ISDN Basic Rate Interface Feature Fundamentals
Avaya Communication Server 1000

Release 7.6
NN43001-580
Issue 06.01
March 2013

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Contents

Chapter 1: New in this release

The following section details what's new in *Avaya ISDN Basic Rate Interface Feature Fundamentals, NN43001-580* for Avaya Communication Server 1000 (Avaya CS 1000) Release 7.6.

Features

There are no updates to the feature descriptions in this document.

Other

Revision History

March 2013	Standard 06.01. This document is up-issued to support Avaya Communication Server 1000 Release 7.6.
October 2011	Standard 05.04. This document is up-issued to remove references to legacy feature and hardware content that is no longer applicable to or supported by Communication Server 1000 systems.
May 2011	Standard 05.03. This document is up-issued to include information about configuring a trunk clock reference source for Communication Server 1000 Release 7.5.
November 2010	Standard 05.02. This document is published to support Communication Server 1000 Release 7.5.
November 2010	Standard 05.01. This document was issued to support Communication Server 1000 Release 7.5.
June 2010	Standard 04.01. This document is up-issued to support Communication Server 1000 Release 7.0.
May 2009	Standard 03.01. This document is up-issued to support Communication Server Release 6.0. This NTP may contain information on or refer to products and naming conventions that are not supported in this release. This information is included for legacy purposes and convenience only. This includes but is not limited to

	items, such as: SSC; ISP 1100; ITG Pentium cards; and Media Cards running certain IP Line applications.
December 2007	Standard 02.02. This document is up-issued to support Communication Server Release 5.5.
June 2007	Standard 01.02. This document is up-issued to remove the Nortel Networks Confidential statement.
June 2007	Standard 01.01. This document is issued to support Communication Server 1000 Release 5.0. This document contains information previously contained in the following legacy document, now retired: ISDN Primary Rate Interface Features (553-3001-369).
March 2007	Standard 00.01. This document is up-issued to reflect changes in the Edited content to Configure a DSL for a line section.
August 2005	Standard 3.00. This document is up-issued to support Communication Server 1000 Release 4.5.
September 2004	Standard 2.00. This document is up-issued for Communication Server 1000 Release 4.0.
October 2003	Standard 1.00. This document is a new technical document for Succession 3.0. It was created to support a restructuring of the Documentation Library, which resulted in the merging of multiple legacy technical documents. This new document consolidates information previously contained in the following legacy documents, now retired:

- *ISDN Basic Rate Interface: Description* (553-3901-100)
- *International ISDN Primary Rate Interface: Feature Description and Administration* (553-2901-301)
- *ISDN Basic Rate Interface: Administration* (553-3901-300)

Chapter 2: Customer service

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Chapter 3: Introduction

This document is a global document. Contact your system supplier or your Avaya representative to verify that the hardware and software described are supported in your area.

Subject

This document describes the following for Integrated Services Digital Network Basic Rate Interface (ISDN BRI):

- features
- concepts
- functional and physical characteristics
- general engineering guidelines on configuring ISDN BRI functions and features

Note:

ISDN BRI trunking is not supported in North America.

Note on legacy products and releases

This document contains information about systems, components, and features that are compatible with Communication Server 1000 software. For more information on legacy products and releases, click the Technical Documentation link under Support & Training on the Avaya home page:

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Applicable systems

This document applies to the following systems:

- Avaya Communication Server 1000E (Avaya CS 1000E)
- Avaya Communication Server 1000M Single Group (Avaya CS 1000M SG)
- Avaya Communication Server 1000M Multi Group (Avaya CS 1000M MG)

System migration

When particular Meridian 1 systems are upgraded to run CS 1000 software and configured to include a Signaling Server, they become CS 1000 systems.Table 1: Meridian 1 systems to CS 1000 systems on page 46 lists each Meridian 1 system that supports an upgrade path to a CS 1000 system.

Table 1: Meridian 1 systems to CS 1000 systems

This Meridian 1 system	Maps to this CS 1000 system
Meridian 1 PBX Chassis	CS 1000E
Meridian 1 PBX Cabinet	CS 1000E
Meridian 1 PBX 61C	CS 1000M Single Group
Meridian 1 PBX 81C	CS 1000M Multi Group

For more information, see one or more of the following documentss:

- *Avaya CS 1000M and Meridian 1 Large System Upgrades Overview, NN43021-458*
- *Avaya Communication Server 1000E Software Upgrades, NN43041-458*
- *Avaya Communication Server 1000E Upgrade - Hardware Upgrade Procedures, NN43041-464*

Intended audience

This document is intended for individuals responsible for administering CS 1000 and Meridian 1 systems.

Conventions

Terminology

In this document, the following systems are referred to generically as system:

- Communication Server 1000E (CS 1000E)
- Communication Server 1000M (CS 1000M)
- Meridian 1

Related information

This section lists information sources that relate to this document.

Technical documentation

The following technical documents are referenced in this document:

- *Avaya Software Input Output Reference - Administration, NN43001-611*
- *Avaya Software Input Output Reference - Maintenance, NN43001-711*

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Chapter 4: Functional description

Contents

This section contains information on the following topics:

Introduction

This chapter describes the following:

- ISDN BRI system functions
- operation of the Multi-Purpose ISDN Signaling Processor (MISP)
- Basic Rate Signaling Concentrator (BRSC)
- S/T Interface Line Card (SILC)
- U Interface Line Card (UILC)
- two forms of packet handlers - external and integrated
- Digital Subscriber Loop (DSL) concepts
- how to initialize and assign service attributes to ISDN BRI terminals and trunks assigned to a DSL

Operating parameters

Please note the following important operating parameters pertaining to ISDN BRI functionality:

- ISDN BRI trunk access is not supported in North America.

ISDN Basic Rate Interface overview

ISDN Basic Rate Interface (ISDN BRI) is a digital connection that provides three digital channels. These channels consist of two 64 kbps Bearer channels (B-channels) and one 16 kbps signaling channel (D-channel). This 2B+D connection is known as a Digital Subscriber Loop (DSL). The DSL can be configured to provide line access, trunk access, or packet data transmission.

Line Access provides a digital connection from a system ISDN BRI card to ISDN terminals that comply with CCITT, ANSI, ETSI NET-3 and ETS 300 403 (including EuroISDN), INS NET-64 (including Japan D70), National ISDN-1 (NI-1), 1TR6, and Numerics VN2 standards; examples of terminals are telephone sets, FAX machines, personal computers, and video display terminals.

Trunk Access provides Meridian Customer Defined Network (MCDN) TIE trunk connectivity between systems, QSIG ISDN BRI trunk connectivity, and CO/DID trunk connectivity to local exchanges that support Numeris VN3, 1TR6, ETSI NET-3 and ETS 300 403 (EuroISDN), INS NET-64 (including Japan D70), Australia ETSI, and Asia-Pacific protocols.

Note:

ISDN BRI trunk access is not supported in North America.

The system supports both B-channeland D-channel packet data transmission through an external DPN-100 packet handler, or an integrated Meridian 1 Packet Handler (MPH).

B-channels and D-channel

B-channels can be automatically assigned and reassigned to different voice and data terminals in circuit-switched applications, they can be dedicated to specific terminals for packet data applications, or configured for ISDN BRI trunk applications.

For ISDN BRI line access, the ability to dynamically connect different terminals on one DSL provides more flexibility, connectivity, and service diversity than the conventional "hard wired" connections where each channel is dedicated to one terminal. The D-channel is used for signaling and low speed packet data transmission.

For ISDN BRI trunk access, one ISDN BRI trunk can be assigned to each B-channel in the 2B +D configuration. The D-channel is used for signaling.

ISDN BRI interfaces to the system

ISDN BRI provides two types of interfaces to the system - the S/T interface or a U interface. The DSLs can be configured for either an S/T or U interface, and can be configured for either line or trunk access.

The S/T interface is a globally accepted standard interface. This interface is provided by the SILC line card, which supports eight DSLs.

The U interface is implemented as an ANSI standard interface only (2B1Q line encoding). This interface is provided by the UILC line card, which also supports eight DSLs.

Data link and network processing

The ISDN BRI data link and network layering processing functions are carried out by the Multi-Purpose ISDN Signaling Processor (MISP), or the Basic Rate Signaling Concentrator (BRSC); these functions, as well as the functions of the other system hardware, are detailed in the "Engineering guidelines" section of this document.

Note:

The BRSC cannot be used for trunk access. Therefore, the BRSC only performs the data link processing.

General ISDN BRI capabilities

The most important capabilities of ISDN BRI are:

- for line access
    - simultaneous voice and circuit-switched data over a single DSL
    - B-channel and/or D-channel packet data transmission over a single DSL
    - multiple physical terminals connected to a single DSL
    - multiple logical devices associated with each DSL
    - diverse ISDN-compliant third party terminals (compliant with CCITT, ANSI, ETSI NET-3 and ETS 300 403, INS NET-64, National ISDN-1, 1TR6, Numeris VN2, and EuroISDN standards)
- for trunk access
    - MCDN ISDN BRI TIE trunk connectivity
    - QSIG ISDN BRI TIE trunk connectivity
    - CO/DID trunk connections to local exchanges that support Numeris VN3, 1TR6, ETSI NET-3 and ETS 300 403 (EuroISDN), INS NET-64 (Japan D70), Australia ETSI, and Asia-Pacific protocols

The OSI model

ISDN standards follow the Open System Interconnect (OSI) protocol model to control ISDN functions. The OSI model defines seven layers required to perform all ISDN functions from establishing an end-to-end connection between two terminals or trunks, to making a decision about the type of application that is to be activated. Figure 1: OSI model on page 53 shows the seven layer OSI model. ISDN BRI utilizes only the first three layers. These are:

- Physical layer (layer 1) that provides a network-to-terminal or trunk connection (SILC/UILC).
- Data link layer (layer 2) that provides signaling used to establish a communication link between terminals or trunks over ISDN (point-to-point signaling for terminals, and point-to-multipoint signaling for trunks). It also performs some error checking and error recovery (MISP/BRSC).

  The BRSC applies to terminals only.
- Network layer (layer 3) that controls initialization procedures and assigns service attributes to terminals and trunks. It also controls the call processing procedures (MISP, MPH).

Once these three layers are established, the functional role of ISDN BRI in the OSI protocol sequence is complete. The higher layers of the OSI protocol model are handled by the end-user application software.

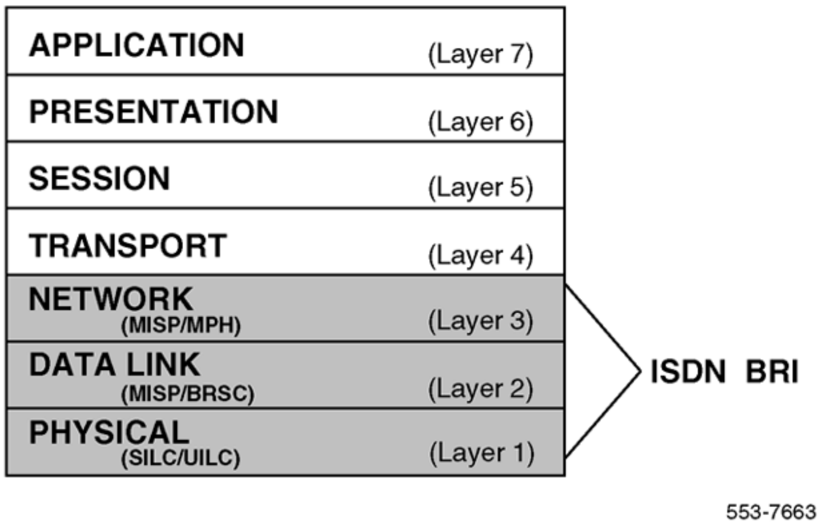


Figure 1: OSI model

ISDN BRI line access

ISDN BRI line access provides 2B+D ISDN service to terminating equipment such as ISDN telephone-sets and data terminals. ISDN BRI line connections are configured on a per DSL basis; that is, DSL line connections can be configured on any given DSL for any SILC or UILC.

The B-channels are dynamically assigned to different voice and data terminals in circuit-switched line applications. In case of packet data applications, the B-channels are dedicated connections. The D-channel is used for signaling and dynamic D-channel packet data.

S/T interface configured for line access

Figure 2: ISDN Basic Rate Interface S/T interface configured as a line on page 54 illustrates a typical ISDN BRI configuration, showing a system with an ISDN BRI S/T interface and the ISDN BRI terminals connected to it; the terminals must comply with CCITT, ANSI, ETSI NET-3 (including EuroISDN), INS NET-64 (including Japan D70), National ISDN-1, 1TR6, and Numeris VN2 standards.

The S/T interface is a four-wire, polarity-sensitive interface which, configured for line application, can support eight physical voice and data terminals, and up to 20 logical terminals on one DSL. A physical terminal is any device directly connected to a DSL. The terminals labeled 1, 2 and 3 in Figure 2: ISDN Basic Rate Interface S/T interface configured as a line on page 54 are physical terminals.

A logical terminal (terminal 4 in Figure 2: ISDN Basic Rate Interface S/T interface configured as a line on page 54) is any terminal that can communicate with the system over a DSL. It can be directly connected to the DSL through its own physical termination or it can be indirectly connected through a common physical termination. Please refer to the "Terminal addressing and service profile assignment" section of the Engineering Guidelines chapter for an illustration of how a single physical termination can actually connect multiple logical terminals. All of the logical terminals connected to the DSL share the two B-channels provided by the S/T interface.

Note:

The Terminal Adapter (TA) that is shown in the illustration is used to adapt non-ISDN BRI terminals to ISDN BRI line interface standards.

The length of an S/T interface DSL depends on the specific terminal configuration and the DSL wire gauge; however, it should not exceed 1 km (3280 ft).

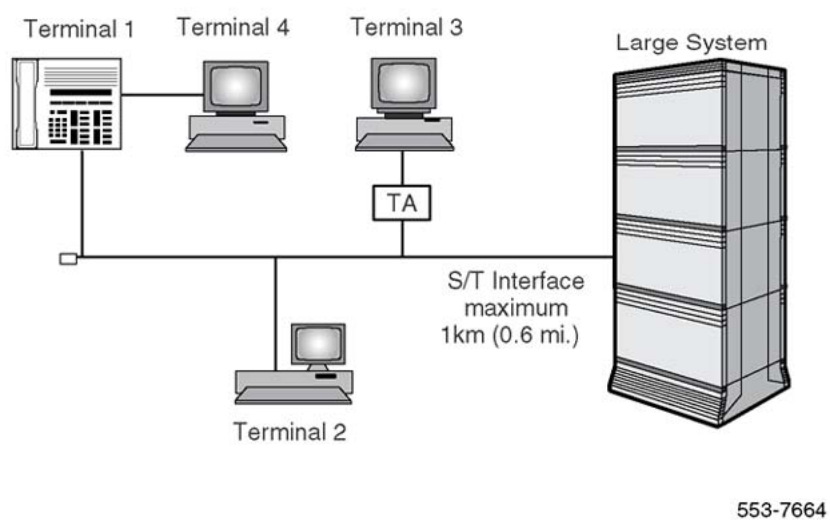


Figure 2: ISDN Basic Rate Interface S/T interface configured as a line

U interface configured for line access

The U interface is a two-wire interface that provides point-to-point connection over a DSL. Each U interface provides two B-channels and one D-channel and supports only one physical termination. This termination can be to a Network Termination 1 (NT1) or directly to a single U interface terminal that contains an internal NT1. Normally this physical termination is to an NT1, which provides a conversion from a U interface to an S/T interface that allows up to eight physical terminals to be connected.

The length of a UILC DSL depends on the specific terminal configuration and the DSL wire gauge; however, it should not exceed 5.5 km (3.3 mi). When connected to an NT1 the DSL length is effectively extended to 6.5 km (3.9 mi) and utilizes the multi-terminal capability of an S/T interface.

Figure 3: ISDN Basic Rate Interface U interface configured as a line on page 55 illustrates a typical ISDN BRI configuration showing a system with an ISDN BRI U interface. ISDN BRI terminals can be connected to the NT1 through the S/T interface; the terminals must comply with CCITT, ANSI, ETSI NET-3 (including EuroISDN), INS NET-64 (including Japan D70), National ISDN-1, 1TR6, Numeris VN2 standards), Australia ETSI, and Asia-Pacific protocols.

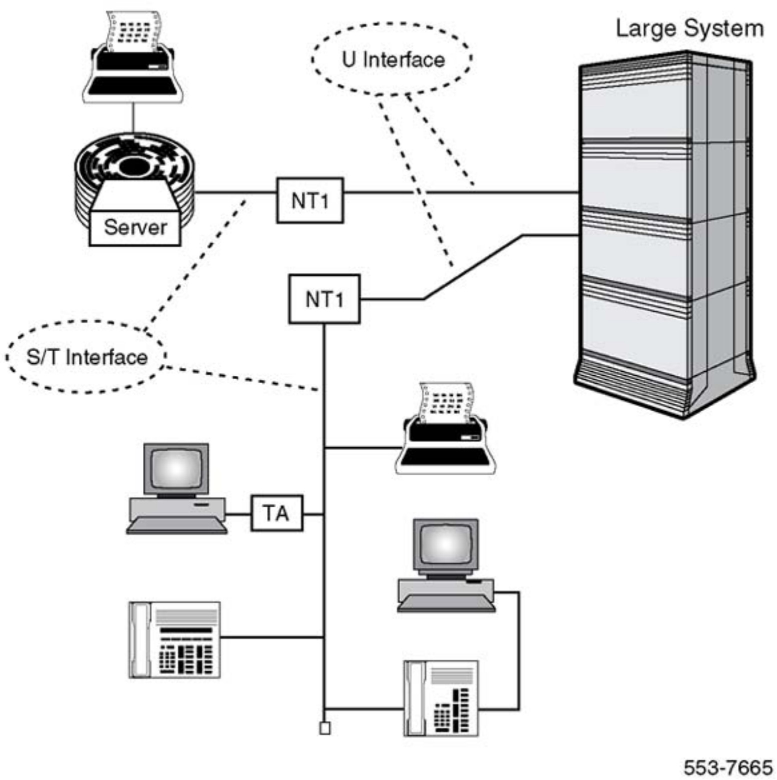


Figure 3: ISDN Basic Rate Interface U interface configured as a line

1TR6 terminal connectivity

1TR6 terminals can be connected to the system using a protocol adapter, which has been specifically designed to interface with ISDN BRI and 1TR6 terminals. Its main function is to convert the 1TR6 protocol sent from the 1TR6 ISDN Terminal Equipment into the European Telecommunication Standard Institute (ETSI) protocol required for ISDN BRI, and vice versa. This conversion is necessary because the layer 3 requirements for 1TR6 and ETSI are different.

Figure 4: An example of an ISDN BRI/1TR6 terminal connectivity on page 56 shows a 1TR6 ISDN BRI terminal connected to the protocol adapter, which is used to access the system through an S/T interface.

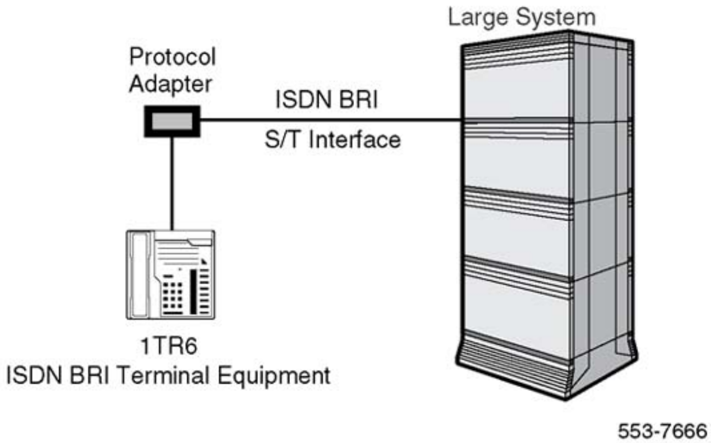


Figure 4: An example of an ISDN BRI/1TR6 terminal connectivity

ISDN BRI packet data transmission

Packet switching differs from circuit switching in that the content of the call is switched rather than the call itself. The message that is being transmitted is broken down into packets that are then sent to their destination through the fastest route.

ISDN BRI on the system supports both B-channel and D-channel packet data transmission. Packet data transmission is done using either an external packet handler, via Avaya Data Packet Network (DPN-100), or the integrated Meridian 1 Packet Handler (MPH).

Packet data transmission using the DPN-100

B-channel packet data transmission

The B-channel packet data from each DSL is transmitted over dedicated connections from the SILC or UILC card to a PRI card, and then over PRI B-channels to the external packet data handler (DPN-100).

The B-channels on a DSL are dedicated to packet data transmission during ISDN BRI service configuration. The number of B-channel connections is limited to the number of available ISDN PRI channels.

D-channel packet data transmission

D-channel packet data from each DSL is transmitted to the MISP or BRSC for separation. The MISP or BRSC separates the packet data from signaling and transmits the packet data to the ISDN PRI. From the ISDN PRI, the data is transmitted to the external packet handler over 64 kbps clear channels.

If the system is to uniquely identify the transmitted and received D-channel packet data for each terminal on a DSL, the system must use an internal identification number. This identification number is called Logical Terminal Identifier (LTID), which must be used together with the Logical Terminal End-point Identifier (LTEI) number during ISDN BRI configuration to uniquely define a logical terminal on a DSL. The LTEI is configured in Overlay 27.

D-channel packet data service is determined separately for each MISP or BRSC during ISDN BRI service configuration.

Figure 5: Packet data transmission using the external packet handler on page 58 illustrates the Meridian 1 external packet handling flow diagram, showing the B-channel and the D-channel packet data routes starting at the line cards and arriving at the external packet handler through dedicated B-channels and dedicated D-channels. Figure 6: Packet data transmission using the external packet handler configured with a BRSC on page 59 presents the same concept, with the addition of a BRSC.

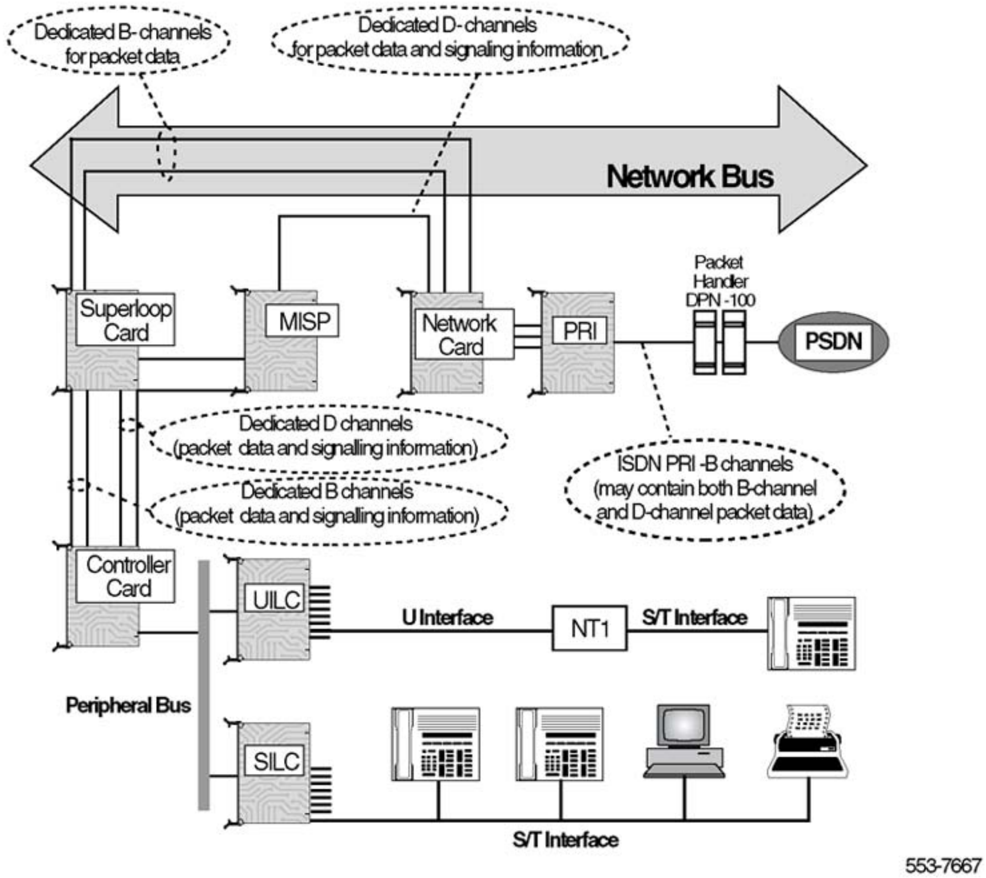


Figure 5: Packet data transmission using the external packet handler

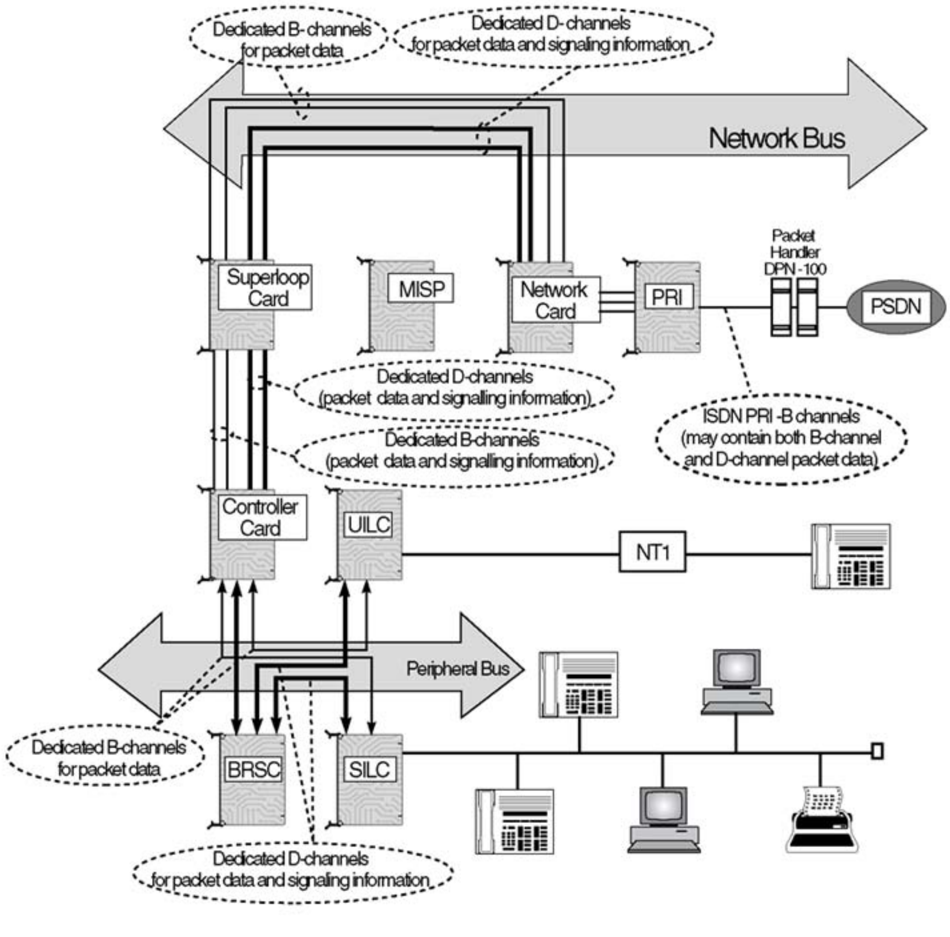


Figure 6: Packet data transmission using the external packet handler configured with a BRSC

Packet data transmission using the Meridian 1 Packet Handler

The Meridian 1 Packet Handler (MPH) uses a dedicated MISP as a hardware platform to run the packet handler application. Routing from the MPH to the Packet Switched Data Network PSDN is through dedicated connections either from a Meridian Communication Unit (MCU) data module and a synchronous modem, or from a dedicated ISDN PRI B-channel (64 kbps clear).

The MPH supports both B-channel and D-channel packet data transmission.

D-channel packet data transmission with the MPH

The MISP line application or the BRSC(s) separate the D-channel packet data from signaling and transmit the data packets on a dedicated PRI D-channel over its network connection to the MPH.

When using an MPH, the Digital Subscriber Loop (DSL) and the Terminal Endpoint Identifier (TEI) are used to identify the Logical Terminal instead of the Logical Terminal Identifiers (LTIDs) which are used by the DPN-100 configuration.

The TEI uniquely identifies to the MPH the transmitted and received packet data for each terminal on a DSL. The TEI number is entered during ISDN BRI configuration to uniquely define a logical terminal on a DSL for the MISP. To configure a terminal for D-channel packet data service, a specific TEI is assigned to an unused static TEI and this information is sent to the MISP.

D-channel packet data service is determined separately for each MISP or BRSC during ISDN BRI configuration. When this data transmission method is selected during system configuration, the user is prompted to enter the MPH loop number to specify the dedicated connection to the MPH. The system CPU sets up the dedicated D-channel to the MPH and informs the MISP or BRSC on which network connection the dedicated D-channel is located.

B-channel packet data transmission with the MPH

B-channel packet data is routed to the MPH directly through dedicated connections. The MPH routes B-channel packet data to the Packet Switched Data Network (PSDN) by means of dedicated channels through either the PRI B-channel (64 kbps clear) or through a Meridian Communication Unit (MCU) with a synchronous modem.

The B-channel packet data from each DSL is routed to the Controller and Superloop network cards. From the Superloop network card, the data is sent to the MPH.

The B-channels on a DSL are dedicated to the packet data transmission by assigning the internal packet data call type to one or more B channels on one or more DSLs during ISDN BRI configuration. These dedicated channels cannot be released by the B-channel packet data terminal. The user is prompted to enter the MPH loop and channel number to specify the dedicated connection to the MPH.

Packet data transmission between the MPH and the PSDN

If using an MPH with an ISDN PRI loop, configure the ISDN PRI loop (LD 17), define an ISDN customer (LD 15), define a TIE trunk route for packet data (LD16), and define a TIE trunk for packet data (LD 14). Then, configure the MISP for an MPH (LD 27).

If using an MPH with an MCU data module, define a TIE trunk route for packet data (LD 16), define a TIE trunk for packet data (LD 14), and configure the MCU (LD 11). Then, configure the MISP for an MPH (LD 27).

Figure 7: Packet data transmission using the MPH on page 61 illustrates packet data transmission for ISDN BRI line application, using the MPH.

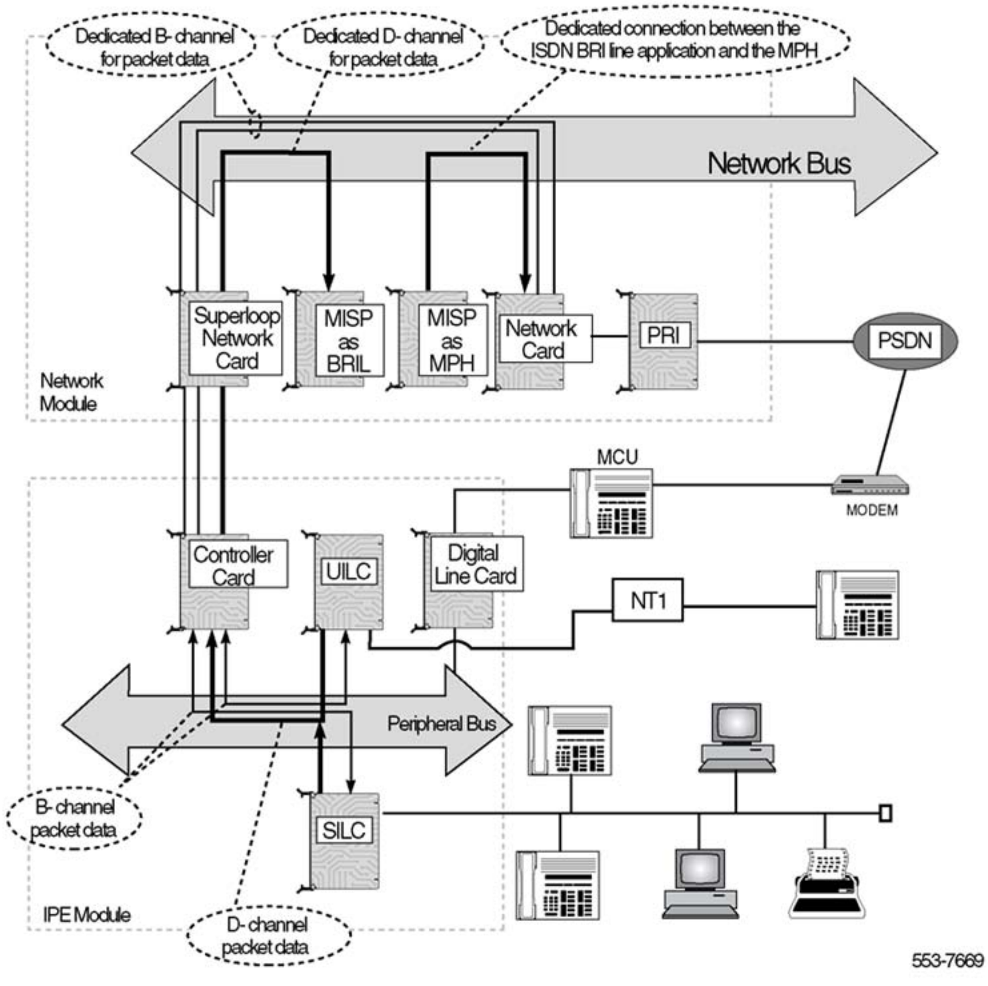


Figure 7: Packet data transmission using the MPH

Figure 8: Packet data transmission using the MPH configured with a BRSC on page 62 illustrates packet data transmission for an ISDN BRI line application, using the MPH and configured with a BRSC.

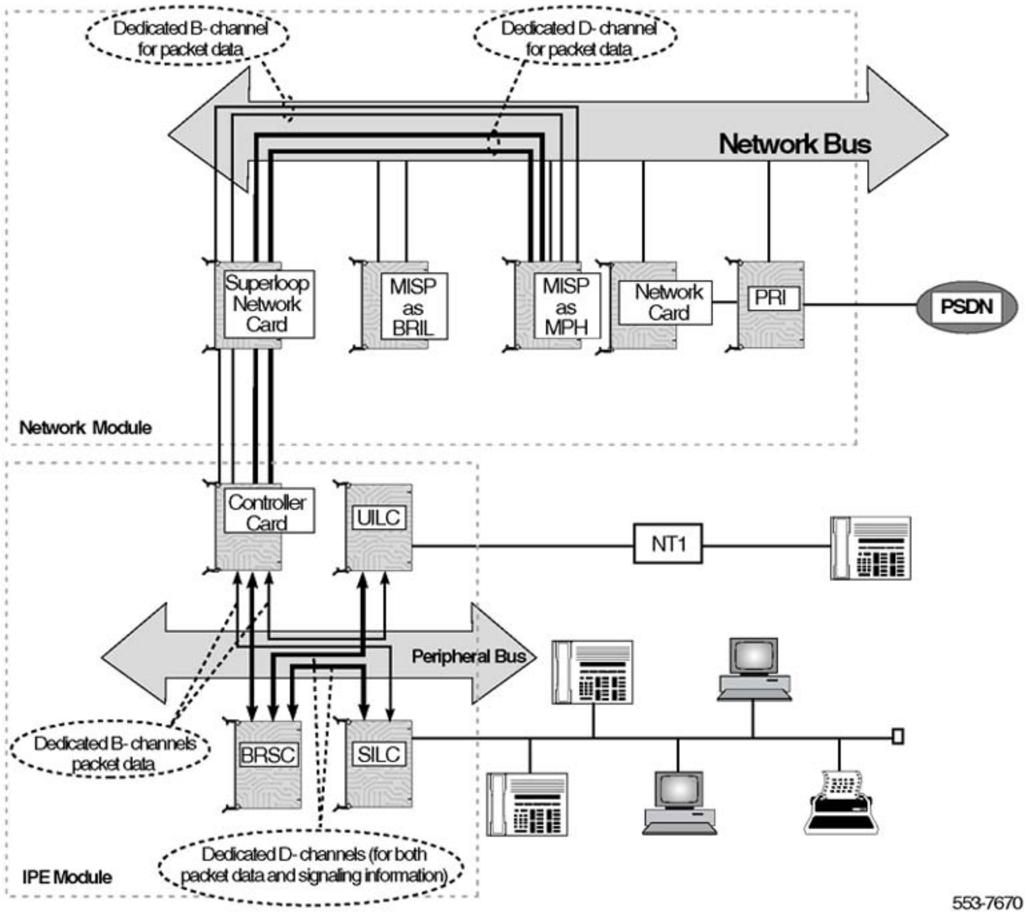


Figure 8: Packet data transmission using the MPH configured with a BRSC

ISDN BRI trunk access

Note:

ISDN BRI trunk access is not supported in North America.

