied
- 6 **Enabled Permanently Busy** - the channel is more permanently busied for outgoing calls
- 7 **Disabled but resetting**
- 8 **Enabled Idle but Resetting**
- 9 **Enabled Software Busy but Resetting**
- 10 **Enabled Maintenance Busy but Resetting**
- 11 **Enabled Permanently Busy but Resetting**

Manual maintenance

Overlay 75 commands

Maintenance requirements are handled by overlay 75, which can be loaded from a teletype or from a maintenance set. Overlay 75 allows you to do the following:

- Disable and enable individual channels when the DCHI is ENBL ACTIVE.

Obtain the status of a DCHI, DDCCS or channel.

Clear the maintenance display and/or minor alarm.

Command	Description
ENL DDSL n	Enables DCHI n
ENL DDCS 1	Enables the PRI loop 1
ENL DTRC 1 c	Enables real channel 1 c
ENL DTVC 1 c	Enables virtual channel 1 c
DIS DDSL n	Disables DCHI n
DIS DDCS 1	Disables the DDCS loop 1
DISI DDCS 1	Disables all channels on loop 1 as they become idle
DIS DTRC 1 c	Disables real channel 1 c
DIS DTVC 1 c	Disables virtual channel 1 c
STAT DDSL	Gives status of all DCHIs
STAT DDSL n	Gives status of DCHI n
STAT DDCS	Gives status of all DDCS loops
STAT DDCS 1	Gives the status of DDCS loop 1, as well as a count of the number of channels in each state
STAT DTRC 1 c	Gives status of real channel 1 c
STAT DTVC 1 c	Gives status of virtual channel 1 c
STRT n	Starts DCHI n
CDSP	Clears the maintenance display
CMIN U	Clears the minor alarm for customer U

Bringing an IDA link into service

In overlay 75:

- 1 Enable the DDCS loop.
- 2 Enable the DCHI link. This puts it into the ENBL IDLE state.
- 3 Start the DCHI link. This makes it go from ENBL STARTING to ENBL ACTIVE.

Both ends of the link should be started within 5 minutes of each other.

Taking a link out of service

In overlay 75:

- 1** Disable all the channels
- 2** Disable the DCHI link. This makes it go from ENBL IDLE to DSBL.
- 3** Disable the DDCS loop

Step 1 can be omitted, but calls will be force disconnected by Step 2. If an enabled card is removed, the system initializes.

DPNSS1/DASS2 alarms

The software currently monitors the following alarms associated with the DPNSS1/DASS2 link.

- TBF - Transmit Buffer Full
- FAE - Frame Alignment Error
- HER - High Error Rate
- TSF - Transmit Signaling Failure
- AIS - Alarm Indication Signal
- LOI - Loss of Input
- DAI - Distant Alarm Indication

There are two criteria:

- An alarm is present for more than the 'persistence time' defined for that alarm.
- An alarm occurs more times than the 'reset count threshold' within the period defined by the 'monitor time' for that alarm.

In either case, the link is stopped, and a minor alarm is raised. When all alarms are cleared, the link is restarted. Various diagnostic messages are issued for alarms.

Test Messages

Test messages are sent to all DCHIs every 30 seconds in order to check the level 3/level 2 interface. The test patterns are echoed back unchanged. If the number of failed tests since midnight exceeds the "Test Message Threshold" (TMT) defined, then the link is reset and a minor alarm is raised.

TMT may be set to infinity if 255 is entered, in which case the link is not reset due to tests failing.

A check is also carried out every 30 seconds on the DCHI states perceived by the hardware and software. If there is a difference, then the link is reset and a minor alarm is raised.

Various diagnostic messages are issued for these faults.

Initialize

When the system is initialized, all PRIs and DCHIs are tested and reset. If the initialize follows a system reload or is the result of pressing the manual initialize button, then all links are brought into service. If the initialize occurs for any other reason, the links which are not disabled are reset.

Link Reset

When certain faults are detected, the DCHI is reset. This amounts to taking the link of service (so that the DCHI is disabled) and then bringing it back into service. This sequence may fail, in which case the link could be left disabled or idled.

Channel Reset

The software may reset a channel:

- when there is difficulty clearing a call;
- each time a channel is enabled;
- when the channel buffer on the card overflows.

The channel is disabled (any call in progress is force disconnected), and the associated lap is reset; the channel is re-enabled when the reset is completed.

The DDCS may itself initiate a channel reset, i.e. when there is difficulty communicating with the far end.

Various diagnostic messages are issued for a channel reset.

If the number of channel resets since midnight exceeds the "Channel Reset Threshold" (CRT) defined, then the link is reset and a minor alarm is raised.

CRT may be set to infinity if 255 is entered, in which case the link is not reset due to channel reset failure.

Stop Count

A count is kept of the number of times since midnight that a link is stopped due to an alarm or for a link reset. If the count exceeds the "Stop Count Threshold" (SCT) then the link is disabled. It remains disabled until it is brought back into service manually.

A diagnostic message is issued for this.

SCT may be set to infinity if 255 is entered, in which case the link is not disabled due to excessive stopping.

Channel Configuration

DPNSS1 and DASS2 are message based signaling systems which use a common signaling channel in timeslot 16. Each traffic channel has an associated LAP; the LAPs operate in parallel over the signaling channel. Each traffic channel (together with its LAP) represents one trunk.

A DCHI is informed of the configuration of its LAPs each time it is enabled. If a discrepancy between the hardware and software configurations is detected during call processing, the software attempts to correct the DCHI's configuration.

Level 2 - Level 3 Communication

If Level 3 (i.e. software) cannot send a message to level 2 (i.e. the DCHI) because no output buffer is available, a diagnostic message is issued. If level 2 cannot send a message to level 3 because no input buffer is available, there is no immediate message. Both cases are included in traffic printouts.

Input messages received by level 3 are checked for validity (i.e. the length must be consistent with the message type). A diagnostic message is issued for any discrepancy.

Chapter 11 — ISL Implementation and Maintenance

Overview

This chapter provides the information needed to implement ISL on the Meridian 1 Option 11 system. It provides:

- hardware and software installation
- programming all components needed to provide basic call service

ISDN features are treated separately in the chapter devoted to ISDN feature implementation.

This chapter builds on the ESN programming that is assumed to be already in place. It is also built on the assumption that you have a basic understanding of NARS and CDP.

Two modes of ISL are available: shared and dedicated. This chapter covers ISL installation in dedicated mode using dedicated and leased lines. Shared mode installations are done according to the instructions supplied for PRI, with the noted exceptions.

ISL Hardware Requirements

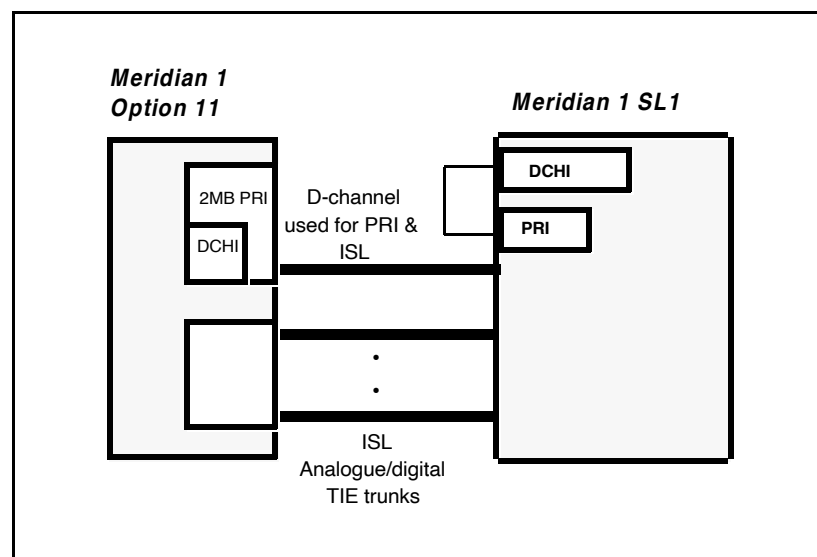
ISL, regardless of the mode of operation, uses analogue or digital TIE lines as B-channels.

ISL in Shared mode

In shared mode, PRI hardware is required in addition to the existing TIE lines interface cards.

- NT8D15 Analogue trunk card(s)
- NTAK02 SDI/DCH, NTAK10 2.0 Mb DTI, NTAK79 2.0 Mb PRI, or NTBK50 2.0 Mb PRI card(s)

Figure 132
ISL in Shared Mode:

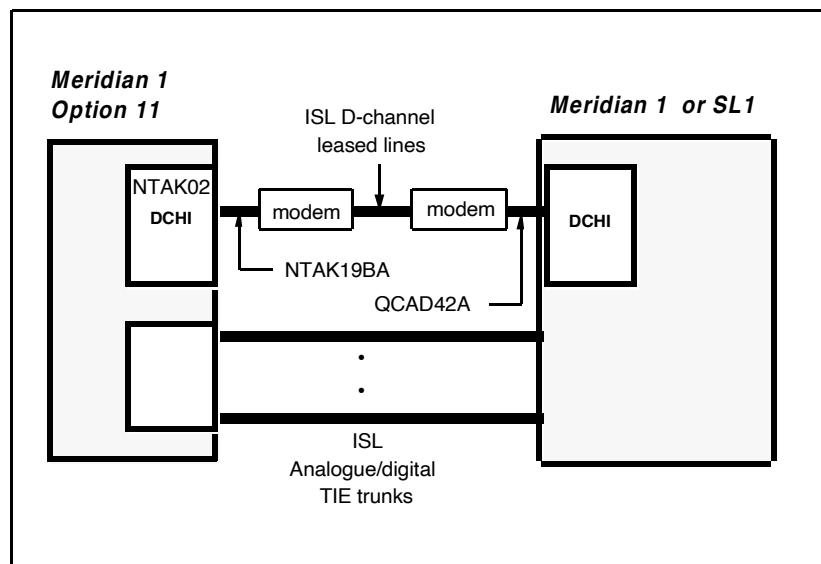


ISL in Dedicated Mode using Leased Line

The following hardware is required:

- NTAK02 SDI/DCH D-channel Handler Interface
- Modem capable of the following: (such as Ventel 2400-33 or 2400 Plus II)
 - minimum of 2400 baud
 - synchronous operation
 - must support leased line (also known as private line or point-to-point) operation
- NTAK19BA 4-Port SDI Cable

Figure 133
ISL in Dedicated Mode using Leased Line

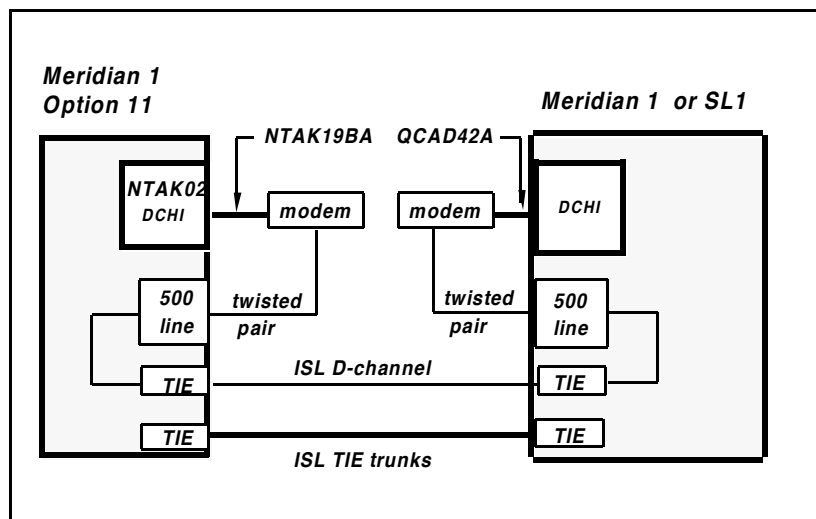


ISL in Dedicated Mode using Dial-up modem

The requirements are as follows:

- NTAK02 SDI/DCH D-channel Handler Interface
- Modems such as Ventel 2400, Hayes 2400 (the Hayes Smartmodem 2400 cannot be used on leased lines) or Gandalf 2400 that can support 2 or 4-wire leased line operation (4-wire operation must be specified when ordering otherwise modems are factory shipped for 2-wire operation).:
 - autodial capability
 - minimum of 2400 baud
 - synchronous operation
 - programmable so that one modem originates the call while the other auto-answers
- NTAK19BA 4 Port SDI Cable
- NT8D09 500 set line card

Figure 134
ISL in Dedicated Mode using Dial-up Modem



ISL hardware installation

Shared mode

The hardware installation is identical to the PRI installation, with the addition of analogue or digital TIE trunks (or both).

Dedicated mode

Step 1: Install the NTAK02.

The NTAK02 connects to the modem through the NTAK19BA four port cable. Only ports 1 and 3 are available for use as DCHIs.

- a Set option switches/jumpers on the DCHI card as shown below, for mode of operation (i.e. RS232 or RS422 and DTE or DCE)
- b Install the NTAK02 in any spare slot 1-9 of the main cabinet
- c Install the NTAK19BA four port cable on the 50 pin Amphenol connector associated with the slot holding the NTAK02.

NTAK02 switch settings

Port 0	Port 1	SW1-1	SW1-2
SDI	DCH	OFF	OFF
SDI	DPNSS1	OFF	ON
—	ESDI	ON	ON

Port 2	Port 3	SW1-3	SW1-4
SDI	DCH	OFF	OFF
SDI	DPNSS1	OFF	ON
—	ESDI	ON	ON

NTAK02 Jumper settings

Unit	Jumper location	Strap for DTE	Strap for DCE	Jumper location	RS422	RS232
Unit 0	J10	C - B	B - A			
Unit 1	J7 J6	C - B C - B	B - A B - A	J9 J8	C - B C - B	B - A B - A
Unit 2	J5	C - B	B - A			
Unit 3	J4 J3	C - B C - B	B - A B - A	J2 J1	C - B C - B	B - A B - A

Step 2: set up the D-channel

- a** If this is a dedicated mode installation using leased line modems, the D-channel connects the DCHI with the far-end modem over a dedicated leased line. Synchronous modems with a minimum 2400 baud data rate must be configured. Modems must support leased line capability and synchronous mode. The Hayes Smartmodem 2400 cannot be used on leased lines
- b** If this a dedicated mode using dial-up modems (modems such as Hayes 2400, Ventel 2400 or Gandalf 2400 can be used): in this configuration, the DCHI is connected to a modem which is connected to a 500 set line card. The call is connected to the far-end through the 500 set-to-TIE trunk path.

To set-up the D-channel, program the modem at one end in the auto-dial mode, so it automatically initiates a call to the other end at power up. The auto-dial number must be coordinated with the far end switch. The originate modem has this auto-dial number stored internally as part of the modem configuration routine.

The far-end modem need only be set-up for auto-answer.

ISL software programming

There are two modes of ISDN Signaling Link (ISL) operation, Shared mode and Dedicated mode.

Shared mode

The NTAK79 on-board DCHI supports ISDN PRI signaling and ISL trunks. The configuration is basically the same as the PRI D-channel, with the D-channel also supporting ISL trunks.

The DTI/PRI software programming sequence can be used (refer to the appropriate chapters for more information) with the following exceptions:

LD 17:

Prompt	Response	Description
USR	SHA	D-channel for ISL in “shared” mode, used for both ISDN PRI and ISL.
ISLM	1-240	number of ISL B-channel (trunks) controlled by the D-channel (no default value).

LD 16:

Prompt	Response	Description
IFC	SL1	interface type must be SL1 (this is the only type supported for ISL).
MODE	ISLD	TIE route used for ISL members.

LD 14:

Prompt	Response	Description
CHID	1-240	channel identifier for ISL channels. Must be coordinated with the far-end.

Dedicated mode

The DCHI uses the NTAK02 circuit card, and does not support ISDN PRI signaling. The DCHI is reserved for ISL use only. The D-channel can communicate with the far end by means of a dedicated leased line modem, or dial-up modem.

Note that the following programming relates to analogue TIE trunks being used as B-channels. In the case where DTI/PRI trunks are also used then LD 17 digital loop (2 Mb PRI) and LD73 (2 Mb PRI/SYTI) must also be configured with the appropriate clocking and threshold settings.

For ISL dedicated mode using a dial-up modem, a 500 set, TIE trunk route and member must be programmed (used for D-channel).

Configuring Basic ISL Capability

Step	Overlay	Action
1	LD 17 (CFN)	Configure the D-channel for ISL use.
2	LD 15 (CDB)	Enable ISDN option
3	LD 16 (RDB)	Enable the ISL option on a per route basis, assign a D-channel for each route
4	LD 14 (TDB)	Assign a channel identification to each trunk with the ISL option.

Step 1: Respond to the following prompts in Overlay 17.

Prompt	Response	Description
REQ	CHG	
TYPE	CFN	configuration data block
ADAN	NEW DCH 1-15	add primary D-channel
CTYP	DCHI	D-channel card type
CDNO	1-9	Card slot in which the card supporting the DCHI resides
PORT	1	must be set to 1
USR	PRI	D-channel for ISDN PRI only
IFC	SL1	Interface type
DCHL	1-9	PRI2 card number. (Must match entry for CDNO.)
SIDE	NET (USR)	Net: network, the controlling switch User: slave to controller
RLS	XX	X11 software release of far-end. This is the current software release of the far end. If the far end has an incompatible release of software, it prevents the sending of application messages, i.e. for Network Ring Again.
CLOK		D-channel clock type for signaling
	EXT	source of D-channel clock is external to DCHI card (in this case the DTI/PRI circuit card). Normally, EXT is used for PRI/ISL. Note: do not confuse this clock with the E1 span Clock Controller found on the NTAK10/79. This clock is in reference to the DCHI synchronous mode of operation. (Note: If directly connecting two DCHI ports with out the use of modems, set "CLOK" to "EXT" on one side and "INT" on the other).
LAPD	YES,(NO)	change LAPD parameters. Simply carriage return if timers are to be left at default value. The following timers are prompted only if LAPD is set to YES. The following can all be left at default during initial set-up.

Prompt	Response	Description
T23	1-(20)-31	interface guard timer checks how long the interface takes to respond. In units of 0.5 seconds (default 20 = 10 seconds).
T200	2-(3)-40	retransmission timer in units of 0.5 seconds (default 3 = 1.5 seconds).
N200	1-(3)-8	maximum number of retransmissions
N201	4(260)	maximum number of octets in information field
K	1-(7)-32	maximum number of outstanding unacknowledged frames (NAKS)

Step 2: Use Overlay 15 to enable the ISDN option.

Prompt	Response	Description
REQ	NEW,CHG	
TYPE	CDB NET_DATA	Customer Data Block Release 21 gate opener
CUST	0-31	customer number
ISDN	YES	customer is equipped with ISDN
PNI	1-32700	customer private network identifier. MUST be unique to this customer in the private network. Used as part of the setup message for feature operation such as Network Ring Again and Network ACD...
HNPA	NPA	telephone area code for this Meridian 1. Sent as part of setup message as CLID
HNXX	NXX	telephone local exchange code for this Meridian 1. Sent as part of setup message as calling line identification.
HLOC	XXX	home location code (NARS)
LSC	1-9999	one to four digit Local Steering Code established in the Coordinated Dialing Plan (CDP). The LSC prompt is required for Calling Line I.D. and Network ACD.
AC2		Access Code 2. Enter call types (type of number) that use access code 2. Multiple responses are permitted. This prompt only appears on NARS equipped systems. If a call type is not entered here, it is automatically defaulted to access code 1.
	NPA	E.164 National
	NXX	E.164 Subscriber
	INTL	International
	SPN	Special Number
	LOC	Location Code

Step 3: Use Overlay 16 to enable the ISL option on a per route basis.

Prompt	Response	Description
REQ	NEW,CHG	
TYPE	RDB	route data block
CUST	0-31	customer number
ROUT	0-127	route number
TKTP	TIE	TIE trunk route
DTRK	YES/NO	enter YES if this is a Digital Trunk Interface (DTI or PRI)
ISDN	YES	ISDN option
MODE	ISLD	Route for ISL application
DCHI	XX	DCHI port no. in CFN which will carry the D-channel for this TIE trunk route.
PNI	1-32700	customer private network identifier- must be the same as the CDB PNI at the far end.
IFC		interface type
	SLI	- select Meridian 1 to Meridian 1
CTYP	<CR>	Call Type. Enter the call type to be associated with the outgoing route for direct dialing using the trunk access code (instead of NARS access code) See the "X11 Software Guide - Including Supplementary Features" for a list of valid responses
INAC	YES	Insert Access Code. Permits the NARS AC1 or AC2 access code to be re-inserted automatically on an incoming ESN call.

Step 4: Respond to the following prompts in Overlay 14.

Prompt	Response	Description
REQ	NEW,CHG	
TYPE	TIE	TIE trunk type
TN	CC UU	Card (CC) and Unit (UU) to be associated with this trunk data
RTMB	RR MM	route (created in step 3) (RR) and member number (MM)
CHID	1-240	channel identifier for ISL channels (remove with Xnn)—must be coordinated with far end (no default value)

ISL maintenance

DCHI Quick Reference

Below is a quick reference list of D-channel commands (LD 96).

Command	Action
DIS DCHI N	disable DCHI port N
ENL DCHI N	enable DCHI port N
EST DCH N	establish D-channel N
PLOG DCHI N	print D-channel statistics log N
RLS DCH N	release D-channel N
SDCH DCH N	release a D-channel and switch D-channels
RST DCH N	reset D-channel N
STAT DCH (N)	print D-channel status (link status)
TEST-100/101	DCH tests
STAT SERV	Print the current service and service acknowledge message for DCHI N
ENL SERV N	enable service messages for DCHI N
DIS SERV N	disable service messages for DCHI N

Maintenance messages

D-channel status and error conditions are reported as DCH messages. These messages can be found in the Option 11 *Software Guide - Maintenance Messages*.

Maintenance testing

ISL Back to Back (without Modems)

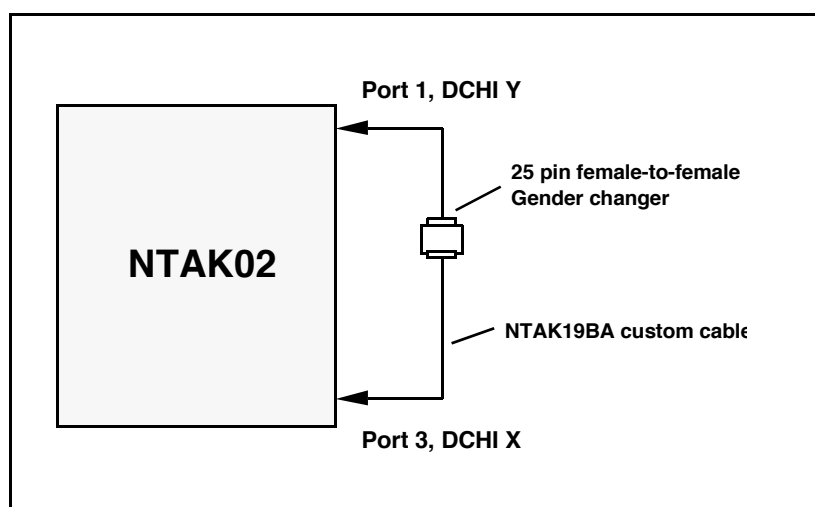
For maintenance reasons or testing purposes it is sometimes necessary to connect ISL back to back (without modems). This connection is normally done within the same Meridian 1 system in a lab environment.

Hardware requirements:

A 25-pin female-to-female gender changer is required to connect the NTAk19BA cable back-to-back. The gender changer is not supplied with the Option 11 system. Figure 135 illustrates the connection.

Note: Protocol converters AO378652 and AO381016 supplied with the system are not gender changers.

Figure 135
ISL back-to-back connection



Port settings:

- For port 3, DCHI X, program the following settings in LD 17:
 - CLOK = INT (Internal clock)
 - SIDE = SLAV (Slave)

Set NTAk02 port 3 jumper plugs to DCE and RS-422.

- For port 1, DCHI Y, program the following settings in LD 17:
 - CLOK = EXT (External clock)
 - SIDE = MAS (Master)

Set NTAK02 port 1 jumper plugs to DTE and RS-422.

Chapter 12 — APNSS Implementation and Maintenance

Overview

This chapter provides information on implementing and maintaining APNSS in an Option 11 system.

APNSS hardware requirements

APNSS requires the following hardware:

- NTAK02 SDI/DCH card (each SDI/DCH card can handle two D-channels)
- NT5K19 Analog Tie trunk card
- modem

Modem settings for APNSS applications

Two modems can be used in the United Kingdom for Analog Private Network Signaling System (APNSS) applications:

- BT 4242VSX modem
- Datel 4960FTX modem

BT 4242VSX modem

If the BT 4242VSX modem is used, set the option strap settings as follows:

SW1	ON	OFF	OFF	ON	OFF	OFF	ON	OFF
SW2	OFF	OFF	OFF	OFF	OFF	OFF	OFF	OFF
SW3	OFF	OFF	OFF	OFF	OFF	OFF	OFF	OFF
SW4	OFF	OFF	OFF	OFF	OFF	OFF	OFF	OFF
SW5	ON	ON	ON	OFF	OFF			

Datel 4960FTX modem

If the Datel 4960FTX modem is used, set the option strap settings as follows:

00	0	29	0
01	1	30	0
07	0	31	1
09	1	32	0
10	1	33	0
11	0	34	0
12	0	35	0
13	0	38	1
14	0	39	1
15	0	72	0
16	0	74	0
17	0	75	0
24	1	76	0
25	1	77	0
26	1	79	0
27	0	80	0
28	0		
SW1: 1, 2, 3, and 4 ON			
5, 6, 7, and 8 OFF			
SK3 must be installed			

APNSS software programming

Follow the steps below to install an APNSS link. The steps must be followed in sequence. The prompts and responses for these steps are explained in the text that follows. Responses in parentheses are default values throughout the procedure.

Step	Overlay	Action
1	LD 17 Configuration Record	Configure the APNSS D Channel Interface Handler, and the unused loop for Virtual channels
2	LD 74 DDSL Data Block	Define the data blocks used for the APNSS protocols
3	LD 16 Route Data Block	Initialize the service routes to be used
4	LD 14 Trunk Data Block	Initialize the channels within the service routes
5	LD 75 IDA Trunk Maintenance	Enable the APNSS link

Step 1: Use overlay 17 to configure the APNSS signaling link (DDSL) and unused loop for virtual channels. The system must be initialized to invoke changes made to DTIB and DTOB.

Prompt	Response	Comment
Req	chg	Change existing data base
Type	cfn	Configuration data block
DPNS	YES	Digital Private Network Signaling. Default is NO
DCHI	0-15	D-channel Port number This prompt is only given if DPNS is YES.
CDNO	1-9	Card number for NTAK02 SDI/DCH card

Prompt	Response	Comment
PORT	1,3	ESDI port number on the NTAK02 SDI/DCH card. Ports 1 and 3 are valid
DTIB	35-1000	Size of IDA trunk input buffers for system (determined according to traffic). Note: The system must be initialized to invoke changes to DTIB.
DTOB	4-100	Size of IDA trunk output buffers per DTSL (determined according to traffic). Note: The system must be initialized to invoke changes to DTOB.
...		
cequ	yes	Change to CE parameters
...		
APVL	1-9	Spare loop for APNSS virtual channels. Cannot conflict with TDS, Digital Trunk card or a PRI card. The recommended position is that of the SDI/DCH card. One specified APVL can be used for more than one APNSS link as long as different channels are used for each link.

Step 2: Use overlay 74 to set protocols for the link.

Prompt	Response	Description
REQ	NEW	
TYPE	LSSL	Low speed signaling link, identifies this channel as APNSS
LSSL	nn	Link number identifying the D-channel to be used
SIGL	DA	Level 2 signaling (DASS2)
RATE	EXT	clock from modem
SIDE	AET, BNT	A or B end of APNSS link
CNTL	YES, (NO)	Change link control parameters.
ALRM	TBF PP MM CC FAE PP MM CC HER PP MM CC TSF PP MM CC AIS PP MM CC LOI PP MM CC DAI PP MM CC	Enter the desired persistence time (PP), monitor time (MM), and repeat count threshold (CC) for one of the seven types of alarms. The alarm condition thresholds are shown below.
CNTR	0- 255 (CRT) (TMT) (SCT)	Only prompted if CNTL=YES. Enter the desired threshold for one of the three counters in the range 0-254. If 255 is entered, the threshold is set to infinity. The defaults are: CRT (channel reset threshold) 120 TMT (test message threshold) 50 SCT (stop count threshold) 20

Alarm condition thresholds

Alarm Mnemonic	PP	MM	CC
TBF	0-15 secs (5)	0-24 hrs (0)	0-15 (1)
FAE	0-15 secs (2)	0-24 hrs (1)	0-15 (4)
HER	0-15 mins (1)	0-24 hrs (1)	0-15 (10)
TSF	0-15 secs (0)	0-24 hrs (0)	0-15 (0)
AIS	0-15 mins (1)	0-24 hrs (1)	0-15 (4)
LOI	0-15 secs (0)	0-24 hrs (0)	0-15 (0)
DAI	1-15 mins (1)	0-24 hrs (1)	0-15 (5)

Step 3: Use overlay 16 to initialize the routes.

Prompt	Response	Description
REQ	NEW, CHG	
TYPE	RDB	
CUST	0-99	Customer number
ROUT	XXX	Route number
TKTP	IDA	Trunk type = IDA (Integrated Digital Access)
SIGL	APNSS	APNSS route
...		
ICOG	IAO ICT OGT	Incoming and Outgoing trunk Incoming trunk Outgoing trunk
ACOD	XXXX	Access code for direct access to the route. The ACOD has to be set, but after the set-up of a network numbering scheme it is only used for test access.

Step 4: Use overlay 14 to initialize the channels within the routes. On APNSS links, high and low priority must be different at each end.

Prompt	Response	Comment
REQ	NEW, CHG	
TYPE	RAC VAC	Real Analog Channel Virtual Analog Channel
TN	CC UU Ill ch	TN for real channels on NT5K19 E&M routes. Channel on APVL - virtual channels (Not prompted for VAC)
XTRK	XFEM	Always XFEM cards for APNSS
DTSL	0-15	DDSL number for APNSS link. Not prompted for VAC
CUST	0-99	Customer number
.		
RTMB	XX YY	Route and member number
CHID	1-15, 17-30	Channel ID for each TN
INC	(YES), NO	Member number (increased) or decreased with channel number
PRI0	(XHP) YLP	High priority on channel seizure Low priority on channel seizure

Step 5: Use overlay 75 to bring the link into service. Both ends of the link should be started within 5 minutes of each other.

- 1 Enable the DCHI link (ENL DDSL n). This puts the DCHI into the ENBL IDLE state.
- 2 Start the DCHI link (STRT n). This makes the DCHI go from ENBL STARTING to ENBL ACTIVE.

APNSS maintenance

APNSS maintenance is nearly the same as DPNSS1 maintenance, except where noted below:

Overlay 32

The STAT command (STAT L S C and STAT L S C U) displays “IDA” to indicate APNSS.

Overlay 36

As in overlay 32, the STAT command displays “IDA”.

Overlay 60

The status of DTI2 channels will display the status of IDA trunks (as for other trunk types).

Overlay 75

The APNSS link is managed in the same manner as DTSL/DDSL:

ENL LSSL xto enable LSSL x

DIS LSSL xto disable LSSL x

STAT LSSL xto print a status of link x

The status of the channels, both real and virtual, can be obtained from the STAT command as follows:

STAT LSRC x (y)status of real channel y on LSSL x

STAT LSVC x (y)status of virtual channel y on LSSL x

Chapter 13 — ISDN Feature Implementation

Overview

The information found in this section can be used to program option features associated with ISDN, and assumes that the basic 2 Mb PRI feature, as described in "2 Mb PRI Implementation", has been implemented completely.

Note: In this section, responses in parentheses are the default responses.

510 Trunk Route Member Expansion

To configure the 510 Trunk Member Expansion feature:

LD 16 - Define the trunk route.

LD 14 - Define the trunk type, and the route member number.

LD 27 - For ISDN BRI trunking, define the trunk type as BRIT, and the route member number on a Digital Subscriber Loop (DSL).

LD 16 - Define the trunk route.

Prompt	Response	Description
REQ	NEW CHG	Add a new data. Change existing data.
TYPE	RDB	Route Data Block.
CUST	xx	Customer number, as already defined in LD 15. xx = 0-31 for Option 11C.
ROUT	0 - 511 0-127	Route Number for Options 51C-81C. Route Number for Option 11C.
DES	x...x	Designator field for trunk (0-16 alphanumeric characters).
TKTP		Supported trunk type.
	COT TIE DID FEX WAT FGDT	Central Office. TIE Direct Inward Dial. Foreign Exchange. Wide Area Telecommunications Service. Feature Group D.
...		
DTRK	(NO) YES	Define whether or not the trunk route is digital. Enter NO for analog, YES for digital.
- DGTP	PRI	For digital trunks, enter PRI as the digital trunk type. This prompt appears only if DTRK = YES.
...		

ISDN	(NO) YES	Integrated Services Digital Network option. Enter NO for analog trunks or YES for digital trunks.
- MODE	aaaa	Mode of operation, where aaaa can be: APN = Analog Private Network. ISLD = ISDN Signaling Link. PRA = Primary Rate Access (ISDN must be YES).
- IFC	aaaa	Interface type, where aaaa can be: D100 = DMS-100. D250 = DMS-250. SL1 = Meridian 1 (the default value).
...		
ICOG	aaa	Incoming or outgoing trunk type, where aaa can be: ICT = Incoming only. OGT = Outgoing only. IAO = Both incoming and outgoing.
...		
PTUT	0-510	Preference Trunk Usage Threshold.
...		

LD 14 - Define the trunk type, and the route member number.

Prompt	Response	Description
REQ	NEW CHG	Add new data. Change existing data.
TYPE	COT TIE DID FEX WAT FGDT	Trunk type. Central Office. TIE. Direct Inward Dial. Foreign Exchange. Wide Area Telecommunications Service. Feature Group D.
...		
TN	c u c ch	TN address, for Option 11C, where: c = card u = unit. Card and channel for digital trunks, for Option 11C. c = card ch = channel
...		
CUST	xx	Customer number, as already defined in LD 15. xx = 0-31 for Option 11C.
...		
RTMB	0-127 1-510	Route number and member number, for Option 11C.
...		

LD 27 - Define the trunk type as BRIT, and the route member number on a Digital Subscriber Loop (DSL).

Prompt	Response	Description
REQ	NEW CHG	Add new data. Change existing data.
TYPE	DSL	Administer the DSL data block.
DSL		DSL address for Option 11C.
	c dsl#	c (card) = 1-9. dsl# (DSL location) = 0-7.
DES	d...d	DSL designator (1 to 6 alphanumeric characters).
APPL	aaaa	Basic Rate Interface application (BRIT or BRIE).
...		
B1	(NO) YES	(Do not) change B-Channel 1 configuration.
- MEMB	1-510	Member number of BRI route.
...		
B2	(NO) YES	(Do not) change B-Channel 2 configuration.
- MEMB	1-510	Member number of BRI route.
...		

Advice of Charge

The AOC capability may be configured for the following Central Office connectivities:

- AXE-10 Australia (at end of call)
- EuroISDN (at call set-up, during a call, end-of-call)
- INS1500 (Japan D70) (at end of call)
- Numeris (at end of call)
- Numeris (during a call)
- SwissNet (during a call)
- 1TR6 (at end of call)

AXE-10 Australia (end of call)

LD 17 – Select PPM functionality and CDR format.

Prompt	Response	Description
REQ	CHG	Change.
TYPE	CFN	Configuration record.
...		
PARM	YES	Change system parameters.
MTRO	PPM	Periodic Pulse Metering.

LD 15 – Allow Charge Display.

Prompt	Response	Description
REQ	CHG	Change.
TYPE	PPM	Periodic Pulse Metering data.
CUST	xx	Customer number.
...		
- UCST	x	Unit cost, x = (0)-9999.

LD 15 – Allow CDR Charge.

Prompt	Response	Description
REQ	CHG	Change.
TYPE	FTR	Customer featuresdata.
CUST	xx	Customer number.
...		
OPT	CHDA	Charge display allowed. Release 21 gate opener is FTR.
...		

LD 16 – Allow AOC on the route.

Prompt	Response	Description
REQ	CHG	Change.
TYPE	RDB	Route data block.
...		
ISDN	YES	Configure ISDN data.
IFC	AXEA	ISDN Interface for Australia. Must be configured in LD 17 for the applicable D-channel.
...		
CDR	YES	Call Detail Recording.
...		
OTL	YES	CDR on outgoing toll calls.
...		
OAN	YES	CDR on all answered outgoing calls.
...		
MR	ENDC	AOC End of Call allowed for international ISDN interfaces.
RUCS	1	Route unit cost value received is treated as charge.
RUCF	1 0	Route unit conversion factor, no conversion required.

LD 10/11– Assign meters to sets.

Prompt	Response	Description
REQ	NEW, CHG	New, or change.
TYPE	aaaa	Telephone type where aaaa = 500,2500, SL1, 2006, 2008, 2009, 2016, 2018, 2112, 2216, 2317, 2616, or 3000.
TN	c u	Terminal Number.
...		
CLS	MRA	Message Registration Allowed.

LD 12 – Assign a Meter Recall key to the Attendant Console.

Prompt	Response	Description
REQ	NEW, CHG	New, or change.
TYPE	aaaa	Type of console, where aaaa = 1250 for M1250 console, 2250 for M2250 console, ATT for QCW3, QCW4 console, or PWR if the TN is used for power or Attendant Supervisory Module ASM.
TN	c u	Terminal Number.
...		
KEY	xx MTR	Assign a Meter recall key on the attendant console.

EuroISDN (call set-up, during call, end of call)

LD 17 – The following steps assume an ISDN interface is already set up.

Prompt	Response	Description
REQ	CHG	Change.
TYPE	ADAN	Change I/O devices.
- ADAN	CHG DCH XX	Change D-channel, xx = 0-15.
...		
- IFC	EURO	EuroISDN interface.
- CNTY	AUS DEN EIR (ETSI) FIN GER ITA NOR POR SWE DUT	Austria. Denmark. Ireland. ETS 300-102 basic protocol. Finland. Germany. Italy. Norway. Portugal. Sweden. Holland.

LD 15 – Allow Charge Display and CDR Charge.

Prompt	Response	Description
REQ	CHG	Change.
TYPE	CDB	Customer data block.
CUST	xx	Customer number.
...		
OPT	CHDA	Charge display allowed. Release 21 gate opener is FTR.
...		
- UCST	x	Unit cost, x = (0)-9999. Release 21 gate opener is PPM.

LD 16 – Allow AOC on the route.

Prompt	Response	Description
REQ	CHG	Change.
TYPE	RDB	Route data block.
CUST	xx	Customer number.
ROUT	xx	Route number.
...		
ISDN	YES	Configure ISDN data.
...	...	Other ISDN sub-prompts.
-IFC	EURO	EuroISDN interface.

-- CNTY	AUS DEN EIR (ETSI) FIN GER ITA NOR POR SWE DUT	Austria. Denmark. Ireland. ETS 300-102 basic protocol. Finland. Germany. Italy. Norway. Portugal. Sweden. Holland.
...		
MR	(NO) ENDC DURC STAC	No AOC service. AOC-E subservice activated. AOC-D subservice activated. AOC-S subservice activated. Not printed for Denmark and Sweden.
RUCS	0-9999	Route unit cost. Not printed for Denmark and Sweden
RURC	0-9999 (0)-3	Route unit reference cost. Note that the formula for the route unit reference cost is: $X \cdot 10^{(-Y)}$, where $X=0-9999$, and $Y=0-3$. The default value for X is identical to the previously entered RUCS value. Not printed for Denmark and Sweden.
RUCF	0-(1)-9999 (0)-3	Route unit conversion factor. Note that the formula for the route unit reference cost is: $X \cdot 10^{(-Y)}$, where $X=0-9999$, and $Y=0-3$. The default value for X is identical to the previously entered RUCS value. Not printed for Denmark and Sweden.

LD 10/11 – Assign meters to sets.

Prompt	Response	Description
REQ	NEW,CHG	New, or change.
TYPE	aaaa	Telephone type where aaaa = 500, 2500, SL1, 2006, 2008, 2009, 2016, 2018, 2112, 2216, 2317, 2616, or 3000.
TN	c u	Terminal Number.
CLS	MRA	Message Registration allowed.

LD 12 – Assign a Meter Recall key to the Attendant Console.

Prompt	Response	Description
REQ	NEW, CHG	New, or change.
TYPE	aaaa	Type of console, where aaaa = 1250 for M1250 console, 2250 for M2250 console, ATT for QCW3, QCW4 console, or PWR if the TN is used for power or Attendant Supervisory Module ASM.
TN	c u	Terminal Number.
...		
KEY	xx MTR	Assign a Meter recall key on the attendant console.

INS1500 (Japan D70) (end of call)

LD 17 – Select PPM functionality and CDR format.

Prompt	Response	Description
REQ	CHG	Change.
TYPE	CFN	Configuration record.
PARM	YES	Parameters for Interface and transmission mode.
...		
FCDR	NEW	Use New CDR format (recommended for Japan).
...		
MTRO	PPM	Periodic Pulse Metering.

LD 15 – Use this overlay to allow Charge Display and CDR Charge.

Prompt	Response	Description
REQ	CHG	Change.
TYPE	CDB	Customer data block.
CUST	xx	Customer number.
...		
OPT	CHDA	Charge display allowed. Release 21 gate opener is FTR.
...		
- UCST	x	Unit cost, x = (0)-9999. Release 21 gate opener is PPM.

LD 16 – Allow AOC on the route.

Prompt	Response	Description
REQ	CHG	Change.
TYPE	RDB	Route data block.
...		
ISDN	YES	Configure ISDN data.
IFC	D70	ISDN Interface for Japan.
...		
CDR	YES	Call Detail Recording.
...		
OTL	YES	CDR on outgoing toll calls.
...		
OAN	YES	CDR on all answered outgoing calls.
...		
MR	ENDC	AOC End of Call allowed for international ISDN interfaces.
RUCS	1	Route unit cost value received is treated as charge.
RUCF	1 0	Route unit conversion factor, no conversion required.

LD 10/11– Assign meters to sets.

Prompt	Response	Description
REQ	NEW, CHG	New, or change.
TYPE	aaaa	Telephone type where aaaa = 500, 2500, SL1, 2006, 2008, 2009, 2016, 2018, 2112, 2216, 2317, 2616, or 3000.
TN	c u	Terminal Number.
...		
CLS	MRA	Message Registration Allowed.

LD 12 – Assign a Meter Recall key to the Attendant Console.

Prompt	Response	Description
REQ	NEW, CHG	New, or change.
TYPE	aaaa	Type of console, where aaaa = 1250 for M1250 console, 2250 for M2250 console, ATT for QCW3, QCW4 console, or PWR if the TN is used for power or Attendant Supervisory Module ASM.
TN	c u	Terminal Number.
...		
KEY	xx MTR	Assign a Meter recall key on the Attendant Console.

Numeris (end of call)**LD 17**– Change Configuration Record to allow PPM.

Prompt	Response	Description
REQ	CHG	Change.
TYPE	CFN	Configuration record.
PARM	YES	Parameters for Interface and transmission mode.
...		
MTRO	PPM	Periodic Pulse Metering.

LD 12 – Assign a Meter recall key on the Attendant Console.

Prompt	Response	Description
REQ	CHG	Change existing data.
TYPE	aaaa	Type of console, where aaaa = 1250 for M1250 console, 2250 for M2250 console, ATT for QCW3, QCW4 console, or PWR if the TN is used for power or Attendant Supervisory Module ASM.
TN	c u	Terminal number.
...		
KEY	xx MTR	key number, Meter key.

LD 15 – Configure the Customer Data Block for AOC.

Prompt	Response	Description
REQ	CHG	Change existing data.
TYPE	CDB FTR	Customer data block. Features and options. Release 21 gate opener.
CUST	xx	Customer number.
...		
ICI	xx MTR	ICI number, Meter Recall.

LD 15 – Configure Periodic Pulse Metering.

Prompt	Response	Description
REQ	CHG	Change.
TYPE	CDB PPM	Customer data block. Periodic Pulse Metering. Release 21 gate opener.
CUST	xx	Customer number.
HMTL	(YES) NO	Hotel/Motel environment.
PCDL	YES	PPM output on CDR Link. Additional three words added to tape record.
UCST	x	Unit cost for Periodic Pulse Metering, x=(0)-9999.
ATCH	(NO) YES	Attendant display of call charge.

LD 16 – Modify the Trunk Route for AOC.

Prompt	Response	Description
REQ	NEW, CHG	New, or change.
TYPE	aaaa	Type of data block.
CUST	xx	Customer number.
...		
OAL	YES	CDR on Outgoing calls.
- OTL	YES	CDR on Outgoing toll calls.
...		
CCO	YES	Printing CDR records for no PPM or AOC count.
...		
MR	ENDC	The AOC information is decoded at the end of the call.
...		
RUCF	x y	Route Unit Conversion Factor.

Numeris and SwissNet (during call)

LD 17 – Select PPM functionality and CDR format.

Prompt	Response	Description
REQ	CHG	Change.
TYPE	CFN	Configuration record.
PARM	YES	Parameters for Interface and transmission mode.
...		
MTRO	PPM	Periodic Pulse Metering.

LD 15 – Allow Charge Display and CDR Charge.

Prompt	Response	Description
REQ	CHG	Change.
TYPE	CDB PPM	Customer data block. Periodic Pulse Metering. Release 21 gate opener.
CUST	xx	Customer Number.
HMTL	(YES) NO	Hotel/Motel environment.
PCDL	YES	Allow PPM output on CDR Link
...		
- UCST	x	Unit cost, x = (0)-9999.
...		
- OPT	CHDA	Charge display allowed. Release 21 gate opener is FTR.

LD 16 – Allow AOC on the route.

Prompt	Response	Description
REQ	CHG	Change.
TYPE	RDB	Route data block.
CUST	xx	Customer number.
ROUT	xx	Route number.
...		
ISDN	YES	Configure ISDN data.
IFC	NUME SWIS	ISDN Interface for Numeris (France). ISDN Interface for Swis (Switzerland).
CDR	YES	Call Detail Recording.
OTL	YES	CDR on outgoing toll calls.
OAN	YES	CDR on all answered outgoing calls.
MR	DURC	AOC during a call.
CCO	TES	Disable the printing of CDR N records.
RUCS	1	Route unit cost value received is treated as charge.
RUCF	1 0	Route unit conversion factor, no conversion required.

LD 10/11 – Assign meters to sets.

Prompt	Response	Description
REQ	NEW, CHG	New, or change.
TYPE	aaaa	Telephone type where aaaa = 500, SL1, 2006, 2008, 2009, 2016, 2018, 2112, 2216, 2317, 2616, or 3000.
TN	c u	Terminal Number.
...		
CLS	MRA	Message Registration Allowed.

LD 12 – Assign a Meter Recall key to the Attendant Console.

Prompt	Response	Description
REQ	NEW, CHG	New, or change.
TYPE	aaaa	Type of console, where aaaa = 1250 for M1250 console, 2250 for M2250 console, ATT for QCW3, QCW4 console, or PWR if the TN is used for power or Attendant Supervisory Module ASM.
TN	c u	Terminal Number.
...		
KEY	xx MTR	Assign a Meter recall key on the attendant console.

1TR6 (end of call)

LD 12 – Assign a Meter recall key on the Attendant Console.

Prompt	Response	Description
REQ	CHG	Change existing data.
TYPE	ATT	QCW3 or QCW4 Attendant Console data block.
TN	c u	Terminal Number.
...		
KEY	x MTR x NULL	key number, Meter key. Remove a Meter key.

LD 15 – Configure the Customer Data Block for AOC.

Prompt	Response	Description
REQ	CHG	Change existing data.
TYPE	CDB NET	Customer data block. ISDN and ESN Networking options. Release 21 gate opener.
CUST	xx	Customer number.
...		
ISDN	YES	Integrated Services Digital Network allowed for customer.
...		
- PFX1	xxxx	Prefix 1.
- PFX2	xxxx	Prefix 2.

LD 15 – For nodes with 1TR6 connections, call charging and charge recording must be configured.

Prompt	Response	Description
REQ	CHG	Change existing data.
TYPE	CDB ATT	Customer data block. Attendant Console options. Release 21 gate opener.
CUST	xx	Customer number.
...		
ICI	x MTR	ICI number, Meter Recall.

LD 15 – For nodes with 1TR6 connections, call charging and charge recording must be configured.

Prompt	Response	Description
REQ	CHG	Change existing data.
TYPE	CDB PPM	Customer data block. Periodic Pulse Metering. Release 21 gate opener.
CUST	xx	Customer number.
HMTL	(YES) NO	Hotel/Motel environment.
PCDL	YES	PPM output on CDR Link. Additional three words added to tape record.
UCST	(0) - 9999	Unit Cost for Periodic Pulse Metering.
ATCH	(NO) YES	Attendant display of call Charge.
SCDL	(0) 1 2 3	Schedule for printing Message Registration and PPM data. No scheduled printing. Daily printout. Weekly printout. Monthly printout.
- WKDY	1-7	Week Day for weekly printout. Where 1 = Sunday.
- DAY	0-28	Day of month for printout.
- HOUR	hh or hh hh	Hour of day for printout.
- MCLR	(NO) YES	Meter Clear after printing.
- PTTY	(0)-15	PPM TTY number for printing meters, one per switch.

LD 17 – Change LD 17 to allow for AOC.

Prompt	Response	Description
REQ	CHG	Change.
TYPE	ADAN	All input/output devices.
ADAN	CHG DCH x	Action device and number, x = 0-15.
...		
ISDN	YES	Integrated Services Digital Network.
...		
- IFC	1TR6	1 TR6 for Germany.
- CNEG	(1) 2	Channel Negotiation option.
- LAPD	YES	Link Access Protocol for D-channel. Change LAPD parameters.
- - T203	2-(10)-40	Maximum Time allowed without frames being exchanged in seconds.

LD 16 – Change the Trunk Route for AOC.

Prompt	Response	Description
REQ	NEW, CHG	New, or change
TYPE	RDB	Route data block.
CUST	xx	Customer number
ROUT	xx	Route number.
...		
ISDN	YES	Integrated Services Digital Network
- IFC	1TR6	1TR6 for Germany

LD 16 – For nodes with 1TR6 connections configuring call-charge metering and printing

Prompt	Response	Description
REQ	NEW, CHG	New, or change.
TYPE	RDB	Route data block.
CUST	xx	Customer number.
ROUT	xx	Route number.
...		
CDR	YES	Call Detail Recording.
- OAL	YES	CDR on outgoing calls.
- - OTL	YES	CDR on Outgoing Toll calls.
CCO	(NO) YES	
RUCS	0-9999	Route Unit Cost.

Attendant and Network Wide Remote Call Forward

Set-based Configuration

LD 15 – Set the Station Control Password Length.

Prompt	Response	Description
REQ	CHG	Change.
TYPE	CDB FFC_DATA	Customer data block. Release 21 gate opener.
CUST	0-31	Customer number.
...		
-SCPL	0-8	Station Control Password Length (must be consistent network wide).
...		

LD 15 – Configure a Special Prefix Number (SPRE) for the customer.

Prompt	Response	Description
REQ	CHG	Change.
TYPE	CDB FTR_DATA	Customer data block. Release 21 gate opener.
CUST	0-31	Customer number.
...		
- SPRE	xxxx	Special Prefix Number.
...		

LD 10 – Set the Station Control Password and allow Call Forward.

Prompt	Response	Description
REQ	CHG	Change.
TYPE	500	Telephone type.
TN	c u	Card, and unit.
SCPW	xxxxxxx	Station Control Password (0 to 8 digits, defined in LD 15).
KEY	xx CFW 4-(16)-23	Assign Call Forward key (xx) and set the forwarding DN length.

LD 11 – Set the Station Control Password and allow Call Forward.

Prompt	Response	Description
REQ	CHG	Change.
TYPE	aaaa	Telephone type, where aaaa = SL1, 2006, 2008, 2009, 2016, 2018, 2112, 2216, 2317, 2616, or 3000.
TN	c u	Card, and unit.
SCPW	xxxxxxx	Station Control Password (0 to 8 digits, defined in LD 15).
KEY	xx CFW 4-(16)-23	Assign Call Forward key (xx) and set the forwarding DN length.

LD 57 – Define Remote Call Forward FFCs and set FFCT.

Prompt	Response	Description
REQ	CHG	Change.
TYPE	FFC	Flexible Feature Codes.
CUST	0-31	Customer number.
FFCT	(NO), YES	Confirmation tone is (is not) to be given after an FFC.
CODE	RCFA	Remote Call Forward Activate.
RCFA	xx	xx = RCFA code.
CODE	RCFD	Remote Call Forward Deactivate.
RCFD	xx	xx = RCFD code.
CODE	RCFV	Remote Call Forward Verify.
RCFV	xx	xx = RCFV code.

Attendant-based Configuration

LD 12 – A new Flexible Attendant feature key, RFW, has been added to this overlay. Configuration of the key on the Attendant Console is required to allow attendant access to the RCFW feature. Configuration of the RFW key is only allowed if the ARFW package is equipped.

Prompt	Response	Description
REQ	NEW,CHG	New, or change.
TYPE	ATT, 1250, 2250	Attendant type – the RFW key can be configured on QCW4, M1250, and M2250 Attendant Consoles.
TN	c u	Card, and unit.
...		
KEY	xx RFW	Key number assigned as Attendant Remote Call Forward key.

LD 15 – Configure the Attendant RCFW password.

Prompt	Response	Description
REQ	CHG	Change.
TYPE	CDB ATT_DATA	Customer data block. Release 21 gate opener.
CUST	0-31	Customer number.
...		
- IRFR	(NO), YES	Internal Remote Call Forward Password required.
-- IRFP	xxxxxxx	Internal RCFW Password (only prompted if the response to IRFR is YES). The password length is one to eight digits; the password is numeric only.
- XRFR	(NO), YES	External Remote Call Forward Password required.
-- XRFP	xxxxxxx	External RCFW password (only prompted if the response to XRFR is YES). The password length is one to eight digits; the password is numeric only.
...		

Attendant Blocking of DN

LD 12 – Configure the SACP key on the Attendant Console.

Prompt	Response	Description
REQ	CHG	Change.
TYPE	ATT, 1250, 2250	Attendant Console type.
TN	c u	Terminal Number for the Option 11.
...		
KEY	xx SACP	Semi-automatic Camp-on key.

LD 15 – Enable the Attendant Blocking of DN feature.

Prompt	Response	Description
REQ	CHG	Change.
TYPE	CDB ATT	Customer Data Block. Release 21 gate opener.
CUST	0-31	Customer number.
...		
ATT	YES	Change Attendant Console options.
...		
- SACP	(NO), ALL, SNGL	Semi-automatic Camp-on not allowed. Semi-automatic Camp-on for all Camp-on occurrences. Semi-automatic Camp-on an a per call basis.
- ABDN	(NO), YES	Activation of the Attendant Blocking of DN feature is (not) allowed. The ABDN prompt only appears when the SACP package is equipped.

Attendant Through Dialing Networkwide

LD 15 – Allow Attendant Through Dialing Networkwide.

Prompt	Response	Description
REQ:	CHG	Change existing data.
TYPE:	ATT	Attendant Console data block.
CUST	xx	Customer number.
OPT	(ATDA)	Attendant Through Dialing Allowed (default). ATDD = Attendant Through Dialing Denied.

Note: The configuration of Improved End-to-End Signaling in Overlay 15 and Calling Line Identification in Call Detail Recording Record are optional. Improved End-to-End Signaling sends the digits dialed by the calling party on the established link in a more efficient manner than End-to-End Signaling. A Call Detail Recording record on the outgoing trunk node shows the outgoing trunk in the ID field and the calling Directory Number in the CLID field if the outgoing trunk is on the attendant's node.

LD 15 – Configure Improved End-to-End Signaling.

Note: In Release 19 and later, Improved End-to-End Signaling is provided when EEST = YES and DTMF = NO.

Prompt	Response	Description
REQ:	CHG	Change existing data.
TYPE:	FTR	Customer Features and options.
CUST	xx	Customer number.
...		
EEST	YES	Send feedback tone to the originator of End-to-End Signaling.
- DTMF	NO	For Release 19 and later, use Improved End-to-End Signaling for single tone feedback.

LD 17 – Allow Calling Line Identification (CLID) field in Call Detail Recording (CDR) records.

Prompt	Response	Description
REQ	CHG	Change existing data.
TYPE	PARM	System parameters.
...		
- FCDR	(OLD) NEW	Format for Call Detail Recording OLD CDR format (default). NEW CDR format.
...		
- CLID	YES	Calling Line Identification in Call Detail Recording.

Australia ETSI

Primary Rate Configuration

LD 17 - Configure a PRI2 loop for the ETSI Australian ISDN connectivity.

Prompt	Response	Description
REQ	CHG	Change existing data.
TYPE	CEQU	Make changes to Common Equipment parameters.
...		
- PRI2	1-9	PRI2 loop number.
...		

LD 17 - Configure the D-channel for ETSI Australian ISDN connectivity.

Prompt	Response	Description
REQ	CHG	CHG = Change existing data.
TYPE	ADAN	Action device and number.
- ADAN	NEW DCH xx	Add a D-channel on logical port 0-15 (Option 11C).
- CTYP	MSDL	Downloadable D-Channel Daughterboard (Option 11C).
- CDNO	1-9	The card number of the Downloadable D-Channel Daughterboard.
- PORT	1	Port number for the Downloadable D-Channel Daughterboard.
...		
- USR	PRI	This D-channel is used for Primary Rate only.
- IFC	EURO	EuroISDN interface.
- - CNTY	EAUS	Australia ETSI.
- - PINX_CUST	0-99	The customer number to be used for the DN address translation associated with call independent connection messages received on the D-Channel.

...		
- - DCHL	1-9	PRI2 loop number for D-channel (Option 11C).
- CNEG	(1) 2	Options for outgoing Channel Negotiation. Option 1: Channel is non-negotiable. Option 2: The Channel listed is preferred, but negotiable.
- RLS	xx	Software Release of the far-end switch.
- RCAP		Remote capabilities, prompted to configure the Connected Line ID Presentation supplementary service. Multiple entries are allowed if separated by a space.
	(COLP) XCOL MCID XMCI	CLID Presentation supported. To remove COLP. Allow Malicious Call Trace Remove Malicious Call Trace.
- RCAP	aaaa	Remote capabilities is reprompted to enable the user to enter a <CR>, exiting from this prompt, or to change an existing remote capability value.
- OVLS	YES	Allow Overlap Sending.
- - OVLT	(0)-8	Duration of time, in seconds, that the sending side has to wait between INFO messages are sent. "0" means send immediately
- TIMR	YES	Change programmable timers. Only supported for interfaces supporting one of the following timers.
- - T310	(30)-100	Maximum time in seconds between an incoming CALL PROCEEDING message and the next incoming message.
- - INC_T306	0-(120)-240	Variable timer, in seconds, for received DISCONNECT message on incoming calls allowing in-band tone to be heard. The network will stop sending after this timer times out. The value is stored in two-second increments, which are rounded up.
- - OUT_T306	0-(120)-240	Variable timer, in seconds, for received DISCONNECT message on outgoing calls allowing in-band tone to be heard. The network will stop sending after this timer times out. The value is stored in two-second increments, which are rounded up.

- LAPD	(NO) YES	(Do not) allow the changing of the layer 2 timer.
...		

LD 16 - Configure the ETSI Australian ISDN PRI2 Route Data Block.

Prompt	Response	Description
REQ	NEW CHG	Add new data. Change existing data.
TYPE	RDB	Route Data Block.
CUST	xx	Customer number. xx = 0-99 (Options 51C-81/81C). xx = 0-31 (Option 11C).
ROUT	0-127	Route number (Option 11C)
TKTP	TIE COT DID	TIE trunk type. Central Office Trunk type. Direct Inward Dialing trunk type.
...		
DTRK	YES	Digital trunk route.
- DGTP	PRI2	2.0 MBit/s PRI digital trunk type.
...		
ISDN	YES	Integrated Services Digital Network.
- MODE	PRA	ISDN PRI route.
- IFC	EURO	EuroISDN interface.
- - CNTY	EAUS	Australia ETSI.
...		
ICOG	IAO ICT OGT	Incoming and/or Outgoing trunk. The trunk is Incoming and Outgoing. The trunk is Incoming only. The trunk is Outgoing only.
...		

ACOD	x...x	The Access Code for the trunk route. The Access Code must not conflict with the numbering plan.
...		
MCTS	YES	Enable MCT signaling.
...		
- MCTM	(0)-30	Malicious Call Trace disconnect delay timer (this timer overrides the T306 timer for calls originating or terminating on sets with MCT Class of Service).
- MTND	(NO) YES	(Do not) apply a Malicious Call Trace disconnect delay for tandem calls.

LD 14 - Configure the Australia ETSI ISDN PRI2 trunks.

Prompt	Response	Description
REQ	NEW CHG	Add new data. Change existing data.
TYPE	TIE COT DID	TIE trunk data block. Central Office Trunk data block. Direct Inward Dialing trunk data block. Note: Must match TKTP defined in LD 16.
TN	c u	Loop and channel for digital trunks, where: cu = card and unit (Option 11C).
CUST	xx	Customer number, as defined in LD 15.
RTMB	0-127 1-510	Route number and member number (Option 11C).
...		

LD 17 - Configure Advice of Charge for Australia ETSI.

Note: The MR package 101 must be equipped on the Meridian 1.

Prompt	Response	Description
REQ	CHG	Change existing data.
TYPE	PARM	Change system parameters,
...		
OCAC	(NO) YES	(Do not) support the Original Carrier Access Code format.
MTRO	PPM	Use Periodic Pulse Metering as the metering option. The default is MR, for Message Registration.
...		

LD 15 - Allow Charge Display and CDR Charge.

Prompt	Response	Description
REQ:	CHG	Change existing data.
TYPE:	PPM	Periodic Pulse Data.
CUST	xx	Customer number defined for this feature. xx = 0-99 for Options 51C-81/81C. xx = 0-31 for Option 11C.
OPT	CHDA	Charge Display Allowed.
...		
UCST	(0)-9999	Unit cost for PPM.
...		

LD 10 - Assign meters to analog (500/2500) sets.

Prompt	Response	Description
REQ	NEW CHG	Add new data. Change existing data.
TYPE	500	Analog set.
TN		TN address
	cu	c u = card and unit (Option 11C).
DES	x...x	ODAS Station Designator.
CUST	xx	Customer number, as defined in LD 15.
...		
DN	x...x yyy	Directory Number (x...x) and CLID entry (yyy).
...		
TGAR	0 - (1) - 30	Trunk Group Access Restriction The default of 1 automatically blocks direct access.
...		
CLS	MRA	Message registration Allowed.
...		

LD 11 - Assign meters to Meridian proprietary sets.

Prompt	Response	Description
REQ	NEW CHG	Add new data. Change existing data.
TYPE	aaa	Telephone set type.
TN		TN address
	cu	c u = card and unit (Option 11C).
DES	x...x	ODAS Station Designator.
CUST	xx	Customer number, as defined in LD 15.
...		
TGAR	0 - (1) - 30	Trunk Group Access Restriction The default of 1 automatically blocks direct access.
...		
CLS	MRA	Message registration Allowed.
...		
KEY	xx aaa yyy	Telephone function key assignments.
...		

LD 16 - Allow Advice of Charge on the route configured for Australia ETSI.

Prompt	Response	Description
REQ	NEW CHG	Add new data. Change existing data.
TYPE	RDB	Route Data Block.
CUST	xx	Customer number, as defined in LD 15. xx = 0-31 (Option 11C).
ROUT	0-127	Route number, as previously configured (Option 11C).
TKTP	TIE COT DID	TIE trunk type. Central Office Trunk type. Direct Inward Dialing trunk type.
...		
DTRK	YES	Digital trunk route.
- DGTP	PRI2	2.0 MBit/s PRI digital trunk type.
...		
ISDN	YES	Integrated Services Digital Network.
- MODE	PRA	ISDN PRI route.
- IFC	EURO	EuroISDN interface.
- - CNTY	EAUS	Australia ETSI.
...		
CDR	YES	Include AOC information in the CDR ticket.
...		
- OAL	YES	CDR on all answered outgoing calls.
- - OTL	YES	CDR on all outgoing toll calls.
...		

MR	STAC DURC ENDC	Define AOC at call set-up. Define AOC during the call. Define AOC at end of call.
DSPD	(NO) YES	(Do not) display the charge during the call.
...		
RUCS	0-9999	Route unit cost.
RURC	X Y	Route unit reference cost. Formula is $X \cdot 10^{(-Y)}$ where X = 0-9999, Y = 0-3. The default value for X is the value that is entered for RUCS.
RUCF	1 0	Route unit conversion factor. 0 = No conversion is required.
DSPT	0-(10)-60	Charge display timer.
...		

ISDN BRI configuration

LD 27 - Define a Link Access Procedure on the D-channel (LAPD) protocol group. The protocol configuration procedures define the protocols used by ISDN BRI DSLs to communicate over ISDN. These protocol groups support various ISDN communication standards used in Europe, and other continents and countries.

Prompt	Response	Comment
REQ	NEW	Add an ISDN protocol group
TYPE	LAPD	LAPD Protocol group
PGPN	0-15 <CR>	Protocol group number <CR> =Stops this prompt from being displayed again
LAPD	(NO) YES	LAPD parameters. (NO) = Does not prompt the LAPD parameters and assigns the default values shown in () to these parameters. YES = Define or modify the LAPD parameters.
USER	(NO) YES	(Do not) print groups selected at PGN prompt.
- T200	(2)-40	Retransmission timer specifies the time delay before the system retransmits the information. Delay is in increments of 0.5 seconds.
- T203	4-(20)-80	Maximum time between transmission frames Delay is in increments of 0.5 seconds.
- N200	1-(3)-8	Maximum number of retransmissions of unsuccessfully transmitted information.
- N201	4-(260)	Maximum number of contiguous octets or bytes of information.
- K	(1)-32	Maximum number of outstanding negative acknowledgment (NAKs) allowed before alarming the system.
PGPN	<CR>	Press <CR> to prevent repetition of all the parameters starting with LAPD.

LD 16 - Configure Route Data Block parameters for the ISDN BRI Trunk access capability.

Prompt	Response	Comment
REQ	NEW	Add new data (ISDN BRI protocol group settings).
TYPE	RDB	Route Data Block.
CUST	xx	Customer number, as defined in LD 15. xx = 0-99 (Options 51C-81/81C). xx = 0-31 (Option 11C).
ROUT	0-511	Route number, as previously configured (Options 51C-81/81C).
	0-127	Route number, as previously configured (Option 11C).
TKTP	TIE	TIE trunk type.
	COT	COT Central Office Trunk trunk type.
	DID	Direct Inward Dialing trunk type.
...		
DTRK	YES	BRI Digital Trunk Route
BRIP	NO	ISDN BRI packet handler route (NO is entered, since packet data is not required).
- DGTP	BRI	Digital trunk type.
ISDN	YES	Integrated Services Digital Network.
...		
- IFC	EURO	EuroISDN interface.
- - CNTY	EAUS	Australia ETSI.
- CNEG	(NO) YES	(Do not) allow Channel Negotiation.
...		
OVLS	(NO) YES	(Do not) allow Overlap Sending.
- OVLT	(0)-8	Overlap Timer in seconds. This timer controls the interval between the sending of INFORMATION messages. "0", the default, means send immediately.

Prompt	Response	Comment
- PGPN	0-15	Protocol Group Number, as defined in LD 27.
- RCAP		Remote capabilities, prompted to configure the Connected Line ID Presentation supplementary service. Multiple entries are allowed if separated by a space.
	(COLP)	CLID Presentation supported.
	XCOL	To remove COLP.
	MCID	Allow Malicious Call Trace
	XMCI	Remove Malicious Call Trace.
- RCAP	aaaa	Remote capabilities is reprompted to enable the user to enter a <CR>, exiting from this prompt, or to change an existing remote capability value.
...		
ISDN	YES	ISDN.
...		
- TIMR	YES	Change programmable timers. Only supported for interfaces supporting one of the prompted timers.
- - INC_T306	0-(120)-240	Variable timer, in seconds, for received DISCONNECT message on incoming calls allowing in-band tone to be heard. The network will stop sending after this timer times out. The value is stored in two-second increments, which are rounded up.
- - OUT_T306	0-(120)-240	Variable timer, in seconds, for received DISCONNECT message on outgoing calls allowing in-band tone to be heard. The network will stop sending after this timer times out. The value is stored in two-second increments, which are rounded up.
...		
MCTS	YES	Enable MCT signaling.
...		
- MCTM	(0)-30	Malicious Call Trace disconnect delay timer (this timer overrides the T306 timer for calls originating or terminating on sets with MCT Class of Service).

Prompt	Response	Comment
- MTND	(NO) YES	(Do not) apply a Malicious Call Trace disconnect delay for tandem calls.
...		

LD 27 - Configure for a Multi-purpose ISDN Signaling Processor (MISP) for an ISDN BRI trunk.

Prompt	Response	Comment
REQ	NEW CHG	Add new data. Change existing data.
TYPE	MISP	Multi-purpose ISDN Signaling Processor.
LOOP	0-158 1-9	MISP loop number (Options 51C-81/81C). MISP loop number (Option 11C).
APPL	BRIE	Application type. Enter BRIE for Australia ETSI.
APPL	<CR>	To end configuration procedure.

LD 27 - Configure an S/T Interface (SILC) or U-Interface (UILC) line card, for an ISDN BRI trunk.

Prompt	Response	Comment
REQ	NEW CHG	Add new data Change existing data.
TYPE	CARD	SILC or UILC configuration.
TN	lll s cc	Card location (Options 51C-81/81C). lll (superloop) = 0-156 (must be an even number, divisible by 4) s (shelf) = 0-1 cc (card) = 0-15
	c	Card location (Option 11C). c (card) = 1-9
MISP	0-158 1-9	MISP loop number (Options 51C-81/81C). MISP loop number (Option 11C). Must be an even loop number that has already been configured.
CTYP	SILC UILC	SILC line card is to be added or changed. UILC line card is to be added or changed. Note: Remove any DSLs configured for this line card before changing the card type.

LD 27 - Configure a Digital Subscriber Loop (DSL) for an ISDN BRI trunk.