ISDN BRI trunks can be configured for either local exchange/CO/DID trunk connectivity, MCDN TIE trunk connectivity, or QSIG trunk connectivity. ISDN BRI trunk connections are configured on a per-DSL basis; that is, DSL trunk connections can be configured on any given DSL for any SILC or UILC. The trunks can be accessed using both ISDN BRI terminals and non-ISDN BRI devices (such as digital and analog telephones).

ISDN BRI Local exchange CO/DID connectivity is accomplished via a MISP card and an S/T interface, using the SILC line card. This connectivity is supported for Numeris VN3, 1TR6, ETSI NET-3 and ETS 300 403 (EuroISDN), INS NET-64 (Japan D70), Australia ETSI, and Asia-Pacific protocols. Refer to Figure 9: ISDN BRI trunk access for local exchange connectivity on page 64.

ISDN BRI MCDN TIE trunk connectivity is achieved via a MISP card and either S/T or U interfaces, using the SILC and UILC line cards respectively. This connectivity can be:

- between two PBXs through a local exchange acting as a passive facility; the local exchange must support Numeris VN3, 1TR6, ETSI NET-3 and ETS 300 403 (EuroISDN), INS NET-64 (Japan D70), Australia ETSI, or Asia-Pacific protocols; refer to Figure 10: ISDN BRI trunk access TIE trunk connectivity - first configuration on page 70.
- directly between two PBXs; refer to Figure 11: ISDN BRI TIE trunk connectivity - second configuration on page 71 and Figure 12: ISDN BRI TIE trunk connectivity - third configuration, as described above on page 72.

ISDN BRI QSIG connectivity is achieved through an MISP card and either S/T or U interfaces, using the SILC and UILC line cards respectively. This connectivity is supported within a Private Telecommunications Network (PTN) between two Private Telecommunications Network Exchanges (PTNXs). Examples can be a Centrex-to-Centrex connection, or a Centrex-to- PBX connection. Refer to Figure 13: QSIG ISDN BRI trunk connectivity on page 73 in the section ISDN BRI QSIG connectivity on page 72.

Note:

The BRSC cannot be utilized for ISDN BRI trunk access.

These configurations are explained in greater detail in the sections that follow.

ISDN BRI Local Exchange/CO/DID connectivity

ISDN BRI Local Exchange connectivity is supported in markets equipped with local exchanges that conform to Numeris VN3, 1TR6, ETSI NET-3 and ETS 300 403 (EuroISDN), INS NET-64 (Japan D70), Australia ETSI, or Asia-Pacific protocols.

Figure 9: ISDN BRI trunk access for local exchange connectivity on page 64 illustrates the ISDN BRI Local Exchange connectivity. The ISDN BRI Local Exchange DSL is connected to a Network Termination (NT1) device, which is physically located on the same premises as the system. The NT1 device connects to the Local Exchange via a U interface. (The NT1 device is typically owned by the Local Exchange/Post Telegraph and Telephone allowing the Local Exchange/PTT to use any type of U interface, including proprietary implementations). The distance limitation of the NT1 from the Local Exchange depends on the distance supported by the Local Exchange.

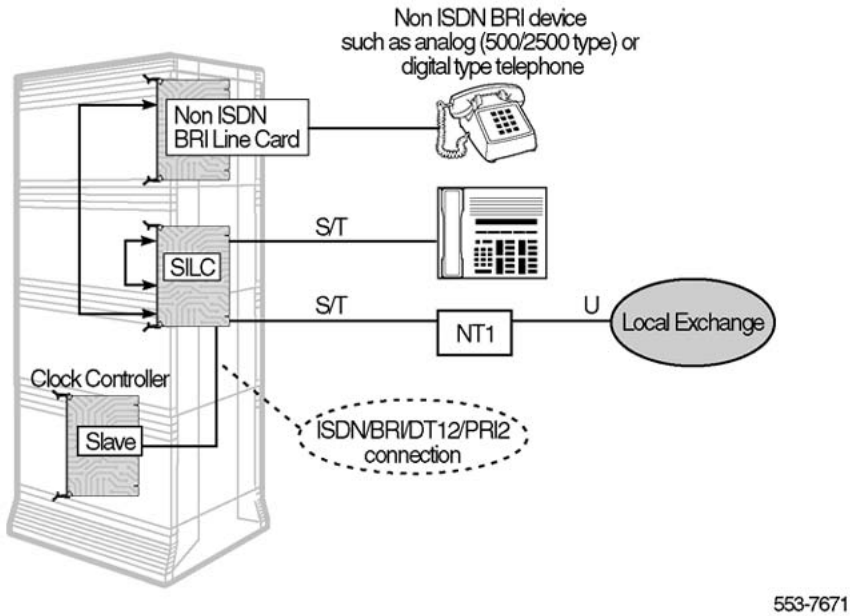


Figure 9: ISDN BRI trunk access for local exchange connectivity

1TR6 local exchange connectivity

1TR6 local exchange connectivity provides 2B+D connectivity to a local exchange that supports 1TR6 protocol via an S/T interface. The ISDN BRI 1TR6 local exchange connectivity provides the following basic call and supplementary services:

Note:

Support for any feature is dependent upon the terminal equipment being used.

- Basic call service
- Circuit switched voice and data on the B-channel
- Calling Line Identification Presentation and Restriction
- Connected Number Delivery
- Support for TIE, COT, DID, DOD trunk types
- Channel negotiation

Note:

In cases where several ISDN BRI trunks (and hence several DSLs) are configured on a route, if Channel Negotiation fails to yield an acceptable channel on any of these DSLs, it is not possible to use another channel on another DSL.

- Overlap sending
- Flexible Numbering Plan
- Indication of Call Charging to the calling party
- Network-wide interworking with ISDN BRI ETSI terminals

Numeris VN3 local exchange connectivity

The Numeris local exchange connectivity provides 2B+D connectivity through an S/T interface to a local exchange that supports Numeris VN3 protocol. The ISDN BRI/Numeris VN3 local exchange connectivity provides the following basic call and supplementary services:

- Basic call service
- Circuit switched voice and data on the B-channel
- Called/calling party subaddress (network-wide)
- Support for TIE, COT, DID, DOD trunk types
- Channel negotiation
- In cases where several ISDN BRI trunks (and hence several DSLs) are configured on a route, if Channel Negotiation fails to yield an acceptable channel on any of these DSLs, it is not possible to use another channel on another DSL.
- 64 kbps clear bearer capability
- Flexible Numbering Plan
- Advice of charge during call and at end of call
- Network-wide interworking with ISDN BRI Numeris terminals

Japan D70 (INS NET-64) local exchange connectivity (non-Asia Pacific protocol)

The Japan D70 (INS NET-64) local exchange connectivity (non-Asia Pacific protocol) provides 2B+D connectivity through an S/T interface to a local exchange that supports the D70 protocol (D70 is the Japanese version of the INS NET-64 protocol).The ISDN BRI/Japan D70 local exchange connectivity provides the following basic call and supplementary services:

- Basic call service
- Circuit switched voice and data on the B-channel
- Called/calling party subaddress (network-wide)
- Support for TIE, COT, DID, DOD trunk types
- 64 kbps clear bearer capability
- Flexible Numbering Plan

- Advice of charge at end of call
- Channel Negotiation

Note:

In cases where several ISDN BRI trunks (and hence several DSLs) are configured on a route, if Channel Negotiation fails to yield an acceptable channel on any of these DSLs, it is not possible to use another channel on another DSL.

EuroISDN connectivity

The EuroISDN connectivity provides an interface between PBXs and Central Offices/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, Belgium, Commonwealth of Independent States (Russia and the Ukraine), Denmark, Finland, Germany, Holland, Ireland, Italy, Norway, Portugal, Spain, Sweden, Switzerland, and the United Kingdom.

The system on the EuroISDN connectivity provides the following call services, for the complying markets:

- Basic call service
- Circuit switched voice and data on the B Channel
- Calling Line Identification Presentation and Restriction (CLIP and CLIR)
- Connected Line Presentation and Restriction
- Calling and connected sub-addresses
- Support for TIE, COT, DID, and DOD trunk types
- Overlap sending and receiving
- Overlap and enbloc dialing
- Flexible Numbering Plan
- Channel negotiation

Note:

In cases where several ISDN BRI trunks (and hence several DSLs) are configured on a route, if Channel Negotiation fails to yield an acceptable channel on any of these DSLs, it is not possible to use another channel on another DSL.

Asia-Pacific connectivity

The ISDN PRI to Asia Pacific connectivity provides ISDN Primary Rate Interface (PRI) connectivity between the system 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); this also applies 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, Channel Negotiation is not supported for Singapore. The CNEG option must be set to 1 (the default) in LD 17.

Note:

The Asia Pacific interface does not support the Meridian 1 Packet Handler (MPH) across the CO.

Clock synchronization, automatic switching and recovery for ISDN BRI to local exchange connectivity

System clock synchronization can be achieved by having the system slave to the local exchange; the clock source can be derived either from the ISDN BRI Local Exchange connection or from other ISDN BRI/PRI/DTI local exchange connections if available.

The clock source is input to the Clock Controller card on the system, and the system clock is then synchronized with the network clock. This functionality is shown in Figure 10: ISDN BRI trunk access TIE trunk connectivity - first configuration on page 70.

Refer to *Avaya ISDN Primary Rate Interface Features Fundamentals, NN43001-318* for procedures on how to provide clock references to the Clock Controller in an ISDN BRI local exchange connectivity.

Automatic switching

The S/T software cannot detect misframes, loss of signal, and bit error rates, conditions that would cause automatic switching of the clock source; rather, the S/T relies on the clock controller to provide feedback on the quality of the clock, and performs automatic switching and recovery as appropriate. If a clock controller error is detected, the system switches to the backup clock controller, without affecting the reference clock that is being tracked.

If the SILC DSL with clock reference is disabled, and re-enabled, clock tracking is restored as follows:

- If DSL #0 has been assigned as the primary reference clock, but the clock controller is tracking on the secondary reference or is in free run mode, the clock is restored to tracking on primary.
- If DSL #1 has been assigned as the secondary reference clock, but the clock controller is in free run mode, the clock is restored to tracking on secondary.

Tracking on the primary or secondary reference clock is automatically switched as follows:

- If the system software is unable to track on the assigned primary reference clock, it switches to the secondary reference clock.
- If the system software is unable to track on the assigned secondary reference clock, it switches to free run mode.

Clock recovery

The SILC is configured in the slave-slave mode when acting as a trunk interface. This is set up through the Maintenance Signaling Channel (MSC). The microcontroller configures the S/T chips on the SILC as appropriate.

Automatic clock recovery is done upon the expiration of the free run guard timer. Tracking is restored to the primary reference clock, if defined. If the primary reference clock is disabled, tracking is restored to the secondary reference clock, if defined.

T0 (2B+D) backup for T2 (30B+D)

Note:

Japan D70 connectivity uses T0 (2B+D) for T1 (23B+D) backup.

When configuring clock synchronization, ISDN BRI trunks can be configured as backup for ISDN PRI trunks using either ESN Route Selection, or Route Hunting. Clock synchronization can be set up with the ISDN BRI clock source configured as a secondary clock reference for an existing ISDN PRI clock source.

Although the reverse is supported; that is, the ISDN BRI clock source can be configured as the primary clock reference with the ISDN PRI clock source as the secondary reference, this configuration is not recommended because an ISDN PRI clock source is more reliable than an ISDN BRI clock source.

ISDN BRI MCDN TIE trunk connectivity

MCDN ISDN BRI TIE trunk connectivity can have three implementations. In the first configuration (refer to Figure 10: ISDN BRI trunk access TIE trunk connectivity - first configuration on page 70), a Meridian Customer Defined Networking (MCDN) TIE trunk connection can be implemented by connecting two systems to the ISDN BRI leased line through the local exchange via two SILC cards. The S/T interface is connected to the local exchange using the NT1 supplied by the PTT.

There is no distance limitation on this configuration. System clock synchronization can be achieved by having the system slave to the local exchange; the clock source can be derived either from the ISDN BRI local exchange connections or from other ISDN BRI/PRI/DTI local exchange connections if available.

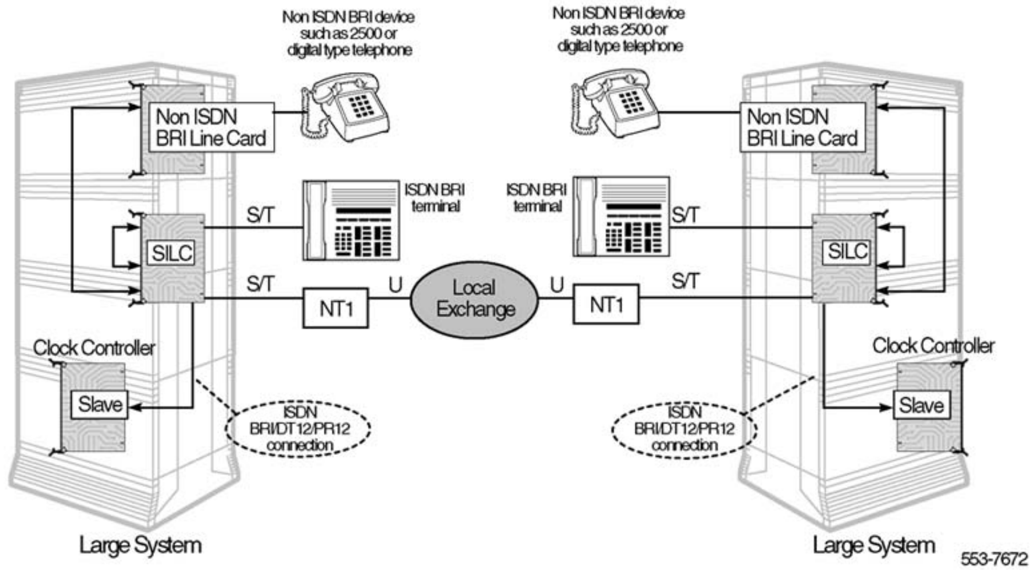


Figure 10: ISDN BRI trunk access TIE trunk connectivity - first configuration

In the second configuration (Figure 11: ISDN BRI TIE trunk connectivity - second configuration on page 71), an MCDN TIE trunk connection can be achieved by connecting two systems through an NT1 device. With this configuration, there is a distance limitation of 6.5 km (3.9 miles), without any signal amplification device.

System clock synchronization can be achieved by having the system, equipped with the SILC, derive clock reference from the ISDN BRI TIE trunk connection or from other ISDN BRI/PRI/DTI connections if available. The system equipped with the UILC interface can be allowed to operate in free-run mode or derive the clock source from other ISDN BRI/PRI/DTI connections if available.

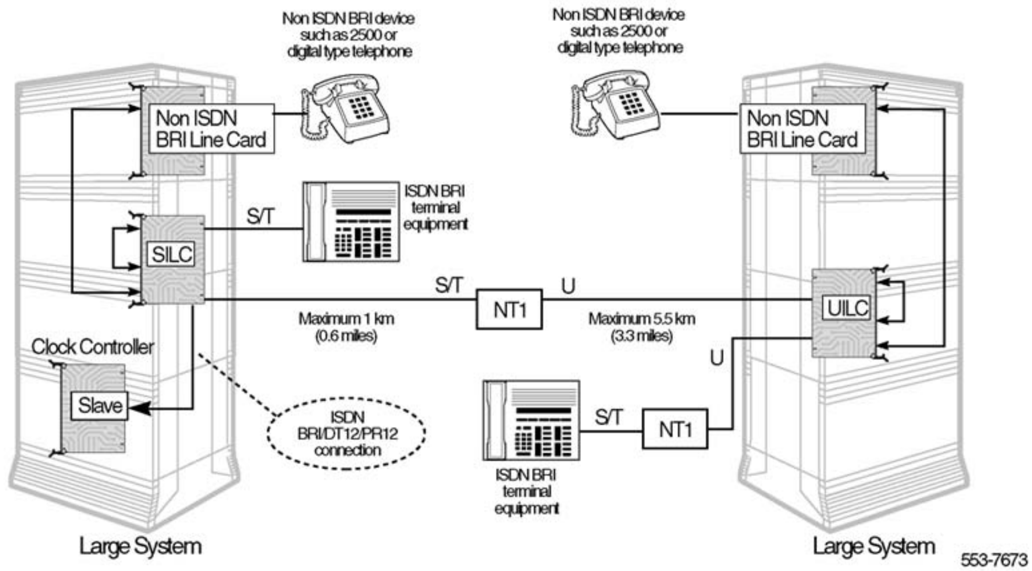


Figure 11: ISDN BRI TIE trunk connectivity - second configuration

The third configuration (refer to Figure 12: ISDN BRI TIE trunk connectivity - third configuration, as described above on page 72), although not recommended because of the lack of protection devices and because of the distance limitation of 1 km (0.6 mile), can establish a MCDN TIE trunk link by connecting two systems through a direct line between two back-to-back SILC interfaces.

This configuration would be applied only to multiple buildings on a contiguous property with no exposed cable (with the 1 km limitation between buildings). System clock synchronization can be achieved by having one of the systems derive clock reference from the ISDN BRI TIE trunk connection or from other ISDN BRI/PRI/DTI connections if available. The other system can be allowed to operate in free-run mode or derive the clock source from other ISDN BRI/PRI/DTI connections if available.

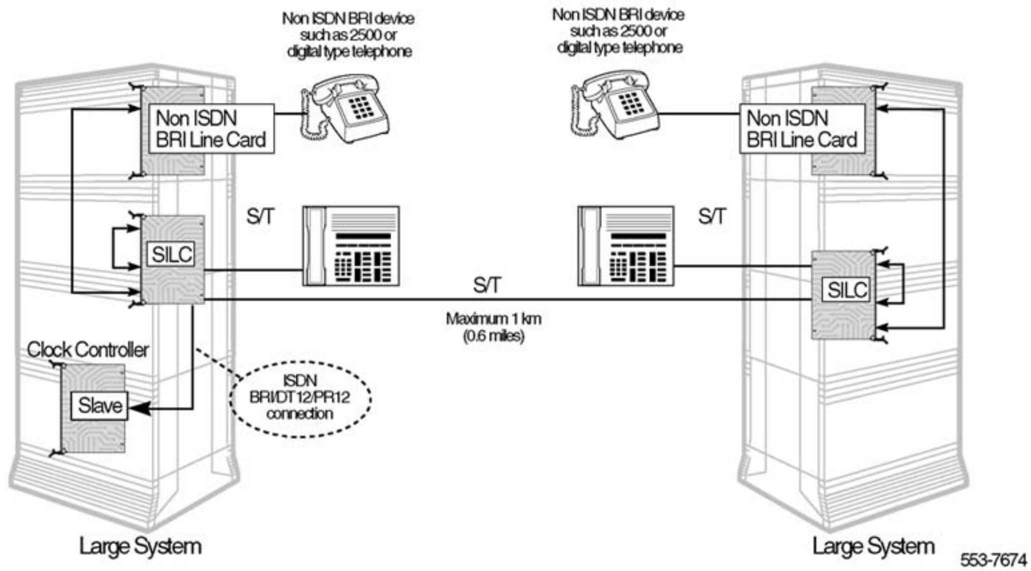


Figure 12: ISDN BRI TIE trunk connectivity - third configuration, as described above

ISDN BRI QSIG connectivity

The European Computer Manufacturers' 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 global PTNX manufacturers supports ISDN BRI (as well as ISDN PRI) connectivity based on the ISDN QSIG (ETSI and ISO) standard.

QSIG is oriented towards signaling and services that occur between peer-to-peer connectivity, that is, between two PBXs, between two Centrex, or between a PBX and a Centrex. The signaling for services would be exchanged across a "Q" reference point. Figure 13: QSIG ISDN BRI trunk connectivity on page 73 illustrates an example of QSIG trunk connectivity. For ISDN BRI, the QSIG interface provides the following capabilities:

Compliant Multi-vendor PBX/Centrex Private ISDN interworking (connectivity between the Private ISDN PBXs can be through PRI or ISDN BRI trunks)

- ETSI or ISO version of basic call service
- 64 kbps clear data
- Overlap Sending/Receiving
- Channel Negotiation

- In cases where several ISDN BRI trunks (and hence several DSLs) are configured on a route, if Channel Negotiation fails to yield an acceptable channel on any of these DSLs, it is not possible to use another channel on another DSL.
- Calling Line Identification Presentation (CLIP)
- Calling Line Identification Restriction (CLIR)
- Connected Line Identification Presentation (COLP)
- Connected Line Identification Restriction (COLR)
- Flexible Numbering Plan
- Support for TIE trunk call types
- Transit Count information transmitted when ISDN Call Connection Limitation (ICCL) is present (supported for ETSI QSIG only)
- Party Category (partially supported on ETSI QSIG)

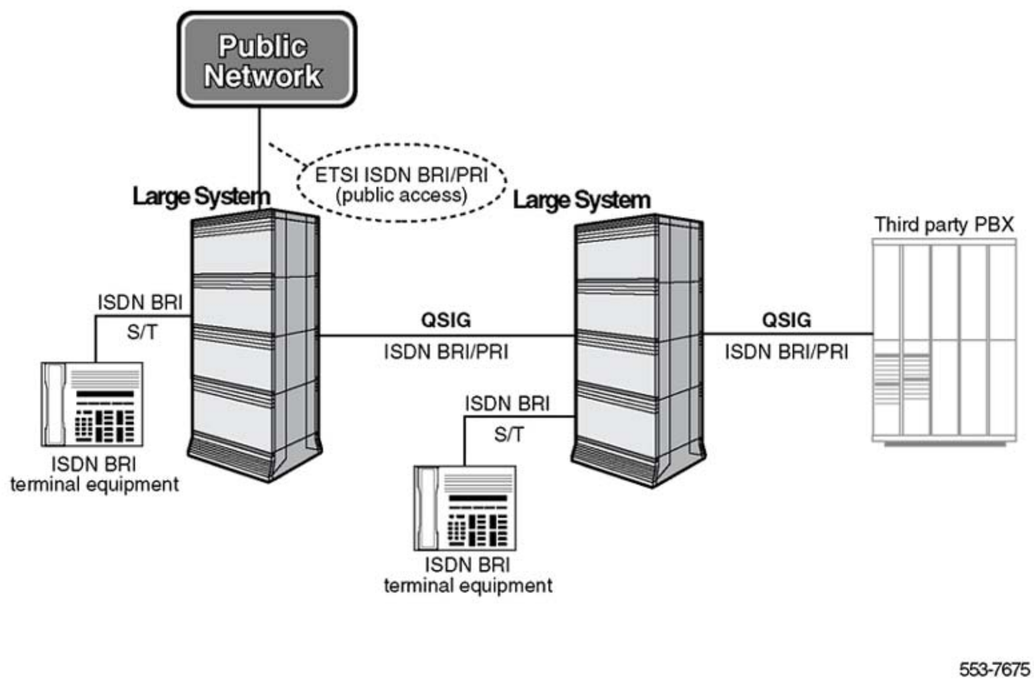


Figure 13: QSIG ISDN BRI trunk connectivity

Chapter 5: Engineering guidelines

Contents

This section contains information on the following topics:

Compatible ISDN BRI terminals on page 126

Introduction

This chapter describes engineering guidelines that should be used to configure an ISDN BRI system. It describes hardware requirements, system capacity, configuration guidelines, Digital Subscriber Loop transmission characteristics, and interface specifications. For further information about general engineering guidelines, refer to the following documents:

- *NN43021-220;*
- *NN43011-220;*

Hardware requirements

Hardware requirements for ISDN BRI line, packet handling, and trunk applications are as follows (please note that a functional description of each component will follow):

Note:
The Mean Time Between Failures (MTF) is given as appropriate.

Line application

- MISP circuit card - NT6D73; MTF = 29 years and NTBK22; MTF = 29 years
- Basic Rate Signaling Concentrator (BRSC) card (optional) - NT6D72; MTF = 17 years
- SILC circuit card - NT6D70AA (-48V North American) MTF = 47.3 years or NT6D70BA (-40V International); MTF = 47.3 years
- UILC circuit card— NT6D71 (ANSI 2B1Q line encoding); MTF = 46.8 years
- Terminating resistor - A0378866
- ISDN BRI terminals
  - M5317TDX - Meridian 1 proprietary set equipped with voice and data transmission options and a hands-free feature; supports B-channel and D-channel packet data.
  - M5209TDcp - Meridian 1 proprietary set equipped with voice and data transmission options and a hands-free feature; supports B-channel and D-channel packet data.
  - Other terminals (any other terminal deemed compatible by Avaya)

- ISDN Terminal Adapter - M5000TD-1, required if connecting non-BRI terminals to the ISDN BRI line interface
- Network Termination 1 (NT1) Needed when conversion from a U to an S/T interface is required.

Figure 14: ISDN BRI line application architecture, configured with a BRSC on page 77 illustrates a basic ISDN BRI line application architecture, configured with a BRSC.

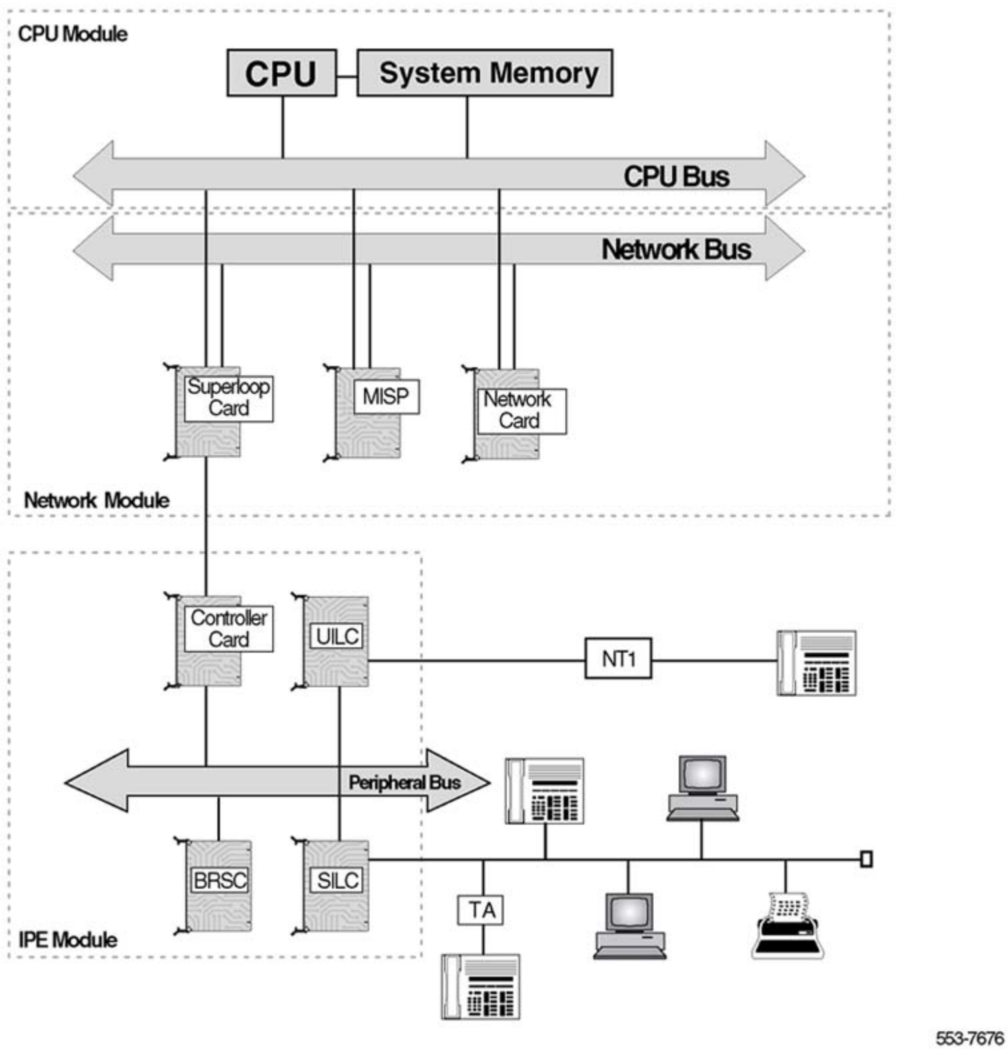


Figure 14: ISDN BRI line application architecture, configured with a BRSC

Packet handling

External packet handler (DPN-100)

- MISP circuit card - NT6D73; MTF = 29 years.NTBK22; MTF = 29 years
- Basic Rate Signaling Concentrator (BRSC) card (optional) - NT6D72; MTF = 17 years
- SILC circuit card - NT6D70AA (-48V North American) or NT6D70BA (-40V International); MTF = 47.3 years
- UILC circuit card - NT6D71 (ANSI 2B1Q line encoding); MTF = 46.8 years
- Terminating resistor - A0378866
- 1.5 PRI circuit card , the QPC720 or the dual-port NT5D12, NTAK09 cards.
- PRI2 circuit - the NT8D72, or NT5D97 dual-port DT12/PR12 , the NTAK79 or the NTBK50 cards.
- DPN-100 External Packet Handler
- ISDN BRI terminals
    - M5317TDX - Digital set equipped with voice and data transmission options and a hands-free feature; supports B-channel and D-channel packet data.
    - M5209TDcp - Digital set equipped with voice and data transmission options and a hands-free feature; supports B-channel and D-channel packet data.
    - Other terminals (any other terminal deemed compatible by Avaya)
- ISDN Terminal Adapter - M5000TD-1, required if connecting non-BRI terminals to the ISDN BRI line interface.
- Network Termination 1 (NT1) Needed when conversion from a U to an S/T interface is required.

Figure 15: A basic ISDN BRI DPN-100 packet data architecture, with a BRSC on page 79 illustrates a basic ISDN BRI DPN-100 packet data architecture, with a BRSC.

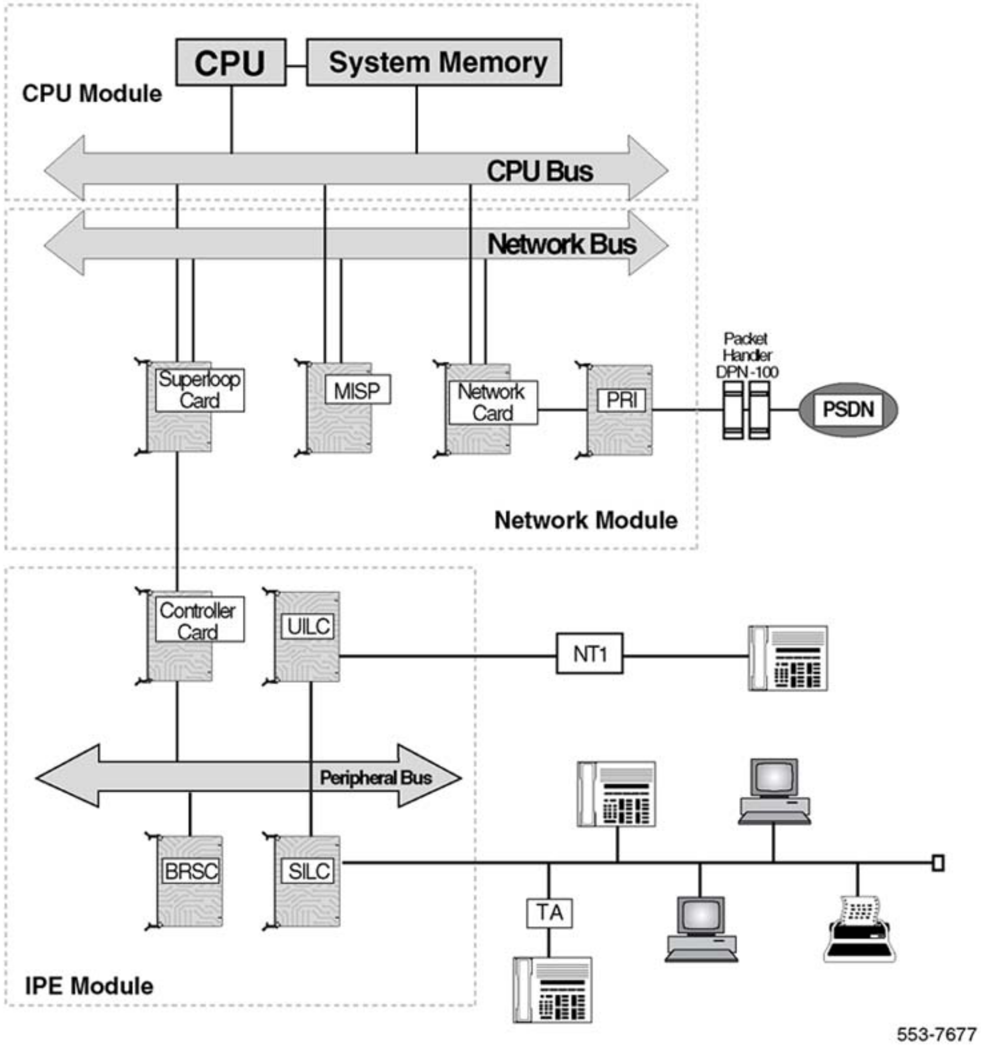


Figure 15: A basic ISDN BRI DPN-100 packet data architecture, with a BRSC

Meridian 1 Packet Handler (MPH)

- MISP circuit card - NT6D73; MTF = 29 years
- Basic Rate Signaling Concentrator (BRSC) card (optional) - NT6D72; MTF = 17 years
- SILC circuit card - NT6D70AA (-48V North American) or NT6D70BA (-40V International); MTF = 47.3 years
- UILC circuit card— NT6D71 (ANSI 2B1Q line encoding); MTF = 46.8 years
- Terminating resistor - A0378866
- 1.5 PRI circuit card - QPC720 or the dual-port NT5D12

- PRI2 circuit card - NT8D72, or NT5D97 dual-port DT12/PR12 card

  (the use of a PRI card is optional, used if a PRI channel is used to access the PSDN instead of an MCU data module)

  or
- Meridian Communication Unit (MCU) (optional, used if an MCU data module is used to access the PSDN instead of a PRI channel)
- modem or Digital Interface Unit (optional, required if an MCU is used)
- Meridian 1 Packet Handler (MPH) (downloadable to the MISP)
- ISDN BRI terminals
    - M5317TDX - Digital set equipped with voice and data transmission options and a hands-free feature; supports B-channel and D-channel packet data.
    - M5209TDcp - Digital set equipped with voice and data transmission options and a hands-free feature; supports B-channel and D-channel packet data.
    - Other terminals (any other terminal deemed compatible by Avaya)
- ISDN Terminal Adapter - M5000TD-1, required if connecting non-BRI terminals to the ISDN BRI line interface
- Network Termination 1 (NT1) Needed when conversion from a U to an S/T interface is required.

Figure 16: A basic ISDN BRI MPH packet data architecture, with a BRSC and an MCU on page 81 illustrates a basic MPH packet data architecture, with a BRSC and an MCU.

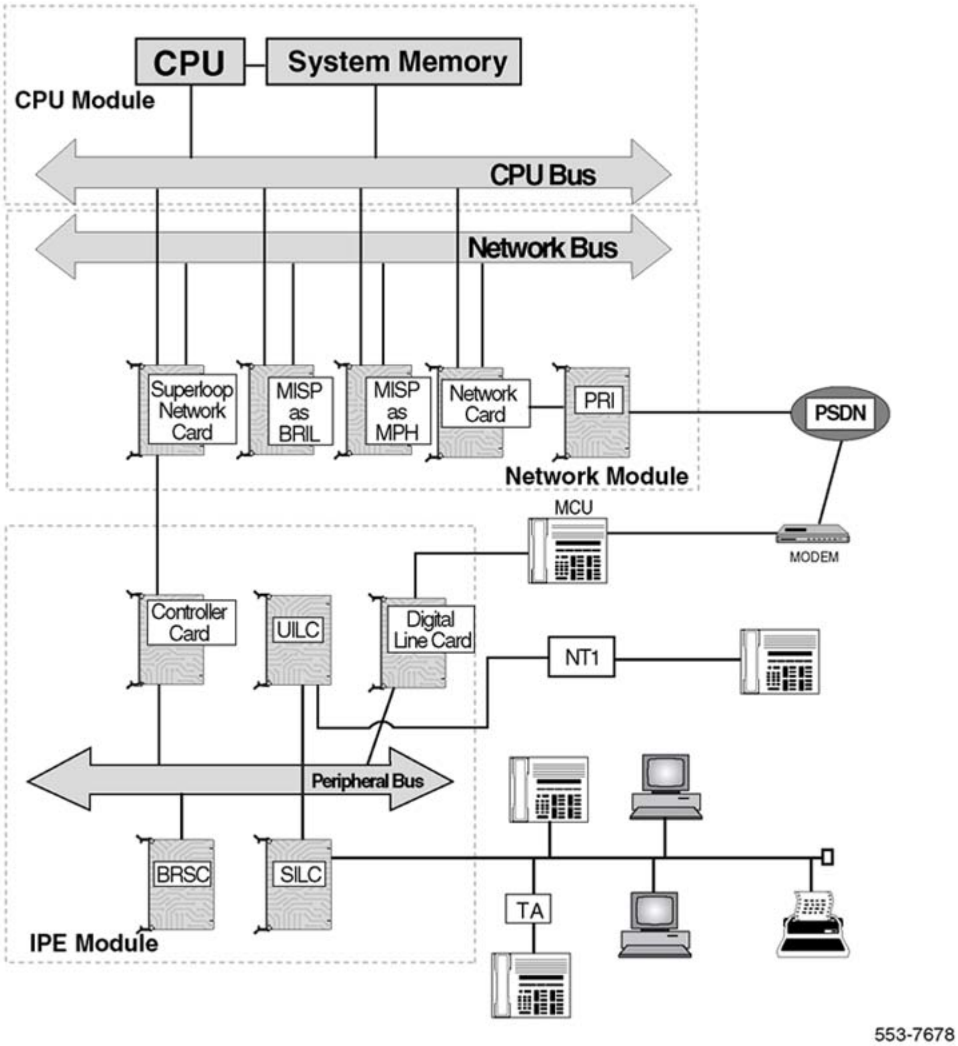


Figure 16: A basic ISDN BRI MPH packet data architecture, with a BRSC and an MCU

Trunk application

- MISP circuit card - NT6D73; MTF = 29 years. NTBK22; MTF = 29 years.
- SILC circuit cardfor CO/TIE connectivity; IPE card; NT6D70AA (-48V North American) or NT6D70BA (-40V International); MTF = 47.3 years
- UILC circuit card for TIE connectivity; NT6D71 (ANSI 2B1Q line encoding); MTF = 46.8 years
- Terminating resistor - A0378866
- Clock Controller - the NTRB53 resides in the Network shelf; required if the clock source is to be drawn from DSL0 or DSL1 of the SILC; DSL0 can only be configured as the

primary source, while DSL1 can only be configured as the secondary source. The CC connects to the SILC clock port via cables)

Note:

NTRB53 is required for EuroISDN and Numeris VN2 applications.

- NTAK20AB (Stratum 3 CC daughter board), or the NTAK20BB (Stratum 4 CC daughter board).
- Clock Controller reference cables - NTD70, NTND71, NTND72
- ISDN BRI terminals
    - M5317TDX - Digital set equipped with voice and data transmission options and a hands-free feature; supports B-channel and D-channel packet data.
    - M5209TDcp - Digital set equipped with voice and data transmission options and a hands-free feature; supports B-channel and D-channel packet data.
    - Other terminals (any other terminal deemed compatible by Avaya)
- ISDN Terminal Adapter - M5000TD-1, required if connecting non-BRI terminals to the ISDN BRI line interface
- Network Termination 1 (NT1) - needed for conversion from a U to an S/T interface

Figure 17: A basic ISDN BRI trunking architecture (CO connectivity) on page 83 illustrates a basic ISDN BRI trunking architecture, with the system connecting to a CO that supports a Numeris VN3, 1TR6, ETSI NET-3 (EuroISDN), INS NET-64 (including Japan D70), Australia ETSI, or Asia-Pacific protocol.

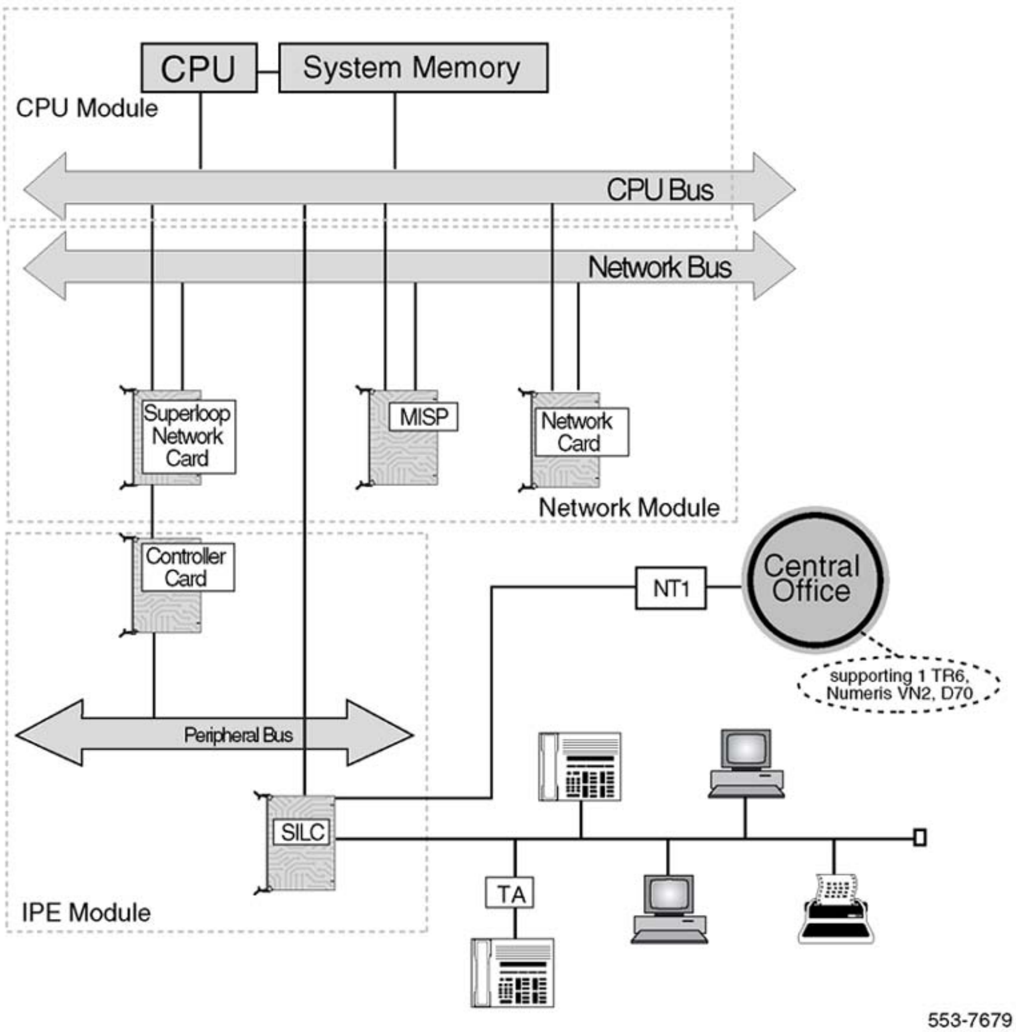


Figure 17: A basic ISDN BRI trunking architecture (CO connectivity)

Hardware functional descriptions

NT6D72 Basic Rate Signaling Concentrator Card (BRSC)

The Basic Rate Signaling Concentrator (BRSC) Card can be used to process data link layer signaling messages from all ISDN BRI line cards and send the resulting network layer messages to the MISP. The BRSC also filters out D-channel Packet Switched Data (DSPD) from signaling information and routes it to the packet handler.

With a BRSC configured, fewer MISPs are needed for the same number of DSLs. Each BRSC can support a combination of 15 SILCs/UILCs per IPE Module, with a maximum of eight UILCs.

The BRSC can route the packet data from the line cards to a DPN100, the external packet handler, or to the MPH, the internal packet handler.

NTBK22 Multi-Purposes ISDN Signaling Processor (MISP) card.

The MISP card (NTBK22) performs Data Link (Layer 2) and Network (Layer 3) processing associated with the OSI protocol.

Each MISP can support four line cards (UILC or SILC or any combination of the two). Each line card supports eight DSLs, therefore each MISP supports 32 DSLs. Since each DSL uses two B-channels and one D-channel, the MISP supports 64 B-channels and 32 D-channels. If the MISP is carrying packet data, it must dedicate one of its D-channels to communicate with the external packet handler. In this case the MISP support only 31 DSLs. The MISP supports the downloading of ISDN applications from the system software daughter board. The MISP is downloaded with the appropriate application code:

- on the first enabling of the MISP card
- when software is upgraded
- when MISP Applications are added/changed

The applications for the MISP are copied from the software cartridge into RAM on the MISP card. Only the new/different applications are downloaded. This information is then copied into the Flash ROM on the MISP for storage. This process requires approximately 10 minutes to complete and is carried out while the MISP pack is operational. The next time the system or MISP card resets, the application is loaded from the MISP Flash ROM provided there are no new or different applications on the software cartridge.

Use the equation below to calculate the number of MISPs required to control SILCs and UILCs.

(SILC + UILCs) Ö 4 = MISPs If the result is a fraction, round it off to the next highest number.

NT6D73 Multi-Purpose ISDN Signaling Processor (MISP), for Large Systems

The NT6D73 MISP is a microprocessor controlled signaling processor that provides a communication interface between the CPU and the peripheral devices. It utilizes the network and the CPU buses to communicate with the CPU, the SILCs, and the UILCs. Both buses are located on the Network Module backplane.

If a BRSC card is not configured in a system, each MISP can support 32 D-channels and therefore can support 32 DSLs since each DSL has a D-channel. This is true only if the D-channels are not configured to carry packet data. If they are carrying packet data to the DPN-100, the MISP can support only 31 DSLs.

Note:

Without a BRSC, one MISP can support up to four SILCs, which results in a maximum of 32 DSLs. Please note that even though every DSL on each SILC may not be defined, you cannot add another SILC (even though you are within the maximum number of 32 DSLs). This is due to the fact the MISP has two nail-ups for each SILC, and each nail-up controls four DSLs in sequential order. Therefore, there is no space in the MISP block to store more than four SILCs.

A BRSC card provides increased capacity, supporting up to 120 DSLs in the same IPE Module. One MISP can serve up to eight BRSCs and two line cards. This increases DSL capacity for the MISP from 32 to 976. This figure is derived as follows:

1 MISP supports 8 BRSCs and 2 line cards (SILC/UILCs)

1 BRSC supports 15 SILC/UILC cards, each having 8 ports:

- total (8 x 15) = 120

  1 SILC/UILC card has 8 ports

- total (8 x 120) = 960

  2 SILC/UILC cards each has 8 ports

- total (8 x 2) = 16

  Therefore, total number of DSLs = 960+16 = 976.

Each MISP can support directly the following combinations:

- four ISDN BRI line cards without any BRSCs
- three ISDN BRI line cards and one BRSC, or
- two ISDN BRI line cards and a maximum of eight BRSCs

The main functions of the MISP are to:

- communicate with the CPU to report ISDN BRI status and receive downloaded application software and configuration parameters
- execute Open System Interconnect (OSI) data link and network layer protocols
- provides the platform for the Meridian 1 Packet Handler
- process the signaling information received on the D-Channels from DSLs. D-channels may also carry user packet data, which the MISP separates from signaling information and forwards to the external DPN-100 packet handler or internal packet handler (MPH)
- control terminal initialization and addressing

- assign B-channels for switched voice and data transmission
- send call control messages to ISDN BRI links over the D-channel

NT6D70AA/NT6D70BA S/T Interface Line card (SILC)

The SILC cards (NT6D70AA -48V North America, NT6D70BA -40 V International) provide a globally accepted standard interface. The SILC circuit cards support the OSI physical layer (layer 1) protocol.

The SILC provides eight S/T four wire full duplex polarity sensitive interfaces. Each S/T interface provides two B-channels and one D-channel and supports a maximum of eight physical connections that may be configured for ISDN BRI terminals or for CO/TIE trunk connectivity.

For line connectivity, each S/T interface can link up to 20 logical terminals on one DSL. A logical terminal is any terminal that can communicate with the system over a DSL. It may be directly connected to the DSL through its own physical termination or be indirectly connected through a common physical termination.

The length of a DSL depends on the specific terminal configuration and the DSL wire gauge; however, it should not exceed 1 km (0.6 mi.).

The SILC interface uses a 4-conductor cable that provides a differential Transmit and Receive pair for each DSL. The SILC has options to provide a total of 2 Watts of power on the Transmit or Receive leads, or no power at all. When this power is supplied from the S/T interface, the terminal devices must not draw more than the 2 Watts of power. Any power requirements beyond this limit must be locally powered.

A terminating resistor (AO378866) must be placed at the end of each DSL associated with an S/T interface to ensure proper operation.

Other functions of the SILC are:

- support point-to-point and point-to-multi-point DSL terminal connections
- execute instructions received from the MISP to configure and control the S/T interfaces
- provide channel mapping between ISDN BRI format (2B+D) and the system bus format
- enable and disable DSLs
- provide loopback control of DSLs
- for trunking applications, provide a reference clock to the clock controller

SILCs required for non-blocking conditions

Use the equations below to calculate the number of SILCs required to provide interfaces for the S/T type ISDN BRI terminals for non-blocking traffic conditions. To provide a non-blocking traffic condition on a DSL, a maximum of 2 B-channel terminals can be connected.

SILC B-channel terminals Ö 16 = SILCs

Note:

A physical terminal that can use two or more B-channels simultaneously, such as circuit-switched voice and data, should be counted as two terminals for the purpose of this calculation.

SILC D-channel terminals Ö 8 (See Note) = SILCs

Note:

This assumes one D-channel terminal per DSL, however, you can install more than one such terminal if their combined packet data transmission speeds do not exceed the D-channel throughput of 16 kbps.

If the result is a fraction, round it off to the next highest number. The larger of the two results obtained from the above two equations defines the number of SILCs required.

SILCs required in blocking conditions

If you accept blocking traffic conditions on DSLs, you have the ability to install any combination of B-channel and D-channel terminals on a DSL as long as the total number of physical terminations connecting these terminals to the DSL does not exceed eight and the number of logical terminals does not exceed 20. The greater the number of terminals on a DSL, the greater the traffic blocking.

To calculate the number of SILCs for a combination of terminals on a DSL, follow the equations below:

- Total SILC B-channel terminals Ö (number of B-channel terminals per DSL x 8) = SILCs
- Total SILC D-channel terminals Ö (number of D-channel terminals per DSL x 8) = SILCs

If the result is a fraction, round it off to the next highest number. The larger of the two results obtained from the above two equations represents the SILCs for blocking conditions.

Voltage:

DANGER OF ELECTRIC SHOCK

Foreign and surge voltage protection

In-circuit protection against power line crosses or lightning is not provided on the SILC line card. When the SILC line card is used in TIE trunk applications in which the cabling is exposed to outside plant conditions, an NT1 module certified for such applications must be used. Check local regulations before providing such service.

NT6D71 U Interface Line Card (UILC)

The NT6D71 UILC card supports the OSI physical layer (layer 1) protocol. The UILC is an ANSI defined standard interface.

The UILC provides eight two-wire full duplex (non polarity sensitive) U interfaces that are used to connect ISDN BRI compatible terminals over DSLs to the system. Each U interface provides two B-channels and one D-channel and supports one physical termination. This termination may be to a Network Termination (NT1) or directly to a single U interface terminal. Normally this physical termination is to an NT1, which provides an S/T interface that allows up to 8 physical terminals to be connected.

The length of a U DSL depends on the specific terminal configuration and the DSL wire gauge; typically, however, it should not exceed 5.5 km (3.3 mi). Refer toTable 14: UDSL cable calculations on page 119 for maximum recommended U DSL length.

The main functions of the UILC are:

- provide eight ISDN U interfaces conforming to ANSI standards
- support point-to-point DSL terminal connections
- provide channel mapping between ISDN BRI format (2B+D) and system bus format
- enable and disable DSLs
- provide loopback control of DSLs

To calculate the number of NT1s for non-blocking operation, take the larger resulting number from the two equations below:

S/T B-channel terminals Ö 2 = NT1s

Note:

A physical terminal that can use two B-channels simultaneously, such as circuit-switched voice and data, should be counted as two terminals for the purpose of this calculation.

S/T D-channel terminals Ö 6 = NT1s. This calculation is application sensitive. Up to 18 logical D-channel terminals can be connected as long as through-put does not exceed 16 kbps on the D-channel.

Note:

The S/T D-channel terminals are divided by 6, since it is assumed that for non-blocking operation, the maximum number of physical connections is 8, and that 2 are used for B-channels.

To calculate the number of NT1s where blocking is acceptable to allow maximum number of terminations on a DSL, use the two equations below:

S/T B-channel terminals Ö 16 = NT1s

Note:

This equation assumes that each S/T interface connects eight physical terminals where each physical terminal can use two B-channels simultaneously such as circuit-switched voice and data.

S/T D-channel terminals Ö 20 = NT1s (maximum of 20 TEIs per DSL)

In both cases use the larger of the two results. If the result is a fraction, round it off to the next highest number. Add the number of NT1s to the number of true U interface terminals to determine the total number of UILC-supported terminals as follows:

Total UILC terminals = Number of NT1s + Number of true U interface terminals

For the sake of this calculation it is fair to assume that each true U interface terminal represents an actual physical termination on a U interface type DSL.

To calculate the required number of UILCs to support the total number of UILC terminals (number of NT1s + number of true U interface terminals) in the system, use the following equation:

UILCs = Total UILC terminals Ö 8

If the result is a fraction, round it off to the next highest number.

NTAK09/NTAK10/NTBK50 PRI card.

Note:

This item is required for packet data implementation only.

Note:

Vintage NTBK50AA is required for Downloadable D-channel applications. When setting the timers for EuroISDN PRI2 loops in Overlay 73, the following settings are required:

If the system is connected to a local exchange that supports CRC-4 multiframing, enter CRC-4 in response to the MFF prompt, enter yes in response to the ACRC prompt (to select automatic CRC error reporting), and enter ALT in response to the ALRM prompt (to select alternate alarm mode).

If the system is connected to a local exchange that does not support CRC-4 multiframing, enter AFF in response to the MFF prompt, and enter ALT in response to the ALRM prompt (to select alternate alarm mode);

In response to the PERS prompt, enter 50 to set the alarm persistence timer to 100 ms.

In response to the CLRS prompt, enter 1 to set the clearance persistence timer to 2 ms.

ISDN PRI is required for packet data implementation to connect the system to the external packet handler (DPN-100). B and/or D-channel packet data is transmitted over clear 64 kbps

PRI B-channels to the packet handler (a D-channel daughter board is not required). The maximum number of ISDN PRI channels available for communication with the packet handler should not exceed 23 with 1.5 Mb PRI or 30 with 2 Mb PRI.

NT8D72/NT5D97 PRI2 card or QPC720/NT5D12 PRI card, for Large Systems

Note:

This item is required for packet data implementation only.

For 2.0 MBit applications, the NT8D72 ISDN PRI2 card, the dual-port NTCK43 PRI2 card, or the NT5D97 dual-port DTI2/PRI2 card is required for packet data implementation, to connect the system to the external packet handler (DPN-100). For 1.5 MBit applications, the QPC720 ISDN PRI card or the dual-port NT5D12 PRI card is required. If the MPH is used for packet data without the MCU data module, the PRI card is used to provide a PRI channel to access the PSDN.

B and/or D-channel packet data is transmitted over clear 64 kbps PRI/PRI2 B-channels to the packet handler (a D-channel daughter board is not required). The maximum number of ISDN PRI channels available for communication with the packet handler should not exceed 30 with PRI2 or 23 with PRI (D-channel connections are not required for ISDN PRI access).

Note:

Vintage NT8D72BA is required for EuroISDN applications. When setting the timers for EuroISDN PRI2 loops in Overlay 73, the following settings are required:

- If the system is connected to a local exchange that supports CRC-4 multiframing, enter CRC-4 in response to the MFF prompt, enter yes in response to the ACRC prompt (to select automatic CRC error reporting), and enter ALT in response to the ALRM prompt (to select alternate alarm mode).

If the system is connected to a local exchange that does not support CRC-4 multiframing, enter AFF in response to the MFF prompt, and enter ALT in response to the ALRM prompt (to select alternate alarm mode);

- In response to the PERS prompt, enter 50 to set the alarm persistence timer to 100 ms;
- In response to the CLRS prompt, enter 1 to set the clearance persistence timer to 2 ms.

Data Packet Network (DPN-100)

Note:

The DPN-100 is required for external packet data implementation only when the system does not process X.25 packets.

The Data Packet Network (DPN-100) is used as the external packet handler to process the B and/or D-channel packet data sent to it over ISDN PRI B-channels.

Meridian 1 Packet Handler

Note:

The MPH is required for packet data implementation only.

The MPH provides an alternative to the DPN-100, the external packet handler, for processing packet data. The MPH application resides on the MISP circuit pack. The MPH uses the dedicated MISP as the hardware platform to run the packet handler application. The system supports its administration, utilities and maintenance.

The MPH supports packet data on ISDN BRI B-channels and D-channels. D-channel packet data is routed to the MPH by the MISP line application or BRSC(s). B-channel packet data is routed to the MPH directly through dedicated connections. The MPH routes packet data to the PSDN by means of dedicated channels through either the Primary Rate Interface (PRI) or through a Meridian Communication Unit (MCU) with a synchronous modem. The MPH can support a combination of PRI or MCU connections, to a maximum of three.

The MPH can support three types of calls:

- local calls between packet data terminals connected to the same MPH without PSDN involvement
- calls between packet data terminals on separate MPH applications which must go through the PSDN
- calls to destinations not local to the MPH which are routed to the PSDN

A single MPH provides basic packet data handling functionality for up to 100 D-channels and 19 B-channel packet data terminals.

Numbering Plan supported by the MPH

The MPH supports the CCITT X.121 Numbering Plan, which consists of up to 14 digits to specify the Data Network Address (DNA) of a Data Terminal Equipment (DTE). The DNA consists of a four digit Data Network Identification Code (DNIC) and a one-10 digit National Terminal Number (NTN). The DNIC consists of a three digit Data Country Code (DCC) and a single Network Digit (ND).

In summary, the X.121 DNA is composed as follows:

DNA = DNIC (DCC+ND) + NTN

- DNIC = zxxx (z can be 2-7; the digits 0 and 1 are reserved, and 8 and 9 are used for Telex; x can be 0-9)
- NTN = 0000000001-9999999999 (1-10 digits)

Note:

The DTA may be prefixed by a single digit (0-9), which, while transparent to the MPH, may have a local significance at the PSDN interface (typically used for international calls). This prefix may be entered in response to the PRFX prompt in LD 27, when configuring the MISP for the MPH.

Only one DNIC can be configured for each MPH. Tables of DNAs can be configured and assigned to PSDN interfaces to allow packet data terminals access to the PSDN.

The MPH and Permanent Virtual Circuits and Switched Virtual Circuits

The MPH allows Permanent Virtual Circuits (PVCs) and Switched Virtual Circuits (SVCs) to be established between two local terminals or between a local terminal and a remote destination in the PSDN. The MPH can support 64 simultaneous packet switched data calls.

A Logical Channel Number (LCN) is a numeric identifier at Layer 3. It identifies the particular call (SVC or PVC) where a packet belongs. This allows multiple packet data calls to be established across a single interface.

For a PVC, a permanent logical connection is established by the MPH between the two endpoints. PVCs are mapped by LCNs at each interface. The LCN and the interface are the only identifiers used for routing packets across a PVC. A PVC establishes a permanent call between the two endpoints using the specified LCNs at each interface (without the use of call setup packets). An MPH supports a maximum of four PVCs. Each PVC is defined in the MPH configuration in LD 27. SVCs are established by call request packets originating from an ISDN BRI terminal or the PSDN. The MPH identifies the appropriate destination based on the called DNA in the call request packet.

The MPH dynamically allocates an LCN at the destination interface. The MPH does not support dynamic Layer 2 establishment (that is, terminals using either B-channels or D-channels must have all parameters (Layer 2 and Layer 3) configured and operational to receive and/or transmit calls).

The MPH and tandem connections

The MPH allows the system with access to the PSDN, to pass packet data transparently from other system switches in a private network to the PSDN by means of tandem connections. This allows the private network to make optimum use of the number of links to the PSDN. Figure 18: Tandem connections with the MPH on page 93 illustrates tandem connections, which involves dedicating a channel on two separate PRI loops on the same switch.

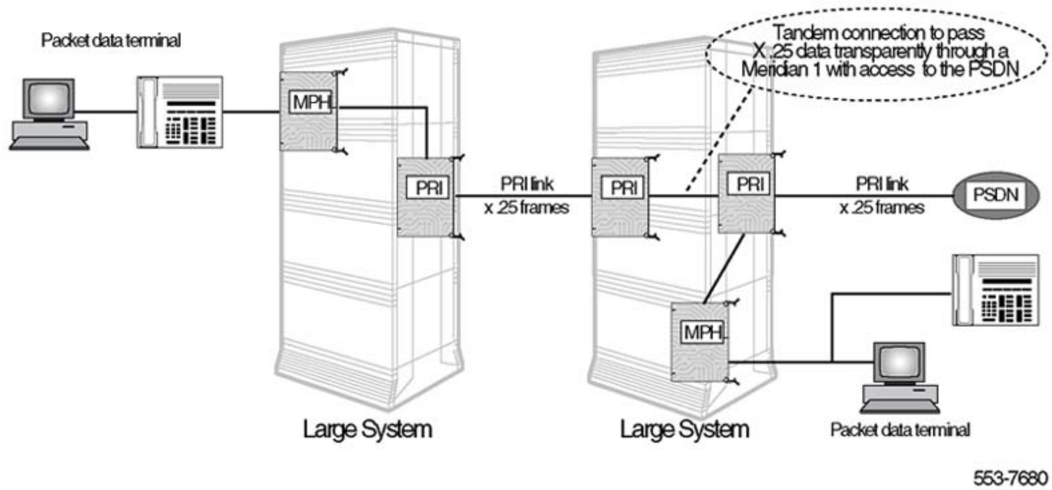


Figure 18: Tandem connections with the MPH

Call Detail Recording for MPH

MPH has two CDR record types: internal and external. The internal packet data call is a data call within an MPH which may cross different customer numbers. The external packet data call is a data call which goes to/comes from the PSDN. This includes calls between two different MPH applications on the same switch. An internal record is generated when internal CDR is equipped on either one or both for the originating data packet terminal or the terminating data packet terminal. If both data packet terminals have internal CDR equipped, then a single record is generated.

Characteristics of the MPH application

The following list summarizes the main characteristics and capabilities of the MPH.

- The MPH application and the ISDN BRI line application must reside on separate MISPs
- There is no routing between MPH applications. The MPH supports routing only between terminals logically attached to it and the PSDN
- The MPH supports the X.121 numbering plan only
- Each MPH application supports a maximum of three links to the PSDN
- There is a maximum of eight D-channel packet data separators, MISPs or BRSCs
- Each MPH application supports a maximum of 100 D-channel terminals
- The MPH can process input from a PRI/PRI2 pack over a 64Kbps or 56Kbps link
- The maximum number of PVCs for each MPH is four

- The maximum simultaneous Packet Switched Data calls for each MPH, including PVCs, is 64
- The maximum B-channel terminations for each MPH application is 19
- Tandem connections apply to PRI links only
- An MPH network interface supports a maximum of four DNA tables
- PVCs have no CDR because there is no call establishing process involved
- The packet size supported is 128 or 256
- The maximum window size is seven

Meridian Communication Unit (MCU)

The MCU is a data module used to interface between the MPH and the PSDN, when an NT8D72 ISDN PRI2 card is not used (the MCU is the only data module that supports the MPH/PSDN interface, due to the proprietary protocol used between the MPH and the MCU that allows X.25 data to pass through the interface).

The MCU requires a nailed-up connection to be established between the MPH and the MCU. The MCU is connected to a modem or Digital Interface Unit (DIU), which in turn is connected to the PSDN via a voice grade or leased line. The communication between the MCU and modem uses data packets framed in HDLC format through a dial-up synchronous modem connection. An Avaya proprietary protocol allows the X.25 data to pass through the MCU and the modem. The maximum data rate supported by any single connection is 64 Kbps.

Network Termination 1 (NT1)

The stand-alone NT1 product, which is typically installed at the user's work area, consists of the following units:

- the NT1 unit
- the optional NT1 power supply
- a mounting plate

The stand-alone NT1 unit is a two-part molded housing 210 mm (8.27 in.) by 108 mm (4.25 in.), its depth tapering from about 50 mm (2 in.) to about 32 mm (1.25 in.). On the unit's housing are four LED status indicators and three connectors.

The bottom of the unit holds four rubber feet for desk-mounting the unit, and four slides that are used to attach the unit to the mounting plate. The unit contains the single NT1 circuit pack assembly.

The stand-alone NT1 is powered by one of two methods:

1. The NT1 power supply unit which converts 110 V ac input to provide -48 V dc for the NT1, and optionally for the TEs on the S/T bus.
2. A customer-provided -48 V dc supply rated a 2 W minimum for NT1 powering. Additional power may be provided to power the TEs on the S/T bus.

The NT1 power supply unit is virtually identical to the NT1 unit. It is a two-part molded housing of 210 mm (8.27 in.) by 108 mm (4.25 in.), its depth tapering from about 50 mm (2 in.) to about 32 mm (1.25 in.). On the units housing are three connectors, one of which is a captive power cord.

The bottom of the unit holds four rubber feet for desk-mounting the unit, and four slides that are used to attach the unit to the mounting plate. The unit contains a single circuit pack assembly.

Two cables are provided with the NT1 power supply unit:

- a 178 mm (7 in.) cable (A0346581) for connecting between the power supply and the NT1 unit
- a captive power cord for connection the an ac power outlet

Terminating resistor

A terminating resistor (A0378866) is required at the end of each DSL to reduce signal reflection.

Physical capacity

Physical capacity without the BRSC

Without a BRSC, the number of network loops depends on the number of Superloop Network cards in the Network Module(s). Each Superloop Network card supports up to two IPE Modules or 32 conventional time compression multiplexing (TCM) line and trunk cards, or up to 512 ports.

The number of total ports supported by the same system option with ISDN BRI services is smaller and depends on the ratio of conventional ports to ISDN BRI ports.

The reduction of the total number of ports exists because:

- each MISP supports any combination of four SILCs and UILCs
- each SILC and UILC has eight ports

To illustrate these two points, the physical capacity for ISDN BRI systems is calculated for the following two traffic conditions assuming that ISDN BRI ports make up 10% of all the ports in the system:

- non-blocking, where each loop has a voice and data terminal and there is no contention for the B-channels on a DSL
- average traffic load of 6 CCS for voice and 12 CCS for data, where there are more terminals than the system can simultaneously connect

Table 2: Physical capacity without the BRSC on page 96 shows the number of ISDN BRI ports and the number of TCM ports supported by each system, assuming 10% ISDN BRI and 90% TCM ports.

Table 2: Physical capacity without the BRSC

	Half Group systems			Single group systems			Multi Group systems	
Capacity	NB	B		NB	B		NB	B
Groups	1/2	1/2		1	1		5	5
Modules	1	1		2	2		10	10
Connections	360	240		720	600		3600	3000
MISPs	1	2		1	4		7	19
Super-loops	3	2		6	5		30	25
DSLs	32	64		32	128		224	608
TCM Loops	133	648		310	1514		1541	7594

Note:

* NB = non-blocking, B = blocking

The system with ISDN BRI reduces the total number of combined ports; however, ISDN BRI DSL ports can connect up to eight physical terminations that can comprise up to 20 logical terminals. This provides greater port capacity than the conventional TCM voice and data ports at the cost of call blocking on the DSL.

Physical capacity with the BRSC

With the BRSC, the physical capacity of the network also determines ISDN BRI system capacity. The exception is Meridian 1 81C; where the system CPU Real Time impact is the deciding factor for overall system capacity. ISDN BRI Real Time impact is equivalent to a digital telephone. The capacity for an ISDN BRI switch (all ISDN BRI loops) in terms of the number of DSLs that can be supported is shown in Table 3: Physical capacity with the BRSC on page 97. For Half Group and Single Group systems, the numbers are the physical capacity of those machines calculated assuming 15% trunking, 25% circuit switched data, and traffic at 6 CCS per voice line.

Table 3: Physical capacity with the BRSC

	Half Group systems	Single Group systems	Multi Group systems
Blocking capacity DSLs	648	1296	4200

Memory capacity

The following tables describe the ISDN BRI requirements only. Refer to the following documents for complete instructions on how to upgrade your system:

- *Avaya CS 1000M and Meridian 1 Large System Upgrades Overview, NN43021-458*
- *Avaya Communication Server 1000E Upgrades, NN43041-458*

Refer to the following documents for details on memory and capacity:

- *Avaya Communication Server 1000M and Meridian 1 Large System Planning and Engineering, NN43021-220*

Protected data store

Protected data in the main system memory contains terminal identification and service profile data.Table 4: Protected data memory requirements for line application on page 97 andTable 5: Protected data memory requirements for trunk application on page 98 show protected data memory requirements for ISDN BRI line application and trunk application, respectively.