Prompt	Response	Comment
REQ	NEW CHG	Add new data Change existing data.
TYPE	DSL	Digital Subscriber Loop data block.
DSL	lll s cc dsl#	DSL Terminal Number location (Options 51C-81/81C). lll (superloop) = 0-156 (must be zero or a number divisible by 4) s (shelf) = 0-1 cc (card) = 0-15 dsl# (DSL location) = 0-7
	cc sdl	DSL Terminal Number location (Option 11C). cc (card) = 1-20 dsl# (DSL number) = 0-7
APPL	BRIE	BRI trunk application for Australia ETSI.
CUST	xx	Customer number, as defined in LD 15. xx = 0-99 (Options 51C-81/81C). xx = 0-31 (Option 11C).
CTYP	SILC	Interface card type is SILC.
MISP	0-158 0-9	MISP loop number for Options 51C-81/81C MISP loop number for Option 11C.
MODE	TE	Enter TE (user side) as the mode for Australia ETSI.
- MTFM	(NO) YES	BRI multiframe option.
TKTP	TIE COT DID	TIE trunk type. Central Office Trunk type. Direct Inward Dialing trunk type. Must be the same entry as defined in LD 16.
CLOK	(NO) YES	(Do not) use the DSL as the clock source.
PDCA	(1)-16	Pad table number.

Prompt	Response	Comment
ROUT	0-511 0-127	Route number (both B-Channels must belong to the same route) (Options 51C-81/81C). Route number (both B-Channels must belong to the same route) (Option 11C).
TIMR	(NO) YES	(Do not) change timer values.
- - T310	(30)-100	Maximum time in seconds between an incoming CALL PROCEEDING message and the next incoming message.
B1	(NO) YES	(Do not) change the configuration parameters for B-Channel 1.
- MEMB	1-510	Route member number.
...		
B2	(NO) YES	(Do not) change the configuration parameters for B-Channel 2.
- MEMB	1-510	Route member number, for Options 51C-81/81C.
...		

LD 27 - Assign meters to a DSL (this step is required for Advice of Charge).
The MR/PPM package 101 must be equipped on the Meridian 1.

Prompt	Response	Comment
REQ	NEW	Add an ISDN protocol group
TYPE	DSL	LAPD Protocol group
DSL	Ill s cc dsl#	DSL Terminal Number location (Options 51C-81/81C). Ill (superloop) = 0-156 (must be zero or a number divisible by 4) s (shelf) = 0-1 cc (card) = 0-15 dsl# (DSL location) = 0-7
	cc sdl	DSL Terminal Number location (Option 11C). cc (card) = 1-20 dsl# (DSL number) = 0-7
...		
CLS	MRA	Allow Message Registration on the DSL.

LD 16 - Allow Advice of Charge on the route configured for the Australia ETSI.

Prompt	Response	Comment
REQ	NEW	Add new data (ISDN BRI protocol group settings).
TYPE	RDB	Route Data Block.
CUST	xx	Customer number, as defined in LD 15. xx = 0-99 (Options 51C-81/81C). xx = 0-31 (Option 11C).
ROUT	0-511 0-127	Route number, as previously configured (Options 51C-81/81C). Route number, as previously configured (Option 11C).
TKTP	TIE COT DID	TIE trunk type. COT Central Office Trunk trunk type. Direct Inward Dialing trunk type.
...		
DTRK	YES	BRI Digital Trunk Route
BRIP	NO	ISDN BRI packet handler route (NO is entered, since packet data is not required).
- DGTP	BRI	Digital trunk type.
ISDN	YES	Integrated Services Digital Network.
...		
- IFC	EURO	EuroISDN interface.
- - CNTY	EAUS	Australia ETSI.
...		
CDR	YES	Include AOC information in the CDR ticket.
...		
OAL	YES	CDR on all answered outgoing calls.
- OTL	YES	CDR on all outgoing toll calls.

Prompt	Response	Comment
...		
MR	STAC DURC ENDC	Define AOC at call set-up. Define AOC during the call. Define AOC at end of call.
DSPD	(NO) YES	(Do not) display the charge during the call.
...		
RUCS	0-9999	Route unit cost.
RURC	X Y	Route unit reference cost. Formula is $X \cdot 10^{(-Y)}$ where X = 0-9999, Y = 0-3. The default value for X is the value that is entered for RUCS.
RUCF	1 0	Route unit conversion factor. 0 = No conversion is required.
DSPT	0-(10)-60	Charge display timer.
...		

Automatic Redial

LD 13 – Define Tone Detector Units.

Prompt	Response	Description
REQ	NEW	Add.
TYPE	TDET	Tone Detector data block.
TN	c u	Terminal Number.

LD 15 – Define Automatic Redial.

Prompt	Response	Description
REQ:	CHG	Change.
TYPE:	FTR	Change features and options.
CUST	xx	Customer number.
...		
-ARDL_ATEMPT	1-(30)-60	Number of Automatic Redial attempts.
REQ:	CHG	Change.
TYPE:	TIM	Change Timers.
CUST	xx	Customer number.
...		
- ARDL_ACCEPT	0-(20)-60	Automatic Redial Acceptance Timer in seconds. Odd number entries are rounded up to the next even number and echoed back with a message.
- ARDL_RETRY	10-(30)-60	Automatic Redial Retry Timer in seconds. Odd number entries are rounded up to the next even number and echoed back with a message.

LD 16 – Define Automatic Redial Tone Detector Response Timer.

Prompt	Response	Description
REQ	NEW, CHG	Add, or Change.
TYPE	RDB	Route Data Block.
CUST	xx	Customer number.
ROUT	0-511	Route number.
...		
CNTL	YES	Changes to controls or timers.
TIMR	RTD 0-(12)-60	Tone Detector Response Timer in seconds. Odd number entries are rounded up to the next even number.

LD 87– Define Automatic Redial Network Route Selection.

Prompt	Response	Description
REQ	NEW, CHG	Add, or Change.
CUST	xx	Customer number.
FEAT	NCTL	Network Control Feature.
...		
NCOS	(0) - 99	Network Class of Service group number.
- ARDL	(A) I	A = Automatic Redial network route selection allowed from all route sets (initial and extended). I = Automatic Redial network route selection allowed from initial set of routes only.

LD 11 – Assign Automatic Redial Class of Service and Key.

Prompt	Response	Description
REQ:	NEW, CHG	Add, or Change.
TYPE:	xxxx	Telephone type, where: xxxx = SL1, 2006, 2008, 2016, 2216, or 2616.
...		
CLS	RDLA	Automatic Redial allowed (default). RDLD = Automatic Redial denied.
KEY	xx RGA	Ring Again key assignment.
KEY	xx RGA	Ring Again key assignment for Multi-Automatic Redial capability.

Backup D-channel

Basic PRI or ISL administration must be performed before the backup D-channel is defined. Changes in the following service-change program are required to configure the Backup D-channel feature.

Overlay	Action
LD 17 Configuration Record	Configure the backup D-channel and active the recovery to primary D-channel feature.

Use LD17 and perform one of the following procedures depending on your software version.

Prior to X11 release 18

Follow this procedure and use LD 17 to configure the backup D-channel for prior to X11 release 18 software.

Prompt	Response	Description
REQ	CHG	
TYPE	CFN	Configuration Data Block
ISDN	YES	Change the ISDN parameters
.		
BCHI	1-15	backup DCHI port number
CDNO	1 - 9	Card number of the PRI loop which will support the Backup D-channel
PORT	1	Must be set to 1 for the NTAK79
.		
BCHL	1 - 9	2 Mb PRI loop number for the BCHI. This will be the same as the response to the CDNO prompt.
USR		

Prompt	Response	Description
RCVP	YES, (NO)	See Table 67 before continuing. Recovery to Primary D-channel option. When RCVP is YES, the primary D-channel becomes the active D-channel. Defining this prompt has to be coordinated with the far end. Both sides either have to be YES or NO. If the two sides do not match, both sides default to NO.

X11 release 18 and later

Follow this procedure and use LD17 to configure the backup D-channel for X11 release 18 and later software.

Prompt	Response	Description
REQ	CHG	Change
TYPE	CFN	Configuration Record
ADAN	NEW BDCH 0-63	Add a backup D-channel (also CHG, MOV, and OUT BDCH)
PDCH	1-15	Primary D-channel
CTYP	DCHI, MSDL	Card type
DNUM	0-15	Device number: physical port (odd) for D-channel on DCH, physical card address for MSDL
_PORT	1	Port number on MSDL card
USR	PRI, ISLD, SHA	D-channel mode
_ISLM	1-382	Number of ISL trunks controlled by the D-Channel
DCHL	0, 2, 4,...9	PRI loop number for DCHI
OTBF	1-(32)-127	Number of output request buffers
_BPS	xxxx	Baud rate for ISL D-channel on MSDL port (default 64000)
_PARM	R232,(R422) DCE, (DTE)	ISL D-channel interface and transmission mode (MSDL port only)

Prompt	Response	Description
BCHL	0,2,4,...9	PRI loop number for back-up D-channel
RCVP	Yes, (No)	Auto-recovery to primary D-channel option
ADAN	<cr>, ****	Go on to next prompt or exit overlay

Table 67**Recovery to primary D-channel RCVP prompt responses**

RCVP = YES	RCVP = NO
primary D-channel up—active backup D-channel up	primary D-channel up—active backup D-channel up
primary D-channel down backup D-channel up—active (see below)	primary D-channel down backup D-channel up—active
primary D-channel up—active backup D-channel up	primary D-channel up backup D-channel up—active

When RCVP is YES, the primary D-channel is down, and the backup D-channel is up, and the following occurs. First the switch tries to re-establish the primary D-channel connection. If this cannot be done successfully, then the backup D-channel is switched in. When the primary D-channel is brought up again, the primary D-channel becomes the active D-channel.

If RCVP is NO and the primary D-channel is down, the backup D-channel remains active when the primary D-channel is brought up. It is important to note that the backup D-channel remains the active D-channel.

Bearer Capability in Call Detail Recording

Follow these steps to configure Bearer Capability in CDR:

- Configure the new format for CDR records in LD 17.
- Enable printing of the Bearer Capability in CDR information in LD 15.

LD 17 – Configure the CDR records.

Prompt	Response	Description
REQ	CHG	Change existing data.
TYPE	PARM	Add system parameters data.
...		
FCDR	NEW	Enable new format for CDR record.

LD 15 – Enable printing of the Bearer Capability information in the CDR ticket.

Prompt	Response	Description
REQ:	CHG	Change existing data.
TYPE:	CDR	CDR and Charge Account options.
CUST	xx	Customer number defined for this feature.
CDR	YES	Allow CDR records for the customer.
...		
PORT	0-15	Serial Data Interface Port Monitor.
CNI	a..a	Calling Number Identification. The response determines which field of the CDR record will contain the Calling Number Identification for this customer. • (DGTS) - CNI is recorded in the digits field. • CLID - CNI of R2MFC is recorded in CLID field. • NONE - CNI is not recorded in CDR.
- BCAP	YES	Enable Bearer Capability in CDR ticket.

Call Forward, Break-In and Hunt Internal or External Network Wide

LD 15 – Define the Network Wide definition of calls as internal.

Prompt	Response	Description
REQ:	CHG	Change.
TYPE:	FTR	Customer Features and options.
CUST	xx	Customer number.
- IDEF	YES	Calls will be treated as internal for Network-wide internal for Call Forward/Hunt by Call Type, Internal Call Forward and Break-In Indication Prevention.

LD 16 – Modify Route Data Block to Define Internal/External Calls.

Prompt	Response	Description
REQ	CHG	Change.
TYPE	RDB	Route data block.
CUST	xx	Customer number.
ROUT	0-511	Route number.
...		
RCLS	(EXT) INT	Route Class marked as External or Internal.
IDEF	(NET) LOC	Internal/External Definition. If NET is entered, any call over the selected route will receive network treatment according to available network information. If LOC is entered, the route class of the selected route will supercede any other information. A call over this route will receive internal treatment if the route class is set to internal, otherwise it will receive external.
Note: The prompt IDEF will output only if IDEF = YES in the Customer Data Block. If IDEF = NO, any information that was entered previously at the IDEF prompt will not influence the treatment received by a call.		

Call Page Network Wide

LD 16 – Configuring Paging Route.

Prompt	Response	Description
REQ	NEW	New.
TYPE	RDB	Route data block.
CUST	xx	Customer Number associated with route.
...		
TKTP	PAG	Paging Route.
NACC	(PGNR) PGNC PGNU	Call Page Network Wide Restricted (default). Call Page Network Wide controlled (ISDN only) Call Page Network Wide uncontrolled (PGNU is equivalent to ISDN/Analog media.)
ICOG	OGT	Outgoing trunk.
TARG	1-15	Trunk Access Restriction Group.

LD 16 – Configuring BRI Trunk Route.

Prompt	Response	Description
REQ	NEW	New.
TYPE	RDB	Route data block.
CUST	xx	Customer number.
...		
DTRK	YES	Digital Trunk route.
DGTP	BRI	Digital Trunk type.
RCAP	NAC	Remote capability where: NAC = Class of Service data. XNAC = removes Class of Service as a remote capability (default).

LD 14 – Paging Trunk Configuration.

Prompt	Response	Description
REQ	NEW	New.
TYPE	PAG	Trunk type.
TN	c u	Terminal number.
RTMB	0 - 511 1- 254	Route number, Member number.

LD 10 – Assign Class of Service to Analog (500/2500 type) telephones.

Prompt	Response	Description
REQ:	NEW, CHG	New, or change.
TYPE:	500	Type telephone.
...		
CLS	PGNA	Call Page Network Wide Allowed. (PGND) = Call Page Network Wide Denied (default).

LD 11 – Assign Class of Service to Meridian 1 Proprietary Telephones.

Prompt	Response	Description
REQ:	NEW, CHG	New, or change.
TYPE:	xxxx	Telephone type, where xxxx= SL1, 2006, 2008, 2009, 2016, 2018, 2112, 2216, 2317, 2616, or 3000.
...		
CLS	PGNA	Call Page Network Wide Allowed. (PGND) = Call Page Network Wide Denied (default).

LD 27 – Assign Class of Service to ISDN BRI sets.

Prompt	Response	Description
REQ	NEW, CHG	New, or change.
TYPE	DSL	Type of data block.
DSL	l s c dsl	Digital Subscriber Loop address.
CLS	PGNA	Call Page Network Wide Allowed. (PGND) = Call Page Network Wide Denied (default).

LD 17 – D-channel Message Configuration.

Prompt	Response	Description
REQ	CHG	Change.
TYPE	ADAN	Input or Output Devices.
ADAN	CHG DCH x	Change D-channel. x = 0-63.
- CTYP	MSDL	Downloadable D-Channel Daughterboard.
RCAP	NAC	Remote capability where: NAC = Class of Service data. XNAC = removes Class of Service as a remote capability (default).

Call Park Network Wide

LD 15 – Enable Call Park Network Wide.

Prompt	Response	Description
REQ:	CHG	Change.
TYPE:	FTR	Features and options. Release 21 gate opener.
CUST	xx	Customer number.
- OPT	CPN	Enable Call Park Network Wide. CPA = Enables Call Park. CPD = Disables Call Park (default).

LD 50 – Add/Change Customer Call Park Data.

Prompt	Response	Description
REQ	NEW, CHG, PRT	New, or change, or print.
TYPE	CPK	Call Park data block.
CUST	xx	Customer number.
BLOC	1-5	Call Park data block number. Primary Call Park (block 1) must be defined for Call Park operation. Block 1 must be initially defined before attempting to remove.
CPTM	30-(45)-480	Call Park Timer (in seconds).
RECA	(NO) YES	Call Park Recall to Attendant.
SPDN	(0)-100 xx...x	Number of contiguous system park DNs and first DN of that range.
MURT	0-511	Music Route number for parked call.

LD 10 – Enable Call Park for Analog (500/2500 type) telephones.

Prompt	Response	Description
REQ:	NEW, CHG	New, or change.
TYPE:	500	Telephone type.
TN	c u	Terminal Number.
CLS	XFA	Call Transfer Allowed. XFD = Call Transfer Denied.

LD 11 – Add/Change Call Park Key on Meridian 1 proprietary telephones.

Prompt	Response	Description
REQ:	NEW, CHG	New, or change.
TYPE:	xxxx	Telephone type where: aaaa = SL1, 2006, 2008, 2009, 2016, 2018, 2112, 2216, 2317, 2616 or 3000.
TN	c u	Terminal Number.
KEY	xx PRK xx TRN xx AO3 xxAO6	Key assignment for Call Park. Key number for Transfer. Three Party Conference. Six Party Conference.

LD 12 – Add/Change Call Park Key on Attendant Consoles.

Prompt	Response	Description
REQ	NEW, CHG	New, or Change.
TYPE	xxxx	Console type.
TN	c u	Terminal Number.
KEY	xx PRK	Key number, Call Park.

Calling Line Identification

Basic PRI or ISL administration must be performed before Calling Line Identification (CLID) is defined (see PRI Configuration and ISL data administration). Changes in the following service-change programs are required to configure the CLID feature.

CAUTION

Third-party software is not guaranteed if this feature is activated. If the third-party software does not recognize the CLID field, it may shut down

Step 1: Respond to the following prompt in Overlay 17.

Prompt	Response	Description
•		
•		
MANU		
CLID	YES,(NO)	enable (do not enable) CLID in CDR Note: Package 118, CLID in CDR, is required.

Step 2: Respond to the following prompt in Overlay 15.

Prompt	Response	Description
•		
ISDN		
•		
•		
CNTP	(PDN) LDN	calling number type—determines the class of service associated with CLID for all sets, except those configured differently in Overlays 10 and 11. Note: Attendant consoles have only a Listed Directory Number (LDN). —the CLID feature displays the set's Prime Directory Number —the CLID feature displays the customer's Listed Directory Number

Step 3: Respond to these prompts in Overlays 10 and 11.

Prompt	Response	Description
REQ	NEW, CHG, MOV, OUT	
TYPE	set type	as appropriate
TN	c u	terminal number address —card (1-10) —unit (0-7)
XLST		
CLS	(PDN) LDN	—the set's Prime Directory Number —the customer's Listed Directory Number

Call Charging under 1 TR 6 Connections

In overlay 12:

- For nodes with 1 TR 6 connections, respond to the TYPE prompt with ATT. Respond to the KEY prompt with 0-9 MTR for each attendant console to have a Meter key. (Respond with 0-9 NULL to remove a Meter key.)

In overlay 15:

- Respond to the ISDN prompt with YES. This gives rise to the PFX1 and PFX2 prompts. Respond to these with Prefix 1 and Prefix 2 respectively (each a maximum of four digits). This information is sent out as part of the Call Setup message.

For nodes with 1 TR 6 connections, call charging and charge recording must be configured in overlay 15:

- Respond to the ICI prompt with *xx*MTR to assign a meter recall key on an ICI position at attendant consoles.
- Respond to the PPMD prompt with YES to enter periodic pulse metering (PPM) data, or NO to continue without being prompted for PPM data. If you are entering PPM data, respond to the HMTL prompt with YES to select the Hotel/Motel option, or NO to suppress this option. Respond to the PCDL prompt with YES to generate call charging information in the CDR Link tape records, or NO to suppress this option. Define the call charge unit cost by responding to the UCST prompt with a value from 1 to 9999. (Entering a value of 0 disables this option.) Respond to the ATCH prompt with YES to display call charge information on the attendant console, or NO to prevent display.

- If PPMD was enabled, select schedules for the output of Message Registration and Call Charging information. Respond to the SCDL prompt with 0 to prevent scheduled printing, 1 to get a daily printout, 2 to get a weekly printout or 3 to get a monthly printout. If you did not enter 0, the WKDY prompt appears. Respond to this with a number from 1 to 7; this indicates the day of the week (1 being Sunday) on which weekly printouts are generated. Respond to the DAY prompt with a number from 1 to 28; this indicates the day of the month on which monthly printouts are generated. Respond to the HOUR prompt with a number from 0 to 23; this indicates the hour of the day during which daily printouts are generated. (If you enter a space and then a second hour, printouts are generated twice daily, at the specified times.) Once the schedule has been completed by responding to these prompts, the MCLR prompt appears. Respond with YES to have meters cleared after printing, or NO to leave the meters uncleared. Respond to the PTTY prompt with a number from 0 to 15. This is the port number for automatic printing.

In overlay 17:

- Respond to the ISDN prompt with YES.
- Respond to the IFC prompt with either
AXEA (to configure for an AXE-10 for Australia)
AXES (to configure for an AXE-10 for Sweden)
SS12 (to configure for a SYS-12)
SWIS (to configure for SwissNet), or
1TR6 (to configure for a 1 TR 6 channel protocol connection).
- Respond to the CNEG (outgoing calls only) prompt with either 1, to deny channel negotiation, or 2 to allow it.
- Respond to the LAPD prompt with YES.
- Respond to the T203 prompt with a value between 2 and 40. (This is the maximum time, in seconds, to be allowed without the exchange of frames.)

In overlay 16:

- Respond to the ISDN prompt with YES.
- Respond to the IFC prompt with either
AXEA (to configure for an AXE-10 for Australia)
AXES (to configure for an AXE-10 for Sweden)
SS12 (to configure for a SYS-12)
SWIS (to configure for SwissNet), or
1TR6 (to configure for a 1 TR 6 channel protocol connection).

For nodes with 1 TR 6 connections, call-charge metering and printing must be configured in overlay 16:

- Respond to the CDR prompt with YES.
- Respond to the OAL prompt with YES.
- Respond to the OTL prompt with YES.
- Respond to the CCO prompt with YES to prevent the printing of CDR records containing no call charging information, or NO to allow this printing.
- Respond to the RUCS prompt with a number between 0 and 9999, to define the call charge unit cost on a route basis.

Call Connection Restriction

To configure the operation of the Call Connection Restriction product improvement, enter responses to the following prompts in overlay 15.

RCNT: This limits the number of call redirections to be allowed in a call connection. Respond to this prompt with the limit, from 0 to 5.
(The default is 1.)

PSTN: This limits to one the number of PSTNs allowed in a call connection. Respond to this prompt with YES to place the limit; respond with NO to allow an unlimited number of PSTNs in a call connection.

TNDM: This limits the number of tandem connections allowed in a call connection. Respond to this prompt with the limit, from 0 to 31
(the default value is 15).

PCMC: This limits the number of μ /A-Law conversions to be allowed in a call connection. Respond to this prompt with the limit, from 0 to 31.
(the default value is 15).

SATD: This limits the number of Satellite delays to be allowed in a call connection. Respond to this prompt with the limit, from 0 to 5.
(the default value is 1).

The above prompts only appear if the ISDN prompt in overlay 15 is set to YES. For more information on the prompts presented in overlay 15, refer to the Option 11 software guides.

Call Forward/Hunt Override via Flexible Feature Code

LD 57 – Define FFC for Call Forward/Hunt Override.

Prompt	Response	Description
REQ	NEW, CHG	New, or change.
TYPE	FFC	Flexible Feature Code.
...		
CODE	CFHO	Call Forward/Hunt Override Via FFC.
CFHO	nnnn	Call Forward/Hunt FFC.
...		

LD 10 – Set class of service for the Forward/Hunt Override Via FFC feature.

Prompt	Response	Description
REQ	NEW, CHG	New, or change.
TYPE	500	Type of telephone set.
...		
CLS	(CFHD), CFHA	Call Forward/Hunt Override Via FFC is (denied) or allowed.
...		

LD 11 – Set class of service for the Forward/Hunt Override Via FFC feature for Meridian 1 propriety sets.

Prompt	Response	Description
REQ	NEW, CHG	New, or change.
TYPE	xxxx	Type of telephone set.
...		
CLS	(CFHD), CFHA	Call Forward/Hunt Override Via FFC is (denied) or allowed.
...		

LD 18 – Configure ABCD key for the Forward/Hunt Override Via FFC feature.

Prompt	Response	Description
REQ	NEW, CHG	New, or change.
TYPE	ABCD	Modifying 16-button DTMF.
...		
PRED	YES	Function table for pre-dial.
A	CFHO*FFC*	CFHO is assigned to key A.
B	CFHO*FFC*	CFHO is assigned to key B.
C	CFHO*FFC*	CFHO is assigned to key C.
D	CFHO*FFC*	CFHO is assigned to key D.

Call Pickup Network Wide

Use the following procedure to configure Call Pickup Network Wide.

Note: minimum software vintage of Generic X11 Release 21 in order for Call Pickup Network Wide to be functional. The software Release ID is defined in LD 17.

LD 10 – Assign a Ringing Pickup Group number to an analog (500/2500 type) telephone, and define with the Class of Service which of the Call Pickup features are allowed to be activated from this station.

Prompt	Response	Description
REQ	CHG	Change.
TYPE	500	Telephone type.
TN	c u	Terminal Number for the Option 11.
...		
RNPG	(0)-4095	Ring Number Pickup Group (RNPG). If the RNPG is set to 0 (the default) on a station, it is not possible to pick up any call ringing this station from any other set. Enter 0 to remove a station from the RNPG. Note: If the RPNG is two or more digits in length, it must be configured in the Customer Data Block (LD 15), so that it may use the Group Pickup key. See the procedure which follows LD 11.
...		
CLS	(PUD) PUA	Ring Number Pickup (denied) allowed.
	(DPUD) DPUA	DN pickup (denied) allowed.
	(GPUD) GPU A	Group pickup (denied) allowed.

LD 11 – Assign a pickup group to a Meridian 1 proprietary telephone. Define with the Class of Service which of the Call Pickup features are allowed to be activated from this station.

The configuring of different pickup keys is optional, since the Call Pickup and Directed Call Pickup features can be activated by dialing the SPRE + xx or by dialing a Flexible Feature Code. The Digit Display key is needed for the Display Call Pickup feature.

Prompt	Response	Description
REQ	CHG	Change.
TYPE	aaaa	Telephone type, where: aaaa = SL1, 2006, 2008, 2009, 2016, 2018, 2112, 2216, 2317, 2616, and 3000.
TN	c u	Terminal Number for the Option 11.
...		
RNPG	(0)-4095	Ringing Number Pickup Group (RNPG). If the RNPG is set to 0 (the default) on a station, it is not possible to pick up any call ringing this station from any other set. Enter 0 to remove a station from the RNPG. Note: If the RPNG is two or more digits in length, it must be configured in the Customer Data Block (LD 15), so that it may use the Group Pickup key. See the procedure which follows.
...		
CLS	(PUD), PUA (DPUD), DPUA (GPUD), GPU A	Ringing Number Pickup (denied) allowed. DN pickup (denied) allowed. Group pickup (denied) allowed.
...		
KEY	xx RNP yyyy	Key number, Ringing Number Pickup, Pickup Group number (optional). If the Group number is not entered, the key will pick up calls in the group assigned to the station. If the Group number is entered, the key will pick up calls in the specified group yyyy.

KEY	xx GPU	Key number, Group Number Pickup.
KEY	xx DPU	Key number, DN Pickup.
...		
KEY	xx DSP	Key number, Digit Display.

LD 15 - If the customer's Ringing Number Pickup Group is two or more digits in length (as configured in LD10/11, in response to the RNPG prompt), it must be configured in the Customer Data Block (in response to the PKND prompt); otherwise the Group Pickup key cannot be used by the group.

Prompt	Response	Comment
REQ:	CHG	Change existing data block
TYPE:	FTR_DATA	Customer Features and options
CUST	0-31	Customer number
- OPT	aaa	Options
- DGRP	(0)-2046	Maximum number of Dial Intercom Groups
- IRNG	(NO) YES	Intercom Ring
- PKND	(1)-4	Number of digits Dialed for Group Pickup
...		

LD 57 – Define the Flexible Feature Codes (FFCs) used to activate Call Pickup and Directed Call Pickup from a an analog (500/2500 type) telephone. The FFCs may also be used on Meridian 1 proprietary telephone sets.

Prompt	Response	Description
REQ	CHG	Change.
TYPE	FFC	Flexible Feature Code.
CUST	0-31	Customer number.
...		
CODE	mmmm	Specific FFC type.
- PURN	xxxx	Pickup Ringing Number code.
- PUGR	xxxx	Pickup Group code.
- PUDN	xxxx	Pickup DN code.

LD 18 – Configure a Speed Call List. This Speed Call List will be used by a Ringing Number Pickup Group (RNPG) or a set of RNPGs as a search list to scan the MCDN network.

The Speed Call List entries should contain digits which can be used to route a network Call Pickup request to a remote node (e.g., mainly the ISDN PINX DNs of the remote nodes which will be scanned after a network Call Pickup request). There must be no gaps in the Speed Call List (i.e., each Speed Call List entry should be present). If during the network scanning a missing Speed Call List entry is encountered, the network scanning is stopped (as if the Speed Call List was exhausted).

Due to the time it takes to scan the remote node, it is strongly recommended to configure less than six entries in the Speed Call List.

Prompt	Response	Description
REQ	NEW, CHG, OUT	New, change, or remove.
TYPE	SCL	Speed Call List.
LSNO	0-8190	Speed Call List number
...		
DNSZ	4-(16)-31	Maximum size of DNs allowed for Speed Call list.
SIZE	1-1000	Maximum number of DNs allowed in Speed Call list. Note: The size cannot be greater than the value entered against the STOR prompt below.
...		
STOR	000-319 xxxx	Speed Call List entry number and digits (PINX DN) stored against it. Note: The STOR entry cannot be blank.

LD 18 – On a per customer basis, define the Ringing Number Pickup Groups that are network wide by assigning them to one of the previous defined Speed Call Lists. Different RNPGs may be assigned to different Speed Call Lists.

Prompt	Response	Description
REQ	NEW, CHG, OUT	New, change, or remove.
TYPE	CPNW	Call Pickup Network Wide data.
CUST	0-31	Customer number.
LSNO	0-8190	A Speed Call List associated with Call Pickup network wide groups.
- GRP	0-4095	Ringing Number Pickup Group (RNPG) using this Speed Call List. Repeat for all groups sharing the same list. Enter <CR> to reprompt LSNO.
	X	Enter X to remove an RNPG.

LD 17 – Define the software Release ID at the far end of the D-Channel. Both the target and destination switches must be equipped with a minimum software vintage of Generic X11 Release 21 in order for Call Pickup Network Wide to be functional.

Prompt	Response	Description
REQ	CHG	Change I/O device.
ADAN	CHG DCH 0-63	Change D-channel information
- RLS	21	Release ID of the switch at the far end of the D-channel.
...		Note: Both the target and destination switches must be equipped with a minimum software vintage of Release 21.

LD 15 – Define the customer's Private Integrated Network Identifier (PNI) and Private Integrated Services Network Exchange (PINX) DN.

Prompt	Response	Description
REQ	CHG	Change.
TYPE	NET_DATA	Networking data.
CUST	0-31	Customer number.
...		
ISDN	YES	Allow ISDN for this customer.
- PNI	1-32700	Define the Private Network Identifier.
- PINX_DN	xx...x	Node DN, up to seven digits with DN Expansion package 150.
	X	Enter X to remove.

LD 15 – Define the special prefix code (SPRE) to be able to activate the Call Pickup features by dialing SPRE + xx.

Prompt	Response	Description
REQ	CHG	Change.
TYPE	FTR_DATA	Customer Features and options
CUST	0-31	Customer number.
...		
- SPRE	xxxx	Special Prefix number for this customer.
...		

LD 16 – Configure the PNI on the route. This PNI must correspond to the one configured in LD 15 on the target (remote) node.

Prompt	Response	Description
REQ	CHG	Change.
TYPE	RDB	Route data block.
ROUT	0-511	Route number.
...		
ISDN	YES	ISDN is allowed on this route.
- PNI	1-32700	Define the Private Network Identifier.

Calling Party Privacy

LD 57 – Define the FFC for CPP feature. FFC package 139 is required for loading LD 57.

Prompt	Response	Description
REQ	NEW, CHG	Add, or change.
TYPE	FFC	Flexible Feature Code.
CUST	xx	Customer number.
FFCT	(NO) YES	Flexible Feature Confirmation Tone.
...		
CODE	CPP	FFC type to be altered. CPP entry is only allowed if the CPP package is equipped. <CR> means that no FFC types are prompted.
...		
CPP	(*67) nnnn	Calling Party Privacy code. CPP is prompted only if the CPP package is equipped. Any arbitrary digit sequence up to four digits can be specified. For Meridian 1 proprietary set, an “*” can be entered as the first digit. CPP will be prompted until a <CR> is entered.

LD 16 – Define Privacy Indicators. CPP is only prompted if the CPP package is equipped, the OPAO package 104 is **not** equipped, the trunk outgoing (OGT) or incoming and outgoing (IAO), non-ISDN option and the trunk route type is COT, DID, FEX, or WAT.

Prompt	Response	Description
REQ	NEW, CHG	Add, or change.
TYPE	RDB	Route Data Block.
CUST	xx	Customer number.
TKTP	COT DID FEX WAT	Central Office Trunk data block. Direct Inward Dialing trunk data block. Foreign Exchange trunk data block. Wide Area Telephone Service trunk data block.
...		
DTRK	YES	Digital trunk route.
DGTP	DTI DTI2	Digital trunk type.
ISDN	YES	ISDN PRI option.
...		
ACOD	nnnn	Trunk Access Code.
CPP	YES	Calling Party Privacy. YES = This trunk route is enabled for the recognition of the Calling Party Privacy feature. CPP is only prompted if the following conditions are met: the CPP package is equipped, the OPAO package is not equipped, OGT (outgoing) or IAO (incoming and outgoing) trunk, non ISDN option and trunk route type is COT/DID/FEX/WAT. The default value for the CPP prompt is NO.
TCPP	(NO) YES	CPP for an incoming trunk call tandemed to this trunk route. YES = An incoming non-ISDN trunk call tandemed to this trunk route will carry the Privacy Indicator. The default value for the TCPP is NO.

- DTPI	nnnn	Privacy Indicator for a digitone trunk. DTPI is prompted only if CPP is set to "YES" and the trunk route is non-ISDN. If CPP is changed from NO to YES, the default is *67. Any arbitrary digit sequence (0-9) up to four digits can be specified. An asterisk "*" is allowed to be the first digit only if the outgoing call goes to a Public Network.
- DPPI	nnnn	Privacy Indicator for a dial pulse trunk. DPPI is prompted only if CPP is set to "YES" and the trunk route is non-ISDN. If CPP is changed from NO to YES, the default is 1167. Any arbitrary digit sequence (0-9) up to four digits can be specified.
- PII	(NO) YES	Calling Party Privacy Indicator is honored. Calling Party Privacy Indicator is ignored. PII is prompted with X11 Release 23 and later. Note: PII is only prompted when the CPP package is equipped; the trunk route type is COT, DID, FEX, or WAT; the ISDN option is set to YES; the ISDN Interface (IFC) is D100, D250, ESS4, ESS5, or NI2; and the route is Incoming and Outgoing (IAO) or Incoming Only Trunk (ICT).

LD 10/11 – Activate Calling Party per-line blocking. CLBA Class of Service activates Calling Party per-line blocking. CLBD Class of Service deactivates Calling Party per-line blocking; however, the user can still request Calling Party Privacy by dialing the CPP code.

Prompt	Response	Description
REQ	NEW, CHG	Add, or change.
TYPE	nnnn	Type of telephone.
TN	c u	Terminal Number.
...		
CLS	CLBA	Activate Calling Party per-line blocking. Enter CLBD to deactivate Calling Party per-line blocking (default). Note: CLBA Class of Service activates Calling Party per-line blocking. CLBD Class of Service deactivates Calling Party per-line blocking; however, the user can still request Calling Party Privacy by dialing the CPP code.

Display of Access Prefix on CLID

The following procedures are required to configure the Display of Access Prefix on CLID feature:

- Enable the Display of Access Prefix on CLID feature for digital sets in LD 11.
- Enable the Display of Access Prefix on CLID for attendant consoles in LD 12.
- Configure the access prefix table in LD 15.
- Enable the Display of Access Prefix on CLID feature in LD 16.

LD 11 - Enable the Display of Access Prefix on CLID feature for digital sets.

Prompt	Response	Description
REQ:	NEW CHG	Add new data. Change existing data.
TYPE:	aaaa	Type of Meridian 1 proprietary set.
TN	c u	Terminal Number. c = card, u = unit for Option 11C.
DES	x...x	Office Data Administration System Designator.
CUST	xx	Customer Number as defined in LD 15.
...	...	
CLS	(DAPA)	Status of Display of Access Prefix on CLID. (DAPA) = Default enables the Display of Access Prefix on CLID. DAPD = Disable Display of Access Prefix on CLID.

LD 12 - Enable the Display of Access Prefix on CLID for attendant consoles.

Prompt	Response	Description
REQ	NEW CHG	Add new data. Change existing data.
TYPE	aaaa	Attendant type. Where aaaa = 1250 or 2250.
TN	c u	Terminal Number. c = card, u = unit for Option 11C.
SETN	c u	Second Terminal Number. c = card, u = unit for Option 11C.
CUST	xx	Customer Number as defined LD 15.
ANUM	1-63	Attendant Number.
...	...	
DAPC	(DAPA)	Status of Display of Access Prefix on CLID. (DAPA) = Enable Display of Access Prefix on CLID. DAPD = Disable Display of Access Prefix on CLID.

LD 15 - Configure the access prefix table.

Prompt	Response	Description
REQ:	CHG	Change existing data block.
TYPE:	FTR	Customer Features and options.
CUST	xx	Customer Number associated with this feature.
...	...	
DAPC	YES	Configure Display of Access Prefix table entry option. (NO) is the default.
-TBL	1-15	Table number. Table 0 is non configurable. Precede table number with "X" to remove.

--NPI	aaaa	Numbering Plan Identification. Valid entries for aaaa include: UNKN = Unknown. E164 = Numbering Plan based on E164. PRIV = Private. E163 = Numbering Plan based on E163. TELX = Telex. X121 = Data X121. NATL = National.
--TON	aaaa	Type of number. Valid entries for aaaa include: UNKN = Unknown. INTL = International. NATL = National. ESPN = ESN_SPN. LOCL = Local. ELOC = ESN_LOC. ECDP = ESN_CDP.
---PREF	0-9999	Up to four digit Access Prefix for a unique NPI/TON combination in the table. Carriage return is taken as NIL Access Prefix value.
	#	Wild Character for replacement of any digit. The entry of “#” for wild card character is stored and displayed as “*”. Note: The entry of “*” would be misinterpreted by the overlay supervisor for the standard overlay operation.)
	X	Reset the access prefix value to NIL.
--TON	...	Repeat for every value of TON for the particular NPI.
--NPI	...	Repeat for every value of NPI.

LD 16 - Enable the Display of Access Prefix on CLID feature.

Prompt	Response	Description
REQ	NEW CHG	Add new data. Change existing data.
TYPE	RDB	Route Data Block.
CUST	xx	Customer Number as defined in LD 15.
ROUT	xxx	Route number. xxx = 0-511 for Options 51C, 61C, 81, and 81C. xxx = 0-127 for Option 11C.
...	...	
ISDN	YES	Integrated Services Digital Network route.
...	...	
DAPC	YES	Enable feature at the route data block Level. YES = Enable Display Access Prefix CLID. (NO) = (default) Disable feature at the route data block level.
-TBL	1-15	Prefix table number as defined in LD 15.

Note 1: Set related printing is done in Overlay 20. Overlay 20 is modified to allow the printing of the Class of Service for Access Prefix display allowed or denied.

Note 2: To print the table number associated with the route data block, the feature is enabled at the route data block level.

LD 21- Print the table number associated with the route data block.

Prompt	Response	Description
REQ	PRT	Print route/customer data block.
TYPE	FTR_DATA/FT R	Route data block/Prefix table
TBL	0-15	Prefix table number. When a <CR> is entered all the configured tables for that customer are printed.

Display of Calling Party Denied

LD 10 – Deny Directory Number Display and Name Display for Analog (500/2500 type) sets.

Prompt	Response	Description
REQ:	CHG	Change.
TYPE:	500	500/2500 type set data block.
TN	c u	Terminal Number.
...		
CLS	DDGD	Digit Display Denied DDGA = Digit Display Allowed (default).
CLS	NAMD	Name Display Denied NAMA = Name Display Allowed (default).

LD 11 – Deny Digit Display and Name Display for Meridian 1 proprietary sets.

Prompt	Response	Description
REQ:	CHG.	Change.
TYPE:	xxxx	Telephone type, where: xxxx = SL1, 2006, 2008, 2009, 2016, 2018, 2112, 2216, 2317, 2616 or 3000.
TN	c u	Terminal Number.
...		
CLS	DDGD	Digit Display Denied DDGA = Digit Display Allowed (default).
CLS	NAMD	Name Display Denied NAMA = Name Display Allowed (default).

E.164/ESN Numbering Plan Expansion

LD 10 — For analog (500/2500 type) sets, configure the expansion pertaining to the Autodial, Call Forward All Calls, Default Call Forward (for Phantom TNs) and Internal Call Forward DN.

Prompt	Response	Description
REQ:	NEW, CHG	Add new data, or change existing data.
TYPE:	500	Type of telephone set.
TN	c u	Terminal Number.
...		
FTR	ADL nn x..x	Autodial where nn = number of digits up to a maximum of 31 in Autodial DN and x..x = Autodial DN.
	CFW (4)-31	Enter the maximum number of digits in the Call Forward All Calls DN. Note that there is no default value.
	DCFW ll xxx...x	Enter the maximum number of digits in the Default Call Forward DN for a Phantom DN. ll = 4,8,12,16,20,24,28,31 (entries are rounded up to the next valid length.) xx...x = the Default Call Forward All Calls DN.
	ICF (4)-31	Enter the maximum number of digits in the Internal Call Forward DN for a Phantom DN.

LD 11 – For Meridian 1 proprietary sets, configure the expansion pertaining to the Autodial, Call Forward All Calls, Default Call Forward (for Phantom TNs) and Internal Call Forward DN.

Prompt	Response	Description
REQ:	NEW, CHG	Add new data, or change existing data.
TYPE:	xxxx	Type of telephone set. xxxx = SL1, 2006, 2008, 2009, 2016, 2018, 2112, 2116, 2317, 2616, M3000.
TN	c u	Terminal Number.
KEY	nn ADL ll xxx...x	Telephone function key assignments. Enter the maximum number of Autodial digits that may be stored. nn = Key number ll = 4,8,12,(16),20,24,28,31,NUL (for M3000 sets, the accepted range is (4)-28) Entries are rounded up to the next valid length. An entry of NUL will disable the Autodial feature xx...x = the Autodial target DN (this is an optional entry.)
	nn CA ll xxx...x	Enter the maximum number of digits that may be stored for the combined Conference-Autodial DN. nn = Key number ll = 4,8,12,(16),20,24,28,31 (entries are rounded up to the next valid length.) xx...x = the Conference-Autodial target DN (this is an optional entry.)
	nn CFW ll xxx...x	Enter the maximum number of digits that may be stored for the Call Forward All Calls DN. nn = Key number ll = (4)-31 (for M2317 and M3000 sets, the accepted range is (4)-23) xx...x = the Call Forward All Calls DN (this is an optional entry.)
	nn ICF ll xx...x	Enter the maximum number of digits that may be stored for the Internal Call Forward DN. nn = Key number ll = (4)-31 xx...x = the Internal Call Forward DN (this is an optional entry.)

LD 12 – For attendant consoles, configure the expansion pertaining to the maximum length of the Autodial DN.

Prompt	Response	Description
REQ	NEW, CHG	Add new data, or change existing data.
TYPE	xxxx	Type of attendant console xxxx = 1250, 2250.
TN	c u	Terminal Number.
KEY	nn ADL xxx...x	Attendant console function key assignments. Enter the maximum number of Autodial digits that may be stored. nn = Key number xxx...x = the Autodial DN, up to 31 digits (this is an optional entry.)

LD 15 - Configure the expansion pertaining to the Customer Call Forward DN.

Prompt	Response	Description
REQ:	NEW, CHG	Add new data, or change existing data.
TYPE:	RDR	Redirection data.
CUST	xx	Customer number.
...		
- CCFWDN	xxx...x X	Enter the Customer Call Forward DN, up to 23 digits (this entry is made for a new CCFWDN, or if the length of an existing CCFWDN has been expanded.) Enter X for no entry.

LD 23 - Configure the expansion pertaining to the ACD Night Call Forward DN and the ACD Interflow DN.

Prompt	Response	Description
REQ	NEW, CHG	Add new data, or change existing data.
TYPE	ACD	Automatic Call Distribution data block.
CUST	xx	Customer number.
ACDN	xxx...x	ACD DN.
...		
NCFW	xxx...x	Enter the ACD Night Forward DN, up to 31 digits (this entry is made for a new Night Forward DN, or if the length of an existing Night Forward DN has been expanded.)
...		
IFDN	xxx...x	Enter the ACD Interflow DN, up to 31 digits (this entry is made for a new Interflow DN, or if the length of an existing Interflow DN has been expanded.)

LD 86 - For the ESN data block, configure the expansion pertaining to the maximum number of Supplemental Digit Restriction and Recognition blocks, and the maximum number of Digit Manipulation tables.

Prompt	Response	Description
REQ	NEW, CHG	Add new data, or change existing data.
CUST	xx	Customer number.
FEAT	ESN	ESN data block feature.
...		
MXSD	(0)-1500	Enter the maximum number of Supplemental Digit Restriction and Recognition (SDRR) blocks, for BARS or NARS. Enter 0 (the default) if no SDRR blocks are required.
...		
MXDM	(0)-1000 (0)-256 (0)-32	Enter the maximum number of Digit Manipulation tables. (0)-1000 = for Flexible Numbering Plan (0)-256 = for BARS/NARS (0)-32 = for Coordinated Dialing Plan Enter 0 (the default) if no Digit Manipulation tables are required.

LD 86 - For the Digit Manipulation data block, configure the expansion pertaining to the maximum number of leading digits to be deleted and inserted.

Prompt	Response	Description
REQ	NEW, CHG	Add new data, or change existing data.
CUST	xx	Customer number.
FEAT	DGT	Digit Manipulation data block feature.
...		
DMI	1-999 1-255 1-31	Enter the Digit Manipulation table. 1-999 = for Flexible Numbering Plan 1-255 = for BARS/NARS 1-31 = for Coordinated Dialing Plan. The maximum number of Digit Manipulation tables is defined using prompt MXDM, in the ESN data block.
DEL	(0)-19	Enter the number of leading digits to be deleted. Enter 0 (the default) for none.
INST	x...x x...x*y...y X	Enter the number of leading digits to be inserted. x...x = 1-31 x...x*y...y = for Special Common Carriers (SCCs), 1-30, including access number (x...x), delimiter (*), and authorization code (y...y) Enter X for none.

LD 86 - For the Route List data block, configure the expansion pertaining to the minimum Overlap Digit Length.

Prompt	Response	Description
REQ	NEW, CHG	Add new data, or change existing data.
CUST	xx	Customer number.
FEAT	RLB	Route List data block feature.
RLI	0-999	Route List Index.
DMI	1-999 1-255 1-31	Enter the Digit Manipulation table. 1-999 = for Flexible Numbering Plan 1-255 = for BARS/NARS 1-31 = for Coordinated Dialing Plan. The maximum number of Digit Manipulation tables is defined using prompt MXDM, in the ESN data block.
...		
OVLL	(0)-24	Enter the minimum Overlap Digit Length, pertaining to Overlap Sending. If 0 (the default) is entered, then the Flexible Digit Number Length (FLEN) determines whether Overlap Sending takes place.

LD 87 - For a Free Special Number Screening Index, configure the expansion pertaining to the maximum number of Special Numbers to be screened.

Prompt	Response	Description
REQ	NEW, CHG	Add new data, or change existing data.
CUST	xx	Customer number.
FEAT	FSNS	Free Special Number Screening.
FSNI	1-255	Free Special Number Screening Index.
SPN	1-19	Special Number to be screened.

LD 87 - For a Trunk Steering Code, configure the expansion pertaining to the maximum number of Flexible Numbers.

Prompt	Response	Description
REQ	NEW, CHG	Add new data, or change existing data.
CUST	xx	Customer number.
FEAT	CDP	Coordinated Dialing Plan.
TYPE	TSC	Trunk Steering Code.
TSC	1-7	Trunk Steering Code.
- FLEN	0-24	Flexible Number.

LD 90 - For Network Translation tables, configure the expansion pertaining to the maximum number of Special Numbers digits, the maximum number of digits for Flexible Numbers, the maximum number of Supplemental Digit Restriction and Recognition digits per entry, and the addition of “allow” (ALLOW) as a new Supplemental Digit Restriction and Recognition type.

Prompt	Response	Description
REQ	NEW, CHG	Add new data, or change existing data.
CUST	xx	Customer number.
FEAT	NET	Network Translation Tables data.
...		
TYPE	SPN	Special Number Translation.
SPN	xxxx xxxx x...x	Enter the Special Number Translation digits. Up to 19 digits may be entered. The SPN digits must be entered in groups of at most four digits, separated by a space. Up to five groups can be entered. The restriction is removed of allowing one less digit if the first digit of a digit string is not “1”.
- FLEN	(0)-24	Enter the number of Flexible Digits (the number of digits that the system expects to receive before accessing a trunk, and outpulsing the digits.)
...		
- SDRR	DENY LDID LDDD DID DDD ITED ARRN STRK ALLOW	Type of Supplemental Digit Restriction and Recognition. Restricted codes. Recognized local DID codes. Recognized local DDD codes. Recognize remote DID codes. Recognize remote DDD codes. Incoming trunk group exclusion digits. Alternate routing remote number. Allowed codes for ADM/MDM. Allowed codes.

		The maximum number of digits entered in response to the DENY, LDID, LDDD, DID, DDD and ALLOW prompts, which follow, must be less than 10, or 7-m (8-m for 1+ dialing) for Central Office translation data (NXX), 10-m (11-m for 1= dialing) for Numbering Plan Area Code translation data (NPA), or 19-m for Special Numbers Translation data (SPN), where m = the number of digits entered for the prompt NXX, NPA or SPN. These numbers are no longer required to be leftwise unique. For non leftwise unique numbers, the longer number takes precedence over the shorter number. However, the exact same numbers (non leftwise unique and the same length) are still blocked.
- - DENY	x...x	Restricted number to be denied.
- - LDID	x...x	Local DID number to be recognized.
- - LDDD	x...x	Local DDD number to be recognized.
- - DID	x...x	Remote DID number to be recognized.
- - DDD	x...x	Remote DDD number to be recognized.
- - ALLOW	x...x	Code to be allowed.

Electronic Lock Network Wide/Electronic Lock on Private Lines

Electronic Lock Network Wide

Use the following overlays to implement Electronic Lock Network Wide.

LD 15 – Modify Customer Data Block for Controlled Class of Service data as follows

Prompt	Response	Description
REQ:	CHG	Change existing data block.
TYPE:	CCS_DATA	Controlled Class of Service data.
CUST	0-31	Customer number.
- CCRS	aaa	Controlled class of service (CCOS) Restricted Service.
- ECC1	aaa	Enhanced Controlled Class of Service level 1.
- ECC2	aaa	Enhanced Controlled Class of Service level 2.
- CNCS	0-99	Network Controlled Class Of Service for Electronic Lock.
- PELK	YES	Electronic Lock on Private Lines.

LD 15 – Modify Customer Data Block for Flexible Feature Code data as follows:

Prompt	Response	Description
REQ:	CHG	Change existing data block.
TYPE:	FFC_DATA	Flexible Feature Code data.
CUST	0-31	Customer number.
- CCRS	aaa	Controlled class of service (CCOS) Restricted Service.
- SCPL	0-8	Station Control Password Length.
- SBUP	(YES) NO	Enable use of station control passwords for set based administration user level access

- - PWD2	xxxx	PWD2 password for confirmation
- FFCS	(NO) YES	Change Flexible Feature Code end-of-dialing indicator
- - STRL	1-3	String Length of end-of-dial indicator
- - STRG	xxx	String to indicate end-of-dialing
- ADLD	(0)-20	Auto Dial Delay in seconds
- SCPL	0-8	Station Control Password Length

LD 10 – Set the Station Control Password and Controlled Class of Service Allowed for each analog (500/2500 type) telephone.

Prompt	Response	Description
REQ	NEW,CHG	Add,Change.
TYPE	500	Telephone type.
CUST	0-31	Customer Number.
TN	c u	Card, and unit.
SCPW	xxxxxxxx	Station Control Password (1 to 8 digits, as defined in LD 15).
CLS	CCSA	Controlled Class of Service Allowed.

LD 11 – Set the Station Control Password and Controlled Class of Service Allowed for each Meridian 1 proprietary telephone.

Prompt	Response	Description
REQ	NEW,CHG	Add,Change.
TYPE	aaaa	Telephone type, where aaaa = SL1, 2006, 2008, 2009, 2016, 2018, 2112, 2216, 2317, 2616, or 3000.
CUST	0-31	Customer Number.
TN	c u	Card, and unit.
SCPW	xxxxxxxx	Station Control Password (1 to 8 digits, as defined in LD 15).

CLS	CCSA	Controlled Class of Service Allowed.

LD 57 – Configure the Flexible Feature Codes for Electronic Lock.

Prompt	Response	Description
REQ	NEW/CHG	Enter new/change data.
TYPE	FFC	Flexible Feature Codes.
CUST	0-31	Customer Number.
FFCT	YES	Provide FFC confirmation tone.
CODE	ELKA	New/change Electronic Lock Activate FFC.
ELKA	xxxx	Enter the new or changed Electronic Lock Activate FFC
ELKA	<cr>	
CODE	ELKD	New/change Electronic Lock Deactivate FFC.
ELKD	xxxx	Enter the new or changed Electronic Lock Deactivate FFC
ELKD	<cr>	

LD 87 – In the ESN 2 overlay, define all NCOS to be used.

Prompt	Response	Description
REQ	NEW/CHG	Enter new/change data.
CUST	0-31	Customer Number.
FEAT	NCTL	Network Control.
...		
NRNG	0-99	NCOS range.
NCOS	0-99	Network Class of Service group number.
...		
– FRL	(0)-7	Facility Restriction level.

Electronic Lock on Private Lines

Use the following overlays to implement Electronic Lock Network on Private Lines.

LD 15 – Modify Customer Data Block for Controlled Class of Service data as follows

Prompt	Response	Description
REQ:	CHG	Change existing data block.
TYPE:	CCS_DATA	Controlled Class of Service data.
CUST	0-31	Customer number.
- CCRS	aaa	Controlled class of service (CCOS) Restricted Service.
- ECC1	aaa	Enhanced Controlled Class of Service level 1.
- ECC2	aaa	Enhanced Controlled Class of Service level 2.
- CNCS	0-99	Network Controlled Class Of Service for Electronic Lock.
- PELK	YES (NO)	Electronic Lock on Private Lines (for a private line); otherwise enter NO.

LD 15 – Modify Customer Data Block for Flexible Feature Code data as follows

Prompt	Response	Description
REQ:	CHG	Change existing data block.
TYPE:	FFC_DATA	Flexible Feature Code data.
CUST	0-31	Customer number.
- CCRS	aaa	Controlled class of service (CCOS) Restricted Service.
- SCPL	0-8	Station Control Password Length.
- SBUP	(YES) NO	Enable use of station control passwords for set based administration user level access
- - PWD2	xxxx	PWD2 password for confirmation

- FFCS	(NO) YES	Change Flexible Feature Code end-of-dialing indicator
- - STRL	1-3	String Length of end-of-dial indicator
- - STRG	xxx	String to indicate end-of-dialing
- ADLD	(0)-20	Auto Dial Delay in seconds
- SCPL	0-8	Station Control Password Length

LD 16 – Set the Facility Restriction Level (FRL) number and the New Flexible Code Restriction (NFCR) tree number for private lines.

Prompt	Response	Description
REQ	CHG	Change.
TYPE	RDB	Route Data Block.
...		
PRIV	YES	Private Line Route.
...		
FRL	0-7 0-254	Facility Restriction Level (FRL) and New Flexible Code Restriction (NFCR) tree number.

LD 10 – Set the Station Control Password and Controlled Class of Service Allowed for each analog (500/2500 type) telephone.

Prompt	Response	Description
REQ	NEW,CHG	Add,Change.
TYPE	500	Telephone type.
CUST	0-31	Customer number.
TN	c u	Card, and unit.
SCPW	xxxxxxx	Station Control Password (1 to 8 digits, as defined in LD 15).
CLS	CCSA	Controlled Class of Service Allowed.

LD 11 – Set the Station Control Password and Controlled Class of Service Allowed for each Meridian 1 proprietary telephone.

Prompt	Response	Description
REQ	NEW,CHG	Add,Change.
CUST	0-31	Customer number.
TYPE	aaaa	Telephone type, where aaaa = SL1, 2006, 2008, 2009, 2016, 2018, 2112, 2216, 2317, 2616, or 3000.
TN	c u	Card, and unit.
SCPW	xxxxxxx	Station Control Password (1 to 8 digits, as defined in LD 15).
CLS	CCSA	Controlled Class of Service Allowed.
...		
KEY	xx PVR yyyy(yyy)	Key number and DN for Private Line Ringing, where xx = the key number and yyyy(yyy) = the Private Line Ringing DN (up to four digits; up to seven digits if the Directory Number Expansion (DNXP) package 150 is equipped). Private Line Directory Number (PRDN) must be defined in LD 14.

Equi-distribution NAS routing

Use the following procedure to configure Equi-distribution NAS routing.

Procedure 1

Configuring equi-distribution for NAS routing

Step	Overlay	Action
1	LD 15 Customer Data Block	Define the efficiency factor loading level.
2	LD 93 Multi-Tenant Service	Define the efficiency factor loading level.
3	LD 86 ESN	Define the attendant alternatives for NAS routing.
4	LD 86 ESN	Enable/disable Drop Back Busy option.

Step 1

Enter the following command in Overlay 15.

Prompt	Response	Description
EFL	0 — 8064	The efficiency factor loading level. If the default value of 0 is entered, then the Equi-distribution NAS Routing feature is not activated, and normal NAS routing occurs.

Step 2

Enter the following command in Overlay 93.

Prompt	Response	Description
EFL	0 — 8064	The efficiency factor loading level. If the default value of 0 is entered, then the Equi-distribution NAS Routing feature is not activated, and normal NAS routing occurs.

Step 3

Enter the following command in Overlay 93.

Prompt	Response	Description
ALST	1 - 7	Attendant alternatives for NAS routing (up to 4 alternatives may be defined for each schedule period).

Step 4

Enter the following command in Overlay 93.

Prompt	Response	Description
DBK	YES, (NO)	YES = Drop Back Busy allowed, NO = Drop Back Busy denied.

Error Handling on ISDN Signaling Link Analog E&M TIE Trunks

LD 16 – Configure ISL on Analog Trunks.

Prompt	Response	Description
REQ	NEW, CHG	Add, or change.
TYPE	RDB	Route data block.
...		
DTRK	NO	Digital trunk route.
ISDN	YES	ISDN Primary Rate Interface option.
MODE	ISLD	Mode of D-channel controlling the route.
...		
FALT	YES	Enables the error handling and recognition of E lead error for ISL E&M Tie Trunks. If FALT= NO, then no error handling with occur.

EuroISDN 7kHz/Videotelephony Teleservices

ISDN PRI2 implementation

LD 17 - Configure D-Channel for the ETS 300 403 interface.

Prompt	Response	Description
REQ:	CHG	Change existing data.
TYPE:	ADAN	Action Device and Number.
- ADAN	CHG DCH xx	Change existing D-channel data.
...		
- IFC	E403	EuroISDN Interface conforming to the ETS 300 403 or ETS 300 102 protocol.
- - CNTY	xxxx	Enter country pertaining to the E403 interface.

LD 16 - Configure the 7 kHz/Videotelephony teleservices for a PRI2 EuroISDN E403 route.

Note: The EuroISDN route using the 7 kHz/Videotelephony teleservices must be configured as using the interface type of ETS 300 403 (IFC = E403), even if the link to the Central Office is still be based on ETS 300 102 implementation.

Prompt	Response	Description
REQ	CHG	Change existing data.
TYPE	RDB	Route Data Block.
CUST	xx	Customer number, as defined in LD 15. xx = 0-31 for Option 11C.
ROUT	0-127	Route number.
TKTP	aaaa	Trunk type.
...		
DTRK	YES	EuroISDN route is digital.

- DGTP	PRI2	Digital trunk type. PRI2 = 2.0 Mbps Primary Rate Interface.
- IFC	E403	EuroISDN Interface conforming to the ETS 300 403, or the extended ETS 300 102 version for country-specific interfaces. Refer to the Note above.
- - CNTY		Enter country pertaining to the E403 interface. If either ETSI or NET are entered, the interface is fully compliant with ETS 300 403. The entered value must match the entry in LD 17.
	ETSI	ETS 300 403 for the user side.
	NET	ETS 300 403 for the network side. If any of the countries listed on the following page are entered, the interface functions with the extended ETS 300 102 capabilities.
	AUS	Austria.
	DEN	Denmark.
	FIN	Finland.
	GER	Germany.
	ITA	Italy.
	NOR	Norway.
	POR	Portugal.
	SWE	Sweden.
	EIR	Ireland.
	DUT	Holland.
	SWI	Switzerland.
	BEL	Belgium.
	ESP	Spain.
	UK	United Kingdom.
	FRA	France.
	CIS	Commonwealth of Independent States (Russia and the Ukraine).
...		
- DSEL		Data selection.
	7VOD	The route supports voice and data calls, and the telephony 7 kHz/Videotelephony teleservices.
	7DTA	The route supports data calls, and the telephony 7 kHz/Videotelephony teleservices.
...		

LD 14 - Configure the 7 kHz/Videotelephony teleservices PRI2 EuroISDN E403 trunks.

Prompt	Response	Description
REQ	NEW CHG	Add new data. Change existing data.
TYPE	aaaa	Trunk type. Must match the trunk type configured in LD 16.
CUST	xx	Customer number, as defined in LD 15. xx = 0-31 for Option 11C.
RTMB	0-127 1-510	Route number and member number
...		

ISDN Basic Rate Interface implementation

LD 16 - Configure the 7 kHz/Videotelephony teleservices for an ISDN BRI EuroISDN E403 route, in the Route Data Block.

Note: The EuroISDN route using the 7 kHz/Videotelephony teleservices must be configured as using the interface type of ETS 300 403 (IFC = E403), even if the link to the Central Office is still be based on ETS 300 102 implementation.

Prompt	Response	Description
REQ	CHG	Change existing data.
TYPE	RDB	Route Data Block.
CUST	xx	Customer number, as defined in LD 15. xx = 0-99 for Options 51C-81/81C xx = 0-31 for Option 11C.
ROUT	0-127	Route number.
TKTP	aaaa	Trunk type.
...		
DTRK	YES	EuroISDN route is digital.
- DGTP	BRI	Digital trunk type is Basic Rate Interface.
- IFC	E403	EuroISDN Interface conforming to the ETS 300 403, or the extended ETS 300 102 version for country-specific interfaces. Refer to the Note above.
- - CNTY		Enter country pertaining to the E403 interface. If either ETSI or NET are entered, the interface is fully compliant with ETS 300 403. The entered value must match the entry in LD 17.
	ETSI	ETS 300 403 for the user side.
	NET	ETS 300 403 for the network side.
		If any of the countries listed on the following page are entered, the interface functions with the extended ETS 300 102 capabilities.

	AUS DEN FIN GER ITA NOR POR SWE EIR DUT SWI BEL ESP UK FRA CIS	Austria. Denmark. Finland. Germany. Italy. Norway. Portugal. Sweden. Ireland. Holland. Switzerland. Belgium. Spain. United Kingdom. France. Commonwealth of Independent States (Russia and the Ukraine).
...		
- DSEL		Data selection.
	7VOD	The route supports voice and data calls, and the telephony 7 kHz/Videotelephony teleservices.
	7DTA	The route supports data calls, and the telephony 7 kHz/Videotelephony teleservices.
...		

LD 27 - Configure the ETSI ISDN BRI sets, on the Digital Subscriber Loop (DSL), for voice and data.

Prompt	Response	Comment
REQ	NEW CHG	Add new data Change existing data.
TYPE	DSL	Digital Subscriber Loop data block.
DSL	cc sdl	DSL Terminal Number location, Option 11C. cc (card) = 1-20 dsl# (DSL number) = 0-7
CUST	xx	Customer number, as defined in LD 15. xx = 0-31 for Option 11C.
CTYP	SILC UILC	Interface card type is SILC or UILC.
APPL	BRIL	Basic Rate Interface.
MISP	0-9	MISP loop number for Option 11C.
MODE	(TE) NT	Mode, user side (TE) or network side (NT).
B1CT	VCE DTA	B-Channel 1 call type is voice and data. Note: At least one of the B-Channels must be configured for voice and data.
B2CT	VCE DTA	B-Channel 2 call type is voice and data. Note: At least one of the B-Channels must be configured for voice and data.
...		

LD 27 - Configure the Terminal Service Profile (TSP) for voice and data, and the 7 kHz/Videotelephony teleservices on the ETSI ISDN BRI sets.

Prompt	Response	Comment
REQ	NEW CHG	Add new data Change existing data.
TYPE	TSP	Terminal Service Profile data block.
DSL	cc sdl	DSL Terminal Number location, Option 11C. cc (card) = 1-20 dsl# (DSL number) = 0-7
...		
BCH	1 2	B-Channel (either 1 or 2) to which the TSP is associated.
...		
- CT	VCE DTA	The call type that the TSP supports is voice and data.
...		
SSRV_ETSI	VID7 XVID7	The ETSI ISDN BRI set supports the 7kHz/Videotelephony teleservices. Precede with an X to remove the configured 7kHz/Videotelephony teleservices.
...		

EuroISDN Continuation

LD 10 – Configure Fax Allowed Class of Service for Analog (500 type) set.

Prompt	Response	Description
REQ:	NEW	Add.
TYPE:	500	Telephone type.
...		
CLS	FAXA	Fax Class of Service allowed for set or modem. ISDN call is generated with 3.1 KHz bearer capability. FAXD = Fax Class of Service denied.

LD 17 – Create a new D-channel.

Prompt	Response	Description
REQ	NEW	Add.
TYPE	ADAN	Action Device and Number.
- ADAN	NEW DCH x	New D-channel at port number x.
- CTYP	MSDL	Downloadable D-Channel Daughterboard.
- PORT	1	Port number on Downloadable D-Channel Daughterboard.
- USR	PRI	D-channel for ISDN PRI only. Precede with X to remove.
IFC	EURO	Interface type for D-channel, where: EURO = EuroISDN.
	SWIS	IFC SWIS is for PRI2 (SN2).

-- CNTY	AUS DEN (ETSI) FIN GER ITA NOR POR SWE EIR DUT SWI BEL ESP UK	Austria. Denmark. ETS 300-102 basic protocol. Finland. Germany. Italy. Norway. Portugal. Sweden. Ireland. Holland. Switzerland. Belgium. Spain. United Kingdom.
...		
-- DCHL	0-159	PRI loop number.
- RCAP	COLP	Enter COLP to support Connected Line Identification Presentation as a remote capability. Enter XCOL to remove Connect Line Identification Presentation.
PINX_CUST	0-99	This customer number will be used for DN address translation, associated with bearer independent connection messages received on this D-channel.
...		
PARM	CHG	Change system parameters.
BCAP	(SPEE) 31KH	Speech (the default). 3.1 Khz bearer capability setting for outgoing voice calls.

LD 16 – Configure new DID routes (the same responses to the IFC and CNTY prompts, as were used in LD 17, must be entered.)

Prompt	Response	Description
REQ	NEW	Add.
TYPE	RDB	Route Data Block.
CUST	xx	Customer number.
ROUT	0-511	Route number.
TKTP	DID	Trunk type, where: DID = Direct Inward Dialing.
...		
DTRK	YES	Digital trunk route.
- DGTP	PRI	Digital trunk type for route.
- IFC	EURO SWIS	Interface type for route, where: EURO = EuroISDN. IFC SWIS is for PRI2 (SN2).
- - CNTY	AUS DEN (ETSI) FIN GER ITA NOR POR SWE EIR DUT SWI BEL SPA UK	Austria. Denmark. ETS 300-102 basic protocol. Finland. Germany. Italy. Norway. Portugal. Sweden. Ireland. Holland. Switzerland. Belgium. Spain. United Kingdom.

CLID	OPT0	Prefix = 0, for North American dialing plan. This is the default value for ESIG and ISIG interfaces.
	OPT1	Prefix = 1, for international PFXs in CLID, any numbering type supported. This is the default value for all EuroISDN interfaces.
	OPT2	Prefix = 2, for international PFXs in CLID, CCITT numbering type supported: UKWN, INTL, NPA, and NXX. Default value for CO/DID routes for the Telecom New Zealand interface.
	OPT3	Prefix = 3, for international PFXs in CLID, only NXX number type supported. Default value for TIE routes for the Telecom New Zealand interface.
	OPT4	For international COs, if the call originates from a CO trunk type, add nothing. Otherwise, add PFX1 and PFX2. This is the default value for the Hong Kong, Singapore, and Thailand interfaces.
	OPT5	This is the same as OPT4, except it supports a maximum of 10 digits in the CLID. This is the default value for the Austrian interface.
PROG	NCHG	Send PROGRESS (default value for all interfaces, except Austria).
	MALE	Send ALERT after CALL PROCEEDING.
	MCON	Send CONNECT after CALL PROCEEDING (this is the default value for the Austrian interface).
...		
RCAP	COLP	Connected Number IE Presentation is supported on the far end (enter XCOLP to remove COLP).
- CPFXS	(YES) NO	Customer-defined Prefixes option. This prompt is added as one of the sub-prompts of ISDN = YES. It is added at the end of the ISDN sub-prompts. If CPFXS = YES, when constructing the Calling or Connected Line Identification, the prefixes are retrieved from the Customer Data Block via the PFX1 and PFX2 prompts in LD 15, as is currently done. This is the default response. If CPFXS = NO, when constructing the Calling or Connected Line Identification, the prefixes are retrieved from the Route Data Block via the HNTN and HLCL prompts in LD 16.

- - HNTN	0-9999	Home National Number. This number is similar to the PFX1 number prompted in LD 15. It is added to this overlay so that this prefix can be configured on a route basis as required in some countries (e.g., Italy). As is the case with PFX1, the HNTN prefix can be from one-to-four digits long. This prompt is displayed only if CPFXS = NO. If only a <CR> is entered, this prompt keeps its previous configuration. If no value was configured previously, no value will be configured. Enter X to delete the digits.
HLCL	0-9999	Home Location Number. This number is similar to PFX2 number prompted in LD 15. It is added to this overlay so that this prefix can be configured on a route basis as required in some countries (e.g., Italy). As is the case with PFX2, the HLCL prefix can be from one-to-four digits long. This prompt is displayed only if CPFXS = NO. If only a <CR> is entered, this prompt keeps its previous configuration. If no value was configured previously, no value will be configured. Enter X to delete the digits.
ADDP	(NO), YES	YES = the prefixes 0 (national) or 00 (international) are added to the Calling Party Number if the Type of Number (TON) is public. NO = the Calling or Connected Party Number displayed is not modified.

LD 14 – Configure new DID trunks.

Prompt	Response	Description
REQ	NEW xx	Add a new data block to the system. Follow NEW with a value of 1-255 to create that number of consecutive trunks.
TYPE	DID	Direct Inward Dialing data block.
TN	c u	Terminal number.
...		
RTMB	0-511 1-254	Route number; Member number.

EuroISDN Continuation Phase III

LD 15 – Configure the Optional Sending of Forwarding CLID functionality (in response to the OCLI prompt).

Prompt	Response	Description
REQ:	CHG	Change existing data.
TYPE:	NET	Networking data.
CUST	xx	Customer number.
...		
ISDN	YES	Integrated Services Digital Network.
PNI	(0)-32700	Private Network Identifier.
...		
SATD	0-(1)-5	Satellite Delays.
OCLI	(NO)	NO = No manipulation is done on outgoing CLID for calls forwarded over EuroISDN links. EXT = The last forwarding DN is sent as CLID information for incoming calls over a standard ISDN (EuroISDN, APAC, NI-2) DID or CO link to a gateway Meridian 1 node, and redirected back over the standard ISDN (EuroISDN, APAC, NI-2) DID or CO link. ALL = The same as for EXT. Moreover, the last forwarding DN is sent as CLID information for internal calls from a local set or over ISDN TIE trunks, except for DPNSS1 calls which are redirected from a gateway node over a standard ISDN (EuroISDN, APAC, NI-2) DID or CO link. If the redirection does not occur at the gateway node, the redirecting information number, if present in the incoming SETUP message received at the gateway node from the private network, is sent as CLID.
	EXT	
	ALL	
....		

LD 17 – Create a new D-channel for the EuroISDN interface.

Prompt	Response	Description
REQ	CHG	Change existing data.
TYPE	ADAN	Action Device and Number.
- ADAN	NEW DCH x	New D-channel at port number x.
- CTYP	MSDL	Card type, where MSDL = Downloadable D-Channel Board for Option 11C.
- PORT	1	Port number on the Downloadable D-Channel Daughterboard.
- USR	PRI	D-channel for ISDN PRI only. Precede with X to remove.
- IFC		DCH interface type.
	EURO	EURO = EuroISDN
- - CNTY		Enter country pertaining to EuroISDN interface.
	AUS	Austria
	DEN	Denmark
	(ETSI)	ETS 300-102 basic protocol
	FIN	Finland
	GER	Germany
	ITA	Italy
	NOR	Norway
	POR	Portugal
	SWE	Sweden
	EIR	Ireland
	DUT	Holland
	SWI	Switzerland
	BEL	Belgium
	ESP	Spain
	UK	United Kingdom
	FRA	France
	CIS	Commonwealth of Independent States (Russia and the Ukraine).
PINX_CUST	xx	The customer number used for the DN address.
...		

DCHL	0-159	The PRI loop number for the D-Channel.
- - CNEG	(1) 2	Channel Negotiation option. (1) = Channel is indicated and no alternative is acceptable. This is the default value for all EuroISDN Interfaces except FRA. 2 = Channel is indicated and any alternative is acceptable. This is the default value for the FRA interface.
- RCAP	COLP XCOL	Enter COLP to support Connected Line Identification Presentation as a remote capability. This is the default value for the ESIG, ISIG, NI2, and EURO interfaces. Enter XCOL to remove Connect Line Identification Presentation as a remote capability. This is the default value for the APAC, AUS, EIR, DEUT, ESP, BEL, and FRA interfaces.

LD 16 – Configure new DID routes (the same responses to the IFC and CNTY prompts, as were used in LD 17, must be entered).

Prompt	Response	Description
REQ	NEW CHG	Add new data. Change or delete existing data.
TYPE	RDB	Route Data Block.
CUST	xx	Customer number.
ROUT	0-127	Route Number.
TKTP	DID COT	Trunk type. DID = Direct Inward Dialing COT = Central Office.
...		
DTRK	YES	Digital trunk route.
- DGTP	PRI2 BRI	Digital trunk type for route.
...		

- IFC	EURO	Interface type. EURO = EuroISDN.
- - CNTY	AUS DEN (ETSI) FIN GER ITA NOR POR SWE EIR DUT SWI BEL ESP UK FRA CIS	Enter country pertaining to EuroISDN and Asia Pacific interface type. Austria Denmark ETS 300-102 basic protocol Finland Germany Italy Norway Portugal Sweden Ireland Holland Switzerland Belgium Spain United Kingdom France Commonwealth of Independent States (Russia and the Ukraine).
ICOG	IAO OGT ICT	Incoming and Outgoing trunk. Outgoing trunk only. Incoming trunk only.
ACOD	x..x	Access code for this trunk route.

- CLID	OPT0	Prefix = 0, for North American dialing plan. This is the default value for ESIG and ISIG interfaces.
	OPT1	Prefix = 1, for international PFXs in CLID, any numbering type supported. This is the default value for all EuroISDN interfaces.
	OPT2	Prefix = 2, for international PFXs in CLID, CCITT numbering type supported: UKWN, INTL, NPA, and NXX. Default value for CO/DID routes for the Telecom New Zealand interface.
	OPT3	Prefix = 3, for international PFXs in CLID, only NXX number type supported. Default value for TIE routes for the Telecom New Zealand interface.
	OPT4	For international COs, if the call originates from a CO trunk type, add nothing. Otherwise, add PFX1 and PFX2. This is the default value for the Hong Kong, Singapore, and Thailand interfaces.
	OPT5	This is the same as OPT4, except it supports a maximum of 10 digits in the CLID. This is the default value for the Austrian interface.
- PROG		Progress Signal.
	NCHG	Send PROGRESS (default value for all interfaces, except Austria).
	MALE	Send ALERT after CALL PROCEEDING.
...	MCON	Send CONNECT after CALL PROCEEDING (this is the default value for the Austrian interface).
- RCAP	COLP	Enter COLP to support Connected Line Identification Presentation as a remote capability. This is the default value for the ESIG, ISIG, NI2, and EURO interfaces.
	XCOL	Enter XCOL to remove Connect Line Identification Presentation as a remote capability. This is the default value for the APAC, AUS, EIR, DEUT, ESP, BEL, and FRA interfaces.
...		

- CPFXS	(YES) NO	Customer-defined Prefixes option. This prompt is added as one of the sub-prompts of ISDN = YES. It is added at the end of the ISDN sub-prompts. If CPFXS = YES, when constructing the Calling or Connected Line Identification, the prefixes are retrieved from the Customer Data Block via the PFX1 and PFX2 prompts in LD 15, as is currently done. This is the default response. If CPFXS = NO, when constructing the Calling or Connected Line Identification, the prefixes are retrieved from the Route Data Block via the HNTN and HLCL prompts in LD 16.
- - HNTN	0-9999	Home National Number. This number is similar to the PFX1 number prompted in LD 15. It is added to this overlay so that this prefix can be configured on a route basis as required in some countries (e.g., Italy). As is the case with PFX1, the HNTN prefix can be from one-to-four digits long. This prompt is displayed only if CPFXS = NO. If only a <CR> is entered, this prompt keeps its previous configuration. If no value was configured previously, no value will be configured. Enter X to delete the digits.
HLCL	0-9999	Home Location Number. This number is similar to PFX2 number prompted in LD 15. It is added to this overlay so that this prefix can be configured on a route basis as required in some countries (e.g., Italy). As is the case with PFX2, the HLCL prefix can be from one-to-four digits long. This prompt is displayed only if CPFXS = NO. If only a <CR> is entered, this prompt keeps its previous configuration. If no value was configured previously, no value will be configured. Enter X to delete the digits.
ADDP	(NO) YES	If ADDP = NO, the Calling or Connected Party Number displayed is not modified. If ADDP = YES, the prefixes 0 (national) or 00 (international) are added to the Calling Party Number if the Type of Number (TON) is public.

LD 17 – Configure the Numbering Plan Identification (NPI) and Type of Number (TON) fields to be included in the CDR tickets, in addition to the CLID, for EuroISDN calls (this is automatically done by entering YES in response to the CLID prompt).

Prompt	Response	Description
REQ	CHG	Change existing data.
TYPE	PARM	Change system parameters.
...		
- FCDR	NEW	Format for Call Detail Recording. Enter NEW for new format.
...		
- CLID	(NO) YES	Enter YES to include the TON and NPI fields in the CDR ticket, in addition to the CLID.
...		

EuroISDN ETS 300 403 Compliance Update

ISDN PRI2 implementation

LD 16 - Configure a PRI2 loop for the ETS 300 403 interface..

Prompt	Response	Description
REQ	CHG	Change existing data.
TYPE	CEQU	Make changes to Common Equipment parameters.
...		
- PRI2	0-159	PRI2 loop number.
...		

LD 17 - Configure a D-Channel for the ETS 300 403 interface..

Prompt	Response	Description
REQ:	CHG	Change existing data.
TYPE	ADAN	Action Device and Number.
- ADAN	NEW DCH xx	Add a D-Channel on port 0-15 (Option 11C).
...		
- CTYP	MSDL	Card type, where MSDL = Downloadable D-Channel card for Option 11C.
- PORT	1	Port number on Downloadable D-Channel Daughterboard.
- USR	PRI	D-channel for ISDN PRI only. Precede with X to remove.
- IFC	E403	EuroISDN Interface conforming to the ETS 300 403, or the extended ETS 300 102 version for country-specific interfaces.

- - CNTY	ETSI NET	Enter country pertaining to the E403 interface. If either ETSI or NET are entered, the interface is fully compliant with ETS 300 403. ETS 300 403 for the user side. ETS 300 403 for the network side. If any of the countries listed on the following page are entered, the interface functions with the extended ETS 300 102 capabilities.
	AUS DEN FIN GER ITA NOR POR SWE EIR DUT SWI BEL ESP UK FRA CIS	Austria. Denmark. Finland. Germany. Italy. Norway. Portugal. Sweden. Ireland. Holland. Switzerland. Belgium. Spain. United Kingdom. France. Commonwealth of Independent States (Russia and the Ukraine).
...		
- RLS	xx	Software Release of the far-end switch.
- RCAP	aaaa Xaaaa	Remote capabilities. Precede with an X to remove an existing configuration.
- OVLS	(NO) YES	(Do not) allow Overlap Sending.
- - OVLT	(0)-8	Duration of time, in seconds, that the sending side has to wait between INFO messages are sent. "0" means send immediately
- TIMR	(NO) YES	(Do not) change programmable timers. Only supported for interfaces supporting one of the following timers.
- - T310	(30)-100	Maximum time in seconds between an incoming CALL PROCEEDING message and the next incoming message.

- - INC_T306	0-(2)-30	Variable timer, in seconds, for received DISCONNECT message on incoming calls allowing in-band tone to be heard. The network will stop sending after this timer times out. The value is stored in two-second increments, which are rounded up.
- - OUT_T306	0-(30)	Variable timer, in seconds, for received DISCONNECT message on outgoing calls allowing in-band tone to be heard. The network will stop sending after this timer times out. The value is stored in two-second increments, which are rounded up.
- LAPD	(NO) YES	(Do not) change the parameters for the Link Access Protocol the D-Channel.
...		

LD 16 - Configure a new route for the E403 interface (the same responses to the IFC and CNTY prompts, as were used in LD 17, must be entered).

Prompt	Response	Description
REQ	NEW	Add new data.
TYPE	RDB	Route Data Block.
CUST	xx	Customer number, as previously defined in LD 15. xx = 0-99 for Options 51C-81/81C. xx = 0-31 for Option 11C.
ROUT	0 - 511 0-127	Route Number for Options 51C-81/81C. Route Number for Option 11C.
TKTP	TIE DID COT	Trunk type. TIE. Direct Inward Dialing. Central Office Trunk.
...		
DTRK	YES	Digital trunk route.
- DGTP	PRI2	2.0 MBit/s PRI digital trunk type.
...		
ISDN	YES	Integrated Services Digital Network.
- MODE	PRA	ISDN PRI route.
...		
- IFC	E403	EuroISDN Interface conforming to the ETS 300 403, or the extended ETS 300 102 version for country-specific interfaces.
- - CNTY		Enter country pertaining to the E403 interface. If either ETSI or NET are entered, the interface is fully compliant with ETS 300 403.
	ETSI NET	ETSI 300 403 for the user side. ETSI 300 403 for the network side. If any of the countries listed on the following page are entered, the interface functions with the extended ETS 300 102 capabilities.

	AUS DEN FIN GER ITA NOR POR SWE EIR DUT SWI BEL ESP UK FRA CIS	Austria Denmark Finland Germany Italy Norway Portugal Sweden Ireland Holland Switzerland Belgium Spain United Kingdom France Commonwealth of Independent States (Russia and the Ukraine).
ICOG	IAO OGT ICT	Incoming and Outgoing trunk. Outgoing trunk only. Incoming trunk only.
ACOD	xx..x	Access code for this trunk route.
- CLID	OPT0	Prefix = 0, for North American dialing plan. This is the default value for ESIG and ISIG interfaces.
	OPT1	Prefix = 1, for international PFXs in CLID, any numbering type supported. This is the default value for all EuroISDN interfaces.
	OPT2	Prefix = 2, for international PFXs in CLID, CCITT numbering type supported: UKWN, INTL, NPA, and NXX. Default value for CO/DID routes for the Telecom New Zealand interface.
	OPT3	Prefix = 3, for international PFXs in CLID, only NXX number type supported. Default value for TIE routes for the Telecom New Zealand interface.
	OPT4	For international COs, if the call originates from a CO trunk type, add nothing. Otherwise, add PFX1 and PFX2. This is the default value for the Hong Kong, Singapore, and Thailand interfaces.
	OPT5	This is the same as OPT4, except it supports a maximum of 10 digits in the CLID. This is the default value for the Austrian interface.

- PROG	NCHG	Progress Signal. Send PROGRESS (default value for all interfaces, except Austria).
	MALE	Send ALERT after CALL PROCEEDING.
	MCON	Send CONNNECT after CALL PROCEEDING (this is the default value for the Austrian interface).
...		
- RCAP	aaaa Xaaaa	Remote capabilities. Precede with an X to remove an existing configuration.
...		
- CPFXS	(YES) NO	Customer-defined Prefixes option. This prompt is added as one of the sub-prompts of ISDN = YES. It is added at the end of the ISDN sub-prompts. If CPFXS = YES, when constructing the Calling or Connected Line Identification, the prefixes are retrieved from the Customer Data Block via the PFX1 and PFX2 prompts in LD 15, as is currently done. This is the default response. If CPFXS = NO, when constructing the Calling or Connected Line Identification, the prefixes are retrieved from the Route Data Block via the HNTN and HLCL prompts in LD 16.
- - HNTN	0-9999	Home National Number. This number is similar to the PFX1 number prompted in LD 15. It is added to this overlay so that this prefix can be configured on a route basis as required in some countries (e.g., Italy). As is the case with PFX1, the HNTN prefix can be from one-to-four digits long. This prompt is displayed only if CPFXS = NO. If only a <CR> is entered, this prompt keeps its previous configuration. If no value was configured previously, no value will be configured. Enter X to delete the digits.

HLCL	0-9999	Home Location Number. This number is similar to PFX2 number prompted in LD 15. It is added to this overlay so that this prefix can be configured on a route basis as required in some countries (e.g., Italy). As is the case with PFX2, the HLCL prefix can be from one-to-four digits long. This prompt is displayed only if CPFXS = NO. If only a <CR> is entered, this prompt keeps its previous configuration. If no value was configured previously, no value will be configured. Enter X to delete the digits.
ADDP	(NO) YES	If ADDP = NO, the Calling or Connected Party Number displayed is not modified. If ADDP = YES, the prefixes 0 (national) or 00 (international) are added to the Calling Party Number if the Type of Number (TON) is public.

LD 14 - Configure the PRI2 trunks for the E403 interface.

Prompt	Response	Description
REQ	NEW CHG	Add new data. Change existing data.
TYPE	TIE COT DID	TIE trunk data block. Central Office Trunk data block. Direct Inward Dialing trunk data block. Note: Must match TKTP defined in LD 16.
TN	l ch c u	Loop and channel for digital trunks, where: l = Previously defined PRI2 loop. ch = channel 1-30 cu = card and unit, for Option 11C.
CUST	xx	Customer number, as previously defined in LD 15. xx = 0-99 for Options 51C-81/81C. xx = 0-31 for Option 11C.
RTMB	0-511 1-510 0-127 1-510	Route number and member number, for Options 51C-81/81C. For Option 11C.
TGAR	0 - (1) - 31	Trunk Group Access Restriction The default of 1 automatically blocks direct access.

ISDN BRIT implementation

LD 16 - Configure a BRIT route for the E403 interface.

Prompt	Response	Description
REQ	CHG	Change existing data.
TYPE	RDB	Route Data Block.
CUST	xx	Customer number, as previously defined in LD 15. xx = 0-99 for Options 51C-81C. xx = 0-31 for Option 11C.
ROUT	0 - 511 0-127	Route Number for Options 51C-81C. Route Number for Option 11C.
TKTP	TIE DID COT	Trunk type. TIE. Direct Inward Dialing. Central Office Trunk.
DTRK	YES	Digital Trunk Route.
BRIP	NO	ISDN BRI packet handler route (NO is entered, since packet data is not required).
- DGTP	BRI	Digital trunk type.
ISDN	YES	Integrated Services Digital Network.
...		
- IFC	E403	EuroISDN Interface conforming to the ETS 300 403, or the extended ETS 300 102 version for country-specific interfaces.
- - CNTY	ETSI NET	Enter country pertaining to the E403 interface. If either ETSI or NET are entered, the interface is fully compliant with ETS 300 403. ETS 300 403 for the user side. ETS 300 403 for the network side. If any of the countries listed on the following page are entered, the interface functions with the extended ETS 300 102 capabilities.

	AUS DEN FIN GER ITA NOR POR SWE EIR DUT SWI BEL ESP UK FRA CIS	Austria Denmark Finland Germany Italy Norway Portugal Sweden Ireland Holland Switzerland Belgium Spain United Kingdom France Commonwealth of Independent States (Russia and the Ukraine).
...		
- RCAP	aaaa Xaaaa	Remote capabilities. Precede with an X to remove an existing configuration.
...		

EuroISDN Malicious Call Identification

LD 17 - Configure remote D-channel capability for EuroISDN interface for PRI and BRI.

Prompt	Response	Description
REQ	CHG	Change existing data.
TYPE	ADAN	Action Device and Number.
- ADAN	CHG DCH xx	Change D-channel.
...		
- IFC	EURO	EuroISDN interface for D-channel.
...		
RCAP	MCID	Add Malicious Call Identification as a new remote capability.

LD 16 - Configure remote capability for EuroISDN interface for PRI.

Prompt	Response	Description
REQ	CHG	Change existing data.
TYPE	RDB	Route Data Block.
...		
DTRK	YES	Digital Trunk Route.
- IFC	EURO	EuroISDN interface.
...		
RCAP	MCID	Add Malicious Call Identification as a new remote capability.

LD 16 - Configure remote capability for EuroISDN interface for BRI.

Prompt	Response	Description
REQ	CHG	Change existing data.
TYPE	RDB	Route Data Block.
...		
DTRK	YES	Digital Trunk Route.
...		
- DGTP	BRI	Basic Rate Interface Digital Trunk Type.
- IFC	EURO	EuroISDN interface.
RCAP	MCID	Add Malicious Call Identification as a new remote capability.

LD 15 - Modify system and software parameters.

Prompt	Response	Description
REQ:	CHG	Change existing data.
TYPE:	FTR	Customer Features and options.
...		
OPT	MCTA	Malicious Call Trace signal is allowed for attendants. MCTD = Malicious Call Trace is denied (default)
...		
MCDC	YES	Malicious Call DN/CLID printing allowed.

LD 10 - Configure analog (500/2500 type) sets.

Prompt	Response	Description
REQ:	CHG	Change.
TYPE:	500	Telephone type.
TN	c u	Terminal Number.
...		
CLS	MCTA	Malicious Call Trace allowed. (MCTD) = Malicious Call Trace denied.

LD 11 - Configure Meridian 1 proprietary sets.

Prompt	Response	Description
REQ:	CHG	Change existing data.
TYPE:	xxxx	Telephone type, where: xxxx = SL1, 2006, 2008, 2009, 2016, 2018, 2112, 2216, 2317, 2616, 3000
TN	c u	Terminal Number.
...		
CLS	MCTA	Malicious Call Trace allowed. MCTD = Malicious Call Trace denied (default).
KEY	xx TRC	Trace key number.

LD 12 - Attendant Consoles.

Prompt	Response	Description
REQ	CHG	Change existing data.
TYPE	xxxx	Type of Attendant Console.
TN	c u	Terminal Number.
...		
CLS	MCTA	Malicious Call Trace allowed. MCTD = Malicious Call Trace denied (default).
KEY	xx TRC	Trace key number.

LD 57 - Configure Flexible Feature Codes for Analog (500/2500 type) sets.

Prompt	Response	Description
REQ	NEW CHG	Add new data. Change existing data
TYPE	FFC	Flexible Feature Code data block.
CUST	xx	Customer number.
...		
CODE	MTRC	Malicious Call Trace code
- MTRC	xxxx	Enter Flexible Feature Code for Malicious Call Trace.

LD 16 - Configure Malicious Call Trace Timer and Tandem Delay.

Note: This configuration is only required to handle Malicious Call Trace during call disconnection.

Prompt	Response	Description
REQ	CHG	Change existing data.
TYPE	RDB	Route Data Block.
...		
MCTS	YES	Malicious Call Trace Signal. (NO) = default.
- MCTM	(0) - 30	Malicious Call Trace Request Timer ID in seconds.
- MTND	YES	Malicious Call Trace disconnect delay for tandem calls on AXE-10 Australia and EuroISDN.

EuroISDN Trunk - Network Side

The following steps are required to configure the EuroISDN Trunk - Network Side feature:

For ISDN PRI access:

- In LD 17, for the network side EuroISDN Trunk - Network Side connectivity, configure the D-Channel for the network side EuroISDN ETSI protocol; and
- In LD 16, for the Bearer Capability-Based Routing capability, configure the route for the call type (voice, voice with 3.1 KHz audio, data, data with 3.1 KHz audio, or both voice and data with 3.1 KHz audio).

For ISDN BRI access:

- In LD 16, for the network side EuroISDN Trunk - Network Side connectivity, configure the route for the network side EuroISDN ETSI protocol.

For ISDN PRI access:

LD 17 — For the network side EuroISDN Trunk - Network Side connectivity, configure the D-Channel for the network side EuroISDN ETSI protocol.

Prompt	Response	Description
REQ	CHG	Change the existing data.
TYPE	ADAN	Action device and number.
- ADAN	CHG DCH x	Change a specified D-Channel, where x = 0-15.
- CTYP	MSDL	Card type, where MSDL = Downloadable D-Channel Daughterboard.
...		
- IFC	EURO	Interface type. Enter EURO for EuroISDN.
- - CNTY	NET	Country pertaining to EuroISDN interface. Enter NET for network side.
...		

For ISDN BRI access:

LD 16 — Configure the ISDN BRI link for the network side EuroISDN ETSI protocol.

Prompt	Response	Description
REQ	CHG	Change the existing data.
TYPE	RDB	Route Data Block.
CUST	xx	Customer number.
ROUT	0-511	Route number.
...		
DTRK	YES	Digital trunk.
...		
- IFC	EURO	Interface type. Enter EURO for EuroISDN.
- - CNTY	NET	Country pertaining to EuroISDN interface. Enter NET for network side.
...		

LD 16 — For an ISDN or DPNSS1/DASS2 trunk, configure the route for the Bearer Capability-Based Routing call type.

Prompt	Response	Description
REQ	CHG	Change the existing data.
TYPE	RDB	Route Data Block.
CUST	xx	Customer number.
ROUT	0-511	Route number.
TKTP		Supported trunk type.
	CO	Central Office trunk.
	DID	Direct Inward Dial trunk.
	TIE	TIE trunk.
	IDA	Integrated Digital Access trunk.
...		
DTRK	YES	Digital trunk.
...		
- DSEL		Data Selection.
	(VOD)	Route is for both voice and data
	VCE	Route is voice only
	DTA	Route is data only
	3VCE	Route is voice and 3.1 KHz
	3DTA	Route is data and 3.1 KHz
	TDN	Transparent Data Network

Flexible Numbering Plan

To configure the operation of the Flexible Numbering Plan feature, complete the following tasks.

To enable or disable Vacant Number Routing, respond to the following prompts in overlay 15.

VNR : This specifies that Vacant Number Routing is enabled or disabled. Respond to this prompt with YES to enable, or NO to disable.

RLI: This specifies the Route List Index to be used for Vacant Number Routing. Respond to this prompt with the index number, from 0 to 999.

CDPL: This specifies the (flexible) length of VNR CDP digits to be expected. Respond to this prompt with the expected digit length, from 0 to 10. CDPL should be set to the length of the longest number in use. (The default is 10.)

LOCL: This specifies the flexible length of VNR LOC digits to be expected (digit length from 0 to 10). LOCL should be set to the length of the longest number in use. (The default is 10.)

Note: The RLI, CDPL and LOCL prompts do not appear if VNR is set to NO.

NIT: This specifies the value of the Network Interdigit Timer. Respond to this prompt with the setting for the timer, from 2 to 8 seconds. (The default is 8.)

To configure routing, respond to the following prompts in overlay 86.

RLI: This specifies the Route List Index to be accessed. Respond to this prompt with the index number, from 0 to 999.

DMI: This specifies the Digit Manipulation Index to be used. Respond to this prompt with the index number, from 0 to 999.

FSNS: This specifies the Free Special Number Screening Index. Respond to this prompt with the index number, from 0 to 255. (The default index number is 0.)

AC1: This specifies Access Code 1. Enter the 1- to 4-digit network access code at this prompt.

AC2: This specifies Access Code 2. Enter the 1- to 4-digit network access code at this prompt.

Respond to the following prompts in overlay 87.

DMI: This specifies the Digit Manipulation Index. Respond to this prompt with the index number, from 0 to 999.

RLI: This specifies the Route List Index to be accessed. It must be set separately for the distant steering code and trunk steering code (under TYPE=DSC and TYPE=TSC respectively). Respond to this prompt with the index number, from 0 to 999.

FLEN: This specifies the length (flexible) of digits to be expected. Respond to this prompt with the expected digit length, from 0 to 16 for Trunk Steering Code, and from 0 to 10 for Distant Steering Code. (The default is 0.)

DSP: This specifies that either the Local Steering Code (LSC), Home Location Code (HLOC) or Directory Number (DN) is to be used for the Calling Station ID, in Group Dialing Plan calls. Respond to this prompt with either LSC, LOC or DN.

FSNS: This specifies the Free Special Number Screening Index (prompted under TYPE=FSNS). Respond to this prompt with the index number, from 1 to 255.

SPN: This specifies the special-number code to be screened. Respond to this prompt with the 1- to 11-digit code to be used.

XXX: This specifies whether routing codes are to be denied or allowed. Respond to this prompt with either DENY to deny these codes, or ALLOW to allow them.

DENY: This is prompted if the XXX prompt is set to DENY. Respond to it with the routing codes or range of routing codes to be denied.

ALLOW: This is prompted if the XXX prompt is set to ALLOW. Respond to it with the routing codes or range of routing codes to be allowed.

To configure translation-table data, respond to the following prompts in overlay 90.

RLI: This specifies the Route List Index to be used. Respond to this for each of TYPE=LOC, TYPE=NPA, TYPE=NXN, TYPE=SPN. Respond to this prompt with the index number, from 0 to 999.

FLEN: This specifies the length (flexible) of digits to be expected. Respond to this prompt with the expected digit length, from 0 to 16 for Special Number, and from 0 to 10 for Location Code. (The default is 0.)

Flexible Numbering Plan Enhancement

LD 15 – Enable/disable the Flexible Numbering Plan feature.

Prompt	Response	Description
REQ:	CHG	Change existing data.
TYPE:	NET	Networking data.
CUST	xx	Customer number.
...		
AC2	NPA NXX INTL SPN LOC	Access Code 2. E.164 National E.164 Subscriber International Special Number Location Code
FNPN	YES	Enable the Flexible Numbering Plan feature (default).
ISDN	(NO) YES	ISDN option.
...		
VNR	(NO) YES	Vacant Number Routing enabled (disabled). VNR is only prompted when FNP = YES. When FNP = NO, VNR is automatically set to NO and is, therefore, restricted.
- RLI	0-999	Route List Index. Enter the route list, defined in LD 86, to be used by Vacant Number Routing.
- FLEN	1-(16)	Flexible length of digits expected.
- CDPL	1-(10)	Coordinated Dialing Plan Length. Enter the maximum number of Coordinated Dialing Plan (CDP) digits expected by Vacant Number Routing.
- LOCL	1-(10)	Location Code Length. Enter the maximum number of Location (LOC) digits expected by Vacant Number Routing.

Idle Extension Notification

In overlay 12, designate the Semi-Automatic Camp-On key. This key is also used with the Idle Extension Notification feature.

Prompt	Response	Description
KEY	xx SACP	Key number, Semi-Automatic Camp-On/Idle Extension Notification

Use overlay 15 (for customer data) or overlay 93 (for tenant data) to configure a new ICI key. When an extension that is under Idle Extension Notification supervision becomes idle, the recall is presented on this new ICI key at the attendant console.

Prompt	Response	Description
ICI	xx IEN	ICI number, Idle Extension Notification

Use overlay 16 to configure the Idle Extension Notification block timer.

Prompt	Response	Description
CNTL	YES	
TIMR	IENB 2-(5)-10	Idle Extension Notification block timer, in minutes. The default value is five minutes. Both SACP and NAS packages must be equipped.

Incoming Trunk Programmable Calling Line Identification

Use the following procedure to configure Incoming Trunk Programmable Calling Line Identification.

Step 1 - Use **Overlay 16** to configure the incoming route.

Prompt	Response	Description
REQ	NEW, or CHG	New, or Change.
TYPE	RDB	Route data block.
...		
BILN	(NO), YES	Billing number (is not) is required.
BLN	1-(10)-16	Billing number length. The default is 10 digits.
BNUM	x...x	Billing number (1 to 16 digits).
BDSP	(NO), YES	(Do not) display Billing Number.

Step 2 - Use **Overlay 16** to configure outgoing route.

Prompt	Response	Description
REQ	CHG	Change.
TYPE	RDB	Route data block.
...		
ISDN	(NO), YES	ISDN route must be YES.
...		
IFC	xxxx	Select the appropriate interface type for route (cannot be SL1 or IDA).
SBN	(NO), YES	(Do not) send billing number. Must be set to YES.

International 1.5/2.0 Mb/s Gateway

Use this procedure to configure International 1.5/2.0 Mb/s Gateway on an Option 11 system.

Pad and coding configuration

LD 14 – Select the pad category table and coding law for the 1.5 Mb/s PRI/DTI trunks used in the gateway.

Prompt	Response	Description
REQ	CHG	Change.
PDCA	0-16	pad category table. Enter 0 if no pad value is to be used.
PCML	M, A	coding law (M μ - or A-law)

Pad configuration

LD 73 – Set the pad category values for Meridian Digital sets, for the 2 Mb/s PRI/DTI XPE Analog Public Exchange trunks, XPE Analog Tie trunks, and digital pads for trunk connections between 1.5 Mb/s PRI/DTI and 2 Mb/s PRI/DTI trunks.

Prompt	Response	Description
REQ	CHG	Change.
TYPE	PRI, DTI	1.5 Mb/s PRI or DTI
DSET	x y	pad category value for the new Meridian Digital Set
PRI2	x y	pad category value for the 2 Mb/s PRI/DTI trunks
XUT	x y	pad category value for the XPE Analog Public Exchange trunks
XEM	x y	pad category value for the XPE Analog TIE trunks
PRI	x y	pad category value for the 1.5 Mb/s PRI/DTI trunks

Intercept Computer Dial from Directory

LD 15 – Enable Intercept Computer Dial from Directory.

Prompt	Response	Description
REQ	NEW, CHG	New, or change.
TYPE	CDB ICP_DATA	Customer data block. Release 21 gate opener.
...		
- ICP	(NO), YES	Intercept Computer.
...		
- ICPD	(0)-9	ICP Padding Digit
- ICTD	(NO), YES	Intercept Computer Treatment Dial from directory. This prompt allows an intercept attendant position to dial an extension DN from the Intercept Computer Terminal. It is only prompted if ICP is set to "YES".

ISDN PRI Central Office connectivity

This feature allows the interconnection of the Meridian 1 with public exchanges in the international markets that support the following protocols:

- AXE-10 Australia (non-Asia Pacific ISDN Connectivity)
- AXE-10 (Sweden)
- NEAX-61 (New Zealand) (non-Asia Pacific ISDN Connectivity)
- Numeris VN3 (France)
- SwissNet (Switzerland)
- SYS-12 (Norway)
- 1TR6 (Germany)
- EuroISDN
- Japan D70 (non-Asia Pacific ISDN Connectivity)
- Asia-Pacific, consisting of:
 - Australia (private or alternative carrier)
 - China
 - Hong Kong
 - India
 - Indonesia
 - Japan
 - Malaysia
 - New Zealand
 - Philippines
 - Singapore
 - Taiwan
 - Thailand

These implementation procedures for these connectivities are described in the sections which follow.

AXE-10 (Australia), non-Asia Pacific connectivity

LD 17 – Define system support for the AXE-10 Australia connectivity.

Prompt	Response	Description
REQ	CHG	Change existing data block.
TYPE	CFN	Configuration record.
- ADAN	CHG DCH x	Change a D-channel on logical port 0-15.
...		
- IFC	AXEA	Ericsson AXE-10 for Australia.
- CNEG	(1) 2	The required channel negotiation option, either exclusive (the default) or preferred.

LD 16 – Configure the routes for the AXE-10 Australia connectivity.

Prompt	Response	Description
REQ	NEW, CHG	New, or change.
TYPE	RDB	Route Data Block.
CUST	xx	Customer number associated with this route.
ROUT	xx	Route number.
...		
IFC	AXEA	Ericsson AXE-10 for Australia.

AXE-10 (Sweden), non-Asia Pacific connectivity

This feature allows ISDN PRI connectivity between the Meridian 1 and the

LD 17 – Define the system support for the AXE-10 Sweden connectivity.

Prompt	Response	Description
REQ	CHG	Change existing data block.
TYPE	CFN	Configuration record.
- ADAN	CHG DCH x	Change a D-channel on logical port 0-15.
...		
- IFC	AXES	Ericsson AXE-10 for Sweden.
- CNEG	(1) 2	The required channel negotiation option, either exclusive (the default) or preferred.

LD 16 – Configure the routes for the AXE-10 Sweden connectivity.

Prompt	Response	Description
REQ	NEW, CHG	New, or change.
TYPE	RDB	Route Data Block.
CUST	xx	Customer number associated with this route.
ROUT	xx	Route number.
...		
IFC	AXES	Ericsson AXE-10 for Sweden.

NEAX-61 (New Zealand), non-Asia Pacific connectivity

LD 17 – Configure an ISDN interface supporting the Telecom New Zealand ISDN protocol.

Prompt	Response	Description
REQ	CHG	Change.
TYPE	CFN	Configuration data block.
- ADAN	CHG DCH x	Change a D-channel on logical port 0-15 (all other machine types).
- CTYP	MSDL	TCNZ requires the D-channel be defined on an MSDL.
- CDNO	1-9	The card number of the Downloadable D-Channel Daughterboard.
- DES	aaaaaa	Alphanumeric designator.
- DPNS	(NO) YES	Digital Private Network Signaling.
- USR	PRI	D-channel mode. This D-channel is used for Primary Rate only.
- IFC	TCNZ	Telecom New Zealand.
-DCHL	0-159 (0)-3	TCNZ loop and interface ID – non Option 11.
	1-10 (0)-3	TCNZ loop and interface ID – Option 11.
- PRI2	0-159 0-3	Secondary PRI2 loops for nB+D, plus sequence. Note that the D-channel is not necessarily on IFC ID 0; this is set by service change.
- OTBF	1-(32)-127	Number of output request buffers.
- CNEG	(1) 2	Options for outgoing Channel Negotiation. Option 1: Channel is non-negotiable. Option 2: The Channel listed is preferred, but negotiable.
- RCAP	aaa	Remote DCH capabilities, as applicable.
- OVLS	YES	Allow overlap sending.
- - OVLT	1-8, 0	Time the sending side has to wait between INFO messages. "0" means send immediately.

- TIMR	(NO) YES	Change protocol timer value.
- - T310	10-60	Timer used to determine how long the switch can wait for the response message when the QSIG outgoing call is in the outgoing call processing state.
- - INC_T306	0,(2),...,T306 (T306 is the duration of the T306 Network timer in seconds)	Variable timer for received DISCONNECT message on incoming calls, allowing in-band tone to be heard when sent by the network. The network will stop sending tone after T306 expires, so the maximum time will be T306. Entered in two-second increments. TCNZ interface: T306 = 30 sec.
- - OUT_T306	0,2,...,(T306) (T306 is the duration of the T306 Network timer in seconds)	Variable timer for received DISCONNECT message on outgoing calls, allowing in-band tone to be heard when sent by the network. The network will stop sending tone after T306 expires, so the maximum time will be T306. Entered in two-second increments. TCNZ interface: T306 = 30 sec.
- LAPD	(NO) YES	Change the Link Access Protocol for D-channel parameters.

LD 16 – Configure a COT, DID/DOD, or TIE route using the Telecom New Zealand (TCNZ) protocol.

Prompt	Response	Description
REQ	NEW CHG	New, or change.
TYPE	RDB	Route Data Block.
CUST	xx	Customer number.
ROUT	xx	Route number.
...		
DGTP	PRI2	Type of digital route.
ISDN	YES	Default for a PRI2 route.
MODE	PRA	Primary Rate Access.
...		
IFC	TCNZ	Telecom New Zealand (NEAX-61).
...		
CTYP	UKWN, INTL, NPA, NXX	Only these call types are supported for TCNZ interface TIE routes.
...		
OPD	(YES) NO	Outpulsed digits in CDR.
CDRX	(YES) NO	X records produced on call transfer.
NDP	INC 1-32 EXC 1-32	Number of printed digits; output the first 1-32 digits. Suppress the last 1-32 digits. If REQ = NEW, output all digits and suppress none, if REQ = CHG, leave unchanged.
CDRY	(NO) YES	CDR Public Network Feature Invoke records will (will not) be generated when the feature is invoked. Default is NO.
OPA	(NO) YES	Generate CDR records for PPM pulses.
...		

MCTS	(NO) YES	Send a Malicious Call Trace request to the public network if MCT is invoked in the private network on current switch.

Numeris VN3 (France)

LD 17 – Define system support for NUMERIS.

Prompt	Response	Description
REQ	CHG	Change existing data block.
TYPE	CFN	Configuration record.
- ADAN	CHG DCH x	Change a D-channel on logical port 0-15 (all other machine types).
...		
- IFC	NUME	Numeris for France.
- CNEG	(1) 2	The required channel negotiation option, either exclusive (the default) or preferred.

LD 16 – Configure routes for NUMERIS.

Prompt	Response	Description
REQ	NEW, CHG	New, or change
TYPE	RDB	Route Data Block.
CUST	xx	Customer number associated with this route
ROUT	xx	Route number.
...		
IFC	NUME	Numeris for France

LD 73 – Define the alarm handler in the firmware, disable or enable the grade of service processing in the software, and define the number of seconds before the firmware sends an ERR message.

Prompt	Response	Description
REQ	NEW, CHG	New, or change.
TYPE	PRI2	2.0 Mbit Primary Rate Interface data block.
...		
ALRM	REG	Regular firmware alarm handler (RAI transmission is controlled by software).
	ALT	Alternate firmware alarm handler (Immediate transmission of RAI by firmware).
...		
NOOS	(NO)	Grade of service feat.
	YES	Enables current grade of service feat.
...		
RATS	1-(10)-15	The number of consecutive seconds the firmware has to check and validate error rate condition.

SwissNet 2 (Switzerland)

LD 17 – Define the ISDN PRI to SwissNet connectivity.

Prompt	Response	Description
REQ	CHG	Change.
TYPE	CFN	Configuration Record.
- ADAN	CHG DCH x	Change a D-channel on logical port 0-15 (all other machine types).
...	...	...
- IFC	SWIS	Interface type for SwissNet connectivity.
- CNEG	(1) 2	The required channel negotiation option, either exclusive (the default) or preferred.

LD 16 – Define the ISDN PRI to SwissNet connectivity.

Prompt	Response	Description
REQ	CHG	Change.
TYPE	RBD	Route Data Block.
CUST	xx	Customer number.
ROUT	xx	Route number.
...	...	...
IFC	SWIS	Interface type for SwissNet connectivity.

SYS-12 (Norway)**LD 17** – Define the ISDN PRI to SYS-12 connectivity.

Prompt	Response	Description
REQ	CHG	Change.
TYPE	CFN	Type of data block.
- ADAN	CHG DCH x	Change a D-channel on logical port 0-15 (all other machine types).
...	...	...
- IFC	SS12	Interface type for SYS-12 connectivity.
- CNEG	(1) 2	The required channel negotiation option, either exclusive (the default) or preferred.

LD 16 – Define the ISDN PRI to SYS-12 connectivity.

Prompt	Response	Description
REQ	NEW, CHG	New, or change.
TYPE	RDB	Route Data Block.
CUST	xx	Customer number.
ROUT	xx	Route number.
...	...	...
IFC	SS12	Interface type for SYS-12 connectivity.

1TR6 (Germany)

LD 17 – Define the ISDN PRI to 1TR6 connectivity.

Prompt	Response	Description
REQ	CHG	Change.
TYPE	CFN	Configuration record.
- ADAN	CHG DCH xx	Change a D-channel on logical port 0-15 (all other machine types).
...	...	...
- IFC	1TR6	Interface type for 1TR6 connectivity.
- CNEG	(1) 2	The required channel negotiation option, either exclusive (the default) or preferred.

LD 16 – Define the ISDN PRI to 1TR6 connectivity.

Prompt	Response	Description
REQ	NEW, CHG	New, or change.
TYPE	RDB	Route Data Block.
CUST	xx	Customer number.
ROUT	xx	Route number.
...	...	...
IFC	1TR6	Interface type for 1TR6 connectivity.

EuroISDN

LD 15 – Configure the Optional Sending of Forwarding CLID functionality (in response to the OCLI prompt).

Prompt	Response	Description
REQ:	CHG	Change existing data.
TYPE:	NET	Networking data.
CUST	xx	Customer number.
...		
ISDN	YES	Integrated Services Digital Network.
PNI	(0)-32700	Private Network Identifier.
...		
SATD	0-(1)-5	Satellite Delays.
OCLI	(NO)	NO = No manipulation is done on outgoing CLID for calls forwarded over EuroISDN links. EXT = The last forwarding DN is sent as CLID information for incoming calls over a standard ISDN (EuroISDN, APAC, NI-2) DID or CO link to a gateway Meridian 1 node, and redirected back over the standard ISDN (EuroISDN, APAC, NI-2) DID or CO link. ALL = The same as for EXT. Moreover, the last forwarding DN is sent as CLID information for internal calls from a local set or over ISDN TIE trunks, except for DPNSS1 calls which are redirected from a gateway node over a standard ISDN (EuroISDN, APAC, NI-2) DID or CO link. If the redirection does not occur at the gateway node, the redirecting information number, if present in the incoming SETUP message received at the gateway node from the private network, is sent as CLID.
	EXT	
	ALL	
....		

LD 17 – Create a new D-channel for the EuroISDN interface.

Prompt	Response	Description
REQ	CHG	Change existing data.
TYPE	ADAN	Action Device and Number.
- ADAN	NEW DCH x	New D-channel at port number x.
- CTYP	MSDL	Card type, where MSDL = the Downloadable D-Channel Daughterboard.
- PORT	1	The port number of the Downloadable D-Channel Daughterboard.
- USR	PRI	D-channel for ISDN PRI only. Precede with X to remove.
- IFC		DCH interface type.
	EURO	EURO = EuroISDN
- - CNTY		Enter country pertaining to EuroISDN interface.
	AUS	Austria
	DEN	Denmark
	(ETSI)	ETS 300-102 basic protocol
	FIN	Finland
	GER	Germany
	ITA	Italy
	NOR	Norway
	POR	Portugal
	SWE	Sweden
	EIR	Ireland
	DUT	Holland
	SWI	Switzerland
	BEL	Belgium
	ESP	Spain
	UK	United Kingdom
	FRA	France
	CIS	Commonwealth of Independent States (Russia and the Ukraine).
PINX_CUST	xx	The customer number used for the DN address.
...		
DCHL	0-159	The PRI loop number for the D-Channel.

- - CNEG	(1) 2	Channel Negotiation option. (1) = Channel is indicated and no alternative is acceptable. This is the default value for all EuroISDN Interfaces except FRA. 2 = Channel is indicated and any alternative is acceptable. This is the default value for the FRA interface.
- RCAP	COLP XCOL	Enter COLP to support Connected Line Identification Presentation as a remote capability. This is the default value for the ESIG, ISIG, NI2, and EURO interfaces. Enter XCOL to remove Connect Line Identification Presentation as a remote capability. This is the default value for the APAC, AUS, EIR, DEUT, ESP, BEL, and FRA interfaces.

LD 16 – Configure new DID routes (the same responses to the IFC and CNTY prompts, as were used in LD 17, must be entered).

Prompt	Response	Description
REQ	NEW CHG	Add new data. Change or delete existing data.
TYPE	RDB	Route Data Block.
CUST	xx	Customer number.
ROUT	0 - 511 0-127	Route Number. For Option 11C.
TKTP	DID COT	Trunk type. DID = Direct Inward Dialing COT = Central Office.
...		
DTRK	YES	Digital trunk route.
- DGTP	PRI2 BRI	Digital trunk type for route.
...		

- IFC	EURO	Interface type. EURO = EuroISDN.
- - CNTY	AUS DEN (ETSI) FIN GER ITA NOR POR SWE EIR DUT SWI BEL ESP UK FRA CIS	Enter country pertaining to EuroISDN and Asia Pacific interface type. Austria Denmark ETS 300-102 basic protocol Finland Germany Italy Norway Portugal Sweden Ireland Holland Switzerland Belgium Spain United Kingdom France Commonwealth of Independent States (Russia and the Ukraine).
ICOG	IAO OGT ICT	Incoming and Outgoing trunk. Outgoing trunk only. Incoming trunk only.
ACOD	x..x	Access code for this trunk route.

- CLID	OPT0	Prefix = 0, for North American dialing plan. This is the default value for ESIG and ISIG interfaces.
	OPT1	Prefix = 1, for international PFXs in CLID, any numbering type supported. This is the default value for all EuroISDN interfaces.
	OPT2	Prefix = 2, for international PFXs in CLID, CCITT numbering type supported: UKWN, INTL, NPA, and NXX. Default value for CO/DID routes for the Telecom New Zealand interface.
	OPT3	Prefix = 3, for international PFXs in CLID, only NXX number type supported. Default value for TIE routes for the Telecom New Zealand interface.
	OPT4	For international COs, if the call originates from a CO trunk type, add nothing. Otherwise, add PFX1 and PFX2. This is the default value for the Hong Kong, Singapore, and Thailand interfaces.
	OPT5	This is the same as OPT4, except it supports a maximum of 10 digits in the CLID. This is the default value for the Austrian interface.
- PROG		Progress Signal.
	NCHG	Send PROGRESS (default value for all interfaces, except Austria).
	MALE	Send ALERT after CALL PROCEEDING.
	MCON	Send CONNECT after CALL PROCEEDING (this is the default value for the Austrian interface).
...		
- RCAP	COLP	Enter COLP to support Connected Line Identification Presentation as a remote capability. This is the default value for the ESIG, ISIG, NI2, and EURO interfaces.
	XCOL	Enter XCOL to remove Connect Line Identification Presentation as a remote capability. This is the default value for the APAC, AUS, EIR, DEUT, ESP, BEL, and FRA interfaces.
...		

- CPFXS	(YES) NO	Customer-defined Prefixes option. This prompt is added as one of the sub-prompts of ISDN = YES. It is added at the end of the ISDN sub-prompts. If CPFXS = YES, when constructing the Calling or Connected Line Identification, the prefixes are retrieved from the Customer Data Block via the PFX1 and PFX2 prompts in LD 15, as is currently done. This is the default response. If CPFXS = NO, when constructing the Calling or Connected Line Identification, the prefixes are retrieved from the Route Data Block via the HNTN and HLCL prompts in LD 16.
- - HNTN	0-9999	Home National Number. This number is similar to the PFX1 number prompted in LD 15. It is added to this overlay so that this prefix can be configured on a route basis as required in some countries (e.g., Italy). As is the case with PFX1, the HNTN prefix can be from one-to-four digits long. This prompt is displayed only if CPFXS = NO. If only a <CR> is entered, this prompt keeps its previous configuration. If no value was configured previously, no value will be configured. Enter X to delete the digits.
HLCL	0-9999	Home Location Number. This number is similar to PFX2 number prompted in LD 15. It is added to this overlay so that this prefix can be configured on a route basis as required in some countries (e.g., Italy). As is the case with PFX2, the HLCL prefix can be from one-to-four digits long. This prompt is displayed only if CPFXS = NO. If only a <CR> is entered, this prompt keeps its previous configuration. If no value was configured previously, no value will be configured. Enter X to delete the digits.
ADDP	(NO) YES	If ADDP = NO, the Calling or Connected Party Number displayed is not modified. If ADDP = YES, the prefixes 0 (national) or 00 (international) are added to the Calling Party Number if the Type of Number (TON) is public.

LD 17 – Configure the Numbering Plan Identification (NPI) and Type of Number (TON) fields to be included in the CDR tickets, in addition to the CLID, for EuroISDN calls (this is automatically done by entering YES in response to the CLID prompt).

Prompt	Response	Description
REQ	CHG	Change existing data.
TYPE	PARM	Change system parameters.
...		
- FCDR	NEW	Format for Call Detail Recording. Enter NEW for new format.
...		
- CLID	(NO) YES	Enter YES to include the TON and NPI fields in the CDR ticket, in addition to the CLID.
...		

Japan D70, non-Asia Pacific connectivity

LD 17 – Define the ISDN PRI to D70 connectivity.

Prompt	Response	Description
REQ	CHG	Change.
TYPE	CFN	Configuration record.
- ADAN	CHG DCH x	Change a D-channel on logical port 0-15 (all other machine types).
...	...	...
- IFC	D70	Interface type for Japan D70 connectivity.
- CNEG	(1) 2	The required channel negotiation option, either exclusive (the default) or preferred.

LD 16 – Define the ISDN PRI to D70 connectivity.

Prompt	Response	Description
REQ	NEW, CHG	New, or change.
TYPE	RDB	Route Data Block.
CUST	xx	Customer number.
ROUT	xxx	Route number.
...	...	...
IFC	D70	Interface type for Japan D70 connectivity.

Asia Pacific ISDN PRI Central Office Connectivity

LD 15 — Configure the Customer Data Block for Asia Pacific with Calling Line Identification.

Prompt	Response	Description
REQ:	CHG	Change existing data.
TYPE:	NET	Networking data (if REQ = CHG only.)
CUST	xx	Customer number. xx = 0-99 for Options 51C-81C. xx = 0-31 for Option 11C.
OPT	a...a	Options.
AC2	aaaa	Access Code 2 as defined in LD 86 (aaaa = NPA, NXX, INTL, SPN, or LOC).
...		
CLID	YES	Configure CLID option for Asia Pacific.
- SIZE	0-(256)-4000	The maximum number of CLID entries needed for a customer. Note: Do not define a size much larger than actually needed. This entry may be increased or decreased as required.
- INTL	0-9999	Country code, for international number.

- ENTRY	aaaa Xaaaa Xaaaa Xbbbb	aaaa = CLID entry to be configured. Xaaaa = CLID entry to be deleted. Xaaaa Xbbbb = CLID entries to be deleted. aaaa and bbbb must be a value between 0 and (SIZE-1). The ENTRY prompt is repeated until <CR> is entered as a response. If REQ = CHG, as many entries as needed may be created, changed or deleted. The action for the entry is saved to system memory after the CLID entry has been completely configured. That is, after the LSC prompt has been answered. If a new CLID entry is created, or an existing CLID entry is changed, the message "ENTRY aaaa SAVED" is displayed after the LSC prompt. If a CLID entry or CLID entries is/are deleted, the message "ENTRY aaaa DELETED" or "ENTRIES aaaa-bbbb DELETED" is displayed after the LSC prompt.
- - HNTN	0-999999	National code for home national number (1-6 digits).
- - HLCL	0-999..9	Local code for home local number or Listed Directory Number (1-12 digits).
- - DIDN	(YES) NO SRCH	How to use the DN as a DID when constructing a CLID national or local number. The CLID is constructed using the digits defined in HLCL followed by the DN of the active key (default). Construct the CLID using the digits defined in HLCL. Search on the set, from key 0 - upwards, to find a CLID entry which has the DIDN set to YES. Use the found CLID to construct the local number.
- - HLOC	0-9999999	Home location code (ESN), 1-7 digits.
- - LSC	0-9999999	Local steering code, 1-7 digits.
ISDN	YES	Integrated Services Digital Network.
- PNI	1-327000	Private Network Identifier.
ENTRY aaaa SAVED ENTRY aaaa DELETED ENTRIES aaaa-bbbb DELETED		Displayed message appears after each entry is input.

LD 17 – Configure the Asia Pacific D-channel.

Prompt	Response	Description
REQ	CHG	Change existing data.
TYPE	CFN	Configuration data block.
- ADAN	NEW DCH xx	Add a D-channel on logical port 0-63 (51C-81C).
	NEW DCH xx	Add a D-channel on logical port 0-15 (Option 11C).
- CTYP	MSDL	Card type where: MSDL = The NT6D80 Multi-purpose Serial Data Link or the NTB51AA Downloadable D-Channel Daughterboard for Options 51C-81C. MSDL = The NTB51BA Downloadable D-Channel Daughterboard for Option 11C.
- GRP	0-4	Network group number for Option 81/81C systems.
- DNUM	0-15	Device number for I/O ports. All ports on the MSDL card share the same DNUM. The MSDL card address settings must match the DNUM value.
- CDNO	1-9	The card number of the Downloadable D-Channel Daughterboard, for Option 11C.
- PORT	0-3 1	Port number on MSDL cards. For Option 11C.
- DES	a...a	Designator. DES is used to identify the link and can be up to 16 alphanumeric characters: 0-9, and upper case A-Z. Characters "*" and "#" are not allowed.
- USR	PRI	This D-channel is used for Primary Rate only.
- IFC	APAC	Asia Pacific ISDN interface.

- - CNTY	AUST CHNA HKNG INDI INDO JAPN MSIA PHLP SING TAIW TCNZ THAI	Country associated with the APAC interface. Australia. China. Hong Kong. India. Indonesia. Japan. Malaysia. Philippines. Singapore. Taiwan. New Zealand. Thailand.
- - ISDN_MCNT	60-(300)-350	Layer 3 call control message count per 5 second time interval.
CLID	OPT4	OPT4 is the default for all Asia Pacific interface.
PROG	NCHG MALE MCON	Progress signal. No Change. This is the default for all Asia Pacific interfaces except Singapore, Japan, and Australia. Alert message. Connect message. This is the default for the Australia interface. Note: The PROG prompt should not be configured for Japan and Singapore, since these countries do not support the Progress signal.
- DCHL	0-159 1-9	PRI/PRI2 loop number for D-channel (Options 51C - 81C). PRI/PRI2 loop number for D-channel (Option 11C).
- CNEG	(1) 2	Channel negotiation option. Channel is indicated and no alternative acceptable, exclusive. Channel is indicated and any alternative acceptable, preferred. Note: for the Singapore interface, CNEG must be set to 1.
- RLS	xx	Software release of the far end switch. If the far end has an incompatible release, it prevents the sending of application messages.

- RCAP	(XCOL) COLP	Remote capabilities, to configure the Connected Line ID Presentation supplementary service. To remove COLP. CLID Presentation supported. This prompt appears if CNTY = Indonesia, India, Taiwan, and the Philippines.
...		
- OVLR	YES	Allow Overlap Receiving. This prompt applies only if CNTY = THAI, MSIA, INDI, INDO, or CHNA.
- - DIDD	(0)-15	Number of leading digits to delete from DID trunks.
- OVLS	(NO) YES	(Do not) allow Overlap Sending. Enter NO for Japan and Philippines. Enter YES for all other interfaces.
- - OVLT	(0)-8	Overlap Timer in seconds. This timer controls the interval between the sending of INFORMATION messages. "0," the default, means send immediately.
- TIMR	YES	Change programmable timers. Only supported for interfaces supporting one of the following timers.
- - T310	10-(10)(30)-60	Maximum time in seconds between an incoming CALL PROCEEDING message and the next incoming message. Not supported for Australia. Default values are as follows: CHNA = 30 seconds. TAIW = 30 seconds. PHLP = 10 seconds. HKNG = 10 seconds. INDI = 10 seconds. INDO = 10 seconds. JAPN = 10 seconds. MSIA = 10 seconds. SING = 10 seconds. TCNZ = 10 seconds. THAI = 10 seconds.

- - INC_T306	0-(2)-240	Variable timer for received DISCONNECT message on incoming calls allowing in-band tone to be heard when the network sends in-band tone. T306 is the duration of the network timer in seconds. The network will stop sending after T306 times out, so the maximum time will be T306. The value is stored in two-second increments, which are rounded up.
- - OUT_T306	0-(30)-240	Variable timer for received DISCONNECT message on outgoing calls allowing in-band tone to be heard when the network sends in-band tone. T306 is the duration of the network timer in seconds. The network will stop sending after T306 times out, so the maximum time will be T306. The value is stored in two-second increments, which are rounded up.
- LAPD	YES	Allows the changing of the layer 2 timer. NO = Do not allow the changing of the layer 2 timer (default).
...		
PARM	YES	Change system parameters.
FCDR	NEW	Use new CDR format (recommended for Japan).
OCAC	YES	Support the Original Carrier Access Code format. NO = Do not support the Original Carrier Access Code format (default).
MTRO	(MR) PPM	Message Registration (default). Periodic Pulse Metering.

LD 17 – Configure a Back-up D-Channel for Hong Kong.

Prompt	Response	Description
REQ	CHG	Change existing data.
TYPE	ADAN	Action device and number.
- ADAN	NEW BDCH xx	Add a Back-up D-Channel on logical port 0-63, for Options 51C-81/81C
	NEW BDCH xx	Add a Back-up D-Channel D-Channel on logical port 0-15, for Option 11C.
- PDCH	xx	Primary D-Channel associated with the Backup D-Channel. The value must be the same as the one entered for the Primary D-Channel. xx = 0-63 for Options 51C-81/81C. xx = 0-15 for Option 11C.
- CTYP	MSDL	Card type where: MSDL = The NT6D80 Multi-purpose Serial Data Link or the NTB51AA Downloadable D-Channel Daughterboard for Options 51C-81/81C. MSDL = The NTB51BA Downloadable D-Channel Daughterboard for Option 11C.
- GRP	0-4	Network group number for Option 81/81C systems.
- DNUM	0-15	Device number for I/O ports. All ports on the MSDL card share the same DNUM. The MSDL card address settings must match the DNUM value.
- CDNO	1-9	The card number of the Downloadable D-Channel Daughterboard, for Option 11C.
- PORT	0-3	Port number on the NT6D80 MSDL card, if the MSDL is used for D-Channel handling on large systems (Options 51C-81/81C).
...		

	0-1	Port number of the NTB51AA, if the NTB51AA is used for D-Channel handling on large systems (Options 51C, 61C, 81, 81C). Port 0 of the NTB51AA can only be defined to work with Loop 0 of the NT5D97AA DDP2 card, and Port 1 of the NTB51AA can only be defined to work with Loop 1 of the NT5D97AA. This relationship must be reflected in the BCHL prompt, which follows later (either 0 or 1 must be entered when specifying the loop number used by the Backup D-Channel).
- RCV	1	Port number of the NTB51BA, for Option 11C.
	YES	Auto-recovery to primary D-Channel.
	1-9	PRI/PRI2 loop number for D-channel (Option 11C).
...		
- USR	PRI	This D-Channel is used for Primary Rate only.
- IFC	APAC	Asia Pacific ISDN interface.
- - CNTY	HKNG	Hong Kong
...		
PROG	NCHG	Progress signal. NCHG = No Change. This is the default for all Asia Pacific interfaces except Australia (and Japan and Singapore, which do not support the Progress signal for PRI).
	MALE	MALE = Alert message.
	MCON	MCON = Connect message. This is the default for the Australia interface.
		Note: The PROG prompt should not be configured for Japan and Singapore, since these countries do not support the Progress signal.
- RCV	YES	Auto-recovery to primary D-Channel.
- BCHL	0-159	PRI2 loop number for the Back-up D-channel, for large systems (Options 51C-81/81C). If the NTB51AA is used for D-Channel handling, only loop 0 or 1 can be configured.

...	1-9	PRI/PRI2 loop number for D-channel (Option 11C).

LD 17 – Configure nB+D for Hong Kong, Japan, New Zealand, and Malaysia.

Note: The International nB+D (INBD) package 255 must be enabled to configure nB+D.

Prompt	Response	Description
REQ	CHG	Add existing data Change existing data.
TYPE	ADAN	Action device and number.
- ADAN	NEW DCH xx	Add a primary D-Channel on logical port 0-63, for Options 51C-81C.
	NEW DCH xx	Add a primary D-Channel D-Channel on logical port 0-15, for Option 11C.
- CTYP	MSDL	Card type where: MSDL = The NT6D80 Multi-purpose Serial Data Link or the NTB51AA Downloadable D-Channel Daughterboard for Options 51C-81/81C. MSDL = The NTB51BA Downloadable D-Channel Daughterboard for Option 11C.
- GRP	0-4	Network group number for Option 81/81C systems.
- DNUM	0-15	Device number for I/O ports. All ports on the MSDL card share the same DNUM. The MSDL card address settings must match the DNUM value.
	1-10	The card number of the Downloadable D-Channel Daughterboard, for Option 11C.
- PORT	0-3	Port number on the NT6D80 MSDL card, if the MSDL is used for D-Channel handling, for Options 51C-81/81C.

- DES	0-1	Port number of the NTB51AA, if the NTB51AA is used for D-Channel handling on large systems (Options 51C-81/81C). Port 0 of the NTB51AA can only be defined to work with Loop 0 of the NT5D97AA DDP2 card, and Port 1 of the NTB51AA can only be defined to work with Loop 1 of the NT5D97AA. This relationship must be reflected in the BCHL prompt, which follows later (either 0 or 1 must be entered when specifying the loop number used by the Backup D-Channel).
	1	Port number of the NTB51BA, for the small system (Option 11C).
	aaa...a	Designator. DES is used to identify the link and can be up to 16 alphanumeric characters: 0-9, and upper case A-Z. Characters "*" and "#" are not allowed.
	PRI	This D-Channel is used for Primary Rate only.
	APAC	Asia Pacific ISDN interface.
- USR	HKNG	Hong Kong.
- IFC	MSIA	Malaysia.
- - CNTY	JAPN	Japan.
...	TCNZ	New Zealand.
- DCHL	LOOP ID 0-159 (0)-15	Primary PRI loop number and interface identifier for the D-Channel (Options 51C-81/81C). Note: the INBD package 255 must be enabled to allow this range of entry. If the INBD package is not enabled, the allowable entry would be the PRI loop number (0-159 for Options 51C-81/81C, 1-9 for Option 11C). Also, the subsequent PRI prompt(s) would not be generated.
	LOOP ID 1-9 (0)-15	Primary PRI loop number and interface identifier for the D-Channel (Option 11C).

- PRI	LOOP ID 0-159 (0)-15	Secondary PRI loop number and interface identifier (Options 51C-81/81C). Prompted if INBD package 255 is enabled. The values entered must be different than those entered for the loop number and interface identifier at the DCHL prompt. The PRI prompt is generated until <CR> is entered.
	LOOP ID 1-9 (0)-15	Secondary PRI loop number and interface identifier (Option 11C). Prompted if INBD package 255 is enabled. The values entered must be different than those entered for the loop number and interface identifier at the DCHL prompt.
- PRI	<CR>	

LD 17 - Configure the Bearer Capability for the Asia Pacific connectivity D-Channel.

Prompt	Response	Description
REQ	CHG	Change existing data.
TYPE	PARM	Make changes to system parameters.
...		
- BCAP	a...a	Bearer Capability. a...a = SPEECH (the default) or 3.1 KHz data.
...		

LD 16 – Configure Asia Pacific ISDN PRI Route Data Block.

Prompt	Response	Description
REQ	NEW CHG	Add new data. Change existing data.
TYPE	RDB	Route Data Block.
CUST	xx	Customer number, as defined in LD 15.
ROUT	0-511 0-127	Route number, for Options 51C-81/81C. For Option 11C.
TKTP	TIE COT DID	TIE trunk type. Central Office Trunk type. Direct Inward Dialing trunk type.
...		
DTRK	YES	Digital trunk route.
- DGTP	PRI PRI2	PRI digital trunk type for TAIW, HKNG and JAPN. PRI digital trunk type for AUST, CHNA, INDI, INDO, SING, THAI, TCNZ, MSIA, and PHLP.
...		
- ISDN	YES	Integrated Services Digital Network.
- MODE	PRA	ISDN PRI route.
- IFC	APAC	Asia Pacific ISDN interface.

- - CNTY	AUST CHNA HKNG INDI INDO JAPN MSIA PHLP SING TAIW TCNZ THAI	Country associated with the APAC interface. Australia. China. Hong Kong. India. Indonesia. Japan. Malaysia. Philippines. Singapore. Taiwan. New Zealand. Thailand.
...		
CPFXS	NO (YES)	Customer-defined Prefixes option. If CPFXS = NO, when constructing the Calling or Connected Line Identification, the prefixes are retrieved from the Route Data Block via the HNTN and HLCL prompts which follow. Enter NO for APAC. If CPFXS = YES, when constructing the Calling or Connected Line Identification, the prefixes are retrieved from the Customer Data Block via the HNTN and HLCL prompts in LD 15, as is currently done. This is the default response.
HNTN	0-9999	This prompt applies to APAC only if CPFXS = NO. Home National Number. This number is similar to the PFX1 number prompted in LD 15. It is added to this overlay so that this prefix can be configured on a route basis. As is the case with PFX1, the HNTN prefix can be from one-to-four digits long.
HLCL	0-9999	This prompt applies to APAC only if CPFXS = NO. Home Location Number. This number is similar to the PFX2 number prompted in LD 15. It is added to this overlay so that this prefix can be configured on a route basis. As is the case with PFX2, the HLCL prefix can be from one-to-four digits in length.
...		

ICOG	IAO ICT OGT	Incoming and/or Outgoing trunk. The trunk is Incoming and Outgoing. The trunk is Incoming only. The trunk is Outgoing only.
...		
ACOD	x...x	The Access Code for the trunk route. The ACOD must not conflict with the existing numbering plan.
CDR	YES	Call Detail Recording if the AOC display is desired in the call record.
- - OTL	YES	CDR on outgoing toll calls.
- - OAN	YES	CDR on all answered outgoing calls.
...		
MR	ENDC	AOC at end of call.
RUCS	1	Route unit cost - value received is treated as charged.
RUCF	1 0	Route unit conversion factor - no conversion required.
...		
MCTS	YES	Enable Malicious Call Trace signaling for AUST.
- MCTM	(0)-30	Malicious Call Trace request timer is defined in seconds. This is the disconnection delay which is used. It overrides T306 for calls to/from Malicious Call Trace capable sets (for AUST).
- MTND	(NO) YES	Malicious Call Trace disconnect delay for tandem calls (for AUST).

LD 14 – Configure Asia Pacific ISDN PRI trunks.

Prompt	Response	Description
REQ	NEW CHG	Add new data. Change existing data.
TYPE	TIE COT DID	TIE trunk data block. Central Office Trunk data block. Direct Inward Dialing trunk data block. Note: Must match TKTP defined in LD 16.
TN	l ch c u	Loop and channel for digital trunks, where: l = Previously defined PRI/PRI2 loops. ch = channel 1-24 for 1.5 Mbit DTI/PRI or 1-30 for 2.0 Mbit DTI/PRI For Option 11C.
CUST	xx	Customer number, as defined in LD 15. xx = 0-99 for Options 51C-81C xx = 0-31 for Option 11C.
RTMB	0-511 1-510 0-127 1-510	Route number and member number. For Option 11C.
TGAR	0 - (1) - 31	Trunk Group Access Restriction The default of 1 automatically blocks direct access.

LD 73 – Define the frame format and grade of service timers for a PRI/PRI2 loop.

Prompt	Response	Description
REQ	CHG	Change existing data.
TYPE	PRI PRI2	PRI loop. PRI2 loop.
FEAT	LPTI	Set the timers for PRI2 loop.
LOOP	xxx	PRI/PRI2 loop number.
MFF	(AFF) CRC	Alternate frame format (default). CRC-4 frame format.
ACRC	(NO) YES	NO = No automatic reporting of CRC-4 errors during transmission. Automatic reporting of CRC-4 errors during transmission.
ALRM	(REG) ALT	Regular mode of firmware alarm version (default). Alternate mode of firmware alarm version.
RAIE	(NO) YES	Disable RAIE Group II alarm state (default). Enable RAIE Group II alarm state.
...		
PERS	0-(50)-256	Persistence timer for Group II problems (2 ms increments).
CLEA	0-(50)-256	Clearance timer for Group II problems (2 ms increments).

ISDN Calling Line Identification enhancements

LD 15 — Construct the CLID entry for a customer.

Prompt	Response	Description
REQ:	NEW, CHG	Add new data, or change or delete existing data.
TYPE:	NET	Networking data (if REQ = CHG only.)
CUST	xx	Customer number.
...		
ISDN	YES	Integrated Services Digital Network.
- PNI	1-327000	Private Network Identifier.
- CLID	(NO) YES	CLID option. YES = configure a CLID table for the customer. NO = (the default) do not configure a CLID table. In this case, the remaining prompts are not generated, and no CLID is sent for the customer.
- - SIZE	0-(256)-4000	The maximum number of CLID entries needed for a customer. If REQ = NEW, you may select the default value (256) by entering <CR> in response to this prompt. It is advised that you not define a size much larger than actually needed. This entry may be increased or decreased as required.
- - INTL	0-9999 X	Country code, for international number. Enter X to delete digits.