Table 4: Protected data memory requirements for line application

Data	Memory Requirements
Protocol Groups Data	48 words per system maximum
USID Map Data	16 words per DSL maximum
TSP Data	1,072 words per DSL maximum
Other Data	approximately 10 words per system
MISP loop block	37 words
MSDL MISP block	22 words
Socket ID table	49 words
Physical IO block	5 words

Data	Memory Requirements
IO polling table	1 word per MISP
TN line basic block	21 words
Office Data Administration System (ODAS)	3 words
Class of Service (EFD, HUNT, EHT)	12 words (4x3)
DSL data	14 words (as a non-key function)
LTID data	40 words (as a non-key function)
USID map	16 words
Template (base)	15 words
Template (features - LTID, EFD, HUNT, EHT)	1 word each
TSP data block	66 words
ISDN BRI block	7 words for each ISDN BRI DN

Table 5: Protected data memory requirements for trunk application

Data	Memory Requirements
Protected Trunk Block (Trunk DSLs use the Protected Trunk Block instead of the Protected BCS Block)	3 words added
Route Data Block	16 words added for RURC prompt and 1 word added for STAT prompt, for Advice of Charge for Euro ISDN. 6 words added, to store the country for EuroISDN.
ISDN BRI Protocol Data Block	2 words added
Configuration Record	6 words added, to store the country for EuroISDN.
Protected D-channel block	10 words added to store the protocol-specific D-channel information, for the Universal ISDN Protocol Engine (UIEP).
Protected MSDL/MISP block	24 words added, to support the increase from 8 to 32 of the number of supported applications per MSDL/MISP.

Unprotected data store

Unprotected data in the main system memory is used for data link (layer 2) and network (layer 3) information message storage during system operation.Table 6: Unprotected data memory requirements for line application on page 99 andTable 7: Unprotected data memory requirements for trunk application on page 100 show memory requirements for temporary data storage during system operation for ISDN BRI, for line application and trunk application, respectively.

Table 6: Unprotected data memory requirements for line application

Data	Memory Requirements
MISP input buffer	140 words per system
MISP expedited input buffer	128 words per system
MISP loop block	82 words
MISP output buffer (transmit receive)	260 words
MISP expedited output buffer	512 words
MISP output request buffer	80 words
MSDL MISP block	95 words
Socket ID table	48 words
Meridian 1 expedited receive buffer	128 words
Meridian 1 receive buffer	140 words
Meridian 1 expedited transmit buffer	528 words
MISP traffic accumulating block	30 words
MISP traffic holding block	30 words
TN line block	32 words (16x2)
Incoming call reference table	33 words
Outgoing call reference table	33 words
Incoming call reference usage map	4 words
Outgoing call reference usage map	4 words
Incoming message call reference table	33 words
Outgoing message call reference table	33 words
DSL data block	3 words

Table 7: Unprotected data memory requirements for trunk application

Data	Memory Requirements
MISP Call Register Data Block	2 words
Global Variable Data Block	8 words
Trunk Card Block (all ISDN BRI unprotected card block use the Trunk Card Block)	For Advice of Charge For EuroISDN4: For AOC at call set-up: - 24 words to store the Start of Time value of the Real Time Clock - type of charging - charged item - charging rate - rate type - currency identifier - currency amount - multiplier - length of time unit - scale of time unit - granularity - scale of granularity For AOC during the call: no. of words - type of charging information - recorded charges - currency identifier - currency amount - multiplier - number of charging unit - billing identification For AOC at the end of the call: no. of words - type of charging information - recorded charges - number of charging unit - currency identifier - currency amount - multiplier - number of charging unit - billing identification
Trunk Line Block	6 words
DCH global area, for messaging	1 word
MISP global area, for messaging	2 words
Unprotected MSDL/MISP block	185 words added, to support the increase from 8 to 32 of the number of supported applications per MSDL/MISP.

ISDN BRI configuration guidelines

Throughout this section, please note the following:

- ISDN BRI trunking is not supported in North America.

System modular design permits flexible engineering. Systems can be tailored to a customer's system size and port type requirements. ISDN BRI line cards can be mixed with conventional TCM line and trunk cards in the same IPE module.

The I/O panels on the IPE modules are the same regardless of the type of line cards installed in the module. Therefore, the external communication cables between system and Main Distribution Frame (MDF) are the same for DSLs and the conventional telephone sets.

Physical parameters

The physical parameters specific to ISDN BRI are as follows:

- MISP location in the Network module for Large Systems.
- BRSC location in the IPE module
- DSL configuration (line application or trunk access)

MISP location in the Network module (Large Systems)

For Large Systems a Network module can contain one or more MISPs. This number is governed by contention for the network card slots and network loop addresses between the MISP and the Superloop Network cards, among other network cards.

Each MISP occupies one network card slot and two network loop addresses. The Superloop Network card also occupies one network card slot, but it requires four network loops, two of its own and two of the adjacent card slot.

Since the MISP uses one network loop for communication with ISDN BRI cards, the MISP must be located in the same Network module as the Superloop Network card. To prevent conflict between the MISP and the Superloop Network card, the MISP should always be installed into a card slot with loop addresses that are not used by a Superloop Network card.

BRSC location in the IPE module

The BRSC can be located in any IPE module, provided it is in the same module as the ISDN BRI line card it serves.

DSL configuration (line application)

Digital Subscriber Loops configured for a line application connect the system to ISDN BRI terminals. A DSL consists of a cable connecting the ISDN BRI DSL port to the Main Distribution Frame (MDF). From the MDF, the loop is cross-connected to the office wiring, which terminates into 8-pin modular jacks (wall outlets). From these outlets, module cables of a maximum length of 10 meters (33 feet) connect to ISDN BRI terminals.

To provide reliable voice and data transmission between the system and ISDN BRI terminals, DSLs must be engineered with the following basic considerations in mind:

- number of terminals connected
- loop length (cable type and wire gauge)
- distribution of terminals on a DSL
- type of terminals connected to a DSL

Refer to "Transmission characteristics" in this chapter for a detailed description of the DSL bus configurations and their characteristics. These types are:

- point-to-point SILC DSL (see Figure 20: Point-to-point DSL on page 111)
- short passive SILC DSL (see Figure 21: Short passive SILC DSL on page 112)
- extended passive SILC DSL (see Figure 22: Extended passive SILC DSL on page 114)
- branched passive SILC DSL (see Figure 23: Branched passive SILC DSL on page 115)
- point-to-point UILC DSL (see Figure 24: Point-to-Point UILC DSL on page 118)

Loop Length

Loop lengths for specific configurations are controlled by the differential round-trip time delay to the SILC. A short passive bus uses fixed timing and must maintain the differential round-trip delay between 10 and 14 microseconds. An extended or branched passive loop uses adaptive timing to maintain the differential round-trip delay within two microseconds. A point-to-point passive bus uses adaptive timing with the delay from 10 to 42 microseconds. Please refer toTable 8: Cable types and characteristics on page 110 for the loss and delay parameters of the various cable types used in determining the S/T DSL configuration limits.

Types of cables

Some commonly used types of cable with different wire gauges are shown in Table 8: Cable types and characteristics on page 110. Refer to this table for maximum recommended cable lengths.

The SILC interface supports a four-wire S/T bus consisting of a transmit pair and a receive pair per DSL. The same wiring polarity must be maintained for all physical terminals on the S/T bus. The SILC supplies up to 2 Watts of power per DSL as an optional terminal power source PS1 by providing -48 volts (ANSI) or -40 volts (international) DC on the transmit pair signal with respect to receive pair. One additional pair of wires may be used in the office wiring to supply power from an auxiliary power source PS2.

The UILC interface supports a two-wire point-to-point loop consisting of a twisted pair engineered for 2B1Q line encoding on the DSL. One physical termination is allowed at the end of the loop. This may be an NT1 device that interfaces with an S/T bus on the subscriber's premises. The U interface is not polarity sensitive. The UILC does not provide power to the terminal.

The cables used to connect the terminals to ISDN BRI cards normally come in one of three wire gauges: 22, 24, and 26 AWG. The larger gauge wire such as 22 AWG has less transmission loss. This means it is actually able to provide a DSL of almost twice the length of the 26 AWG wire for the same loop configuration.

Distribution of terminals

Distribution of terminals on a DSL depends on the type of loop used and the type of interface connected to the loop. For the SILC interface, use the following rules:

- On a short passive bus, up to eight terminals can be distributed anywhere on the loop.
- On an extended passive bus, the terminals must be clustered at the far end of the loop and no more than 4 terminals should be connected.
- On a branched passive bus, two branches may exist at the end of the S/T loop. Up to two terminals per branch can be connected.

Note:

For each of the bus configurations described above, a terminating resistor (Part number A0378866) must be used to connect the last device.

- Where there is only one terminal connected to the loop in a point-to-point configuration, the terminal must be connected at the end of the loop at the terminating resistor jack.

Type of terminals

Type of terminal used depends on the customer requirements. The ISDN BRI terminals can be circuit-switched voice or data, B-channel packet data, or D-channel packet data terminals. A DSL can support up to eight physical terminations each linking one or more terminals to the DSL.

The following recommendations should be considered when connecting terminals to DSLs:

- The total number of physical terminations on an S/T DSL may not exceed eight. Up to 20 logical terminals may be connected to an S/T DSL. A logical terminal may be directly connected to the DSL through its own physical termination, or it may be indirectly connected through a common physical termination.

  For non-blocking traffic conditions: Two B-channel circuit-switched voice or data terminals may be connected on each S/T DSL. More than two B-channel terminals can be connected, however, only two will be able to communicate simultaneously. If more than two terminals are connected this could create a blocking condition where the terminals are contend for available B-channels. Any other terminals connected to this DSL can only be D-channel terminals. You can install more than one D-channel terminal if their combined packet data transmission speeds do not exceed the D-channel throughput of 16 kbps.

  For blocking traffic conditions: If you accept blocking traffic conditions on DSLs, you have the ability to install any combination of B-channel and D-channel terminals as long as the total number of physical terminations connecting these terminals to the DSL does not exceed eight. These physical terminations can link up to 20 logical terminals The greater the number of terminals on a DSL, the greater the probability for traffic blocking.
- Only one termination can be connected at the end of a UILC DSL. This termination can be to a Network Termination (NT1) or directly to a single U interface terminal. Normally this physical termination is to an NT1, which provides an S/T interface that allows up to

8 physical terminals to be connected. These terminals communicate to the system through the NT1 and the UILC interface.

- Determine the type of DSL bus configuration you wish to use to connect your terminals keeping in mind the wire type, the length, and the layout of your office wiring

DSL configuration (trunk application)

When configuring DSLs for trunk access the following basic considerations must be kept in mind:

- type of trunk connection to be configured (TIE or CO/DID)
- whether to draw the system reference clock source from the ISDN BRI trunk connection, and the associated clock reference cabling connections
- whether the ISDN BRI trunks are to be configured as the backup trunks for ISDN PRI trunks

Type of trunk connection - Digital Subscriber Loops configured for trunk access allows either a TIE trunk connection from either an S/T interface or U interface, or a CO/DID trunk connection from an S/T interface (please refer to the detailed information described earlier in the "ISDN BRI trunk access" section of the "Functional description" chapter).

System reference clock source - If the system reference clock source is drawn from the ISDN BRI trunk connection (the system on the USER side), the connection from the SILC line card to the Clock Controller must be from DSL#0 or DSL#1 of the SILC card (please refer to the detailed information described earlier in the "ISDN BRI trunk access" section of the "Functional description" chapter). The clock reference cables have to be connected following the procedures described in *Avaya ISDN Basic Rate Interface Installation and Commissioning, NN43001-318*.

ISDN BRI trunks as backup for ISDN PRI trunks - ISDN BRI trunks can be configured as backup trunks for ISDN PRI trunks, through ESN Route Selection or Route Hunting. As well, an ISDN BRI clock source can be configured as a secondary clock reference for an existing ISDN DTI/PRI clock source.

Functional parameters

Functional parameters must be considered during ISDN BRI configuration procedures. These procedures are used to create an ISDN BRI database and to configure ISDN BRI functions and features when administering the system. These functional parameters apply to:

- ISDN BRI database generation
- DSL addressing

ISDN BRI database generation

When installing ISDN BRI in the system for the first time, configure these components in the order listed below. SeeISDN BRI implementation on page 129 for detailed procedures on configuring ISDN BRI.

When changing existing ISDN BRI service, following this order is unnecessary. However, be aware of the relationship of one component to another and whether changing one component necessitates changing other components.

Configurations for an ISDN BRI line application are:

- LAPD protocol group
- Multi-purpose ISDN Signaling Processor (MISP)
- Basic Rate Signaling Concentrator (BRSC) (optional)
- S/T line card/U line card (optional)
- Digital Subscriber Loop (DSL)
- Terminal Service Profile (TSP) on DSL
- Terminals (for example, M5317TDX, M5209TDcp)
- Traffic (LD 02) (optional)

To add an MPH to a system with an existing ISDN BRI configuration, perform the following configurations in the order shown. Use LD 27 except where noted otherwise:

- Configure the LAPD protocol group.
- Configure the LAPB protocol group.
- Configure the X.25 packet protocol group.
- Configure the DNA table associated with the MPH network interface.
- Configure the ISDN Primary Rate Interface (PRI) for packet data:
    - the ISDN PRI loop (LD 17).
    - the ISDN customer (LD 15).
    - the TIE trunk route for packet data (LD 16).
    - the TIE trunk for packet data (LD 14).

OR, if an MCU is used instead of a PRI pack:

- Configure
    - the TIE trunk route for packet data (LD 16).
    - the TIE trunk for packet data (LD 14).

- the Meridian Communication Unit (MCU) (LD 11).

- Perform changes at a centra or at a remon tandem connections (optional) (LD 14).
- Configure a Multi-purpose ISDN Signaling Processor (MISP) for an MPH.
- Disable the MISP and modify the MISP for line cards or BRSCs to support D-channel packet data.
- Modify the TSP for D-channel packet data.
- Modify the DSL for B-channel packet data and define new TSP.
- Configure Permanent Virtual Circuits (PVCs) (optional).
- Change terminal configurations (M5317TDX, M5209TDcp, M5000TD-1).
- Configure Customer Data Record (CDR) (LD 15 and LD 27). (optional)

To add an external packet handler (DPN-100) to a system with an existing ISDN BRI configuration, configure the following steps in the order shown. Use LD 27 except where noted otherwise:

- LAPD protocol group
- Packet data transmission:
- the ISDN PRI loop (LD 17)
- the ISDN customer (LD 15)
- the TIE trunk route for packet data (LD 16)
- the TIE trunk for packet data (LD 14)
- Multi-purpose ISDN Signaling Processor (MISP)
- Basic Rate Signaling Concentrator (BRSC) Card (optional)
- S/T Interface Line Card (SILC)/U Interface Line Card (UILC) (optional)
- Digital subscriber loop (DSL)
- Terminal Service Profile (TSP) on DSL
- Terminals (for example, M5317TDX, M5209TDcp)

Configurations for an ISDN BRI trunk application are:

- ISDN customer (LD 15)
- Trunk pad tables (LD 73) (optional)

Note:

The digital pad provides gain or attenuation values to condition the level of the digitized transmission signal according to the network loss plan. This determines transmission levels for the B-channel circuit-switched voice calls.

- LAPD Protocol Group
- ISDN BRI trunk route data block (LD 16)
- MISP
- SILC (for CO/TIE connectivity) and/or UILC (for TIE connectivity)
- Trunk DSL
- If the SILC clock is configured, enter the ISDN BRI trunk clock reference (LD 73)

DSL addressing

ISDN BRI DSL addressing corresponds to standard TN addressing. A DSL address is uniquely defined as TN (l s c dsl#), where:

- l is the number of the network superloop by which it is controlled
- s is the number of the IPE module (shelf number) where it is located
- c is the card slot position of the SILC/UILC in the module
- dsl# is the port number on the card

Transmission characteristics

ISDN BRI provides two different types of interfaces: the S/T interface provided by the SILC and the U interface provided by the UILC. Each interface has unique physical and transmission characteristics and requires different considerations when configuring DSLs for lines or trunks.

SILC DSL line configuration

The SILC supports both point-to-point and point-to-multipoint transmission. The maximum DSL length must not exceed 1 km (3,280 ft); however, the actual length depends on the cable wire gauge, the number of terminals connected to the loop, and the differential round-trip time delay limits.

When you are configuring DSLs for lines, follow these basic rules:

- Maintain wiring polarity for both the transmit and receive pairs.
- Use a maximum of 10 m (33 ft) modular cable to connect each ISDN BRI terminal to the DSL.
- Keep the length of the cable stub (distance between the RJ-45 receptacle and the DSL cable) to less then 1 m (3.3 ft).

- Do not allow bridge taps or split pairs on the DSL and make sure that the differential pairs (Tx-/Tx+ and Rx-/Rx+), each consist of a twisted pair along the entire length of the DSL.
- Make sure ISDN BRI terminals powered through the DSL do not exceed the total power of 2 Watts.
- Connect a maximum of two ISDN BRI terminals requiring B-channel transmission or one terminal using both circuit-switched voice and data to each DSL for a non-blocking configuration. For maximum concentration per DSL, connect up to 8 physical BRI terminals, and up to 20 logical terminals to use two B-channels and the 16 kbps capacity of the D-channel.
- Select the appropriate Network Terminal line sampling mode (NTFS for fixed and NTAS for adaptive) when configuring the DSL in LD 27.

  NT Mode Fixed Sampling (NTFS) may be selected when the device is in an NT on a passive bus wiring configuration up to approximately 200 meters in length (depending on cable type). In this mode, multiple terminals (up to eight) may be connected anywhere along the passive bus.

  NT Mode, Adaptive Sampling (NTAS) should be selected when the device is in an NT on any wiring configuration up to the maximum specified length for operation. Multiple terminals, if required, must be grouped within approximately 100 meters of each other (depending on cable type).
- Place the terminating resistor (A0378866) at the end of the loop, depending on the configuration (refer to Figure 19: S/T digital subscriber loop wiring example for ISDN BRI lines on page 109).

Figure 19: S/T digital subscriber loop wiring example for ISDN BRI lines on page 109 shows a wiring example of an SILC DSL with multiple physical terminations and the terminating resistor at the end of the loop. An SILC DSL consists of a six-wire twisted pair cable, but only four wires carry the signal and two wires provide conduit for an auxiliary power source. This external power source can be used when the total power consumption exceeds 2 Watts on each DSL. In this case the terminals need to be configured to use the auxiliary power source (PS2) or other auxiliary power supplies as part of their product packages.

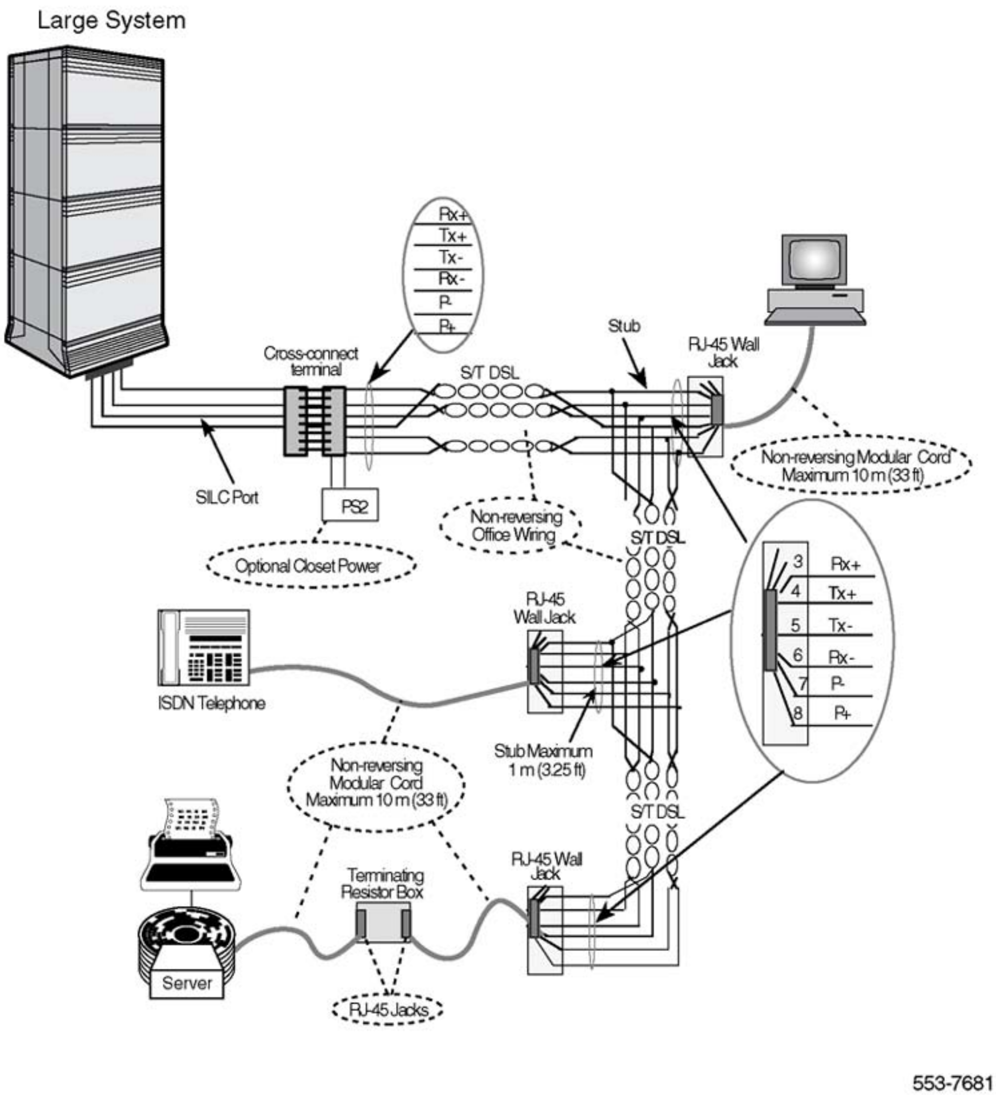


Figure 19: S/T digital subscriber loop wiring example for ISDN BRI lines

Cable characteristics

Table 8: Cable types and characteristics on page 110 lists the parameters of the various cable types used in determining the S/T DSL configuration limits. The cables listed are those used for telephony wiring applications, and the characteristics listed are for Avaya-supplied cable at 96 kHz and 20°C (68° F).

Table 8: Cable types and characteristics

Cable type	Gauge AWG	Loss dB/km (dB/kft.)	Delay μs/km (μs/ kft.)
Outside PIC	22	1.6 (5.4)	1.7 (5.5)
Outside pulp	22	1.8 (6.0)	1.6 (5.3)
Outside PIC	24	2.3 (7.6)	1.7 (5.6)
Outside pulp	24	2.5 (8.2)	1.7 (5.5)
Outside PIC	26	3.3 (11.0)	1.8 (5.9)
Outside pulp	26	3.3 (11.0)	1.7 (5.7)
Inside riser	22	1.6 (5.2)	1.6 (5.2)
Inside riser	24	2.3 (7.5)	1.7(5.6)
Inside riser	26	3.2 (10.5)	1.8 (5.9)
Inside Z station (FT1)	22	1.6 (5.2)	1.8 (5.9
Inside Z station (FT4)	22	2.0 (6.6)	2.0 (6.6)
Inside type D (3 and 4 pair)	24	2.6 (8.5)	1.9 (6.2)
Inside type D (25 pair)	24	2.9 (9.5)	2.0 (6.6)

The following examples show some typical SILC DSL configurations. These are:

- point-to-point DSL
- short passive DSL
- extended passive DSL
- branched passive DSL

Point-to-point SILC DSL

This configuration is shown in Figure 20: Point-to-point DSL on page 111. It represents the simplest type of bus configuration.

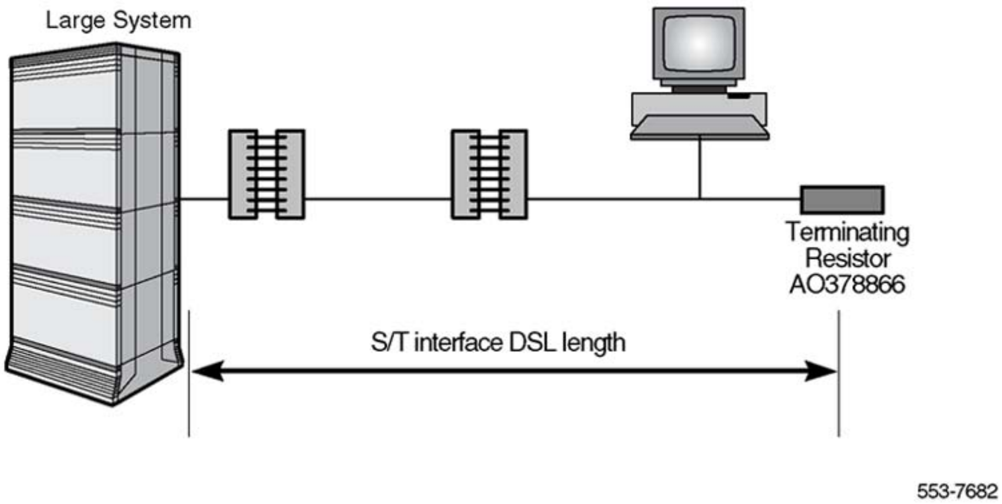


Figure 20: Point-to-point DSL

The Point-to-Point bus provides the longest SILC DSL length. Recommended rules:

- Configure the DSL as adaptive mode through overlay 27 (MODE = NTAS)
- Use a terminating resistor (A0378866) at the end of the DSL.
- Connect only one terminal.
- Cable loss must not exceed 6 dB.

Maximum DSL length depends on the cable type and wire gauge. For a point-to-point bus, the SILC DSL length is as shown in Table 9: Cable types and point-to-point DSL lengths on page 111.

Table 9: Cable types and point-to-point DSL lengths

Cable type	Gauge AWG	Maximum DSL length m (ft.)
Outside PIC	22	1110 (3640)
Outside PIC	24	790 (2590)
Outside PIC	26	540 (1770)
Outside pulp	22	1000 (3280)
Outside pulp	24	730 (2390)
Outside pulp	26	540 (1770)
Inside riser	22	1150 (3770)
Inside riser	24	800 (2620)
Inside riser	26	570 (1870)

Cable type	Gauge AWG	Maximum DSL length m (ft.)
Inside Z station (FT1)	22	1150 (3770)
Inside Z station (FT4)	22	910 (2980)
Inside type D (3 and 4 pair)	24	700 (2300)
Inside type D (25 pair)	24	630 (2070)

Short passive SILC DSL

This configuration is shown in Figure 21: Short passive SILC DSL on page 112.

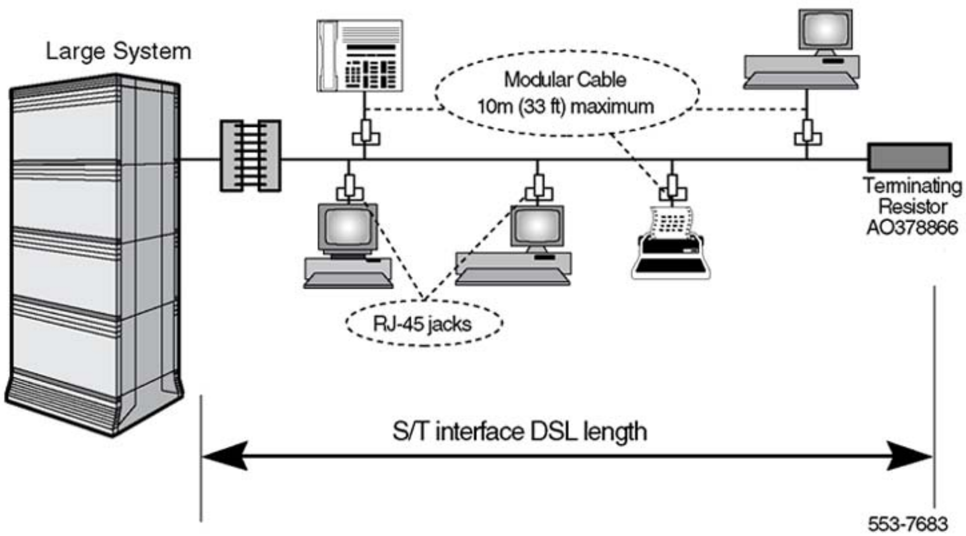


Figure 21: Short passive SILC DSL

In the short passive SILC DSL configuration the system and terminals may be located anywhere along the SILC DSL. This configuration has the shortest length, but the maximum number of terminals are allowed with no restrictions on the location of the system and the terminals.

Recommended rules:

- Configure the DSL as fixed timing mode through overlay 27 (MODE = NTFS).
- A maximum of eight physical terminals may be connected.
- Use a 100 ½ terminating resistor (A0378866) at the end of the DSL.
- Terminate both ends of the DSL if the NT is not located at the end of the DSL. In this case the distance between the system and the SILC DSL should not exceed 30 ft (9 m).
- The maximum round trip delay for the selected DSL cable is 2 µs.

Maximum DSL length depends on the cable type and wire gauge. For a short passive SILC DSL, the length is as shown in Table 10: Cable types and short passive SILC DSL lengths on page 113.

Table 10: Cable types and short passive SILC DSL lengths

Cable type	Gauge AWG	Maximum DSL length m (ft.)
Outside PIC	22	170 (560)
Outside PIC	24	165 (540)
Outside PIC	26	155 (510)
Outside pulp	22	170 (560)
Outside pulp	24	170 (560)
Outside pulp	26	160 (520)
Inside riser	22	180 (590)
Inside riser	24	165 (540)
Inside riser	26	150 (490)
Inside Z station (FT1)	22	150 (490)
Inside Z station (FT4)	22	140 (460)
Inside type D (3 and 4 pair)	24	150 (490)
Inside type D (25 pair)	24	145 (480)

Extended passive SILC DSL

This bus configuration is shown in Figure 22: Extended passive SILC DSL on page 114.

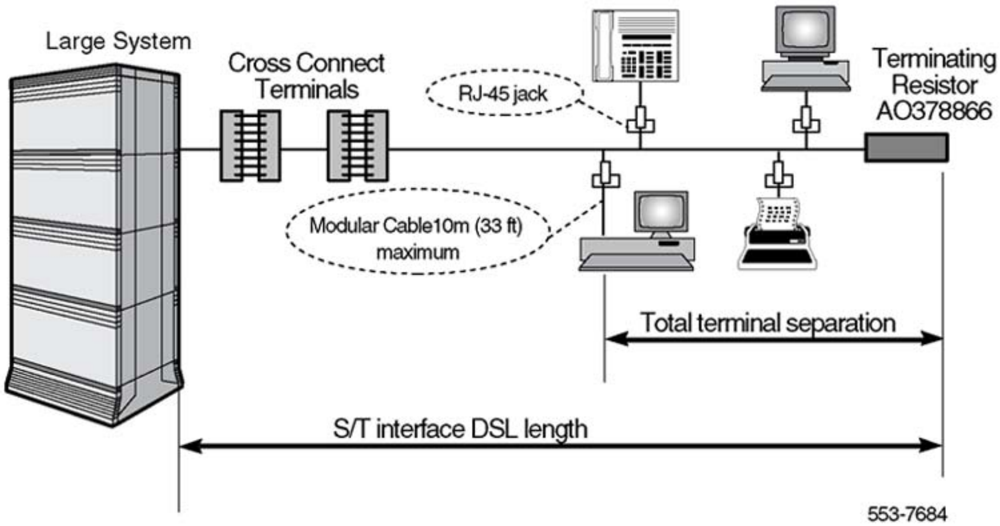


Figure 22: Extended passive SILC DSL

The extended passive bus is designed to allow up to four terminals to be located a long distance from the SILC. The length of the SILC DSL and the separation between each terminal are the significant factors in this configuration.

Recommended rules:

- Configure the DSL in adaptive mode through overlay 27 (MODE = NTAS).
- Use a 100 ¾ terminating resistor (A0378866) at the end of the SILC DSL.
- Configure no more than four terminals.
- The cable loss must not exceed 3.8 dB.

The maximum SILC DSL length and separation between terminals is given in Table 11: Cable types and extended passive SILC DSL lengths on page 114. Configure the first terminal at the end of the terminated SILC DSL, then calculate the distance from the farthest to the nearest terminal.

For every terminal (less than four) not configured, you may add 15 ft (5 m) to the distance of total terminal separation.

Table 11: Cable types and extended passive SILC DSL lengths

Cable type	Gauge AWG	Maximum DSL length m (ft.)	Total terminal separation m (ft.)
Outside PIC	22	700 (2300)	40 (130)
Outside PIC	24	500 (1640)	40 (130)
Outside PIC	26	340 (1110)	36 (120)

Cable type	Gauge AWG	Maximum DSL length m (ft.)	Total terminal separation m (ft.)
Outside pulp	22	630 (2070)	42 (140)
Outside pulp	24	460 (1510)	40 (130)
Outside pulp	26	340 (1110)	36 (120)
Inside riser	22	730 (2390)	42 (140)
Inside riser	24	500 (1640)	38 (125)
Inside riser	26	360 (1180)	35 (115)
Inside Z station (FT1)	22	730 (2390)	35 (115)
Inside Z station (FT4)	22	570 (1870)	33 (110)
Inside type D (3 and 4 pair)	24	440 (1443)	35 (115)
Inside type D (25 pair)	24	400 (1310)	35 (115)

Branched passive SILC DSL

This bus configuration is shown in Figure 23: Branched passive SILC DSL on page 115.

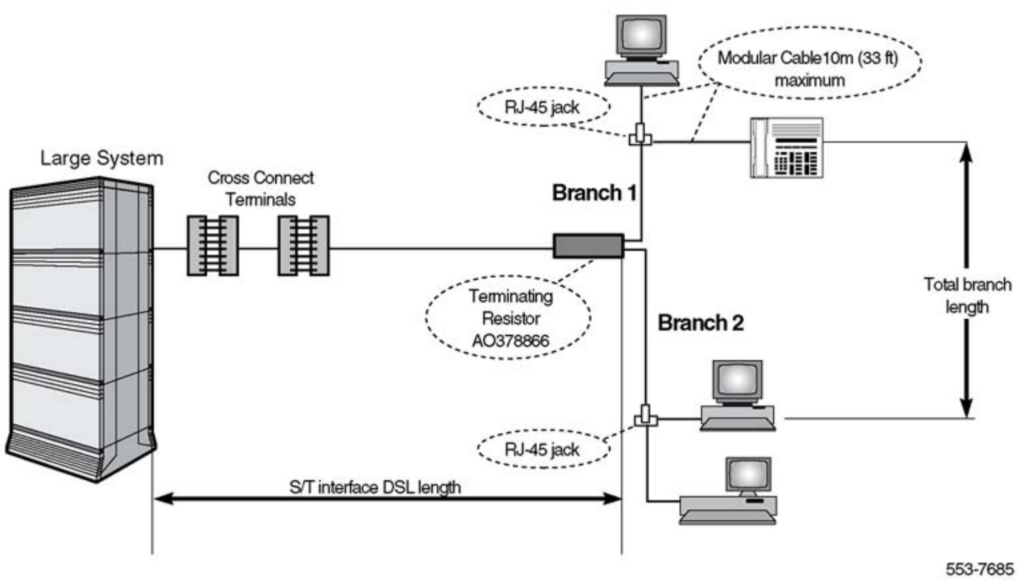


Figure 23: Branched passive SILC DSL

The branched passive bus configuration uses the existing building wiring where the SILC DSL is terminated in a telephone wiring closet. The significant factors in this configuration are the maximum SILC DSL length, the total length of the two branches, and the difference between two branch lengths.