- - ENTRY	aaaa Xaaaa Xaaaa Xbbbb <CR>	aaaa = CLID entry to be configured. Xaaaa = CLID entry to be deleted. Xaaaa Xbbbb = CLID entries to be deleted. aaaa and bbbb must be a value between 0 and (SIZE-1). The ENTRY prompt is repeated until <CR> is entered as a response. If REQ = NEW, only one new entry may be created. The entry will be saved to system memory when the configuration for the entire overlay is completed. If REQ = CHG, as many entries as needed may be created, changed or deleted. The action for the entry will be saved to system memory after the CLID entry has been completely configured, that is, after the LSC prompt has been answered. If a new CLID entry is created, or an existing CLID entry is changed, the message "ENTRY aaaa SAVED" is displayed after the LSC prompt. If a CLID entry or CLID entries is/are deleted, the message "ENTRY aaaa DELETED" or "ENTRIES aaaa-bbbb DELETED" is displayed after the LSC prompt.
- - - HNTN	0-999999 X	National code for home national number (1-6 digits). X = delete digits.
- - - HLCL	0-999..9 X	Local code for home local number or Listed Directory Number (1-12 digits). X = delete digits.
- - - DIDN	(YES) NO SRCH	How to use the DN as a DID when constructing a CLID national or local number. YES = The default. The CLID is constructed using the digits defined in HLCL followed by the DN of the active key. NO = Construct the CLID using the digits defined in HLCL. SRCH = Search on the set, from key 0 - upwards, to find a CLID entry which has the DIDN set to YES. Use the found CLID to construct the local number.
- - - HLOC	0-9999999 X	Home location code (ESN), 1-7 digits. X = delete digits.
- - - LSC	0-9999999 X	Local steering code, 1-7 digits. X = delete digits.
ENTRY aaaa SAVED ENTRY aaaa DELETED ENTRIES aaaa-bbbb DELETED		Displayed message.

It is advised that after the CLID table has been built in LD 15, a printout be generated, using LD 21, to verify the table entries defined in LD 15.

LD 21 — Print the CLID table for a customer.

Prompt	Response	Description
REQ:	PRT	Print the customer data block.
TYPE:	CLID	CLID entry data.
CUST	xx	Customer number.
SIZE	0-4000	The value that was defined for SIZE in LD 15 is printed automatically after the customer number has been entered.
RNGE	aaaa aaaa bbbb <CR>	CLID entry to be printed. Range of CLID entries to be printed (from aaaaa to bbbbb), aaaa and bbbb must be between 0 and the value defined for SIZE-1 in LD 15. Printing begins when <CR> is entered.
INTL	0-9999	Country code, for international number, is printed.
ENTRY	aaaa	CLID entry number in CLID table is printed. If a range of entries is entered in the RNGE prompt, all of the entries in the specified range are printed in sequence.
- HNTN	0-999999	National code for home national number, 1-6 digits, is printed.
- HLCL	0-999..9	Local code for home local number or Listed Directory Number, 1-12 digits, is printed.
- DIDN	YES NO SRCH	The desired choice of how to use the DN as a DID, when constructing a CLID national or local number, will be printed.
- HLOC	0-9999999	Home location code (ESN), 1-7 digits, is printed
- LSC	0-9999999	Local steering code, 1-7 digits, is printed.
...		

LD 10 — Define the CLID entry for analog (500/2500 type) sets.

Prompt	Response	Description
REQ:	NEW, CHG	Add new data, or change existing data.
TYPE:	500	500 set.
TN	c u	Terminal Number.
...		
DIG	xx yy	Dial Intercom group number and member number.
DN	xxxx (0)-N	DN and CLID entry. N = CLID SIZE-1 (SIZE defined in LD 15).
...		

LD 11 — Define the CLID entry for DN keys for Meridian 1 proprietary sets.

Note: When assigning a CLID entry to an ACD set, you cannot use the same position ID already on the set. The set must be first outted, or the ACD key must be nulled and then rebuilt with the table entry number.

Prompt	Response	Description
REQ:	NEW, CHG	Add new data, or change existing data.
TYPE:	xxxx	Type of telephone set.
TN	c u	Terminal Number.
...		

KEY	xx MCN yyyy (0)-N/D xx MCR yyyy (0)-N/D xx PVN yyyy (0)-N/D xx PVR yyyy (0)-N/D xx SCN yyyy (0)-N/D xx SCR yyyy (0)-N/D	Telephone function key assignments. xx = key number. MCN = Multiple Call Non-ringing key. MCR = Multiple Call Ringing key. PVN = Private Line Non-Ringing key PVR = Private Line Ringing key SCN = Single Call Non-ringing key. SCR = Single Call Ringing key. yyyy = DN. (0)-N = CLID entry, with N = CLID SIZE-1 (SIZE defined in LD 15). D = CLID entry. Search for a CLID entry from key 0 upwards, to find a DN key. The found CLID is used as the CLID entry for the active DN key.
KEY	xx HOT D dd yyy...y zzzz m (0)-N/D	Two-way Hotline Direct key, where: xx = key number. dd = number of digits dialed. yyy...y = target number (terminating DN, maximum of 31 digits). zzzz = two-way hotline DN. m = one of the following Terminating Modes: H = Hotline (default) N = Non-ringing R = Ringing V = Voice (0)-N = CLID entry, with N = CLID SIZE-1 (SIZE defined in LD 15). D = CLID entry. Search for a CLID entry from key 0 upwards, to find a DN key. The found CLID is used as the CLID entry for the active DN key.

KEY	xx HOT L bbb zzzz (0)-N/D	Two-way Hotline List key, where: xx = key number. bbb = Hot Line List entry (0-999). zzzz = two-way hotline DN. (0)-N = CLID entry, with N = CLID SIZE-1 (SIZE defined in LD 15). D = CLID entry. Search for a CLID entry from key 0 upwards, to find a DN key. The found CLID is used as the CLID entry for the active DN key.
KEY	xx ACD aaaa 0-N/D bbbb	ACD key, where: xx = key number. aaaa = ACD DN or Message Center DN. 0-N = CLID entry, with N = CLID SIZE-1 (SIZE defined in LD 15). D = CLID entry. Search for a CLID entry from key 0 upwards, to find a DN key. The found CLID is used as the CLID entry for the active DN key. bbbb = ACD agent's position ID. Please refer to the note at the top of the overlay table, on page 1212, that pertains to assigning a CLID entry to an ACD set.
...		

LD 27 — Define the CLID entry for ISDN BRI sets.

Prompt	Response	Description
REQ	NEW, CHG	Add new data, or change existing data.
TYPE	TSP	Administer the Terminal Service Profile on the Digital Subscriber Loop.
...		
- SPID	xxxxxxx	Service Profile Identifier.
DN	xxxx (0)-N	xxxx = DN to be associated with the TSP. (0)-N = CLID entry, with N = CLID SIZE-1 (SIZE defined in LD 15).

ISDN QSIG Basic Call

ISDN PRI implementation

LD 16 – Configure Automatic Trunk Maintenance in Route Data Block.

Prompt	Response	Description
REQ	NEW, CHG	New, or change.
TYPE	RDB	Route Data Block.
...		
DGTP	PRI PRI2 BRI	1.5 Mbit DTI. 2.0 Mbit DTI. Basic Rate Interface.
IFC	ESGF ISGF	ESGF = ETSI QSIG interface with GF platform. ISGF = ISO QSIG interface with GF platform. The ESGF and ISGF prompt can be entered if both QSIG and QSIGGF packages are included.

LD 17 – Configuration Record.

Prompt	Response	Description
REQ	CHG	Change.
TYPE	ADAN	
ADAN	aaa aaa x	Add, Move or Change Input/Output Device
...		
- IFC	ESGF ISGF	ESGF = ETSI QSIG interface with GF platform. ISGF = ISO QSIG interface with GF platform.
PINX_CUST	xx	Customer number.
ISDN_MCNT	60-(300)-350	Layer 3 call control message count five per second time interval.

ISDN BRI implementation

LD 27 - Configure DSL for ISDN BRI trunk configuration.

Prompt	Response	Description
REQ	NEW, CHG	New, or change.
TYPE	DSL	Digital Subscriber Loop.
DSL	c dsl#	Card and DSL Number.
APPL	BRIE	Basic Rate Trunk Application.
ISDN_MCNT	20-(300)-350	Layer 3 call control message count five per second time interval.
CUST	xx	Customer number.
...		

ISDN QSIG Alternate Routing

LD 86 – Configure the QSIG Alternate Routing options.

QSIG Alternate Routing may be configured for each of the 512 different routes.

Prompt	Response	Description
REQ	CHG	Change existing data.
CUST	xx	Customer number, as configured in LD 15. xx = 0-31 for Option 11C.
FEAT	RLB	Configure the Route List as a feature.
MXLC	0-1000	Maximum number of NARS Location Codes.
RLI	xxx	Route List Index to be accessed. xxx = 0-127 if a Coordinated Dialing Plan is used. xxx = 0-255 if NARS is configured. xxx = 0-999 if the Flexible Numbering Plan is configured.
ENTR	0-63	Entry number for the NARS route list.
ROUT	0-127	Route Number for Option 11C.
...		
SBOC		Step Back On Congestion option.
	(NRR)	No re-routing.
	RRO	Re-route if congestion is encountered at the originating node; if congestion is encountered at a transit node, drop-back to the originating node, so that the originating node decides if re-routing is needed (the IDBB prompt in this LD 86 must be set to DBA or DBI).
	RRA	Re-route whether congestion is encountered at the originating or tandem node.

- COPT		Conditions that may cause QSIG Alternate Routing. These values are contained in signaling messages that trigger alternate routing. The COPT prompt is generated only if SBOC = RRO or RRA.
	(1)	QSIG Alternate Routing is supported due to the following causes: — Cause 34, "No Channel or Circuit Available" — Cause 38, "Network Out of Order" — Cause 42, "Congestion"
	2	QSIG Alternate Routing is supported due to the following causes: — Cause 27, "Destination is Out of Service" — Cause 34, "No Channel/Circuit Available" — Cause 38, "Network Out of Order" — Cause 42, "Congestion"
IDBB	(DBD)	ISDN Drop Back Busy options.
	DBA	Deny Drop Back Busy.
	DBI	Allow Drop Back Busy on, if all route sets are busy.
...		Allow Drop Back Busy, if all Initial route sets are busy.

ISDN QSIG-BC and QSIG-GF Compliance Update

LD 17 – Configure the updated interface type for the D-channel.

Prompt	Response	Description
REQ	CHG	Change.
TYPE	ADAN	Action Device and Number.
- ADAN	NEW DCH xx	New D-channel number.
...	...	
- IFC		Interface type for D-channel.
	EGF4	Updated interface type (X11 Rls 24.2x or later). Only available for PRI2 loops.
...	...	
- DCHL	xxx	PRI2 loop number for D-channel.
...	...	

LD 16 – Configure the updated interface type for the trunk route.

Prompt	Response	Description
REQ	NEW/CHG	Add or Change.
TYPE	RDB	Route Data Block.
...	...	
- IFC		Interface type for D-channel.
	EGF4	Updated interface type (X11 Rls 24.2x or later).
...	...	

ISDN QSIG/EuroISDN Call Completion

LD 17– Configure Remote Capabilities to Call Completion for QSIG or EuroISDN on Primary Rate Interface.

Prompt	Response	Description
REQ	CHG	Change.
TYPE	ADAN	Configuration Record.
- ADAN	CHG DCH x	Change D-channel.
...		
- IFC	xxxx	Interface type for route where xxxx is: ESGF = Interface ID for ETSI QSIG with GF capability. ISGF = Interface ID for ISO QSIG with GF capability. EURO = Interface ID for EuroISDN.
- - CNTY	xxx	Country. Prompted only when IFC = EURO. Where xxx is: ETSI = ETS 300 102 basic protocol (default). GER = Germany. DEN = Denmark. NET = Master Mode. SWI = Switzerland.
...		
- RCAP	CCBS	Add Call Completion to Busy Subscriber as remote capability.
- RCAP	CCNR	Add Call Completion No Response. This prompt is only applicable to QSIG interfaces.

LD 16 – Configure Remote Capability to Call Completion QSIG or EuroISDN on Basic Rate Interface.

Prompt	Response	Description
REQ	CHG	Change.
TYPE	RDB	Route Data Block.
...		
DTRK	YES	Digital Trunk Route.
- DGTP	BRI	Basic Rate Interface. Allowed if TKTP = TIE, COT, or DID.
...		
- IFC	xxxx	Interface type for route where xxxx is: ESGF = Interface ID for ETSI QSIG with GF capability. ISGF = Interface ID for ISO QSIG with GF capability. EURO = Interface ID for EuroISDN.
- - CNTY	xxx	Country. Prompted only when IFC = EURO. Where xxx is: ETSI = ETS 300 102 basic protocol (default). GER = Germany. DEN = Denmark. NET = Master Mode. SWI = Switzerland.
...		
- RCAP	CCBS	Add Call Completion to Busy Subscriber as a remote capability. The response XCCBS removes this capability.
- RCAP	CCNR	Add Call Completion No Response as a remote capability. This prompt is only applicable to QSIG interfaces.

LD 10 – Add or Change Ring Again on Analog (500/2500 type) telephones.

Prompt	Response	Description
REQ:	CHG, NEW	Change, or Add.
TYPE:	500	Telephone type.
TN	c u	Terminal Number.
...		
CLS	XRA	Ring Again allowed. XRD = Ring Again denied (default).

LD 11 – Add or Change Ring Again on Meridian 1 proprietary telephones.

Prompt	Response	Description
REQ:	CHG, NEW	Change, or Add.
TYPE:	aaaa	Telephone type where aaaa = SL1, 2006, 2008, 2009, 2016, 1018, 2112, 2216 2317, 2616 or 3000.
TN	c u	Terminal Number.
...		
KEY	xx RGA	Assign Ring Again key where xx = key number. On M2317 and M3000 telephones xx = 27.

LD 57 – Define Ring Again Flexible Feature Codes (FFCs) and FFC Confirmation Tone.

Prompt	Response	Description
REQ	CHG	Change.
TYPE	FFC	Flexible Feature Code.
CUST	xx	Customer number.
FFCT	YES	Confirmation tone provided after FFC is activated. NO = Confirmation tone not provided.
CODE	RGAA	Ring Again activate.
RGAA	xx	Ring Again code.
CODE	RGAD	Ring Again deactivate.
RGAD	xx	Ring Again deactivation code.
CODE	RGAV	Ring Again verification.
RGAV	xx	Ring Again verification code.

LD 15 – Configure a Private Integrated Services Network Exchange (PINX) DN.

Prompt	Response	Description
REQ:	CHG	Change.
TYPE:	NET	Networking.
CUST	xx	Customer number.
- PINX_DN	xx...x	Node DN to a maximum of seven digits with DN extension.

ISDN QSIG/EuroISDN Call Completion Enhancement

ISDN PRI2 implementation

LD 17 - Configure the method of Operation Coding for the Call Completion to a Busy Subscriber supplementary service, for the associated QSIG or EuroISDN D-Channel.

Note 1: When configuring the interface type, please consider the following. In order to support countries that have not yet upgraded to the ETS 300 403 standard (as introduced by the Release 24 EuroISDN ETS 300 403 Compliance Update feature), the Meridian 1 still interworks with Central Offices conforming to the ETS 300 102 standard. So, when programming the D-Channel for PRI2 trunks (in LD 17) or a PRI2 route for ISDN trunks (in LD 16) for an ETS 300 403 interface, the following applies:

- If IFC = E403 and CNTY = ETSI (ETS 300 403 for the user side) or NET (ETS 300 403 for the network side), then the interface is fully compliant with ETS 300 403.
- If IFC = E403 and CNTY = any of the supported country entries except ETSI and NET, then the interface behaves like an ETS 300 102 extended version, that is, in addition to the existing ETS 300 102 capabilities, the bearer capability and High Layer Compatibility selection procedures (fall-back mechanism), and the basic telecommunication service identification are also implemented in order to take advantage of new teleservices being offered in Release 24, such as 7kHz telephony and Videotelephony.

Also note that a user may still configure an interface fully compliant with the ETS 300 102 standard if IFC = EURO and CNTY = any of the supported country values.

Note 2: Consider the following when configuring the method of Operation Coding for a QSIG interface. When the QSIG interface is ISO (IFC = ISGF), operations are mostly coded with Integer Values (RCAP = CCBI or CCNI). For ETSI interfaces (IFC = ESGF), the operation coding depends on the standard version. The current versions are likely to use Integer Values (RCAP = CCBI or CCNI), whereas some older versions may use Object Identifies (RCAP = CCBO or CCNO).

Prompt	Response	Description
REQ	CHG	Change existing data.
TYPE	ADAN	Action Device and Number.
- ADAN	NEW DCH xx	Add a D-Channel on port 0-63, for Options 51C-81/81C.
	NEW DCH xx	Add a D-Channel on port 0-14, for Option 11C.
...		
- IFC	ESGF ISGF EURO E403	ETSI QSIG interface with GF platform. ISO QSIG interface with GF platform. EuroISDN. EuroISDN Interface conforming to the ETS 300 403, or the extended ETS 300 102 version for country-specific interfaces. See Note 1 on page 1224.
- - CNTY		Enter country pertaining to the E403 interface. If either ETSI or NET are entered, the interface is fully compliant with ETS 300 403. See Note 1 on page 1224.
	ETSI NET	ETS 300 403 for the user side. ETS 300 403 for the network side. If any of the countries listed on the following page are entered, the interface functions with the extended ETS 300 102 capabilities. See Note 1 on page 1224.

		Enter country pertaining to EuroISDN interface (this prompt is generated if IFC = EURO).
	AUS	Austria.
	DEN	Denmark.
	(ETSI)	ETS 300-102 basic protocol.
	FIN	Finland.
	GER	Germany.
	ITA	Italy.
	NOR	Norway.
	POR	Portugal.
	SWE	Sweden.
	EIR	Ireland.
	DUT	Holland.
	SWI	Switzerland.
	BEL	Belgium.
	ESP	Spain.
	UK	United Kingdom.
	FRA	France.
	CIS	Commonwealth of Independent States (Russia and the Ukraine).
...		
- RCAP		Enter the Operation Coding method for the QSIG/ETSI CCBS supplementary service.
	CCBO	Coding by Object ID.
	CCBI	Coding by Integer Value.
	XCCB	Remove the CCBO or CCBI value.
		Note: CCBO and CCBI are mutually exclusive.
...		

LD 17 - Configure the method of Operation Coding for the Call Completion on No Reply supplementary service, for the associated QSIG D-Channel.

The value is configured as a remote capability. It may be either by Object Identifier, used for ETSI, or by Integer Value, used for ISO.

Prompt	Response	Description
REQ	CHG	Change existing data.
TYPE	ADAN	Action Device and Number.
- ADAN	NEW DCH xx	Add a D-Channel on port 0-63, for Options 51C-81/81C.
...	NEW DCH xx	Add a D-Channel on port 0-14, for Option 11C.
- IFC	ESGF	ETSI QSIG interface with GF platform.
...	ISGF	ISO QSIG interface with GF platform.
- RCAP		Enter the Operation Coding method for the QSIG CCNR supplementary service.
...	CCNO	Coding by Object ID.
	CCNI	Coding by Integer Value.
	XCCN	Remove the CCBO or CCBI value.
		Note: CCNO and CCNI are mutually exclusive.
...		

LD 10 - Configure Ring Again Class of Service on analog (500/2500) sets.

Prompt	Response	Description
REQ	NEW CHG	Add new data. Change existing data.
TYPE	500	Type of set.
TN	l s c u c u	Terminal Number address, for Option 51C-81/81C. l = loop, s = shelf, c = card, u = unit. Terminal Number address, for Option 11C. c = card, u = unit.
DES	d...d	Office Data Administration System (ODAS) Station Designator, 1-6 alphanumeric characters.
CUST	xx	Customer number, as defined in LD 15. xx = 0-99 for Option 51C-81/81C. xx = 0-31 for Option 11C.
CLS	(XRD) XRA	Deny or allow Ring Again Class of Service.
...		

LD 15 - Configure a Special Prefix Number (SPRE) for a customer.

Prompt	Response	Description
REQ:	CHG	Change existing data.
TYPE	FTR_DATA	Features and options data.
CUST	xx	Customer number. xx = 0-99 for Options 51C-81/81C. xx = 0-31 for Option 11C.
...		
SPRE	xxxx	Special Prefix Number (xxxx = 1-4digits).
...		

LD 15 - Configure the Ring Again on No Answer and Call Completion on No Response (CCNR) option, and a Private Integrated Services Network Exchange (PINX) node Directory Number.

Prompt	Response	Description
REQ:	CHG	Change existing data.
TYPE	NET	Networking data.
CUST	xx	Customer number. xx = 0-99 for Options 51C-81/81C. xx = 0-31 for Option 11C.
OPT	(RND) RNA	Deny or allow Ring Again on No Answer or Call Completion on No Response.
ISDN	YES	Integrated Services Digital Network.
...		
- PINX_DN	xx...x	Private Integrated Services Network Exchange node Directory Number (up to seven digits). Precede the existing value with an X to delete it. If the Call Completion to a Busy Subscriber (CCBS) service does not provide CLID to the MCDN or DPNSS1 gateway, the PINX_DN is used to build the calling number. On an incoming CCBS request, the PINX_DN is used to determine the length of the DN extension.
...		

LD 57 - Configure the Flexible Feature Code for Calling Private Privacy, and define whether Flexible Feature Code confirmation tone is to be provided.

Prompt	Response	Description
REQ	CHG	Change existing data.
TYPE	FFC	Flexible Feature Code data.
CUST	xx	Customer number, as defined in LD 15. xx = 0-99 for Options 51C-81/81C. xx = 0-31 for Option 11C.
FFCT	(NO) YES	(Do not) provided FFC confirmation tone after the FFC is dialed.
CODE	RGAA	The type of FFC is Ring Again Activate.
- RGAA	xxxx	Enter the Ring Again Activate FFC.
CODE	RGAA	The type of FFC is Ring Again Activate.
- RGAD	xxxx	Enter the Ring Again Deactivate FFC.
CODE	RGAA	The type of FFC is Ring Again Activate.
- RGAV	xxxx	Enter the Ring Again Verify FFC.
...		

LD 17 - Configure Network Attendant Service on an MCDN D-Channel.

Prompt	Response	Description
REQ	CHG	Change existing data.
TYPE	ADAN	Action device and number.
- ADAN	CHG DCH xx	Change a D-channel on logical port 0-63, for Options 51C-81/81C.
	CHG DCH xx	Change a D-channel on logical port 0-15, for Option 11C.
...		
- IFC	SL1	MCDN interface.
...		
- NASA	YES	Allow Network Attendant Service.
...		

ISDN BRI implementation

LD 16 - Configure the method of Operation Coding for the Call Completion to a Busy Subscriber supplementary service, for the associated QSIG or EuroISDN BRI trunk route.

Prompt	Response	Description
REQ	NEW	Add new data (ISDN BRI protocol group settings).
TYPE	RDB	Route Data Block.
CUST	xx	Customer number, already defined in LD 15. xx = 0-99 for Options 51C-81/81C xx = 0-31 for Option 11C.
ROUT	0-511 0-127	Route number, for Options 51C-81/81C. For Option 11C.
TKTP	aaa	Trunk type.
DTRK	YES	Digital Trunk Route.
BRIP	NO	ISDN BRI packet handler route (NO is entered, since packet data is not required).
- DGTP	BRI	Digital trunk type.
ISDN	YES	Integrated Services Digital Network.
...		
- IFC	ESGF ISGF EURO E403	ETSI QSIG interface with GF platform. ISO QSIG interface with GF platform. EuroISDN. EuroISDN Interface conforming to the ETS 300 403, or the extended ETS 300 102 version for country-specific interfaces. See Note 1 on page 1224.

- - CNTY	ETSI NET	Enter country pertaining to the E403 interface. If either ETSI or NET are entered, the interface is fully compliant with ETS 300 403. See Note 1 on page 1224. ETS 300 403 for the user side. ETS 300 403 for the network side. If any of the countries listed on the following page are entered, the interface functions with the extended ETS 300 102 capabilities. See Note 1 on page 1224. Enter country pertaining to EuroISDN interface (this prompt is generated if IFC = EURO).
...		
RCAP	AUS DEN (ETSI) FIN GER ITA NOR POR SWE EIR DUT SWI BEL ESP UK FRA CIS	Austria. Denmark. ETS 300-102 basic protocol. Finland. Germany. Italy. Norway. Portugal. Sweden. Ireland. Holland. Switzerland. Belgium. Spain. United Kingdom. France. Commonwealth of Independent States (Russia and the Ukraine).
...		
	CCBO CCBI XCCB	Enter the Operation Coding method for the QSIG/ETSI CCBS supplementary service. Coding by Object ID. Coding by Integer Value. Remove the CCBO or CCBI value. Note: CCBO and CCBI are mutually exclusive.
...		

LD 16 - Configure the method of Operation Coding for the Call Completion on No Reply supplementary service, for the associated QSIG ISDN BRI trunk route.

Prompt	Response	Description
REQ	NEW	Add new data (ISDN BRI protocol group settings).
TYPE	RDB	Route Data Block.
CUST	xx	Customer number, as defined in LD 15. xx = 0-99 for Options 51C-81/81C xx = 0-31 for Option 11C.
ROUT	0-511 0-127	Route number, for Options 51C-81/81C. For Option 11C.
TKTP	aaa	Trunk type.
DTRK	YES	Digital Trunk Route.
BRIP	NO	ISDN BRI packet handler route (NO is entered, since packet data is not required).
- DGTP	BRI	Digital trunk type.
ISDN	YES	Integrated Services Digital Network.
...		
- IFC	ESGF ISGF	ETSI QSIG interface with GF platform. ISO QSIG interface with GF platform.
...		
RCAP		Enter the Operation Coding method for the QSIG CCNR supplementary service.
	CCNO CCNI XCCN	Coding by Object ID. Coding by Integer Value. Remove the CCBO or CCBI value.
...		Note: CCNO and CCNI are mutually exclusive.

ISDN QSIG Call Diversion Notification

The QSIG Call Diversion Notification feature requires the configuration of one of the following call redirection features: Call Forward All Calls, Call Forward/Hunt Override Via Flexible Feature Codes, Call Forward Remote (Attendant and Networkwide), Call Forward No Answer/Flexible Call Forward No Answer, Call Forward No Answer Second Level, Call Forward Internal Calls and User Selectable Call Redirection. The implementation of these features is not modified.

Remote Capability Meanings

Table 68 indicates the remote capabilities meanings for both D-channel and BRI routes. When using Table 68 consider the following:

- Only nodes subject to be Originating, Served, Diverted or Rerouting nodes with respect to QSIG Call Diversion Notification need to have diversion remote capability configured. Transmit nodes pass the information transparently.
- When choosing the Operation Coding Choice, the interface type should be considered. When the QSIG interface used is ISO (IFC ISGF), operations are mostly coded with Integer Values. For ETSI interfaces (IFC ESGF), the operation coding depends on the standard version: the last ones are likely to use Integer Values, whereas some older ones may use Object Identifiers.

Only one remote capability allows the QSIG Diversion configuration on a D-channel/BRI route. This remote capability gathers the three following possibilities for the D-channel/BRI route:

- 1 coding of operations is sent to the remote switch, which can be coded as either as Object Identifier or as Integer Value. If coded as Object Identifier, the remote capability ends with an 'O', whereas for Integer Value, the remote capability ends with an 'I'. This means that remote capabilities explained below in 2 and 3 are defined twice.
- 2 sending of QSIG Diversion Notification Information to the remote switch: this information is sent only if the remote capability is of first or third type, i.e. DV1x or DV3x, where the x is either 'I' or 'O' as explained in 1.

- 3 treatment of Rerouting requests received from the remote switch: a rerouting request is only processed if the remote capability is of second or third type, i.e. DV2x or DV3x, where x is either 'I' or 'O' as explained above in 1.

Table 68
Remote Capability Meanings

Remote capability	Meaning for Operation Coding	Meaning for Notification Information	Meaning for Rerouting request
None of the following remote capabilities.	Not applicable (nothing sent)	Not <u>sent</u>	Not processed when <u>received</u>
DV1O	<u>Sent</u> coded as Object Identifier	<u>Sent</u>	Not processed when <u>received</u>
DV1I	<u>Sent</u> coded as Integer Value		
DV2O	<u>Sent</u> coded as Object Identifier	Not <u>Sent</u>	Processed when <u>received</u>
DV2I	<u>Sent</u> coded as Integer Value		
DV3O	<u>Sent</u> coded as Object Identifier	<u>Sent</u>	Processed when <u>received</u>
DV3I	<u>Sent</u> coded as Integer Value		

LD 17 – Configure or remove remote capabilities D-channel for QSIG Call Diversion Notification.

Prompt	Response	Description
REQ:	CHG	Change existing data.
TYPE:	ADAN	Action device and number.
ADAN	CHG DCH x	Configure new remote capabilities on D-channel x.
...		
IFC	xxxx	QSIG interface type where xxxx is: ESGF = interface ID for ETSI QSIG with GF capability. ISGF = Interface ID for ISO with GF capability.
...		
RLS	23	Release 23 and later.
...		
RCAP		QSIG SS Call Diversion Notification remote capability. Configure sending of QSIG Diversion Notification Information, treatment of Rerouting request and coding of operations. If coded as Object Identifier, the remote capability ends with 'O', whereas for Integer Value, the remote capability ends with 'I'. Only one remote capability is allowed. Refer to Table 1 for further information.
	DV1I	Diversion information is sent to remote switch.
	DV1O	Diversion information is sent to remote switch.
	DV2I	Rerouting requests from remote switch are processed.
		Rerouting requests from remote switch are processed.
	DV2O	Diversion information is sent to remote switch. Rerouting requests from remote switch are processed.
		Diversion information is sent to remote switch. Rerouting requests from remote switch are processed.
	DV3I	Diversion information is sent to remote switch.
	DV3O	Rerouting requests from remote switch are processed.
		Precede with 'X' to remove capability.

LD 16 – Configure Remote Capability for a route with QSIG Generic Functional Interface.

Prompt	Response	Description
REQ	CHG	Change existing data.
TYPE	RDB	Route Data Block.
CUST	xx	Customer number.
ROUT	0-127	Route Number.
...		
IFC	xxxx	QSIG interface type where xxxx is: ESGF = interface ID for ETSI QSIG with GF capability. ISGF = Interface ID for ISO with GF capability.
...		
RCAP		QSIG SS Call Diversion Notification remote capability. Configure sending of QSIG Diversion Notification Information, treatment of Rerouting request and coding of operations. If coded as Object Identifier, the remote capability ends with 'O', whereas for Integer Value, the remote capability ends with 'I'. Only one remote capability is allowed. Refer to Table 1 for further information.
	DV1I	Diversion information is sent to remote switch.
	DV1O	Diversion information is sent to remote switch.
	DV2I	Rerouting requests from remote switch are processed.
	DV2O	Rerouting requests from remote switch are processed.
	DV3I	Diversion information is sent to remote switch.
	DV3O	Rerouting requests from remote switch are processed.
		Precede with 'X' to remove capability.

LD 10 – Configure QSIG Diversion Notification on analog (500/2500 type) telephones

Prompt	Response	Description
REQ:	NEW CHG	Add new data. Change existing data.
TYPE:	500	Telephone type.
...		
TN	c u	Terminal Number.
...		
DN	x...x yyy	Directory Number and Calling Line Identification entry (Range is (0) - value entered for SIZE prompt in LD 15 minus one).
...		
- CPND	aaa	Calling Party Name Display where: NEW = Add data block. CHG = Change existing data block. OUT = Remove existing data block.
- -CPND_LANG	aaa	Calling Party Name Display Language where: (ROM) = Roman. KAT = Katakana.
- - NAME	aaaa bbbb	Calling Party Name Display Name where: First name Last name.
...		
CLS	(DNO3) DNO1 DNO2	QSIG Call Diversion Notification options for calling party. Notification with diverted-to party's number and name. No notification. Notification without diverted-to party's number and name.

	(DNDY) DNDN	QSIG Call Diversion Notification options concerning forwarded- to party. Notification with Called party's number and name. Notification without Called party's number and name.
	CNDA	Call Party Name Display Allowed
	CFXA	Call Forward to External DN Allowed.
	CFTA	Call Forward by Call Type Allowed.
	DDGA	Directory Number Display on other set Allowed.
	FNA	Call Forward No Answer Allowed.
	HTA	Hunting Allowed.
	NAMA	Name Display on other set Allowed or (Denied).

LD 11— Configure QSIG Diversion Notification on Meridian 1 proprietary telephones.

Prompt	Response	Description
REQ:	NEW CHG	Add new data. Change existing data.
TYPE:	xxxx	Telephone type, where: xxxx = SL1, 2006, 2008, 2009, 2016, 2018, 2112, 2216, 2317, 2616 or 3000.
...		
TN	c u	Terminal Number.
...		
FDN	x...x	Flexible Call Forward No Answer DN for an Internal Directory Number.
...		
CLS	(DNO3) DNO1 DNO2	QSIG Call Diversion Notification options for calling party. Notification with diverted-to party's number and name. No notification. Notification without diverted-to party's number and name.

	(DNDY) DNDN	QSIG Call Diversion Notification options concerning forwarded- to party. Notification with Called party's number and name. Notification without Called party's number and name.
	CNDA	Call Party Name Display Allowed
	CFXA	Call Forward to External DN Allowed.
	CFTA	Call Forward by Call Type Allowed.
	DDGA	Directory Number Display on other set Allowed.
	FNA	Call Forward No Answer Allowed.
	HTA	Hunting Allowed.
	NAMA	Name Display on other set Allowed or (Denied).
...		
RCO	(0) - 2	Ringing Cycle Option for Call Forward No Answer.
EFD	x...x	Call Forward No Answer Directory Number for external calls
...		
- CPND	aaa	Calling Party Name Display where: NEW = Add data block. CHG = Change existing data block. OUT = Remove existing data block.
- -CPND_LANG	aaa	Calling Party Name Display Language where: (ROM) = Roman. KAT = Katakana.
- - NAME	aaaa bbbb	Calling Party Name Display Name where: First name Last name.

LD 27 – Configure Basic Rate Interface telephones.

Prompt	Response	Description
REQ	NEW CHG	Add new data. Change existing data.
TYPE	TSP	Terminal Service Profile.
DSL	l s c dsl	Digital Subscriber Loop address.
USID	x	User Service Identifier.
...		
DN	xxx...x	Directory Number associated with TSP (1 to 7 digits).
- CTI	VCE DTA	Call type where VCE = circuit switched voice and DTA = circuit switched data.
...		
FEAT	CFXA	Call Forward Allowed to external Directory Number.
FEAT	xxxx	QSIG Call Diversion Notification for calling party where xxxx: DNO1 = no notification DNO2 = notification without forwarded-to (diverted) party's number and name (DNO3) = notification with forwarded-to (diverted) party's number and name when available.
FEAT	xxxx	QSIG Call Diversion Notification for forwarded-to (diverted) party where xxxx: DNDN = no notification of called party's number and name notification (DNDY) = notification with called party's number and name when available (default).
SSRV_ETSI	aaa mmm nnn	ETSI Supplementary Service where: VCFW = Voice Call Forward and DCFW = Data Call Forward.

LD 95 – Modify Call Party Name Display data block.

Prompt	Response	Description
REQ	CHG	Change existing data.
TYPE	CPND	Call Party Name Display data block.
CUST	xx	Customer number.
...		
RESN	YES	Display Redirection reason allowed. (NO) = Display Redirection reason denied (default).
- CFWD	aaaa	Mnemonic for Call Forward All Call display. (F) = default
- CFNA	aaaa	Mnemonic for Call Forward No Answer display (N) = default.
- HUNT	aaaa	Mnemonic for Call Forward Busy display. (B) = default

LD 95 – Define Name for telephones.

Prompt	Response	Description
REQ	NEW	Add new data.
TYPE	NAME	Name.
CUST	xx	Customer number.
...		
DN	x..x	Directory Number.
- NAME	a..a	Calling Party Name Display in ASCII characters.

LD 11 – Configure the name that is to be displayed on the Forwarded-to (diverted) Party's set.

Prompt	Response	Description
REQ	CHG	Change existing data.
TYPE	xxxx	Telephone type, where: xxxx = SL1, 2006, 2008, 2009, 2016, 2018, 2112, 2216, 2317, 2616 or 3000.
TN	c u	Terminal Number.
...		
CLS	DNDA	Dialed Name Display Allowed. (DNDD)= Dialed Name Display Denied (default).

Note: LD 81 may be used to print counting or listing queries pertaining to the new Classes of Service, by entering CNT or LST against the REQ prompt, and the appropriate COS against the FEAT prompt.

Note: LD 83 may be used may be used to print the new Classes of Service in the TN blocks, by entering TNB against the REQ prompt, and the appropriate COS against the CLS prompt.

ISDN QSIG Call Diversion Notification Enhancements

Remote Capability Meanings, for PRI D-Channels and BRI routes

Table 68 indicates the remote capabilities meanings for both PRI2 D-Channels and BRI routes. When using Table 68 consider the following:

- Only nodes subject to be Originating, Served, Diverted or Rerouting nodes with respect to QSIG Call Diversion Notification need to have diversion remote capability configured. Transmit nodes pass the information transparently.
- When choosing the Operation Coding Choice, the interface type should be considered. When the QSIG interface used is ISO (IFC ISGF), operations are mostly coded with Integer Values. For ETSI interfaces (IFC ESGF), the operation coding depends on the standard version: the last ones are likely to use Integer Values, whereas some older ones may use Object Identifiers.

Only one remote capability allows the QSIG Diversion configuration on a D-Channel/BRI route. This remote capability gathers the three following possibilities for the D-Channel/BRI route:

- 1** coding of operations is sent to the remote switch, which can be coded as either as Object Identifier or as Integer Value. If coded as Object Identifier, the remote capability ends with an 'O', whereas for Integer Value, the remote capability ends with an 'I'. This means that remote capabilities explained below in 2 and 3 are defined twice.
- 2** sending of QSIG Diversion Notification Information to the remote switch: this information is sent only if the remote capability is of first or third type, i.e. DV1x or DV3x, where the x is either 'I' or 'O' as explained in 1.
- 3** treatment of Rerouting requests received from the remote switch: a rerouting request is only processed if the remote capability is of second or third type, i.e. DV2x or DV3x, where x is either 'I' or 'O' as explained above in 1.

Table 69
Remote Capability Meanings

Remote capability	Meaning for Operation Coding	Meaning for Notification Information	Meaning for Rerouting request
None of the following remote capabilities.	Not applicable (nothing sent)	Not <u>sent</u>	Not processed when <u>received</u>
DV10	<u>Sent</u> coded as Object Identifier	<u>Sent</u>	Not processed when <u>received</u>
DV11	<u>Sent</u> coded as Integer Value		
DV20	<u>Sent</u> coded as Object Identifier	Not <u>Sent</u>	Processed when <u>received</u>
DV21	<u>Sent</u> coded as Integer Value		
DV30	<u>Sent</u> coded as Object Identifier	<u>Sent</u>	Processed when <u>received</u>
DV31	<u>Sent</u> coded as Integer Value		

ISDN PRI2 implementation

LD 17 - Configure or remove remote capabilities D-Channel for QSIG Call Diversion Notification.

Prompt	Response	Description
REQ:	CHG	Change existing data.
TYPE	ADAN	Action Device and Number.
- ADAN	NEW DCH xx	Add a D-Channel on port 0-63 (Options 51C-81C).
	NEW DCH xx	Add a D-Channel on port 0-14 (Option 11C).
...		
- IFC	ESGF ISGF	ETSI QSIG interface with GF platform. ISO QSIG interface with GF platform.
...		
RLS	23	Release 23 and later.
...		
RCAP		QSIG SS Call Diversion Notification remote capability. Configure sending of QSIG Diversion Notification Information, treatment of Rerouting request and coding of operations. If coded as Object Identifier, the remote capability ends with 'O', whereas for Integer Value, the remote capability ends with 'I'. Only one remote capability is allowed. Refer to Table 1 for further information.
	DV1I	Diversion information is sent to remote switch.
	DV1O	Diversion information is sent to remote switch.
	DV2I	Rerouting requests from remote switch are processed.
		Rerouting requests from remote switch are processed.
	DV2O	Diversion information is sent to remote switch. Rerouting requests from remote switch are processed.
		Diversion information is sent to remote switch. Rerouting requests from remote switch are processed.
	DV3I	Diversion information is sent to remote switch.
	DV3O	Rerouting requests from remote switch are processed.
		Precede with 'X' to remove capability, but do not specify the coding type . For example, XDV3 or XDV1 (and not XDV3I or XDV1O).

LD 10 - Configure QSIG Diversion Notification Class of Service, on analog (500/2500 type) sets.

Prompt	Response	Description
REQ:	NEW CHG	Add new data. Change existing data.
TYPE:	500	Set type.
...		
TN	l s c u c u	Terminal Number, for Options 51C-81C. l = loop, s = shelf, c = card, u = unit. Terminal Number for Option 11C. c = card, u = unit.
...		
DN	xx...x yyy	Directory Number and Calling Line Identification entry (yyy range is (0) - value entered for SIZE prompt in LD 15 minus one).
...		
- CPND	aaa	Calling Party Name Display where aaa can be: NEW = Add data block. CHG = Change existing data block. OUT = Remove existing data block.
- -CPND_LANG	aaa	Calling Party Name Display Language where: (ROM) = Roman. KAT = Katakana.
- - NAME	aaaa bbbb	Calling Party Name Display Name where: aaaa = First name bbbb = Last name.
...		
CLS	(DNO3) DNO1 DNO2	QSIG Call Diversion Notification options for calling party. Notification with diverted-to party's number and name. No notification. Notification without diverted-to party's number and name.

	(DNDY) DNDN	QSIG Call Diversion Notification options for diverted-to party. DNDY = Notification with called party's number and name. DNDN = Notification without called party's number and name.

LD 11 - Configure QSIG Diversion Notification Class of Service, on Meridian 1 proprietary sets.

Prompt	Response	Description
REQ:	NEW CHG	Add new data. Change existing data.
TYPE:	xxxx	Set type, where: xxxx = SL1, 2006, 2008, 2009, 2016, 2018, 2112, 2216, 2317, 2616 or 3000.
...		
TN	l s c u c u	Terminal Number, for Options 51C-81C. l = loop, s = shelf, c = card, u = unit. Terminal Number for Option 11C. c = card, u = unit.
...		
FDN	x...x	Flexible Call Forward No Answer DN for an Internal Directory Number.
...		
CLS	(DNO3) DNO1 DNO2 (DNDY) DNDN	QSIG Call Diversion Notification options for calling party. Notification with diverted-to party's number and name. No notification. Notification without diverted-to party's number and name. QSIG Call Diversion Notification options for diverted-to party. Notification with called party's number and name. Notification without called party's number and name.

...		
- CPND	aaa	Calling Party Name Display where aaa can be: NEW = Add data block. CHG = Change existing data block. OUT = Remove existing data block.
- -CPND_LANG	aaa	Calling Party Name Display Language where aaa can be: (ROM) = Roman. KAT = Katakana.
- - NAME	aaaa bbbb	Calling Party Name Display Name where: aaaa = first name bbbb = last name.

LD 95 - Configure the QSIF Call Diversion Notification mnemonics, for Call Forward All Calls, Call Forward No Answer, and Call Forward Busy, in the Call Party Name Display data block.

Prompt	Response	Description
REQ	CHG	Change existing data.
TYPE	CPND	Call Party Name Display data block.
CUST	xx	Customer number, as defined in LD 15. xx = 0-99 for Options 51C-81C. xx = 0-31 for Option 11C.
...		
RESN	YES	Display Redirection reason allowed. (NO) = Display Redirection reason denied (default).
- CFWD	aaaa	Mnemonic for Call Forward All Call display. (F) = default
- CFNA	aaaa	Mnemonic for Call Forward No Answer display (N) = default.
- HUNT	aaaa	Mnemonic for Call Forward Busy display. (B) = default

LD 95 – Define a Calling Party Name to be displayed on a receiving set.

Prompt	Response	Description
REQ	NEW	Add new data.
TYPE	NAME	Name.
CUST	xx	Customer number, as defined in LD 15. xx = 0-99 for Options 51C-81C. xx = 0-31 for Option 11C.
...		
DN	x..x	Directory Number.
- NAME	a..a	Calling Party Name Display in ASCII characters.

LD 11 - On a Meridian proprietary set, configure whether or not the name of the originally-called party is to be displayed on the diverted-to set.

Note 1: LD 81 may be used to print counting or listing queries pertaining to the new Classes of Service, by entering CNT or LST against the REQ prompt, and the appropriate COS against the FEAT prompt.

Note 2: LD 83 may be used to print the new Classes of Service in the TN blocks, by entering TNB against the REQ prompt, and the appropriate COS against the CLS prompt.

Prompt	Response	Description
REQ	CHG	Change existing data.
TYPE	xxxx	Telephone type, where: xxxx = SL1, 2006, 2008, 2009, 2016, 2018, 2112, 2216, 2317, 2616 or 3000.
TN	c u	Terminal Number for Option 11C. c = card, u = unit.
...		

CLS	(DNDD)	DNDD = Dialed Name Display Denied (the name of the originally-dialed party is not displayed on the set). DNDA = Dialed Name Display Allowed (the name of the originally-dialed party is displayed on the set).
	DNDA	

ISDN BRIT implementation

LD 16 - Configure the QSIG Call Diversion Notification Remote Capability for a BRI route with QSIG Generic Functional Interface.

Prompt	Response	Description
REQ	CHG	Change existing data.
TYPE	RDB	Route Data Block.
CUST	xx	Customer number, as defined in LD 15. xx = 0-31 for Option 11C.
ROUT	0-127	Route number, for Option 11C.
TKTP	aaa	Trunk type.
DTRK	YES	Digital Trunk Route.
BRIP	NO	ISDN BRI packet handler route (NO is entered, since packet data is not required).
- DGTP	BRI	Digital trunk type.
ISDN	YES	Integrated Services Digital Network.
...		
- IFC	ESGF ISGF	QSIG interface type where xxxx is: ESGF = interface ID for ETSI QSIG with GF capability. ISGF = Interface ID for ISO with GF capability.
...		
RCAP		QSIG SS Call Diversion Notification remote capability. Configure sending of QSIG Diversion Notification Information, treatment of Rerouting request and coding of operations. If coded as Object Identifier, the remote capability ends with 'O', whereas for Integer Value, the remote capability ends with 'I'. Only one remote capability is allowed. Refer to Table 1 for further information.

	DV1I DV1O DV2I DV2O DV3I DV3O	Diversion information is sent to remote switch. Diversion information is sent to remote switch. Rerouting requests from remote switch are processed. Rerouting requests from remote switch are processed. Diversion information is sent to remote switch. Rerouting requests from remote switch are processed. Diversion information is sent to remote switch. Rerouting requests from remote switch are processed. Precede with 'X' to remove capability, but do not specify the coding type. For example, XDV3 or XDV1 (and not XDV3I or XDV1O).

LD 27 - Configure the Terminal Service Profile (TSP), to implement the QSIG Call Diversion options on BRI sets.

Prompt	Response	Description
REQ	NEW CHG	Add new data. Change existing data.
TYPE	TSP	Terminal Service Profile.
DSL	cc sdl	DSL Terminal Number location, for Option 11C: cc (card) = 1-20 dsl# (DSL number) = 0-7
USID	x	User Service Identifier.
...		
DN	xxx...x	Directory Number associated with TSP (1 to 7 digits).
...		
FEAT	xxxx	QSIG Call Diversion Notification for calling party where xxxx: DNO1 = no notification DNO2 = notification without forwarded-to (diverted) party's number and name (DNO3) = notification with forwarded-to (diverted) party's number and name when available.

FEAT	xxxx	QSIG Call Diversion Notification for diverted-to party where xxxx: DNDN = no notification of called party's number and name notification (DNDY) = notification with called party's number and name when available (default).
...		

ISDN QSIG Call Transfer Notification

ISDN PRI2 implementation

LD 15 - Configure a Private Integrated Services Network Exchange (PINX) DN for a customer.

Prompt	Response	Description
REQ	CHG	Change existing data.
TYPE	NET	Networking data.
...		
CUST	xx	Customer number, as defined in LD 15. xx = 0-99 for Options 51C-81C. xx = 0-31 for Option 11C.
...		
- PINX_DN	xx...x	Private Integrated Services Network Exchange DN (1-7 digits).
...		

LD 17 – Configure the method of Operation Coding for the QSIG Call Transfer Notification feature, for the associated QSIG D-Channel.

The value is configured as a remote capability. It may be either by Object Identifier, used for ETSI, or by Integer Value, used for ISO.

Prompt	Response	Description
REQ	CHG	Change existing data.
TYPE	ADAN	Action Device and Number.
- ADAN	NEW DCH xx	Add a D-Channel on port 0-63, for Options 51C-81C.
	NEW DCH xx	Add a D-Channel on port 0-14, for Option 11C..
...		
- IFC	ESGF	ETSI QSIG interface with GF platform.
	ISGF	ISO QSIG interface with GF platform.
...		
- RCAP		Enter the Operation Coding method for the QSIG Call Transfer Notification feature.
	CTI	Call Transfer Operation Coding method is by Integer Value.
	XCTI	Remove the Call Transfer Coding by Integer Value.
	CTO	Call Transfer Operation Coding method is by Object ID.
	XCTO	Remove the Call Transfer Coding by Object ID.
		Note: CTO and CTI are mutually exclusive.
...		

LD 95 – Configure the Reason for Redirection, and the Call Transfer mnemonic, for a customer.

Prompt	Response	Description
REQ	NEW CHG	Add new data. Change existing data.
TYPE	CPND	Configure the Calling Party Name Display data block.
CUST	xx	Customer number, as defined in LD 15. xx = 0-99 for Options 51C-81C. xx = 0-31 for Option 11C.
...		
RESN	(NO) YES	(Do not) display the reason for redirection. Enter YES.
- XFER	(T) aaaa	1-4 character mnemonic for Call Transfer. The default is T.

LD 95 – Define a name string for a customer.

Prompt	Response	Description
REQ	NEW CHG	Add new data. Change existing data.
TYPE	NAME	Create a new Calling Party Name string.
CUST	xx	Customer number, as previously entered in LD 95 when configuring the CPND data block. xx = 0-99 for Options 51C-81C. xx = 0-31 for Option 11C.
...		
DN	xxx...x	Directory Number (1 to 7 digits) to which the name string is associated.
- NAME	a...a	CPND name, in ASCII characters.

LD 10 - Configure an analog (500/2500) transferring set, using LD 10, so that after Call Transfer, the originating set display and the transferred-to set display show the name and number associated with the transferred-to set and originating set, respectively.

Prompt	Response	Description
REQ:	NEW CHG	Add new data. Change existing data.
TYPE:	aaa	Type of analog set. aaa = 500 or 2500.
TN	l s c u	Terminal Number, for Options 51C-81C. l = loop, s = shelf, c = card, u = unit.
	c u	Terminal Number for Option 11C. c = card, u = unit.
...		
CUST	xx	Customer number associated with this set, as previously entered in LD 15. xx = 0-99 for Options 51C-81C. xx = 0-31 for Option 11C.
...		
DN	xxxx yyy	Directory Number and CLID entry (range for yyy is 0 - value entered in the SIZE prompt in LD 15).
- CPND	NEW CHG OUT	Add new Calling Party Name Display data. Change existing Calling Party Name Display data. Remove Calling Party Name Display data.
- - CPND_LAN	aaa	Calling Party Name Display language.
- - NAME	aaaa bbbb	Calling Party Name Display name. aaaa = first name, bbbb = last name.
...		
CLS		Class of Service.
	(XFD) XFA	Transfer (denied) allowed on this set.
	(CNDD) CNDA	Name Display (denied) allowed on this set.

CLS	(NAMD) NAMA	Name Display (denied) allowed on other sets.
...	(DDGD) DDGA	DN display (denied) allowed on other sets.
	<CR>	The CLS prompt is generated until <cr> is entered.

LD 11 - Configure a Meridian 1 proprietary transferring set, using LD 11, so that after Call Transfer, the originating set display and the transferred-to set display show the name and number associated with the transferred-to set and originating set, respectively.

Prompt	Response	Description
REQ:	NEW CHG	Add new data. Change existing data.
TYPE:	aaa	Type of Meridian 1 proprietary set.
TN	l s c u	Terminal Number, for Options 51C-81C. l = loop, s = shelf, c = card, u = unit.
...	c u	Terminal Number for Option 11C. c = card, u = unit.
CUST	xx	Customer number associated with this set, as defined in LD 15. xx = 0-99 for Options 51C-81C. xx = 0-31 for Option 11C.
...		
DN	xxxx yyy	Directory Number and CLID entry (range for yyy is 0 - value entered in the SIZE prompt in LD 15).
- CPND	NEW CHG OUT	Add new Calling Party Name Display data. Change existing Calling Party Name Display data. Remove Calling Party Name Display data.
- - CPND_LAN	aaa	Calling Party Name Display language.

- - NAME	aaaa bbbb	Calling Party Name Display name. aaaa = first name. bbbb = last name.
...		
KEY	xx TRAN	Assign the Transfer key.
...		
CLS		Class of Service.
	(CNDD) CNDA	Name Display (denied) allowed on this set.
	(NAMD) NAMA	Name Display (denied) allowed on other sets.
	(DDGD) DDGA	DN display (denied) allowed on other sets.
CLS	<CR>	The CLS prompt is generated until <cr> is entered.
...		

LD 11 - Configure a Meridian 1 proprietary set, using LD 11, so that if it originates or receives a transferred call over a QSIG link, it displays the redirected/connected name and number, after it is received.

Prompt	Response	Description
REQ:	NEW CHG	Add new data. Change existing data.
TYPE:	aaa	Type of Meridian 1 proprietary set.
TN	l s c u	Terminal Number, for Options 51C-81C. l = loop, s = shelf, c = card, u = unit.
	c u	Terminal Number for Option 11C. c = card, u = unit.
...		
CUST	xx	Customer number associated with this set, as defined in LD 15. xx = 0-99 for Options 51C-81C. xx = 0-31 for Option 11C.

...		
DN	xxxx yyy	Directory Number and CLID entry (range for yyy is 0 - value entered in the SIZE prompt in LD 15).
- CPND	NEW CHG OUT	Add new Calling Party Name Display data. Change existing Calling Party Name Display data. Remove Calling Party Name Display data.
- - CPND_LAN	aaa	Calling Party Name Display language.
- - NAME	aaaa bbbb	Calling Party Name Display name. aaaa = first name. bbbb = last name.
...		
CLS		Class of Service.
	(CNDD) CNDA	Name Display (denied) allowed on this set.
	(NAMD) NAMA	Name Display (denied) allowed on other sets.
	(DDGD) DDGA	DN display (denied) allowed on other sets.
CLS	<CR>	The CLS prompt is generated until <cr> is entered.
...		

ISDN BRIT implementation

LD 16 - Configure the method of Operation Coding for the QSIG Call Transfer Notification, for the associated QSIG ISDN BRI trunk route.

Prompt	Response	Description
REQ	NEW	Add new data (ISDN BRI protocol group settings).
TYPE	RDB	Route Data Block.
CUST	xx	Customer number, as defined in Ld 15. xx = 0-99 for Options 51C-81C xx = 0-31 for Option 11C.
ROUT	0-511	Route number.
TKTP	aaa	Trunk type.
DTRK	YES	Digital Trunk Route.
BRIP	NO	ISDN BRI packet handler route (NO is entered, since packet data is not required).
- DGTP	BRI	Digital trunk type.
ISDN	YES	Integrated Services Digital Network.
...		
- IFC	ESGF ISGF	ETSI QSIG interface with GF platform. ISO QSIG interface with GF platform.
...		
- RCAP		Enter the Operation Coding method for the QSIG Call Transfer Notification feature.
	CTI XCTI CTO XCTO	Call Transfer Operation Coding method is by Integer Value. Remove the Call Transfer Coding by Integer Value. Call Transfer Operation Coding method is by Object ID. Remove the Call Transfer Coding by Object ID.
		Note: CTO and CTI are mutually exclusive.

ISDN QSIG Channel ID Coding

LD 17 – Configure the QCHID parameter for the QSIG PRI2 interface.

Prompt	Response	Description
REQ	CHG	Change.
TYPE	ADAN	Action Device and Number.
- ADAN	CHG DCH 0-15	Change D-channel.
...	...	
- IFC		Interface type for D-channel.
	ESGF	ETSI QSIG interface with GF capability.
	ISGF	ISO QSIG interface with GF capability.
	EGF4	ETSI QSIG interface with GF capability (RIs 24. 2x or later).
...	...	
- DCHL	xxx	PRI2 loop number for D-channel.
...	...	
- RLS	XX	Release ID of the switch at the far end of the D-channel.
- QCHID	NO	Timeslot values 17-31 of the PRI2 loop are associated with channel number 16-30 of the Channel ID IE.
	(YES)	Timeslot values 17-31 of the PRI2 loop are associated with channel number 17-31 of the Channel ID IE (same as previous operation).
		This prompt does not appear for PRI (1.5 Mbps) links.
...	...	

ISDN QSIG Generic Functional Transport

ISDN PRI implementation

LD 16 – Configure Automatic Trunk Maintenance in Route Data Block.

Prompt	Response	Description
REQ	NEW, CHG	New, or change.
TYPE	RDB	Route Data Block.
...		
DGTP	PRI PRI2 BRI	1.5 Mbit DTI. 2.0 Mbit DTI. Basic Rate Interface.
IFC	ESGF ISGF	ESGF = ETSI QSIG interface with GF platform. ISGF = ISO QSIG interface with GF platform. The ESIG and ISIG prompt can be entered if both QSIG and QSIGGF packages are included.

LD 17 – Configuration Record.

Prompt	Response	Description
REQ	CHG	Change.
TYPE	ADAN	
ADAN	aaa aaa x	Add, Move or Change Input/Output Device
...		
- IFC	ESGF ISGF	ESGF = ETSI QSIG interface with GF platform. ISGF = ISO QSIG interface with GF platform.
PINX_CUST	xx	Customer number.
ISDN_MCNT	60-(300)-350	Layer 3 call control message count five per second time interval.

ISDN BRI implementation**LD 27** - Configure DSL for ISDN BRI trunk configuration.

Prompt	Response	Description
REQ	NEW, CHG	New, or change.
TYPE	DSL	Digital Subscriber Loop.
DSL	c dsl#	Card and DSL Number.
APPL	BRIE	Basic Rate Trunk Application.
ISDN_MCNT	20-(300)-350	Layer 3 call control message count five per second time interval.
CUST	xx	Customer number.
...		

ISDN QSIG Name Display

LD 95 – Modify the Call Party Name Display (CPND) data block.

Note: Call Party Name Display is also defined at the set level in either LD 10 or LD 11 depending on type of set.

Prompt	Response	Description
REQ	NEW	Create Call Party Name Display data block.
TYPE	CPND	Call Party Name Display data block.
CUST	xx	Customer number.
....		
REQ	NEW	Create new name strings.
TYPE	NAME	Name Strings.
CUST	xx	Customer number.
...		
DN	xxxx	Directory Number.

LD 10 – Define Class of Service Name Display Allowed on Analog (500/2500 type) sets.

Prompt	Response	Description
REQ:	NEW, CHG	Add, or Change.
TYPE:	500	Type of telephone.
CUST	xx	Customer number.
...		
CLS	CNDA	Called Party Number Display Allowed. Allows called/connected party name to be viewed by calling party. (CNDD) = Call Number Display Denied (default).

CLS	NAMA	Name Display Allowed on the far end. (NAMD) = Name Display Denied (default).
FTR	CPND	Allow Call Party Name Display name assignment.

LD 11 – Define Class of Service Name Display Allowed on Meridian 1 proprietary sets.

Prompt	Response	Description
REQ:	NEW, CHG	Add, or Change.
TYPE:	xxxx	Telephone type, where: xxxx = SL1, 2006, 2008, 2009, 2016, 2018, 2112, 2216, 2317, 2616 or 3000.
CUST	xx	Customer number.
...		
CLS	CNDA	Call Party Name Display Allowed. Allows display of original DN name on redirection. (CNDD) = Call Party Name Display Denied (default).
CLS	DNDA	Dialed Name Display Allowed. This displays the originally dialed DN Name on redirected calls. (DNDD) = Dialed Name Display Denied.
CLS	NAMA	Name Display Allowed on the far end (default). (NAMD) = Name Display Denied.

LD 12 – Define Class of Service Name Display Allowed on Attendant Consoles.

Prompt	Response	Description
REQ	NEW, CHG	Add, or Change.
TYPE	xxxx	Console type where xxxx =1250 or 2250.
TN	c u	Terminal Number.
...		
CPND	CNDA	Call Party Name Display Allowed. (CNDD) = Call Party Name Display Denied (default).

LD 27 – Define data for ISDN BRI sets.

Prompt	Response	Description
REQ	NEW, CHG	Add, or Change.
TYPE	TSP	Terminal Service Profile.
DSL	c dsl	Digital Subscriber Loop address.
USID	x	User Service Identifier.
...		
DN	xxx...x	Directory Number associated with TSP (1 to 7 digits).
...		
- CLIP	YES	Calling Line Identification Presentation for Incoming Calls Allowed. NO = Denied.
- PRES	YES	Presentation of Name and Calling Line Identification (CLID) to far end on outgoing calls allowed. NO = Denied.

LD 16 – QSIG BRI route configuration.

Prompt	Response	Description
REQ	CHG	Change.
TYPE	RDB	Route Data Block.
CUST	xx	Customer number.
...		
DTRK	xxxx	Digital Trunk Type for route.
- IFC	xxxx	Interface type for PRI route where xxxx is: ESGF = Interface ID for ETSI QSIG with GF capability. ISGF = Interface ID for ISO QSIG with GF capability.
- RCAP	NDS	Allow Name Display Services (NDS) as new remote capability.

LD 16 – Configure Remote Capability for an MCDN BRI route.

Prompt	Response	Description
REQ	CHG	Change.
TYPE	RDB	Route Data Block.
CUST	xx	Customer number.
...		
DTRK	xxxx	Digital Trunk Type for route.
- IFC	xxxx	Interface type for D-channel where xxxx is: ESGF = Interface ID for ETSI QSIG with GF capability. ISGF = Interface ID for ISO with GF capability.
- RCAP	ND3	Allow Name Display 3 (ND3) remote capability.
- NCNA	YES	Network Calling Name Allowed.

LD 17 – Configure Remote Capability for QSIG D-channel.

Prompt	Response	Description
REQ	CHG	Change.
TYPE	ADAN	Action Device and Number.
- ADAN	aaa dch x	Change or New D-channel where aaa = NEW or CHG.
...		
- IFC	SL1	Interface type for D-channel.
...		
- RCAP	NDS	Allow Name Display Services (NDS) as new remote capability.

LD 17 – Configure Remote Capability for MCDN D-channel.

Prompt	Response	Description
REQ	CHG	Change.
TYPE	ADAN	Action Device and Number.
- ADAN	aaa dch x	Change Input/Output Device.
...		
- IFC	SL1	Interface type for D-channel.
...		
- RCAP	ND3	Allow Name Display 3 (ND3) as new remote capability.

LD 57 – Define the Calling Party Privacy Flexible Feature Code for Meridian 1 proprietary and Analog (500/2500 type) sets.

Prompt	Response	Description
REQ	NEW, CHG	Add, or Change.
TYPE	FFC	Flexible Feature Code data block.
CUST	xx	Customer number.
FFCT	YES	Flexible Feature Confirmation Tone.
...		
CPP	xxxx	Calling Party Privacy Flexible Feature Code (typically *67). CPP is only prompted if the CPP package is equipped.

ISDN Name Display Enhancement

ISDN PRI implementation

LD 95 – Configure Calling Party Name Display data block for a customer.

Prompt	Response	Description
REQ	NEW CHG	Add new data. Change existing data.
TYPE	CPND	Configure the Calling Party Name Display data block.
CUST	xx	Customer number, as defined in LD 15. xx = 0-99 for Options 51C-81C. xx = 0-31 for Option 11C.
...		

LD 95 – Define a name string for a customer.

Prompt	Response	Description
REQ	NEW CHG	Add new data. Change existing data.
TYPE	NAME	Create a new Calling Party Name string.
CUST	xx	Customer number, as previously entered in LD 95 when configuring the CPND data block. xx = 0-99 for Options 51C-81C. xx = 0-31 for Option 11C.
...		
DN	xxx...x	Directory Number (1 to 7 digits) to which the name string is associated.
...		

LD 10 - Define the Calling Name Class of Service for analog (500/2500) sets.

Prompt	Response	Description
REQ:	NEW CHG	Add new data. Change existing data.
TYPE:	500	Type of analog set.
TN	l s c u	Terminal Number, for Options 51C-81C. l = loop, s = shelf, c = card, u = unit.
	c u	Terminal Number for Option 11C. c = card, u = unit.
...		
DES	d...d	Designator The response d...d represents an Office Data Administration System (ODAS) Station Designator of 1-6 alphanumeric characters.
CUST	xx	Customer number, as previously entered in LD 95. xx = 0-99 for Options 51C-81C. xx = 0-31 for Option 11C.
...		
DN	x..x	Directory Number.
- CPND	NEW CHG OUT	Calling Party Name Display Add data block. Change existing data block. Remove existing data block.
- NAME	aaaa yyyy	Calling Party Name Display name. aaaa = first name, yyyy = last name.
- XPLN	xx	Expected name length.
...		
CLS	NAMA	Allow calling name display at the far end. (NAMD) = Name Display Denied is the default.
...		

LD 11 - Define the Calling Name Class of Service for Meridian 1 proprietary sets.

Prompt	Response	Description
REQ:	NEW CHG	Add new data. Change existing data.
TYPE:	aaaa	Type of set.
TN	l s c u	Terminal Number, for Options 51C-81C. l = loop, s = shelf, c = card, u = unit.
	c u	Terminal Number for Option 11C. c = card, u = unit.
...		
DES	d...d	Designator The response d...d represents an Office Data Administration System (ODAS) Station Designator of 1-6 alphanumeric characters.
CUST	xx	Customer number, as previously entered in LD 95 when configuring the CPND data block. xx = 0-99 for Options 51C-81C xx = 0-31 for Option 11C.
...		
CLS		Enter each non-default Class of Service entry separated by a space.
	CNDA	Allow the calling name to be displayed on this set. (CNDD) = Deny calling name display is default.
	DNDA	Allow the name of the original caller to be displayed on this set, for redirected calls. (DNDD) = Dialed Name Display Denied is default.
	NAMA	Allow calling name display at the far end. (NAMD) = Name Display on other set Denied is default.
KEY	xx aaa yyyy	Telephone function key and assignments.

- CPND	NEW CHG OUT	Calling Party Name Display Add data block. Change existing data block. Remove existing data block.
- NAME	aaaa yyyy	Calling Party Name Display name. aaaa = first name, yyyy = last name.
...		

LD 12 – Define the Calling Name Class of Service for attendant consoles.

Prompt	Response	Description
REQ:	NEW CHG	Add new data Change existing data.
TYPE:	aaaa	Type of attendant console.
TN	l s c u	Terminal Number, for Options 51C-81C. l = loop, s = shelf, c = card, u = unit.
	c u	Terminal Number for Option 11C. c = card, u = unit.
SETN	l s c u	Second Terminal Number. Must have same loop, shelf and card as Terminal Number. l = loop, s = shelf, c = card, u = unit for Option 51-81C.
	c u	c = card, u = unit for Option 11C.
CUST	xx	Customer number, as previously entered in LD 95 when configuring the CPND data block. xx = 0-99 for Options 51C-81C. xx = 0-31 for Option 11C.
...		
ANUM	1-63	Attendant Number.
...		
CPND	CNDA	Allow the calling name to displayed on this console. (CNDD) = Call Party Name Display feature Denied is the default.
...		

LD 17 – Configure the method of Operation Coding for the Name Display supplementary services, for the associated QSIG D-Channel.

The value is configured as a remote capability. It may be either by Object Identifier, used for ETSI, or by Integer Value, used for ISO.

Prompt	Response	Description
REQ	CHG	Change existing data.
TYPE	ADAN	Action Device and Number.
- ADAN	CHG DCH xx	Change a D-Channel on port 0-63 (Options 51C-81C).
	CHG DCH xx	Change a D-Channel on port 0-15 (Option 11C).
...		
- IFC	ESGF ISGF	ESGF = ETSI QSIG interface with GF platform. ISGF = ISO QSIG interface with GF platform.
...		
- RCAP		Enter the Operation Coding method for the QSIG Name Display supplementary services.
	NDO NDI	NDO = Coding by Object ID (IFC should be set to ESGF). NDI = Coding by Integer Value (IFC should be set to ISGF). Precede the entry with an 'X' to delete it.
		Note: NDO and NDI are mutually exclusive.
...		

LD 57 – Configure the Flexible Feature Code for Calling Private Privacy.

Prompt	Response	Description
REQ	CHG	Change existing data.
TYPE	FFC	Flexible Feature Code data.
CUST	xx	Customer number, as previously entered in LD 95 when configuring the CPND data block. xx = 0-99 for Options 51C-81C. xx = 0-31 for Option 11C.
...		
CODE	CPP	Configure a Flexible Feature Code for Calling Party Privacy.
CPP	xxxx	Enter the Flexible Feature Code for Calling Party Privacy.
...		

ISDN BRI implementation

LD 16 - Configure the method of Operation Coding for the Name Display supplementary services, for the associated QSIG ISDN BRI trunk route.

Prompt	Response	Description
REQ	NEW	Add new data (ISDN BRI protocol group settings).
TYPE	RDB	Route Data Block.
CUST	xx	Customer number, as defined in LD 15. xx = 0-99 for Options 51C-81C xx = 0-31 for Option 11C.
ROUT	0-511 0-127	Route number, for Options 51C-81C. Route number, for Option 11C.
TKTP	aaa	Trunk type.
DTRK	YES	Digital Trunk Route.
BRIP	NO	ISDN BRI packet handler route (NO is entered, since packet data is not required).
- DGTP	BRI	Digital trunk type.
ISDN	YES	Integrated Services Digital Network.
...		
IFC	ESGF ISGF	ESGF = ETSI QSIG interface with GF platform. ISGF = ISO QSIG interface with GF platform.
...		
RCAP		Enter the Operation Coding method for the QSIG Name Display supplementary services.
	NDO NDI	NDO = Coding by Object ID (IFC should be set to ESGF). NDI = Coding by Integer Value (IFC should be set to ISGF). Precede the entry with an 'X' to delete it.
		Note: NDO and NDI are mutually exclusive.
...		