Recommended rules:

- Configure the DSL as adaptive mode through overlay 27 (MODE = NTAS).
- Configure no more than four terminals.
- Configure no more than two terminals per branch.
- Use a 100 ¾ terminating resistor at the end of the SILC DSL.
- The cable loss must not exceed 3.8 dB.

Maximum SILC DSL length depends on the cable type and wire gauge. For a branched passive bus, the DSL length is typically:

- 1,919 ft (585 m) for Inside Z station type cable 22 AWG
- 1,197 ft (365 m) for Inside type D cable 24 AWG
- 1,033 ft (315 m) for Inside riser type cable 26 AWG

Calculate the length of the SILC DSL, the sum of the branches, and difference in the length of the branches. Refer to Table 12: Cable types and branched passive SILC DSL lengths on page 116 for maximum allowable limits.

Table 12: Cable types and branched passive SILC DSL lengths

Cable type	Gauge AWG	Maximum DSL length m (ft.)	Branched lengths m (ft.)
Outside PIC	22	340 (1115)	105 (345)
Outside PIC	24	270 (885)	105 (345)
Outside PIC	26	215 (705)	100 (330)
Outside pulp	22	315 (1030)	110 (360)
Outside pulp	24	255 (835)	105 (345)
Outside pulp	26	215 (705)	95 (310)
Inside riser	22	350 (1150)	110 (360)
Inside riser	24	270 (885)	105 (345)
Inside riser	26	220 (720)	100 (330)
Inside Z station (FT1)	22	350 (1150)	100 (330)
Inside Z station (FT4)	22	295 (965)	95 (310)
Inside type D (3 and 4 pair)	24	250 (820)	100 (330)
Inside type D (25 pair)	24	770 (235)	95 (310)

SILC DSL trunk configuration

When you are configuring trunk DSLs, follow these basic rules:

- spread the two B-channels out as members of the Route Data Block, so that if a trunk DSL is out of service, the search for an idle trunk will not be impaired.
- do not exceed the maximum loop length of 1km (0.6 mi) for the S/T interface.
- if the system reference clock source is to be drawn from the SILC, the connection to the clock controller can only be from DSL#0 and DSL#1 of the SILC card. DSL#0 may only be configured as the primary clock reference and DSL#1 as the secondary clock reference. Also, ensure that the proper cable connections are made between the SILC and the clock controller, following the procedures described in *Avaya ISDN Basic Rate Interface Installation and Commissioning, NN43001-318*.

UILC DSL line configuration

The UILC supports only point-to-point transmission. The maximum length of the U interface DSL is determined by the maximum loop loss, but typically should not exceed 5.5 km (3.3 mi).Table 13: Maximum recommended U DSL length on page 118 gives the maximum recommended U DSL length. The maximum loss is 46 dB @ 40 kH. However, to meet the mandatory bit error rate performance of 10^{-7} or better in all cases, a maximum limit of 40 dB cable loss is recommended.

Note:

This interface is designed to utilize most of the existing non-loaded twisted pair wiring in North America. Not all the twisted pair cables are suitable for ISDN BRI application. Before you use a section of this cable, you must verify its suitability by performing the following tests:

- determine the type and length of the cable, and the total signal loss of the DSL (calculated using Table 14: UDSL cable calculations on page 119)
- determine total signal loss contributed by the bridge taps on the DSL (1.7 db/kft(km) and up to 5.1 db for each bridge tap length)
- verify that there are no consistent or random noise sources that may affect the transmission quality (that is, using a Bit Error Rate tester, the result should be better than 10^{-7} error rate for at least 30 minutes)

Note:

All calculated losses above should total less than 40dB @ 40 kH (BER ð 10^{-7}).

- determine the outlet pinout at the terminal location

Figure 24: Point-to-Point UILC DSL on page 118 shows a typical U interface DSL with an NT1 terminating the DSL and providing an S/T interface to an ISDN BRI terminal.

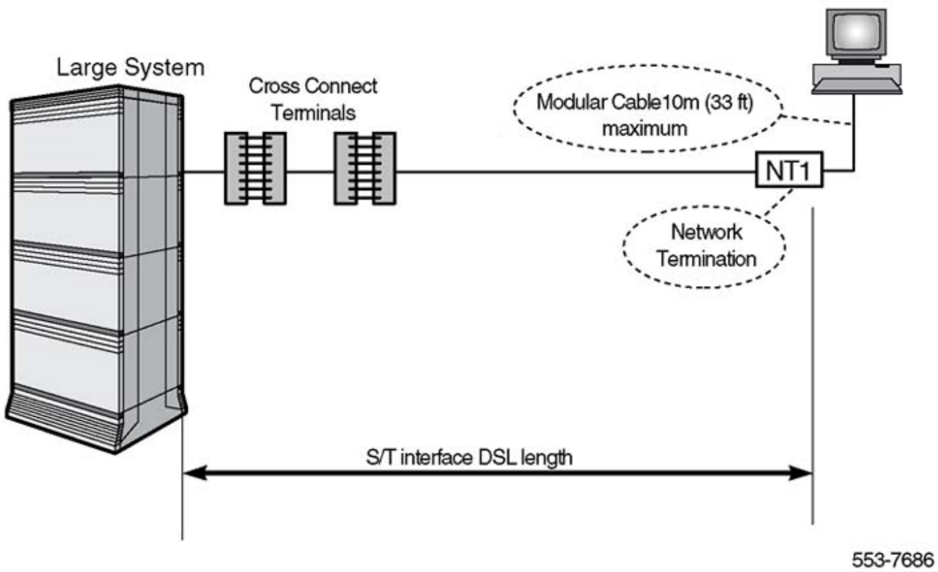


Figure 24: Point-to-Point UILC DSL

Table 13: Maximum recommended U DSL length on page 118 lists the maximum recommended loop length which should be considered in U interface DSL installations without any bridge taps (refer to Table Table 13: Maximum recommended U DSL length on page 118 for a guideline for calculating DSL loop loss for U interface DSL installations with bridge taps).

Table 13: Maximum recommended U DSL length

Cable gauge AWG	Maximum recommended length km (kft.)
26	4.40 (14.5)
24	6.55 (21.5)
22	8.80 (29.0)
mix	5.50 (18.0)

Table 14: UDSL cable calculations on page 119 is recommended as a guideline for calculating selected U interface DSL loop loss. Use the Length km (kft.) and Loss dB columns to record your calculations. Select the corresponding loss (dB/kft) based on the type of cable used and multiply by the cable length. After calculating all the losses due to the DSL used, calculate losses due to bridge taps. For any bridge tap with length exceeding 3 kft, only add 5.1 dB. Add

all the losses due to the DSL and due to bridge taps, and record your calculations. The total loss should not exceed the recommended maximum loss of 40 dB.

Note:

The bridge taps are not terminated and are left unused. A complete knowledge about the characteristics of the DSL selected is recommended. Keep the DSL as simple and as short as possible to obtain maximum performance.

Table 14: UDSL cable calculations

Item	Gauge AWG	Insulation type	Loss dB/km (dB/kft.)	Length km (kft.)	Loss dB
1	19	PIC	1.0 (3.3)		
2	19	pulp	1.1 (3.6)		
3	22	PIC	1.4 (4.6)		
4	22	PIC	1.5 (4.9)		
5	24	PIC	1.8 (5.9)		
6	24	pulp	1.9 (6.3)		
7	26	PIC or pulp	2.8 (9.2)		
8	Customer premises wiring		1.8 (5.9)		
9	Local exchange wiring		2.8 (9.2)		
10	bridge tap 1				
11	bridge tap 2				
12	bridge tap 3				
13	bridge tap 4				
14	bridge tap 5				
15	bridge tap 6				
Total loss in dB (add items 1 through 15)					
Recommended maximum loss = 40 dB					

Figure 25: UILC DSL used as an extension for an S/T interface loop on page 120 shows a DSL extension where the U interface is used to extend the loop to an NT1 and from the NT1 it shows an S/T interface connecting two ISDN BRI terminals.

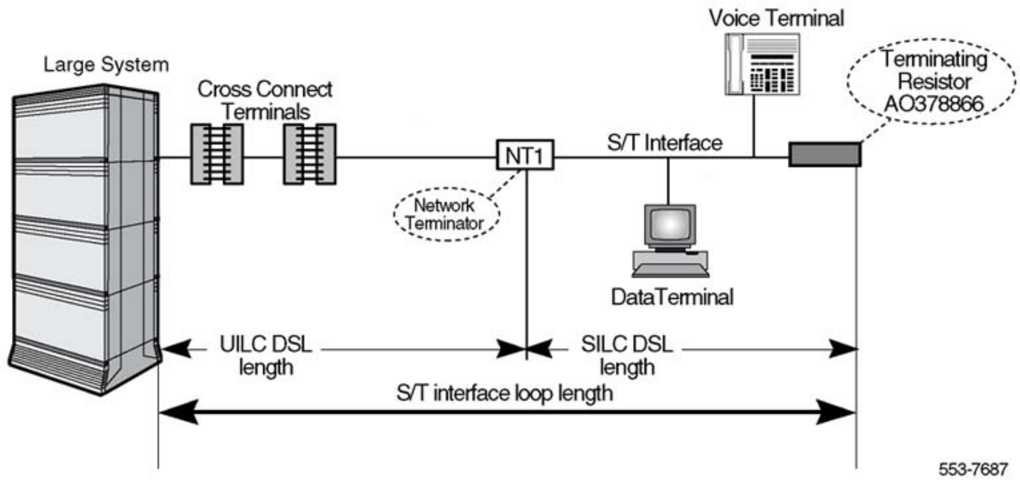


Figure 25: UILC DSL used as an extension for an S/T interface loop

Terminal addressing and service profile assignment

Terminal Service Profiles (TSPs) are service profile specifications stored in the database that can be associated with various terminals during terminal initialization and that define the terminal DN, Class of Service, call restriction levels, and other service and feature attributes. ISDN Terminal initialization occurs when a terminal is installed, every time the system is sysloaded, or when the MISP or the line card to which the terminal is connected is replaced.

ISDN BRI terminal addressing

An ISDN BRI terminal connected to a DSL is addressed by using both the physical address and the logical address, where:

- the physical address is l s c dsl# representing the physical (layer 1) identifier
- the logical address is defined as Terminal End-point Identifier (TEI), which is a data link (layer 2) identifier and the Terminal Service Profile (TSP), which is a network (layer 3) identifier

The User Service Identifier (USID) uniquely identifies the Terminal Service Profile (TSP) number (there may be up to 16 TSPs per DSL.) The Service Profile ID (SPID) is a reference to the TSP, which contains the DN.

The SPID is an identification number (any combination of 1-20 alphanumeric characters) that is configured in overlay 27. The SPID is entered in overlay 27, in association with the TSP, and is also entered on the terminal keypad during initialization. All the terminals on a DSL that share the same TSP have the same USID.

Assigning the Terminal End-point Identifier

A Terminal End-point Identifier (TEI) is associated with establishing the data link (layer 2) connection between a terminal and the network. The TEI is a terminal logical address that is used by the MISP to address a terminal during the exchange of layer 2 information messages with that terminal. Each logical terminal is associated with one unique TEI. Up to 20 TEIs can be assigned to the logical terminals on one DSL.

The system provides two types of TEIs based on their assignment method. These are:

- dynamic TEI, automatically assigned by the MISP
- static TEI, entered into the terminal by the user on the terminal key pad

Dynamic TEI

Terminals supporting the dynamic TEI assignment receive their TEI automatically when the terminal is connected to the DSL. The MISP detects the terminal on the loop and assigns to it an unassigned TEI. The range of the automatically assignable TEI numbers is from 64 to 126. TEI 127 is used for sending broadcast messages. A different TEI may be dynamically assigned by the system every time it is initialized.

Static TEI

The terminals that do not support dynamic TEI assignment use static TEI assignment. The TEI can be uniquely identified at the data link layer 2). It can be assigned to one logical terminal at a time, that is, there is a one-to-one mapping of TEI to logical terminal.

The static TEI assignment is performed by entering an unassigned TEI number from 0 to 63 directly into the terminal using its key pad. This TEI is assigned to that terminal as long as the terminal is operational. If the terminal becomes inoperative, the associated DSL has to be reinitialized.

Note:

A packet data terminal must be of the static TEI type.

Figure 26: Multiple logical terminals on one physical termination on page 122 illustrates how a single physical termination may actually connect multiple logical terminals. Each ST-interface DSL can support up to eight physical terminations, and up to 20 logical terminals. Each logical terminal has assigned one unique TEI, which represents the layer 2 logical address for that terminal.

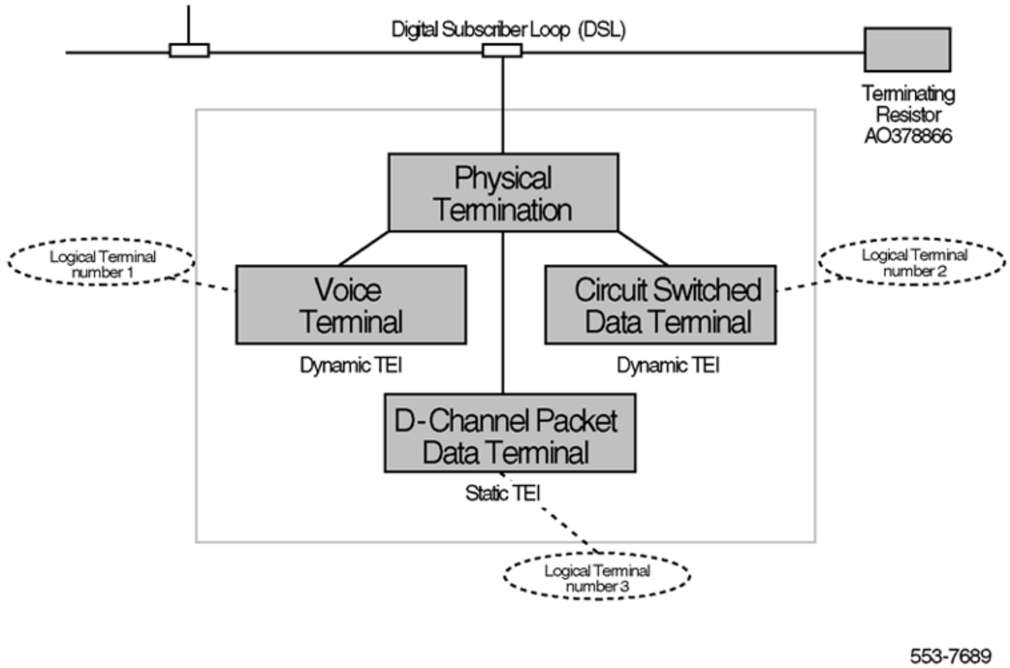


Figure 26: Multiple logical terminals on one physical termination

Types of ISDN BRI terminals

ISDN BRI terminals are divided into four categories based on layer 3 and layer 2 initialization procedures:

- initializing terminal with dynamic TEI assignment
- initializing terminal with static TEI assignment
- non-initializing terminal with dynamic TEI assignment
- non-initializing terminal with static TEI assignment

Initializing terminal with dynamic TEI assignment

Each initializing terminal has an identification number called a Service Profile ID (SPID) that is entered into the terminal by the user when the terminal is installed. This number is usually the directory number with one or two alphanumeric characters appended to it, although it can be any alphanumeric number up to 20 digits long. The SPID is used by the MISP to identify the terminal and to assign to it specific service attributes during layer 3 initialization.

Before layer 3 terminal initialization can start, layer 2 must be fully established, which includes TEI assignment. The TEI may be Dynamic (the MISP assigns an unassigned TEI) or Static (the TEI in manually entered on the terminal key pad). The terminal must then have its SPID number entered at the terminal key pad.

Layer 3 initialization with dynamic TEI assignment starts when the terminal transmits its SPID to the MISP using an information message. The MISP acknowledges the message and sends an end-point identifier message that contains two identification parameters; the User Service Identifier (USID) and the Terminal Identifier (TID).

Initializing terminal with static TEI assignment

For an initializing terminal that does not support the dynamic TEI assignment, the end-point identification parameters USID and TID are not automatically assigned by the MISP. Before layer 3 terminal initialization can start, the terminal must have its SPID entered at the terminal keypad.

Layer 3 initialization starts when the terminal transmits its SPID to the MISP using an information message. The MISP acknowledges the message and assigns a TSP to the terminal.

Non-initializing terminal with dynamic TEI assignment

A non-initializing terminal does not support the dynamic TEI assignment and is not associated with a SPID number. However, non-initializing terminals may support dynamic TEI assignment where the MISP automatically assigns an unassigned TEI when the terminal is installed or when the system or the cards are reset.

The range of the automatically assignable TEI numbers is from 64 to 126. Because these terminals do not support layer 3 initialization procedures, the MISP assigns the same default TSP to all terminals of this type on a specific DSL. The default TSP is defined by specifying USID = 0 in Overlay 27.

Non-initializing terminal with static TEI assignment

A non-initializing terminal does not support the dynamic TEI assignment and is not associated with a SPID number. The non-initializing terminals may support static TEI assignment where the user assigns an unassigned TEI by entering the TEI number on the terminal key pad when the terminal is installed or when the system or the cards are initialized.

The static TEI assignment is performed by entering an unassigned TEI number from 0 to 63 directly into the terminal using its key pad. Because these terminals do not support layer 3 initialization procedures, the MISP assigns the same default TSP to all terminals of this type on a specific DSL. The default TSP is defined by specifying USID = 0 in overlay 27.

Figure 27: Terminal initialization and service profile assignment examples on page 124 shows different types of terminals and their relationship to each other when they are connected to the same DSL. It also shows how the terminal initialization parameters are handled for different types of terminals.

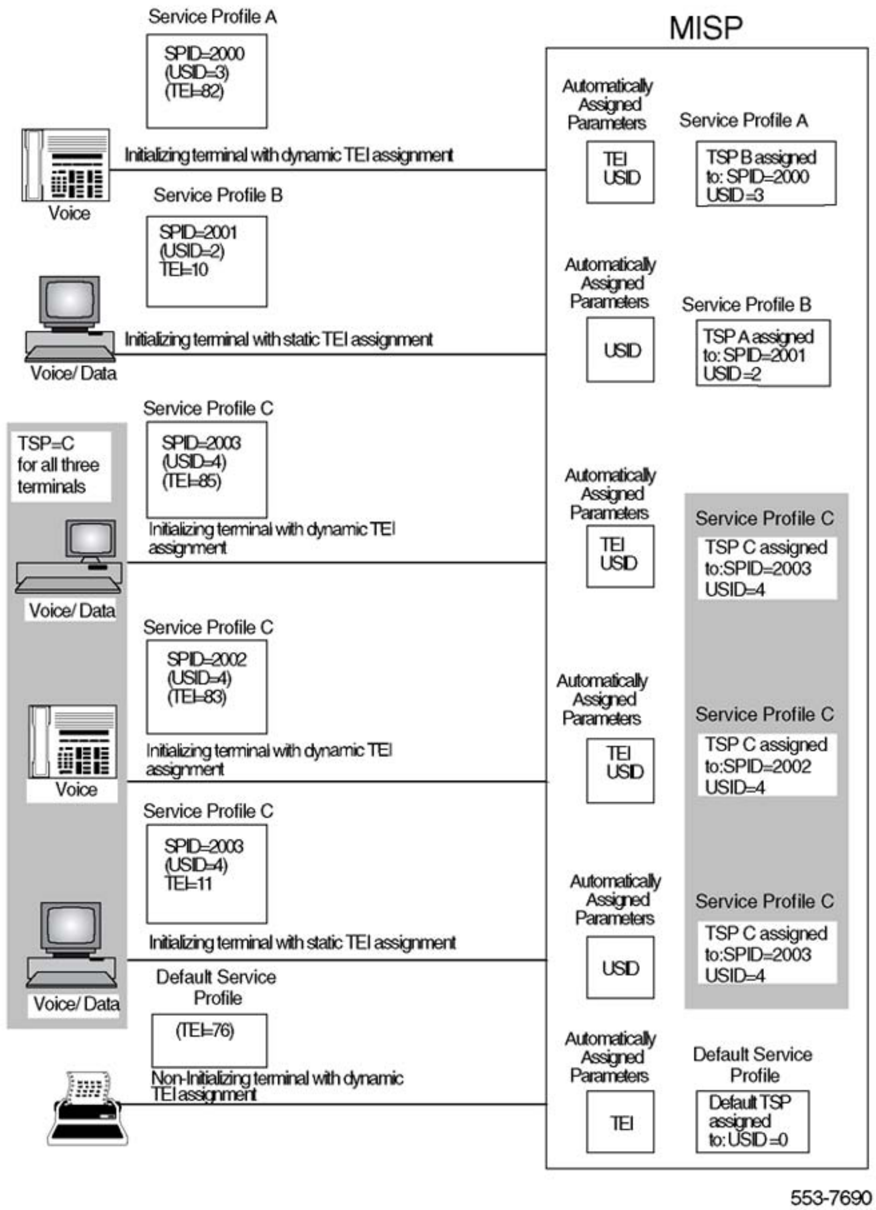


Figure 27: Terminal initialization and service profile assignment examples

ISDN BRI terminal interface specification

ISDN BRI provides two types of interfaces. They are:

- S/T interface
- U interface

A terminal connected to an interface over a DSL must meet the interface specification requirements. Each terminal must provide a jack of the appropriate type and with the appropriate pinouts for the interface.

S/T interface specification

The S/T interface uses an 8-conductor modular cable terminated with an 8-pin RJ-45 plug. An 8-pin RJ-45 jack located on the terminal is used to connect the terminal to the DSL using this modular cable.Table 15: S/T interface connector specification, from NT1 to terminal on page 125 shows the connector pin assignment for the jack and the plug, from the NT1 side to the terminal. It also shows the signal names for each interface pin at the SILC and at the terminal.

Note:

Power Sink 2 provides an optional means of powering the terminal from a common supply in the wiring closet. Power Source 3 provides the power from the terminal to the NT1 if the NT1 does not have a local power source. Up to 2 watts of power is supplied by the SILC to the terminals on the DSL. This power is simplexed over the Tx and Rx pairs provided by -48 V (-40 V for Europe) supply on the SILC. The Rx pair is positive with respect to the Tx pair.

Table 15: S/T interface connector specification, from NT1 to terminal

Pin number	Terminal pin signal name	SILC pin signal name
1	Power Source 3	Not applicable
2	Power Source 3	Not applicable
3	Tx +	Rx +
4	Rx +	Tx +
5	Rx -	Tx -
6	Tx -	Rx -
7	Power Sink 2 (-)	Not applicable
8	Power Sink 2 (+)	Not applicable

U interface specification

The U interface uses a 2-conductor twisted pair cable terminated with an RJ-45 jack. An RJ-45 jack located on the terminal is used to connect the terminal to the DSL using this twisted pair cable.

The connector pin assignments for the jack and plug are shown in Table 16: U interface connector specification on page 126. The table also shows the signal names for each interface pin at the UILC and at the terminal.

Note:

The U interface meets all the safety protection requirements specified by UL (1459), CSA, TUV, and FCC (68.302 and 68.304). These requirements provide protection against inside and outside plant foreign voltages. In addition to other protection components used on the board, 1-Amp (125V voltage rating) fuses are used in series to ensure all the safety requirements. These fuses must be replaced if needed with the same type and rating only to continuously protect against the risk of fire.

Table 16: U interface connector specification

Pin Number	Terminal Pin Signal Name	UILC Pin Signal Name
1	Not used	Not applicable
2	Not used	Not applicable
3	Not used	Not applicable
4	Transmit or Receive	Transmit or Receive
5	Transmit or Receive	Transmit or Receive
6	Not used	Not applicable
7	Not used	Not applicable
8	Not used	Not applicable

Compatible ISDN BRI terminals

The list of terminals deemed compatible can change without notice. Refer to Table 17: ISDN BRI terminals on page 126. To obtain the latest list of compatible terminals and ordering codes, contact your Avaya representative.

Table 17: ISDN BRI terminals

Terminal type	Description
M5317TDX	A Meridian 1 proprietary telephone equipped with voice transmission options and circuit-switched or packet data options. **Note:** M5317TX is voice only.
M5209TDcp	A Meridian 1 proprietary telephone equipped with voice transmission options and circuit-switched or packet data options.

Terminal type	Description
	Note: M5209T is voice only.
M5000TD-1	ISDN Terminal Adapter provides a connection to an analog (500/2500-type) telephone and supports circuit-switched or packet data.

Chapter 6: ISDN BRI implementation

Contents

This section contains information on the following topics:

Task summary list

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

1. LD 73 - Configure pad tables (optional).
2. LD 27 - Add or change an LAPD protocol group for a line.
3. LD 27 - Remove an LAPD protocol group for a line.

4. LD 27 - Print an LAPD protocol group.
5. LD 27 - Add or change a MISP for a line.
6. LD 27 - Remove a MISP configured for a line.
7. LD 22 - Print a MISP configured for a line.
8. LD 27 - Add or change a BRSC for a line.
9. LD 27 - Remove a BRSC card configured for a line.
10. LD 22 - Print a BRSC configured for a line.
11. LD 27 - Add or change a SILC or UILC configured for a line.
12. LD 27 - Remove a SILC or UILC.
13. LD 27 - Print a SILC or UILC configured for a line.
14. LD 27 - Add or change a DSL for a line.
15. LD 27 - Remove a DSL configured for a line.
16. LD 27 - Print a DSL configured for a line.
17. LD 27 - Add or change a TSP for a line.
18. LD 27 - Remove a TSP configured for a line.
19. LD 27 - Print a TSP configured for a line.
20. LD 27 - Add or change an LAPD protocol group.
21. LD 27 - Remove an LAPD protocol group.
22. LD 27 - Print an LAPD protocol group.
23. LD17 - Add an ISDN PRI loop for an external packet handler.
24. LD15 - Define an ISDN customer for an external packet handler.
25. LD 16 - Configure a TIE trunk route for packet data for an external packet handler.
26. LD14 - Configure the TIE trunk for packet data for an external packet handler.
27. LD 27 - Add or change a MISP configured for an external packet handler.
28. LD 27 - Remove a MISP configured for an external packet handler.
29. LD 22 - Print a MISP configured for an external packet handler.
30. LD 27 - Add or change a BRSC for an external packet handler.
31. LD 27 - Remove a BRSC configured for an external packet handler.
32. LD 22 - Print a BRSC configured for an external packet handler.
33. LD 27 - Add or change a SILC or UILC for an external packet handler.
34. LD 27 - Remove a SILC or UILC configured for an external packet handler.
35. LD 27 - Print a SILC or UILC configured for an external packet handler.

36. LD 22 - Add or change a DSL for an external packet handler.
37. LD 27 - Remove a DSL configured for an external packet handler.
38. LD 27 - Print a DSL configured for an external packet handler.
39. LD 27 - Add or change a TSP for an external packet handler.
40. LD 27 - Remove a TSP configured for an external packet handler.
41. LD 27 - Print a TSP configured for an external packet handler.
42. LD 27 - Add or change an LAPD protocol group.
43. LD 27 - Remove an LAPD protocol group.
44. LD 27 - Print an LAPD protocol group.
45. LD 27 - Add or change an LAPB protocol group.
46. LD 27 - Remove an LAPB protocol group.
47. LD 27 - Print an LAPB protocol group.
48. LD 27 - Add or change an X.25 protocol group.
49. LD 27 - Remove an X.25 protocol group.
50. LD 27 - Print an X.25 protocol group.
51. LD 27 - Add or change a DNA table for an MPH.
52. LD 27 - Remove a DNA table.
53. LD 27 - Print a DNA table configured for an MPH.
54. LD17 - Add an ISDN PRI loop for an external MPH.
55. LD15 - Define an ISDN customer for an MPH.
56. LD 16 - Configure a TIE trunk route for packet data for an MPH.
57. LD14 - Configure the TIE trunk for packet data for an MPH.
58. LD 16 - Configure a TIE trunk route for an MCU to MPH interface.
59. LD14 - Configure the TIE trunk for an MCU to MPH interface.
60. LD 11 - Add or change an MCU to MPH interface.
61. LD11 - Remove an MCU to MPH interface.
62. LD 27 - Add or change a MISP configured for an MPH.
63. LD 27 - Remove a MISP configured for an MPH.
64. LD 22 - Print a MISP configured for an MPH.
65. LD 27 - Add or change a BRSC configured for an MPH.
66. LD 27 - Remove a BRSC configured for an MPH.
67. LD 27 - Print a BRSC configured for an MPH.

68. LD 27 - Add or change a SILC or UILC for an MPH.
69. LD 27 - Remove a SILC or UILC configured for an MPH.
70. LD 27 - Print a SILC or UILC configured for an MPH.
71. LD 27 - Add or change a DSL for an MPH.
72. LD 27 - Remove a DSL configured for an MPH.
73. LD 27 - Print a DSL configured for an MPH.
74. LD 27 - Add or change a TSP for an MPH.
75. LD 27 - Remove a TSP configured for an MPH.
76. LD 27 - Print a TSP.
77. LD 27 - Add or change PVC connection configuration.
78. LD 27 - Remove PVC connection configuration data.
79. LD 27 - Print PVC connection configuration data.
80. LD14 - Add or change a tandem connection configuration.
81. LD14 - Remove a tandem connection configuration.
82. LD14 - Print a tandem connection configuration.
83. LD15 - Add Call Detail Recording for the MPH.
84. LD15 - Define a customer for a trunk.
85. LD 73 - Configure pad tables (optional).
86. LD 27 - Add or change an LAPD protocol group for a trunk.
87. LD 27 - Remove an LAPD protocol group for a trunk.
88. LD 27 - Print an LAPD protocol group for a trunk.
89. LD 16 - Configure ISDN BRI trunk route parameters.
90. LD 27 - Add or change trunk for a trunk.
91. LD 27 - Remove a MISP configured for a trunk.
92. LD 22 - Print a MISP configured for a trunk.
93. LD 27 - Add or change a SILC or UILC for a trunk.
94. LD 27 - Remove a SILC or UILC configured for a trunk.
95. LD 27 - Print a SILC or UILC configured for a trunk.
96. LD 27 - Add or change a DSL for a trunk.
97. LD 27 - Remove a DSL configured for a trunk.
98. LD 27 - Print a DSL configured for a trunk.

99. LD 73 - Configure a trunk clock reference source, for 1.5 Mb PRI/DTI.
100. LD 73 - Configure trunk clock reference source, for 2.0 Mb PRI/DTI.

Introduction

I line and packet data implementation, and ISDN BRI trunk access implementation. It lists the order in which these procedures should be performed and provides a detailed description of each procedure showing the prompts that are displayed and the responses to each prompt.

ISDN BRI line application is configured using Overlays 27 and 73 (optionally used for configuring a pad table). This includes any requirements needed to configure ISDN BRI sets for ISDN features which are non-BRI specific. For example, ISDN BRI sets are supported on the ISDN QSIG Call Diversion feature, so Overlay 27 shows those prompts that are required to configure ISDN BRI sets for this feature. The Terminal Service Profile (TSP) configuration is to be used.

The ISDN BRI Trunk Access capability is administered using Overlays 16, 27, and 73 (optionally used for configuring a pad table and setting clock referencing). This includes any requirements needed to configure ISDN BRI trunks for ISDN features which are non-BRI specific. For example, ISDN BRI trunks are supported on the ISDN QSIG Call Diversion feature, so Overlay 16 shows those prompts that are required to configure ISDN BRI trunks for this feature.

Configure ISDN BRI line application

Configuration order for line application

You must configure the following components in the order listed below to configure ISDN BRI lines.

Please note that when changing existing ISDN BRI service, following this order is unnecessary. Be aware, though, of the relationship of one component to another and whether changing one component necessitates changing other components.

1. Configure a pad table using LD 73 (optional).

   This step is optional; if no pad values are configured the default values will be used.

Note:

Pad table are used for lines only when the protocol to be used on a DSL is set to ETSI NET-3, INS NET 64, QSIG, or Numeris.

2. Configure a Link Access Procedure on the D-channel (LAPD) Group using LD 27.
3. Configure the MISP using LD 27.
4. Configure the BRSC using LD 27 (optional).
5. Configure the SILC or UILC using LD 27.

   This step is optional. The SILC or UILC can also be configured when configuring the DSL (see next step).
6. Configure the DSL using LD 27.
7. Configure the TSP using LD 27.
8. Program ISDN BRI terminals (M5317TDX, M5209TDcp).

Configure pad tables (optional)

This step is optional; if no pad values are configured the default values will be used.

Note:

Pad table are used for lines only when the protocol to be used on a DSL is set to ETSI NET-3, INS NET 64, QSIG, or Numeris.

The digital pad provides gain or attenuation values to condition the level of the digitized transmission signal according to the network loss plan. This determines transmission levels for the B-channel circuit-switched voice calls.

Table 18: LD 73 - Configure pad tables. (optional)

Prompt	Response	Description
REQ	NEW	New settings
TYPE		Pad table type.
	BRIL	Enter BRIL
FEAT	PAD	Set the pad values used for ISDN BRIL
PDCA	1-16	Pad category table.
DFLT	(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
		The following prompts define the pad levels. The receiving pad code is r and the transmission pad code is t. These entries have the range 0-26. 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
OPX	r ... t	Off-premises extension
DTT	r ... t.	Digital TIE trunks
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	2Mb DTI digital TOLL office trunks
ACO	r ... t	Analog local exchange or WATS trunks
AFX	r ... t	Analog FEX trunks
ADD	r ... t	Analog DID trunks
SATT	r ... t	Analog satellite TIE trunks
ATO	r ... t	Analog TOLL office trunks
PRI2	r ... t	2Mb PRI trunk (prompted only if the 1.5/2Mb Gateway feature is equipped and TYPE=2Mb PRI)
XUT	r ... t	Analog local exchange trunk (prompted only if the 1.5/2Mb Gateway feature is equipped and TYPE=PRI2)
XEM	r ... t	Analog TIE trunk (prompted only if the 1.5/2Mb Gateway feature is equipped and TYPE=PRI2)
BRIL	r ... t	ISDN BRIL pad values. Valid inputs are 0-26. Refer to Table 19: ISDN BRI trunk pad codes and values on page 137.
ADD	r ... t	Analog DID trunks
SATT	r ... t	Analog satellite TIE trunks
ATO	r ... t	Analog TOLL office trunks
PRI2	r ... t	2Mb PRI trunk (prompted only if the 1.5/2Mb Gateway feature is equipped and TYPE=2Mb PRI)

Prompt	Response	Description
XUT	r ... t	Analog local exchange trunk (prompted only if the 1.5/2Mb Gateway feature is equipped and TYPE=PRI2)
XEM	r ... t	Analog TIE trunk (prompted only if the 1.5/2Mb Gateway feature is equipped and TYPE=PRI2)
BRIL	r ... t	ISDN BRIL pad values. Valid inputs are 0-26. Refer to Table 19: ISDN BRI trunk pad codes and values on page 137.

Table 19: ISDN BRI trunk pad codes and values on page 137 shows ISDN BRI pad codes and their values. Positive dB represents loss and negative dB represents gain.

Table 19: ISDN BRI trunk pad codes and values

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					

Configure an LAPD protocol group for line

Add a protocol group by using LD 27 and specifying its protocol group number. You can also change its LAPD parameters as needed or accept the default values. LAPD is a transmission protocol that specifies the transmission timers, the maximum number of retransmissions, the size of the data frame, and the number of negative acknowledgments allowed before the system issues an alarm.

Table 20: LD 27 - Add or change an LAPD protocol group for a line.

Prompt	Response	Description
REQ	NEW	Add an ISDN protocol group.
TYPE	LAPD	LAPD Protocol group

Prompt	Response	Description
PGPN	0-15 <cr>	Protocol group number The values for this prompt are: 0-15 = Adds a specified protocol group <cr> = Stops this prompt from being displayed again
LAPD	YES NO	LAPD parameters —The values for this prompt are: YES = Define or modify the LAPD parameters NO = Does not prompt the LAPD parameters and assigns the default values shown in ( ) to these parameters.
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.
N2X4	0-(10)-20	For 1TR6 connectivity - number of status inquiries when the remote station is in peer busy state.
PGPN	<cr>	Press <cr> to prevent repetition of all the parameters starting with LAPD.

Remove an LAPD protocol group for a line

You can remove an LAPD protocol group as long as it is not assigned to a DSL. If a protocol group is assigned to a DSL, delete the DSL before removing the protocol group.

Table 21: LD 27 - Remove an LAPD protocol group for a line.

Prompt	Response	Description
REQ	OUT	Remove an ISDN BRI component.
TYPE	LAPD	Protocol group
PGPN		Protocol group number
	0-15 ALL <cr>	0-15 = Removes a specified protocol group from 0-15 ALL = Removes all protocol groups <cr> = No change the protocol group is not removed. A protocol group cannot be removed if it is assigned to a DSL.
...	...	

Print an LAPD protocol group for a line

Configuration information for a specific LAPD protocol group or for all protocol groups can be printed.

Table 22: LD 27 - Print an LAPD protocol group.

Prompt	Response	Description
REQ	PRT	Prints an ISDN BRI component
TYPE	LAPD	Protocol group
PGPN		Protocol group number
	0-15 <cr>	0-15 = Prints a specified protocol group from 0-15 <cr> = Prints all protocol groups and the number of DSLs in each group
USER	YES (NO)	YES = Print the LAPD group(s) selected in the PGPN prompt and the DSLs that are using it (them). NO = Do not print the LAPD user information.
REQ	...	

Configure a MISP for a line

The ISDN BRI line application or the Meridian 1 Packet Handler (MPH) application can be downloaded to the MISP hardware. If the hardware runs only the ISDN BRI line application, it functions as a stand-alone MISP. In this configuration, the MISP can support the signaling processing for four ISDN BRI line cards without association with a BRSC or it can support up to 120 ISDN BRI line cards with the maximum eight BRSCs.

The procedure which follows indicates how to add or change a MISP for line application. To add or change a MISP, specify its even loop number.

The MISP must be enabled by using the **`ENLL l`** command in Network and IPE Diagnostic Program LD 32.

Table 23: LD 27 - Add or change a MISP for a line.

Prompt	Response	Description
REQ	NEW CHG	Add or change an ISDN BRI MISP **Note:** The defaults apply to adding, not changing, a MISP.
TYPE	MISP	MISP

Prompt	Response	Description
LOOP	0-254	MISP loop number; must be an even number, with the next odd loop number unequipped.
APPL	BRIL XBRIL <cr>	BRIL = ISDN BRI line application XBRIL = Remove the ISDN BRI line application <cr> = None. Enter BRIL for ISDN BRI line application. APPL is prompted until <cr> is entered.
DSPD	YES (NO)	YES = D-channel Packet Switched Data NO = No D-channel Packet Switched Data. Use the default value NO. Subsequent prompts are skipped.

Remove a MISP configured for a line

Before removing the MISP which has been configured for a line:

- Remove all BRSCs associated with it, if applicable.
- Remove all DSLs connected to SILCs and UILCs associated with it.
- Disable the MISP loop with the DISL l command in LD 32.

Table 24: LD 27 - Remove a MISP configured for a line.

Prompt	Response	Description
REQ	OUT	Remove an ISDN BRI component
TYPE	MISP	Enter MISP to remove the MISP.
LOOP	0-254	Loop number of the MISP to be removed (must be an even number). The MISP must be disabled before being removed. All BSRCs, SILC and/or UILC DSLs associated with the MISP must be removed before removing the MISP. See "Remove a BRSC configured for a line" or "Remove a SILC or UILC configured for a line" in this chapter.

Print a MISP configured for a line

Print the configuration information for a MISP which has been configured for a line by specifying its network loop number. If the MISP network loop number is not known, use LD 22 to print the system configuration.

Table 25: LD 22 - Print a MISP configured for a line.

Prompt	Response	Description
REQ	PRT	Prints an ISDN BRI component
TYPE	MISP	MISP **Note:** BRSC TNs associated with the MISP are also printed.
LOOP	0-254	Loop number (must be an even number).

Configure a BRSC for a line

The Basic Rate Signaling Concentrator (BRSC) enhances the capacity of the ISDN BRI lines on the system by off-loading some of the signaling processing from the MISP to the BRSC. Each BRSC can support 120 DSLs. This increases DSL capacity for the MISP from 32 to 976.

Without a BRSC configured, a MISP can support up to four SILCs and UILCs in any combination. With a BRSC configured, a MISP can support the following maximum combinations:

- three line cards, one BRSC
- two line cards, eight BRSCs

The maximum number of DSLs that a MISP configured with a BRSC is 976. This figure is derived as follows:

- 1 MISP supports 8 BRSCs and 2 line cards (SILC/UILCs)
- 1 BRSC supports 15 SILC/UILC cards, each having 8 ports:
  - total (8*15) = 120
- 1 SILC/UILC card has 8 ports
  - total (8*120) = 960
- 2 SILC/UILC cards each having eight ports
  - total (8*2) = 16

Therefore, total number of DSLs = 960+16 = 976.

To configure a BRSC:

- Disable the MISP under the following conditions:
  - The first BRSC is configured.

- You add the first BRSC to an IPE module with two or more configured ISDN BRI line cards. The MISP does not have to be disabled when adding BRSCs 2-8 to this configuration.
- You change from a configuration with three line cards and two BRSCs to a configuration with two line cards and up to eight BRSCs.

- Disable all ISDN BRI line cards in an IPE Module
- Configure a BRSC in the IPE Module; specify its superloop number, shelf number, and card number. Since the BRSC handles Dchannel Packet Switched Data (DPSD), specify the PRI loop and channel numbers for routing of the DPSD to an external packet handler.
- Select a MISP that can accommodate the BRSC
- Enable the MISP
- Enable the BRSC with the **`ENLC lll s cc`** command in LD 32

Table 26: Add or remove configured BRSCs on page 142 lists the possible cases for adding or removing BRSCs with a system already configured for ISDN BRI.

Table 26: Add or remove configured BRSCs

Initial configuration				
Configured line cards installed	**BRSCs**	**Action**	**Same IPE Module?**	**Disable MISP?**
0 0	0 1	Add BRSC 1 Add BRSC 2-8	NA NA	Yes No
1 1	0 1	Add BRSC 1 Add BRSC 2-8	Yes/No NA	Yes No
2 2	0 1	Add BRSC 1 Add BRSC 2-8	Yes/No Yes/No	Yes No
3	0	Add BRSC 1	Yes/No	Yes
3 2 0-2	0 1 2	Add BRSC 1 Add BRSC 2 Add BRSC 3-8	No Yes Yes/No	Yes Yes No
4 3 0-2	0 1 3-8	Add BRSC 1 Add BRSC 2 Add BRSC 3-8	Yes Yes Yes/No	Yes Yes No
2 2 3 3	2-8 1 1 0	Disable BRSC 1 Add line card 3 Delete BRSC Add line card 4	N/A N/A N/A N/A	No Yes No Yes
2 2 3	2-8 0 0	Delete BRSCs Add line card 3 Add line card 4	N/A N/A N/A	No Yes No

To add or change a BRSC for a line application, specify its superloop number, shelf number, and card number.

Enable the BRSC with the **`ENLC lll s cc`** command in LD 32.

Table 27: LD 27 - Add or change a BRSC for a line.

Prompt	Response	Description
REQ	NEW CHG	Add or change a SILC or UILC line card
TYPE	BRSC	SILC or UILC line card
BRSC	lll s cc	Card location The values for this prompt are: lll (loop) = 0-156 (must be an even number divisible by 4) s (shelf) = 0-1 cc (card) = 0-15
MISP	0-254	MISP loop number (must be an even number that has already been configured)
DPSD	YES (NO)	YES = D-channel Packet Switched Data NO = No D-channel Packet Switched Data Use the default value NO. All subsequent prompts are repressed.

Remove a BRSC configured for a line

Remove a BRSC which has been configured for a line by specifying its loop number. Before removing the BRSC, disable all line cards associated with it.

Before removing the BRSC, disable the BRSC loop with the **`DISC lll s cc`** command in LD 32; also, disable all line cards associated with the BRSC.

Table 28: LD 27 - Remove a BRSC card configured for a line.

Prompt	Response	Description
REQ	OUT	Remove an ISDN BRI component
TYPE	BRSC	Remove the BRSC data
BRSC	lll s cc	Card location The values for this prompt are: lll (loop) = 0-156 (must be an even number divisible by 4) s (shelf) = 0-1 cc (card) = 0-15
REQ	...	

Print a BRSC configured for a line

Print the configuration information for a BRSC configured for a line by specifying its network loop number. If the BRSC network loop number is not known, use LD 22 to print the system configuration.

To print all BRSCs associated with a MISP, enter <cr> at the BRSC and MISP prompts.

Table 29: LD 22 - Print a BRSC configured for a line.

Prompt	Response	Description
REQ	PRT	Prints an ISDN BRI component
TYPE	BRSC	Print BRSC data **Note:** BRSC TNs associated with the MISP are also printed.
BRSC	lll s cc	Card location The values for this prompt are: lll (loop) = 0-156 (must be an even number divisible by 4) s (shelf) = 0-1 cc (card) = 0-15
MISP	0-254 <cr>	MISP loop number (must be an even number that has already been configured). If <cr> is entered, all BRSCs configured in the system are printed; otherwise, all BRSCs associated with the MISP loop are printed.
REQ	...	

Configure a SILC or UILC for a line

Add or change a new SILC or UILC for ISDN BRI line application by specifying its location, card type, and the MISP network loop that this card uses to transmit and receive signaling and D-channel packet data.

Note:

This step may be skipped and the card type specified when configuring the DSL in the procedure "Add a DSL for a line."

The following procedure is used when configuring the SILC or UILC cards without configuring their DSLs.

Note:

If there is a BRSC configured in the IPE module, the MISP prompt is skipped and the MISP lll and the BRSC lll s cc is displayed.

Table 30: LD 27 - Add or change a SILC or UILC configured for a line.

Prompt	Response	Description
REQ	NEW CHG	Add or change a SILC or UILC line card
TYPE	CARD	SILC or UILC line card
TN		Terminal Number
	l s c	Format, where: l = • 0-156 (must be an even number divisible by 4) s = • 0–1 c = • 0-15 for Large System, 1-4, 7-10 for chassis, 1-10 for cabinet and MG 1010
MISP	0-254	Loop number (must be an even number that has already been configured) for Large Systems **Note:** If there is a BRSC configured in the IPE module, the MISP prompt is skipped and the MISP lll and the BRSC lll s cc is displayed.
CTYP	SILC UILC	Card type to be added or changed. Remove any DSLs configured for this line card before changing the card type.
REQ	...	

Remove a SILC or UILC configured for a line

Remove a SILC or UILC which has been configured for a line by specifying its card location. Before removing the SILC or UILC, all configured DSLs must first be removed from the card by using the procedure "Remove a DSL configured for a line". When the last DSL is removed, the card is automatically deleted.

When removing the card, the database information is also deleted from the data block. Use LD 20 to list cards that have been removed.

Table 31: LD 27 - Remove a SILC or UILC.

Prompt	Response	Description
REQ	OUT	Remove an ISDN BRI component
TYPE	CARD	ISDN BRI line card

Prompt	Response	Description
TN		Terminal Number
	l s c	Format, where: l = • 0-156 (must be an even number divisible by 4) s = • 0–1 c = • 0-15 for Large System, 1-4, 7-10 for chassis, 1-10 for cabinet and MG 1010

Print a SILC or UILC configured for a line

To print the configuration information for a SILC or UILC, specify its card location.

Table 32: LD 27 - Print a SILC or UILC configured for a line.

Prompt	Response	Description
REQ	PRT	Print an ISDN BRI component
TYPE	CARD	ISDN BRI line card
TN		Terminal Number
	l s c	Format, where: l = • 0-156 (must be an even number divisible by 4) s = • 0–1 c = • 0-15 for Large System, 1-4, 7-10 for chassis, 1-10 for cabinet and MG 1010

Configure a DSL for a line

To add or change a DSL for a line application, specify its port location and its DSL characteristics. DSL location specifies a SILC/UILC port connected to a DSL.

Table 33: LD 27 - Add or change a DSL for a line.

Prompt	Response	Description
REQ	NEW CHG	Add or change a DSL. **Note:** The defaults apply to adding, not changing, a DSL.
TYPE	DSL	Digital Subscriber Loop
DSL	lll s cc dsl#	DSL location, where: • lll (superloop) = 0-156 (must be zero or a number divisible by 4) • s (shelf) = 0-1 • cc (card) = 0-15 and for CS 1000E 1–20 • dsl# (DSL location) = 0-7 Assign 31 DSLs for each MISP if DCH or BDCH were specified at the PH prompt in "MISP configuration procedures".
APPL	BRIL	ISDN BRI line application.
DES	x...x	Designator to assign to a DSL (ex. BUILD2) x...x = 1 to 6 alphanumeric DSL designator
CUST	xx	Customer number, as defined in LD 15
CTYP	SILC UILC	Card type. This prompt is displayed only if the SILC or UILC has not been previously configured using the "SILC or UILC configuration procedures," or "Add a DSL" procedure when configuring another DSL on the same SILC/UILC.
MISP	0-254	Loop number (must be an even number of a MISP that has already been configured).
	1-9	MISP card slot number has to be 1-4, 7-10 for Media Gateway 1000 1-10 for Media Gateway 1010
MODE	NTAS NTFS	Network terminal line sampling mode (this prompt is displayed only if you specified the card type as SILC). The values for this prompt are: NTAS=Adaptive sampling Extended passive bus, Branched passive bus, Point-to-point bus, U interface DSL. NTFS=Fixed sampling Short passive bus.
B1CT	(VCE) (DTA)	B-channel 1 call type
	<cr>	VCE = circuit switched voice DTA = circuit switched data Enter <cr> to select voice and data as defaults.
B2CT	(VCE) (DTA)	B-channel 2 call type
	<cr>	VCE = Circuit switched voice DTA = Circuit switched data

Prompt	Response	Description
		Enter <cr> to select voice and data as defaults.
LDN	0-3 (NO)	Departmental listed directory number 0-3 = Departmental listed DN specified in LD15 NO = No departmental listed DN associated with the DSL
XLST	(0)-254	Pretranslation group (if configured in customer data block).
MTEI	1-(8)-20	Maximum number of Terminal Endpoint Identifiers, both static and dynamic combined assigned to the logical terminals on this DSL.
MCAL	2-(16)-32	Maximum number of calls on the DSL at one time. This includes calls waiting and on hold. Warning is received if less than 8 is specified.
MTSP	1-(8)-16	Maximum number of TSPs allowed for a DSL
PGPN	0-15	Protocol group number (no default value) The protocol group should be previously added as described in "Configure a protocol group for a line."
PRID	1- 6	Defines the protocol to be used on the DSL Selection of the protocol ID is terminal dependent. The values for this prompt are: 1 = ANSI 2 = ETSI 3 = DMS 4 = NET64 5 = NUMERIS 6 = NI-1 **Note:** A response of 6 allows the ISDN BRI Conference feature to be configured in the TSP of the DSL.
PDCA	1-16	Pad category table, defined in LD 73. Prompted if PRID = 2 or 4.
FDN	n...n	Flexible CFNA directory number. Enter a 1-13 digit DN.
EFD	n...n	Flexible external call CFNA DN. Enter a 1-13 digit DN.
HUNT	n...n	Hunt directory number. Enter a 1-13 digit DN.
EHT	n...n	Hunt external call directory number. Enter a 1-13 digit DN.
TGAR	(0)-31	Trunk group access restriction
NCOS	(0)-99	Network Class Of Service
SGRP	(0)-999	Scheduled Access Restriction Group Number.
CLS		Class of Service access restrictions. More than one Class of Service can be entered by separating each entry with a space. Default features shown in parenthesis are selected by pressing <cr>.
	(ABDD) ABDA	Abandoned call record and time to answer denied Abandoned call record and time to answer allowed
	(ICDD) ICDA	Internal Call Detail Recording (Denied) Allowed

Prompt	Response	Description
	(MRD) MRA	Message Restriction (Denied) Allowed
	(UDI) RDI	(Unrestricted) Restricted DID
	(UNR) (CTD) CUN FR1 FR2 FRE SRE TLD ICDA (ICDD) BRTA (BRTD) <cr>	(Unrestricted) Conditionally Toll Denied Conditionally Unrestricted Fully Restricted class 1 Fully Restricted class 2 Fully Restricted Semi-Restricted Toll Denied Internal Call Detail Recording allowed (Internal Call Detail Recording denied) Brent ISDN2 set allowed (Brent ISDN2 set denied) Enter <cr> to select the defaults. More than one Class of Service may be selected by separating each entry with a space.
REQ	...	

Remove a DSL configured for a line

Remove a DSL configured for a line by specifying its location. To remove a DSL, first remove all the TSPs assigned to this DSL. When the last configured DSL on a card is removed, the card is removed automatically.

Table 34: LD 27 - Remove a DSL configured for a line.

Prompt	Response	Description
REQ	OUT	Remove an ISDN BRI component
TYPE	DSL	Digital Subscriber Loop
DSL		DSL location
	lll s cc dsl#	DSL location, where: • lll (superloop) = 0-156 (must be zero or a number divisible by 4) • s (shelf) = 0-1 • cc (card) = 0-15 and for CS 1000E 1–20 • dsl# (DSL location) = 0-7 Assign 31 DSLs for each MISP if DCH or BDCH were specified at the PH prompt in "MISP configuration procedures".

Print a DSL configured for a line

Print the configuration information for a single DSL by specifying its location.

Table 35: LD 27 - Print a DSL configured for a line.

Prompt	Response	Description
REQ	PRT	Print an ISDN BRI component.
TYPE	DSL	DSL
DSL		DSL location
	lll s cc dsl# lll s cc lll s lll	Format for Large System and CS 1000E system, where: lll s cc dsl# = Prints information for the specified dsl# lll s cc = Prints information for DSLs on the specified card lll s = Prints information for DSLs in the specified shelf lll = Prints information for DSLs on the specified loop
DATE	(<cr>) x y z	Print data and display the last active date, where x = day (1-31), y = month (Jan-Dec), and z = year (1979-9999) specifies the starting date of the data to be displayed or printed.
PAGE	YES (NO)	YES = prints one DSL per page NO = prints without paging
DES	x...x <cr>	1-digit to 6-digit alphanumeric DSL designator No designator for DSLs
NACT	YES (NO)	Activity date is updated to current date.
...	...	

Configure a TSP for a line

The TSP configuration procedures define the service profiles for ISDN BRI terminals connected to a DSL. A service profile specifies the type of transmission, the call restrictions, and the features the terminal can use.

To add or change a TSP to a DSL, specify the DSL location, its transmission characteristics, and the Class of Service for terminals connected to the DSL. If the default value is desired, press the ENTER key.

Table 36: LD 27 - Add or change a TSP for a line.

Prompt	Response	Description
REQ	NEW CHG	Add or change a TSP.
TYPE	TSP	Assign TSP to a DSL
DSL	lll s cc dsl#	DSL location, where: • lll (superloop) = 0-156 (must be zero or a number divisible by 4) • s (shelf) = 0-1

Prompt	Response	Description
		• cc (card) = 0-15 and for CS 1000E 1–20 • dsl# (DSL location) = 0-7 Assign 31 DSLs for each MISP if DCH or BDCH were specified at the PH prompt in "MISP configuration procedures".
USID	0-15	User service identifier 0 is the TSP assigned to non-initializing terminals. The total number of TSPs defined for a DSL cannot exceed the maximum number of TSPs allowed for a DSL as specified by the MTSP prompt in the "DSL configuration procedures." A TSP should be configured for non-initializing terminals. This is done by assigning USID=0 to the TSP.
MPHC	(YES) NO	Route D-channel packet switched data to the Meridian Packet Handler. Enter NO.
SPID	aaa...a <cr> Xaaa...a	Service profile ID aaa...a = any combination of 1-20 alphanumeric characters. <cr> = Stops this prompt from being displayed again. A maximum of 8 valid SPIDs per TSP are allowed. Xaaa...a removes the specified SPID. This prompt appears only if USID = 1-15. It repeats until <cr> is entered, but only up to 8 SPIDs may be entered. This SPID must be entered in the initializing terminal to associate the terminal with a USID.
FEATID	aaa mmm nnn <cr> Xaaa	ID associated with feature aaa, as follows: A03 = 3-party Conference A06 = 6-party conference mmm = Feature Activation ID(1-127) nnn = Feature Indication ID (1-127) (optional; if not entered, the value entered for mmm is assumed) <cr> = Skip the FEATID entry Xaaa = Delete the feature. Feature Activation ID and Feature Indication ID are feature key number assignments configured at the terminal level. Recommended terminal assignments are: - for the M5317TDX: A06 15 - for the M5209TDcp: A06 9
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). The DN cannot be shared by a non ISDN BRI terminal. This prompt is repeated until <cr> is pressed. At least one DN and a maximum of 8 DNs can be assigned to a DSL. The directory number can be associated with multiple TSP.
...		
BCH	1-2	B-Channel (either 1 or 2) to which the TSP is associated.
...		
CT	VCE DTA	Directory number call type VCE=Circuit switched voice DTA=Circuit switched data

Prompt	Response	Description
		One or more call types can be entered by separating each entry with a space. The call types entered must have been specified for the B1CT and B2CT prompts in "DSL configuration procedures."
...		
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.
...		
MCAL	1-(4)-8	Maximum number of calls per DN at one time Defines the maximum number of calls allowed for a directory number, which includes the total number of active calls, calls waiting, and calls on hold.
CLIP	(YES) NO	Calling Line Identification presentation service YES = displays of calling party DN on incoming calls NO = does not display of calling party DN on incoming calls
PRES	(YES) NO	Allows display of calling line identification to far end on outgoing calls. YES = present this DN to the called party on outgoing calls NO = do not present this DN to the called party on outgoing calls
COLP	(NO) YES	Connected Number Information Elements (IEs) is (not) passed from the system to the Terminal Adapter (S_o).
TRANS	(NO) YES	CLID and Connected Number Information Element (IE) are (not) passed from the system to the Terminal Adapter (S_o), if presentation is restricted.
FEAT		Class of Service features
	HTA (HTD) FNA (FND) SFA (SFD) CFTA (CFTD) MWA (MWD) FBA (FBD) HBTA (HBTD)	HTA = Hunt allowed (always assign if terminal has CWT capability) HTD = Hunt denied FNA = Call forward no answer allowed FND = Call forward no answer denied SFA = Second level call forward no answer allowed SFD = Second level call forward no answer denied CFTA = Call forward by call type allowed CFTD = Call forward by call type denied MWA = Message waiting allowed MWD = Message waiting denied FBA = Call forward busy allowed FBD = Call forward busy denied HBTA = Hunting by call type allowed HBTD = Hunting by call type denied
		DNO1/DNO2/(DNO3) = QSIG Call Diversion Notification for calling party where:
	DNO1 DNO2	DNO1 = no notification DNO2 = notification without forwarded-to (diverted) party's number and name
	(DNO3)	(DNO3) = notification with forwarded-to (diverted) party's number and name when available (default).

Prompt	Response	Description
		DNDN/(DNDY) = QSIG Call Diversion Notification for forwarded-to (diverted) party where:
	DNDN	DNDN = no notification of called party's number and name notification
	(DNDY)	(DNDY) = notification with called party's number and name when available (default).
		More than one Class of Service can be entered by separating each entry with a space. Press <cr> to select multiple default features shown in parenthesis.
DFDN	n...n	Default directory number Enter a 1-digit to 7-digit DN. This DN must be defined at the preceding DN prompt A DN can be associated with multiple TSPs. Only one default DN can be defined for a TSP. This DN is sent in the outgoing setup if the terminal does not send a calling line identification number with the outgoing call.
REQ	...	

Remove a TSP configured for a line

Before removing a TSP configured for a line, disable the B-channel.

To remove a single TSP from a DSL, specify the DSL location and the user service identifier. Remove all TSPs from a DSL by entering **ALL** at the USID prompt.

Removal of the TSP disconnects all calls associated with the TSP's D-channel packet switched data and circuit switched voice data.

Table 37: LD 27 - Remove a TSP configured for a line.

Prompt	Response	Description
REQ	OUT	Remove an ISDN BRI component
TYPE	TSP	Terminal Service Profile **Note:** Removal of the TSP disconnects all calls associated with the TSP's D-channel packet switched data and circuit switched voice data.
DSL	lll s cc dsl#	DSL location, where: • lll (superloop) = 0-156 (must be zero or a number divisible by 4) • s (shelf) = 0-1

Prompt	Response	Description
		• cc (card) = 0-15 and for CS 1000E 1–20 • dsl# (DSL location) = 0-7 Assign 31 DSLs for each MISP if DCH or BDCH were specified at the PH prompt in "MISP configuration procedures".
USID	0-15 ALL	User service identifier 0-15 = Removes a specified TSP from 0 to 15 ALL = Removes all TSPs for the specified DSL
REQ	...	

Print a TSP configured for a line

Configuration information can be printed for a TSP which has been configured for a line based on characteristics such as user service identifier, service profile ID, and directory number.

Table 38: LD 27 - Print a TSP configured for a line.

Prompt	Response	Description
REQ	PRT	Print an ISDN BRI component.
TYPE	TSP	Terminal Service Profile
DSL	lll s cc dsl#	DSL location, where: • lll (superloop) = 0-156 (must be zero or a number divisible by 4) • s (shelf) = 0-1 • cc (card) = 0-15 and for CS 1000E 1–20 • dsl# (DSL location) = 0-7 Assign 31 DSLs for each MISP if DCH or BDCH were specified at the PH prompt in "MISP configuration procedures".
OPT	USID SPID SUID DN DNS NTN	USID = Prints the TSP with the specified user service ID SPID = Prints the TSPs with the specified service profile ID SUID = Prints the specified user profile ID and the User Service ID map DN = Prints the TSP(s) that contains the specified DN DNS = Prints all the directory numbers defined for the DSL NTN = Prints the TSPs that contain the specified NTN.
	<cr>	<cr> = Prints all the TSPs defined for the DSL
- USID	0-15	User service identifier
- SPID	aa..aa	Service profile ID Enter a 1-20 alphanumeric service profile ID.
- DN	xxxx (0)-N	xxxx = Directory number associated with the TSP (0)-N = CLID entry, N = SIZE-1 (SIZE defined in LD 15).

Prompt	Response	Description
- NTN	nn..nn	National Terminal Number (1-10 digits)
REQ	...	

Initialize ISDN BRI terminals

After configuring the TSPs, initialize the ISDN BRI terminals by entering the required parameter values at the terminal key pad or keyboard. The user manual shipped with each terminal provides instructions for initializing the terminal for a specific application.

Information on configuring the M5317TDX and the M5209TDcp ISDN BRI terminals follows. Procedures are also given on how to configure NI-Conference on the M5317TDX and the M5209TDcp.

Set up ISDN BRI terminal parameters

The ISDN BRI terminal requires that Layer 2 and Layer 3 parameters be programmed at the terminal. Refer to your terminal documentation for complete instructions. In general, the following parameters are needed:

- TEI: Voice and circuit switched data calls require dynamic TEI assignment, which automatically assigns a TEI (range 64 - 126) when the terminal is connected to the DSL.

  Packet data requires static TEI assignment, which is performed manually by entering an unassigned TEI number (range 0 - 63) directly on the terminal. This TEI remains assigned to the terminal as long as it remains operational. Procedures for configuring terminals for voice and circuit switched data calls (dynamic TEI assignment), and for packet data (static TEI assignment) soon follow in this section.
- SPID: Enter the voice SPID, data SPID. Each SPID should match the one entered in LD 27 in the TSP which contains this terminal's DN. A different SPID can be configured for the voice and data, allowing two different TSPs to be configured in LD 27.
- DN: Enter voice DN and data DN. These DNs should match those defined in the associated TSPs.
- Some terminals may allow the user to make other selections such as Aµ law or bearer capability. Refer to the set menu for details.

Program the M5317TDX for line application

Program the M5317TDX for a line on page 156 provides the steps to program terminal parameters for the M5317TDX, firmware version 2.3a and later.

Program the M5317TDX for a line

1. Unpack and plug in the terminal. Hold down the **RLS** and **HOLD** keys as it powers up.
2. Press **MAINROM**.
3. Press **INSTALL**.
4. Press **ENGLISH** or **FRANCAIS** for your choice of language.
5. Ensure that the terminal TEI is set to dynamic, by pressing the **terminal softkey** and entering *****.
6. Press **OK** when you have finished setting the TEIs.
7. Enter the terminal (voice) SPID as configured in LD 27 (the default SPID does not apply). Press **OK** when finished.
8. Enter the data SPID as configured in LD 27 (the default SPID does not apply). Press **OK** when finished.
9. Enter the data DN. Press **OK** when finished.
10. Press **HEADSET** until `REAR` is displayed.
11. Press **SIGTYPE** until `MER1` is displayed. If NI-Conference is desired, set SIGTYPE = NI.
12. Press **A/MU** until `µ-Law` is displayed.
13. Press **MORE**.
14. Ensure **DIALPLAN** is configured for **NATIONAL**.
15. Press **EXIT**.
16. Press **NO for execute SPM** (Service Profile Management).
17. Enter **YES** to enter new data fill, and **YES** to delete existing datafill.
18. Press **KEY#** then enter on the keypad the key number you wish to program.
19. Press **EDIT DN** then enter on the keypad the DN digits.
20. After each DN entry, press **OK**, then **SAVE**.
21. Repeat steps 17-19 for each voice DN desired. Press **EXIT** when finished.
22. To install NI-Conference on an M5317TDX terminal, press **KEY#**, then enter a number on the keypad (15 is recommended). Press **TYPE** until `FA` is displayed. Press **FEATURE** until `FCC` is displayed. Press **SAVE**.

    Note:

    The KEY# must correspond to the Feature Activation ID entered for the terminal's TSP in LD 27.
23. If you made a mistake, press **INSTALL** again and begin at step 3; otherwise, press **EXIT**.

24. Wait a few minutes. `Error code 23` displays and clears. `DATE AND TIME` flashes. Push **setup**. Set **clock**. Set **date**.
25. Make a voice call to ensure that the terminal is operational.

The following features are supported on M5317TDX terminals:

- AutoDial Keys (up to 9 keys each with 20 digit numbers)
- Inspect Key
- Data Port Configuration from the menu
- Hands Free
- Conference-Soft Key (Recommended: Key #15)
- DiscData Soft key for Disconnecting Data Calls
- English/French Language Support
- Set based Ringing Patterns
- Set based Clock

Program the M5209TDcp for a voice call

Program the M5209TDcp for voice call on page 157 provides the steps to program terminal parameters for voice calls on the M5209TDcp, firmware version 2.28 and later.

Program the M5209TDcp for voice call

1. Hold the **Release** and **Hold** keys simultaneously and wait for the menu screen. To advance to the next option, press ***** and to select an option, press **#**
2. `MAIN MENU CONFIG`
   - for TEI/SPID/DN/FEAT configuration, press **#**
3. `ENTER PASSWORD`
   - enter a number for `isdn` (using keypad to spell out isdn# = 4736#)
4. `CONFIGURATION MENU TEI`
   - press **#** to select the TEI menu
5. `ENTER TEI VOICE AUTO`
   - press **#** to select AUTO (dynamic TEI assignment) for voice calls
6. `ENTER TEI PSD AUTO`
   - press **#** to select AUTO (dynamic TEI assignment) for packet switch data calls
7. `ENTER TEI CSD AUTO`
   - press **#** to select AUTO (dynamic TEI assignment) for circuit switch data calls

8. `CONFIGURATION MENU SPID`
    - press # to select the SPID menu
9. `ENTER SPID VOICE aaaa`
    - enter the SPID for voice call and press # to accept
10. `ENTER SPID CSD aaaa`
    - enter the SPID for circuit switch data call and press # to accept
11. `CONFIGURATION MENU SWITCH`
    - press # to select the SWITCH menu
12. `SELECT SWITCH TYPE NISDN`
    - press # to select National ISDN-1 type
13. `CONFIGURATION MENU EKTSmode`
    - press ***** to take the default call handling mode (BASIC) and advance to the next option
14. `CONFIGURATION MENU DEF KEYS`
    - press the first key from the bottom to define a primary DN appearance
15. `DEFINE KEY 1 CALL APP nnnn`
    - enter the `primary DN` (key 1 can be a DN only)
    - If you made a mistake and must re-enter the DN, press ***** to clear and enter the correct DN
    - press # to accept

    Note:

    Call appearance can use the same DN on multiple keys. For example, you may have one DN and two keys to invoke Conference.
16. `CONFIGURATION MENU DEF KEYS`
    - press key 2 to 9 to enter another DN appearance or a feature
17. `DEFINE KEY 2 CALL APP nnnn`
    - enter a `secondary DN` appearance, OR
    - press ***** to scroll through the feature options
    - press # to accept
18. `CONFIGURATION MENU DEF KEYS`
    - press ***** to advance to the next option
19. `CONFIGURATION MENU CSD DN`
    - press # to select the circuit switch data DN menu

20. `ENTER DN FOR CSD nnnn`
    - enter the `CSD DN`
    - press ***** to re-enter
    - press **#** to accept
21. `CONFIGURATION MENU EXIT`
    - press **#** to exit
    - press***** to review/revise the above entries
22. `MAIN MENU EXIT`
    - press ***** to scroll through the main menu options until `EXIT`
23. `SAVING DATA`
    - wait for screen to refresh
24. `SELF TEST PASSED`
    - wait for self test to finish

It is recommended that the following keys be configured on the M5209TDcp:

- Two DN call appearances at a minimum
- Speed Call List
- Stored Number Redial
- Disconnect Data Call
- Conference

Program NI-Conference on the M5209TDcp

Configure NI-Conference on the M5209TDcp terminal on page 159 provides the steps to configure the M5209TDcp terminal for the NI-Conference feature.

Configure NI-Conference on the M5209TDcp terminal

1. `MAIN MENU CONFIG`
    - for FEAT configuration, press **#**
2. `ENTER PASSWORD`
    - enter `isdn#` (for example, 4736#)
3. `CONFIGURATION MENU TEI`
    - press ***** to scroll through the main menu until `DEF KEYS`

4. `CONFIGURATION MENU DEF KEYS`

   • press key 2 to 9 to configure a feature

5. `DEFINE KEY 9 CONF`

   • to define Conference, press ***** to scroll through the feature options until `CONF`

   • press **#** to accept

   Note:

   NI-Conference can be defined on keys 2-9. The key number which is selected must also be entered as the Feature Activation ID of A03/A06 in the TSP configuration in LD 27. Also, at least two DN appearances must be configured on the terminal.

Program the M5209TDcp for a data call

To configure a data call for the M5209TDcp terminal, use Configure a data call for the M5209TDcp terminal on page 160 to select circuit switched data (CSD) mode and to program CSD parameters.

Important:

Ensure that the TEI and/or SPID is configured for Circuit Switched Data as described in the voice call configuration.