LD 27 - Configure the Calling Line Presentation (CLIP) service, and Calling Line Identification presentation option in the Terminal Service Profile, for an ISDN BRI QSIG Digital Subscriber Loop (DSL).

Prompt	Response	Description
REQ	NEW CHG	Add new data. Change existing data.
TYPE	TSP	Terminal Service Profile.
DSL	l s c dsl	Digital Subscriber Loop address, for Options 51C-81C. l (superloop) = 0-156 (must be zero or a number divisible by 4) s (shelf) = 0-1 c (card) = 0-15 dsl# (DSL location) = 0-7.
	c dsl	Digital Subscriber Loop address, for Option 11C. c (card) = 1-20. dsl# (DSL location) = 0-7.
...		
DN	xxx...x	Directory Number (1 to 7 digits) associated with the TSP.
- CLIP	(YES) NO	Allow or deny Calling Line Identification Presentation for Incoming Calls.
- PRES	(YES) NO	Allow or deny Presentation of Calling Line Identification to far end, for outgoing calls.
...		

ISDN QSIG Path Replacement

LD 15 – Configure Private Integrated Services Network Exchange Directory Number.

Prompt	Response	Description
REQ:	CHG	Change existing data.
TYPE:	NET	Networking.
CUST	xx	Customer number.
...		
ISDN	YES	Integrated Services Digital Network.
...		
- PINX_DN	xx...x	Private Integrated Services Network Exchange Directory Number, up to seven digits.

LD 16 – Configure New Path Replacement parameters for an ISDN PRI or BRI trunk.

Prompt	Response	Description
REQ	CHG NEW	Change existing data. Add new data.
TYPE	RDB	Route Data Block.
CUST	xx	Customer number.
ROUT	0-127	Route Number.
...		
DTRK	YES	Digital Trunk Type.
- DGTP	PRI BRI	PRI trunk. BRI trunk.
- IFC	ESGF ISGF	ESGF = ETSI QSIG interface with GF platform. ISGF = ISO QSIG interface with GF platform.

...		
RCAP	PRI PRO	Add Path Replacement as a remote capability. PRI = The encoding method uses Integer values. PRO = The encoding method uses Object Identifier. PRI and PRO are mutually exclusive, and cannot be configured together at the same time on the same link. Precede PRI and PRO with X to remove. The previously configured PR parameters will be valid, if PR is deactivated and then reactivated.
...		
PR_TRIGS	<CR>	Path Replacement Triggers. <CR> = If REQ = NEW, the parameters are set to the default values, which are: DIV 2 3 (two PR attempts, with a three minute delay for Diversion triggers) CNG 2 3 (two PR attempts, with a three minute delay for Congestion triggers) XCON (Connected Number is not a trigger). If REQ = CHG, the Path Replacement parameters are not modified.
	DIV xx-y CNG xx-y CON xx-y	DIV = Diversion is used to trigger Path Replacement. CNG = Congestion is used to trigger Path Replacement. A Connected number different from a called number is used to trigger Path Replacement. where: xx = 0-(2)-15, the number of Path Replacement attempts (the default is two). If 0 is entered, the number will be set to infinite. yy = 1-(3)-7, the delay, in minutes, between two consecutive Path Replacement attempts (the default is three minutes). Precede DIV, CNG, and CON with X to remove the Path Replacement trigger.
PR_RTN	YES (NO)	Retain option is (is not) supported by the far end PINX.
...		

LD 17 – Configure QSIG Path Replacement for a QSIG ETSI or ISO interface.

Prompt	Response	Description
REQ	CHG	Change existing data.
TYPE	CFN	Configuration Record.
ADAN	CHG DCH X	Change D-Channel number.
...		
- IFC	ESGF ISGF	ESGF = ETSI QSIG interface with GF platform. ISGF = ISO QSIG interface with GF platform.
...		
- RCAP	aaa	Add Path Replacement as remote capability where: PRI = Integer values as encoding method. PRO = Object identifier as encoding method. Precede with X to remove.
...		

PR_TRIGS	<CR> DIV xx-y CNG xx-y CON xx-y	Path Replacement Triggers. <CR> = If REQ = NEW, the parameters are set to the default values, which are: DIV 2 3 (two PR attempts, with a three minute delay for Diversion triggers) CNG 2 3 (two PR attempts, with a three minute delay for Congestion triggers) XCON (Connected Number is not a trigger). If REQ = CHG, the Path Replacement parameters are not modified. DIV = Diversion is used to trigger Path Replacement. CNG = Congestion is used to trigger Path Replacement. CON = A Connected number different from a called number is used to trigger Path Replacement. where: xx = 0-(2)-15, the number of Path Replacement attempts (the default is two). If 0 is entered, the number will be set to infinite. yy = 1-(3)-7, the delay, in minutes, between two consecutive Path Replacement attempts (the default is three minutes).
PR_RTN	YES (NO)	Retain option is (is not) supported by the far end PINX.

ISDN Semi Permanent Connections for Australia

Following subscription to the ISDN Semi Permanent Connections (ISPC) service from a Telecom entity, the overlays listed below are used to configure ISPC links when they are used to convey D-channel signaling or used for ISL calls.

For both cases:

- at least one Phantom DTI2 loop must be defined in LD 17; and
- a specific SICA Table must be defined in LD 73.

ISPC links used for ISL calls:

- Configure a TIE ISL Route in LD 16; and
- For each ISPC Link, a TN on the Phantom DTI2 loop must be defined in LD 14.

ISPC links used to convey D-channel signaling:

- The configuration is asymmetrical. One side of the link is defined as MASTER and the other is defined as SLAVE.
- On the MASTER SIDE:
 - Configure Dial Tone Detection Feature in LD 13;
 - Configure Direct Inward Dialing route (non-ISL) in LD 16;
 - Configure Meridian Communication Adaptor or Meridian Communication Unit in LD 11; and
 - Define one TN on Phantom DTI2 loop in LD 14
- On the SLAVE SIDE:
 - Configure Direct Inward Dialing route (non-ISL) in LD 16;
 - Configure Data Interface in LD 11; and
 - Define one TN on the Phantom DTI2 loop in LD 14.

It is assumed that the ISL D-channel has been configured (LD 16). The following settings are recommended: RS-232 for 19.2 kbps and below, HS (RS-422) for speeds above 19.2 kbps, MCA/MCU configured as 64 kbps and DTE device configuration.

Note: ISPC uses a new set of commands in LD 96 to monitor the SLAVE SIDE. These new commands are STAT ISPC lp ch, EST ISPC lp ch and RLS ISPC lp ch. Additionally, in LD 60, the STAT command identifies the channel used as ISPC link to a PBX by indicating the ISPC mnemonic.

LD 11 – Configure Data Interface.

Prompt	Response	Description
REQ:	NEW	Add.
TYPE:	aaaa	Meridian Modular Telephones (MMT) types equipped with Meridian Communication Adapter (MCA) or Meridian Communications Unit (MCU). Where aaaa = 2006, 2008, 2216, 2616 or MCU.
...		
CLS	DTA	Class of Service Data Terminal.
...		
DTAO	MCA	Data Option software for Meridian Communications Adaptor (MCA) that determines the downloaded data, system and operating parameters. The DTAO prompt appears for all Meridian Modular Telephone data ports except MCU.
...		
OPE	YES	Change data port parameters.
- PSDS	NO	Public Switched Data Service (PSDS) option.
- TRAN	SYN	Synchronous data transmission.
...		
<i>The following is for the MCA/MCU used on the MASTER side of the ISPC link.</i>		
- HOT	ON	Enable Hotline for data port.
...		
- BAUD	0-(11)-12	Data rate in bps for the data port. Baud rate 12 (64 kbps) is recommended.

...		
KEY	0 SCR yyyy	Single Call Ringing key used by Hotline, where: 0 = Key number, SCR = Single Call Ringing, yyyy = DN.
KEY	3 ADL yy z...z	Autodial key used by hotline feature, where: 3 = Key number, ADL = Autodial, yy = maximum DN length and z...z = DN preceded by route access code of the Data Interface on the ISPC SLAVE side of link.
<i>The following is for the MCA/MCU used on the SLAVE side of the ISPC link.</i>		
- HOT	OFF	Hotline feature disabled.
- AUT	ON	Autoanswer.
- BAUD	12	Data rate for 64 kbps.
...		
KEY	0 SCR yyyy	Single Call Ringing key, where: 0 = Key number, SCR = Single Call Ringing, yyyy = DN. The Autodial Key on the MASTER side must be configured to access this DN.

LD 13 – Configure Dial Tone Detection on Master Side Meridian 1

Prompt	Response	Description
REQ	NEW	Add data block for Digitone Receiver.
TYPE	aaa	Type of data block, where aaa = DTD (Dial Tone Detector) or XTD (Extended Dial Tone Detector).
TN	c u	Terminal number of tone detector card.

LD 73 – Configure Signaling Categories used for ISPC links

Prompt	Response	Description
REQ	NEW	Create new data block.
TYPE	DTI2	2.0 Mbps DTI data block.
FEAT	ABCD	ABCD Signaling category feature.
SICA	2 - 16	Signaling Category table number.
DFLT	1	Default signaling category used for default tables. For phantom DTI2 trunks choose Table 1 as default table.
Prompts for Incoming/Outgoing Calls.		
IDLE (S)	1001	Idle (sender).
IDLE (R)	1001	Idle (receiver).
Incoming Calls		
...		
SEZA (S)	N	Seize signal acknowledgment sent. For phantom DTI2 trunks, SEZA (S) is not configured.
...		
E CONN (S)	(1001)	Connect send. CONN (S) must be the same as IDLE (S) for phantom DTI2 trunks.
CONN (R)	(1001)	Connect received. E CONN (R) must be the same as IDLE (R) for phantom DTI2 trunks.
...		
C CLRB (S)	N	Clear Back send signal. For phantom DTI2 trunks, there is no C CLRB signal.
...		

<i>Outgoing Calls</i>		
...		
SEZA (R)	N	Seize signal acknowledgment received. For phantom DTI2 trunks, SEZA (R) is not configured.
...		
CONN (S)	1001	Connect send. E CONN (S) must be the same as IDLE (S) for phantom DTI2 trunks.
E CONN (R)	1001	Connect received. E CONN (R) must be the same as IDLE (R) for phantom DTI2 trunks.
...		
C CLR (R)	N	Clear Back.

LD 17 – Configure Phantom Loops .

Prompt	Response	Description
REQ	CHG	Change.
TYPE	CEQU	Change Common Equipment Parameters.
...		
- DTI2	N	2.0 Mbps Digital Trunk Interface or DTI loop number. Precede loop with an "N" to configure this loop as a phantom loop for trunks.

LD 16 – Configure Route Data Block for ISPC link.

Prompt	Response	Description
REQ	NEW	New route.
TYPE	RDB	Route Data Block.
CUST	xx	Customer number.
ROUT	0-511	Route Number.
<i>The following prompts must be answered when ISPC link is used for ISL calls</i>		
TKTP	TIE	TIE Trunk Route Type. TIE trunks must be used for ISPC links used for ISL calls.
...		
DTRK	YES	Digital Trunk Route.
...		
- DGTP	DTI2	2.0 Mbps Digital Trunk Interface. ISL route that uses Phantom Trunk TN's must be configured as DTI2 routes.
ISDN	YES	Integrated Services Digital Network. ISDN must be set to YES for ISL signaling to control ISPC link usage.
- MODE	ISLD	Route uses ISDN Signaling Link. ISDN is allowed only if ISDN = YES and ISL package 147 is equipped.
- DCH	1 - 63	D-channel Port Number that controls ISPC links usage.
- PNI	(0) - 32700	Private Network Identifier. Each customer data block must use a unique PNI when equipped with multi-customer option. It must match the PNI in the far end Customer Data Block to support Network Ring Again, Network ACD and Network Message Service features.
- IFC	SL1	Interface type for PRI route. Connection is direct to another Meridian 1 PBX.
...		

ICOG	IOA ICT OGT	Incoming and outgoing trunk, Incoming trunk only, or Outgoing trunk only.
...		
ACOD	xxxx	Trunk Route Access Code. ACOD must not conflict with numbering plan. The code entered can be up to four digits or up to seven digits, if the DNXP package 150 is equipped.
<i>The following prompts must be answered when ISPC link is used to convey D-channel signaling</i>		
TKTP	DID	Trunk Route Type. DID must be used when associated with an ISPC link used to convey D-channel signaling.
...		
DTRK	YES	Digital Trunk Route.
...		
- DGTP	DTI2	2.0 Mbps Digital Trunk Interface.
ISDN	NO	Integrated Services Digital Network PRI option.
- DSEL	DTA	Data selection. DTA is configured as "data only" when ISPC link is used to convey D-channel signaling.
ICOG	IOA OGT ICT	Incoming and outgoing trunk, Outgoing trunk only, or Incoming trunk only. ICOG configuration is dependent on the side of the ISPC link with which the route is to be associated. On the MASTER side (SMAS = YES in LD 14), ICOG = OGT or IAO. On the SLAVE side (SMAS = NO in LD 14), ICOG = ICT or IAO.
...		
ACOD	xxxx	Trunk Route Access Code. ACOD must not conflict with numbering plan. The code entered can be up to four digits or up to seven digits, if the DNXP package 150 is equipped.

...		
CNTL	YES	Change to Controls or Timers. CNTL must to answered to trigger DTD prompt.
...		
- DTD	YES	Dial Tone Detection provided. On MASTER side of ISPC link (SMAS = YES 0n LD 14)
...		
NEDC	ETH	Near End Disconnect Control.
FEDC	ETH	Far End Disconnect Control.
CPDC	NO	Meridian 1 is not the only controlling party on incoming calls.
DLTN	NO	Dial Tone on originating call provided on this route. On SLAVE side of ISPC link (SMAS = NO in LD 14)
...		
PRDL	BSY	Busy signal is sent on time-out. The PRDL prompt must be set to "BSY" in order to disconnect ISPC phantom DT12 trunks when an incomplete or invalid DN is received on the ISPC SLAVE D-channel.

LD 14 – Configure Phantom Trunk Terminal Numbers.

Note: ISPC Reference Numbers supplied by Telecom entity must be associated with Phantom Trunk TNs defined in LD 14.

Prompt	Response	Description
REQ	CHG, END, NEW x, OUT x	Change existing data block, exit overlay program, add new data block (1-255) or remove data block (1-255).
<i>The following answers are required when ISPC link is used for ISL calls.</i>		
TYPE	TIE	Trunk type.
TN	c u	Terminal Number.
...		

SICA	1 - 16	Signaling Category Table Number configured in LD 73.
...		
RTMB	0-127 1-126	Route number, Member number.
CHID	1-382	ISL Channel Identification for trunk. A different channel is required for each phantom TN used for the ISPC links. The same Channel ID must be configured for the ISPC reference number on both Meridian 1 PBXs linked by the connection.
...		
TKID	nnnnnnnn	Trunk Identifier.
SREF	XXXXXXX	Semi Permanent Connection Reference Number as defined by the Telecom administration at subscription. Reference number can be a maximum of seven digits. SREF is prompted only when Trunk TN is on a phantom loop.
SDCH	NO	The ISPC link is not used to convey D-channel signaling.
...		
<i>The following answers are required when ISPC link is used to convey D-channel signaling</i>		
TYPE	DID	Trunk type.
...		
SICA	1 - 16	Signaling Category Table Number configured in LD 73.
...		
RTMB	0-127 1-126	Route number, Member number. ISPC link requires ISL TIE Route.
...		
CLS	DTN	Digitone Class of Service. CLS = DTN for ISPC link to convey D-channel signaling.

...		
SREF	yy..yy	ISPC Reference Number(s) supplied by Telecom entity to a maximum of 7 digits.
SDCH	YES	ISPC link used to convey D-channel signaling.
SMAS	(NO) YES	Data Interface over the ISPC link is MASTER or SLAVE. YES = ISPC link is on MASTER side. NO = ISPC link is on SLAVE side. One side of the ISPC link must be configured with SMAS = YES and the other side with SMAS = NO.

LD 96 – D-Channel Message Monitoring

Commands	Description
STAT ISPC l ch	Get status of data interface establishment process at an ISPC SLAVE side of an ISPC link (where l = loop and ch = channel) which is configured to convey D-channel signaling. The status may be: 1. DISABLED = ISPC link is disabled 2. AWAITING ESTABLISHMENT = the data interface establishment process is waiting to receive DTMF digits. While waiting, dial tone is provided to the ISPC link at the SLAVE side. 3. RELEASED = the data interface establishment process is stopped because either an “RLS ISPC” command has been entered or because the maximum number of tries has been reached. 4. ESTABLISHED = data interfaces at both ends of the ISPC link are established. 5. AWAITING RELEASE = the ISPC link is being disconnected. Once disconnected, the link will transition to the “RELEASED” state. The mode may be: 1. MODE MANUAL N = the data interface establishment process is running with up to “N” number of tries 2. MODE AUTOMATIC = the Meridian 1 provides dial tone to the ISPC slave D-channel.

RLS ISPC l ch	Establish data interface process at the ISPC SLAVE side of an ISPC link at loop (l) and channel (ch), where “N” = the “number of tries” counter. The actual range of N = 0-(1)-255. When N = 0, the mode is set to AUTOMATIC. This mode requires one TDS slot and one DTR unit. This mode is not recommended for use on small systems or systems loaded with heavy traffic. When N > 0, the process is activated for a period of up to N * 30 minutes. The available range goes from 30 minutes (N = 1) to 128.5 hours or 5 days (N = 255). If no value is entered for N, it defaults to a value of 1. When N = 1, one attempt will be made to establish the data interface process before the ISPC slave D-channel is released. Stop the data interface establishment process. When this command is entered, the mode becomes MANUAL and the number of attempts to be performed = 0.

ISDN Signaling Link

Configuring ISL

To configure ISL, perform the following steps:

- 1** LD17 Configuration Record
Configure the D-channel for ISL use.
X11 RELEASE 17 AND EARLIER
- OR*
- 2** LD17 Configuration Record
Configure the D-channel for ISL use.
X11 RELEASE 18 AND LATER
- 3** LD15 Customer Data Block
Define a PRI customer
- 4** LD16 Route Data Block
Enable the ISL option on a per route basis
- 5** LD 14
Assign a D-channel for each route

LD 17 – Configure the D-Channel for ISL (X11 Release 17 and earlier).

Prompt	Response	Description
REQ	CHG	
TYPE	CFN	Configuration record
...		
PAM		
ISDN	YES, (NO)	
DCHI	1-15	DCHI port number (remove with Xnn)—only odd numbers allowed
BCHI	1-15	Back-up DCHI port number
USR		Change DCHI parameters
	PRA	D-channel for ISDN PRI only
	ISLD	D-channel for ISL in “dedicated” mode
	SHA	D-channel for ISL in “shared” mode, used for both ISDN PRI and
		Note 1: One of the above responses is mandatory and there is no default.
		Note 2: Package 147, ISDN Signaling Link, is required.
DCHL	0-159	PRI loop for DCHI, not prompted with D-channel in dedicated mode (D-channel does not require a channel on the PRI)
BCHL	0-159	PRI loop number for BCHI
OTBF	1-(16)-127	Number of output request buffers—use default (16)
DRAT		D-channel transmission rate
	56K	56 Kbps, default when LCMT (LD17) is AMI
	64KC	64 Kbps clear, default when LCMT (LD17) is B8S
	64KI	64 Kbps inverted HDLC (64 Kbps restricted)

PRI	loop ID	In X11 release 15 and later DRAT is not prompted when configuring ISL D-channels because the speed is controlled by the modem baud rate Not prompted if the D-channel is in dedicated mode (the D-channel does not require a channel on the PRI card)
ISLM	1-382	Number of ISL trunks controlled by the D-channel (no default value)
CLOK	(EXT), INT	D-channel clock type EXT = Source of primary clock is external to DCHI card. Normally, EXT is used with PRI/ISL. INT = Source of primary clock is internal to DCHI card. INT is used only during DCHI loop back testing, where one side is set to INT, the other side is set to EXT.

LD 17 – Configure the D-Channel for ISL (X11 Release 18 and later).

Prompt	Response	Description
REQ	CHG	Change
TYPE	CFN	Configuration Record
ADAN	NEW DCH 0-63	Add a primary D-channel (can also CHG and OUT DCH)
CTYP	MSDL	Card type, where MSDL = the Downloadable D-Channel Daughterboard for Option 11C.
- PORT	1	Port number on the Downloadable D-Channel Daughterboard.
USR	PRI SLD SHA	D-channel mode
- ISLM	1-382	Number of ISL trunks controlled by the D-Channel
DCHL	0, 2, 4,...159	PRI loop number for DCHI
OTBF	1-(32)-127	Number of output request buffers

- BPS	xxxx	Baud rate for ISL D-channel (default 64000)
- PARM	R232 (R422) DCE (DTE)	ISL D-channel interface and transmission mode
PRI	0-159 2-15	Additional PRI loops using the same D-channel, and interface ID
DRAT	(56K), 64KC, 64KI	D-channel transmission
IFC	aaa	Interface type: (D100), D250, ESS4, ESS5, SL1, S100
SIDE	NET (USR)	Meridian 1 node type
RLS	xx	Release ID of the switch at the far end of the D-Channel
RCAP	aaa	Remote D-channel capabilities: MSL, NCT, ND1, ND2, RVQ
- CLOK	INT, (EXT)	Internal or external clock on ISL D-channels
LAPD	Yes, (No)	D-channel LAPD parameters
- T23	1-(20)-31	Interface guard timer (DCHI only)
- T200	2-(3)-40	Retransmission timer
- N200	1-(3)-8	Maximum number of retransmissions
- N201	4-(260)	Maximum number of octets in information field
- T203	2-(10)-40	Maximum time (in seconds) without frames being exchanged
- K	1-(7)	Maximum number of outstanding frames
ADAN	NEW BDCH 0-63	Add a backup D-channel
PDCH	0-63	Primary D-channel
CTYP	MSDL	Card type, where MSDL = Downloadable D-channel daughterboard.
- PORT	1	Port number on the Downloadable D-Channel Daughterboard for Option 11C.

BCHL	0,2,4,...,-159	PRI loop number for back-up D-channel
RCVP	Yes, (No)	Auto-recovery to primary D-channel option
ADAN	<cr>,****	Go on to next prompt or exit overlay

LD 15 – Define a PRI customer and networking data.

Prompt	Response	Description
REQ	NEW, CHG	Add or change a customer
TYPE	NET	Networking
CUST	0-31	Customer number
...		
- ISDN	YES	ISDN
- PNI	1-32700	Private Network Identifier (for CLID)
- PFX1	xxxx	Prefix (area) code for International PRA (for CLID)
- PFX2	xxxx	Central Office Prefix for IPRA (for CLID)
- HNPA	100-319	Home Number Plan Area code (for CLID)
- HNXX	100-319	Prefix for Central Office (for CLID)
- HLOC	100-3199	Home Location Code (ESN) (for CLID)
- LSC	xxxx	Local steering code (for CLID)
- CNTP	LDN, (PDN)	Default for Calling Line ID (CLID)
- RCNT	0-(5)	Maximum inter-node hops in a network redirection call

LD 16 – Enable ISL on a per route basis.

Prompt	Response	Description
REQ	NEW	Add a PRI trunk route
TYPE	RDB	Type of data
CUST	0-31	Customer number
ROUT	0-511	Route number
TKTP	ISL	Trunk route type:
ESN	Yes, (No)	ESN signaling
BRIP	Yes, (No)	ISDN BRI packet handler route
ISDN	Yes, (No)	ISDN PRI option
- MODE	ISLD	Mode of D-Channel that controls the route
- DCHI	1-15	DCHI port number Prompted if Mode = ISLD
- PNI	0-32700	Customer Private ID (unique to a customer)
- IFC	SL1	Interface machine type for this PRI route
- NCNA	Yes	Network call name allowed
- NCRD	Yes	Network Call Redirection
- TRO	(No), Yes	Trunk optimization allowed (denied) on the route
- NSF	(No), Yes	Network Service Facility
- TIER	0-511	Incoming Tie route number
- CHTY	ABCH, (BCH)	Signaling type for B-channel digital routes
- CTYP	aaa	Call type for outgoing direct dialed Tie route
- INAC	Yes, (No)	Insert ESN access code to incoming private network call
- NCOS	(0)-99	Network Call Of Service
- CLS	aaa	Class of service restriction for tie route
- TGAR	0-31	Trunk group access restrictions

- IEC	000-319	Interexchange carrier ID
PTYP	aaa	Port type at far end
ICOG	IAO, ICT, OGT	Incoming and/or outgoing trunk
SIGO	aaa	Signaling arrangement
NEDC	ORG, ETH	Near end disconnect control
FEDC	ETH, FEC, JNT,(ORG)	Far end disconnect control
CPDC	Yes, (No)	Meridian 1 is the only controlling party on incoming calls
DLTN	Yes, (No)	Provide dial tone to the far-end
PSEL	TLNK, (DMDM)	T-link or DM-DM protocol selection
EQAR	Yes, (No)	Enable Equal Access on this route
- GCR	(Yes), No	Use General Carrier Restriction for Equal Access calls
- NTOL	ALLOW, (DENY)	Allow or deny North American toll calls (1+ calls)
- ITOL	ALLOW, (DENY)	Allow or deny international toll calls (011+ calls)
- SCR	Yes, (No)	Use Selective Carrier Restriction for Equal Access calls
AUTH	Yes, (No)	Authcode to be prompted for incoming tie callers

LD 14 – Assign a channel identification to each trunk with the ISL option.

Prompt	Response	Description
REQ	NEW	
TYPE	TIE	Trunk type allowed
TN	loop ch	Use loop number created in LD17
RTMB	0-511 1-254	The route defined as an ISA route in LD16
CHID	1-382	Channel identifier for ISL channels (remove with Xnn)—must be coordinated with far end (no default value)

Japan D70 nB+D

LD 17 – Configure nB+D for D70.

Prompt	Response	Description
REQ	CHG	Change.
TYPE	CFN	Configuration Record.
ADAN	NEW DCH 0-15	Add a primary D-channel on logical port.
CTYP	aaaa	Card type.
USR	PRI	D-channel mode.
IFC	D70	Interface type for Japan D70.
DCHL	1-9 1-126	PRI loop number and interface identifier for the DCHI when IFC=D70.
...		
OTBF	1-(32)-127	Number of output request buffers.
PRI	1-9 1-126	Loop number and interface ID. Note that the interface ID range is larger when IFC=D70.
	<cr>	End definition of PRI loops.
DRAT	(56K), 64KC, 64KI	D-channel transmission.
...		
ADAN	NEW BDCH 1-15	Add a primary D-channel on logical port.
PDCH	0 -15	Primary D-channel.
CTYP	MSDL	Card type, where MSDL = Downloadable D-Channel Daughterboard.
BCHL	1-9 1-126	PRI loop number and interface identifier for the DCHI when IFC=D70.
...		

Japan Telecommunication Technology Committee (JTTC) Common Channel Signaling

LD 73 – Define a new Digital Data Block (DDB) with a defined threshold.

Prompt	Response	Description
REQ	NEW	Add new data.
TYPE	DDB	Digital Data Block.
CLKN	1-9	Card number for Clock Controller.
- PREF	1-9	Primary Reference Source card for Clock Controller. Enter the PRI loop number or card number that is defined at the DLOP prompt in LD 17. This prompt must be defined for the SLAV Meridian 1 only.
- - SREF	1-9	Secondary Reference (prompted only when PREF is not free-run). Source card for Clock Controller (Option 11C). This prompt must be defined for the SLAV Meridian 1 only.
TRSH	0-15	Threshold set.

LD 17 – Configure a digital loop number for Japan TTC.

Prompt	Response	Description
REQ	CHG	Change existing data.
TYPE	CEQU	Common Equipment parameters.
...		
- DLOP	c dd ff	Digital Trunk Interface Loop, where: c = DTI/PRI card number (1-9) dd = number of voice or data calls (0-24) ff = frame format D2, D3, D4, or ESF
- MODE	PRI	Primary Rate Interface mode.
- - TRSH	xx	Digital Trunk Interface threshold. The TRSH value must match the value defined in LD 73.

LD 17 – Configure a new D Channel with Japan TTC Common Channel Signaling interface.

Prompt	Response	Description
REQ	CHG	Change existing data.
TYPE	ADAN	Action Device and Number.
ADAN	NEW DCH x	Create a new D-channel. x = 0-15
- CTYP	MSDL	Card type, where MSDL = the Downloadable D-Channel Daughterboard for Option 11C.
- PORT	1	Port number on the Downloadable D-Channel Daughterboard.
- USR	PRI ISLD	D Channel mode. Primary Rate Interface. Integrated Services Digital Link Dedicated.
- IFC	JTTC	Interface ID for Japan TTC.
- -ISDN_MCNT	60-(300)-350	Layer 3 call control message count per 5 second time interval.
- - ISLM	1-382	Integrated Services Signaling Link Maximum.
- -DCHL	x	PRI loop number for D-channel, as defined in LD 17 at the DLOP prompt.
...		
- BPS	xxxxx	Asynchronous baud rates (bits per second), where xxxxx is: 1200, 2400, 4800, 9600, 19200, 48000, 56000, 64000.
- SIDE	(USR) NET	Meridian 1 node type. Slave to the controller. Network, the controlling switch.
- RLS	23	Release ID of the switch at the far-end of the D-channel. Supported on Release 23 and later.

LD 16 – Configure Japan TTC interface.

Prompt	Response	Description
REQ	NEW CHG	Add new data. Change existing data.
TYPE	RDB	Route Data Block.
CUST	xx	Customer number.
ROUT	0-127	Route Number.
TKTP	TIE	TIE trunk type. Used for PBX-to-PBX interface.
DTRK	YES	Digital trunk route.
- DGTP	PRI	Digital trunk type.
ISDN	YES	Integrated Services Digital Network.
- MODE	PRA ISLD	Mode of operation. D-channel mode is Primary Rate Interface. D-channel mode is Integrated Services Digital Link Dedicated.
- DCH	0-159	D channel number, as defined in LD 17.
...		
- IFC	JTTC	Interface for Japan TTC.
- - PNI	(0)-32700	Private Network Identifier. Must match far end PBX.

LD 14 – Configure trunks for Japan TTC.

Prompt	Response	Description
REQ	NEW CHG	Add new data. Change existing data.
TYPE	TIE	TIE trunk data block.
TN	c u	Loop and channel for digital trunk.
CUST	xx	Customer number, as defined in LD 15.
SICA	(1)-16	Signaling category table number. The default is 16 if the loop type = JDML.
RTMB	0-127 1-254	Route number and member number. The Route number is specified in LD 16.
TGAR	0 - (1) - 31	Trunk Group Access Restriction. The default value (1) automatically blocks direct access.

MCDN Alternate Routing

LD 86 – Configure the MCDN Alternate Routing options.

Note: To ensure that MCDN Alternate Routing occurs for all nodes in a system, define all nodes as SBOC = RRA. Conversely, to prevent MCDN Alternate Routing from occurring on any nodes, define all nodes as SBOC = NRR.

Prompt	Response	Description
REQ	CHG	Change existing data.
CUST	xx	Customer number, as configured in LD 15. xx = 0-99 for Options 51C-81C. xx = 0-31 for Option 11C.
FEAT	RLB	Configure the Route List as a feature.
RLI	xxx	Route List Index to be accessed. xxx = 1-127 if a Coordinated Dialing Plan or Basic Alternate Route Selection is used. xxx = 1-255 if Network Alternate Route Selection is configured. xxx = 0-999 if the Flexible Numbering Plan is configured.
ENTR	0-63	Entry number for the NARS or BARS route list.
ROUT	0-511 0-127	Route Number for Options 51C-81C. Route Number for Option 11C.
SBOC		Step Back On Congestion option. (NRR) = No re-routing
	RRO	Re-route at the originating node if congestion is encountered. If congestion is encountered at a transit node, drop back to the preceding node so that the preceding node determines if re-routing is needed.
	RRA	Re-route at any node, originating or transit (tandem), when congestion is encountered.
...		
ISET	(0)-64	Initial Set. Number of entries in Initial Set for route list block.

NALT	1-(5)-10	Number of alternate routing attempts. Prompt appears once per RLI.
MFRL	(MIN) 0-7	Set Maximum Facility Restriction Level used to determine autocode prompting. Use default of MiN to set to the minimum FRL value.

MCDN End to End Transparency

To implement the MCDN End to End Transparency feature, follow these steps:

- Configure a D-Channel for a PRI interface using the QSIG transport and select the NAS, NACD and/or NMS MCDN features In LD 17.

or

- Configure a route for an ISDN BRI interface using the QSIG transport and add the NAS, NACD and/or NMS MCDN features in LD 16.

After selecting NAS in LD 17 or LD 16, to configure NAS for a QSIG link, follow these steps:

- Configure attendant consoles with NAS key in LD 12 (optional).
- Enable or disable network attendant control, NAS control and define trunk ICI keys and NAS routing thresholds in LD 15.
- Define the NAS routing table in LD 86.

After selecting NACD in LD 17 or LD 16, to configure NACD for a QSIG link, follow these steps:

- Define a CDP or UDP between the two nodes in LD 87 and LD 90.
- Configure a CLID table in LD 15.
- Configure the ACD DN queue at source and target nodes in LD 23.
- Configure the NACD routing table in LD 23.
- Define a Meridian 1 proprietary ACD set at source and/or target node in LD 11.

After selecting NMS in LD 17 or LD 16, to configure NMS-MC for a QSIG link, follow these steps:

- Define a CDP (DSC or LSC) or UDP between the two nodes in LD 87 and LD 90.
- Allow Message Waiting Center access in LD 15.
- Define a Meridian 1 proprietary set at source and/or target node in LD 11.
- Define a Meridian 1 proprietary NMS-MC user set in LD 11.

After selecting NMS in LD 17 or LD 16, to configure NMS-MM for a QSIG link, follow these steps:

- Configure the primary voice messaging ACD queue at the prime location (where Meridian Mail is installed) in LD 23.
- Add agents to the primary agent queue in LD 11.
- Configure ACD parameters for all voice service queues in LD 23.
- Configure the Voice Services DN (VSDN) table in NMS-MM administration terminal. Refer to Meridian Mail documentation for this configuration information.

ISDN PRI implementation using the QSIG transport

LD 17 – Configure a D-Channel for an ISDN PRI interface using the QSIG transport and add MCDN features.

Prompt	Response	Description
REQ	CHG	Change existing data.
TYPE	CFN	Configuration database.
...		
ADAN	CHG DCH aaa	Change D-Channel information.
...		
- IFC	ESGF ISGF	ESGF = ETSI QSIG interface with GF platform. ISGF = ISO QSIG interface with GF platform.
...		
- RCAP	MQC	Add MCDN QSIG Conversion as a new remote capability. XMQC removes MCDN QSIG Conversion as a remote capability.
-- MQC_FEAT		MCDN QSIG feature type. Prompted if RCAP = MQC. Precede MQC feature type with X to remove.
	NAS	Enable NAS on QSIG. XNAS disables NAS on QSIG.
	NACD	Enable NACD on QSIG. XNACD disables NACD on QSIG.
	NMS	Enable NMS on QSIG. XNMS disables NMS on QSIG.

ISDN BRI implementation using the QSIG transport

LD 16 – Configure a route for an ISDN BRI interface using the QSIG transport and add MCDN features.

Prompt	Response	Description
REQ	NEW CHG	Add new data. Change existing data.
TYPE	RDB	Route Data Block.
TKTP	a...a	Trunk Type.
...		
DTRK	(NO) YES	Digital Trunk Route.
- BRIP	(NO) YES	ISDN BRI Packet handler route.
- DGTP	a...a	Digital Trunk Type for route.
	BRI	Basic Rate Interface (Allowed if TKTP = TIE, COT or DID and BRIP = NO.)
...		
- IFC	ESGF ISGF	ESGF = ETSI QSIG interface with GF platform. ISGF = ISO QSIG interface with GF platform.
...		
- RCAP	MQC	Add MCDN QSIG Conversion as a new remote capability. XMQC removes MCDN QSIG Conversion as a remote capability.
-- MQC_FEAT		MCDN QSIG feature type. Prompted if RCAP = MQC. Precede MQC feature type with X to remove.
	NAS	Enable NAS on QSIG. XNAS = disable NAS on QSIG.
	NACD	Enable NACD on QSIG. XNACD = disable NACD on QSIG.
	NMS	Enable NMS-MC and NMS-MM on QSIG. XNMS = disable NMS-MC and NMS=MM on QSIG.

NAS implementation using the QSIG transport

To configure NAS using the QSIG transport, follow these steps:

- Configure an ISDN PRI interface using the QSIG transport and select NAS as an MCDN feature in LD 17, or configure an ISDN BRI interface using the QSIG transport and select NAS as an MCDN feature in LD 16.
- Configure NAS for a QSIG link:
 - Configure attendant consoles with NAS key (optional) in LD 12.
 - Enable or disable network attendant control, NAS control and define trunk ICI keys and NAS routing thresholds in LD 15.
 - Define the NAS routing table in LD 86.

LD 12 – Configure Attendant Consoles with NAS keys. (optional)

Prompt	Response	Description
REQ	NEW CHG	Add new data. Change data.
TYPE	ATT aaaa	Type of console. aaaa = 1250 or 2250.
TN	l s c u	Terminal Number. l = loop, s = shelf, c = card, u = unit for Options 51-81C. c = card, u = unit for Option 11C.
...		
KEY	xx NAS	xx = the key number assigned the NAS function. Each attendant console can have only one NAS key defined. This key is optional.

LD 15 – Enable or disable network attendant control, NAS routing, and define a trunk ICI key.

Prompt	Response	Description
REQ:	NEW CHG	Add new data. Change existing data.
TYPE:	CDB	Customer Data Block.
TN	I s c u	Terminal Number. I = loop, s = shelf, c = card, u = unit for Options 51-81C. c = card, u = unit for Option 11C.
...	...	...
- ISDN	YES	Integrated Services Digital Network.
...	...	...
- ICI	0-19 NCO NDID NTIE NFEX NWAT	Key number. Followed with a space and the trunk type. Network CO trunk. Network DID trunk. Network TIE trunk. Network FEX trunk. Network WAT trunk.
...	...	...
- CWCL	(0)-255 (0)-255	Call Waiting Call Limit. Lower and upper thresholds.
- CWTM	(0)-511 (0)-511	Call Waiting Time. Lower and upper thresholds (in seconds).
...	...	...
- NAS ATCL	YES (NO)	Allow/deny attendant control for call extension.
- NAS ACTV	YES (NO)	Allow/deny NAS routing.

LD 86 – Define NAS routing tables.

Prompt	Response	Description
REQ	NEW CHG	Add new data. Change existing data.
CUST	xx	Customer number. xx = 0-99 for Options 51C-81C. xx = 0-31 for Option 11C.
FEAT	NAS	Type of data.
TBL	0-63	Routing table number. Without Multi-Tenant Service, 0 is the customer routing table. With Multi-Tenant Service enabled, NAS tables 1-63 may be associated with Console Presentation Groups (CPGs) 1-63.
ALT	1-7	An alternative attendant or routing table. (To clear an old number, type an X before typing the new number. The old number cannot be cleared if it is associated with a schedule period. Reach TODS by pressing the return key.)
ID	xx...x <CR>	The dialed digits (including the network access code) needed to reach an attendant associated with the alternative number. Respond with a string of up to 16 digits to change the attendant ID. Press the return key to leave the ID unchanged, exit the prompt, and return to ALT.
TODS	0-31	Time of Day schedule where; 0 = default to handle all time periods not defined in 1 through 31. Press the return key to continue the NAS feature setup process.

- PER	hr: mm hr: mm <CR>	Specify start and stop times for the period using 24-hour format. Start time must be before stop time. mm = 00 or 30 only. <CR> = leave times unchanged; move to the DAYS prompt.
- DAYS	a,a...a	Specify applicable days of the week for the time period. Input a number representing each day for which the schedule is active (where 1=Monday, 2=Tuesday... 7=Sunday).
ALST	1-7	Alternatives list to be used for the schedule period.
DBK	(NO) YES	Disable/enable Drop Back busy option.
QUE	(NO) YES	Disable/enable queuing to a route.

NACD implementation using the QSIG transport

To configure NACD using the QSIG transport, follow these steps:

- Configure an ISDN PRI interface using the QSIG transport and select NACD as an MCDN feature in LD 17, or configure an ISDN BRI interface using the QSIG transport and select NACD as an MCDN feature in LD 16.
- Configure NACD for a QSIG link:
 - Define a CDP or UDP between the two nodes.
 - Configure a CLID table in LD 15.
 - Configure the ACD DN queue at source and target nodes in LD 23.
 - Configure the NACD routing table in LD 23.
 - Define a Meridian 1 proprietary ACD set at source and/or target node in LD 11.

LD 15 – Configure a CLID table.

Prompt	Response	Description
REQ:	NEW CHG	Add new data. Change existing data.
TYPE:	NET	Networking data (if REQ = CHG only).
CUST	xx	Customer number. xx = 0-31 for Option 11C.
...		
- CLID	YES (NO)	CLID option. YES = configure a CLID table for the customer. NO = (the default) do not configure a CLID table. Remaining prompts are not generated and no CLID is sent for the customer.
...		
-- ENTRY	y	CLID entry number.

LD 23 – Configure ACD Directory Number queues at source and target nodes.

Prompt	Response	Description
REQ	NEW CHG	Add new data. Change existing data.
TYPE	ACD	Automatic Call Distribution data block.
CUST	xx	Customer number. xx = 0-31 for Option 11C.
...		
ACDN	xxxx	ACD Directory Number.

LD 23 – Configure an NACD Routing Table.

Prompt	Response	Description
REQ	NEW CHG	Add new data. Change existing data.
TYPE	aaaa	Type of data block. Enter NACD for Network ACD.
CUST	xx	Customer number. xx = 0-31 for Option 11C.
ACDN	xxxx	ACD Directory Number.
TABL	a	Day or Night Table. a = D or N.
- OUTS	xxxx xxxx	Routing Table entries to be removed.
- TRGT	xxxx tttt	Target ACD DN and the timer in seconds.

LD 11 – Define a Meridian 1 proprietary ACD set at source and/or target node.

Note: When assigning a CLID entry to an ACD set, you cannot use the position ID already on the set. You must out the set first or null the ACD key and then rebuild with the table entry number..

Prompt	Response	Description
REQ:	NEW CHG	Add new data. Change data.
TYPE:	aaaa	Type of set. aaaa = SL1, 2006, 2008, 2009, 2016, 2018, 2112, 2216, 2317, 2616, 3000.
TN	c u	Terminal Number. c = card, u = unit for Option 11C.
...		
CLS	(AGN) SPV	Class of Service group. AGN = (default) ACD Agent. SPV = ACD Supervisor.
...		
KEY	xx ACD yyyy 0-N/D zzzz	ACD key, where: xx = key number (<i>must be key 0</i>). yyyy = ACD DN or Message Center DN. 0-N = CLID entry, with N = CLID SIZE-1 (SIZE defined in LD 15). D = CLID entry, Search for a CLID entry from key 0 upwards, to find a DN key. the found CLID is used as the CLID entry for the active DN key. zzzz = ACD agent's position ID. Please refer to the note at the top of LD 11 on assigning a CLID entry to an ACD set.
KEY	xx MSB yyyy (0)-N/D xx NRD yy (0)-N/D	Telephone function key assignments. xx = key number. MSB = Make Set Busy key. NRD = Not Ready key.

LD 10 – Define an analog (500/2500) set at source and/or target node.

Prompt	Response	Description
REQ:	NEW CHG	Add new data. Change existing data.
TYPE:	aaaa	Type of set. aaaa = analog (500 or 2500) set.
TN	c u	Terminal Number. c = card, u = unit for Option 11C.
...		
CLS	AGTA	ACD services for analog (500/2500) sets allowed.
...		
FTR	ACD xxxx 0-N/D zzzz	ACD feature where: aaa = ACD. xxxx = ACD DN. 0-N/D = CLID entry. zzzz = ACD agent's position ID.

NMS implementation using the QSIG transport

To configure NMS-MC using the QSIG transport, follow these steps:

- Configure an ISDN PRI interface using the QSIG transport and select NMS as an MCDN feature in LD 17, or configure an ISDN BRI interface using the QSIG transport and select NMS as an MCDN feature in LD 16.
- Configure NMS-MC for a QSIG link:
 - Define a CDP (DSC or LSC) or UDP between the two nodes.
 - Allow Message Waiting Center access in LD 15.
 - Define a Meridian 1 proprietary ACD agent set at source and/or target node in LD 11.
 - Define a Meridian 1 proprietary NMS-MC user set in LD 11.

LD 15 – Allow Message Waiting Center access.

Prompt	Response	Description
REQ:	NEW CHG	Add new data. Change existing data.
TYPE:	FTR_DATA	Feature data.
CUST	xx	Customer number. xx = 0-31 for Option 11C.
OPT	(MCX) MCI	Message Center excluded. Message Center included.
	(MWUD) MWUA	Message Waiting Unconditional Denied. Message Waiting Unconditional Allowed.

LD 11 – Define a Meridian 1 proprietary set at source and/or target node.

Prompt	Response	Description
REQ:	NEW CHG	Add new data. Change existing data.
TYPE:	aaaa	Type of set. aaaa = SL1, 2006, 2008, 2009, 2016, 2018, 2112, 2216, 2317, 2616, 3000.
TN	c u	Terminal Number. c = card, u = unit for Option 11C.
...		
CLS	(MWD) MWA	Class of Service options. MWD = Message Waiting Denied MWA = Message Waiting Allowed.
...		
KEY	xx MIK xx MCK	Telephone function key assignments. xx = key number. MIK = Message Indication Key. MCK = Message Cancellation Key.

LD 11 – Define a Meridian 1 proprietary NMS-MC user set.

Prompt	Response	Description
REQ:	NEW CHG	Add new data. Change existing data.
TYPE:	aaaa	Type of set. aaaa = SL1, 2006, 2008, 2009, 2016, 2018, 2112, 2216, 2317, 2616, 3000.
TN	c u	Terminal Number. c = card, u = unit for Option 11C.
CLS	AGTA	ACD services for analog (500/2500) sets allowed.
KEY	xx MWK x...x	Telephone function key assignments where: xx = key number. MWK = Message Waiting Key x...x = NMS-MC DN.

To configure NMS-MM on QSIG, follow these steps:

- Configure an ISDN PRI interface using the QSIG transport and select NMS as an MCDN feature in LD 17, or configure an ISDN BRI interface using the QSIG transport and select NMS as an MCDN feature in LD 16.
- Configure NMS-MM for a QSIG link:
 - Configure the primary voice messaging ACD queue at the prime location (where Meridian Mail is installed) in LD 23.
 - Add agents to the primary agent queue in LD 11.
 - Configure ACD parameters for all voice service queues in LD 23.
 - Configure the Voice Services DN (VSDN) table in NMS-MM administration terminal.

LD 23 – Create primary voice messaging ACD queue.

Prompt	Response	Description
REQ	NEW	Add new data.
TYPE	ACD	Type of data block. ACD = Automatic Call Distribution data block.
CUST	xx	Customer number. xx = 0-31 for Option 11C.
ACDN	xxxx	ACD DN = Express Messaging DN (the DN which users dial to access their mailboxes).
MWC	YES	ACD DN is a message center DN.
NCFW	xx...x	Night Call Forward DN (up to 23 digits).
...		

LD 11 – Add agents to the primary agent queue.

Note: When assigning a CLID entry to an ACD set, you cannot use the position ID already on the set. You must out the set first or null the ACD key and then rebuild with the table entry number.

Prompt	Response	Description
REQ:	NEW	Add new data.
TYPE:	SL1	ACD data block.
TN	c u	Terminal Number. c = card, u = unit for Option 11C.
...		
CLS	VMA	Allow voice messaging.
...		
KEY	xx ACD yyyy 0-N/D zzzz	ACD key, where: xx = key number (<i>must be key 0</i>). yyyy = ACD DN or Message Center DN. 0-N = CLID entry, with N = CLID SIZE-1 (SIZE defined in LD 15). D = CLID entry, Search for a CLID entry from key 0 upwards, to find a DN key. the found CLID is used as the CLID entry for the active DN key. zzzz = ACD agent's position ID. Please refer to the note at the top of LD 11 on assigning a CLID entry to an ACD set.

	xx SCN yyyy (ccc or D).	Single Call Non-Ringing DN where: yyyy = DN ccc = CLID entry of (0)-N, where N = the value entered at the SIZE prompt in LD 15 minus 1. D = the character D may be entered to search a CLID entry from key 0 and up to find a DN key. The CLID associated with the found DN key will then be used. The DN can be up to 4 digits, up to 7 digits with Directory Number Expansion (DNXP) package 150. With X11 Release 18 and later, once the SCN key has been defined, MARO is prompted.
	xx MSB	Make Set Busy key.
	xx NRD	Not Ready key.
	xx TRN	Call Transfer key.
	xx A03	A03 = Three-Party Conference key.
	xx A06	A06 = Six-Party Conference key.
	xx RLS	Release key. Requires CLS = LUXA. Key/lamp pair is not required.

LD 23 – Configure ACD parameters for all voice service queue.

Prompt	Response	Description
REQ	NEW	Add new data.
TYPE	ACD	Automatic Call Distribution data block.
CUST	xx	Customer number. xx = 0-31 for Option 11C.
ACDN	x...x	ACD Directory Number.
MWC	(NO) YES	Message Waiting Center.
...		
MAXP	xxxx	Maximum number of positions. Enter 1.
...		
NCFW	xx...x	Night Call Forward DN = DN of the primary voice messaging queue in network format. xx...x = up to 23 digits.

Malicious Call Trace enhancements

Malicious Call Trace enhancements

LD 10 – In order to activate Malicious Call Trace from an analog (500/2500 type) telephone, the user has to dial SPRE + two-digit access code (83) or the MCT FFC.

Prompt	Response	Description
REQ	NEW, CHG	New, or change.
TYPE	500	
TN	c u	
CLS	MCTA, MCTD	Malicious Call Trace is allowed if class is MCTA.

LD 11 – In order to activate Malicious Call Trace from a Meridian 1 propriety telephone, it should have CLS MCTA, and the TRC key should be defined. However, the same function can be achieved using a transfer or conference key and the SPRE + 83 or the MCT FFC.

Prompt	Response	Description
REQ	NEW, CHG	New, or change.
TYPE	xxxx	
TN	c u	Terminal number for Option 11
CLS	(MCTD), MCTA	
		Malicious Call Trace is allowed if class is MCTA.

LD 17 – In order to print the MCT record on a dedicated MCT TTY port, USER type MCT has to be defined.

Prompt	Response	Description
REQ	CHG	Change.
TYPE	CFN	
ADAN	xxx TTY yy	xxx = new or change. yy = port number 0-63 or 0-15.
USR	MCT	Dedicated TTY port for MCT record.

LD 16 – Setup the recorder route.

Prompt	Response	Description
REQ	NEW, CHG	New, or change.
TYPE	RDB	Route data block.
ROUT	xxx	Route number.
TKTP	RCD	Recorder trunk data block.
ACOD	xxxx	Recorder route access code.

LD 14 – Setup the recorder trunk.

Prompt	Response	Description
REQ	NEW, CHG	New, or change.
TYPE	RCD	Recorder trunk.
TN	c u	Card and unit.
CUST	0-31	Customer number.
RTMB	xxx xxx	Trunk route and member number for RCD.

LD 15 – Setup the recorder and alarm options.

Prompt	Response	Description
REQ	CHG	Change.
TYPE	CDB FTR_DATA	Customer data block. Features and options data (Release 21 gate opener).
CUST	0-31	Customer number.
...		
- ALDN	xxxxxxx	DN for the alarm.
- ALRM	(NO), YES	The ALRM prompt appears only if ALDN is defined. ALRM has to be set to YES if the alarm is to be rung for any call (external or internal) when MCT is activated.
- TIME	0-(15)	Time is prompted only if ALRM is set to YES. Time for the alarm is set in one-minute increments from 1 to 15.
- INT	(NO), YES	INT is prompted only if ALRM is set to YES. In addition, INT must be YES if the alarm is to be rung when MCT is activated against internal calls.
- RECD	(NO), YES	If the user wants the recorder, set RECD to YES. This prompt does not appear when a new customer is being defined.
- - MCRT	0-511	The user has to use the recorder route number defined in LD 16. It will only be prompted if the RECD is set to YES.

LD 16 – Set up the alarm for external calls.

Prompt	Response	Description
REQ	NEW, CHG	New, or change.
TYPE	RDB	Route data block.
TKTP	DID, COT	Direct Inward Dial or Central Office Trunks.
ALRM	(NO), YES	Malicious call trace is allowed for external calls when the response is YES.

LD 57 – In order to activate malicious call trace from an analog (500/2500 type) set without using the SPRE and 83, the MCT FFC has to be defined.

Prompt	Response	Description
REQ	CHG	Change.
TYPE	FFC	Flexible Feature Code.
CUST	0-31	Customer number.
CODE	MTRC	Malicious Call Trace.
MTRC	xxxx	Flexible Feature Code for Malicious Call Trace.

LD 16 – For analog and 1.5 Mbit digital trunks, the flash range to be sent to the central office is configured using the FLH timer. In order to send the string to the central office, MCCD has to be defined.

Prompt	Response	Description
REQ	NEW, CHG	New, or change.
TYPE	RDB	Route data block.
RCLS	(EXT), INT	Class marked route as (external) or internal.
...		
CNTL	YES	Changes control or timers.

TIMR	FLH <space> 60-(510)-1536	Flash timer in msec. The range of the Centrex switch hook flash timer is 60-(510)-1536. The FLH value is rounded down to the nearest 10 msec. tick. If the value entered is 128 or 129, then it is set to 130 msec. <i>Software controlled flash</i> 60-127 msec. Digit 1 will be sent. 128-1536 msec. software controlled switch hook flash. <i>Firmware controlled flash</i> The user can enter any value from 60 to 1536 msec. 90 msec. is the hardcoded firmware flash for an XFCOT pack; the craftsperson should enter 90 msec. Note: the FWTM prompt must be set to YES for the trunk associated with this route in LD 14, if firmware timing is to be used.
...		
MCTS	(NO), YES	Enter YES to get the new prompts
MCCD	0-8 digits	The call trace request string can be 0-8 digits in length. Valid digits are 0-9, *, and #.
MCDT	(0)-4	Digit string delay is in seconds, in increments of one second.

LD 14 – The WTM prompt is provided for EXUT and XCOT cards. This prompt should be set to YES if firmware timing is to be done for the flash and the card supports this functionality. If the prompt is set to YES for one unit, it is also set to YES for all other units.

Prompt	Response	Description
REQ	NEW, CHG	New, or change.
TYPE	DID, COT	Trunk type.
TN	c u	card, unit.
XTRK	EXUT, XCOT	Card type
FWTM	(NO), YES	Firmware timing for the trunk hook flash is available. This prompt is set to YES if firmware timing for trunk hook flash is supported by the pack.
CUST	0-31	Customer number.
RTMB	xxx yyy	xxx – Trunk route. yyy – Member number for RCD.

LD 73 – Define the DTI2 flash time range.

Prompt	Response	Description
REQ	NEW, CHG, PRT	New, change, or print.
TYPE	DTI2	
FEAT	abcd	Digital signaling category.
SICA	2-16	SICA table number.
...		
FALT (R)	abcd N	If FALT (receive) signal is not required.
P RRC(S)	abcd	Register recall signal activated by MCT.
TIME	10-(100)-630	Time of RRC(S) signal in milliseconds. This is the flash duration used for 2.0 Mbit DTI trunks. It is programmable in one-millisecond increments from 10 to 630.

LD 16 – Setup MCTM timer and tandem delay (2 Mbit PRI for AXE-10 Australia only).

Prompt	Response	Description
REQ	NEW, CHG	New or Change.
TYPE	CDB	Customer data block.
CUST	0-31	Customer number.
MCTS	YES, NO	
MCTM	(0) - 30	Malicious call trace timer (in seconds).
MTND	(NO), YES	Malicious call trace disconnect delay for tandem calls for AXE10 Australia.

Meridian hospitality voice services

LD 15 – Allow call redirection to Meridian Mail for voice messaging:

Prompt	Response	Description
REQ	CHG	Change.
TYPE	CDB RDR_DATA	Customer data block. Call Redirection data (Release 21 gate opener).
CUST	0-31	Customer number.
...		
- DNDH	YES	Do Not Disturb Hunting. MHVS package (179) must be equipped for this prompt to appear. LD 21 will reflect the DNDH option if MHVS is equipped.

LD 23 – Define the AOP DN:

Prompt	Response	Description
MWC	YES	ACD DN message center DN.
CMS	YES	Command and status link.

Meridian Mail Trunk Access Restriction

LD 15 – Enable Meridian Mail Trunk Access Restriction.

Prompt	Response	Description
REQ:	CHG	Change.
TYPE:	FTR	Customer Features and Options.
CUST	xx	Customer number.
- OPT	MCI	Message Centre Included.
...		
- MTAR	YES	Meridian Mail Trunk Access restricted. NO = Meridian Mail Trunk Access allowed.

Message Waiting Indication (MWI) Interworking with DMS

Use the following steps to configure MWI.

Procedure 2

Use LD 17 to configure remote D-channel capability

Prompt	Response	Description
REQ	CHG	Change system data
TYPE	CFN	Configuration record
ADAN	CHG DCH 0-63	Change Primary D-Channel
RLS	19, 36	Release ID of the switch at the far end of the D-channel interface. 19 is Meridian 1, 36 is DMS.
RCAP	MWI XMWI	Add remote D-channel capabilities for MWI Remove remote D-channel capabilities for MWI

If the Message Center is on a Meridian 1, it must be a Meridian Mail. No other messaging devices are supported for the Meridian 1. The DMS may use Meridian Mail or other messaging devices (Octel) for its Message Center.

If the Message Center is on DMS for MWI Interworking, a Meridian 1 that is connected to DMS must be on Release 19; other Meridian 1 switches that are connected to the Release 19 switch can be Release 16, 17, or 18.

Procedure 3

Use LD15 to enable Message Service

Prompt	Response	Description
REQ	CHG	
TYPE	CDB	Customer Data Block
OPT	MCI	Message Center is included. This prompt must be set to include the message center feature for both the user's telephone and the message center if the message center is on M-1.

Procedure 4

Use LD 23 to configure an ACD Group for the Message Center users in the remote switch

Prompt	Response	Description
REQ	NEW, CHG	Add or change system data
TYPE	ACD	Automatic call distribution
CUST	0–99	Customer number
ACDN	xxxx	ACD Directory Number
MWC	YES	ACD DN message center DN
NCFW	xx ... xx	Message Center DN: — a public DN (10 digits) prefixed by an ESN access code. — an ESN number prefixed by an ESN access code Note: Do not define an agent for this ACD group to allow for the automatic call redirection to the ACD Night Call Forward DN where the Message Center DN is defined.

Procedure 5

Use LD 11 to define the ACD Group as the Message Center DN for each telephone

Prompt	Response	Description
REQ	CHG	
FDN	x..x	Flexible call forward No Answer to Message Center DN
EFD	x..x	Call Forward by Call Type—External No Answer to Message Center DN
EHT	x..x	Call Forward by Call Type—External Hunt to Message Center DN
HUNT	x	Hunt to Message Center DN
KEY	xx MWK x..x	Message Waiting Key, where x..x is the Message Center DN.

Procedure 6**Use LD 15 to configure AC2 to insert ESN access code and PNI**

Prompt	Response	Description
REQ	NEW, CHG, END	
TYPE	CDB NET_DATA	Customer Data Block Networking Data (Release 21 gate opener)
...		
- AC2		Access code 2. Enter call types (type of number) that use access code 2. Multiple call types can be entered. Default is to access code 1.
	NPA	E.164 National
	NXX	E.164 Subscriber
	INTL	International
	SPN	Special number
	LOC	Location code
- ISDN	YES	Change ISDN options
- PNI	1-32700	Customer private identifier—unique to a customer. Within one network, use the same value for PNI in both the Customer Data Block (LD 15) and the Route Data Block (LD 16) in all PBXs.
...		
- HNPA	200- 999 1200- 1999	Home Number Plan Area code defined in LD 90
- HNXX	100-9999	Prefix for the central office
- HLOC	100-999	Home location code (NARS)
- LSC	1-9999	1- to 4-digit Local Steering Code established in the Coordinated Dialing Plan (CDP). The LSC prompt only appears if user has a 5- or 6-digit dialing plan.

For ISDN Networking Features such as NRAG, NMS and MWI, the originating switch must not use the digit manipulation (DMI) to insert the ESN access code. The ESN access code must be inserted at the terminating/tandem node via the INST prompt if it connects to the DMS in the Public Network.

Procedure 7**Use LD 16 to configure PNI and INAC or INST**

Prompt		Response	Description
REQ		NEW,CHG,OUT	
TYPE		RDB	Route Data Block
...	.		
- PNI		1–32700	Customer private identifier—unique to a customer. Within one network, use the same value for PNI in both the Customer Data Block (LD 15) and the Route Data Block (LD 16) in all PBXs.
- INAC		(NO), YES	Insert access code. Permit an ESN access code to be automatically added to an incoming ESN call from a private network. If INAC is YES, the digit insertion option (INST) is bypassed. This prompt only appears if the route type is a TIE trunk.
- INST		(0)–99999999	Digits to be inserted in front of leading digits. Use this prompt if route is used for connection to DMS.

Use “INST” in LD 16 at Meridian 1 location A for incoming tandem/terminating calls. Use “INAC” at Meridian 1 locations B and C. For the message center users on M-1, configure Home Location Code and Home NPA to terminate a Facility message from the message center either on M-1 or on DMS to the message center user’s switch. The Facility message is for turning on/off the message waiting indicator for the user.

For the private ESN Uniform Dialing Plan, use LD 90 to configure Home Location Code (HLOC).

Procedure 8
Use LD 90 to configure Home Location Code

Prompt	Response	Description
REQ	NEW, CHG	Add or change
CUST	0-31	Customer number
FEAT	NET	Network translation tables
TRAN	AC1, AC2	Access code 1 or 2
TYPE	HLOC	Home location code
HLOC	xxx	Home location code (3–7 digits)

For the public numbering plan, a 10 digit number (NPA+NXX+XXXX) is used in the ORIG and DEST IE of the Facility message. Use LD 90 to configure Home NPA and Central Office Translation (NXX) to terminate the Facility message onto the message center user's DN.

Procedure 9
Use LD 90 to configure Home NPA

Prompt	Response	Description
REQ	NEW, CHG	Add or change
CUST	0-31	Customer number
FEAT	NET	Network translation tables
TRAN	AC1, AC2	Access code 1 or 2
TYPE	HPNA	Home NPA Translation
HNPA	xxx	Home NPA

Procedure 10
Use LD 90 to configure Central Office Translation

Prompt	Response	Description
REQ	NEW, CHG	Add or change
CUST	0-31	Customer number
FEAT	NET	Network translation tables
TRAN	AC1, AC2	Access code 1 or 2
TYPE	NXX	Central Office Translation
NXX	xxx	Office code translation
RLI	0-255	Route list index
SDRR	LDID	Recognized local DID codes
DMI	1-255	Digit manipulation table index. DMI should delete 3 digits.
LDID	xxx	Local DID number recognized within the NPA and NXX.

Procedure 11
Use LD 22 to print the configuration record

Prompt	Response	Description
REQ	PRT	Print system data
TYPE	CFN	Configuration record
RLS	xx	Release ID of the switch at the far end of the D-channel
RCAP	MWI	Remote D-channel capabilities

Note: The mnemonic "MWI" is printed for overlay 22.

Network Attendant Service

LD 12 – Configure Attendant Consoles with NAS keys.

Prompt	Response	Description
REQ	NEW, CHG	New, or change.
TYPE	ATT 1250 2250	Type of console
TN	c u	Terminal Number.
KEY	xx NAS	xx is the key number assigned the NAS function. Each Attendant Console can have only one NAS key defined. This key is optional.

LD 15 – Enable or disable network attendant control, Recall to Same Attendant, NAS routing, and define a TRK ICI key.

Prompt	Response	Description
REQ	CHG	Change existing data block.
TYPE	CDB NET	Customer Data Block. Networking Data. Release 21 gate opener.
CUST	xx	Customer number.
- OPT	aaa	Options.
- AC2	aaa	Access Code 2 as defined in LD 86.
- ISDN	YES	Integrated Services Digital Network
...		
- NAS	(YES) NO	Specifies whether to (allow) or deny attendant control for call extension.
- NAS_ ACTV	(NO) YES	Specifies whether to allow or (deny) NAS routing.
- NAS_ ATCL	(YES) NO	Specifies whether to allow (YES) or deny attendant control for call extension.

- NAS_ ACTV	(NO) YES	Specifies whether to allow or (deny) NAS routing.
...		
- ICI	xx NCO NDID NTIE NFEX NWAT	Respond with the key number, from 0 to 19, followed by a space, followed by the trunk type. Network CO trunk Network DID trunk Network TIE trunk Network FEX trunk Network WAT trunk
- RTSA	(RSAD) RSAA RSAX	Recall to Same Attendant Denied Recall to Same Attendant Allowed Recall to Same Attendant allowed, with queuing on busy attendant.

LD 17 – Configure the D-channel interface (DCHI) for NAS.

Prompt	Response	Description
REQ	NEW, CHG	New, or change.
TYPE	ADAN	Action Device and Number.
DCHI	xxx	Respond with DCHI number for which NAS signaling is to be allowed or restricted.
NASA	(NO) YES	Specifies whether DCHI is to be allowed or (denied) NAS signaling.

LD 86 – Configure NAS routing.

Prompt	Response	Description
REQ	NEW, CHG	New, or change.
CUST	xx	Customer number.
...		
FEAT	NAS	Specifies type of data to be changed.
TBL	0–63	Routing table number, from 0 to 63. Without Multi-Tenant Service, 0 is the customer routing table. With Multi-Tenant Service enabled, NAS tables 1-63 may be associated with CPGs 1-63.
ALT	1–7	An alternative attendant or routing table, from 1 to 7. (To clear an old number, type an X before typing the new number. The old number cannot be cleared if it is associated with a schedule period. Reach TODS by pressing the return key.)
ID	xxxxxxxxxxxxxxxx	The dialed digits (including the network access code) needed to reach an attendant associated with the alternative number. Respond with a string of up to 16 digits to change the attendant ID; press the return key to leave the ID unchanged, exit the prompt, and return to ALT.
TODS	0–31	Specifies a schedule period, from 0 to 31, where 0 is the default to handle all time periods not defined in 1 through 31. Type an X before a period number to remove the schedule period. (Typing an X before a 0 clears all associated alternatives, leaving the default treatment as local attendant treatment.) Press the return key to continue the NAS feature setup process.
- PER	HR:MIN HR:MIN	Specifies the start and stop times for the period using 24-hour format. Start time must be before the stop time, and minutes can be only 00 or 30. Press the return key to leave times unchanged and move to the DAYS prompt.

- DAYS	D, D, ... D	Specifies applicable days of the week for the time period. Respond by inputting a number representing each day for which the schedule is active (where 1=Monday, 2=Tuesday,...7=Sunday). Type an X before the day number to deactivate the schedule period for that day. No more than seven entries are permitted on an input line. Press the return key to leave this schedule unchanged and move to the next prompt. (If not otherwise specified, a schedule period is assumed valid for all days.)
ALST	xxxx	Alternative list to be used for the schedule period. Respond with up to four alternative numbers in the order in which they are to be attempted. (These numbers are defined using ALT.)
DBK	Z Z Z Z	Alternatives for which “drop back busy” is to be active during this period. Respond with four entries of Y (allow) or N (deny) for the alternative evoked by the previous prompt. Responses are applied in sequence.
QUE	Z Z Z Z	Specifies alternatives for which queuing to a route is to be allowed during this period. Respond with four entries of Y (allow) or N (deny) for the alternatives evoked by ALST. Responses are applied in sequence. If the response is Y, off-hook queuing must already be configured for calls to be queued on this route.

LD 93 – Assign a NAS routing table to Console Presentation Groups.

Prompt	Response	Description
REQ	NEW, CHG	New, or change.
TYPE	CPG	Console Presentation Group.
CUST	xx	Customer number.
CPG	1-63	Console Presentation Group number.
CPGS	(NO) YES	Customer Presentation Group Services.
ROUT	0-511	Route number.
TEN	1-511	Tenant number.
NTBL	(0) - 63	NAS routing table to be used for calls directed to this Attendant Console Group(ACG)/ Console Presentation Group (CPG).

LD 16 – Allow or deny pre-answer tromboning (allow or deny an incoming call to be directly routed back on the same route).

Prompt	Response	Description
REQ	NEW, CHG	New, or change.
TYPE	RDB	Route Data Block.
CUST	xx	Customer number.
ROUT	0-511	Route Number.
TKTP	aaa	Trunk Type requires response when REQ=NEW.
TRMB	(YES) NO	Tromboning. NO = Tromboning denied. Incoming trunk call on route may not be routed back on same route. YES = Tromboning allowed. Incoming trunk call on route may be routed back on the same route. Only applies to calls routed using NARS/BARS or CDP. Does not apply to calls redirected by Hunt, Forward All Calls, or Forward No Answer.

Note: The anti-tromboning capabilities programmed in LD 16 only apply to attendant extended calls.

NAS performs tromboning when required regardless of the LD 16 programming. However, if redundant (or tromboned) trunk connections are to drop after the telephone is answered, this prompt must be set to NO across the network where this might occur.

Network and Executive Distinctive Ringing

LD 10 – Respond to the CLS prompt with EXR0, EXR1, EXR2, EXR3, or EXR4 to define the distinctive ringing cadence/tone to be used for 500/2500-type sets (a value of 0 inhibits distinctive ringing).

LD 11 – Respond to the CLS prompt with EXR0, EXR1, EXR2, EXR3, or EXR4 to define the distinctive ringing cadence/tone to be used for SL-1 sets (a value of 0 inhibits distinctive ringing).

LD 16 – Respond to the DRNG prompt with NO or YES to deny or allow distinctive ringing. Respond to the NDRI prompt with a digit between 0-4 to define the network ring index (a value of 0 results in the index not being defined).

LD 56 – Define the distinctive ringing cadence for 500/2500-type sets and the distinctive ringing tone for SL-1 sets in the Flexible Tone and Cadence (FTC) table:

- Respond to the NDR1 PBX prompt to define the first distinctive ringing cadence for 500/2500-type sets.
- Respond to the NDR1 BCS prompt to define the first distinctive ringing cadence for SL-1 sets.
- Respond to the NDR2 PBX prompt to define the second distinctive ringing cadence for 500/2500-type sets.
- Respond to the NDR2 BCS prompt to define the second distinctive ringing cadence for SL-1 sets.
- Respond to the NDR3 PBX prompt to define the third distinctive ringing cadence for 500/2500-type sets.
- Respond to the NDR3 BCS prompt to define the third distinctive ringing cadence for SL-1 sets.
- Respond to the NDR4 PBX prompt to define the fourth distinctive ringing cadence for 500/2500-type sets.
- Respond to the NDR4 BCS prompt to define the fourth distinctive ringing cadence for SL-1 sets.

Network call redirection

Use this procedure to configure Network Call Redirection (NCRD) on an Option 11 system.

Restrictions and limitations

The Network Call Redirection (NCRD) feature has three components:

- Network Call Forward All Calls (**NCFAC**)
- Network Call Forward No Answer (**NCFNA**).
- Network Hunt (**NHNT**)

The various types of call redirection allowed use all of the basic programming already in place in the Meridian 1 database. For example the Call Forward No Answer feature uses the FNAD, FNAT and FNAL programming to determine which destination DN to use (FDN or HUNT DN) based on the type of incoming calls (DID, External or Internal respectively). The called set must have the proper class of service (FNA, HTA, FBA) as well as Hunt DN and FDN to the appropriate destination and a Call Forward key (or dial access from regular set) for call forward all calls (CFW).

ISDN allows call redirection to take place across the private network instead of being limited to the set's home PBX. Presently supported between Meridian 1/SL-1 and to Centrex via TIE lines.

By allowing call redirection across the network, it becomes important to limit the number of steps that a call can take while hopping from switch to switch. This is achieved by a prompt in the customer data block (**RCNT**) which sets the maximum of inter-node hops allowed for NCRD.

Optionally, the reason for call redirection can be displayed using the call party name display feature (CPND in overlay 95) in the same fashion as before.

Basic PRI or ISL administration must be performed before Network Call Redirection is implemented.

Programming procedure

This procedure contains three major steps:

- 1** Set the number of inter-node hops allowed. Use LD15.
- 2** Allow Network Call Redirection messages on a route. Use LD16.
- 3** Display the reason calls are redirected. Use LD95.

Step 1

Set the number of inter-node hops allowed. Use LD 15.

Prompt	Response	Description
REQ	CHG	
TYPE	CDB NET_DATA	Customer data block Release 21 gate opener
CUST	0-31	Customer number
...		
- RCNT	0-(5)	Maximum number of inter-node hops allowed in a network redirection call. This field must be set to greater than 0 for a network redirection to take place
...		

Step 2Allow Network Call Redirection messages on a route. Use **LD16**.

Prompt	Response	Description
REQ	CHG	
TYPE	RDB	Route data block
CUST	0-31	Customer number
ISDN	yes	Modify ISDN parameters
NCNA	yes	Network Call Name is (is not) allowed. Only required if the name and/or the reason for call redirection is to be displayed. Make sure the Call Party Name Display (CPND) block in overlay 95 is programmed. When connected to a DMS set this prompt to NO.
NCRD	yes	Network Call Redirection. Allows network call redirection reason to be sent (or blocks messages if NCRD =no). Note: Network Call Redirection can occur without answering "yes" to the NCRD prompt. This prompt only controls the sending of Network Call Redirection reason messages for display purposes, not the actual redirection of the call. The message supplied when NCRD = yes provides the information for the CLID display. When NCRD is NO, the call is redirected without the CLID redirection information.
TRO	YES,(NO)	Trunk Optimization

Step 3

Display the reason calls are redirected. Use **LD95**.

Prompt	Response	Description
TYPE	CPND	Call Party Name Display data block
CUST	0-31	Customer number
RESN	yes	Allow display of reason for redirecting calls
CFWD	xxxx, (F)	Display mnemonic for (Network) Call Forward All Calls. Default is "F."
CFNA	xxxx, (N)	Display mnemonic for (Network) Call Forward No Answer display. Default is "N."
HUNT	xxxx, (B)	Display mnemonic for Hunt/Call Forward Busy. Default is "B".
PKUP	xxxx, (P)	Display mnemonic for Call Pickup. Default is "P".
XFER	xxxx, (T)	Display mnemonic for Call Transfer. Default is "T"
AAA	xxxx, (A)	Display mnemonic for Attendant Alternative Answering. Default "A".

Network drop back busy and Off-hook queuing

Note: Drop Back Busy (DBB) and Remote Virtual Queuing (RVQ) are both packaged under the Originating Routing Control/Remote Virtual Queuing (ORC-RVQ) package 192. If DBB are both configured, DBB will take precedence over RVQ. If the user wishes to activate RVQ, DBB must be disabled for a route entry in response to the IDBB prompt. If the user wishes to activate DBB, it must be enabled by entering DBI or DBA.

In Overlay 87, allow OHQ Network Class of Service for the customer.

Prompt	Response	Description
REQ	CHG	Change
CUST	nn	Customer number
FEAT	NCTL	Network Control
SOHQ	YES	Allow system (customer) Off-hook queuing.
- OHTL	nn	Off-hook queue time limit.
SCBQ	YES,(NO	Allow, (disallow) system (customer) Call Back queueing.
NCOS	nn	Network Class of Service number. The originating telephone must have the same value.
- OHQ	YES, (NO)	Off Hook Queueing allowed (not allowed) for this NCOS Both RVQ and OHQ can be enabled on a system. Only one can be activated at a time.
- CBQ	YES, (NO)	Call Back Queueing allowed (not allowed) for this NCOS
- RETT	2-(10)-30	Remote Virtual Queuing Retry Timer (Time between searches, in seconds).
- RETC	4-(5)-16	Remote Virtual Queuing Retry Counter (Number of times RVQ searches the initial set before moving on to the extended set).

In Overlay 86 (Route List Data Block), configure the originating node for Remote Virtual Queuing.

Prompt	Response	Description
REQ	CHG	Change
CUST	0-31	Customer number
FEAT	RLB	Route List data block
RLI	nn	Route List Index
ENTRY	nn	Route List entry number
ROUT	nnn	Route number
...	...	...
IDBB	(DBD) DBI DBA	Enter DBB (Drop Back Busy Disabled). This will disable Drop Back Busy, and enable Remote Virtual Queuing for the customer. DBI = Drop Back if Initial set is busy. DBA = Drop Back if all routes are busy.

Network Individual Do Not Disturb

Use the following procedure to configure the Network Individual Do Not Disturb feature.

Step	Overlay	Action
1	LD 12 Attendant Console	Allow the attendant to override DNDI, and configure the attendant console for DNDI
2	LD 15	Configure intercept treatment for DNDI
3	LD 16	Define the route number for DNDI RAN intercept treatment (the same route number defined in LD 15)
4	LD 14	Define the trunks associated with the RAN route

Step 1: LD 12 – Allow the attendant to override DNDI, and configure the attendant console for DNDI.

Prompt	Response	Description
KEY	xx BKL	Attendant keys The key number on the attendant console assigned to Break-in to DNDI
	xx DDL	The key number on the attendant console assigned for DNDI indication

Step 2: LD 15 – Configure intercept treatment for DNDI.

Prompt	Response	Description
DNDT	(BST) ATT RAN	Busy tone (default) Attendant Recorded announcement
RRT	0-511	Route number for the recorded announcement, prompted only if the DNDI intercept treatment is RAN

Step 3: LD 16 – Define the route number for DNDI RAN intercept treatment (the same route number defined in LD 15).

Prompt	Response	Description
RRT	0-511	Route number for the recorded announcement, prompted only if the DNDI intercept treatment is RAN
TKTP	RAN	The trunks associated with the RAN treatment

Step 4: LD 14 – Define the trunks associated with the RAN route.

Prompt	Response	Description
TKTP	RAN	

Network Intercom (Hot Type D and Hot Type I Enhancements)

Use the following procedure to configure Network Intercom.

Note: When configuring two-way Hot Type D keys in voice mode, it must be taken into account the fact that the CLID does not transmit the originator's Hot Type D DN between ISDN locations; it contains the prime DN of the originating set, and not the originator's Hot Type D DN. Therefore, ringing may occur on the Hot Type D key rather than immediate answer, since a match could not be found for the originator's Hot Type D DN. For this reason also, when the configured mode is either voice, ringing, or non-ringing, a "No Answer Indication" is not left on the set called set of a two-way Hotline. It is therefore recommended that the Hot Type I be used for a two-way Hotline, since it relies only on the prime DN.

LD 11 – Define Hot Type D and I keys and Classes of Service as follows:

Prompt	Response	Description
REQ	NEW, CHG, OUT	Configure, change or remove.
TYPE	aaaa	Telephone type where aaaa cannot = M3000.
TN	c u	Terminal Number for the Option 11.
...		
CLS	FICA (FICD) NAIA (NAID)	Forward Hot Type I allowed. Forward Hot Type I denied (default). No Answer indication allowed. No Answer indication denied (default).
...		

KEY	nn HOT D dd target_num hot_dn R/V/N/(H)	Two-way Hot Line type D Key. nn = key number. dd = number of digits dialed. target_number = terminating DN (31 digits maximum). Hot_dn = two-way Hot Line DN. Termination mode: R = Ringing V = Voice N = Non-ringing, and H = Hot Line (the default).
KEY	nn HOT I dd target_number (V)/N/R	Hot Line type I key. nn = key number. dd = number of digits dialed. target_number = terminating DN (31 digits maximum). Termination mode: V = Voice (the default) N = Non-ringing, and R = Ringing.

LD 95 – Configure the Calling Party Name Display:

Prompt	Response	Description
REQ	NEW, CHG	Configure or change.
TYPE	CPND	Calling Party Name Display.
CUST	0-31	Customer Number.
...		
NITC	aaaa (NI)	Non-Hot Line call. Indicates that the Hot Line call terminated as a normal call.

Network Messaging Service — Message Centre

All these procedures apply to both NMS-MM and NMS-MC except for Procedure 16 which is Meridian Mail only.

Procedure 12

Use LD17 to define the software release of each switch at the far end of each D-channel in the NMS network (X11 Release 17 and earlier)

Prompt	Response	Description
REQ	NEW, CHG	
TYPE	CFN	Configuration database
ISDN	YES	
DCHI	1-15	D-channel to the far end
RLS	xx	Enter the X11 release of the far end. The minimum release for NMS-MC is 15, and for NMS-MM is 16.

Procedure 13

Use LD17 to define the software release of each switch at the far end of each D-channel in the NMS network (X11 Release 18 and later)

Prompt	Response	Description
REQ	CHG	Change
TYPE	CFN	Configuration Record
ADAN	CHG DCH 0-63	Change D-channel information
RLS	xx	Enter the X11 release of the far end. The minimum release for NMS-MC is 15, and for NMS-MM is 16.

Procedure 14
Use LD15 to enable Message Services

Prompt	Response	Description
REQ	CHG	
TYPE	CDB	Customer Data Block
OPT	MCI	Message Center is included. This prompt must be set to include the message center feature for both the user's telephone and the message center.

Procedure 15
Create an ISDN transport signaling database in LD16

Prompt	Response	Description
REQ	NEW, CHG	
TYPE	RDB	Route Data Block
CUST	xx	Customer number or range of customer numbers
ROUT		Route number
ISDN	YES	ISDN option
INAC	(NO), YES	Insert Access Code. INAC = YES is required for ISDN network features. This prompt only appears if the route type is a TIE trunk.
PNI	1-32700	Customer private identifier—unique to a customer. This is the private identifier of the target switch and must be the same number used for PNI in LD15 for remote customer. Matches PNI in LD16 at remote to PNI in LD15 at local site.
NCRD	YES	Network Call Redirection
TRO	YES	Trunk Route Optimization allowed (denied) on the route.

Procedure 16

Use LD23 to define Meridian Mail ACD group in the remote switch (NMS-MM only)

Prompt	Response	Description
REQ	aaa	Action request: NEW, CHG, PRT, END
TYPE	ACD	Automatic Call Distribution data block
CUST	xx	Customer number, 0-31
ACDN	xxxxxxx	The Meridian Mail DN
MWC	YES	Message Center services
MAXP	1	Maximum number of agent positions
NCFW	x..x	Night Call Forward DN, where x...x is the NMS-DN. If network DN, include AC1/AC2.

Note: Do not define any ACD agents for this ACD group to allow automatic redirection to the ACD Night Call Forward DN (NFCW).

Procedure 17

Use LD11 to define Network Message Services DN for each telephone

Prompt	Response	Description
REQ	CHG	
FDN	x..x	Flexible Call Forward No Answer NMS-DN. See Note below.
EFD	x..x	Call Forward by Call Type - External No Answer to NMS-DN.
EHT	x x	Call Forward by Call Type - External Hunt to NMS-DN
HUNT	x	Hunt to NMS-DN
KEY	xx MWK x...x	Message Waiting Key, where x..x is the NMS-DN. See Note below.

Note: The NMS-DN may be the local or the network ACD DN.

Procedure 18**Use LD10 to define Network Message Services DN for each telephone**

Prompt	Response	Description
REQ	CHG	
FTR	FDN x..x	Flexible call forward no answer NMS-DN, x..x is the NMS-DN. The NMS-DN may be the local or the network DN. If network DN, include AC1/AC2.

Procedure 19**Network Message Services DN (applies to both NMS-MC and NMS-MM)**

Call type	Network Message Services DN (NMS-DN)	Number of digits
Private call using Uniform Dialing Plan (UDP)	ACC + LOC + XXXX	8 to 9
Private call using Coordinated Dialing Plan (CDP)	DSC + X..X	10 maximum
Public numbering plan	ACC + (1) + NPA + NXX + XXXX ACC + (1) + NXX + XXXX	14 maximum

Network Ring Again

Follow these steps to configure Network Ring Again:

- 1 LD17 Configuration Record (Procedure 20)
Set the software release ID of the far end.
X11 RELEASE 17 AND EARLIER
- OR*
- 1 LD17 Configuration Record (Procedure 21)
Set the software release ID of the far end.
X11 RELEASE 18 AND LATER
- 2 LD15 Customer Data Block (Procedure 22)
Set up Private Network Identifier (PNI) mapping between call type translator HLOC, LSC, HNPA or HNXX for proper CLID construction.
- 3 LD16 Route Data Block (Procedure 7)
Set up duration timer (NRAG), Private Network Identifier (PNI), insertion of ESN access codes (INAC).

Procedure 20

Use LD17 to define the software release ID (X11 Release 17 and earlier)

Prompt	Response	Description
REQ	CHG	Configuration data block
TYPE	CFN	
ISDN	YES	
DCHI	1-15	D-channel number
.		
.		
RLS	xx	This is the current software release of the far end. If the far end has an incompatible release of software, it prevents the sending of application messages. The minimum X11 release is 14, the minimum BCS release is 28.

Procedure 21**Use LD17 to define the software release ID (X11 Release 18 and later)**

Prompt	Response	Description
REQ	CHG	Change
TYPE	CFN	Configuration Record
ADAN	NEW DCH 0-63	Add a primary D-channel (can also CHG and OUT DCH)
CTYP	MSDL	Card type, where MSDL = Downloadable D-channel daughterboard.
- PORT	1	Port number on the Downloadable D-Channel Daughterboard
RLS	xx	Release ID of the switch at the far end of the D-Channel

Procedure 22**Respond to the following prompts in LD15**

Prompt	Response	Description
REQ	CHG	
TYPE	CDB NET_DATA	Customer data block Networking Data (Release 21 gate opener)
CUST	0-31	Customer Number.
- OPT	RNA	Allow Ring Again No Answer.
_ AC2	aaa	Access Code 2. Enter call types (type of number) that use access code 2. Multiple call types can be entered. Default is to access code 1.
- ISDN	YES	Change ISDN options
- PNI	1-32700	Customer private identifier—unique to a customer. Within one network, use the same value for PNI in both the Customer Data Block (LD15) and the Route Data Block (LD16) in all PBXs.
	100-319	Area code for the Meridian 1 system

Procedure 22
Respond to the following prompts in LD15

Prompt	Response	Description
	100-319	Prefix for the Central Office
	100-319	Home location code (NARS)
	1-9999	1- to 4-digit Local Steering Code established in the Coordinated Dialing Plan (CDP). The LSC prompt only appears if user has a 5- or 6-digit dialing plan.
...		

Procedure 23
Respond to the following prompts in LD16

Prompt	Response	Description
REQ	NEW,CHG,OUT	
TYPE	RDB	Route data block
CNTL	(NO),YES	Changes to controls or timers
TIMR	NRAG (30)-240	Network Ring Again duration timer (T6 and T7 timers)—time is in minutes. Currently, only a value of 30 minutes is supported.
	Note: Package 148, Advanced ISDN Features, is required.	
.		

Procedure 23**Respond to the following prompts in LD16**

Prompt	Response	Description
INAC	(NO), YES	Insert Access Code. Permit an ESN access code to be automatically added to an incoming ESN call from a private network. If INAC is YES, the digit insertion option (INST) is bypassed. This prompt only appears if the route type is a TIE trunk.
PNI	1-32700	Customer private identifier—unique to a customer. Within one network, use the same value for PNI in both the Customer Data Block (LD15) and the Route Data Block (LD16) in all PBXs.

Network-wide Listed Directory Number

LD 10, 11 – The LDN prompt has been changed to accept a greater value (0-5) in these overlays.

LD 15 – Four new prompts have been added for defining the extended LDN numbers and the Listed Attendants (LDAs) belonging to the LDNs. The prompts can be answered in the same way as the prompts LDN0, 1, 2, 3. The LDA prompts only appear if DLDN is set to YES. These store the Attendant Console number associated with the LDN number.

Prompt	Response	Description
REQ	NEW, CHG	New, or change.
TYPE	CDB LDN_DATA	Customer Data Block. Departmental Listed Directory Numbers data (Release 21 gate opener).
CUST	0-31	Customer number
- OPT	aaa	Options
- DLDN	YES	
...		
- LDN3	...	
- LDA4	xx xx..., ALL	xx can be in the range of 1-63 or all attendants. Precede an attendant number with X to remove.
- LDN4	x...x	Listed Directory Number. If the DNXP package is equipped, up to seven digits are allowed; otherwise, only four digits are allowed.
- LDA5	xx xx..., ALL	xx can be in the range of 1-63 or all attendants. Precede an attendant number with X to remove.
- LDN5	x...x	Listed Directory Number. If the DNXP package is equipped, up to seven digits are allowed; otherwise, only four digits are allowed.

LD 15 – Turn on or off the network recognition of the LDNs. The ICI keys can also be assigned to the new LDN values.

Prompt	Response	Description
REQ	NEW, CHG	New, or change.
TYPE	CDB LDN_DATA	Customer Data Block. Departmental Listed Directory Numbers data (Release 21 gate opener).
CUST	0-31	Customer number
- OPT	NLDN	Network Wide LDN allowed
...		
- ICI	xx LD4	New answer to existing prompt, where xx is the key number.
- ICI	xx LD5	New answer to existing prompt, where xx is the key number.

LD 93 – This overlay is used for printing LDN information.

Prompt	Response	Description
TYPE	CPGP	Changes affect the Console presentation group parameters.
...		
LDN3	...	
LDN4	xxxx(xxx)	Listed Directory Number. If the DNXP package is equipped, up to seven digits are allowed; otherwise only four can be entered.
LDN5	xxxx(xxx)	Listed Directory Number. If the DNXP package is equipped, up to seven digits are allowed; otherwise only four can be entered.
...		
ICI	0-19 LD4	New answer to existing prompt.
ICI	0-19 LD5	New answer to existing prompt.

Option 11 Downloadable D-Channel

Add an NTAK93 DCHI or NTBK51 DDCH card. Use Overlay 17.

Prompt	Response	Description
REQ		
TYPE	CFN	configuration data block
ADAN	NEW DCH 1-15 CHG DCH 1-15 OUT DCH 1-15	Add a primary D-Channel port number. Any unused SDI port number Change a primary D-Channel Out the primary D-Channel
CTYP	MSDL	MSDL for Downloadable D-Channel configuration (NTBK51 D-channel). The MSDL base and D-Channel application software are downloadable into the DDCH card.
CDNO	1-9	Card slot number to be used as the primary DDCH
PORT	1	PORT must be set to "1"
ADAN	NEW BDCH 1-15 CHG BDCH 1-15 OUT BDCH 1-15	Add a backup D-Channel port number. Any unused SDI port number Change a backup D-Channel Out a backup D-Channel
PDCH	X	Primary D-Channel X as defined above
CTYP	MSDL	Card type (automatically printed because it must be the same as the primary D-Channel)
CDNO	1-9	Card slot number to be used as the backup D-Channel
PORT	1	PORT must be set to "1"
...	...	...

Overlap Signaling

Use the following procedure to configure Overlap Signaling.

Step	Overlay	Action
1	LD 17 Configuration Record	Enable or disable overlap sending and overlap receiving, and set number of leading digits to delete when receiving, and the interval between the sending of INFORMATION messages.
2	LD 86 Route List Block	Define the minimum overlap digit length

Step 1: Enter the following commands in Overlay 17.

Prompt	Response	Description
REQ		
TYPE	CFN	Configuration Record
.		
.		
.		
ISDN	YES	
DCHI	xx	D-channel interface number
.		
.		
.		
CNEG		
RLS		
OVLRL	(NO), YES	Overlap Receiving (not) allowed

DIDD	(0)-15	Number of leading digits to delete when receiving digits from DID trunk
OVLS	(NO), YES	Overlap sending (not) allowed
OVLT	(0)-8	Overlap timer in seconds (this timer controls the interval between the sending of INFORMATION messages; recommended value is 1)

Step 2: Enter the following commands in Overlay 86.

It may be necessary to adjust the value of FLEN in overlay 90, so that FLEN is greater than OVLL, being the maximum number of digits which can be used for this ESN or CDP steering code.

Prompt	Response	Description
REQ		
TYPE	RLB	Route List Block
RLI	xx	Route List Block number
.		
.		
.		
ISSET	xx	Number of digits in the initial set
MFRL	xx	Minimum FRL required to access this Route List
OVLL	(0)-16	Minimum Overlap Digit Length (number of digits dialed by called, not including ESN access code, before the SETUP message is sent or outpulsing begins). If this value is 0, then FLEN alone will determine when overlap sending takes place. It is recommended that OVLL be at least the average number of digits in the ESN or CDP steering code which will be using the particular Route List Block.

QSIG

LD 17 – Assign the configuration record.

Prompt	Response	Description
ADAN	NEW, CHG, OUT, MOV, DCH xx	New, change out, or move.
...		
IFC	ISIG ESIG	Interface ID for ISO QSIG. Interface ID for ETSI QSIG.
TIMR	(NO), YES	NO = skip timer prompt. YES = change timer value.
T310	10-(30)-60	10-60 seconds (one-second increments). 30 seconds is the default value.

LD 16 – Use this overlay to define a route data block.

Prompt	Response	Description
REQ	NEW, CHG	New, or change.
TYPE	RDB	Route data block.
...		
DGTP	PRI, PRI2, BRI	Digital route type.
IFC	ISIG ESIG	New DCH interface ID.

LD 97 – This overlay defines the Extended Peripheral Equipment (XPE) configuration. However, data relating to the software downloading for the MSDL card will remain in the overlay. Hence, this overlay needs to be modified to accept the parameters required for downloading the PRI application data files.

The parameters basically specify conditional or forced downloading.

Prompt	Response	Description
REQ	CHG, PRT	Change, or print.
TYPE	SYSM	
...		
FDLC	P1, P2	Peripheral Software Download Option. P1 = This will be set to the application and its data files, such as BRIE, and PRIE. P2 = Specifies conditional (C) or forced downloading (F).

Radio Paging Improvement

LD 15 – In order for the this feature to operate network wide, the Recall to Same Attendant (RTSA) prompt must has to be activated on the originating node as follows:

Prompt	Response	Description
REQ	NEW, CHG	New, or change.
TYPE	CDB ATT	Customer data block. Attendant Consoles data (Release 21 fate opener).
CUST	0-31	Customer number
- OPT	aaa	Options
...		
RTSA	RSAA	Recall to same attendant allowed.

LD 87 – Set up remote Radio Paging on originating node.

Prompt	Response	Description
REQ	NEW, CHG	New, or change.
TYPE	TSC, DSC	Trunk/Distant Steering Code (enter RPAX FFC defined on paging node).
RRPA	(NO), YES	Remote Radio Paging option.
RLI		Route List Index of route list block used to route to paging node.

Radio Paging Improvement Continuation

LD 15 – In order for the this feature to operate network -wide, the Recall to Same Attendant (RTSA) prompt must be activated on the originating node as follows:

Prompt	Response	Description
REQ	NEW, CHG	New, or change.
TYPE	CDB ATT	Customer Data Block. Attendant Consoles data. Release 21 gate opener.
CUST	xx	Customer number.
- OPT	aaa	Options.
...		
RTSA	RSAA	Recall to Same Attendant Allowed.

LD 87 – Set up remote Radio Paging on originating node.

Prompt	Response	Description
REQ	NEW, CHG	New, or change.
TYPE	TSC DSC	Trunk Steering Code Distant Steering Code (enter RPAX FFC defined on paging node).
RRPA	(NO) YES	Remote Radio Paging option.
RLI		Route List Index of Route List Block used to route to paging node.

Recorded Announcement for Calls Diverted to External Trunks

Use the following procedure to configure Recorded Announcement for Calls Diverted to External Trunks.

Procedure 24

Configure Recorded Announcement for Calls Diverted to External Trunks

Step	Overlay	Action
1	LD 16	Configure Recorded Announcement for Calls Diverted to External Trunks in Route Data Block.

Step 1 - Configure Recorded Announcement for Calls Diverted to External Trunks

Enter the following commands in **Overlay 16**.

Prompt	Response	Description
REQ	NEW/CHG	New RDB or change data.
TYPE	RDB	Route Data Block.
DLDN	YES	YES if no CPG configured.
...		
FORM	aaa	Signaling format.
ICOG		Incoming and/or outgoing trunk.
RANX	(NO)	(RAN not requested when a call is forwarded to this route)
	YES	RAN is requested when a call is forwarded to this route
	<cr>	RANX value is not changed.
RANR	0-127	The route number for the RAN route (appears only if RANX = YES)
...		

Remote Virtual Queueing

In Overlay 86 (Route List Data Block), configure the originating node for Remote Virtual Queueing.

Prompt	Response	Description
REQ	CHG	Change
CUST	0-31	Customer number
FEAT	RLB	Route List data block
RLI	nn	Route List Index
ENTRY	nn	Route List entry number
- ROUT	nnn	Route number
...	...	...
- IDBB	(DBD) DBI DBA	Enter DBD (Drop Back Busy Disabled). This will disable Drop Back Busy, and enable Remote Virtual Queueing for the customer. DBI = Drop Back if Initial set is busy. DBA = Drop Back if all routes are busy.

In Overlay 87, configure the originating nodes for CBQ.

Prompt	Response	Description
REQ	CHG	Change
CUST	0-31	Customer number
FEAT	NCTL	Network Control
...		
SCBQ	YES	Allow system Call Back queueing for the customer.
NCOS	nn	Network Class of Service number. The originating telephone must have the same value.
...		

Prompt	Response	Description
CBQ	YES, (NO)	Call Back Queueing allowed (not allowed) for this NCOS
RETT	2-(10)-30	Remote Virtual Queueing Retry Timer (Time between searches, in seconds).
RETC	4-(5)-16	Remote Virtual Queueing Retry Counter (Number of times RVQ searches the initial set before moving on to the extended set).

In Overlay 86 (Route List Data Block), configure the outgoing routes of the originating nodes for CBQ (at least one entry of the Initial set of the outgoing route has to allow CBQ).

Prompt	Response	Description
REQ	CHG	Change
CUST	nn	Customer number
FEAT	RLB	Route List data block
RLI	nn	Route List Index
ENTRY	nn	Route List entry number
- ROUT	nnn	Route number
...		
- CBQ	YES)	Allow CBQ
...		
ENTRY	<cr>	
ISSET	nn	Size of the Initial set.

In Overlay 16 (Route Data Block), allow CBQ for the incoming non-ISDN routes on the originating nodes (for RVQ at a conventional Main operation).

Prompt	Response	Description
REQ	CHG	Change
TYPE	RDB	Route datablock
CUST	nn	Customer number
ROUT	nnn	Route number
...		
CBQ	YES	Allow Call Back Queueing
...		

Station Activity Records

LD 10 – Set Class of Service CDMA/CDMD for analog (500/2500 type) of telephone.

Prompt	Response	Description
REQ	NEW, CHG	New, or change.
TYPE	500	Telephone type.
...		
CLS	(CDMD), CDMA	CDMA allows Station Activity Records to be generated for the set (when the trunk is involved in the call). CDMD denies record generation.

LD 11 – Set Class of Service CDMA/CDMD for Meridian 1 propriety telephone.

Prompt	Response	Description
REQ	NEW, CHG	New, or change.
TYPE	xxxx	BCS set type.
...		
CLS	(CDMD), CDMA	CDMA allows Station Activity Records to be generated for the set (when the trunk is involved in the call). CDMD denies record generation.

LD 17 – Define a CDR link for Call Detail Recording.

Prompt	Response	Description
REQ	CHG	Change.
TYPE	CFN	Configuration.
ADAN	NEW TTY xx	Add a port.
USER	CTY	TTY has CTY as the user (for CDR records).

LD 15 – Enable CDR for the customer.

Prompt	Response	Description
REQ	NEW, CHG	New, or change.
TYPE	CDB CDR_DATA	Customer data block. Call Detail Recording data (Release 21 gate opener).
CUST	0-31	Customer number.
- IMPH	(NO) YES	CDR for Incoming Packet data call
- OMPH	(NO) YES	CDR for Outgoing Packet data call
- AXID	(NO) YES	Auxiliary Identification output in CDR record
- TRCR	(NO) YES	Carriage Return sent after each CDR message
- C DPR	(NO) YES	Coordinated Dialing Plan Record option
- ECDR	(NO) YES	End-to-End Signaling digits in CDR record.
- OTCR	(NO) YES	CDR provided, based on Originally dialed Trunk Route
- PORT	0-15	The CDR port number for the customer.

Taiwan R1 Modified Signaling

To configure the Taiwan R1 Modified Signaling feature on a Meridian 1:

- Configure the digital Taiwan R1 trunks, and configure a digital loop for the XCT (Extended Conference/Tone and Digit Switch/Multifrequency Sender) card (required for outgoing calls), using LD 17.
- Configure a Multifrequency Receiver (MFR), using LD 13 (required for incoming calls).
- Configure a customer CLID database (required for outgoing calls), using LD 15.
- Configure a Taiwan R1 route, using LD 16.
- Configure Taiwan R1 trunks, using LD 14.
- Configure a CLID entry for analog (500/2500 type) sets (required for outgoing calls), using LD 10.
- Configure a CLID entry for Meridian 1 proprietary sets (required for outgoing calls), using LD 11.
- Configure a Taiwan Numbering Plan:
 - Configure a Digit Manipulation Index, using LD 86.
 - Configure a Route List Index, using LD 86.
 - Configure ESN data, using LD 86.
 - Configure Special Number Translation data, using LD 90.
 - Define the Special Number (002) for outgoing Taiwan international calls, and configure the appropriate SPN and ITOH values, using LD 90.

LD 17 - Configure the digital Taiwan R1 trunks.

Configure a digital loop for the XCT (Extended Conference/Tone and Digit Switch/Multifrequency Sender) card (required for outgoing calls).

Prompt	Response	Description
REQ	CHG	Change existing data.
TYPE	CEQU	Common Equipment Information.
...		
- XCT	0, 2, 4,...158	Loop number (even-numbered) for the Extended Conference/TDS/MFS card.
...		
- DLOP	ll dd ff	Digital trunk loop options. ll = loop number dd = number of voice or data calls ff = frame format
- MODE	TRK	Mode of operation is digital trunk.
...		

LD 13 – Configure a Multifrequency Receiver (MFR) (required for incoming calls).

Prompt	Response	Description
REQ	NEW CHG	Add new data. Change existing data.
TYPE	MFR	Multifrequency Receiver.
TN	l s c u c u	Terminal Number for the MFR, for Options 51C-81/81C. l = loop. s = shelf. c = card. u = unit. Terminal Number for Option 11C. c = card. u = unit.
...		

LD 15 – Configure the customer CLID database (required for outgoing calls, if ICOG = OGT in LD 16).

Prompt	Response	Description
REQ:	NEW CHG	Add new data Change or delete existing data.
TYPE:	NET	Networking data (if REQ = CHG only.)
CUST	xx	Customer number. xx = 0-99 for Options 51C-81/81C xx = 0-31 for Option 11C.
...		
- CLID	(NO) YES	CLID option. YES = configure a CLID table for the customer. NO = do not configure a CLID table. In this case, the remaining prompts are not generated, and no CLID is sent for the customer.
- - SIZE	0-(256)-4000	The maximum number of CLID entries needed for a customer. If REQ = NEW, you may select the default value (256) by entering <CR> in response to this prompt. It is advised that you not define a size much larger than actually needed. This entry may be increased or decreased as required.
- - INTL	0-9999 X	Country code, for international number. Enter X to delete the digits.
- - ENTRY	aaaa Xaaaa Xaaaa Xbbbb <CR>	CLID entry to be configured. CLID entry to be deleted. Range of CLID entries to be deleted, aaaa and bbbb must be a value between 0 and (SIZE-1). The ENTRY prompt is repeated until <CR> is entered as a response.

		If REQ = NEW, only one new entry may be created. The entry will be saved to system memory when the configuration for the entire overlay is completed. If REQ = CHG, as many entries as needed may be created, changed or deleted. The action for the entry will be saved to system memory after the CLID entry has been completely configured, that is, after the LSC prompt has been answered. If a new CLID entry is created, or an existing CLID entry is changed, the message "ENTRY aaaa SAVED" is displayed after the LSC prompt. If a CLID entry or CLID entries is/are deleted, the message "ENTRY aaaa DELETED" or "ENTRIES aaaa-bbbb DELETED" is displayed after the LSC prompt.
- - - HNTN	0-999999 X	National code for home national number (1-6 digits). Enter X to delete the digits.
- - - HLCL	0-999..9 X	Local code for home local number or Listed Directory Number (1-12 digits). Enter X to delete the digits.
- - - DIDN	(YES) NO SRCH	How to use the DN as a DID when constructing a CLID national or local number. The CLID is constructed using the digits defined in HLCL followed by the DN of the active key. Construct the CLID using the digits defined in HLCL. Search on the set, from key 0 - upwards, to find a CLID entry which has the DIDN set to YES. Use the found CLID to construct the local number.
- - - HLOC	0-9999999 X	Home location code (ESN), 1-7 digits. Enter X to delete the digits.
- - - LSC	0-9999999 X	Local steering code, 1-7 digits. Enter X to delete the digits.
ENTRY aaaa SAVED ENTRY aaaa DELETED ENTRIES aaaa-bbbb DELETED		Displayed message. Refer to Note 3 for the ENTRY prompt.

LD 16 - Configure a Taiwan R1 route, in the Route Data Block.

Prompt	Response	Description
REQ	NEW CHG	Add new data. Change existing data.
TYPE	RDB	Route Data Block.
CUST	xx	Customer number, as already configured in LD 15. xx = 0-99 for Options 51C-81/81C xx = 0-31 for Option 11C.
ROUT	0 - 511 0-127	Route Number for Options 51C-81/81C. Route Number for Option 11C.
TKTP		Trunk type.
	DID	Taiwan R1 trunks must be Direct Inward Dialing (DID).
TW_ROUTE	YES	Configure a Taiwan R1 route. This prompt is generated only if TKTP = DID. It is autoprnted if REQ = CHG.
...		
DTRK	YES	Taiwan R1 routes are digital. The prompt and response are auto-printed, if TW_ROUTE = YES.
DGTP	DTI	Taiwan R1 route trunks are digital. The prompt and response are auto-printed, if TW_ROUTE = YES.
...		
ICOG	ICT OGT	Taiwan R1 trunk is incoming only. Taiwan R1 trunk is outgoing only. If TW_ROUTE = YES, the incoming and outgoing trunk (IAO) option is not allowed.
TW_INC_CLID	(NO) YES	YES = the Meridian 1 sends the CLID request wink signal after the called number has been received. NO = The call is terminated as soon as the called number is received. This prompt appears only if ICOG = ICT.

...		
INST	xx...x	Number of digits to be inserted before the leading digit. (Required for incoming calls). xx...x = 1-99999999.
CNTL	YES	Configure the control timers.
- TIMR	TTO (128)-7040	Configure the Taiwan Time Out value, in milliseconds. This is valid for outgoing trunks, that is, if ICOG = OGT.
- TIMR	SST 3-(5)-15	Configure the Seizure Supervision Timer, in seconds.

LD 14 - Configure Taiwan R1 trunks.

Prompt	Response	Description
REQ	NEW CHG	Add new data. Change existing data.
TYPE	DID	Trunk type. Taiwan R1 trunks must be Direct Inward Dialing (DID).
CUST	xx	Customer number, as already configured in LD 15. xx = 0-99 for Options 51C-81/81C xx = 0-31 for Option 11C.
RTMB	0-511 1-510 0-127 1-510	Route number and member number, for Options 51C-81/81C. For Option 11C.
...		
SIGL	LDR	Level 3 signaling is Loop Dial Repeating for Taiwan R1 trunks. The prompt and response are autoprinted.
...		
STRI	MWNL	Start arrangement for incoming Taiwan R1 trunks is Modified Wink. The prompt and response are autoprinted if ICOG = ICT (incoming trunks only) in LD 16.

STRO	MWNK	Start arrangement for outgoing Taiwan R1 trunks is Modified Wink. The prompt and response are autoprinted if ICOG = OGT (outgoing trunks only) in LD 16.
...		
CLS	MFR	Class of Service for Taiwan M1 trunks must be Multifrequency Receive.
...		

LD 10 – Configure the CLID entry for analog (500/2500 type) sets (required for outgoing calls, if ICOG = OGT in LD 16).

Prompt	Response	Description
REQ:	NEW CHG	Add new data. Change existing data.
TYPE:	500	500 set.
TN	l s c u c u	Terminal Number. l = loop, s = shelf, c = card, u = unit for Options 51C-81/81C c = card, u = unit for Option 11C.
DES	d...d	Office Data Administration System (ODAS) Station Designator of 1-6 alphanumeric characters.
CUST	xx	Customer number as defined in LD 15. xx = 0-99 for Options 51C - 81C. xx = 0-31 for Option 11C.
...		
DIG	xx yy	Dial Intercom group number and member number.
DN	xxxx (0)-N	DN and CLID entry. N = CLID SIZE-1 (SIZE defined in LD 15).
...		

LD 11 - Define the CLID entry for DN keys for Meridian 1 proprietary sets (required for outgoing calls, if ICOG = OGT in LD 16).

Note: When assigning a CLID entry to an ACD set, you cannot use the same position ID already on the set. The set must be first outted, or the ACD key must be nulled and then rebuilt with the table entry number.

Prompt	Response	Description
REQ:	NEW CHG	Add new data. Change existing data.
TYPE:	xxxx	Type of telephone set.
TN	l s c u c u	Terminal Number. l = loop, s = shelf, c = card, u = unit for Options 51C-81/81C c = card, u = unit for Option 11C.
DES	d...d	Office Data Administration System (ODAS) Station Designator of 1-6 alphanumeric characters.
CUST	xx	Customer number as defined in LD 15. xx = 0-99 for Options 51C - 81C. xx = 0-31 for Option 11C.
...		
KEY	xx MCN yyyy (0)-N/D xx MCR yyyy (0)-N/D xx PVN yyyy (0)-N/D xx PVR yyyy (0)-N/D xx SCN yyyy (0)-N/D xx SCR yyyy (0)-N/D	Telephone function key assignments. xx = key number. MCN = Multiple Call Non-ringing key. MCR = Multiple Call Ringing key. PVN = Private Line Non-Ringing key PVR = Private Line Ringing key SCN = Single Call Non-ringing key. SCR = Single Call Ringing key. yyyy = DN. (0)-N = CLID entry, with N = CLID SIZE-1 (SIZE defined in LD 15). D = CLID entry. Search for a CLID entry from key 0 upwards, to find a DN key. The found CLID is used as the CLID entry for the active DN key.

KEY	xx HOT D dd yyy...y zzzz m (0)-N/D	Two-way Hotline Direct key, where: xx = key number. dd = number of digits dialed. yyy...y = target number (terminating DN, maximum of 31 digits). zzzz = two-way hotline DN. m = one of the following Terminating Modes: H = Hotline (default) N = Non-ringing R = Ringing V = Voice (0)-N = CLID entry, with N = CLID SIZE-1 (SIZE defined in LD 15). D = CLID entry. Search for a CLID entry from key 0 upwards, to find a DN key. The found CLID is used as the CLID entry for the active DN key.
KEY	xx HOT L bbb zzzz (0)-N/D	Two-way Hotline List key, where: xx = key number. bbb = Hot Line List entry (0-999). zzzz = two-way hotline DN. (0)-N = CLID entry, with N = CLID SIZE-1 (SIZE defined in LD 15). D = CLID entry. Search for a CLID entry from key 0 upwards, to find a DN key. The found CLID is used as the CLID entry for the active DN key.
KEY	xx ACD aaaa 0-N/D bbbb	ACD key, where: xx = key number. aaaa = ACD DN or Message Center DN. 0-N = CLID entry, with N = CLID SIZE-1 (SIZE defined in LD 15). D = CLID entry. Search for a CLID entry from key 0 upwards, to find a DN key. The found CLID is used as the CLID entry for the active DN key. bbbb = ACD agent's position ID. Please refer to the note at the top of the overlay table, on page 1212, that pertains to assigning a CLID entry to an ACD set.

Configure a Taiwan Numbering Plan.

LD 86 – Configure a Digit Manipulation Index (required for incoming calls).

Prompt	Response	Description
REQ	NEW	Add data.
CUST	xx	Customer number, as already configured in LD 15. xx = 0-99 for Options 51C-81/81C xx = 0-31 for Option 11C.
FEAT	DGT	Digit Manipulation Index.
DMI	(0)-999	Digit Manipulation Index number.
DEL	(0)-19	Number of leading digits to be deleted.
...		

LD 86 – Configure a Route List Index.

Prompt	Response	Description
REQ	NEW	Add data.
CUST	xx	Customer number, as already configured in LD 15. xx = 0-99 for Options 51C-81/81C xx = 0-31 for Option 11C.
FEAT	RLB	Route List Index.
RLI	xxx	Route List Index to be accessed.
ENTR	xxx	Entry Number for Route List.
LTER	(NO) YES	Local Termination entry (required for incoming calls). Enter NO.
ROUT	0 - 511 0-127	Route Number for Options 51C-81/81C. Route Number for Option 11C.
DMI	(0)-999	Digit Manipulation Index number, as defined in previous step (required for incoming calls).
...		

LD 86 – Configure ESN data.

Prompt	Response	Description
REQ	NEW	Add data.
CUST	xx	Customer number, as already configured in LD 15. xx = 0-99 for Options 51C-81/81C xx = 0-31 for Option 11C.
FEAT	ESN	ESN data.
...		
AC2	xx	NARS Access Code 2.
...		

LD 90 – Configure Special Number Translation.

Prompt	Response	Description
REQ	NEW	Add data.
CUST	xx	Customer number, as already configured in LD 15. xx = 0-99 for Options 51C-81/81C xx = 0-31 for Option 11C.
FEAT	NET	Network Translation Table.
TRAN	aaa	Translator. aaa = AC1, AC2, or SUM. Enter AC2.
TYPE	SPN	Type is Special Number Translation.
SPN	x...x	Special Number Translation.
- FLEN	(0)-24	Flexible Length (required for outgoing calls).
- ITOH	NO	Do not inhibit Time-out Handler (required for outgoing calls).
- RLI	xxx	Route List Index.

LD 90 - Define the Special Number (002) for outgoing Taiwan international calls, and configure the appropriate SPN and ITOH values.

Prompt	Response	Description
REQ	NEW	Add data.
CUST	xx	Customer number, as already configured in LD 15. xx = 0-99 for Options 51C-81/81C xx = 0-31 for Option 11C.
FEAT	NET	Network Translation Table.
TRAN	aaa	Translator. aaa = AC1, AC2, or SUM. Enter AC2.
TYPE	SPN	Type is Special Number Translation.
SPN	002	Define 002 as the Special Number for outgoing Taiwan R1 international calls.
- FLEN	(0)-24	Flexible Length (required for outgoing calls). Enter a value of at least 16.
- ITOH	NO	Do not inhibit Time-out Handler (required for outgoing calls).
- RLI	xxx	Route List Index.

Trunk Anti-Tromboning

LD 17 — Configure TAT functionality on the D-channel.

Prompt	Response	Description
REQ	CHG	Change existing data.
TYPE	ADAN	Type of change.
- ADAN		Action Device and Number.
	NEW DCH x	Add D-channel x.
	CHG DCH x	Change D-channel x.
- CTYP	MSDL	Card type, where MSDL = the Downloadable D-Channel Daughterboard for Option 11C.
- PORT	1	Port number for the Downloadable D-channel.
...		
- IFC		Interface type for D-channel.
	SL1	SL1.
	S100	SL-100,
	D100	DMS-100.
	D250	DMS-250.
...		
- RLS	xx	Release ID of the switch at the far end of the D-channel. Release 21 or higher must be entered.
- RCAP	TAT	Remote Capabilities. TAT must be entered to enable Trunk Anti-Tromboning.

Trunk Optimization (before answer)

Trunk optimization is configured in LD16.

Procedure 25

Respond to the following prompts in LD16

Prompt	Response	Description
REQ	CHG	
TYPE	RDB	Route data block
CUST	0-31	Customer number
ROUT	nnn	Route number 0-511 for NT, XT, 61, 71, and 81 0-127 for ST, 21, STE, 21E
DTRK	YES, (No)	Digital Trunk Route Must be Yes to prompt ISDN
ISDN	YES, (No)	ISDN option
NCRD	YES, (No)	Network Call Redirection. Allows network call redirection messages to be sent (or blocks messages if NCRD =no). Must be Yes to prompt TRO.
TRO	YES, (No)	Trunk Optimization

Trunk to Trunk Connection

LD 15 – Modifications to Customer Data Block.

Prompt	Response	Description
REQ:	CHG	Change.
TYPE:	ATT	Attendant console prompts.
CUST	xx	Customer number.
RTIM	xxx yyy zzz	Enter defined value for the Slow Answer Recall timer where: xxx = 0-(30)-378 Slow Answer Recall yy y= 0-(30)-510 Camp On Recall zzz = 0-(30)-510 Call Waiting Recall

LD 15 – Modifications to Customer Data Block

Prompt	Response	Description
REQ:	CHG	Change
TYPE	NET	Trunk and network options.
CUST	xx	Customer number.
...		
ISDN	YES	Change the Integrated Services Digital Network options.
- PSTN	NO	Public Switched Telephone Network. Limit the number of PSTNs allowed in a network connection to one PSTN. NO= Puts no limit on the number of PSTN connections. YES = Limits the number of PSTN connections.
...		
DITI	YES	Allow Direct Inward Dialing to TIE connections for customer.

TRNX	YES	YES = Allows transfer on ringing of an external trunk over a supervised analog network TIE trunk across private network. NO= Prevents transfer on ringing of an external trunk over a supervised analog network TIE trunk across private network.
EXTT	YES	YES = Allows connection of supervised external trunks. NO = Prevents connection of supervised external trunks.

Virtual network services

LD 17 – Configure Virtual Network Services.

Prompt	Response	Description
REQ	NEW, CHG	New, or change.
TYPE	ADAN aaa xx	Release 19 gate opener. DCH 0 - 15
...		
- USR	VNS SHAV	To define a D-channel used for Virtual Network Services (or ISLD). To define a D-channel shared between PRI and VNS (and ISLD).
- VNSM	0-100	Define the maximum number of VNS channels over the D-channel.
- VNSC	xx	Virtual Network Services Customer number.
- VNSP	0-32700	Private Network Identifier (PNI) of the far end customer.
- VNCA	YES	Network Call Party Name Display is available over this D-channel for VNS.
- VCRD	YES	Network Call Redirection is available over this D-channel for VNS.
- VTRO	YES	Trunk Route Optimization before answer is available over this D-channel for VNS. This prompt is optional and is not a prerequisite for VNS.

The following steps describe how to:

- Add or create a new single VDN or a block of contiguous VDNs
- Remove a single VDN or a block of contiguous VDNs
- Disable or enable a block of VDNs, or print VNS information for a customer

LD 79 - Add or create a new single VDN or a block of contiguous VDNs.

Prompt	Response	Description
REQ	NEW	Add or create a new single VDN or a block of contiguous VDNs. Note: The CHG command is not supported; you must use the NEW command to enter information.
TYPE	VNS	Virtual Network Services.
CUST	xx	Customer number.
VNDN	xx...x 1-4000 xx...x <CR>	Individual VDN to be added. 1-4000 = number of contiguous VDN to be added xx...x = first VDN to be added. You may add another single VDN or a block of contiguous VDNs by entering <CR> after the VNDN entry. VNDN is prompted until <CR> is entered. In this case, the REQ prompt will appear again. Note: For the above entries, the VDNs must be part of the customer's numbering plan.

LD 79 - Remove a single VDN or a block of contiguous VDNs, or remove all existing VDN data blocks.

Prompt	Response	Description
REQ	OUT	Remove a single VDN or a block of contiguous VDNs, or remove all existing VDN data blocks. Note: You cannot remove only certain VDNs from a block; you have to remove the entire block.
TYPE	VNS	Virtual Network Services.
CUST	xx	Customer number. At least one D-Channel must be configured with <code>USR = VNS</code> or <code>USR = SHAV</code> and having <code>VNS = customer number</code> .
VNDN	XALLVDNS xx...x <CR>	Remove all VNS data blocks. Remove an individual VDN, or the first VDN of a block of contiguous VDNs. You may remove another single VDN, or a block of contiguous VDNs, by entering <CR> after the VNDN entry. VNDN is prompted until <CR> is entered. In this case, the REQ prompt will appear again. If XALLVDNS is entered, the message REMOVE ALL VDN BLOCKS? is then output, followed by the CONF prompt.
CONF	YES	To confirm the removal of all VDN blocks.

LD 79 – Disable or enable single VDN, or a block of contiguous VDNs.

Prompt	Response	Description
REQ	DIS ENL	Disable a single VDN or a block of contiguous VDNs Enable a single VDN or a block of contiguous VDNs
TYPE	VNS	Virtual Network Services.
CUST	xx	Customer number. At least one D-Channel must be configured with USR = VNS or USR = SHAV and having VNS = customer number.
VNDN	xx....x <CR>	Individual VDN, or first VDN of a block of contiguous VDNs to be disabled or enabled. You may enable or disable another single VDN, or a block of contiguous VDNs, by entering <CR> after the VNDN entry. VNDN is prompted until <CR> is entered. In this case, the REQ prompt will appear again.

LD 79 – Print VNS information for a customer.

The information is output after the customer number is entered in response to the CUST prompt.

Prompt	Response	Description
REQ	PRT	Print VDN information.
TYPE	VNS	Virtual Network Services.
CUST	xx	Customer number. At least one D-Channel must be configured with USR = VNS or USR = SHAV and having VNS = customer number. The VNS information is output for the customer. For a range of VDNs, the first VDN is displayed, followed by “—” and the last VDN of the block. On the same line, the number of VDNs in the block is displayed in brackets. For a single VDN, the VDN is displayed followed by “(1)”. If a block is disabled, the above indication is followed by the number of VNS calls still using a VDN in the block. These calls have to be cleared before the VDN block can be removed. At the end, the total number of VDNs configured for the customer is output. A sample output could be: <pre>7676 (1) 8100-8199 (100) TOTAL NUMBER OF VDN FOR CUST 2: 101</pre> If the VDN block 8100-8199, containing 100 VDNs, is disabled, the output would be: <pre>7676 (1) 8100-8199 (100) *DISABLED - VDN USED: 2* TOTAL NUMBER OF VDN FOR CUST 2: 101</pre> In this case, the VDN block is disabled and two VNS calls are still using two VDNs. These two calls must first be cleared before the VDN block 8100-8199 can be removed.

LD 86 – Assign D-channel number and VNS digit manipulation index to be used when signaling on the Bearer trunk and on the D-channel.

Prompt	Response	Description
REQ	NEW, CHG	New, or change.
...		
ENTR	0-63	Entry number for NARS/BARS Route List.
ROUT	0-511	The number of the route to be associated with the VNS Bearer channel.
VNS	YES	Virtual Network Services.
VDCH	0-63	The D-channel used for these VNS calls (must be configured in LD 17).
	0-15	For Option 11.
VDMI	xxx	The Digit Manipulation Table to be used on the VNS D-channel.
	(0)	No digit manipulation required.
	1-31	For Coordinated Dialing Plan feature.
	1-255	For NARS/BARS.
VTRK	1-(20)-100	Number of VNS trunks allowed on the route.
DMI	xxx	The Digit Manipulation table to be used on the VNS Bearer.
	(0)	No digit manipulation required.
	1-31	For Coordinated Dialing Plan feature.
	1-255	For NARS/BARS.

LD 16 – The following TIMR, TIMR, and VRAT prompts have been added to this overlay:

Prompt	Response	Description
REQ	NEW, CHG	New, or change.
TYPE	RDB	Route Data Block.
CUST	xx	Customer number.
ROUT	0-511	The number of the route to be associated with the incoming VNS Bearer channel.
CNTL	YES	Change control of timers.
- TIMR	VSS (0) 1 2-1023	VNS Set Speechpath Timer 0 = Do not answer the bearer channel until the terminating party answers. 1 = Answer the Bearer channel immediately on arrival. 2-1023 = Answer the Bearer channel after specified seconds (rounded down to two-second multiples) if the terminating party has not already answered.
- TIMR	VGd 0-(6)-31	VNS Guard Timer The time allowed for the Bearer trunk call to disconnect, in seconds. This is a guard timer on the associated VNS DN.
...		
VRAT	(NO) YES	VNS Return Attendant Tones Option Do (not) answer an attendant extended call over VNS immediately on the incoming Bearer trunk.

Virtual Network Services Virtual Directory Number Expansion

LD 79 - Add or create a new single VDN or a block of contiguous VDNs.

Prompt	Response	Description
REQ	NEW	Add, or create a new single VDN or a block of contiguous VDNs. Note: The CHG command is not supported; you must use the NEW command to enter information.
TYPE	VNS	Virtual Network Services.
CUST	xx	Customer number. At least one D-Channel must be configured with USR = VNS or USR=SHAV and having VNS = Customer number.
VNDN	xx...x 1-4000 xx...x <CR>	Individual VDN to be added. 1-4000 = number of contiguous VDN to be added xx...x = first VDN to be added. You may add another single VDN or a block of contiguous VDNs by entering <cr> after the VNDN entry (VNDN is prompted until <cr> is entered.) In this case, the REQ prompt will appear again. Note: For the above entries, the VDNs must be part of the customer's numbering plan.

LD 79 - Remove a single VDN or a block of contiguous VDNs, or remove all existing VDN data blocks.

Prompt	Response	Description
REQ	OUT	Remove a single VDN or a block of contiguous VDNs, or remove all existing VDN data blocks. Note: You cannot remove only certain VDNs from a block; you must remove the entire block.
TYPE	VNS	Virtual Network Services.
CUST	xx	Customer number. At least one D-Channel must be configured with <code>USR = VNS</code> or <code>USR = SHAV</code> and having <code>VNS = Customer number</code> .
VNDN	XALLVDNS xx...x <CR>	Remove all VNS data blocks. Remove an individual VDN, or the first VDN of a block of contiguous VDNs. You may remove another single VDN, or a block of contiguous VDNs, by entering <cr> after the VNDN entry (VNDN is prompted until <cr> is entered.) In this case, the REQ prompt will appear again. If XALLVDNS is entered, the message REMOVE ALL VDN BLOCKS? is then output, followed by the CONF prompt.
CONF	YES	To confirm the removal of all VDN blocks.

LD 79 – Disable or enable single VDN, or a block of contiguous VDNs.

Prompt	Response	Description
REQ	DIS ENL	Disable a single VDN or a block of contiguous VDNs Enable a single VDN or a block of contiguous VDNs
TYPE	VNS	Virtual Network Services.
CUST	xx	Customer number. At least one D-Channel must be configured with USR = VNS or USR = SHAV and having VNS = Customer number.
VNDN	xx....x <CR>	Individual VDN, or first VDN of a block of contiguous VDNs to be disabled or enabled. You may enable or disable another single VDN or a block of contiguous VDNs, by entering <CR> after the VNDN entry (VNDN is prompted until <CR> is entered.) In this case, the REQ prompt will appear again.

LD 79– Print VNS information for a customer.

The information is output after the customer number is entered in response to the CUST prompt.

Prompt	Response	Description
REQ	PRT	PRT = print VDN information.
TYPE	VNS	Virtual Network Services.
CUST	xx	Customer number. At least one D-Channel must be configured with USR = VNS or USR = SHAV and having VNS = Customer number. The VNS information is output for the customer. For a range of VDNs, the first VDN is displayed, followed by “—” and the last VDN of the block. On the same line, the number of VDNs in the block is displayed in brackets. For a single VDN, the VDN is displayed followed by “(1)”. If a block is disabled, the above indication is followed by the number of VNS calls still using a VDN in the block. These calls have to be cleared before the VDN block can be removed. At the end, the total number of VDNs configured for the customer is output. A sample output could be: <pre>7676 (1) 8100–8199 (100) TOTAL NUMBER OF VDN FOR CUST 2: 101</pre> If the VDN block 8100-8199, containing 100 VDNs, is disabled, the output would be: <pre>7676 (1) 8100–8199 (100) *DISABLED – VDN USED: 2* TOTAL NUMBER OF VDN FOR CUST 2: 101</pre> In this case, the VDN block is disabled and two VNS calls are still using two VDNs. These two calls must first be cleared before the VDN block 8100-8199 can be removed.

LD 17 – Define the maximum number of VNS channels supported by a D-Channel.

Prompt	Response	Description
REQ	CHG, END	Change, or exit LD 17.
TYPE	ADAN	Action Device and Number.
- ADAN	CHG DCH 0-63	The D-Channel number for Options 51C, 61C, 81, 81C.
	CHG DCH 0-15	For Option 11C.
...		
- USR	VNS SHAV	VNS = dedicated VNS D-Channel SHAV = shared VNS D-Channel
...		
- - VNSM	1-300	Maximum number of VNS channels supported by the D-Channel. Note: This is the potential VNS capability for the D-Channel and is not associated with other restrictions placed on VNS capability, such as the number of VDNs.

LD 86 – Define the maximum number of VNS trunks allowed on the route list entry.

Prompt	Response	Description
REQ	NEW, CHG	New, or change.
CUST	xx	Customer number.
FEAT	RLB	Route list data block feature.
RLI	0-MXRL	The Route List Index to be associated with the VNS Bearer Channel.
ENTR	0-63	The entry within the Route List Index to be associated with the VNS Bearer Channel.
ROUT	0-511	The number of the Route to be associated with the VNS Bearer Channel.
VNS	YES	Virtual Network Services.
...		
VTRK	1-(20)-254	Number of VNS Trunks allowed on the route.

Appendix A — Network Numbering

Overview

Whichever dialing plan is used for call routing over a Network from Meridian 1, the basic principal is the same with all programming being carried out in overlays 86, 87 and 90 which are called Electronic Switched Network (ESN).

Codes for translation are identified as either Local or Distant then route selection and digit manipulation are programmed accordingly.

Local Codes

No route selection is required in this case and only digit manipulation can be carried out.

Distant Codes

A route must be selected and the code points towards a Route List Block.

Route List Block

A Route List Block can be pointed to by one or more distant codes and contains the first and subsequent choice routes to be selected for this call. Certain criteria must be met to allow a choice of Route, the most used ones being the time of day as set in the Time of Day Schedule and the minimum FRL required by the call originator.

A Route can also be defined as an Expensive route in a particular set up case.

Whichever route is selected for a call, a Digit Manipulation Block is selected for that particular case.

Time of Day Scheduling

Up to eight time periods can be set up and if required, a route list block entry can point to a time period for selection.

Digit Manipulation Block

A Digit Manipulation Block can be pointed to from one or more local code or Route List Block entry.

Digit manipulation block number 0 has no effect on digits while other blocks are programmed as required to delete a quantity of leading digits and insert specified digits in front of the remaining digit string.

When selecting a route, the route ACOD does not need to be inserted as route selection is made by the Route List Block.

Network Numbering Plans

The following are descriptions and examples of Network numbering plans available within a Meridian 1 network. The numbering plans available enable connection of Meridian 1 in an existing network although restrictions may be imposed by limitations and working methods of other PBXs.

These numbering plans are fully supported on a Q.931 network.

Certain restrictions exist when Meridian 1 is connected using DPNSS1, DASS2 or APNSS although this does not affect connectivity. These restrictions will be detailed later.

These numbering plans are achieved from two base plans:

- 1** UNIFORM DIALLING - users dial a network access code followed by a location code followed by the required DN number.
- 2** CO ORDINATED DIALLING - The dialled number is used to identify the distant node and may be the whole of the required DN number.

The numbering plans used allow for access to and from the public network either direct or via another node.

Uniform Dialing

Each PBX has its own Location code (LOC) and access from users is by dialing a network access code.

DNs can be the same in each Node.

DNs can be the same as Location codes at other nodes.

Uniform dialing is achieved using the Network Alternate Route Selection feature (NARS).

When using Network Alternate Route Selection (NARS), two network access codes can exist (AC1, AC2).

There is no strict rule as to the use of these access codes but one will be used for the Public network and the other for the Private network.

If required, Dial Tone can be given after the Access Code.

AC1 or AC2 is 1 - 4 digits.

LOC is 1 - 7 digits.

DN is 1 - 7 digits.

** LOC + DN can be maximum 10 digits.

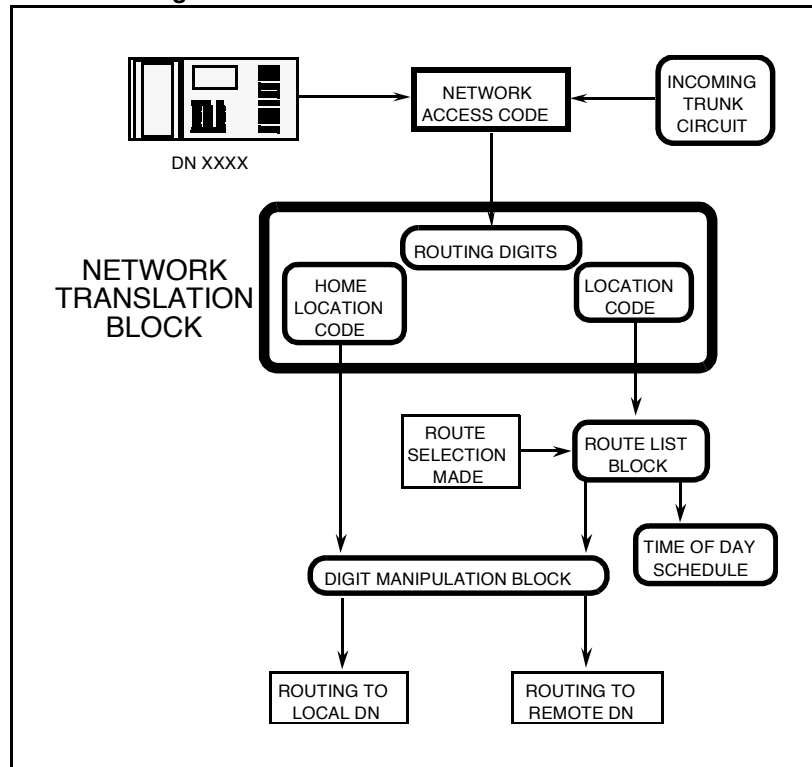
The number of digits expected can be specified for each Location Code.

The access code is used to reach the network translation tables and must be re-inserted if necessary.

Refer to later sections on the following:

- Use of AC2
- Private Network break out to Public Network
- Recognition of On Network numbers

Figure 136
Uniform Dialing Plan Schematic



Programming of Uniform Dialing

Programming for all Network numbering plans is done in overlays 86, 87 and 90 with some areas being unique to the feature (NARS or CDP) and some being common to all (Digit manipulation blocks and Route list blocks). Whichever dialing plan is used, the common areas can be used by both.

The following procedures will cover the necessary overlays and prompt/responses to set up a Uniform Dialing plan:

Step 1: LD 86—ESN Data Block

This sets several general items, and would normally be set once and then changed occasionally. When setting the ESN data block, several other prompts relating to maximums must be answered and should be set to 0 if not used.

Prompt	Response	Description
REQ	NEW, CHG, PRT, OUT	
CUST	XX	Customer number
FEAT	ESN	NARS/ BARS data block
MXLC	0-999	Max. no. of network location codes
MXDM	1-1000	Max. no. of digit manipulation tables
MXRL	0-999	Max. no. of Route list table
AC1	xxxx	1-4 digit Network access code one
AC2	xxxx	Network access code two
DLTN	YES/NO	Dial tone after AC1 or AC2
TODS		Time of day schedule for use from route list blocks
	0-7, 0-23, 00-59, 0-23 00-59	Schedule number, start time, end time

Step 2: LD 87—Network Control Block

This must be set before any translations are programmed even if all values are set to default. This may have been previously set if NCOS and FRL have been linked to set up NFCR.

Prompt	Response	Description
REQ	NEW, CHG, PRT, OUT	
CUST	XX	Customer number
FEAT	NCTL	Network control block

Step 3: LD 86—Digit Manipulation Block

Whichever dialing plan is used, digit manipulation blocks must be programmed first.

Prompt	Response	Description
REQ	NEW, CHG, PRT, OUT	
CUST	XX	Customer number
FEAT	DGT	Digit manipulation data block
DMI	XXX	Digit manipulation table index number
DEL	(0)-15	No. of leading digits to be deleted
INST	xxx....xx	leading digits to be inserted (0-20)

Step 4: LD 86—Route List Data Block

The programming in the Route list block is the part that selects the route for an outgoing call together with the criteria to allow that route to be selected. Each Route List Block can have 64 entries, entry 0 being the first choice route then entry 1 as the 2nd choice etc.)

Prompt	Response	Description
REQ	NEW, CHG, PRT, OUT	
CUST	XX	Customer number
FEAT	RLB	Route list data block
RLI	xxx	Route list index number
ENTR	xx	Routing option (0-63)
ROUT	xxx	Route number
TOD	0-7	Time of day control as set in stage 1
DMI	xxx	Digit manipulation table number

Step 5: LD 15—Customer Data Block

Program as required for the particular setup.

Prompt	Response	Description
REQ	CHG	
CUST	XX	Customer number
AC2	LOC	If AC2 is used for uniform dialing
HLOC	XXX	Home location code to be sent with calling line ID

Step 6: LD 90—Network Translation Block

Prompt	Response	Description
REQ	NEW, CHG, PRT, OUT	
CUST	XX	Customer number
FEAT	NET	Network translation tables
TRAN	AC1/ AC2	Translation is for AC1 or AC2

For Home Location Code:

TYPE	HLOC	Home location code
HLOC	xxxxxxx	Home location code (up to 7 digit)
DMI	xxx	Digit manipulation table

For Distant Location Code:

TYPE	LOC	Network locations
LOC	xxxxxxx	Location code (up to 7 digit)
FLEN	(0)-10	Number of dialled digits following AC1/AC2
RLI	xxx	Route list table index no.
LDN	xxxx	Listed directory of the location Note: Digit(s) must be entered against LDN although they have no significance in the UK

Co-ordinated Dialing

Co-ordinated Dialing Plan (CDP) identifies up to 7 digits of a dialled string and routes the call according to the Network numbering plan.

The differences in a network using CDP is in the digits dialled by the user, the length of the DN numbers and the spread of unique numbers in each Network node.

In all cases, a unique number must be identified to control routing of calls, this number being called a Steering Code.

LOCAL STEERING CODE (LSC) - this is used to route a call locally, the digits normally being received on an incoming trunk circuit although they may be dialled locally.

DISTANT STEERING CODE (DSC) - used when the digits dialled or digits received from an incoming trunk are for a Network DN.

Maximum digits for DSC + DN is 10.

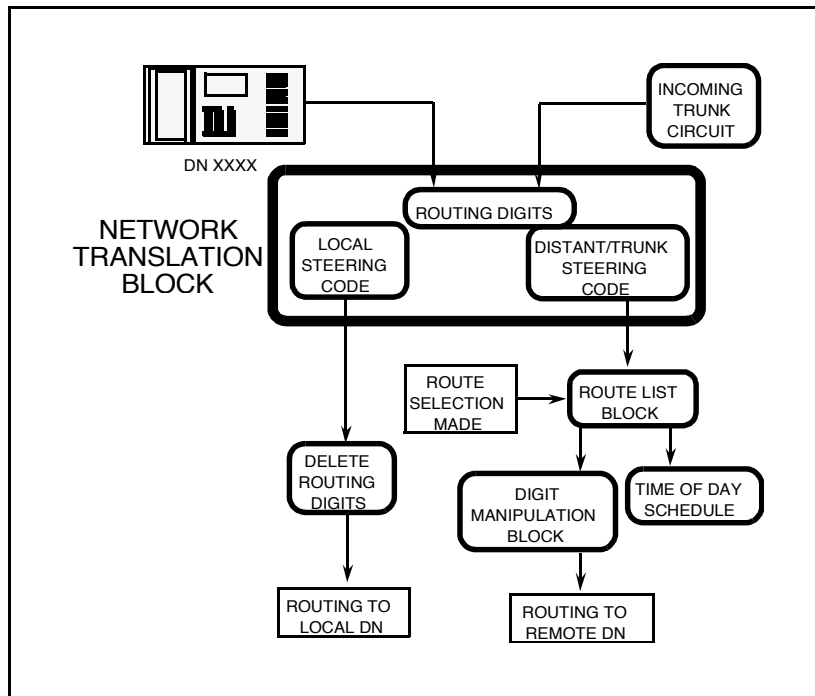
TRUNK STEERING CODE (TSC) - used for the dialing of Off Network numbers that may require breakout over the Private network or dialing to other private networks with unknown digit lengths.