Configure a data call for the M5209TDcp terminal

Follow steps 1-9 to select the Circuit Switched Data (CSD) mode on the M5209TDcp:

1. Press the **Hold** and **Release** keys simultaneously until the `Main Menu` appears on the set display.

   `MAIN MENU CONFIG`

2. Press ***** until the data option appears on the display.

   `MAIN MENU DATA`

3. Press **#** to display the current data mode.

   `DATA MENU MODE`

4. Press **#** to display the current data mode.

   `SELECT DATA MODE PACKET`

5. If PACKET, VCCI or NONE is displayed, press ***** until `CIRCUIT` is displayed. If CIRCUIT is displayed already, proceed to step 6.

   `SELECT DATA MODE CIRCUIT`

6. Press **#** to select circuit-switched mode.

   `DATA MENU PARAMS`

7. Press ***** to scroll through data options until `EXIT` appears on the display.

   `DATA MENU EXIT`

8. Press **#**

   `MAIN MENU EXIT`

9. Press **#**

   Follow steps 10-20 to configure the CSD parameters on the M5209TDcp terminal:

   Note:

   For information on CSD parameters, refer to the *&varUGM5209TDcpDCG;*.

10. Press the **Hold** and **Release** keys simultaneously until the Main Menu appears on the set display.

    `MAIN MENU CONFIG`

11. Press ***** until the data option appears on the display.

    `MAIN MENU DATA`

12. Press **#**.

    `DATA MENU MODE`

13. Press ***** until the `PARAMS` option is displayed.

    `DATA MENU PARAM`

14. Press **#** to select a parameter to change.

    `SELECT PARAMETER`

15. Type the parameter number you wish to change and press **#**.

    `ENTER PARAMETER VALUE 202:3`

    This example, we typed 202. The display shows the current value for parameter 202 as 3(1200bps).

    If you typed an invalid parameter, the display shows the `INV PARM` message. Retype a valid parameter number and press #.

16. Type the new value you wish to assign the parameter and press **#**. If you want to leave the parameter value unchanged, do not enter a value, just press **#**.

    `SELECT PARAMETER ACCEPTED`

    The display shows the `ACCEPTED` message when you enter a valid value. If it displays `INV VALUE`, retype the value and press **#**.

17. Repeat steps 15 and 16 for other parameters. After the M5209TDcp has accepted the last parameter that you wish to change, press **#**.

    `DATA MENU SAVE`

18. To save the changed parameters in the M5209TDcp nonvolatile memory, press **#**. Otherwise, press ***** until the `EXIT` option is displayed.

    `DATA MENU EXIT`

19. Press **#**.

    `MAIN MENU EXIT`

20. Press **#** to finish programming CSD parameter.

The Channel Control registers are used to select the CSD Data Bearer Capability. Hayes AT commands must be used to change the selection:

- AT%A4 = 0 sets it to unrestricted 64kbps
- AT%A4 = 1 sets it to 56kbps (adapted from 64kbps)

The following features are supported on M5209TDcp terminals:

- Last Number Redial (invoked by ##)
- Set based Speed Call (Can store up to five 25 digit numbers)
- Store Number Redial (multiple keys)
- Conference (Key #9)
- DiscData Soft key for Disconnecting Data calls
- English/French Language Support
- Set based Ringing Patterns

Initialize an M5000TD-1 terminal adapter

The M5000TD-1 is a Universal Terminal Adapter (UTA). It adapts a non ISDN BRI data terminal or an analog (500/2500-type) telephone to the ISDN BRI protocol. A terminal must be attached to the M5000 terminal adapter to initialize it.

Use the following procedures to initialize the M5000TD. For additional information, refer to the *&varUGM5000TD1UG;*.

1 Circuit Switched Voice Calls

Service Profile ID (SPID) !C2 = "nnn" Primary Directory Number (DN)!N3 = "nnn" Second DN (optional) !N4 = "nnn" B1 (B2) Channel!C4 (5) = 4 Configure DTE Channel as Voice %A0 = 3

2 Circuit Switched Data Calls

Service Profile ID (SPID) !C6 = "nnn" Directory Number (DN)!N1 = "nnn" B1 (B2) Channel! C4 (5) = 4 Configure DTE Channel as Data%A0 = 2 Configure Data Call as CSD%A1 = 0 Synchronous (Asynchronous)&Q1, 2 (&Q0) Protocol Used% A2 = 0 (No protocol) synch %A2 = 1 (T-Link - Reg S37) synch or asynch %A2 = 2 (V.120) asynch %A2 = 8 (I.515) asynch Speed of channel%A4 = 0 (64kbps) %A4 = 1 (56kbps) Read-Only Error Register %A6

3 Packet Switched Data B-channel

Which B-channel 1 (2)!C4(5) = 1
!Y Registers specify B-channel packet layer parameters
default Packet Size - Receive !Y2 = 7 (Packet Size = 128 bytes) !Y2 = 8 (Packet Size = 256 bytes) Default Packet - Transmit!Y3 = 7 (Packet Size = 128 bytes) !Y3 = 8 (Packet Size = 256 bytes)
X.25 Logical Channel Provisioning (M5000TD-1 supports a maximum of 8 LCNs)
Number of PVCs on b-channel !Y4=n (default is 0) Number of ILCs on B-channel!Y5=n (default is 8) Number of OLCs on B-channel!Y7=n (default is 0)
Transmission Retry Count!B3=n (default is 4, range is 1-255)
Configure DTE Channel as data%A0 = 2 Configure Data Call as PSD%A1 = 1 Specify B1 or B2 Channel%A3 = 3 (B1) %A3 = 5 (B2)

4 Packet Switched Data D-channel

X.25 packet service - D-channel !C3 = 1 (default) Terminal Endpoint Identifier !D0 = n (n is TEI)
!X Registers specify D-channel packet layer parameters
Default Packet Size - Receive!X2 = 7 (Packet Size = 128 bytes) !X2 = 8 (Packet size = 256 bytes) Default Packet Size - Transmit!X3 = 7 (Packet Size = 128 bytes) !X3 = 8 (Packet Size = 256 bytes)
X.25 Logical Channel Provisions (M5000TD-1 supports a maximum of 8 LCNs)
Number of PVCs on D-channel!X4 = n (default is 0) Number of ILCs on D-channel!X5 = n (default is 0) Number of TLCs on D-channel!X6 = n (default is 8) Number of OLCs on D-channel!X7 = n (default is 0)
Configure DTE Channel as Data%A0 = 2 Configure Data Call as PSD%A1 = 1 Specify D-channel%A3 = 9

Note:

The operation of the Packet Assemble/Disassembler (PAD) of the M5000TD-1 is controlled by the %L registers. See the *&varUGM5000TD1UG;* for details.

Note:

After setting/changing the ! registers of the M5000TD-1, type AT%Z1 for the new settings to take effect.

Note:

Type ATD<DN> to dial up another terminal. Type ATA to answer. Type +++ to get into command mode. Type ATH to hang up (from command mode). Type ATO to get back into online mode.

Note:

The &K setting affects flow control on the M5000TD-1:
Value of 0 turns off flow control. Value of 3 turns on XON/XOFF. Value of 4 turns on TTS/CTS hardware flow control.

Note:

The &Q setting affects asynch/synch mode of the M5000TD-1:
Value of 0 -> asynchronous. Value of 1 -> asynch during call setup, synch thereafter.

Note:

Typing at&v displays these settings:
Typing at&4 displays %L settings. Typing &w0 saves active profile as profile 0.
Typing &y0 loads profile 0 on boot up.

Configure packet data

This section contains procedures for adding an external packet handler or the integrated internal packet handler (MPH).

Add an external packet handler

The following procedures, in the presented order, should be followed when configuring an external packet handler. Please note, however, that when changing an existing ISDN BRI packet data configuration, following this order is unnecessary. Be aware, though, of the relationship of one component to another and whether changing one component necessitates changing other components.

1. Configure a Link Access Procedure on the D-channel (LAPD) Protocol Group (LD 27).
2. Configure packet data implementation:
    a. ISDN PRI loop (LD17)
    b. ISDN customer (LD15)
    c. TIE trunk route for packet data (LD 16)
    d. TIE trunk for packet data (LD14)
3. Configure the MISP (LD 27).
4. Configure the BRSC (LD 27) (optional).
5. Configure the SILC or UILC (LD 27).

    Note:

    This step is optional. The SILC or UILC can also be configured when configuring the DSL (see next step).
6. Configure the DSL (LD 27).
7. Configure the TSP on the DSL (LD 27).
8. Initialize ISDN BRI terminals (M5317TDX, M5209TDcp).

Configure an LAPD protocol group for an external packet handler

Add an LAPD protocol group by using LD 27 and specifying its protocol group number. You may also change its LAPD parameters as needed or accept the default values. LAPD is a transmission protocol that specifies the transmission timers, the maximum number of retransmissions, the size of the data frame, and the number of negative acknowledgments allowed before the system issues an alarm.

Table 39: LD 27 - Add or change an LAPD protocol group.

Prompt	Response	Description
REQ	NEW CHG	Add or change an ISDN protocol group.
TYPE	LAPD	LAPD Protocol group
PGPN	0-15 <cr>	Protocol group number The values for this prompt are: 0-15=Adds a specified protocol group <cr>=Stops this prompt from being displayed again
LAPD	YES (NO)	LAPD parameters. The values for this prompt are: YES = Define or modify the LAPD parameters (NO) = Does not prompt the LAPD parameters and assigns the default values shown in ( ) to these parameters.
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.
N2X4	0-(10)-20	For 1TR6 connectivity - number of status inquiries when the remote station is in peer busy state.
PGPN	<cr>	Press <cr> to prevent repetition of all the parameters starting with LAPD.

Remove an LAPD protocol group configured for an external packet handler

You can remove an LAPD protocol group as long as it is not assigned to a DSL. If a protocol group is assigned to a DSL, delete the DSL before removing the protocol group.

Table 40: LD 27 - Remove an LAPD protocol group.

Prompt	Response	Description
REQ	OUT	Remove an ISDN BRI component.
TYPE	LAPD	Protocol group
PGPN		Protocol group number
	0-15 ALL <cr>	0-15 = Removes a specified protocol group from 0-15 ALL = Removes all protocol groups <cr> = No change the protocol group is not removed. A protocol group cannot be removed if it is assigned to a DSL.
...	...	

Print an LAPD protocol group configured for an external packet handler

Configuration information for a specific LAPD protocol group or for all protocol groups can be printed.

Table 41: LD 27 - Print an LAPD protocol group.

Prompt	Response	Description
REQ	PRT	Prints an ISDN BRI component.
TYPE	LAPD	Protocol group
PGPN		Protocol group number
	0-15 <cr>	0-15 = Prints a specified protocol group from 0-15 <cr> = Prints all protocol groups and the number of DSLs in each group
USER	YES (NO)	YES = Print the LAPD group(s) selected in the PGPN prompt and the DSLs that are using it (them) NO = Do not print the LAPD user information.
REQ	...	

Configure ISDN PRI trunk assignments for an external packet handler

Before configuring the MISP, DSL, and TSP with LD 27, the following must be specified to provide a communication link with the packet handler.

- ISDN PRI loop (LD17)
- ISDN customer (LD15)
- TIE trunk route for packet data (LD 16)
- TIE trunk for packet data (LD 14)

Table 42: LD 17 - Add an ISDN PRI loop for an external packet handler.

Prompt	Response	Description
REQ	NEW CHG	Add new data, or change existing data.
TYPE	CFN	Configuration data block
...	...	
CEQU	YES	Change common equipment options
...	...	
DLOP	lll dd ff	PRI loop parameters for Large Systems lll (0-159) = Network loop number dd (0-24) = number of voice or data calls ff (ESF) = frame format: D2, D3, D4, Frame format must match the far end.
DLOP	1-9	1.5 Mbit PRI, for Small Systems.
		2.0 Mbit PRI, for Small Systems
PR12	1-9	
MODE	PRI	Primary Rate Interface mode
PRI	0-159	PRI loop number

Table 43: LD 15 - Define an ISDN customer for an external packet handler.

Prompt	Response	Description
REQ:	CHG	Change existing data.
TYPE:	NET	Networking Data.
CUST		Customer number
	0-99	Range for Large System and CS 1000E system.
	0-31	Range for Small System, Media Gateway 1000B, and Media Gateway 1000T.
ISDN	YES	YES = customer is equipped with ISDN (prompted only with D-channel defined in LD17)
...	...	

Table 44: LD 16 - Configure a TIE trunk route for packet data for an external packet handler.

Prompt	Response	Description
REQ	NEW	Add ISDN BRI protocol group settings
TYPE	RDB	Route data block
CUST	xx	Customer number, as defined in LD 15

Prompt	Response	Description
ROUT		Route number
	0-511	Range for Large System and CS 1000E system.
	0-127	Range for Small System, Media Gateway 1000B, and Media Gateway 1000T.
TKTP	TIE	Trunk route type
DTRK	YES	Digital trunk route
BRIP	YES	Packet handler route Prompted only if DTRK = YES.
ACOD	xxxxxx	Trunk route access code
TARG	<cr>	Access restriction group number
CNTL	<cr>	Changes to control timers
...	...	

Table 45: LD 14 - Configure the TIE trunk for packet data for an external packet handler.

Prompt	Response	Description
REQ	NEW	Enter new trunk data
TYPE	TIE	Trunk type
TN		Terminal Number
	l ch	Format for Large System and CS 1000E system, where l = loop, ch = channel number.
	c u	Format for Small System, Media Gateway 1000B, and Media Gateway 1000T, where c = card, u = unit.
CUST	xx	Customer number, as defined in LD 15
NCOS	<cr>	Network Class Of Service group
RTMB		Route number and Member Number
	0-511 1-4000	Range for Large System and CS 1000E system.
	0-127 1-4000	Range for Small System, Media Gateway 1000B, and Media Gateway 1000T.
MNDN	<cr>	Manual directory number
TGAR	<cr>	Trunk group access restriction
CLS	<cr>	Class of Service
...	...	

Configure a MISP for a external packet handler

To add or change a MISP configured for an external packet handler, specify its even loop number. Also, specify the ISDN PRI loop and channel numbers used to transmit packet data to and from the packet handler.

The MISP must be enabled by using the **`ENLL l`** command in LD 32.

Table 46: LD 27 - Add or change a MISP configured for an external packet handler.

Prompt	Response	Description
REQ	NEW CHG	Add or change an ISDN BRI MISP **Note:** The defaults apply to adding, not changing, a MISP.
TYPE	MISP	MISP.
LOOP	0-158	MISP loop number; must be an even number, with the next odd loop number unequipped.
	1-9	MISP card slot number has to be 1-4, 7-10 for Media Gateway 1000 1-10 for Media Gateway 1010
APPL	MPH	Application type for the MISP (MPH = packet handler).
DPSD	YES (NO)	YES = D-channel Packet Switched Data NO = No D-channel Packet Switched Data. Enter YES for packet switched data.
MPHC	(YES) NO	YES = DPSD are routed to a MPH NO = DPSD are routed to an external packet handler or PSDN. Enter NO to choose an external packet handler.
TN		Terminal Number
	l ch	Format for Large System and CS 1000E system, where l = loop, ch = channel. The values for this prompt are: l = 0-159 ch = 1-23
	c	Format for Small System, Media Gateway 1000B, and Media Gateway 1000T, where c = card.
REQ	...	

Remove a MISP configured for an external packet handler

Before removing a MISP:

- Remove all BRSCs associated with it.
- Remove all DSLs connected to SILCs and UILCs associated with it.
- Disable the MISP loop with the DISL l command in LD 32.
- Remove the PVC or network interface connections.

Remove a MISP by specifying its loop number.

Table 47: LD 27 - Remove a MISP configured for an external packet handler.

Prompt	Response	Description
REQ	OUT	Remove an ISDN BRI component.
TYPE	MISP	MISP
LOOP	0-158	Loop number for Large Systems. Must be an even number.
	1-9	MISP card slot number has to be 1-4, 7-10 for Media Gateway 1000 1-10 for Media Gateway 1010 The MISP must be disabled before removing it. All SILC and/or UILC DSLs associated with the MISP must be removed before removing the MISP.
REQ	...	

Print a MISP configured for an external packet handler

Print the configuration information for a MISP configured for an external packet handler by specifying its network loop number. If the MISP network loop number is not known, use LD 22 to print the system configuration. BRSC TNs which are associated with a MISP are also listed when data of this MISP are printed.

Table 48: LD 22 - Print a MISP configured for an external packet handler.

Prompt	Response	Description
REQ.	PRT	Print an ISDN BRI component.
TYPE	MISP	MISP BRSC TNs associated with a MISP are also listed when data of the MISP are printed.
LOOP	0-158	MISP loop number for Options Large Systems. Must be an even number.

Prompt	Response	Description
	1-9	MISP card slot number has to be 1-4, 7-10 for Media Gateway 1000 1-10 for Media Gateway 1010
REQ	...	

Add or change a BRSC for an external packet handler

To add or change a BRSC for an external packet handler, specify its superloop number, shelf number, and card number. Specify the PRI loop and channel numbers for routing of the DPSD to an external packet handler.

When the BRSC has been added or changed, enable the MISP, enable the BRSC with the **`ENLClll s cc`** command in LD 32, and enable all ISDN BRI line cards in the IPE module.

Note:

Refer to the section "Add or remove BRSCs with a configured ISDN BRI system", which follows, for possible scenarios for adding or removing BRSCs with a system configured for ISDN BRI.

Table 49: LD 27 - Add or change a BRSC for an external packet handler.

Prompt	Response	Description
REQ	NEW CHG	Add or change a SILC or UILC line card.
TYPE	BRSC	SILC or UILC line card
BRSC	lll s cc	Card location The values for this prompt are: lll (loop)=0-156 (must be an even number divisible by 4) s (shelf)=0-1 cc (card)=0-15
MISP	0-158	MISP loop number (must be an even number that has already been configured)
DPSD	YES (NO)	Enter YES for D-channel Packet Switched Data.
- MPHC	(YES) NO	Route D-channel packet switched data to the Meridian Packet Handler. Enter NO.
- PRI	lll nn	lll (1- 159) = PRI loop number that is connected to the external packet handler nn (1-23) = PRI channel number on which the dedicated connection from the BRSC is terminated Enter lll <space> nn on same line PRI CH appears only if DPSD = YES.

Remove a BRSC configured for an external packet handler

Remove a BRSC which has been configured for an external packet handler by specifying its loop number. Before removing the BRSC, disable all line cards associated with it, and disable the BRSC loop with the **`DISC lll s cc`** command in LD 32 before removing the BRSC.

Note:

Refer to the section "Add or remove BRSCs with a configured ISDN BRI system", which follows, for possible scenarios for adding or removing BRSCs with a system configured for ISDN BRI.

Table 50: LD 27 - Remove a BRSC configured for an external packet handler.

Prompt	Response	Description
REQ	OUT	Remove an ISDN BRI component.
TYPE	BRSC	Remove the BRSC data
BRSC	lll s cc	Card location The values for this prompt are: lll (loop)=0-156 (must be an even number divisible by 4) s (shelf)=0-1 cc (card)=0-15
REQ	...	

Add or remove BRSCs with an ISDN BRI system

To add or remove BRSCs with a system configured for ISDN BRI, follow these steps:

- Disable the MISP when the first BRSC is configured.
- If two or more line cards which are served by the MISP are in the IPE Module where the first BRSC is added, disable the MISP once only.
- Disable the MISP when changing from a configuration with three line cards and two BRSC to a configuration with two line cards and up to eight BRSCs.
- Conversely, disable the MISP when changing from a configuration with two line cards and up to eight BRSCs to a configuration with three line cards and one BRSC.

Table 51: Add or remove BRSCs with a configured ISDN BRI system on page 173 lists the possible scenarios for adding or removing BRSCs with a system configured for ISDN BRI.

Table 51: Add or remove BRSCs with a configured ISDN BRI system

Initial configuration			
Configured line cards installed	**BRSCs**	**Action**	**Same IPE Module?**
0 0	0 1	Add BRSC 1 Add BRSC 2-8	NA NA
1 1	0 1	Add BRSC 1 Add BRSC 2-8	Yes/No NA
2 2	0 1	Add BRSC 1 Add BRSC 2-8	Yes/No Yes/No
3	0	Add BRSC 1	Yes/No
3 2 0-2	0 1 2	Add BRSC 1 Add BRSC 2 Add BRSC 3-8	No Yes Yes/No
4 3 0-2	0 1 3-8	Add BRSC 1 Add BRSC 2 Add BRSC 3-8	Yes Yes Yes/No
2 2 3 3	2-8 1 1 0	Disable BRSC 1 Add line card 3 Delete BRSC Add line card 4	N/A N/A N/A N/A
2 2 3	2-8 0 0	Delete BRSCs Add line card 3 Add line card 4	N/A N/A N/A

Print a BRSC configured for an external packet handler

Print the configuration information for a BRSC which has been configured for an external packet handler by specifying its network loop number. If the BRSC network loop number is not known, use LD 22 to print the system configuration.

To print all BRSCs associated with a MISP, enter <cr> at the BRSC and MISP prompts.

Table 52: LD 22 - Print a BRSC configured for an external packet handler.

Prompt	Response	Description
REQ	PRT	Prints an ISDN BRI component.
TYPE	BRSC	Print BRSC data. **Note:** BRSC TNs associated with the MISP are also printed.
BRSC	lll s cc	Card location The values for this prompt are:

Prompt	Response	Description
		lll (loop)=0-156 (must be an even number divisible by 4) s (shelf)=0-1 cc (card)=0-15
MISP	0-158 <cr>	MISP loop number (must be an even number that has already been configured). If <cr> is entered, all BRSCs configured in the system are printed; otherwise, all BRSCs associated with the MISP loop are printed.
REQ	...	

Configure a SILC or UILC for an external packet handler

Add or change a new SILC or UILC to the system by specifying its location, card type, and the MISP network loop that this card uses to transmit and receive signaling and D-channel packet data.

Note:

This step can be skipped and the card type specified when configuring the DSL in the procedure "Add a DSL for an external packet handler."

The following procedure is used when configuring the SILC or UILC cards without configuring their DSLs.

Table 53: LD 27 - Add or change a SILC or UILC for an external packet handler.

Prompt	Response	Description
REQ	NEW CHG	Add or change a SILC or UILC line card.
TYPE	CARD	SILC or UILC line card
TN		Terminal Number
	l s c	Format, where: l = • 0-156 (must be an even number divisible by 4) s = • 0–1 c = • 0-15 for Large System, 1-4, 7-10 for chassis, 1-10 for cabinet and MG 1010
MISP	0-158	Loop number (must be an even number that has already been configured) for Options 51C- 81C. If there is a BRSC configured in the IPE module, the MISP prompt is skipped and the MISP lll and the BRSC lll s cc is displayed.

Prompt	Response	Description
	1-9	MISP card slot number has to be 1-4, 7-10 for Media Gateway 1000 1-10 for Media Gateway 1010
CTYP	SILC UILC	Card type to be added or changed. Remove any DSLs configured for this line card before changing the card type.
REQ	...	

Remove a SILC or UILC configured for an external packet handler

Remove a SILC or UILC by specifying its card location. Before removing the SILC or UILC, all configured DSLs must first be removed from the card by using the procedure "Remove a DSL configured for an external packet handler". When the last DSL is removed, the card is automatically deleted.

When removing the card, the database information is also deleted from the data block. Use LD 20 to list cards that have been removed.

Table 54: LD 27 - Remove a SILC or UILC configured for an external packet handler.

Prompt	Response	Description
REQ	OUT	Remove an ISDN BRI component.
TYPE	CARD	ISDN BRI line card
TN		Terminal Number
	l s c	Format, where: l = • 0-156 (must be an even number divisible by 4) s = • 0–1 c = • 0-15 for Large System, 1-4, 7-10 for chassis, 1-10 for cabinet and MG 1010

Print a SILC or UILC configured for an external packet handler

To print the configuration information for a SILC or UILC, specify its card location.

Table 55: LD 27 - Print a SILC or UILC configured for an external packet handler.

Prompt	Response	Description
REQ	PRT	Print an ISDN BRI component.

Prompt	Response	Description
TYPE	CARD	ISDN BRI line card
TN		Terminal Number
	l s c	Format, where: l (Loop)= • 0-156 (must be an even number divisible by 4) s (Shelf)= • 0–1 c (Card)= • 0-15 for Large System, 1-4, 7-10 for chassis, 1-10 for cabinet and MG 1010

Configure a DSL for an external packet handler

To add or change a DSL, specify its port location and its DSL characteristics. DSL location specifies a SILC/UILC port connected to a DSL.

Table 56: LD 22 - Add or change a DSL for an external packet handler.

Prompt	Response	Description
REQ	NEW CHG	Add or change a DSL **Note:** The defaults apply to adding, not changing, a DSL.
TYPE	DSL	DSL
DSL	lll s cc dsl#	DSL location for Options 51C - 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
	c dsl#	DSL location for Option 11C c (card) = 1-20 dsl# (DSL number) = 0-7 Assign 31 DSLs for each MISP if DCH or BDCH were specified at the PH prompt in "MISP configuration procedures."
DES	x...x	Designator to assign to a DSL (ex. BUILD2) x...x = 1 to 6 alphanumeric DSL designator
CUST	xx	Customer number, as defined in LD 15
CTYP	SILC UILC	Card type. This prompt is displayed only if the SILC or UILC has not been previously configured.
OPT	(BRIL) <cr>	Defaults to ISDN BRI line application (BRIL). Enter <cr>

Prompt	Response	Description
MISP	0-158	Loop number (must be an even number of a MISP that has already been configured). This prompt is displayed only if the MISP has not been assigned to the specified SILC or UILC.
MISP	0-158	Loop number (must be an even number of a MISP that has already been configured) for Large Systems.
	1-9	MISP card slot number has to be 1-4, 7-10 for Media Gateway 1000 1-10 for Media Gateway 1010 This prompt is displayed only if the MISP has not been assigned to the specified SILC or UILC.
MODE	NTAS NTFS	Network terminal line sampling mode (this prompt is displayed only if you specified the card type as SILC). The values for this prompt are: NTAS=Adaptive sampling Extended passive bus, Branched passive bus, Point-to-point bus, U interface DSL. NTFS=Fixed sampling Short passive bus.
B1CT	(VCE) (DTA) PMD XPMD	B-channel 1 call type Enter PMD. PMD = B-channel packet data with dedicated connection from DSL to a PRI channel using external packet handler. For B1CT = PMD, B-channel packet data must have been specified at the PH prompt in LD 27. PMD cannot be combined with any other options. Do not select <cr> which defaults to VCE (circuit switched voice) and DTA (circuit switched data) which may not run concurrently with packet data. XPMD = Delete PMD call types.
TN		Terminal Number TN prompt is given only if call type = PMD. The PRI channel must be configured in LD17 and dedicated only to the connection of an external packet handler.
	l ch	Format for Large System and CS 1000E system, where: l (0-159) = PRI2 loop number which is connected to the external packet handler or the Packet Switched Data Network. ch (1-30) = the PRI2 channel on which the BD-channel dedicated connection from the DSL B-channel is terminated.
	c u	Format for Media Gateway 1000B , where c = card, u = unit.
B2CT	(VCE) (DTA) PMD XPMD	B-channel 2 call type Enter PMD. PMD = B-channel packet data with dedicated connection from DSL to a PRI channel. (B-channel packet data must have been specified at the PH prompt in LD 27). PMD cannot be combined with any other options. Do not select <cr> which defaults to VCE (circuit switched voice) and DTA (circuit switched data) which may not run concurrently with packet data.

Prompt	Response	Description
		For B2CT = PMD, B-channel packet data must have been specified at the PH prompt in LD 27. XPMD = Delete PMD call types
TN		Terminal Number TN prompt is given only if call type = PMD. The PRI channel must be configured in LD 17 and dedicated only to the connection of an external packet handler.
	l ch	Format for Large System and CS 1000E system, where: l (0-159) = PRI2 loop number which is connected to the external packet handler or the Packet Switched Data Network. ch (1-30) = the PRI2 channel on which the BD-channel dedicated connection from the DSL B-channel is terminated.
	c u	Format for Media Gateway 1000B, where c = card, u = unit.
LDN	0-3 (NO)	Departmental listed directory number 0-3 = Departmental listed DN specified in LD15 NO = No departmental listed DN associated with the DSL
XLST	(0)-254	Pretranslation group (if configured in customer data block)
MTEI	1-(8)-20	Maximum number of Terminal Endpoint Identifiers, both static and dynamic combined assigned to the logical terminals on this DSL
LTEI	n1 n2 mm <cr> Xmm	n1 = Logical Terminal Identifier (LTID) n2 = static Terminal Endpoint Identifier (TEI) mm (0-63) = MTEI, the maximum number of LTID and TEI n1 = 0 -15 and n2 = 0-1023 0 0 is invalid. 15 1023 is invalid. Logical Terminal Identifier (LTID) and static Terminal Endpoint Identifier (TEI) pair for D-channel packet data transmission. MTEI = the maximum number of these pairs. LTID = Logical Terminal Group (LTG) and Logical Terminal Number (LTN). LTG and LTN are entered as part of the DPN configuration. The TE I must match the one in the terminal. By entering all three here, the MISP is able to route data from the terminal to the DPN packet switch. Xmm = Deletes the LTID and TEI for the specified TEI. mm = 0-63 (must be unused static TEI). <cr> = Stops this prompt from being displayed again and skips to the next prompt LTEI is displayed only if: an MPH is not used for PSDN and D-channel packet data was specified for an associated MISP or DPSD is configured for the associated BRSC or the PH option is either DCH or BDCH for line cards associated with a MISP directly.
MCAL	2-(16)-32	Maximum number of calls on the DSL at one time. This includes calls waiting and on hold. Warning is received if less than 8 is specified
MTSP	1-(8)-16	Maximum number of TSPs allowed for a DSL

Prompt	Response	Description
PGPN	0-15	Protocol group number The protocol group should be previously added as described in Configure an LAPD protocol group for an external packet handler on page 165
PRID	1- 6	Defines the protocol to be used on the DSL Selection of the protocol ID is terminal dependent. The values for this prompt are: 1 = ANSI 2 = ETSI 3 = DMS 4 = NET64 5 = NUMERIS 6 = NI-1 **Note:** A response of 6 allows the ISDN BRI Conference feature to be configured in the TSP of the DSL.
FDN	n...n	Flexible CFNA directory number. Enter a 1-13 digit DN
EFD	n...n	Flexible external call CFNA DN. Enter a 1-13 digit DN.
HUNT	n...n	Hunt directory number. Enter a 1-13 digit DN.
EHT	n...n	Hunt external call directory number. Enter a 1-13 digit DN.
TGAR	(0)-31	Trunk group access restriction
NCOS	(0)-99	Network Class Of Service
CLS		Class of Service access restrictions. More than one Class of Service can be entered by separating each entry with a space. Default features shown in parenthesis are selected by pressing <cr>.
	(ICDD) ICDA	Internal Call Detail Recording (Denied) Allowed
	(MRD) MRA	Message Restriction (Denied) Allowed
	(UDI) RDI	(Unrestricted) Restricted DID
	(UNR) CTD CUN FR1 FR2 FRE SRE TLD ICDA (ICDD) <cr>	(Unrestricted) Conditionally Toll Denied Conditionally Unrestricted Fully Restricted class 1 Fully Restricted class 2 Fully Restricted Semi-Restricted Toll Denied Internal Call Detail Recording allowed (Internal Call Detail Recording denied) Enter <cr> to select the defaults. More than one Class of Service can be selected by separating each entry with a space.
REQ	...	

Remove a DSL configured for an external packet handler

Remove a DSL by specifying its location. To remove a DSL, first remove all the TSPs assigned to this DSL. When the last configured DSL on a card is removed, the card is removed automatically.

Table 57: LD 27 - Remove a DSL configured for an external packet handler.

Prompt	Response	Description
REQ	OUT	Remove an ISDN BRI component
TYPE	DSL	Digital Subscriber Loop
DSL	lll s cc dsl#	DSL location, where: • lll (superloop) = 0-156 (must be zero or a number divisible by 4) • s (shelf) = 0-1 • cc (card) = 0-15 and for CS 1000E 1–20 • dsl# (DSL location) = 0-7 Assign 31 DSLs for each MISP if DCH or BDCH were specified at the PH prompt in "MISP configuration procedures".

Print a DSL configured for an external packet handler

Print the configuration information for a single DSL by specifying its location.

Table 58: LD 27 - Print a DSL configured for an external packet handler.

Prompt	Response	Description
REQ	PRT	Print an ISDN BRI component.
TYPE	DSL	Digital Subscriber Loop
DSL	lll s cc dsl# lll s cc lll s lll	DSL information for system. lll s cc dsl# = Prints information for the specified dsl# lll s cc = Prints information for DSLs on the specified card lll s = Prints information for DSLs in the specified shelf lll = Prints information for DSLs on the specified loop
DATE	(<cr>) x y z	Print data and display the last active date, where x = day (1-31), y = month (Jan-Dec), and z = year (1979-9999) specifies the starting date of the data to be displayed or printed.
PAGE	YES (NO)	YES = prints one DSL per page NO = prints without paging
DES	x...x <cr>	1-digit to 6-digit alphanumeric DSL designator No designator for DSLs
NACT	YES (NO)	Activity date is updated to current date.

Configure a TSP for an external packet handler

To add or change a TSP to a DSL, specify the DSL location, its transmission characteristics, and the Class of Service for terminals connected to the DSL. If the default value is desired, press the ENTER key.

Table 59: LD 27 - Add or change a TSP for an external packet handler.

Prompt	Response	Description
REQ	NEW CHG	Add or change a TSP.
TYPE	TSP	Assign TSP to a DSL
DSL	lll s cc dsl#	DSL location, where: • lll (superloop) = 0-156 (must be zero or a number divisible by 4) • s (shelf) = 0-1 • cc (card) = 0-15 and for CS 1000E 1–20 • dsl# (DSL location) = 0-7 Assign 31 DSLs for each MISP if DCH or BDCH were specified at the PH prompt in "MISP configuration procedures".
USID	0-15	User service identifier Set USID = 0 to configure a default TSP for non-initializing terminals. Set USID = 1-15 for initializing terminals, for example, the M5317TDX. The total number of TSPs defined for a DSL cannot exceed the maximum number of TSPs allowed for a DSL as specified by the MTSP.
MPHC	(YES) NO	Route D-channel packet switched data to the Meridian Packet Handler. Enter NO.
SPID	aaa...a <cr> Xaaa...a	Service profile ID aaa...a = any combination of 1-20 alphanumeric characters. <cr> = Stops this prompt from being displayed again. A maximum of 8 valid SPIDs per TSP are allowed. Xaaa...a removes the specified SPID. This prompt appears only if USID = 1-15. It repeats until <cr> is entered, but only up to 8 SPIDs may be entered. This SPID must be entered in the initializing terminal to associate the terminal with a USID.
FEATID	aaa mmm nnn <cr> Xaaa	ID associated with feature aaa, as follows: A03 = 3-party Conference A06 = 6-party conference mmm = Feature Activation ID(1-127) nnn = Feature Indication ID (1-127) (optional; if not entered, the value entered for mmm is assumed) <cr> = Skip the FEATID entry Xaaa = Delete the feature. Feature Activation ID and Feature Indication ID are feature key number assignments configured at the terminal level. Recommended terminal assignments are: - for the M5317TDX: A06 15 - for the M5209TDcp: A06 9
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). The DN cannot be shared by a non ISDN BRI terminal. This prompt is repeated until <cr> is pressed. At least one DN and a maximum of 8 DNs can be assigned to a DSL.