Maximum digits for TSC + DN is 16.

Steering codes can be 1 - 7 digits and up to 10,000 steering codes can be specified in each Meridian 1 node.

The expected number of digits can be programmed against each DSC or TSC.

Figure 137
Co-ordinated Dialing Plan Schematic



Vacant Number Routing

The use of a fully transferable numbering scheme as shown in CDP example 3, would require programming in every node of a Network when a DN is moved between sites, as well as extensive programming to set up the Network.

The feature Vacant Number Routing (VNR) allows nodes within the Network to use another 'Database Node' for call routing.

A node that uses VNR must be no further than 1 hop from a Database node although more than 1 database node may exist within the network.

When a node is programmed for VNR, all calls originating within that node to a Vacant number are routed to a Database node for action.

Programming a Co-ordinated Dialing Plan

Programming for the Network numbering plans is done in overlays 86, 87 and 90 with some areas being unique to the feature (NARS or CDP) and some being common to all (Digit manipulation blocks and Route list blocks). Whichever dialing plan is used, the common areas can be used by both.

The following procedures will cover the necessary overlays and prompt/responses to set up a Co-ordinated Dialing plan:

Step 1: LD 86—ESN Data Block

This sets several general items, and would normally be set once and then maybe changed occasionally. Some items are not relevant to all Network set ups. When setting the ESN data block, several other prompts relating to maximums must be answered and should be set to 0 if not used.

Prompt	Response	Description
REQ	NEW, CHG, PRT, OUT	
CUST	XX	Customer number
FEAT	ESN	NARS/ BARS data block
MXDM	1-1000	Max. no. of digit manipulation tables
MXRL	YES/ NO	CDP feature allowed
MXSC	0-10000	Max. no. steering codes
NCDP	3-10	Maximum number of digits in CDP DN
TODS		Time of day schedule for use from route list blocks
	0-7, 0-23, 00-59, 0-23 00-59	Schedule number, start time, end time

Step 2: LD 87—Network Control Block

This must be set before any translations are programmed even if all values are set to default. This may have been previously set if NCOS and FRL have been linked to set up NFCR.

Prompt	Response	Description
REQ	NEW, CHG, PRT, OUT	
CUST	XX	Customer number
FEAT	NCTL	Network control block

Step 3: LD 86—Digit Manipulation Block

Whichever dialing plan is used, digit manipulation blocks must be programmed first.

Prompt	Response	Description
REQ	NEW, CHG, PRT, OUT	
CUST	XX	Customer number
FEAT	DGT	Digit manipulation data block
DMI	XXX	Digit manipulation table index number
DEL	(0)-15	No. of leading digits to be deleted
INST	xxx....xx	leading digits to be inserted (0-20)

Step 4: LD 86—Route List Data Block

The programming in the Route list block is the part that selects the route for an outgoing call together with the criteria to allow that route to be selected. Each Route List Block can have 64 entries, entry 0 being the first choice route then entry 1 the second choice, etc.

Prompt	Response	Description
REQ	NEW, CHG, PRT, OUT	
CUST	XX	Customer number
FEAT	RLB	Route list data block
RLI	xxx	Route list index number
ENTR	xx	Routing option (0-63)
ROUT	xxx	Route number
TOD	0-7	Time of day control as set in stage 1
DMI	xxx	Digit manipulation table number

Step 5: LD 15—Customer Data Block

Program as required for the particular setup.

Prompt	Response	Description
REQ	CHG	
TYPE	CDB	Customer Data Block
CUST	XX	Customer number
HLOC	XXX	Location code sent with calling line ID
LSC	XXX	Local steering code to be sent with calling line ID

Step 6: LD 87—Configuration of CDP

Prompt	Response	Description
REQ	NEW, CHG, PRT, OUT	
CUST	XX	Customer number
FEAT	CDP	Coordinated dialing plan
RLI	xxx	Route list index number
ENTR	xx	Routing option (0-63)
ROUT	xxx	Route number
TOD	0-7	Time of day control as set in stage 1
DMI	xxx	Digit manipulation table number

For Local Steering Code

Prompt	Response	Description
TYPE	LSC	Local steering code
LSC	xxxxxx	Local steering digits (1- 7-digit)
DMI	xxx	Digit manipulation block to use Note: DMI is only prompted with LSCM package
DEL	xxx	No. of leading digits to be deleted

For Distant Steering Code

Prompt	Response	Description
TYPE	DSC	Distant steering code
DSC	xxxxxx	Distant steering digits (1- 7-digit)
FLEN	(0)- 10	Expected dialled digits
DSP	LSC HLOC DN	Display option to destination as set DN only is sent

Configuring Vacant Number Routing (VNR)

The UDP / CDP must have been configured before the VNR feature is configured in the database.

- 1 Configure the required Route List tables in LD 86.

NOTE - Digit manipulation is not allowed when using VNR and the Route list block must point to DMI 0.

- 2 Configure VNR in LD 15 as follows

Prompt	Response	Description
REQ	CHG	
TYPE	CDB	
CUST	xx	
VNR	YES/(NO)	Vacant Number Routing
RLI	xxx	Route list table
CDPL	1-(10)	No. of dialled digits for CDP
LOCL	1-(10)	No. of dialled digits for LOC

Group Dialing

Group dialing is purely a name for a mixture of the uniform and co-ordinated dialing plans.

Groups of PBXs are configured for co-ordinated dialing, and allocated a location code. The groups are configured together as in uniform dialing, users dialing the network access code to make calls outside their own zone.

All the rules of NARS and CDP exist within themselves.

When using NARS and CDP together, certain rules must be followed when routing within the Network translation tables.

Allowed - HLOC routed to Steering code

Not Allowed - LSC routed Steering code

Mixed Length CDP Networks

This type of dialing plan is similar in theory to the Group dialing plan but with the ability to use CDP throughout.

The example below duplicates the previously used group dialing plan with the users dialing the same digits, the network access code being used as part of the steering code.

The one rule which must be strictly followed in this situation is that relating to call routing after analysis as a Local Steering Code.

A local Steering Code must be terminated locally.

Alternate Routing

If a call cannot be made over the first choice route (entry 0 in the route list block), the next entry will be looked at and actioned accordingly.

A typical route list block will be programmed as follows:

- Entry 0 - shortest route to the destination within the Network

- Entry 1 - alternate route within the Network
- Entry 2 - route via PSTN by deleting digits and inserting PSTN code for destination.

Every entry in a route list block will be tried even though all the entries may not be sequential.

During call set up a call cannot return over the same route that it originated from (anti-tromboning) and if this situation occurs, this will be treated as congestion and the next choice will be made from the route list block.

The use of alternate routing in a complex network must be well planned and it is also possible to perform digit conversion to give a dummy steering code to route a call under different control throughout the network.

The use of the Drop back busy feature must also be taken into consideration.

Calling Line Identity

The Calling Line Identity (CLI) or Originating Line Identity (OLI) is the network number belonging to the originator of a call which is sent to the destination. The CLI is used for display purposes if the destination can receive this and is also used for networking features e.g. Ring Again and CLID in CDR. The CLI is formed from the originating extension DN prefixed by a site code programmed in Overlay 15, although there are several options available.

When making calls in a Q931 Private network, the CLI should be prefixed by the letter "H" indicating that the call has been identified as having a valid call type. If the "H" is missing, features will not be able to be invoked.

The above applies to Q.931 trunks only. DPNSS1 trunks have specific rules that are covered later in this chapter.

Private Network Hop Off

Private network hop off or break out is a means of making a call to the public network via a node in the private network that will give the lowest cost call. Each node in the network is programmed to either give immediate break out to PSTN or send the dialled digits towards another node.

This can be achieved with either CDP using Trunk Steering Codes (TSC) or using Basic Alternate Route selection (BARS). Some public exchanges may reject local calls that are accessed using the STD code. This can be avoided by deleting the STD code in digit manipulation at the break out exchange.

Private Network Hop Off using CDP

The PSTN access code (normally 9) followed by the STD code to be identified is programmed as a Trunk Steering Code and route selection will be made as with Distant Steering Codes according to the required routing.

A maximum of 7 digits can be identified as a TSC and the total dialled digits can be up to 16. All codes that could be dialled must be entered as Trunk Steering Codes being either 4 or 5 digits e.g. 9021/90753/9071.

The following is the programming relating to Trunk Steering codes presuming that the control blocks, digit manipulation and route list blocks have been programmed.

Step 1: LD 87—Network Control Block

Prompt	Response	Description
REQ	NEW, CHG, PRT, OUT	
CUST	XX	Customer number
FEAT	CDP	Coordinated dialing plan
TYPE	TSC	Trunk steering code
TSC	xxxxxx	TSC digits (1 - 7 digit)
FLEN	(0) - 16	Expected dialled digits
RLI	xxx	Routing list index number

Private Network Hop Off using BARS

Basic Alternate Route Selection has only one Access code (AC1), although when the NARS software exists then AC1 and/or AC2 can be used for any feature of NARS or BARS. The following are the options available as types of call following a BARS/NARS access code:

- 1** LOC - Private Network Location code
- 2** HLOC - Private Network Home Location code
- 3** NPA - Public Network Area code
- 4** HNPA - Public Network Home Area code
- 5** NXX - Public Network Central Office code
- 6** SPN - Special Number (1-10 digits)

Options 1 and 2 have been described earlier and options 3, 4 and 5 are designed for the North American market.

Special Numbers (SPN) are programmed to give break out within a Network.

The Access code used, AC1 or AC2 is programmed with the Public Network access code (normally 9) then each STD code is programmed as a SPN which is then pointed towards a Route List block to give the required route selection.

Dial tone can be given following AC1/AC2 if required.

AC1/AC2 must be inserted with digit manipulation if the call is routing via the private network.

Identification of On Network Numbers

This is the ability to identify a dialled PSTN number as one that belongs to one of the nodes on the Private network, then performing digit manipulation on the dialled number to route it via the Private network for termination.

The flexibility of the feature could identify a private network number and route via the PSTN.

This is achieved by using Basic Alternate Route selection (BARS), and Special numbers (SPN). The feature called Supplemental Digit Restriction and Recognition is used where numbers dialled following a SPN are compared to those programmed and if a match is found then different routing action is taken.

This feature can be used separately or in conjunction with break out.

Numbers can be identified as:

- A local DID number that should be terminated using part of the dialled number to identify the DN. This is done by pointing the code direct to a Digit Manipulation Block. (LDID)
- A local non DID number that will be routed to the attendant via the Listed directory number (LDN). No digit manipulation is necessary as routing is made automatically. (LDDD)
- Alternate Routing Remote Number which will point the call at an Alternate Route List Index (ARLI) enabling either different routing or routing using different Digit Manipulation. (ARRN)

Overlay 90 also allows numbers following SPNs to be programmed as DID and DDD codes but ARRN gives more flexibility and has been designed for UK applications.

Programming of BARS

The following is the programming for BARS Special numbers to give either network hop off and/or identity of on network numbers. It is presumed that network control blocks, digit manipulation and route list blocks have been previously programmed.

Step 1: LD 90

Prompt	Response	Description
REQ	NEW, CHG, PRT, OUT	
CUST	XX	Customer number
FEAT	NET	Network translation tables
TRAN	SPN	
SPN	XXXX XXXX XX	Code for routing must be entered as groups of 4 digits separated by space (max 10 digits)
FLEN	(0) - 16	Number of digits dialled following AC1/AC2
RLI	XX	Route list index for call routing
SDRR	LDID LDDD ARRN	Enter type of digit recognition required, or <cr> to return to SPN prompt
SPN	XXX	More special numbers or <cr> to exit

Step 2: LD 15

Prompt	Response	Description
REQ	CHG	
TYPE	CDB	Customer Data Block
CUST	XX	Customer number
AC2	SPN	Only if using AC2

Programming for Supplemental Digit Recognition

Local DID Numbers

Step 1 - LD 90

Prompt	Response	Description
SDRR	LDID	Supp. digits are local DID
DMI	XX	DMI to be used with this LDID code
LDID	XXXX	LDID digits
LDID	XXXX	Other digits or <cr> to return to SDRR for more recognition against the same SPN.

Local Non-DID Numbers

Step 1 - LD 90

Prompt	Response	Description
SDRR	LDDD	Supp. digits are local attendant
LDDD	XXXX	LDDD digits
LDDD	XXXX	Other digits or <cr> to return to SDRR for more recognition against the same SPN.

Remote Numbers

Step 1 - LD 90

Prompt	Response	Description
SDRR	ARRN	Supp. digits are for remote routing
ARRN	XXXX	Digits for alternate routing
ARLI	XX	Route list index for this ARRN
ARRN	XXXX	Another number or <cr> to return to SDRR for more recognition against the same SPN.

Number Lengths

With Q.931, the outgoing call setup message will not be sent until certain criteria has been met, which differs slightly between UDP and CDP.

Uniform Dialing - NARS/BARS

The call setup message is sent on one of the following:

- 1 The number of digits is equal to the programmed number length (FLEN).
- 2 The Network interdigit timeout (NIT in overlay 15) has been reached.
- 3 The # key has been pressed.

If FLEN is set to 0, then point 1 will never occur.

The call will not setup if the number of digits dialled is less than the length of the LOC or SPN.

The value of FLEN should always be set to the maximum possible digit length.

Figure 138
CDP Number Lengths

TYPE OF NUMBER		MAXIMUM FLEN	
DSC		10	
TSC		16	

FLEN GREATER THAN 0		
DIGITS DIALLED	FLEN	CALL SET UP
X	LESS THAN X	WHEN FLEN IS REACHED
	X	AFTER X DIGITS
	GREATER THAN X	AFTER NIT OR #
LESS THAN DSC or TSC	ANY VALUE	CANNOT SETUP

NCDP	LSC LENGTH	DEL	RESULT
4	ANY	< 4	OK
		> 3	NOT OK
9	ANY	> 1	OK
		< 2	NOT OK
10	< 3	ANY	NOT OK
	> 2	ANY	OK

Number Lengths using CDP

The call setup message is sent on one of the following:

- 1** The number of digits is equal to the programmed number length (FLEN).
- 2** The Network interdigit timeout (NIT in overlay 15) has been reached.
- 3** The # key has been pressed.
- 4** When the value of NCDP in overlay 86 has been reached and FLEN is set to 0.

The call will not setup if the number of digits dialled is less than the length of the DSC or TSC.

The value of FLEN should always be set to the maximum possible digit length as it overrides the use of NCDP.

CDPL In Overlay 86

As mentioned above, the value of NCDP is overridden by that of FLEN, but the following factors should be taken into account when setting this value:

- 1** The value of NCDP minus the number of digits deleted against any LSC must be greater than 0 and less than 8.
- 2** The value of NCDP minus the length of any LSC must be less than 8.

The value of NCDP cannot be changed if any Steering codes exist.

Figure 139
CDP Number Lengths

TYPE OF NUMBER		MAXIMUM FLEN	
DSC		10	
TSC		16	

FLEN GREATER THAN 0		
DIGITS DIALLED	FLEN	CALL SET UP
X	LESS THAN X	WHEN FLEN IS REACHED
	X	AFTER X DIGITS
	GREATER THAN X	AFTER NIT OR #
LESS THAN DSC or TSC	ANY VALUE	CANNOT SETUP

NCDP	LSC LENGTH	DEL	RESULT
4	ANY	< 4	OK
		> 3	NOT OK
9	ANY	> 1	OK
		< 2	NOT OK
10	< 3	ANY	NOT OK
	> 2	ANY	OK

Access Control and Restrictions

Several parts of the software can control access to trunks and give restrictions, many of which are not envisaged to be used in Europe.

Those used are mentioned below and detail is given where these affect Networking.

TGAR to TARG

Using the route ACOD will always follow TGAR to TARG.

Using CDP overrides any TGAR to TARG.

Using BARS/NARS gives the option to follow or override TGAR to TARG on a customer basis.

In LD 86:

Prompt	Response	Description
REQ	CHG	
CUST	XX	
FEAT	ESN	NARS/ BARS data block
TGAR	YES (NO)	Follow TGAR to TARG barring Override TGAR to TARG

Trunk group busy keys

The operation of trunk group busy keys on the Attendant console will block access to that particular route if the call originator has a TGAR of 7 or less irrespective of the dialing method used.

NFCR trees

A call originators FRL value will still be applied on a route selected with NARS, BARS or CDP and follow the same rules as if the call was made using the ACOD.

If code restrictions are to be applied on a remote site, then the trunks at that site must have CLS=TLN as the call originators CLS is not passed around the network. It is recommended that code restrictions are not applied around the network, but at the point where the call breaks out.

NCOS - FRL

The NCOS of a call originator can be sent around a Q931 network to control any further routing of that call, but if a call is originated from a point without a NCOS then on a transit call the NCOS of an incoming route will be used.

In each node within the network that route selection is made, the NCOS is associated with a FRL.

Under normal circumstances, all NCOS to FRL linking within a network would be the same.

To send NCOS, follow this procedure:

In LD 16 - Route Data Block:

Prompt	Response	Description
REQ	CHG	
TYPE	RDB	
CUST	xx	
ROUTE	xx	
SIGO	ESN3	Sends NCOS to interworking switch. Must be set at both ends

When the NCOS is sent, this can be seen in the digits in call trace. The two digits for the NCOS are always prefixed by 1.

Route list block

Each entry in a route list block can be assigned a minimum FRL to be allowed that particular choice of route.

If the FRL prompt is set to 3 then only calls with a FRL of 3 and above can select that route.

The above mentioned is the normal situation and the prompt 'MFRL' in the Route list block is set to the default 'MIN'.

If the prompt 'MFRL' is set to a value between 0 and 7 then this will be the minimum FRL allowed access to all entries in the Route list block.

Schedule Access restrictions

Schedule Access restrictions control an extensions class of service values which would then be applied to the call in accordance with the above mentioned points.

Access Restriction Tables

Any settings made in overlay 56 for the barring of trunk to trunk connections will apply to all calls.

DPNSS1 Considerations

Meridian 1 is designed to use Q931 as its networking protocol and the main aspects of networking features are designed around this.

In the UK, DPNSS1 is widely used and to meet the requirements of connecting to a DPNSS1 network the X20 version of software has been merged into X11 to handle IDA trunks.

The limitations are still the same as in X20 and the following considerations have to be made:

- 1** NARS is not supported
- 2** Outgoing calls can be made using CDP or BARS special numbers
- 3** Incoming calls must be terminated using CDP local steering codes or direct to DN
- 4** Transit calls should use CDP distant or trunk steering codes
- 5** The OLI/CLI sent is made up of LSC + DN (LSC in overlay 15)
- 6** Any digits programmed against INST in LD 16 will be inserted for routing and also in front of the received OLI/CLI string.

Points 1 to 4 above relate to the routing of a call and fit into the examples covered earlier in this section.

Points 5 and 6 must be closely looked at if Meridian 1 is connecting to a DPNSS1 network which uses anything more complicated than a simple linked numbering plan (CDP in meridian language).

OLI/CLI in a DPNSS1 Network

Several complexed numbering schemes may exist on DPNSS1 networks with different switches having different methods of sending OLI/CLI and some having the ability to carry out digit manipulation on this string.

The OLI/CLI string received must be sufficient to route the call back to the originator and Meridian 1 can insert a digit(s) into this string in overlay 16 (INST) if the full routing digits are not received.

The example opposite shows Meridian 1 connected to an existing network using location code working.

The OLI/CLI within the existing network is 60X+XXX and the receiving node knows that it is a network call so that it inserts the network access digit.

The OLI/CLI from node C to node M is 603+XXX, which will not be enough to route the call back from node M to C. The insert digit on the route makes the OLI/CLI 7+603+XXX, which meets the dialing plan of node M.

Calling from node M into the network, the digit 7 is built into the steering code but we do not want to send an OLI/CLI string to include that digit as it would not match that being sent around the existing network. The LSC as programmed in LD 15 would be set to only the location code of node M.

If the DDSL monitor is being used, then the insert digit would not be seen on the incoming side.

Digit Conversion Interactions

If any combination of digit conversion occurs in a network call, then the schematic opposite can be followed for sequencing the possible conversions.

Local Steering Code Modification

All the previously described use of local steering codes allow only the deletion of digits although a software package exists which will allow both deletion and insertion (package LSCM).

If this package is equipped then both options will be given as shown in the NTP for overlay 87.

This feature has also been used in the field to give a form of pretranslation on locally dialled digits that are not actually used for networking.

Network Attendant Service

Network Attendant Service (NAS) gives the ability to have attendants at any one or more sites within a Network, with routing control and also signaling to control attendant features.

Only network attendant basic routing is supported over DPNSS1 links.

Network Attendant Routing

Routing will take place under one of the following situations:

- 1 The PABX is in night service and a NAS key is operated on the attendant console.
- 2 The PABX is in day service with the NAS key operated and the call waiting threshold has been reached (CW lamp flashing).
- 3 No attendant exists on the PABX and programming is set for permanent NAS.

Once one of the above criteria is met, any call destined for the attendant is then routed according to the programming of the NAS routing tables.

Each attendant or attendant group in the network is given a unique identity number which must not conflict with any other dialled digits. These numbers are then programmed in the Network translation tables and use route list blocks and digit manipulation to point them towards the distant attendants.

If the distant attendant is only one hop away, then it is only necessary to send 0.

If the distant attendant is via another node then the unique ID should be sent to the transit node.

The above two rules should be followed as it is not desirable to use the normal network numbering plans which may have alternate routing set.

Each remote attendant should be accessed by only one possible route.

Routing Control

When a call arrives that is to be routed to a remote attendant, the following procedure takes place:

- 1 Refer to a Time of Day schedule which will give up to four choices of remote attendant.
- 2 Attempt to route the call to the first choice remote attendant.
- 3 If not possible try the next choice(s).
- 4 If no choices available route to local Night service.

Up to seven different remote attendants can be programmed although only four can be selected at any one time of day.

Routing to a next choice attendant will only be attempted if the route is congested or down.

In other cases the call will either:

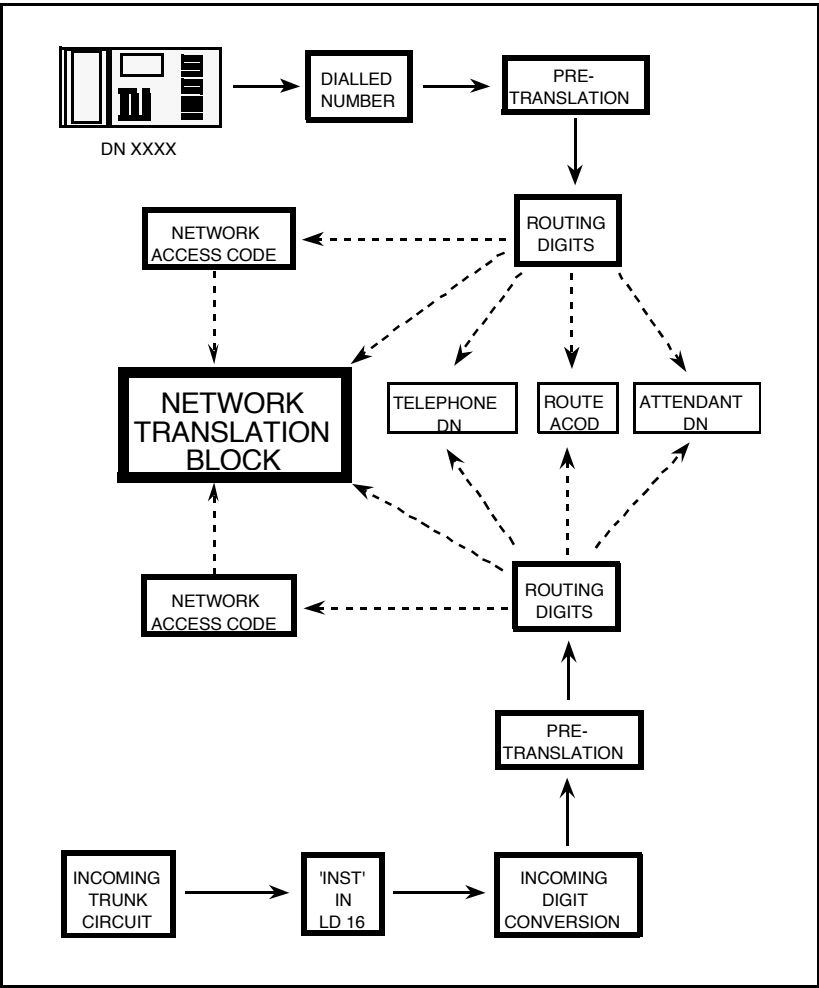
- 1 Join the queue for the attendant choice.
- 2 Route to the night service DN for the attendant choice.
- 3 Follow NAS routing for the attendant choice.

A queuing option does exist on busy routes but it is not envisaged to use this in the UK.

Signaling

Each D channel must be programmed for NAS signaling which sends all relevant control data plus the originating call type enabling the option of appearance on different ICI keys.

Figure 140
Meridian 1/SL-1 Call Routing Schematic



Configuring Network Attendant Service (NAS)

The NARS/CDP networking routing database must be configured first before the NAS feature can be programmed.

Step 1: Use LD 15 to configure control of NAS routing.

Prompt	Response	Description
REQ	CHG	
CUST	XX	
ICI	xx NCO xx NTIE xx NDID	Network COT calls Network TIE calls Network DID calls
NAS ATCL	(NO) YES	(Disallow) allow network attendant control
NAS ACTV	(NO) YES	(Disable) enable NAS routing

Step 2: Use LD 17 to configure NAS signaling.

Prompt	Response	Description
REQ	CHG	
TYPE	CFN	
ISDN	YES	
DCHI	xx	DCHI port number
NASA	YES,(NO)	NAS signaling allowed (disallowed)

Step 3: Use LD 12 to configure Attendant NAS key.

Prompt	Response	Description
REQ	CHG	
TYPE	2250	
TN	XX	
KEY	XX NAS	Network Attn service key

Step 4: Use LD 86 to define NAS routing and time schedule.

Prompt	Response	Description
REQ	NEW/ CHG	
CUST	XX	
FEAT	NAS	Network Attn service
TBL	0-63	Table 0 is for the customer, other tables are for Tenant service NAS
ALT	X	1-7 Attendant alternative no.
ID	XXX	Remote attendant access code. This code is programmed in either CDP or NARS to control routing
ALT	<CR>	Carriage to stop ALT
TODS	0	Default catch-all period
ALST	n n n n	Attn alternative list (up to 4 ALT per schedule). If no entry then local Attendant is used
TODS	1-31	
PER	hh mm hh mm	start and stop time, minute must be 0 or 30 start time must be less than stop time
DAYS	D,-- D	up to 7 days (1-7), 1=mon, 7= Sunday
ALST	n n n n	Attn alternative list (up to 4 ALT per schedule).If no entry then local Attendant is used
DBK	y/n y/n y/n y/n	Drop back to originating node if congestion is received from the above Attn alternative list
QUE	y/n y/n y/n y/n	Queuing to a route for the above Attn alternative list
TODS	1-31	Repeated

Configuring DID to Network Digit Conversion

Step	Overlay	Action
1	LD 49	Configure Conversion table
2	LD 16	Allow Digit conversion

Step 1: Use LD 49 to configure Digit Conversion Tables.

Prompt	Response	Description
REQ	NEW, CHG	
TYPE	IDC	
CUST	XX	
.		
DCNO	XXX	(0- 254), Table No.
IDGT	nnnn xxxxxxxx nnnn xxxxxxxx nnnn xxxxxxxx	Note: nnnn= Up to 4 Incoming digits xxxxxxx= Up to 8 digit local orNetwork DN

Step 2: Use LD 16 to allow Digit Conversion for Route.

Prompt	Response	Description
REQ	NEW, CHG	
TYPE	RDB	
CUST	XX	
ROUT	XXX	
.		
IDC	YES,(NO)	Allow (disallow) Digit conversion
DCNO	XXX	(0- 254), Table No.

Q.931 Drop Back On Busy

The Drop back on Busy feature gives originator control over alternate routing of a call that is being set up over one or more links in a Q931 network and will be used mainly to cover the situation of congestion on a PSTN breakout route, although the feature can be applied to any situation.

When the feature is used, the call will drop back to the originating node, where further routing choices can be made.

To use this feature, NAS signaling must be working on the relevant D channels.

Against any entry in a Route list block, two choices are available as to how congestion will be treated at any point in the network:

- 1 Try every choice available before dropping back to originator.

Answer prompt IDBB with DBA (default)

- 2 Try only a specified number of entries in a route list block before dropping back to the originator. The number of entries is referred to as the Initial set (ISET).

Answer prompt IDBB with DBI

Whichever of the above two choices is made, the relevant information is contained in the call setup message and will not be modified at any tandem node.

To achieve point 2 above, the prompt ISET in a route list block is set with a value between 1 and 32 meaning that dropback will occur if congestion exists after trying that number of entries in the route list block.

If ISET is set to 0 then no entries in that route list block will be tried and drop back will occur immediately.

Appendix B — Basic and Network Alternate Route Selection (BARS/NARS)

Introduction

This appendix describes the Basic Alternate Route Selection (BARS) and Network Alternate Route Selection (NARS) features for X11 software on an SL-1 system. In Canada, BARS/NARS is considered one package; BARS is not offered as a standalone package in Canada. For more information on the BARS/NARS package, contact a Northern Telecom representative.

The BARS or NARS feature enables a customer with a number of switches in different locations to "Tie together" these switches to create a private telecommunications network (see Figure 141). When implemented, the BARS or NARS feature functions to direct a call from a switch in one geographical location to a switch in any other geographical location in a cost-efficient and easy-to-use manner by:

- eliminating long, complex dialing plans and replacing them with an abbreviated Uniform dialing plan (UDP) common to all switches which are part of the network
- providing a means of controlling the number and type of trunks that are available to each network caller, and a method of controlling the time of day that access to a trunk (or group of trunks) is allowed
- selecting automatically the least-cost trunk route available to complete a call between network switches
- providing uniform network access to stations served directly at an SL-1 node (see Note) and stations served at SL-1 mains or conventional mains (see note) connected to an SL-1 node by Tie trunks

- providing the call originator with the option to either accept or refuse call completion over an expensive trunk, if less expensive trunks are not currently available
- providing optional queuing features which enable a call originator (when all trunks are busy) to either remain off-hook until a trunk becomes idle, or hang-up and receive a callback from the SL-1 when a trunk becomes idle.

Switch types

Terminology used in this publication is based on the following definition of switch types:

SL-1 ESN Node

An SL-1 equipped with the NARS or BARS feature.

SL-1 ESN Main

An SL-1 connected by Tie trunks to an SL-1 node and equipped with the Network Signaling (NSIG) feature package. (The connected SL-1 Node must also be equipped with the NSIG feature package.)

Conventional Main

An SL-1 switch connected by Tie trunks to an SL-1 node, but not equipped with the Network Signaling feature package. Conventional Main also applies to any other switch type (for example, step-by-step) connected by Tie trunks to an SL-1 node.

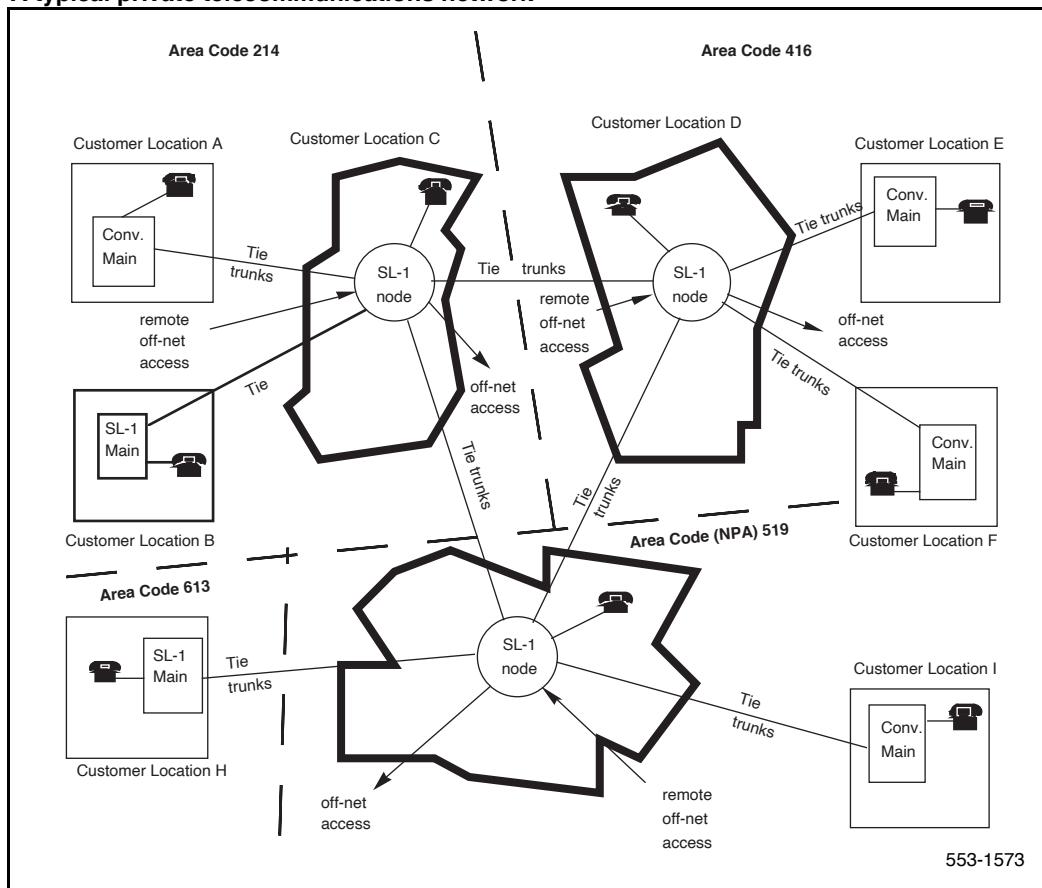
Whether the BARS or NARS feature is used depends on the number and placement of switches within the customer's private network, as well as the volume of traffic the network is to carry. NARS has the ability to translate location codes while BARS translates codes used for on-net dialing as special numbers. Refer to the section on on-net to off-net automatic overflow.

When used in large widely-dispersed applications, NARS forms an integral part of Northern Telecom's Electronic Switched Network (ESN) product. For more complete details, refer to *Electronic Switched Network description* (309-3001-100).

Other optional features that enhance the capabilities of BARS or NARS (Off-Hook Queuing, Network Authorization Codes, for example) are mentioned briefly in this publication. The following publications describe these optional features:

- *Network Queuing description* (552-2751-101)
- *Coordinated Dialing Plan description* (553-2751-102)
- *Basic and Network Authorization Code description* (553-2751-103)

Figure 141
A typical private telecommunications network



Network class of service

Network Class of Service (NCOS) is an integral part of the BARS and NARS features at an SL-1 node, and of the Network Signaling feature at an SL-1 ESN Main.

NCOS provides the means to control:

- which trunk routes are eligible to be accessed to attempt call completion
- whether or not queuing is offered to a call originator
- whether or not the call originator receives a warning tone when an expensive trunk is selected to complete a call
- whether or not the user is allowed to access the Network Speed Call feature.

Table 70 summarizes the NCOS and other parameters for SL-1 nodes and SL-1 mains.

Once each NCOS group is defined through SL-1 service change, then line, trunk, and attendant groups are assigned to the NCOS group which best serves its requirements. The NCOS group to which each line, trunk, or attendant group is assigned is independent of the assigned SL-1 class of service. Tie trunks incoming from SL-1 mains or conventional mains are also assigned to an NCOS group (at the node) which determines their level of access to the network facilities at the node.

Table 70
Summary of node and Main parameters

Parameter (Note 2)	BARS node	NARS node	ESN main
NCOS Groups (Note 3)	0 - 7 (0-99)	0 - 15 (0-99)	0 - 15 (0-99)
Facility Restriction Levels	0 - 7	0 - 7	0 - 7
Digit Manipulation Tables	1 - 255	1 - 255	—
Route Lists	0 - 127	0 - 255	—

Table 70
Summary of node and Main parameters

Parameter (Note 2)	BARS node	NARS node	ESN main
Route List Entries	0 - 7 (0-31)	0 - 7 (0-31)	
FCAS Tables (Note 1)	1 - 127	1 - 255	--
SDR Tables (Note 1)	0 - 255	0 - 511	—
SDRR ⁵⁺ Tables (Note 1)	0 - 255	0 - 511	—
TOD Schedules (Note 1)	0 - 7	0 - 7	—
Note 3: The values in parentheses () are for X11 release 13 and later.			
Note 4: FCAS = Free Calling Area Screening SDR = Supplemental Digit Restriction SDRR = Supplemental Digit Restriction/Recognition TOD = Time-Of-Day			
Note 5: If the NARS and BARS features are equipped in the same switch but for different customers, the highest parameter values apply to that switch. For example, if one customer has NARS and another customer has BARS, the NARS parameters apply to the BARS customer.			
Note 6: If the New Flexible Code Restriction (NFCR) feature is equipped in conjunction with BARS or CDP, the number of available NCOSs is 8. With X11 release 13 and later, this number is expanded to 100. NFCR is described in <i>X11 features and services (553-3001-305)</i> .			

Facility Restriction Level

Included as part of each NCOS group is a Facility Restriction Level (FRL) number which ranges from 0 (low-privilege) to 7 (high-privilege). The FRL is used by the SL-1 software to determine the alternate route selection choices available for network call attempts by users within an NCOS group.

Example

A user assigned to an NCOS group with an FRL of 3 would be allowed access only to alternate route selection choices that have an assigned FRL of 3 or less; access to route choices with an FRL greater than 3 would be denied. Thus, by assigning low-privilege users to an NCOS group with a low FRL, and high-privilege users to an NCOS group with a higher FRL, the customer can control worker/management access to all network facilities.

Expensive Route Warning Tone

In some instances, expensive trunk routes may be assigned an FRL that would enable them to be accessed by certain network users. When this occurs, the originator of the call may be sent an optional Expensive Route Warning Tone (ERWT). (ERWT eligibility is defined through the user's assigned NCOS group.) NCOS groups must have CBQ type A in order to hear ERWT.

The ERWT alerts the user that expensive facilities have been selected to complete the call. The caller may opt to either accept or reject call completion over the expensive facilities. The caller must make this choice within a customer-programmable time. Refer to SL-1 service change programs in the *X11 input/output guide* (553-3001-400), specifically LD86 the ERDT timer, for more information.

Queuing features

Eligibility for the Off-Hook Queuing (OHQ) or Call-Back Queuing (CBQ) features is defined through the user's assigned NCOS group. See *Network Queuing description* (553-2751-101).

Basic/Network Alternate Route Selection

The NARS and BARS features provide comprehensive and flexible networking packages that can be configured to satisfy the specific requirements of a customer's private network. The BARS or NARS packages provide benefits to users through simplified dialing plans and reduced communications costs. Prime elements of the BARS or NARS features are:

- Simple network access codes
- Uniform dialing plan
- Dialing transparency

- Automatic least-cost routing
- Time-of-Day (TOD) routing
- Network controls through Network Class of Service (NCOS), Traveling Class of Service (TCOS), FRL, and regular SL-1 COS
- Routing control through NCOS alterations based on a special TOD schedule
- 1-4 digit translation and 1-7 digit restriction (X11 release 4)
- 1-4 digit translation, 1-7 digit restriction, and 1-7 digit recognition (X11 release 5)
- 1-11 digit translation, restriction, recognition (X11 release 8)
- Free Calling Area Screening (FCAS)
- Expensive Route Warning Tone (ERWT)

Uniform dialing plan

The Uniform dialing plan (UDP) is not a feature of BARS or NARS, but is an operation that enables users at an SL-1 node, SL-1 main, or conventional main to dial all calls in a uniform manner regardless of the location of the calling party or the route which the call will take. (UDP now applies to BARS on X11 release 5 using off-net number recognition and calling HLOC and SPN or NXX codes.) LOCs are programmed as NXXs or SPNs.

UDP for on-net calling

An on-net call is one which terminates at a customer-owned location. To reach any on-net location, the user dials the NARS or BARS on-net access code (AC1), followed by seven digits. The dialing format for this call would be:

AC1 * LOC + XXXX (For NARS)

AC1 * SPN + XXXX (For BARS)

Legend:

AC1 = the 1-digit or 2-digit NARS on-net access code
 = NARS dial tone (optional)

LOC = a 3-digit location code assigned for the destination location

SPN = a 3-digit code assigned for the destination location

XXXX = the extension number of the party to be reached at the destination location.

Each switch which is part of the BARS or NARS network is referenced by a unique three-digit location (LOC) SPN or NXX code assigned at the SL-1 Node. There must be no conflict between the location code number assigned for a switch and Number Plan Area (NPA) codes.

A customer-owned location can be either physically connected to the network (by means of private trunk facilities) or virtually connected to the network (by means of public facilities). If a location is virtually connected to the network, the dialed LOC code is translated and converted by the SL-1 Node into the public number for the virtual location; for example, the Direct Distance Dialing (DDD) number or the Direct Inward Dialing (DID) number. Using digit manipulation for BARS, an SPN can be changed into the public network LDN.

Figure 142 illustrates a private network with a typical UDP. A user at LOC 776 (customer location I) wishing to call extension number 3283 at LOC 777 (customer location H) would first dial 8 (AC1), pause for NARS dial tone (optional), then dial 777-3283. A user at any other customer location that is part of the network would dial these same numbers to reach extension 3283.

UDP for off-net calling

An off-net call is one that does not terminate at a customer-owned location, even though some on-net facilities may be used to complete a portion of the call routing. Referring to Figure 142, a call would be termed off-net if a user at LOC 776 called a station number associated with CO 758-XXXX in the foreign area code 214.

Tables 71 and 72 list the dialing formats for the various types of NARS and BARS UDP calls respectively.

Figure 142
Example of a network with a typical NARS Uniform dialing plan

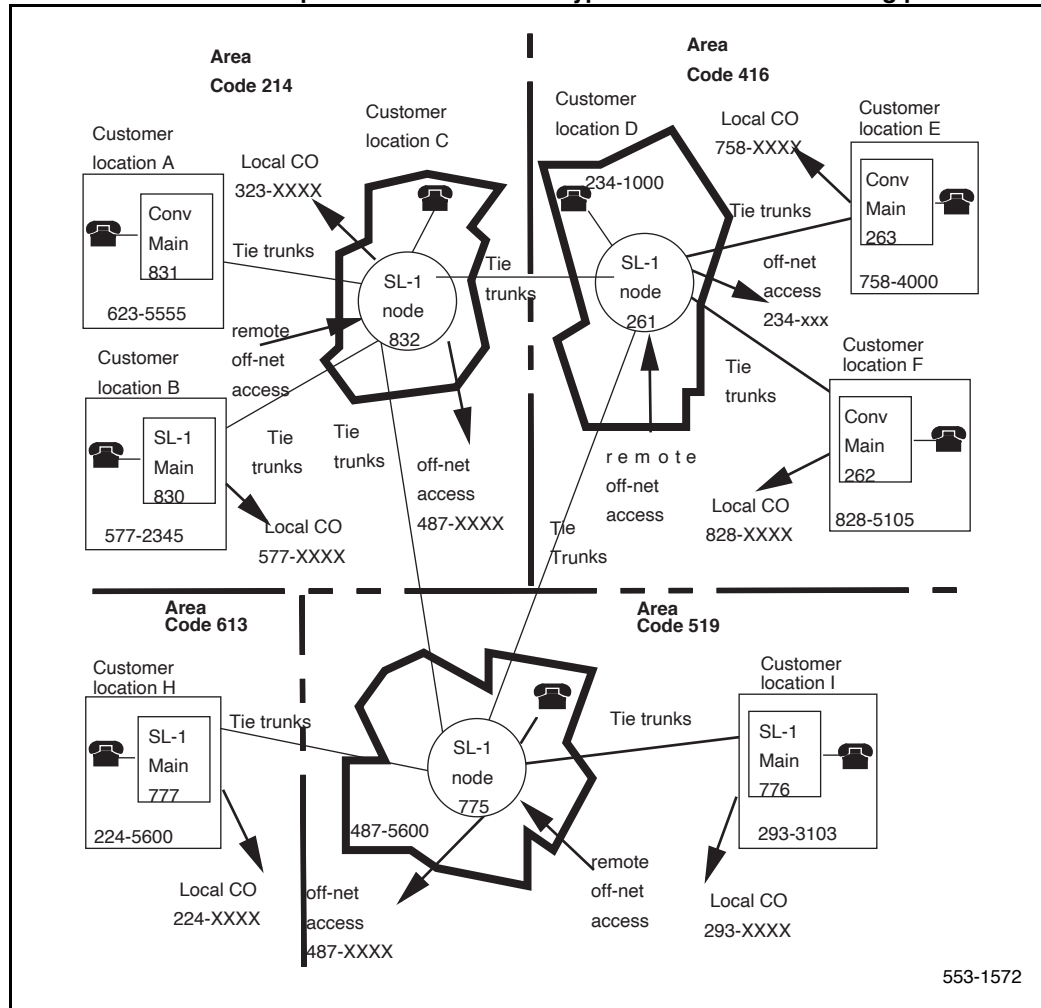


Table 71
Dialing formats for NARS UDP calls

Call type	Dialing format	Code capacity
On-Net (Notes 1, 2, and 3)	AC1 * LOC + XXXX	640
DDD FNPA (Note 1)	AC1 * 1 + NPA + NXX + XXXX	160
Network Speed Call	AC1 * LA + LN	1 - nn
Operator-assisted DDD	AC1 * 0 + NPA + NXX + XXXX	160
International DDD	AC1 * 011 + CC + NN	99
Operator-assisted International DDD	AC1 * 01 + CC + NN	99
DDD HNPA (Note 1)	AC1 or AC2 * 1 + NXX + XXXX	1
DDD Operator	AC2 * 0	1
Local Calls (Note 1)	AC2 * NXX + XXXX	640
Special Local Services	AC2 * SPN	8
Toll-free Calls	AC2 * 800 + NXX + XXXX	1
Toll-free Calls (Note 1)	AC2 * 1 + 800 + NXX + XXXX	1
Toll Special Numbers	AC2 * 900 + NXX + XXXX	1
Toll Special Numbers (Note 1)	AC2 * 1 + 900 + NXX + XXXX	1
Note 1: If 1+ Dialing is used, the On-Net and Local Calls code capacities are increased to 800 and 792 respectively.		
Note 2: If the code 1XX is reserved for future 1+ Dialing use, and not for Network Speed Call codes, then the location code capacity will be reduced to 639 if a three-digit Network Speed Call (NSC) code is used, 632 if a two-digit NSC code is used, or 560 if a one-digit NSC code is used.		
Note 3: When 1+ Dialing is used, Network Speed Call access will be in the form of 2XX-9XX as a subset of the location codes utilized in the UDP. The location code capacity will be reduced to 799 if a three-digit NSC code is used, 792 if a two-digit NSC code is used, or 720 if a one-digit NSC code is used.		
Note 4: See following page for legend of acronyms and dialing format abbreviations.		

Table 71
Dialing formats for NARS UDP calls

Call type	Dialing format	Code capacity
Legend		Legend
AC1	Access code for on-net, long distance and Network Speed Calls. Typically the digit "8" but can be either one or two digits in length.	AC1
AC2	Access code for local calls. Typically the digit "9" but can be either one or two digits in length.	AC2
*	Symbol meaning NARS dial tone (optional).	*
NPA	Numbering Plan Area (NPA) code. Any number of the form N0X or N1X.	NPA
HNPA	Home Numbering Plan Area (HNPA) code. Any number of the form N0X or N1X.	HNPA
FNPA	Foreign Numbering Plan Area (FNPA) code. Any number of the form N0X or N1X.	FNPA

Table 71
Dialing formats for NARS UDP calls

Call type	Dialing format	Code capacity
CC	Country code. Any one, two, or three digits from 2 to 9.	CC
NN	National Number. Depends on national dialing plan; maximum 12 digits including the Country Code.	NN
N	Any of the digits from 2 to 9.	N
X	Any of the digits from 0 to 9.	X
LA	List access code. Any one, two, or three digits from 0 to 9.	LA
LN	List element number. Any one, two, or three digits from 0 to 9 and up to a maximum of 1000 element numbers.	LN
LOC	Three-digit location code for each UDP network location.	LOC
NXX	Local Exchange Code.	NXX
XXXX	Four-digit directory (extension) number.	XXXX
SPN	Special numbers for example, 411, 611 or may be XXXX.	SPN

Table 72
Dialing formats for BARS calls

Call type	Dialing format
On-Net	(1) AC1 * SPN + XXXX (2) AC1 * NXX + XXXX
DDD FNPA	AC1 * 1 + NPA + NXX + XXXX
Operator-assisted DDD	AC1 * 0 + NPA + NXX + XXXX
International DDD	AC1 * 011 + CC + NN
Operator-assisted International DDD	AC1 * 01 + CC + NN
DDD HNPA (if applicable)	(1) AC1 * 1 + NXX + XXXX (2) AC1 * 1 + NPA + NXX-XXXX (see note)
DDD Operator	AC1 * 0
Local Calls	AC1 * NXX + XXXX
Special Local Services	AC1 * 411, 611 etc.
Toll-free Calls	AC1 * 1 + 800 or 1 + 900 + NXX + XXXX
Note 1: 1 + dialing is optional; you can simply enter 1 NPA.	
Legend	

Table 72
Dialing formats for BARS calls

Call type	Dialing format
AC1	Access code for on-net, long distance and Network Speed Calls. Typically the digit "9" but can be either one or two digits in length.
*	Symbol meaning wait for BARS dial tone (optional).
NPA	Numbering Plan Area (NPA) code. Any number of the form N0X or N1X.
HNPA	Home Numbering Plan Area (HNPA) code. Any number of the form N0X or N1X.
FNPA	Foreign Numbering Plan Area (FNPA) code. Any number of the form N0X or N1X.
CC	Country code. Any one, two, or three digits from 2 to 9.
SPN	Special Number. Code used to identify other office locations on the network, or a number such as 411 or 611.
NN	National Number. Depends on national dialing plan; maximum 12 digits including the Country Code.
N	Any of the digits from 2 to 9.
X	Any of the digits from 0 to 9.

BARS access code

Long distance calls and calls to distant company locations ("on-network" calls) from the SL-1 Node are made by dialing the BARS access code (AC1) followed by the desired number. The BARS access code (AC1) is typically the digit 8, but is a customer defined one or two-digit number.

There can be no conflict between the BARS access code and any other part of the dialing plan at the node. Dial tone may or may not be provided to a caller after the BARS access code is dialed at the option of the customer.

Dialing the BARS AC1 triggers the BARS software to perform the call-processing and routing required for call completion. This is accomplished by means of a network translation table associated with the BARS access code. Normal SL-1 translation is used for all other call type; local or off-network calls made by dialing the digit 9.

NARS access codes

To access NARS, the user dials either one of the two customer defined NARS access codes: AC1 or AC2. These access codes are typically 8 for on-net and long distance calls (AC1); and 9 for off-net and local calls (AC2). However, any one-digit or two-digit access codes can be used, provided that AC1 is different from AC2 and there is no conflict with any other part of the dialing plan. Dial tone may or may not be provided to a caller after either NARS access code is dialed, at the option of the customer.

Dialing either of the NARS access codes triggers the NARS software to perform the call-processing and routing required for call completion. This is accomplished by means of network translation tables. There is a network translation table associated with each NARS access code. This translation mechanism is used to implement the NARS Uniform dialing plan for private networks.

Network translation

The normal SL-1 digit translator reads the dialed network access code, determines if the call is to be processed by BARS or NARS, and selects the appropriate Network Translation (see Table 73). BARS or NARS translation determines the method to be used to process the call, refers to Supplementary Digit Restriction and Recognition tables if required, and applies digit restriction or recognition where it is specified. The result of translation is to invoke either route selection with a specified route list, standard call blocking, queuing, or internal recognition.

Any SL-1 switch running a version of X11 software prior to X11 release 8 will translate the first 3 digits after the access code—first 4 digits with 1+ dialing or 4-digit SPN codes. (SPNs can be 1, 2, 3, or 4 digits.)

Only one Route List Index is used for routing NPA/NXX/SPN/LOC codes. In a network with multiple switches sharing DID numbers within the same NXX, Coordinated Dialing Plan is needed to route the calls properly.

11-digit BARS or NARS translation

With 11-digit translation introduced in X11 release 8, the ESN BARS or NARS translation capabilities are expanded from four digits to a maximum of 11 digits for route selection. This means more digits are translated and therefore make similar codes translate differently. This increases the flexibility for network routing, including International calls. For example, NXX1363 becomes 13634, or 13635. More than one code can be handled by the same route list.

By allowing translation of more than four leading digits, unique unconflicted 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, as many Route Lists as are needed to eliminate code conflict or achieve network requirements can be defined by extending translation deeper into the dialed code. Table 73 compares the number of digits that can be translated prior to X11 release 8 with present capabilities.

Table 73
Digit translation

Type	Before X11 release 8	X11 release 8 and later
LOC	3	3 - 7
HLOC	3	3 - 7
NPA	3- 4	3 - 11
HNPA	3- 4	3 - 11
NXX	3- 4	3 - 8
SPN	1- 4	1- 11

Dialing transparency

Extending network access to an SL-1 Main or Conventional Main is accomplished by forming a single Tie trunk access group from the Main to the Node. Users at Main switches access the trunk group to the Node by dialing the BARS or NARS on-net access code (AC1). The SL-1 Node is arranged to insert the digit(s) for AC1 on each incoming call from the Main, thus enabling access to the network facilities of the Node in a transparent fashion. Local calling is arranged through conventional dial "9" CO trunks at the Main.

If an SL-1 node replaces a tandem switch in a Tandem Tie Trunk Network (TTTN), other tandem switches in the network can "tandem through" the SL-1 Node using the same access codes as before. This requires that there be no conflicts between the access codes for the TTTN trunks and the dialing plan implemented at the SL-1 Node.

Automatic least-cost routing

For each network call translated at an SL-1 node, BARS or NARS selects a route from up to eight outgoing alternate routes to complete the call. A list of alternate routes to a particular destination is called a route list and each route specified in the list is termed an entry. Any combination of trunks; for example, CO, FX, or Tie can be specified in a route list. BARS or NARS can select a route from a list of up to 32 outgoing alternate routes.

Use of the Bandwidth Controller (DCA System 9000) improves the capacity of T1 based tandem networks. Its dynamic alternate routing capability can independently choose the optimum path for a voice or data call. However, it will be necessary to coordinate the Bandwidth Controller routing lists with the ESN routing lists.

Typically, the first entries (routes) in a route list are the less expensive routes to a destination and comprise the initial set (I set) of routes in the list. The remaining routes in the list (if any) are the more expensive routes to a destination and comprise the extended set of routes in the list. An initial set marker, defined through service change, determines which routes comprise the initial route set. Refer to *Electronic Switched Network description* (309-3001-100) for more information on I set and extended set routes.

An SL-1 Node equipped with NARS can accommodate 256 (0-255) route lists; an SL-1 node equipped with BARS can accommodate 128 (0-127) route lists. Associated with each entry in a route list is information relevant to:

- the route number (0-511) (depending on the release)
- the minimum FRL required for access
- the time-of-day the route can be accessed
- whether or not queuing (OHQ or CBQ) is allowed on the route
- whether or not the route is to receive ERWT treatment
- a digit manipulation table index number
- a Free Calling Area Screening (FCAS) table index number
- activation/deactivation of conversion from an on-net number to an off-net number (NARS only)

Route eligibility

BARS or NARS translates the number dialed (1-11 digits) after an access code into a route list, and searches sequentially the routes in the list for an available route. Route eligibility for a given call is based on the caller's NCOS, the NCOS-defined FRL, the current time of day, and SL-1 COS.

Because each entry in a route list has a minimum FRL required for access and all network users are assigned an FRL through their NCOS, the network communications manager can restrict the type of calls allowed to particular users. For example, if the minimum FRL for all calls is 1, except for special local services numbers which are assigned an FRL of 0, then a user assigned to an NCOS group with an FRL of 0 would only be able to make calls to the special numbers. In addition, the communications manager can restrict the use of high-cost facilities by assigning a high FRL to the expensive routes in a route list and a lower FRL to a user's NCOS.

Digit manipulation

As mentioned previously, any trunk type can be specified in a route list. However, when certain trunk types are accessed, the digits dialed by the user must be manipulated to conform to the dialing requirements of the trunk. To do this, BARS or NARS uses digit manipulation tables to modify the dialed digits. There can be a maximum of 256 digit manipulation tables, each referenced by a digit manipulation index number, defined at each SL-1 Node. Digit manipulation can delete up to 15 leading digits, and insert up to 20 leading digits.

Example

A user at customer location I (Figure 142) dials 8-613-596-9084 to reach an off-net station in the 613 NPA associated with customer location H. At the SL-1 Node, BARS or NARS selects the appropriate route list for call completion to NPA 613 and finds that the only available route to that NPA is a local CO trunk which requires the insertion of the leading digit "1" for long distance calls. The route list entry for this route specifies a digit manipulation index number (0-255). "0" means no digit manipulation is required. BARS or NARS references the digit manipulation table indicated by the index number, deletes digits as specified in the table (none in this case), and inserts the required digits ("1" in this case), and completes the call on this route.

Time-of-Day routing

BARS or NARS provides for up to eight (0-7) Time-of-Day (TOD) schedules. Each entry (route) in a route list is assigned to the TOD schedule which specifies the hour(s) that the particular entry can be accessed. Thus, based on the current time of day, the most cost-effective route alternatives can be specified. A typical TOD schedule is shown in Table 74.

Table 74
A typical TOD schedule

TOD schedule	Time period
2	00:00 to 07:44 17:30 to 23:59
1	07:45 to 08:59 12:00 to 13:14 16:00 to 17:29
0	09:00 to 11:59 13:15 to 15:59
Note 1: A TOD schedule can be associated with any number of arbitrarily selected 15-minute periods. However, any one 15-minute period can appear only in one TOD schedule.	
Note 2: Normally, the user does not define 0, but lets the SL-1 define it by calculating times of day not mentioned in schedules 1-7.	

Based on the TOD schedule shown in Table 74, a route list entry assigned to TOD schedule 2 would be accessed only between the hours of 00:00 to 07:44 and 17:30 to 23:59. Access to the route at any other time would be denied. TOD schedules can be turned "on" or "off" through SL-1 service change, as traffic conditions warrant. A TOD schedule is turned on for an entry by turning off all other TOD. An X beside the schedule number turns that number off.

Flexible ESN "0" Routing

Flexible ESN "0" Routing (an X11 release 16 feature) allows routing of calls on different routes based on a few predefined non-leftwise unique dialing sequences. This means that each entry in the table cannot match the leftmost 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 list) to be entered into the ESN translation table:

- 0
- 00
- 01
- 011

Flexible ESN “0” Routing is part of the existing BARS (57) and NARS (58) packages and has no interaction with other features besides these. Since NARS has two translation tables, two Flexible ESN “0” Routing datablocks will be included in NARS. This means that a call could be configured to route in two different ways.

This feature is applicable to all route types and network types that are supported by ESN. For information on the appropriate prompts and responses in service change (LD90), refer to Northern Telecom Publication *X11 input/output guide* (553-3001-400).

Automatic on-net to off-net overflow

If all on-net facilities to a location are busy or blocked, NARS can convert a dialed NARS UDP number to the Listed Directory Number (LDN) or DID number of the destination location, and use off-net facilities to complete the call. If the SL-1 Node is equipped with BARS, digit manipulation tables can be used to convert the dialed number to the off-net number (LDN or DID) of the destination location.

Example

A user at customer location I (see Figure 143) dials 8-777-3283 to reach a party with extension number 3283 at customer location H. At the SL-1 Node, NARS translates the dialed LOC number (777) into a route list, and searches all eligible routes in the list. Failing to find an available Tie trunk route, NARS then seizes local off-net facilities and, to complete the call, outputs one of two possibilities:

- 224-3283, if customer location H is arranged for DID
- 224-5600, if customer location H is not arranged for DID

Requirements

Prior to X11 release 5, only one contiguous DID DN range can be defined per location. DNs which lie outside the range are converted to the LDN.

The capability of this feature has been enhanced in X11 release 5 software by "Multiple DID Office Code Screening" and in X11 release 8 by 11-digit translation.

Multiple DID Office Code Screening (X11 release 5+)

Multiple DID Office Code Screening is an enhancement to the On-Net to Off-Net Overflow capability of the NARS feature. This enhancement permits on-net calls which are routed through the public network using on-net to off-net conversion, to terminate at any DN which has been defined in the Location Code data block of memory. For each LOC defined, Multiple DID Office Code Screening will:

- allow the definition of multiple NXX codes,
- allow the definition of multiple ranges of DN within each NXX.

The following arrangements of multiple office codes (NXX) and multiple DN ranges are possible:

- single office code with a single DN range (the only alternative prior to generic X11 release 5)
- single office code with multiple DN ranges
- multiple office codes with a single DN range
- multiple office codes with multiple DN ranges

Requirements

- Prior to X11 release 5, only one NPA per LOC code is allowed.
- Prior to X11 release 5, ranges defined within a LOC code must be unique. Overlapping or duplication of ranges is not permitted.
- The number of digits in each DID range must be 4.
- A maximum of 20 DID ranges may be defined per location code regardless of the number of office codes.

Incoming trunk group exclusion (X11 release 5+)

Incoming Trunk Group Exclusion is an enhancement to the BARS or NARS feature which blocks calls from SL-1 and Conventional Main users who use the network to reach destinations in the home NPA, or other restricted NPAs, NXXs, LOCs and SPNs. When the feature is configured, users cannot use the network to circumvent the restrictions. Instead, they are forced to dial off-net from their own switch and become subject to whatever restrictions are imposed at the Main.

Standard call blocking is applied on outgoing calls to specific NPAs, NXXs, SPNs or LOCs at the SL-1 Node if the call is from a specific incoming trunk group. Two advantages result.

- Loopback routing through the caller's home switch (home NPA, NXX) is prevented. Calls which should have been made off-net from the caller's home switch are blocked outgoing at the node.
- Main users are prevented from using BARS or NARS to make calls to certain NPA, NXX, SPN or LOC that they are restricted from making at the home switch.

Sets of restricted trunk routes to specific NPAs, NXXs, SPNs or LOCs are defined in SL-1 Service Change. There is one Incoming Trunk Group Exclusion Index (maximum 255) for each defined NPA, NXX, SPN or LOC. Each index points to an Incoming Trunk Group Exclusion (ITGE) table. A maximum of 128 restricted routes can be defined in each ITGE table. Incoming Trunk Group Exclusion provides full 10-digit restriction for NPA and SPN codes, seven-digit restriction for NXX codes and three-digit restriction for LOC codes. The code itself may be restricted also.

When a call is received, BARS or NARS tests to see if the dialed code is a restricted type (Supplemental Digit Restriction). If it is, BARS or NARS checks whether or not it has an ITGE restriction and if there is an ITEI number (1-255) associated with it. If an ITEI is defined, the ITGE table corresponding to the dialed code is searched. If the incoming trunk route is a member of the ITGE, the BARS or NARS process is terminated and the call is blocked. Otherwise, call processing continues.

Off-net number recognition (X11 release 5+)

Off-Net Number Recognition eliminates the need for using two extra CO trunks, when a subscriber, using the private network, dials a DID or DDD number that terminates at a BARS or NARS location. Calls are routed directly to the dialed DN (DID calls) or to the LDN (DDD calls), rather than being switched from the terminating switch to the CO and back again.

Off-net number recognition parameters for local and remote DDD and DID locations are defined by the customer in the Network Translation tables and Supplemental Digit Recognition/Restriction blocks (SDRR). Recognition of up to 10 digits can be defined for SL-1 with X11 release 5 or 7 software as follows:

Table 75
SDRR (X11 release 5 or 7)

Call type	Network translation table (number of digits)	SDRR block (number of digits)
NPA	3	1 - 7
1NPA	4	1 - 7
NXX	3	1 - 4
1NXX	4	1 - 4
SPN	4	1 - 6
ISPN	4	1 - 7

With 11-Digit Translation, available on X11 release 8 and later, up to 11 digits can be defined in the following table.

Table 76
DRR (X11, release 8 and up)

Call type	Network translation table (number of digits)	SDRR block (number of digits)
NPA	3 - 10	1 to (10 - N)
1NPA	4 - 11	1 to (11 - N)
NXX	3 - 7	1 to (7 - N)
1NXX	4 - 8	1 to (8 - N)
SPN	1 - 11	1 to (10 - N)
ISPN	1 - 4	1 to (11 - N)
Note 1: The value N is equal to the number of digits defined in the Network Translation table.		

Up to 512 SDRR blocks can be defined for NARS (256 for BARS). Each table can contain up to 64 entries.

Off-Net numbers are recognized at the last intelligent BARS or NARS switch. Translation of the NPA, NXX or SPN identifies the method of treatment for the call. If the data type is SDRR and the index is an SDRR table index, supplemental digit recognition/restriction is applied by comparing the dialed digits with the numbers declared in the SDRR block.

- If no match is found in the SDRR, route selection is called, call processing resumes and the call is routed to the CO of the terminating off-net number.
- If a match is found and the number is in the "denied" block, standard call blocking takes place.

- If a match is found and the number is recognized as a terminating number at the local switch (for example, the last intelligent BARS or NARS switch), the call is terminated at the station DN (DID calls) or at the attendant DN (DDD calls).
- If a match is found and the dialed number is a recognized number terminating at a remote switch, route selection is called, the special digit manipulation takes place and the call is routed directly to the main. DID calls terminate at the dialed station and DDD calls terminate at the attendant DN. Remote recognition applies to Tie trunks only.

Supplemental digit restriction (release dependent)

Supplemental digit restriction blocks (see Table 77) function as follows:

- block (deny) access to certain telephone numbers
- recognize Off-Net calls dialed to On-Net locations
- prevent routing of calls to the home switch of the originating trunk group by either on-net or off-net facilities.

The customer can also specify through SL-1 service change the treatment that blocked calls receive; for example, overflow tone, intercept to attendant, or recorded announcement.

Table 77
Supplemental digit restriction blocks per NARS or BARS

Network package	X11 release 4 and earlier maximum	X11 release 5 and later maximum
NARS	256	512
BARS	32	256

Note 1: For X11 release 4 and earlier, one block can restrict up to 16 numbers. For X11 release 5 and later software, one block can recognize or restrict up to 64 numbers.

Digit translation / restriction / recognition

BARS and NARS provide a 1-digit through 10-digit (11-digit with 1+ dialing) translation/restriction/recognition capability through the use of network translation tables and Supplemental Restriction/Recognition tables. There are two network translation tables with NARS; one associated with each of the network access codes (AC1 and AC2).

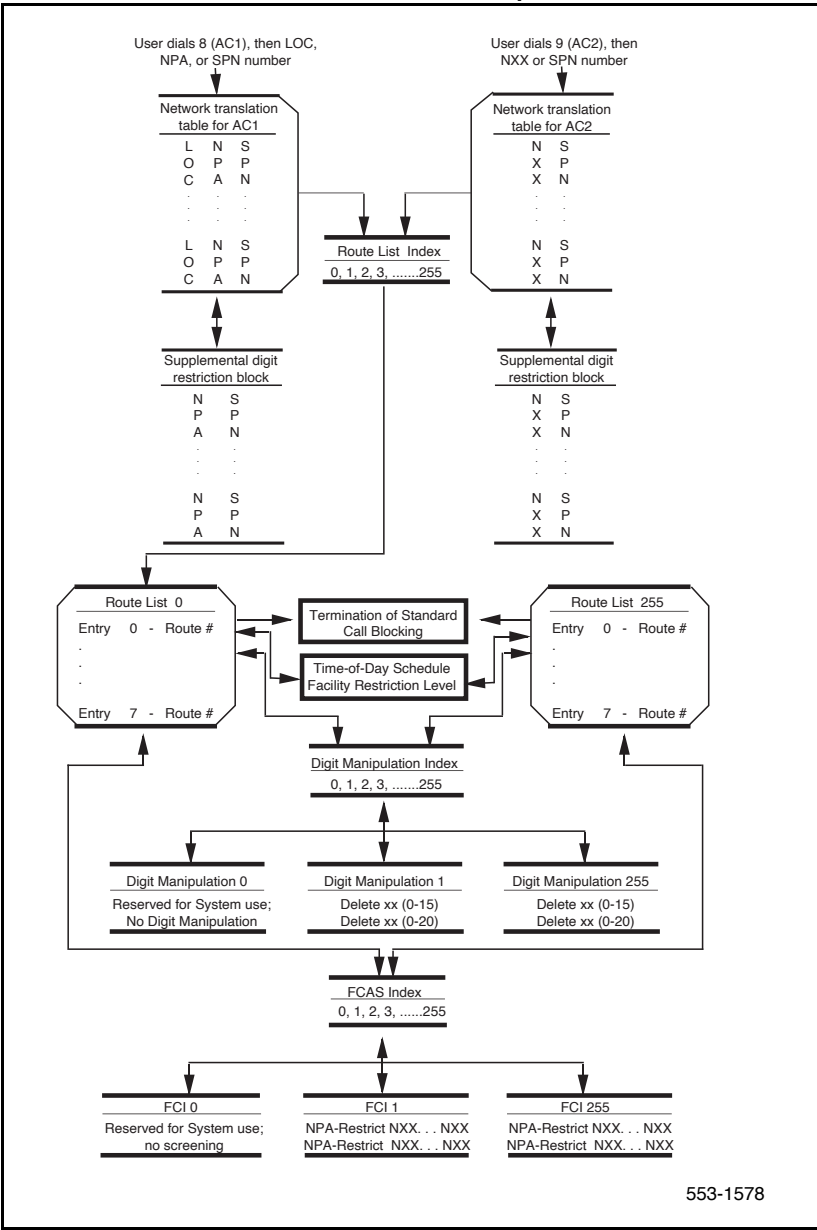
Information contained in the network translation tables is as follows:

- For each NPA entry, excluding the Home NPA (HNPA):
 - a route list index number (0-255 with NARS, 0-127 with BARS) that indicates which route list to use in processing a call to this NPA entry
 - whether or not there are telephone numbers within this NPA entry to which network calls are to be blocked; for example, “denied”
 - whether or not there are telephone numbers within this NPA entry to which network calls are to be blocked because of ITGE restrictions (X11 release 5+)
 - whether or not there are numbers under this NPA entry that are to be recognized as DID or DDD codes to an On-Net location (X11 release 5+)
 - a list (up to 64) of 1- to 7-digit numbers that follow the NPA and are to be blocked or recognized in this NPA.

- For each NXX entry:
 - a route list index number (0-255 with NARS; 0-127 with BARS) that indicates which route list to access in processing a call to this NXX entry
 - whether or not there are telephone numbers within this NXX entry to which network calls are to be blocked
 - whether or not there are telephone numbers within this NXX entry to which network calls are to be blocked because of ITGE restrictions (X11 release 5+)
 - whether or not there are numbers under this NXX entry that are to be recognized as DID or DDD codes to an On-Net location (X11 release 5+)
 - a list of up to 64, 1- to 4-digit numbers that follow the NXX and are to be blocked or recognized in this NXX
- For each LOC entry, excluding the Home LOC code (not applicable to a BARS-equipped switch):
 - a route list index number (0-255) that indicates which route list to access in processing a call to this LOC entry
 - the Listed Directory Number (LDN) to which the LOC entry is to be converted when using off-net DDD facilities
 - the range of DID numbers to which the LOC entry can be converted when using DID facilities
 - whether or not there are LOC entries to which network calls are to be blocked because of ITGE restrictions (X11 release 5+)

- For each SPN (special number) entry:
 - a route list index number (0-255 with NARS; 0-127 with BARS) that indicates which route list to access in processing a call to this SPN
 - whether or not there are digits following SPN numbers to which network calls are to be blocked
 - whether or not there are SPN numbers to which network calls are to be blocked because of ITGE restrictions (X11 release 5+)
 - whether or not there are numbers under this SPN that are to be recognized as DID or DDD codes to an On-Net location (X11 release 5+)
 - a list of up to 64, 1 to 7 numbers that are to be blocked or recognized when following the SPN

Figure 143
NARS elements accessed at an SL-1 node to process a network call



Free Calling Area Screening

Free Calling Area Screening (FCAS) is a BARS or NARS feature which provides the customer with the capability of full six-digit (NPA-NXX) screening to determine the route choice for completion of off-net calls. With FCAS, a customer can allow calls to NXX codes within the "free calling area" surrounding a particular on-net location, and restrict (deny) calls to those NXX codes that would incur long distance charges.

FCAS is implemented in a way similar to the method used for digit manipulation (for example, through FCAS tables). There can be up to 255 FCAS tables defined at an SL-1 node equipped with NARS (BARS can accommodate 127 FCAS tables). Each table can contain up to 15 NPA codes (only 7 NPA codes with BARS). Up to 800 NXX codes can be restricted or allowed within each NPA code. Each FCAS table is referenced by a Free Calling Index (FCI) number (0-255 with NARS, 0-127 with BARS); "FCI = 0" is a system default meaning no Free Calling Area Screening is required. The appropriate FCI number is then assigned to the applicable route list entries.

Whenever a route list entry is being considered for an off-net call (for example, 8-NPA-NXX-XXXX), BARS or NARS checks to see if there is an FCI number (other than "0") referred to by the entry. If an FCI number other than "0" is defined, the appropriate FCAS table for the dialed NPA is found and used for NXX screening. If the dialed NXX is denied in the table, BARS or NARS will not use the route list entry for call completion, but will continue to search for another eligible route list entry. If the dialed NXX is not denied, or specifically allowed in the table, the route list entry is eligible for the call. Calls to the LDN of a location are screened only if the NPA is included as part of the LDN. NXXs allowed in an FCI table are the only ones allowed for that route list entry.

Expensive Route Warning Tone

This feature enables the network communications manager to select certain users to receive an Expensive Route Warning Tone (ERWT). Eligibility for this tone is based on the user's NCOS. The tone (three 256-ms bursts of 440 Hz) notifies the user that BARS or NARS has selected facilities designated as expensive to complete the call. The user then has the choice of either allowing the call to complete over the expensive facilities, or going on-hook to avoid the increased expense, or queuing on the I set routes. (The user must make this choice within a programmable time of 0 to 10 seconds.) The tones must be activated for the customer group and the expensive route cannot be part of the I set in the route list.

If the call originator is located at an SL-1 node or SL-1 Main and the Ring Again feature is defined for the user and the user is CBQ(a) eligible, then Ring Again may be activated to queue the call. See *Network Queuing description* (553-2751-101).

If the SL-1 Node is equipped for Call Detail Recording (CDR), acceptance of an expensive route after ERWT is received is noted in the CDR record.

BARS or NARS bypass control

A customer can allow selected users to bypass the BARS or NARS feature for call completion between any two locations; for example, two locations which share a high community of interest. To do this, routes and trunks are set up between the two locations, and assigned an access code distinct from the AC1 and AC2 codes used to access BARS or NARS. The normal SL-1 trunk controls, like Trunk Group Access Restriction (TGAR), class of service, and code restriction, are then used to enable access only to the selected users. All other users are denied access to the trunk group, and thus, are forced to use BARS or NARS for all calls.

Network Speed Call

The Network Speed Call (NSC) feature enables a user at an SL-1 node who is normally restricted from making certain types of BARS or NARS calls to make such a call if the destination is a company-approved number defined in a System Speed Call (SSC) list. This feature requires that the System Speed Call feature (see *X11 features and services* [553-3001-305]) be equipped, in addition to Network Speed Call. NSC can also be accessed by users at an SL-1 Main or Conventional Main, provided a BARS or NARS access code is used to initiate the call.

Access to the NSC feature is allowed after a NARS or BARS access code is dialed. Upon receipt of BARS or NARS dial tone (optional), the user dials a Network Speed Call access code (one to three digits). The NSC access code must be unique from all LOC, NPA and NXX codes, and special numbers defined in the translator for the BARS or NARS access code.

The NSC access code is associated with a previously defined System Speed Call list (0-253) through service change in the network translation load. If the SCC list has its length (size) changed, the list access code and list number must be deleted and reentered into the NARS translator. Associated with the SSC list is an NCOS number. The NCOS assigned to the SSC list is applied to the call only if the FRL (0-7) is greater than that associated with the call originator's assigned NCOS.

With X11 release 13 and later, the number of SSC lists has expanded from 254 (0-253) to 4096 (0-4095), depending on available system memory and other speed call usage.

If 1+ Dialing is specified for an NPA, NXX or SPN number in a translator, the digit "1" must not be used as the leading digit for Network Speed Call list codes in that translator.

The user then dials the number of the desired entry (1-1000) in the SSC list. Upon completion of dialing, the digits defined for the list entry are passed to BARS or NARS translation for processing. Route and feature (OHQ, CBQ) eligibility for call completion are based on the NCOS associated with the SSC list, if the FRL of the user's assigned NCOS is lower than that of the list.

Network Call Transfer

This feature improves the operation of the existing Call Transfer (XFER) feature between two SL-1s when a call is transferred back to the originating switch. The regular XFER feature requires two Tie trunks to complete the call. With Network Call Transfer (NXFER), if the call is transferred back to the originating switch by means of the same Tie trunk group, the originating switch completes the transfer within itself and the Tie trunks are dropped.

Both SL-1 switches must be equipped with NSIG and NXFER software for this feature to operate.

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 of NXFER is the same as for the existing XFER feature.

As shown in Figure 144, the NXFER feature allows station A at one ESN switch (I) to transfer the Tie trunk call from station B (switch II) to a third party, station C (switch II). In addition to NXFER software, NSIG software is needed at both ends of Tie trunk. If the transfer is allowed, stations B and C are connected on switch II and the Tie trunks are dropped (see Figure 146). In comparison, regular SL-1 Call Transfer (XFER) requires two Tie trunks and both switches to transfer and connect stations B and C.

Figure 144
Connection during NXFER

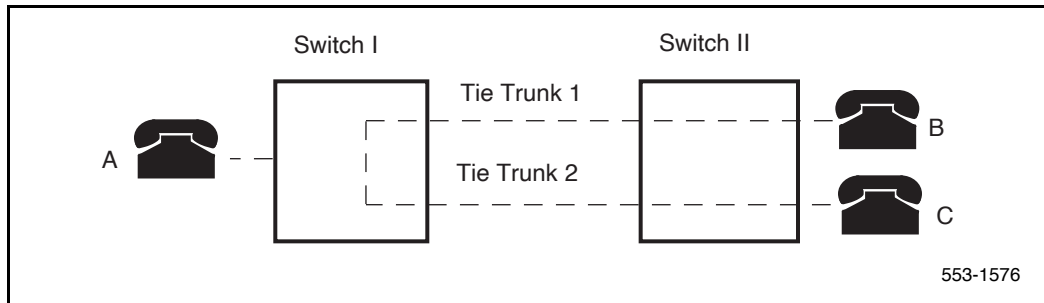


Figure 145
Connection after NXFER

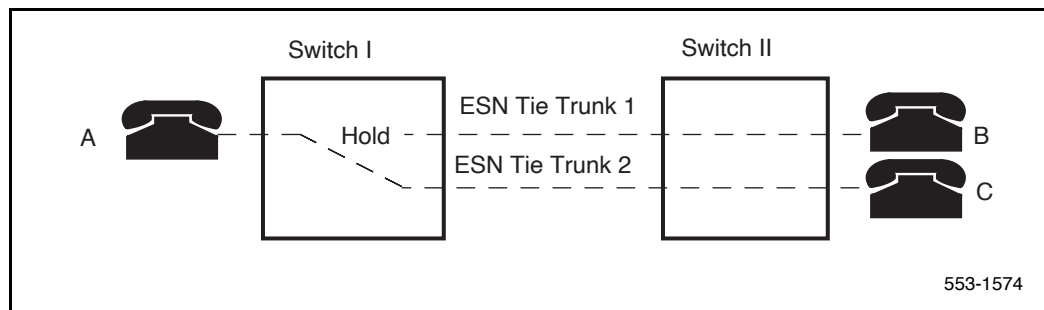
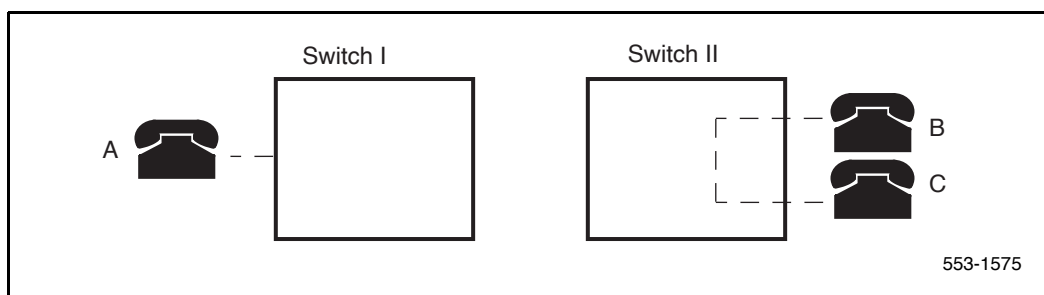


Figure 146
Connection after XFER



1 + dialing

Translation tables With 1+ dialing, the BARS or NARS translation tables are equipped for 4-digit translation (based on the first 1, 2, 3, or 4 digits), or 11-digit translation with X11 release 8 on, thus allowing BARS or NARS access code for long distance calls. NARS provides two translation tables and BARS provides one. Refer to Table 71 for dialing formats for NARS Uniform dialing plan (UDP) calls and Table 72 for BARS calls.

Code ambiguity The 1+ dialing capability also eliminates ambiguity between identical originating and terminating 3-digit NPA, NXX and LOC codes for BARS or NARS-originated calls. Thus, the BARS or NARS customer can route calls to any NPA, NXX or LOC code that conflicts with one of his 3-digit codes.

Limitation If 1+ dialing is specified for an NPA, NXX or SPN number in a translator, the digit "1" must not be used as the leading digit for Network Speed Call list codes in that translator.

Network control

Network control is an enhancement to the Network Class of Service (NCOS) feature that extends NCOS controls to users located at an SL-1 Main. Network control requires that the SL-1 Main and serving SL-1 Node be equipped with the NSIG feature. The SL-1 Main must also be equipped with the NCOS feature.

SL-1 Main NCOS

Users (lines, trunks, attendants) at an SL-1 Main are assigned an NCOS that is used to determine their level of access to network facilities at the serving SL-1 Node. When a user at an SL-1 Main initiates a call to (or through) an SL-1 node, the user's assigned NCOS or TCOS, depending on Tie trunk settings (ESN or ESN2, for example), can be transmitted.

If the user's NCOS is transmitted to the serving node and the node is equipped with BARS, only NCOS 0 through 7 can be assigned at the node. Therefore, only NCOS 0 through 7 should be assigned at the ESN Main even though an ESN Main equipped with NSIG can have an NCOS of 0 through 15. If the node is equipped with NARS, then an NCOS of 0 through 15 can be assigned both at the node and at the ESN Main.

The transmitted NCOS or TCOS overrides the NCOS or FRL assigned to the incoming Tie trunk group at the node, and is used to determine the user's eligibility for network resources/features at the SL-1 Node. Thus, a user at an SL-1 Main has the same network-access capabilities as a user at the SL-1 Node who is assigned the same NCOS.

If the user at the SL-1 Main enters a valid authorization code prior to placing a BARS or NARS call, the NCOS associated with the authorization code is transmitted to the SL-1 Node in place of the user's assigned NCOS.

Calls from a Conventional Main to the SL-1 Node are controlled by the NCOS assigned to the incoming trunk group at the SL-1 Node, since the Conventional Main has no NSIG.

SL-1 node TCOS

Network Control at an SL-1 node can provide a Traveling Class of Service (TCOS) mechanism. TCOS controls route access and Off-Hook Queuing (OHQ) eligibility for calls placed to (or through) another SL-1 node or an associated SL-1 Main. It also enables the SL-1 Node to interface with switches that are part of an Electronic Tie Network (ETN) as long as the SIGO setting on the Tie trunk route is set for ETN at both ends. Nodes can send NCOS if SIGO is set for ESN2, for example, at both ends of the Tie trunk group.

The Traveling Class of Service is, in effect, the Facility Restrictions Level (FRL) of a user's assigned NCOS. When a user at an SL-1 node initiates a call to another SL-1 Node (or an SL-1 ESN Main), the TCOS (for example, the FRL of the user's assigned NCOS) is transmitted to the other SL-1 Node. At the receiving SL-1 Node, the TCOS (0-7) replaces the FRL of the NCOS assigned to the incoming trunk group. Route access and OHQ eligibility for the call are, therefore, based on the NCOS of the incoming trunk group with the modified FRL (for example, TCOS).

The Network Control (NCTL) data block (LD87) is used to define OHQ eligibility on a per FRL (TCOS) basis (for example, if FRL 4 is defined as OHQ eligible, then all users with an NCOS with an FRL of 4 are eligible for OHQ on calls placed to another SL-1 Node or to an associated SL-1 Main).

If a user at an SL-1 Main or conventional main initiates a call that tandems through the serving SL-1 node to another SL-1 node or SL-1 Main, TCOS applies to the call as if the call originated at the serving SL-1 node.

Compatibility with ETN switches

The SL-1 TCOS is equivalent to the Traveling Class Mark (TCM) used at ETN switches. (See Technical Publication 42709, "Tie Trunk Signaling Compatibility for Connecting to a DIMENSION PBX", July 1979.)

When a seven-digit/ten-digit UDP call or a DSC (Distant Steering Code) (CDP) call is made from an SL-1 node to an ETN switch, the dialed digits together with the TCOS number (0-7) are sent to the connected ETN switch. At the ETN switch, the TCOS number received from the SL-1 Node is used as a TCM to determine route access and off-hook queuing eligibility at the ETN switch.

Similarly, when a call is made from an ETN switch to an SL-1 node, the dialed digits together with the TCM number (0-7) are sent to the connected SL-1 node. The SL-1 node interprets the received TCM number as a TCOS number. The received TCM (for example, TCOS) replaces the FRL of the NCOS assigned to the incoming trunk group from the ETN switch. This new FRL (for example, TCM) is then used to determine route access and off-hook queuing eligibility for the call. However, if a DSC (CDP) call is terminated on an SL-1 switch as a Local Steering Code (LSC) call, the TCOS value transmitted by the connected switch will not be collected and saved by this switch.

Network signaling

The Network Signaling (NSIG) feature provides the required signaling protocol to interface SL-1 nodes with SL-1 ESN Mains, SL-1 nodes with other SL-1 nodes, SL-1 nodes with conventional mains, and SL-1 nodes with Electronic Tie Network (ETN) switches.

When equipped with NSIG, an SL-1 Conventional Main is enhanced and becomes an ESN Main. When callers at an ESN Main place calls through a node or nodes with NSIG, their NCOS or TCOS travel with the call and are interpreted at other SL-1 switches that are equipped with NSIG. The Tie trunk settings determine and control the operation of this feature.

When the NSIG feature is equipped at a switch, options are available (route data block, LD16) to define the signaling arrangements between that switch and any other switch that may be connected to it by means of Tie trunks. These options define what call information is to be transmitted to a connected switch and what call information is to be received from a connected switch. The option selected depends on the type of connected switch (node, main, conventional main, ETN) and the options (for example, CCBQ, CBQCM) that are available to the connected switch.

The signaling options are: STD (standard), ESN, ESN2, ESN3, ESN5 (Electronic Switched Network), and ETN (Electronic Tie Network).

STD Arranges the Tie trunk group for transmission/reception of the called number between switches. Sends outpulsed digits.

ESN (X11 release 2 only) Arranges the Tie trunk group for transmission/reception of the call type, NCOS/TCOS, and called number between switches.

ESN2 (X11 release 3 and later) Arranges the Tie group as described for ESN in X11 release 2. Used unless switch has NXFER or Satellite Link Control (SAT).

ESN3 (X11 release 3 or 4) Arranges the Tie group as described for ESN in X11 release 2 and is required on systems equipped with the Network Call Transfer (NXFER) or Satellite Link Control features.

ESN5 (X11 release 5 and later) Arranges the Tie trunk group as described for ESN in X11 release 2; needed with DTI.

ETN Arranges the Tie trunk group for transmission/reception of the called number and TCOS/TCM between switches and is used when connected to an ETN switch. Sends outpulsed digits and TCOS.

Application

Following is a description of how these options would be applied to accommodate the different switch types that can be connected to an SL-1 Main or SL-1 Node that is equipped with the NSIG feature.

SL-1 Node An SL-1 Node can be connected by means of Tie trunks to: another SL-1 Node, an SL-1 Main, a Conventional Main, and/or an ETN Switch.

- If the SL-1 Node connects to another SL-1 Node, both ends of the connecting Tie trunk group are defined with the ETN option (called number plus TCOS/TCM), ESN2, ESN3, or ESN5.
- If the SL-1 Node connects to an SL-1 Main, both ends of the connecting Tie trunk group are defined with the ESN option (call type plus NCOS/TCOS plus called number).
- If the SL-1 Node connects to a Conventional Main, the node-end of the Tie trunk group is defined with the STD option (called number).
- If the SL-1 Node connects to an ETN switch, the node-end of the Tie trunk group is defined with the ETN option (called number plus TCOS-TCM).

SL-1 ESN Main An SL-1 Main can be connected by means of Tie trunks to an SL-1 node and satellite switches.

- For connection to an SL-1 node, both ends of the connecting Tie trunk group are defined with the ESN option (call type plus NCOS/TCOS plus called number).
- If there are satellite switches connected to the SL-1 Main, the main-end of the Tie trunk groups from the satellite switches are defined with the STD option (called number).

Requirements

The following requirements apply:

- As SL-1 Main can connect to only one SL-1 node, both switches must be equipped with the NSIG feature for NSIG-related features.
- Tie trunks between SL-1 nodes and SL-1 mains must be arranged for DTMF sending/receiving and wink-start operation.
- SL-1 Node compatibility with ETN switches is limited to seven-digit on-network, ten-digit off-network and DSC (CDP) calls.

Satellite Link Control

Tandem trunk calls, when connected through more than one communications satellite trunk, are subject to transmission distortion due to propagation to and from communications satellites. The Satellite Link Control feature ensures that the configuration of a call does not include more than one communications satellite trunk.

Requirements

This feature applies to ESN network calls (BARS or NARS/CDP) only.

- ESN Proprietary Signaling (NSIG) is required among ESN switches.
- Routes which receive digits from satellites or send digits to satellites have to be marked as SATELLITE routes for this feature to operate.

Example

When accessing an SCC, the SL-1TD is instructed to look for one of the following tones:

- Busy or overflow tone
- SCC dial tone or regular dial tone
- Ringback tone

Simultaneously a timer is started whose value is an approximation of the period in which ringback tone is normally received. When the timer expires prior to a response from the SL-1TD, a status request message is sent to the SL-1TD by software. The response from the SL-1TD indicates that a tone has been detected but is not yet identified. Based on the sequence in which tones are checked, the software assumes the tone under investigation is ringback tone and can continue the SCC call processing.

Routing control

The Routing Control feature provides a mechanism for changing a user's network-access capabilities when a special TOD schedule is in effect, or when an extended TOD is in effect, or when the user presses a Routing Control key on the console.

NCOS map

With the NARS or BARS feature, TOD schedule 7 is the special TOD schedule. Associated with the special TOD schedule is a Network Class of Service (NCOS) map. The NCOS map lists all NCOS numbers. Associated with each listed NCOS is an alternate NCOS number (greater than, equal to, or smaller than) that replaces the original NCOS number when the special TOD schedule is in effect. Table 78 illustrates a typical NCOS map.

Table 78
A typical NCOS map for special TOD schedule 7

Original NCOS	Alternate NCOS (Note)	Original NCOS	Alternate NCOS (Note)
0	0	8	2
1	0	9	3
2	0	10	3
3	1	11	4
4	1	12	4
5	2	13	5
6	2	14	5
7	2	15	5

When TOD schedule 7 is in effect, the alternate NCOS replaces the user's original NCOS.

Invoking Routing Control

The alternate NCOS numbers associated with special TOD schedule 7 are normally invoked when the time specified for TOD schedule 7 corresponds to the time in the SL-1 system clock. Additionally, the alternate NCOS numbers can be scheduled for implementation (through SL-1 service change) for the full 24-hour period of specified days of the week. This capability enables network-access capabilities to be changed automatically on weekends or company holidays.

The SL-1 attendant can also manually invoke the special TOD schedule through use of a routing control (RTC) key on the console. Pressing the RTC key lights the associated lamp, and invokes the special TOD schedule. To deactivate routing control, the RTC key is pressed again. The associated lamp goes dark and normal TOD schedules are once again in effect.

Authorization code can be used to override the restrictions imposed through routing control. If a user enters a valid Authorization code (AUTH) , the NCOS number associated with the AUTH is applied for the duration of the call.

Network traffic measurements

The Network Traffic (NTRF) feature provides traffic measurement data related to network performance and network traffic at each SL-1 Node and SL-1 Main. Effective use of this data enables the network communications manager to assess the effectiveness of the network, and to identify specific areas of network operation where improvements are warranted.

Areas of network operation that are measured include NCOS, OHQ, CBQ, and routing. These measurements are provided in addition to existing non-network traffic data in *Traffic measurement formats and output* (553-2001-450).

TFN001 routing measurements

The routing measurements provide data related to route list utilization. A route list is a list of alternate trunk routes which are identified through interpretation of the dialed number. For each defined route list, these measurements show:

- how often the list was used

- which routes in the list were used
- the number of calls that were unsuccessful in completing a route list selection or connection
- queuing (OHQ and CBQ) information

These data comprise both the usage and average duration of each call against the route list. Data is output only for route lists upon which calls were attempted during the traffic study interval.

OHQ measurements

The OHQ measurements are associated with route lists, NCOS, and incoming trunk groups. OHQ enables a user to wait off-hook for a network facility to become available. The network communications manager can control the conditions for, and the duration of, the wait. The measurements present both the usage and average duration if the OHQ feature. This data comprises part of the routing measurements.

CBQ measurements

The CBQ measurements are associated with route lists, NCOS, and incoming trunk groups. If the system cannot select a route, the caller may opt to have the system establish the call at a later time when a route becomes available. The measurements indicate usage and the length of time a caller waits for an established call. The communications manager can control the conditions for CBQ and the advancement of a call within the queue. This data comprises part of the routing measurements.

TFN002 NCOS measurements

The network class of service (assigned through service change) indicates the network facilities available to the user. Traffic measurements are accumulated for each NCOS and indicate the grade of service (queuing and blocking delay, for example) provided by the system. The communications manager can change the NCOS if a grade of service is inappropriate for a user category, the definition of the NCOS is inaccurate, or the routing parameters need updating.

TFN003 incoming trunk group measurements

These measurements relate to the incremental traffic that was imposed on the incoming trunk groups by network queuing features (for example, OHQ, CBQ, CCBQ, CBQCM).

The measurements show:

- how often the incoming trunk group was offered OHQ
- the average time in the OHQ
- how often the incoming trunk group was offered CCBQ or CBQCM, and the number of acceptances
- the average time in CCBQ or CBQCM
- how often an access line was blocked when attempting a CCBQ or CBQCM callback
- how often CCBQ or CBQCM callbacks were not answered or were canceled

BARS implementation

This section provides the procedures necessary to configure Basic Alternate Route Selection (BARS). Only the service change information for BARS is shown here. For a complete discussion of prompts and responses, see the Option 11 *Software guide*. The following procedure shows the steps that must be performed to correctly implement your BARS system.

Procedure 1: Implementing BARS

- 1 Gather data for each NCOS group (LD87)
- 2 Gather data to define BARS feature parameters (LD86)
- 3 Gather data for each Digit Manipulation index (LD86)
- 4 Gather data for each Route list associated with a Digit Manipulation table (LD86)
- 5 Gather data for each Incoming Trunk Group Exclusion List (LD86)
- 6 Gather BARS translation data (LD90)
- 7 Gather data to configure a Conventional Main for Off-Net Number Recognition (LD16)
- 8 Gather data to assign a Network Class of Service group number to a list of items (LDs 10, 12, 14, 24, 88)
- 9 Enter data into the database

Configuring BARS

Step 1: Gather data for each NCOS group. Access LD87

REQ	NEW, CHG, OUT	Create, change, or remove data
CUST	0-99	Customer number
FEAT	NCTL	Network control
NCOS	0-7 0-99	NCOS group to be added, changed, or removed
FRL	0-7	Facility restriction level
RWTA	Yes, (No)	Expensive route warning tone

Step 2: Gather data to define BARS feature parameters. Access LD86

REQ	NEW, CHG, OUT	Create, change, or remove data
CUST	0-99	Customer number
FEAT	ESN	Electronic Switched Network data block
MXSD	0-512	Maximum number of supplemental digit restriction blocks
MXIX	0-127	Maximum number of incoming trunk group exclusion tables that can be defined
MXDM	0-256	Maximum number of digit manipulation tables
MXRL	0-128	Maximum number of route lists
AC1	xx	One- or two-digit BARS access code one
DLTN	No, (Yes)	Dial tone after dialing AC1 or AC2
ERWT	No, (Yes)	Expensive route warning tone is not (is) provided

Step 2: Gather data to define BARS feature parameters. Access LD86

REQ	NEW, CHG, OUT	Create, change, or remove data
ERDT	0-(6)-10	Time in seconds that a user has to accept or reject an expensive route after ERWT is given; default is 6 s; range is in 2-second intervals
TODS	x hh mm hh mm	Start and end time-of-day schedules; BARS or NARS = 0-7 x = schedule number hh = hour mm = minutes
TGAR	Yes, (No)	Check for Trunk Group Access Restrictions Yes = examine TGAR/TARG when call is placed No = ignore TGAR/TARG when call is placed

Step 3: Gather data for each Digit Manipulation index (DGT). Access LD86

REQ	NEW, CHG, OUT	Create, change, or remove data
CUST	0-99	Customer number
FEAT	DGT	Digit manipulation index
DMI	1-255	Digit manipulation table index (see note below)
DEL	(0)-15	Number of leading digits to be deleted from the dialed number; default is 0
INST	x...x	Up to 24 leading digits to be inserted
Note 1: The maximum number of Digit manipulation tables is defined by prompt MXDM		

Step 4: Gather data for each Route List associated with A DGT . Access LD86

REQ	NEW, CHG, OUT	Create, change, or remove data
CUST	0-99	Customer number
FEAT	RLB	Route list data block
RLI	0-127	Route list index number
ENTR	0-7 0-99	Route list entry number
RTNO	0-511	Route number associated with the index
TDET	Yes, (No)	Tone detector is (is not) used

Step 4: Gather data for each Route List associated with A DGT .
Access LD86

REQ	NEW, CHG, OUT	Create, change, or remove data
TYPE	TIE, (CC1), CC2	TIE = Tone detector for on-net calls (CC1) = Tone detector for Special Common Carrier, (SCC) Type 1 CC2 = Tone detector for Special Common Carrier, (SCC) Type 2
TONE	DIAL, (SCC)	Type of tone expected from SCC DIAL = normal dial tone SCC = SCC dial tone
TOD	0-7 X0-7	Time-of-Day schedule associated with the entry Turn off a time-of-day schedule
EXP	Yes, (No)	Entry is (is not) classed as expensive
FRL	(0)-7	Minimum Facility Restriction Level a user must have to access the entry
DMI	(0)-255	Index number of the digit manipulation table to be used for the entry; default is 0, no digit manipulation required
FCI	1-127	Free calling area screening table index number (FCAS) (0) = no FCAS required
OHQ	Yes, (No)	Off Hook Queuing is (is not) allowed on the entry
CBQ	Yes, (No)	Call Back Queuing is (is not) allowed on the entry
ISSET	(0)-32	Number of entries in the initial route set
MFRL	(MIN), 0-7	Minimum FRL used to determine autocode prompting

Step 5: Gather data for each Incoming Trunk Group Exclusion Index (ITGE). Access LD86

REQ	NEW, CHG, OUT	Create, change, or remove data
CUST	0-99	Customer number
FEAT	ITGE	Incoming trunk group exclusion data block
ITEI	1-127	Incoming trunk group exclusion index number; if REQ was "OUT" all route entries defined for the entered index are removed
	<return>	Return to REQ prompt
RTNO	0-511	Route number associated with index; precede with an X to delete an existing route
	<return>	Return to REQ prompt

Step 6: Gather BARS translation data. Access LD90

REQ	NEW, CHG, OUT	Create, change, or remove data
CUST	0-99	Customer number
FEAT	NET	Network translation tables
TRAN	AC1, AC2, SUM	Access code 1, 2, or summary tables
TYPE	LOC HNPA NPA HLOC NXX SPN NXCL	Location code Home NPA translation code Number plan area translation code Home location code Central Office translation code Special Number translation code Speed call
	<return>	Return to REQ

Step 6: Gather BARS translation data. Access LD90

REQ	NEW, CHG, OUT	Create, change, or remove data
The following prompt occurs if TYPE is LOC:		
LOC	x...x	Location code (3 digits) or extended LOC (3 to 7)
RLI	0-255	Route list index
ITEI	(0)-255	Incoming trunk group exclusion index
LDN	xx...xx	Up to ten-digit listed directory number including NPA
DID	Yes, (No)	This location arranged for DID
MNXX	Yes, (No)	Multiple NXX codes and ranges
SAVE	1-4	Number of trailing digits to be saved in dialed extension number; must be 4 if MNXX = Yes
OFFC	xxx	NXX of the DID number; prompted if MNXX = Yes
RNGE	0-9999 0-9999	Bottom and top of range of DID numbers
The following prompt occurs if TYPE is HNPA:		
HNPA	xxx	Home NPA
The following prompts occur if TYPE is NPA:		
NPA	xxx, xxx yyy	Area code translation, extended NPA code translation
RLI	0-255	Route list index
SDRR	aaa	Type of supplemental restriction or recognition

Step 6: Gather BARS translation data. Access LD90

REQ	NEW, CHG, OUT	Create, change, or remove data
DMI	1-255	Digit manipulation table index
DENY	x...x	Number to be denied within the HNPA, NPA, NXX, SPN, or SDRR
LDID	x...x	Local DID number recognized within the HNPA, NPA, NXX, or SPN. The maximum number of digits allowed is 10-m (11-m for 1 + dialing).
LDDD	x...x	Local DDD number recognized within the HNPA, NPA, NXX, or SPN. The maximum number of digits allowed is 10-m (11-m for 1 + dialing).
DID	x...x	Remote DID number recognized within the HNPA, NPA, NXX, or SPN. The maximum number of digits allowed is 10-m (11-m for 1 + dialing).
DDD	x...x	Remote DDD number recognized within the HNPA, NPA, NXX, or SPN. The maximum number of digits allowed is 10-m (11-m for 1 + dialing).
ITED	x...x	Incoming trunk group exclusion codes for HNPA, NPA, NXX, or SPN. The maximum number of digits allowed is 7-m.
ITEI	x...x	Incoming trunk group exclusion index
Legend: m = the number of digits entered for the prompt NPA X = a command used to clear a recognized xxx code <return> = A return after each subprompt takes you back to SDRR		
The following prompts appear if TYPE is SPN:		

Step 6: Gather BARS translation data. Access LD90

REQ	NEW, CHG, OUT	Create, change, or remove data
SPN	xxxx xxxx x..	Special number translation
RLI	0-255	Route list index
SDRR	aaa	Type of supplemental restriction or recognition
DMI	1-255	Digit manipulation table index
DENY	x...x	Number to be denied within the HNPA, NPA, NXX, SPN, or SDRR
LDID	x...x	Local DID number recognized within the HNPA, NPA, NXX, or SPN
LDDD	x...x	Local DDD number recognized within the HNPA, NPA, NXX, or SPN
DID	x...x	Remote DID number recognized within the HNPA, NPA, NXX, or SPN
DDD	x...x	Remote DDD number recognized within the HNPA, NPA, NXX, or SPN
ITED	x...x	Incoming trunk group exclusion codes for HNPA, NPA, NXX, or SPN
ITEI	(0)-255	Incoming trunk group exclusion index
The following prompts appear if TYPE is NSCL:		
NSCC	xxx	One- to three-digit network speed call access code
SSCL	0-4095	System speed call list number

Step 7: Gather data to configure a Conventional Main for Off-Net Number Recognition. Access LD 16

REQ	NEW, CHG	Add or change a route
TYPE	RDB	Route data block
CUST	0-99	Customer number
ROUTE	0-511	Route number
TKTP	TIE	Tie trunk
.		
.		
.		
CNVT	Yes, (No)	Route to conventional switch (prompted if the response to TKTP is Tie)
DDMI	(0)-127	Digit manipulation index (prompted if the response to CNVT is Yes)
ATDN	xxxx	Attendant DN of conventional main (prompted if the response to CVNT is Yes)
Note 1: If the DN expansion package is equipped, the attendant DN can have up to 7 digits.; otherwise, only 4 digits can be entered.		

The following procedure involves several load programs. In all cases, the prompt is NCOS; the range is 0-99; the default is 0.

Step 8: Gather data to assign a Network Class of Service group number to a list of items

Gather data to assign a Network Class of Service group number to each of the following items:

- 500/2500 sets (LD10)
- SL-1 telephones, M1000 series telephones, M2000 series digital telephones, M3000 Touchphones (LD12)

- Attendant consoles (LD12)
- Trunks (LD14)
- Direct Inward System Access directory number See LD24)
- Authorization code (LD88)

Step 9: Enter data into the database

The final step in configuring Basic Alternate Route Selection is to enter the data gathered into the database:

- 1** Login.
- 2** Load the appropriate load program.
- 3** Enter data requested by prompts until REQ prompt returns.
- 4** More data to be added/removed?

Yes = go to Step 2

No = perform data dump

A data dump takes approximately 3 minutes to complete. If large amounts of data are being added, perform a data dump periodically. This ensures that entered data is not lost should a system reload occur.

- 5** Print data (respond with PRT to prompt REQ) and verify against data forms.
- 6** Corrections required?

Yes = go to Step 2

No = Refer to 553-2YY1-230 and perform applicable tests.

NARS implementation

This section provides the procedures necessary to configure Network Alternate Route Selection (NARS). Only the service change information for BARS is shown here. For a complete discussion of prompts and responses, refer to the Input/Output guide. The following procedure shows the steps that must be performed to correctly implement your NARS system.

Configuring NARS

Complete the following steps to configure and implement NARS.

- 1 Gather data for each NCOS group (LD87)
- 2 Gather data to define NARS feature parameters (LD86)
- 3 Gather data for each Digit Manipulation index (LD86)
- 4 Gather data for each Route list associated with a Digit Manipulation table (LD86)
- 5 Gather data for each Incoming Trunk Group Exclusion List (LD86)
- 6 Gather NARS translation data (LD90)
- 7 Gather data to configure a Conventional Main for Off-Net Number Recognition (LD16)
- 8 Gather data to assign a Network Class of Service group number to a list of items (LDs 10, 12, 14, 24, 88)
- 9 Enter data into the database

Configuring NARS

Step 1: Gather data for each NCOS group. Access LD87

REQ	NEW, CHG, OUT	Create, change, or remove data
CUST	0-99	Customer number
FEAT	NCTL	Network control

NCOS	0-99	NCOS group to be added, changed, or removed
FRL	(0)-7	Facility restriction level
RWTA	Yes, (No)	Expensive route warning tone

Step 2: Gather data to define NARS feature parameters. Access LD86

REQ	NEW, CHG, OUT	Create, change, or remove data
CUST	0-99	Customer number
FEAT	ESN	Electronic Switched Network data block
MXLC	0-999	Maximum number of supplemental digit restriction blocks
MXSD	0-512	Maximum number of incoming trunk group exclusion tables that can be defined
MXIX	0-256	Maximum number of incoming trunk group exclusion tables
MXDM	0-256	Maximum number of digit manipulation tables
MXRL	0-256	Maximum number of route lists
MXFC	0-256	Maximum number of free calling area screening tables
AC1	xx	One or two digit NARS/BARS access code one
AC2	xx	One or two digit NARS access code two
DLTN	No, (Yes)	NARS/BARS dial tone after dialing AC1 or AC2
ERWT	No, (Yes)	Expensive route warning tone is not (is) provided

Step 2: Gather data to define NARS feature parameters. Access LD86

REQ	NEW, CHG, OUT	Create, change, or remove data
ERDT	0-(6)-10	Time in seconds that a user has to accept or reject an expensive route after ERWT is given; default is 6 s; range is in 2-second intervals
TODS	x hh mm hh mm	Start and end time-of-day schedules; BARS or NARS = 0-7 x = schedule number hh = hour mm = minutes
TGAR	Yes, (No)	Check for Trunk Group Access Restrictions Yes = examine TGAR/TARG when call is placed No = ignore TGAR/TARG when call is placed

Step 3
Gather data for each Digit Manipulation index (DGT). Access LD86

REQ	NEW, CHG, OUT	Create, change, or remove data
CUST	0-99	Customer number
FEAT	DGT	Digit manipulation index
DMI	1-255	Digit manipulation table index (see note below)
DEL	(0)-15	Number of leading digits to be deleted from the dialed number; default is 0
INST	x...x	Up to 24 leading digits to be inserted
Note 1: The maximum number of Digit manipulation tables is defined by prompt MXDM		

Step 4: Gather data for each Route List associated with A DGT. Access LD86

REQ	NEW, CHG, OUT	Create, change, or remove data
CUST	0-99	Customer number
FEAT	RLB	Route list data block
RLI	0-255	Route list index number
ENTR	0-63	Route list entry number (release 15 +)
RTNO	0-511	Route number associated with the index
TDET	Yes, (No)	Tone detector is (is not) used
TYPE	TIE, (CC1), CC2	TIE = Tone detector for on-net calls (CC1) = Tone detector for Special Common Carrier, (SCC) Type 1 CC2 = Tone detector for Special Common Carrier, (SCC) Type 2
TONE	DIAL, (SCC)	Type of tone expected from SCC DIAL = normal dial tone SCC = SCC dial tone
TOD	0-7 X0-7	Time-of-Day schedule associated with the entry Turn off a time-of-day schedule
EXP	Yes, (No)	Entry is (is not) classed as expensive
FRL	(0)-7	Minimum Facility Restriction Level a user must have to access the entry
DMI	(0)-255	Index number of the digit manipulation table to be used for the entry; default is 0, no digit manipulation required
FCI	(0)-255	Free calling area screening table index number (FCAS) (0) = no FCAS required

Step 4: Gather data for each Route List associated with A DGT. Access LD86

REQ	NEW, CHG, OUT	Create, change, or remove data
OHQ	Yes, (No)	Off Hook Queuing is (is not) allowed on the entry
CBQ	Yes, (No)	Call Back Queuing is (is not) allowed on the entry
ISET	(0)-32	Number of entries in the initial route set
MFRL	(MIN), 0-7	Minimum FRL used to determine autocode prompting

Step 5: Gather data for each Incoming Trunk Group Exclusion Index (ITGE). Access LD86

REQ	NEW, CHG, OUT	Create, change, or remove data
CUST	0-99	Customer number
FEAT	ITGE	Incoming trunk group exclusion data block
ITEI	1-127	Incoming trunk group exclusion index number; if REQ was "OUT" all route entries defined for the entered index are removed
RTNO	0-511	Route number associated with index; precede with an X to delete an existing route

Step 6: Enter data into the database

REQ	NEW, CHG, OUT	Create, change, or remove data
CUST	0-99	Customer number
FEAT	NET	Network translation tables
TRAN	AC1, AC2, SUM	Access code 1, 2, or summary tables
TYPE	LOC HNPA NPA HLOC NXX SPN NXCL	Location code Home NPA translation code Number plan area translation code Home location code Central Office translation code Special Number translation code Speed call
	<return>	Return to REQ
The following prompt occurs if TYPE is LOC:		
LOC	x...x	Location code (3 digits) or extended LOC (3 to 7)
RLI	0-255	Route list index
ITEI	(0)-255	Incoming trunk group exclusion index
LDN	xx...xx	Up to ten-digit listed directory number including NPA
DID	Yes, (No)	This location arranged for DID
MNXX	Yes, (No)	Multiple NXX codes and ranges
SAVE	1-4	Number of trailing digits to be saved in dialed extension number; must be 4 if MNXX = Yes
OFFC	xxx	NXX of the DID number; prompted if MNXX = Yes

Step 6: Enter data into the database

REQ	NEW, CHG, OUT	Create, change, or remove data
RNGE	0-9999 0-9999	Bottom and top of range of DID numbers
The following prompt occurs if TYPE is HNPA:		
HNPA	xxx	Home NPA
The following prompts occur if TYPE is NPA:		
NPA	xxx, xxx yyy	Area code translation, extended NPA code translation
RLI	0-255	Route list index
SDRR	aaa	Type of supplemental restriction or recognition
DMI	1-255	Digit manipulation table index
DENY	x...x	Number to be denied within the HNPA, NPA, NXX, SPN, or SDRR
LDID	x...x	Local DID number recognized within the HNPA, NPA, NXX, or SPN. The maximum number of digits allowed is 10-m (11-m for 1 + dialing).
LDDD	x...x	Local DDD number recognized within the HNPA, NPA, NXX, or SPN. The maximum number of digits allowed is 10-m (11-m for 1 + dialing).
DID	x...x	Remote DID number recognized within the HNPA, NPA, NXX, or SPN. The maximum number of digits allowed is 10-m (11-m for 1 + dialing).

Step 6: Enter data into the database

REQ	NEW, CHG, OUT	Create, change, or remove data
DDD	x...x	Remote DDD number recognized within the HNPA, NPA, NXX, or SPN. The maximum number of digits allowed is 10-m (11-m for 1 + dialing).
ITED	x...x	Incoming trunk group exclusion codes for HNPA, NPA, NXX, or SPN. The maximum number of digits allowed is 7-m.
ITEI	(0)-255	Incoming trunk group exclusion index
Legend: m = the number of digits entered for the prompt NPA X = a command used to clear a recognized xxx code <return> = A return after each subprompt takes you back to SDRR		
The following prompts appear if TYPE is NXX		
NXX	xxx, xxx yyy	Office code translation, extended NXX codes translation
RLI	0-255	Route list index
SDRR	aaa	Type of supplemental restriction or recognition
DMI	1-255	Digit manipulation table index
DENY	x...x	Number to be denied within the HNPA, NPA, NXX, SPN, or SDRR. The maximum number of digits allowed is 7-m.
LDID	x...x	Local DID number recognized within the HNPA, NPA, NXX, or SPN
LDDD	x...x	Local DDD number recognized within the HNPA, NPA, NXX, or SPN

Step 6: Enter data into the database

REQ	NEW, CHG, OUT	Create, change, or remove data
DID	x...x	Remote DID number recognized within the HNPA, NPA, NXX, or SPN
DDD	x...x	Remote DDD number recognized within the HNPA, NPA, NXX, or SPN
ITED	x...x	Incoming trunk group exclusion codes for HNPA, NPA, NXX, or SPN
ITEI	(0)-255	Incoming trunk group exclusion index
The following prompts appear if TYPE is SPN:		
SPN	xxxx xxxx x..	Special number translation
RLI	0-255	Route list index
SDRR	aaa	Type of supplemental restriction or recognition
DMI	1-255	Digit manipulation table index
DENY	x...x	Number to be denied within the HNPA, NPA, NXX, SPN, or SDRR
LDID	x...x	Local DID number recognized within the HNPA, NPA, NXX, or SPN
LDDD	x...x	Local DDD number recognized within the HNPA, NPA, NXX, or SPN
DID	x...x	Remote DID number recognized within the HNPA, NPA, NXX, or SPN
DDD	x...x	Remote DDD number recognized within the HNPA, NPA, NXX, or SPN
ITED	x...x	Incoming trunk group exclusion codes for HNPA, NPA, NXX, or SPN
ITEI	(0)-255	Incoming trunk group exclusion index

Step 6: Enter data into the database

REQ	NEW, CHG, OUT	Create, change, or remove data
The following prompts appear if TYPE is NSCL:		
NSCC	xxx	One- to three-digit network speed call access code
SSCL	0-4095	System speed call list number

Step 7: Gather data to configure a Conventional Main for Off-Net Number Recognition. Access LD16

REQ	NEW, CHG	Add or change a route
TYPE	RDB	Route data block
CUST	0-99	Customer number
ROUTE	0-511	Route number
TKTP	TIE	Tie trunk
.		
.		
.		
CNVT	Yes, (No)	Route to conventional switch (prompted if the response to TKTP is Tie)

Step 7: Gather data to configure a Conventional Main for Off-Net Number Recognition. Access LD16

REQ	NEW, CHG	Add or change a route
DDMI	(0)-255	Digit manipulation index (prompted if the response to CNVT is Yes)
ATDN	xxxx	Attendant DN of conventional main (prompted if the response to CVNT is Yes)
Note 1: If the DN expansion package is equipped, the attendant DN can have up to 7 digits.; otherwise, only 4 digits can be entered.		

The following procedure involves several load programs. In all cases, the prompt is NCOS; the range is 0-99; the default is 0.

Step 8: Gather data to assign a Network Class of Service group number to a list of items

- 500/2500 sets (LD10)
- SL-1 telephones, M1000 series telephones, M2000 series digital telephones, M3000 Touchphones (LD12)
- Attendant consoles (LD12)
- Trunks (LD14)
- Direct Inward System Access directory number See LD24)
- Authorization code (LD88)

Step 9: Enter data into the database

Enter the data gathered into the data base:

- 1 Login.
- 2 Load the appropriate load program.
- 3 Enter data requested by prompts until REQ prompt returns.

4 More data to be added/removed?

Yes = go to Step 2

No = perform data dump

A data dump takes approximately 3 minutes to complete. If large amounts of data are being added, perform a data dump periodically. This ensures that entered data is not lost should a system reload occur.

5 Print data (respond with PRT to prompt REQ) and verify against data forms.

6 Corrections required?

Yes = go to Step 2

No = Refer to 553-2YY1-230 and perform applicable tests.

Appendix C — Network Queuing

Introduction

This appendix describes the queuing features and options that are available to switches which operate in a private network environment. The queuing features are

- Off-Hook Queuing (OHQ)
- Call-Back Queuing (CBQ)
- Coordinated Call-Back Queuing (CCBQ)
- Call-Back Queuing to Conventional Mains (CBQCM)

A private network consists of one or more of the following types of switch.

Meridian 1 Node

A Meridian 1 switch that is equipped with either the Basic Alternate Route Selection (BARS) or Network Alternate Route Selection (NARS) software feature packages. NARS and BARS features are described in *Basic and network alternate route selection description* (553-2751-100).

Meridian 1 Main

A Meridian 1 switch that is connected via tie trunks to a Meridian 1 Node and is equipped with the Network Signaling (NSIG) and Network Class of Service (NCOS) software packages. A Meridian 1 Main can be connected to only one Meridian 1 Node. A Meridian 1 Main may also be equipped with the BARS and OHQ features.

Conventional Main

A switch (Meridian 1 or any other type) that is connected via tie trunks to a Meridian 1 Node and is not equipped with any network-related features.

Information presented in this practice assumes the reader is familiar with the terminology and concepts of the alternate route selection capabilities as described in *Basic and network alternate route selection description* (553-2751-100).

Off-Hook Queuing

Off-Hook Queuing (OHQ) is a software feature that can be equipped at a Meridian 1 Node and/or Meridian 1 Main. The feature enables a call originator to remain off-hook for a short time (customer programmable) until a network facility for call completion becomes available at the Node or Main.

OHQ eligibility

Network calls may be placed in an OHQ if all trunk routes (entries) in the initial route set of a route list (see Note) are busy and the following criteria are met.

- At least one of the trunk routes in the initial route set of a route list is defined as being eligible for OHQ.
- The Network Class of Service (NCOS) of the call originator (at a Node or a Main) is defined to permit OHQ.
- The incoming trunk group at the Node or the Main is defined in software to permit OHQ for incoming calls.
- The Traveling Class Mark (TCM) received at the Node from an Electronic Tie Network (ETN) switch is compatible with a Facility Restriction Level (FRL) that is defined to permit OHQ.
- The Traveling Class of Service (TCOS) received at the Node from another Node is compatible with an FRL that is defined to permit OHQ.
- The probability that the call will timeout in the OHQ before a trunk becomes available is below a specific threshold.
- The OHQ feature is enabled for the customer.

A Meridian 1 defined initial set “marker” determines which route list entries are in the initial route set. Typically, the initial route set contains the cheaper routes to a particular destination. The remaining routes in the route list (if any) comprise the extended route set and are usually the more expensive routes to the destination. Only routes in the initial route set should be defined to allow OHQ. OHQ will not be offered by routes in the extended route set even if they are defined to allow OHQ.

Calls which do not meet the preceding requirements for OHQ eligibility may be offered CBQ at this point.

OHQ availability

The OHQ software performs an “availability” test to prevent calls from entering the OHQ when there is a high probability that the call will timeout before a trunk becomes available. The following procedure is used to make this test.

- For each trunk route, a counter is maintained which reflects the current number of calls with the maximum queue priority of 3 queued against it. (This includes all calls in OHQ and those CBQ and Ring Again calls that are currently at priority 3.)
- Each trunk route has a threshold value which indicates the maximum number of priority 3 calls that can be queued against it before OHQ timeout becomes a high probability. Before a call is placed in the OHQ the current queue count is compared with the threshold value for each eligible trunk route in the initial set of routes. If at least one of the trunk routes has a count which is less than or equal to the threshold value, the call is allowed to OHQ against all OHQ-eligible routes.

OHQ offer

If all eligibility and availability checks are successful, the call originator is given a distinctive OHQ offer tone (a 1s burst of 440 Hz tone) when the call enters the OHQ. The call originator either accepts the OHQ offer by remaining off-hook, or declines the offer by going on-hook.

OHQ calls are placed in a priority-ordered queue with all other calls waiting for trunks. OHQ calls are assigned maximum priority (3) because other network facilities may be held while the call is queued. A timer with an initial value equal to the software-defined OHQ time limit is started to limit the duration of the OHQ. (The OHQ time limit is service changeable within the range of 2 to 60 s.)

The queue is examined whenever a trunk becomes idle in a trunk route on which one or more calls are queued. If a call is found which can be terminated on an idle trunk, the available trunk is seized and the call terminated.

If the OHQ timer expires before the call can be terminated, the call is removed from the OHQ. At this time, the remaining eligible routes in the route list (the extended set) are examined and the call is either terminated or given standard blocking treatment if no facilities are available. (CBQ will not be offered at this point as OHQ was already offered.) Expensive Route Warning (ERWT) is not given to calls which have been queued even if terminated on expensive facilities.

OHQ can be offered to call originators located at a Node, Main, Conventional Main or ETN switch. Also, as a call progresses through the network, OHQ can be offered to the call originator from any of the Nodes or Mains that are used to process the call (i.e., OHQ can be offered more than once for a given call).

OHQ traffic measurements

Traffic measurement data related to usage of the OHQ feature is accumulated at a Node or Main that is equipped with Network Traffic (NTRF) feature. Data for OHQ usage is available in the publication *Traffic measurement* (553-2001-450).

Routing measurements (TFN001)

Available at a Node or Main. Reflects on a route list basis the number of calls placed in the OHQ, the average time a call waited in the OHQ and the number of calls abandoned while in the OHQ.

NCOS measurements (TFN002)

Available at a Node or Main. Reflects on a NCOS basis the number of call placed in the OHQ and the average time a call waited in the OHQ.

Incoming trunk group measurements (TFN003)

Available at a Node. (The measurements are also available at a Main if the OHQ feature is equipped and some incoming trunk groups are defined as OHQ eligible.) Reflects on an incoming trunk (or two-way) trunk group basis the number of calls placed in the OHQ and the average time a call waited in the OHQ.

Feature interactions

The OHQ feature has the following interactions with existing Meridian 1 features.

Call modification

Station users are not allowed to activate call modification features (hold, call transfer, conference) while waiting in the OHQ. Switchhook flash used to activate features from 500/2500 sets is ignored. Similarly, operation of SL-1 or digital set feature keys is ignored.

Camp-on, call waiting

If the attendant extends a call to a station that is in the OHQ, the call-waiting tone is not offered to the station. If the attendant releases, the call is camped-on the OHQ station but no warning tone is given. When the camp-on call is recalled to the attendant console, the attendant can repeat the camp-on procedure. Once the OHQ call is in an established state, the camp-on tone is provided.

Attendant functions

- Attendant cannot barge-in during trunk seizure for OHQ calls.
- If the attendant extends a network call for a station user and the call is offered OHQ, the attendant must inform the caller of the OHQ offer before releasing from the call.
- The attendant is not allowed to operate the release key or another loop key if the source call is in conference and the destination call is in the OHQ. Operation of the release destination key is permitted, however, and causes the OHQ call to be abandoned.

Call-Back Queuing

Call-Back Queuing (CBQ) is a Node feature which provides queuing for network calls which encounter busy or blocked facilities at the Node. CBQ enables the calling party to go on-hook after activation of the Ring Again feature [see *Features and services* (553-3001-305)] and receive a callback from a Node when a network facility becomes available.

The CBQ feature is available only to stations located at a Node. Unlike OHQ, CBQ is offered only at the originating Node. Access to CBQ is accomplished through the existing Ring Again feature.

CBQ options

Two options for CBQ eligibility are defined by the call originator's NCOS. The first option, CBQ(i), means the CBQ may be offered after only the initial route set of a route list has been examined for an available route. The second option, CBQ (a), means that CBQ may be offered after both the initial and extended (i.e. all) route sets of a route list have been examined. In either case, a call offered CBQ is queued initially against only the initial route set.

CBQ eligibility

Before offering CBQ to a call originator, the following eligibility tests are performed.

- At least one of the routes in the initial route set is defined as CBQ eligible.
- The user's NCOS is defined as permitting CBQ, either CBQ(i) or CBQ(a).
- The call is not eligible for OHQ. Calls that are eligible for both OHQ and CBQ will be offered OHQ.
- The user's telephone is allowed access to the Ring Again feature and does not have another CBQ or Ring Again call already in the queue.
- The CBQ feature is enabled for this customer.

CBQ(i) eligible

For call originations by a caller defined as CBQ(i) eligible, the system searches the initial route set for an available route. If no available route is found, CBQ is offered to the caller subject to the CBQA eligibility tests.

CBQ(a) eligible

For call originations by a caller defined as CBQ(a) eligible, the system examines the initial route set for an available route. If no available route is found, the extended route set is then searched for an available route. If an available route is not found in the extended set, then CBQ is offered, subject to the CBQ eligibility tests. However, if an available route in the extended route set is found that is designated as expensive, and the user's NCOS allows ERWT, the tone is given and the system delays terminating the call. During this delay the user has the following options.

- Refuse the expensive route by abandoning the call.
- Wait, and allow the call to complete over the expensive route.
- Activate the Ring Again feature (feature key or access code) to place the call in the CBQ. The user must be CBQ(a) eligible; otherwise, operation of the Ring Again feature is ignored.

CBQ offer

The CBQ offer consists of an optional recorded announcement followed by overflow tone. If the station user wishes to accept the CBQ offer, Ring Again must be activated within 30 s. Ring again activation follows present feature operation for SL-1 or digital and 500/2500 sets [see *Features and services* (553-3001-305)]. The CBQ offer can be refused by going on-hook. If the station user neither accepts nor refuses the CBQ offer within 30 s, the call is force disconnected.

CBQ calls are placed in a priority-ordered trunk queue (together with OHQ calls, if any) with a starting priority and maximum priority defined by the call originator's NCOS. At the same time, two timers are started — a queue promotion timer and a route advance timer — each with values defined through the originator's NCOS. At intervals defined by the queue promotion timer, the priority of the call is incremented until it reaches its maximum priority. Each time the call priority is incremented, its position in the CBQ is advanced. If the route advance timer reaches its maximum value before the call can be terminated on a route in the initial set, the extended set or routes is added to the routes that the call is currently queued against.

Expensive route warning tone is not given to calls which have been queued, even if terminated on expensive facilities. Unless canceled by the call originator, CBQ calls remain in the queue until they have been offered a trunk; there is no time limit on CBQ calls.

CBQ callback

When a trunk becomes available for a CBQ call, it is seized to prevent incoming originations during the CBQ callback period. Outpulsing of digits (either those originally dialed by the user or those required as a result of digit manipulation) is started at a slow, fixed rate. The number of digits to be outpulsed determines how long the trunk can be held while CBQ callback is being offered to the originating station. The system computes this time by allowing 10 s before the first digit is outpulsed at 2.56 s between subsequent digits.

The originator of the call is alerted to the CBQ callback by either tone buzzing and winking ring again feature lamp (SL-1 or digital set) or short bursts of ringing (500/2500 set). The set user must accept the callback within the computed value of outpulse time or the service-changeable CBQ time limit of 10 to 30 s, whichever is less. A user with a 500/25000 set must accept the callback within 6 s. Acceptance of the CBQ callback is performed with present Ring Again operating procedures. (When a CBQ callback is answered at a digit display set, the original dialed digits are displayed.) If the user does not answer the callback within the time limit, the call is removed from the queue and discarded. If the user accepts the callback within the time limit, the call is terminated. A CBQ call can be canceled by the originating station via the existing procedures for Ring Again cancellation.

CBQ traffic measurements

Traffic measurement data related to the usage of the CBQ feature is accumulated at a Node or Main equipped with the Network Traffic (NTRF) feature. Data for CBQ usage is available in the publication *Traffic measurement* (553-2001-450).

Routing measurements (TFN001)

Available at a Node or Main. Reflects on a route list basis the number of CBQ offers, the number of CBQ callbacks, the average time in the CBQ and the number of CBQ cancellations.

NCOS measurements (TFN002)

Available at a Node or Main. Reflects on an NCOS basis the number of CBQ calls and the average time in the CBQ.

Feature interactions

The CBQ feature interacts with the following features.

Barge-In, force disconnect

Between the time a trunk is seized for a CBQ call and the user accepts the CBQ callback, the trunk can be stolen by the attendant or force disconnected through service change. If this occurs, there is no guarantee that the call can be terminated when the user accepts the CBQ callback. Under these circumstances, the call is treated like a new origination and NARS/BARS is used to re-attempt termination. This can result in the call being blocked and being offered CBQ a second time.

Hunting, call forward, multiple appearance DN

CBQ callbacks to stations at a Node are offered only to the originating station, regardless of the hunting or call forwarding which may be in effect. Other appearances of a station's Directory Number (DN) are not offered the callback.

Coordinated Call-Back Queuing

The Coordinated Call-Back Queuing (CCBQ) feature enables stations at a Meridian 1 Main to be offered CBQ when network calls are blocked at the serving Meridian 1 Node. When facilities become available at the Node, the call originator at the Main is alerted by a callback from the Node. (This feature requires that Main and associated Node be equipped with the network signalling feature).

CCBQ eligibility

When a station at a Main originates a network call through a Node, the NCOS of the call originator, call type, and whether the station is allowed access to the ring again feature is transmitted to the Node. (If an authcode is entered at the Main prior to dialing a network call, the NCOS associated with the authcode is transmitted to the Node.) When received by the Node, this NCOS is used to determine CCBQ eligibility and is used for the duration of the call, unless further modified by the Authcode Conditionally Last feature [see *Basic and network authorization code description* (553-2751-103)].

The CBQ eligibility tests are performed. In addition, a check is made that the incoming trunk group from the Main is defined (at the Node) to permit CBQ and that the type allows CBQ. CCBQ is offered to the user at the Main if the eligibility tests are successful. If the tests are unsuccessful, standard call blocking is applied to the call.

As for stations at a Node, the call originator at a Main can invoke Ring Again upon receipt of ERWT if the originator's NCOS is defined at the Main as CBQ (a) eligible.

CCBQ offer

The CCBQ offer and acceptance sequences are identical to those for stations at the Node (SBQ offer). The optional recorded announcement and overflow tone are provided by the Node. The offer cannot be accepted until the recorded announcement is completed and the offer tone is started. In addition, after the recorded announcement is provided, the Node transmits a signal to the Main. This signal indicates that the call is in a state which allows Ring Again.

When the call originator at the Main activates ring again the Main assigns a unique “queue identification” number to the call. This number is transmitted to the Node to indicate CCBQ acceptance. At the Main, the call is placed in a holding queue. At the Node, the call (together with the queue identification number) is placed into the trunk queue. The Main to Node tie trunk is released.

CCBQ callback

When an outgoing trunk is seized by the Node for a CCBQ call, slow outpulsing is started to hold the trunk while a callback is made to the call originator at the Main. The Node seizes an available (note) tie trunk to the Main and transmits the “queue identification” number of the call to the Main. The Main then initiates a callback to the call originator. Callback presentation to the call originator is as for standard ring again (CBQ callback).

If no tie trunks to the Main are available, the outgoing trunk is released and can be offered another call. The CCBQ call retains its position in the queue but is not offered another trunk until a tie trunk to the Main becomes available.

When the call originator at the Main accepts the CCBQ callback, answer supervision is sent from the Main to the Node. The Node then completes the call.

If the call originator is equipped with a 500/2500 set and is engaged in a call when the Node initiates a CCBQ callback, a signal is transmitted from the Main to the Node. The Node releases the outgoing trunk and places the CCBQ call into a holding queue for 5 min. No attempt is made to seize another outgoing trunk for the call until the holding time expires. This process occurs only once.

If the originating station is still busy after the holding time has expired, the CCBQ is canceled automatically at the Node. No indication is given to the call originator of the CCBQ cancellation.

To prevent the CCBQ call from remaining indefinitely in the holding queue at the Main, the Main sets a time limit of one hour for CCBQ calls. When this time limit expires, the CCBQ call is canceled automatically. (CCBQ callback to a busy station is as for normal Ring Again).

The call originator at the Main can cancel the CCBQ call at any time. The Node is not aware of the cancellation until the CCBQ callback is attempted.

CCBQ traffic measurements

Traffic measurement data related to usage of the CCBQ feature are accumulated at the Node. The data is reflected in the Incoming Trunk Group Measurements (TFN003). The measurements reflect, on an incoming trunk group basis, the number of CCBQ offers, the number of CCBQ offers that are accepted and the number of CCBQ callbacks not answered or CCBQ cancellations. [Refer to *Traffic measurement* (553-2001-450).]

Feature interactions

The CCBQ feature interacts with the following Meridian 1 features.

Initialization

If the Main initializes while calls are queued at the Node, CCBQ callbacks from the Node are not answered because the initialization has cleared the holding queue at the Main. The Node treats these calls as callback-no-answer calls and cancels the CCBQ automatically. If the Node initializes, CCBQ calls in the trunk queue are lost. The Main cannot detect this situation. To prevent calls from remaining indefinitely in the holding queue at the Main, Main sets a time limit of one hour for CCBQ calls. If a call back from the Node is not received within one hour, the Main cancels the CCBQ calls automatically.

Attendant functions

Attendants at a Main are not offered CCBQ. Attendant barge-in on trunks involved in CCBQ operations results in cancellation of the CCBQ call.

AIOD and ANI

Automatic Identification of Outward Dial (AIOD) and Automatic Number Identification (ANI) facilities can be used to complete CCBQ calls from a Node. The outgoing toll call is billed to the access tie trunk rather than the station at the Main.

Coordinated Dialing Plan

If CCBQ is offered to a CDP call as a result of trunk blocking at the Node, the optional recorded announcement is not given before the CCBQ offer tone.

Call transfer, conference

Stations at a Main can be offered CCBQ when initiating transfer or conference calls. CCBQ is not offered to 500/2500 stations performing these operations.

Feature limitations

A station at a Main is only allowed 10–12 s to activate ring again after the CCBQ offer (or ERWT) rather than the 30 s allowed for stations at the Node. Accordingly, the time interval of the recorded announcement (if provided) must be properly engineered to ensure that the station at the Main has time to activate CCBQ when it is offered.

The time available to answer a CCBQ callback at stations at a Main is 2–3 s less than the time available to stations at the Node, due to the time required for access trunk seizure and signaling.

Tie trunk groups used to provide CCBQ to a Main cannot be arranged for loop start dial repeating signaling and cannot have joint far-end disconnect control as these arrangements do not allow the call to be disconnected after a CCBQ callback.

Tones and recorded announcements used for CCBQ to users at a Main are provided by the Node. Traffic loads imposed on these facilities at the Node must be considered when engineering Nodes.

Call-Back Queuing to Conventional Mains

The Call-Back Queuing to Conventional Mains (CBQCM) feature enables call originators at a Conventional Main to access the CBQ feature at a Node.

CBQCM eligibility

When a station at a Conventional Main originates a network call through a Node, the NCOS assigned to the incoming trunk group is used to determine the CBQCM eligibility. This NCOS, as well as the incoming trunk group, must be defined as CBQ eligible.

If the call originates from an ETN switch, the Traveling Class Mark (TCM) transmitted to the Node, by the ETN switch, must be compatible with a CBQ-eligible Facility Restriction Level (FRL) at the serving Node.

CBQCM offer

The CBQCM offer to call originator at a Conventional Main consists of an optional recorded announcement, followed by special (interrupted) dial tone. (The announcement and tones are provided from the Node.) To accept the CBQCM offer, the call originator dials the extension number associated with the telephone being used for the call. When the last digit of the extension number is dialed, a confirmation tone (three 256 ms bursts of dial tone) is sent from the Node to the call originator. The call is placed in the CBQ at the node when the call originator goes on hook.

The CBQCM offer can be refused by going on-hook any time before the last digit of the extension number is dialed or by remaining off-hook for longer than 30 s after receipt of the confirmation tone. If the CBQCM is neither accepted nor rejected within 30 s, the caller is given overflow tone (from the Node) and the call is disconnected.

CBQCM callback

When an outgoing trunk becomes available at the Node, it is seized and slow outpulsing is started. The Node then seizes a tie trunk (See Note) to the conventional main and outpulses the extension number of the call originator. The call originator must answer the callback before slow outpulsing is completed; otherwise, the callback is canceled and the outgoing trunk is released.

If no tie trunks are currently available to the conventional main, the Node releases the outgoing trunk. The CBQCM call retains its position in the queue but is not offered another outgoing trunk until a tie trunk to the Conventional Main becomes available.

When the call originator answers the CBQCM callback, answer supervision must be transmitted from the Conventional Main to the Node. Upon receipt of answer supervision from the Conventional Main, the Node transmits a tone (three 256ms bursts of dial tone) to notify the call originator that the call is a CBQCM callback, and completes the call.

If the call originator's station is busy, or the originator does not answer when the callback is placed, the Node places the call in a suspended state for 5 min. After 5 min., another callback is attempted if the outgoing trunk is free. If the station which originated the call is still busy or does not answer, the Node cancels the call.

No provision is made for CBQCM cancellation by a call originator at a Conventional Main. Once the CBQCM offer is accepted, the call remains in the queue until the Node initiates a callback.

CBQCM traffic measurements

Traffic measurement data related to usage of the CBQCM feature are accumulated at the Node. This data is outputted as part of the Incoming Trunk Group Measurements (TFN003). The data reflects on an incoming trunk group basis the number of the CBQCM offers and acceptances, average time in CBQCM, number of blocked CBQCM callbacks and the number of CBQCM callbacks not answered.

Feature limitations

Station users at Conventional Mains cannot activate Ring Again to refuse expensive routes after the Expensive Route Warning Tone (ERWT) is given.

The Node seizes the same tie trunk group that was used to initiate CBQCM for the CBQCM callback. Thus, these trunk groups must be two-way (incoming/outgoing).

Conventional Mains must provide answer supervision on tie trunks connected to the Node. These switches must also permit transmission or repetition of station dial pulses for CBQCM operation. This feature cannot be used with systems that operate in senderized mode. Operation may require adjustment of the interdigit timeout on systems that employ simulated cut-through operation.

Conventional Mains must not allow CBQCM callback calls to be modified by call transfer or call forward. Such call modification can result in the tie line not being released upon call completion.

The station user at a Conventional Main must dial the internal DN of his/her station to accept a CBQCM offering, rather than the CDP-DN, if the Node is equipped for the Coordinated Dialing Plan.

Tie trunk groups used to provide CBQCM to a Conventional Main cannot be arranged for loop start dial repeating signaling and cannot have joint far-end disconnect control as these arrangements do not allow the call to be disconnected after a CBQCM callback.

Because tones and recorded announcements for CBQCM are provided by the Node, traffic loads on these facilities at the Node must be considered when engineering Nodes.

Coordinated Call-Back Queuing Against Mains

The Coordinated Call-Back Queuing Against Mains (CCBQAM) feature enables stations at Nodes to be offered CBQ for network calls which are blocked at a Main. When facilities become available at the Main, the call originator at the Node is alerted by a callback from the Main. The CCBQAM otherwise functions identically to CCBQ at the Node.

Meridian 1

Option 11C

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