Prompt	Response	Description
		The directory number can be associated with multiple TSP
CT	VCE DTA	Directory number call type VCE=Circuit switched voice DTA=Circuit switched data One or more call types can be entered by separating each entry with a space. The call types entered must have been specified for the B1CT and B2CT prompts in "DSL configuration procedures."
MCAL	1-(4)-8	Maximum number of calls per DN at one time Defines the maximum number of calls allowed for a directory number, which includes the total number of active calls, calls waiting, and calls on hold.
CLIP	(YES) NO	Calling line identification presentation service YES = displays of calling party DN on incoming calls NO = does not display of calling party DN on incoming calls
PRES	(YES) NO	Allows display of calling line identification to far end on outgoing calls. YES = present this DN to the called party on outgoing calls NO = do not present this DN to the called party on outgoing calls
COLP	(NO) YES	Connected Number Information Elements (IEs) is (not) passed from the system to the Terminal Adapter (S_o).
TRANS	(NO) YES	CLID and Connected Number Information Element (IE) are (not) passed from the system to the Terminal Adapter (S_o), if presentation is restricted.
FEAT		Class of Service features
	HTA	HTA = Hunt allowed (always assign if terminal has CWT capability)
	(HTD) FNA (FND) SFA (SFD) CFTA (CFTD) MWA (MWD) FBA (FBD) HBTA (HBTD)	HTD = Hunt denied FNA = Call forward no answer allowed FND = Call forward no answer denied SFA = Second level call forward no answer allowed SFD = Second level call forward no answer denied CFTA = Call forward by call type allowed CFTD = Call forward by call type denied MWA = Message waiting allowed MWD = Message waiting denied FBA = Call forward busy allowed FBD = Call forward busy denied HBTA = Hunting by call type allowed HBTD = Hunting by call type denied
	DNO1 DNO2 (DNO3)	DNO1/DNO2/(DNO3) = QSIG Call Diversion Notification for calling party where: 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 (default).
	DNDN (DNDY)	DNDN/(DNDY) = QSIG Call Diversion Notification for forwarded-to (diverted) party where: DNDN = no notification of called party's number and name notification (DNDY) = notification with called party's number and name when available (default).

Prompt	Response	Description
		More than one Class of Service can be entered by separating each entry with a space. Press <cr> to select multiple default features shown in parenthesis.
DFDN	n...n	Default directory number Enter a 1-digit to 7-digit DN. This DN must be defined at the preceding DN prompt A DN can be associated with multiple TSPs. Only one default DN can be defined for a TSP. This DN is sent in the outgoing setup if the terminal does not send a calling line identification number with the outgoing call.
REQ	...	

Remove a TSP configured for an external packet handler

Before removing a TSP configured for an external packet handler, disable the B-channel.

To remove a single TSP from a DSL, specify the DSL location and the user service identifier. Remove all TSPs from a DSL by entering ALL at the USID prompt.

Removal of the TSP disconnects all calls associated with the TSP's D-channel packet switched data and circuit switched voice data.

Table 60: LD 27 - Remove a TSP configured for an external packet handler.

Prompt	Response	Description
REQ	OUT	Remove an ISDN BRI component
TYPE	TSP	TSP **Note:** Removal of the TSP disconnects all calls associated with the TSP's D-channel packet switched data and circuit switched voice data.
DSL	lll s cc dsl#	DSL location, where: • lll (superloop) = 0-156 (must be zero or a number divisible by 4) • s (shelf) = 0-1 • cc (card) = 0-15 and for CS 1000E 1–20 • dsl# (DSL location) = 0-7 Assign 31 DSLs for each MISP if DCH or BDCH were specified at the PH prompt in "MISP configuration procedures".
USID	0-15 ALL	User service identifier 0-15 = Removes a specified TSP from 0 to 15 ALL = Removes all TSPs for the specified DSL

Prompt	Response	Description
REQ	...	

Print a TSP configured for an external packet handler

Configuration information can be printed for a TSP based on characteristics such as user service identifier, service profile ID, and directory number.

Table 61: LD 27 - Print a TSP configured for an external packet handler.

Prompt	Response	Description
REQ	PRT	Print configuration data.
TYPE	TSP	Print TSP data
DSL	lll s cc dsl#	DSL location, where: • lll (superloop) = 0-156 (must be zero or a number divisible by 4) • s (shelf) = 0-1 • cc (card) = 0-15 and for CS 1000E 1–20 • dsl# (DSL location) = 0-7 Assign 31 DSLs for each MISP if DCH or BDCH were specified at the PH prompt in "MISP configuration procedures".
OPT	USID SPID SUID DN DNS NTN	USID = Prints the TSP with the specified user service ID SPID = Prints the TSPs with the specified service profile ID SUID = Prints the specified service profile ID and the user service id map DN = Prints the TSP(s) that contains the specified DN DNS = Prints all the directory numbers defined for the DSL NTN = Prints the TSPs that contain the specified NTN. <cr> = Prints all the TSPs defined for the DSL
	<cr>	USID appears only when USID is the response.
USID	0-15	User service identifier
SPID	aa..aa	Service profile ID Enter a 1-20 digit alphanumeric service profile ID.
DN	nn..nn	Directory number associated with the TSP Enter a 1-digit to 7-digit directory number.
NTN	nn..nn	NTN value (1-10 digits)
REQ	...	

Initialize ISDN BRI terminals for an external packet handler

After configuring the TSPs, initialize the ISDN BRI terminals by entering the required parameter values at the terminal key pad or keyboard. The user manual shipped with each terminal provides instructions for initializing the terminal for a specific application.

Program the M5317TDX for an external packet handler

To access the Install Menu

1. Disconnect the line cord from the telephone for a few seconds, then reconnect.
2. Press and **hold** the **Release** key and the Hold key simultaneously until the display shows:

   Select `ROM` to execute `BOOTROM MAINROM`
3. Press the **MAINROM softkey**. After a few seconds, the display shows:

   Select configuration menu `Install Network Lang Exit more...`

Configuration for packet data

1. From the above display, press **Network**. The display shows:

   `SPID TEI SPM X.25DN more...`
2. Press the **TEI softkey**. The display shows:

   `Phone Data X25 ok`

   Note:

   MPH requires static TEI assignment.
3. Press the **X.25 softkey**.
4. Enter your 2 digit `TEI` and press **OK**.
5. Press the **X.25DN softkey**.
6. Press **Clear** to clear out any existing data.
7. Enter your `X.25 address` and press **OK**.
8. Press the **more... softkey**.
9. Press the **Exit softkey**.
10. Press the **Exit softkey** again.

How to use your M5317TDX for packet data

1. You will be in the Main Menu after completing step 10 above. From the Main Menu, press the **Setup softkey**.
2. Press the **Data softkey**.
3. Select your Baud Rate, Character Size and Parity.

   Note:

   You may use the defaults of 9600, 8 bits, no parity.
4. After setting these values, press the **more... softkey**.
5. Make sure the settings for DTR, CD HI, and RTS Hi are all set to YES. Press the **more... softkey**.
6. Press the **Data softkey** until the display shows `X25`.
7. Configure the packet and window sizes. The sizes should be selected to match the service data for the terminal.

   Press the **Pkt/Win softkey** until the display shows `No Neg`. You may select No Neg if you wish to choose the default packet/window size of 128/2 (which is the MPH's default setting). This setting does not result in flow control parameter negotiation.

   To select any other setting (such as 256/2), press the **Pkt/Win softkey** until the display shows the desired setting. This setting, and any other setting other than the default, results in the values being negotiated across the packet data interface, using flow control parameter negotiation.

Program the M5209TDcp for packet data

1. Press the **Hold** and **Release** keys simultaneously until the Main Menu appears on the set display.

   `MAIN MENU CONFIG`
2. Press #.

   `ENTER PASSWORD`
3. Dial `ISDN` and press # to enter configuration mode.

   `CONFIGURATION MENU TEI`
4. Press #.

   `ENTER TEI VOICE AUTO`
5. Press #.

   `ENTER TEI PSD XXX`
6. Enter your `TEI` and press #.

Note:

MPH requires static TEI assignment.

7. Press **#**.

   CONFIGURATION MENU SPID

8. Press ***** until EXIT appears on the display. Press **#**.

   MAIN MENU RING

9. Press ***** until DATA appears on the display. Press **#**.

   DATA MENU MODE

   Note:

   The data mode may also be selected by using a Hayes command across the serial interface of the M5209cp.

10. Press **#**.

    SELECT DATA MODE PACKET

11. Press **#**.

    DATA MENU PARAMS

12. Press ***** until the profile you want to use appears on the display.

    DATA MENU SELECT 0

    SELECT 0 selects PROF 90 profile.

    SELECT 1 selects PROF 91 profile.

    DEFAULT selects the factory default profile.

13. Press **#** to select the desired profile.

    DATA MENU EXIT

14. Press ***** until SAVE appears on the display.

    DATA MENU SAVE

15. Press **#** to save your changes.

16. Press **#**. Press **#** again.

Add an MPH

The following procedures, in the presented order, should be followed when configuring an MPH to a system with an existing ISDN BRI configuration. Please note, however, that when changing an existing ISDN BRI packet data configuration, following this order is unnecessary. Be aware, though, of the relationship of one component to another and whether changing one component necessitates changing other components.

1. Configure the Link Access Procedure on D-channel (LAPD) protocol group (LD 27).
2. Configure the Link Access Procedure - Balanced (LAPB) protocol group (LD 27).
3. Configure the X.25 packet protocol group (LD 27).
4. Configure the DNA table associated with the MPH network interface (LD 27).
5. Configure the ISDN Primary Rate Interface (PRI) for packet data as follows:
    a. ISDN PRI loop (LD 17).
    b. ISDN customer (LD 15).
    c. TIE trunk route for packet data (LD 16).
    d. TIE trunk for packet data (LD 14).

    OR,
6. Configure the Meridian Communication Unit (MCU) for packet data as follows:
    a. TIE trunk route for packet data (LD 16).
    b. TIE trunk for packet data (LD 14).
    c. Meridian Communication Unit (MCU) (LD 11).
7. Configure the MISP for MPH (LD 27).
8. Configure the BRSC for MPH (LD 27) (optional).
9. Configure the SILC/UILC for MPH (optional - The SILC or UILC can also be configured when configuring the DSL - see next step).
10. Configure the DSL for MPH (LD 27).
11. Configure the TSP for MPH (LD 27).
12. Configure Permanent Virtual Circuits (PVCs) (LD 27) (optional).
13. Configure MPH tandem connections (LD 14) (optional).
14. Modify CDR to reflect MPH calls (LD 15) (optional).
15. Initialize the ISDN BRI terminals for MPH (LD 27).

Configure an LAPD protocol group for an MPH

Add an LAPD protocol group by using LD 27 and specifying its protocol group number. You may also change its LAPD parameters as needed or accept the default values. LAPD is a transmission protocol that specifies the transmission timers, the maximum number of retransmissions, the size of the data frame, and the number of negative acknowledgments allowed before the system issues an alarm.

Table 62: LD 27 - Add or change an LAPD protocol group.

Prompt	Response	Description
REQ	NEW CHG	Add, change an ISDN protocol group.
TYPE	LAPD	LAPD Protocol group
PGPN	0-15 <cr>	Protocol group number The values for this prompt are: 0-15=Adds a specified protocol group <cr>=Stops this prompt from being displayed again
LAPD	YES (NO)	LAPD parameters —The values for this prompt are: YES = Define or modify the LAPD parameters (NO) = Does not prompt the LAPD parameters and assigns the default values shown in ( ) to these parameters.
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.
N2X4	0-(10)-20	For 1TR6 connectivity - number of status inquiries when the remote station is in peer busy state.
PGPN	<cr>	Press <cr> to prevent repetition of all the parameters starting with LAPD.

Remove an LAPD protocol group configured for an MPH

You can remove an LAPD protocol group as long as it is not assigned to a DSL. If a protocol group is assigned to a DSL, delete the DSL before removing the protocol group.

Table 63: LD 27 - Remove an LAPD protocol group.

Prompt	Response	Description
REQ	OUT	Remove an ISDN BRI component.
TYPE	LAPD	Protocol group
PGPN	0-15 ALL <cr>	Protocol group number 0-15 = Removes a specified protocol group from 0-15 ALL = Removes all protocol groups <cr> = No change the protocol group is not removed.

Prompt	Response	Description
...	...	A protocol group cannot be removed if it is assigned to a DSL.

Print an LAPD protocol group configured for an MPH

Configuration information for a specific LAPD protocol group or for all protocol groups can be printed.

Table 64: LD 27 - Print an LAPD protocol group.

Prompt	Response	Description
REQ	PRT	Prints an ISDN BRI component
TYPE	LAPD	Protocol group
PGPN	0-15 <cr>	Protocol group number 0-15 = Prints a specified protocol group from 0-15 <cr> = Prints all protocol groups and the number of DSLs in each group
USER	YES (NO)	YES = Print the LAPD group(s) selected in the PGPN prompt and the DSLs that are using it (them). NO = Do not print the LAPD user information.
REQ	...	

Configure an LAPB protocol group for an MPH

Add or change LAPB parameters for B-channel packet data or accept the default values. A set of timers and protocol values are associated with each group. Sixteen protocol parameter set groups can be defined per system. These groups are defined for use on the network and the user interface of MPH.

Table 65: LD 27 - Add or change an LAPB protocol group.

Prompt	Response	Description
REQ	NEW CHG	Add or change configuration data set
TYPE	LAPB	To administer the LAPB protocol group. Package #248 for MPH must be activated.
PGPN	0-15 <cr>	LAPB protocol set group number. Enter <cr> for none. PGPN prompts for another protocol group until <cr> is entered.

Prompt	Response	Description
LAPB	(NO) YES	NO = No change for the LAPB parameters YES = Change the LAPB parameters. Subsequent prompts are given only if LAPB = YES. If REQ = NEW and LAPB = NO, the default values for the timers which follow are assigned.
- T1	2-(6)-130	Response timer (in units of 0.5 seconds). Default is 3 seconds.
- T2	1-(4)-129	Maximum frames delay (in units of 0.5 seconds). Default is 2 seconds. **Note:** T2 must be less than T1.
- T3	0,3-(12)-131	Idle timer (in units of 0.5 seconds). Default is 2 seconds. **Note:** If T3 is not 0, then T3 must be greater than T1.
- N1	23-(135)-263	Maximum I-frame size (octets).
- N2	1-(10)-15	Maximum number of retries.
- K	1-(7)	Window size.
REQ	...	...

Remove an LAPB protocol group configured for an MPH

Note:

Do not remove an LAPB protocol group if it is assigned to an MPH or a TSP.

Table 66: LD 27 - Remove an LAPB protocol group.

Prompt	Response	Description
REQ	OUT	Remove an existing data set.
TYPE	LAPB	To remove an LAPB protocol group.
PGPN	0-15 ALL <cr>	LAPB protocol set group number to be removed. Enter ALL to remove all LAPB protocol groups that are not used. Enter <cr> to remove none. The LAPB protocol group to be deleted must not be referenced by an MPH network interface or a TSP.
...	...	

Print an LAPB protocol group for an MPH

You can print all or some of the LAPB protocol groups, by specifying the group number(s).

Table 67: LD 27 - Print an LAPB protocol group.

Prompt	Response	Description
REQ	PRT	Print the configuration.
TYPE	LAPB	To print an LAPB protocol group set data.
PGPN	0-15 <cr>	LAPB protocol set group number to be printed. Enter <cr> to print all defined LAPB groups, in ascending order.
USER	YES (NO)	YES = Print the LAPB group(s) selected in the PGPN prompt and the TSPs for the MPH network interfaces that are using the group(s). NO = Do not print the LAPB group user information.

Configure an X.25 protocol group for an MPH

The X.25 protocol set group allows for the grouping of sets of timers and protocol values for X.25 packet protocols. Sixteen protocol parameter set groups may be defined per system.

Table 68: LD 27 - Add or change an X.25 protocol group.

Prompt	Response	Description
REQ	NEW CHG	Add or change configuration data set. **Note:** If an X.25 group is changed, all of the active calls associated with the TSPs and MPH network interfaces using this group are dropped.
TYPE	X25P	To administer the X.25 protocol group. Package #248 for MPH must be activated.
PGPN	0-15 <cr>	X.25 protocol set group number. Enter <cr> for none. PGPN prompts for another protocol group until <cr> is entered.
X.25P	(NO) YES	NO = No change for the X.25 parameters YES = Change the X.25 parameters. Subsequent prompts are given only if X.25 = YES. If REQ = NEW and X.25 = NO, the default values for the timers which follow are assigned.
- T10/T20	15-(180)-930	Restart request timer (in seconds).

Prompt	Response	Description
- T11/T21	15-(180)-930	Call request timer (in seconds).
- T12/T22	15-(180)-930	Reset request timer (in seconds).
- T13/T23	0 15-(180)-930	Clear request timer (in seconds). If T13/T23 = 0, it is turned off.
- PSIZ	16 32 64 (128) 256	Default transmit packet size (in octets).
- WSIZ	1-(2)-7	Default transmit window size (in octets).
REQ	...	

Remove an X.25 protocol group

Note:

Do not remove an X.25 protocol group if it is assigned to an MPH or a TSP.

Table 69: LD 27 - Remove an X.25 protocol group.

Prompt	Response	Description
REQ	OUT	Remove an existing data set.
TYPE	LAPB	To remove an X.25 protocol group. Do not remove an X.25 protocol group if it is assigned to an MPH or a TSP
PGPN	0-15 ALL <cr>	X.25 protocol set group number to be removed. Enter ALL to remove all X.25 protocol groups that are not used. Enter <cr> to remove none.
...	...	

Print an X.25 protocol group for MPH

You may print all or some of the X.25 protocol groups, by specifying the group number(s).

Table 70: LD 27 - Print an X.25 protocol group.

Prompt	Response	Description
REQ	PRT	Print the configuration.
TYPE	X.25P	To print an X.25 protocol group set data.

Prompt	Response	Description
PGPN	0-15 <cr>	X.25 protocol set group number to be printed. Enter <cr> to print all defined X.25 groups, in ascending order.
USER	YES (NO)	YES = Print the X.25 group(s) selected in the PGPN prompt and the TSPs for the MMPH network interfaces that are using the group(s). NO = Do not print the X.25 group user information.
...	...	

Configure a Data Network Address table for an MPH

The MPH supports the CCITT X.121 Numbering Plan, which consists of up to 14 digits to specify the Data Network Address (DNA) of a Data Terminal Equipment (DTE). The DNA consists of a four digit Data Network Identification Code (DNIC) and a one-10 digit National Terminal Number (NTN). The DNIC consists of a three digit Data Country Code (DCC) and a single Network Digit (ND).

In summary, the X.121 DNA is composed as follows:

DNA = DNIC (DCC+ND) + NTN

- DNIC = zxxx (z can be 2-7; the digits 0 and 1 are reserved, and 8 and 9 are used for Telex; x can be 0-9)
- NTN = 0000000001-9999999999 (1-10 digits)

Note:

The DTA may be prefixed by a single digit (0-9), which, while transparent to the MPH, may have a local significance at the PSDN interface (typically used for international calls). This prefix may be entered in response to the PRFX prompt in LD 27, when configuring the MISP for the MPH.

The DNA numbers are administered using DNA tables, which are configured in LD 27. DNA tables contain the DNA numbers that are accessible to and by the PSDN. A DNA table number is assigned to each DNA table which is associated with a selected MPH network interface in the MISP configuration (there may be up to 32 DNA tables per MPH network). The DNA table number informs the network of available DNAs on that MPH network interface.

Note:

The craftsperson typically collects the required information, for entry to LD 27, from the MPH network administrator.

Table 71: LD 27 - Add or change a DNA table for an MPH.

Prompt	Response	Description
REQ	NEW CHG	Add or change configuration data set.

Prompt	Response	Description
		Note: You cannot change the DNIC of the DNA table.
TYPE	DNAT	To administer the DNA tables. Package #248 for MPH must be activated.
DNAT	1-32	DNA table number. Note that there is no default value.
DNIC	nnnn	Enter the 4-digit DNIC for the DNAT table. DNIC = zxxx (z can be 2-7; the digits 0 and 1 are reserved, and 8 and 9 are used for Telex; x can be 0-9. **Note:** You cannot change the DNIC of the DNA table.
NTN	nn...nn m...m n Xnn...nn Xm...m n <cr>	nn...nn = Add 1-10 digit National Terminal Number to the selected DNA table. m...m n = Enter a range of National Terminal Numbers to the selected DNA table. Enter the lowest NTN (m...m) in the table range (n, where n = 2-32). For example, entering 96765 20 adds to the DNA table an NTA number of 96765 with a table range of 96765 through 98764. **Note:** If REQ = NEW, leat 1 valid NTN must be entered for the table before using <cr> to exit the DNA table configuration. A maximum of 32 NTNs is allowed per DNA table. Enter Xnn to delete NTN nn from the table. Enter Xmn to delete a range mn NTNs from the table. <cr> = Do not add any NTN to the DNA table.
REQ	...	

Remove a DNA table configured for an MPH

Note:

You cannot remove a DNA table if it is assigned to an MPH or a TSP, or referenced by an MPH network interface.

Table 72: LD 27 - Remove a DNA table.

Prompt	Response	Description
REQ	OUT	Remove an existing data set.
TYPE	DNAT	To remove a DNA table.
DNAT	1-32	DNA table to be removed.

Prompt	Response	Description
...	...	

Print a DNA table configured for an MPH

You can print all or some of the DNA tables which have been configured for an MPH, by specifying the DNA number(s).

Table 73: LD 27 - Print a DNA table configured for an MPH.

Prompt	Response	Description
REQ	PRT	Print the configuration.
TYPE	DNAT	To print DNA table.
DNAT	1-32 <cr>	DNA table number to be printed. Enter <cr> to print all defined DNA tables, in ascending order.
USER	(NO) YES	NO = Do not print the DNA table user information. YES = Print the DNA tables selected in the DNAT prompt and the MMPH network interfaces that are using the DNA tables.
...	...	

Configure ISDN PRI trunk assignments for an MPH interface

The following trunk assignments must be configured for an MPH, when not using a Meridian Communications Unit (MCU).

- ISDN PRI loop (LD 17)
- ISDN customer (LD 15)
- TIE trunk route for packet data (LD 16)
- TIE trunk for packet data (LD 14)

Table 74: LD 17 - Add an ISDN PRI loop for an external MPH.

Prompt	Response	Description
REQ	NEW CHG	NEW for new customer or CHG for an existing customer
TYPE	CFN	Configuration data block
...	...	
CEQU	YES	Change common equipment options
...	...	
DLOP	lll dd ff	PRI loop parameters

Prompt	Response	Description
		lll (0-159) = Network loop number dd (0-24) = number of voice or data calls ff (ESF) = frame format: D2, D3, D4, Frame format must match the far end.
MODE	PRI	Primary Rate Interface mode
...	...	
PRI	0-159	PRI loop number

Table 75: LD 15 - Define an ISDN customer for an MPH.

Prompt	Response	Description
REQ:	NEW CHG	Add new data, or change existing data.
TYPE:	NET	Networking Data
CUST		Customer number
	0-99	Range for Large System and CS 1000E system.
...	...	
ISDN	YES	YES = customer is equipped with ISDN (prompted only with D-channel defined in LD17)
...	...	

Table 76: LD 16 - Configure a TIE trunk route for packet data for an MPH.

Prompt	Response	Description
REQ	NEW	Add ISDN BRI protocol group settings
TYPE	RDB	Route data block
CUST	xx	Customer number, as defined in LD 15
ROUT		Route number
	0-511	Range for Large System and CS 1000E system.
TKTP	TIE	Trunk route type
...	...	
DTRK	YES	Digital trunk route
BRIP	YES	Packet handler route **Note:** Prompted only if DTRK = YES.
ACOD	xxxxxx	Trunk route access code

Prompt	Response	Description
TARG	<cr>	Access restriction group number
CNTL	<cr>	Changes to control timers
...	...	

Table 77: LD 14 - Configure the TIE trunk for packet data for an MPH.

Prompt	Response	Description
REQ	NEW	Enter new trunk data.
TYPE	TIE	Trunk type
TN	l ch	Terminal number, where l = loop, ch = channel number.
CUST	xx	Customer number, as defined in LD 15
NCOS	<cr>	Network Class Of Service group
RTMB		Route number and Member Number
	0-511 1-4000	Range for Large System and CS 1000E system.
MNDN	<cr>	Manual directory number
TGAR	<cr>	Trunk group access restriction
CLS	<cr>	Class of Service
...	...	

Configure a Meridian Communication Unit network interface for an MPH

Use LD 11 to configure an interface between the MCU and the MPH (the MPHI prompt determines if the MCU is to be configured as an MPH interface.)

Before the MCU interface can be configured, the TIE trunk route (using LD 16) and TIE trunk (using LD 14) must be first configured; the procedures follow.

Table 78: LD 16 - Configure a TIE trunk route for an MCU to MPH interface.

Prompt	Response	Description
REQ	NEW	Add ISDN BRI protocol group settings.
TYPE	RDB	Route data block
CUST	xx	Customer number, as defined in LD 15
ROUT		Route number
	0-511	Range for Large System and CS 1000E system.

Prompt	Response	Description
TKTP	TIE	Trunk route type
...	...	
DTRK	YES	Digital trunk route
BRIP	YES	Packet handler route **Note:** Prompted only if DTRK = YES.
ACOD	xxxxxx	Trunk route access code
TARG	<cr>	Access restriction group number
CNTL	<cr>	Changes to control timers
...	...	

Table 79: LD 14 - Configure the TIE trunk for an MCU to MPH interface.

Prompt	Response	Description
REQ	NEW	Enter new trunk data.
TYPE	TIE	Trunk type
TN	l ch	Loop, channel number
CUST	xx	Customer number, as defined in LD 15
NCOS	<cr>	Network Class Of Service group
RTMB		Route number and Member Number
	0-511 1-4000	Range for Large System and CS 1000E system.
	0-127 1-4000	Range for Media Gateway 1000B.
MNDN	<cr>	Manual directory number
TGAR	<cr>	Trunk group access restriction
CLS	<cr>	Class of Service

Table 80: LD 11 - Add or change an MCU to MPH interface.

Prompt	Response	Description
REQ:	NEW CHG	Add new data, or change existing data.
TYPE:	MCU	To administer the MCU.
TN		Terminal number

Prompt	Response	Description
	l s c u	Format for Large System and CS 1000E system, where l = loop, s = shelf, c = card, u = unit.
CDEN	sd dd 4d	Single, double, or quadruple density. **Note:** Not prompted for superloops.
CUST	xx	Customer number, as defined in LD 15
MPHI	YES (NO)	Enter YES, for the MCU to be used with the MPH network interface. All subsequent prompts, except OPE, are skipped. Consequently, only the OPE and related prompts can be changed. NO = Do not use the MCU as the MPH network interface.
...	...	...
REQ		

Remove an MCU to MPH network interface

You may remove an MCU to MPH interface by specifying the loop, shelf, card and unit where it is located.

Table 81: LD 11 - Remove an MCU to MPH interface.

Prompt	Response	Description
REQ:	OUT	Remove an existing data set.
TYPE:	MCU	To remove the MCU interface.
TN		Terminal number
	l s c u	Format for Large System and CS 1000E system, where l = loop, s = shelf, c = card, u = unit.
REQ	...	

Configure a MISP for an MPH

An MPH may be downloaded to the MISP hardware. If the hardware platform runs only the MPH application, the MISP card serves as a dedicated MPH performing basic packet handling functions for the system. This configuration is done by using the procedure which follows.

The MISP must be enabled by using the `ENLL l` command in Network and IPE Diagnostic Program LD 32.

Table 82: LD 27 - Add or change a MISP configured for an MPH.

Prompt	Response	Description
REQ	NEW CHG	Add or change a new data set
TYPE	MISP	Administers the MISP card
LOOP	0-158	MISP loop number The MISP loop number must be an even number and the next odd loop number must be unequipped.
APPL	MPH <cr> XMPH	Enter MPH. <cr> = BRIL application BRIL and MPH may NOT be configured on the same MISP. XMPH = Remove the MPH application The MPH application may be removed only if there are: • no D-channel packet data separators and • no B-channel or D-channel terminals and • no network interfaces and • no PVC connections
PRFX	0-9 NO	Prefix to be used by the DNA tables of the MPH. NO = No prefix is selected. If the PSDN with which the MPH is interfacing uses a prefix, enter the prefix digit here.
NTNO	(YES) NO	YES = PSDN presents only the NTN in the called address field of incoming call packets. NO = PSDN presents both DNIC and NTN in the called address field of incoming call packets. Before changing the response to PRFX, NTNO, and DNIC, disable the MPH application.
DNIC	xxxx	Enter a 4-digit DNIC for the DNAs used with the MPH. Each MPH can support DNAs of only one DNIC. The DNIC may be changed only when the MPH has removed its association with any DNA table and TSP.
NWIF	n Xn <cr>	Enter the MPH network interface identifier for configuration (n = 1-3.) Xn = Remove the MPH network interface, n <cr> = none, completes the MISP configuration. Before changing the parameters TN, PVC, IC, TC, and OC, disable the associated network interface with the DSIF command in LD 32.
- TN	l s c u lll ch	Loop, shelf, card, unit of the MCU where the dedicated connection from the MPH is terminated. lll = PRI loop number which is connected to the MPH ch = the PRI channel on which the dedicated connection from the MPH is terminated.

Prompt	Response	Description
		Enter the TN of either the MCU or the PRI channel on which the dedicated connection from the MPH is terminated. The PRI loop must have been configured in LD 17 and it must be dedicated for the MPH or PSDN. The PRI channel must be configured for ISDN BRI packet data route in LD 16 and LD 14. The MCU must have been configured in LD 11. The associated network interface must be disabled prior to changing the PRI loop and channel number to a different value. If the PRI channel or the MCU has been chosen to serve as a network interface for an MPH, it can be selected to serve as a network interface for another MPH or as another network interface for the same MPH. If the channel is not free, the TN prompts for another PRI loop number and channel number or the same loop number with a free channel number.
- RATE	56 (64)	The PSDN communicates at either 56K or 64K across the PRI channel network interface. This prompt appears only if the input TN is on a PRI loop.
- LAPB	0-15	The LAPB protocol set group number to be used on the MPH network interface.
- X25P	0-15	The X.25 packet protocol set group number to be used on the MPH network interface.
- PVC	n1 n2	The range of Permanent Virtual Circuit Logical Channel Numbers. n1 = lowest PVC LCN (1-4095) n2 = highest PVC LCN (1-4095)
	<cr>	<cr> = none The Logical Channel Numbers values must follow this format: PVC must be less than IC which must be less than TC which must be less than OC. LCNs cannot be shared by PVC, IC, TC, or OC. Not all LCNs must be used.
- IC	n1 n2	The range o Incoming Logical Channel Numbers n1 = lowest Incoming LCN (1-4095) n2 = highest Incoming LCN (1-4095)
	<cr>	<cr> = none
- TC	n1 n2 <cr>	The range of Two-way Logical Channel Numbers. n1 = lowest Two-way LCN (1-4095). n2 = highest two-way LCN (1-4095). <cr> = none
- OC	n1 n2 <cr>	The range of Outgoing Logical Channel Numbers. n1 = lowest Outgoing LCN (1-4095). n2 = highest Outgoing LCN (1-4095) <cr> = none
- DNAT	nn Xnn <cr>	Associate the DNA table with the MPH network interface. nn = 1-32 Xnn = Remove DNA table from the MPH network interface. <cr> = none

Prompt	Response	Description
		Because only one DNIC can be used in an MPH, the DNA table number must have the same DNIC number as the MPH. To change a DNA table number, remove the old table number and then add the new. The table number to be removed must be associated with the MPH network interface. Each MPH network interface can be associated with a maximum of four DNA tables.
...		

Remove a MISP configured for an MPH

Before removing a MISP which has been configured for an MPH:

- Remove all BRSCs associated with it.
- Remove all DSLs connected to SILCs and UILCs associated with it.
- Disable the MISP loop with the DISL l command in LD 32.
- Remove the PVC or network interface connections.

Remove an MISP by specifying its loop number.

Table 83: LD 27 - Remove a MISP configured for an MPH.

Prompt	Response	Description
REQ	OUT	Remove an ISDN BRI component.
TYPE	MISP	MISP
LOOP	0-158	Loop number Must be an even number. The MISP must be disabled before removing it. All SILC and/or UILC DSLs associated with the MISP must be removed before removing the MISP.
REQ	...	

Print a MISP configured for an MPH

Print the configuration information for a MISP which has been configured for an MPH by specifying its network loop number. If the MISP network loop number is not known, use LD 22 to print the system configuration. BRSC TNs which are associated with the MISP are also listed when data of this MISP are printed.

Table 84: LD 22 - Print a MISP configured for an MPH.

Prompt	Response	Description
REQ	PRT	Print an ISDN BRI component.
TYPE	MISP	MISP BRSC TNs associated with a MISP are also listed when data of the MISP are printed.
LOOP	0-158	MISP loop number Must be an even number.
REQ	...	

Configure a BRSC for an MPH

To configure a BRSC for an MPH, follow these steps:

1. Disable the MISP to configure the first BRSC.
2. Disable all ISDN BRI line cards in an IPE Module.
3. Configure a BRSC in the IPE Module and select a MISP that can accommodate the BRSC.
4. Set up the connection to the internal packet handler, the Meridian 1 Packet Handler (MPH).
5. Enable the MISP.
6. Enable the BRSC with the ENLClll s cc command in LD 32.
7. Enable all ISDN line cards in the IPE Module.

To add or change a BRSC, specify its superloop number, shelf number, and card number, and specify the MISP loop for routing of the DPSD to the MPH.

Note:

Refer to the section "Add or remove BRSCs with a configured ISDN BRI system", which follows, for possible scenarios for adding or removing BRSCs with a system configured for ISDN BRI.

Table 85: LD 27 - Add or change a BRSC configured for an MPH.

Prompt	Response	Description
REQ	NEW CHG	Add or change BRSC.
TYPE	BRSC	BRSC
BRSC	lll s cc	Card location The values for this prompt are: lll (loop)=0-156 (must be an even number divisible by 4) s (shelf)=0-1 cc (card)=0-15

Prompt	Response	Description
MISP	0-158	MISP loop number (must be an even number that has already been configured)
DPSD	YES (NO)	Enter YES for D-channel Packet Switched Data NO = No D-channel Packet Switched Data
MPHC	(YES) NO	Enter YES or <cr> for DPSD to be routed to an MPH card. NO = DPSD are routed to an external packet handler.
MPH	0-158	MISP loop number for MPH application The MISP with MPH application where D-channel packet data are being sent. This prompt appears only if DPSD = YES and MPHC = YES. No more prompts of the current transaction appear for the MPH application.

Remove a BRSC configured for an MPH

In order to remove a BRSC which has been configured for an MPH:

- Disable all the line cards.
- Remove a BRSC by specifying its superloop number, shelf number, and card number.
- Disable the BRSC with the DISC lll s cc command in LD 32

Note:

After the BRSC is removed, the line cards must be associated with the MISP again.

Table 86: LD 27 - Remove a BRSC configured for an MPH.

Prompt	Response	Description
REQ	OUT	Remove an ISDN BRI component.
TYPE	BRSC	Remove the BRSC data.
BRSC	lll s cc	Card location The values for this prompt are: lll (loop)=0-156 (must be an even number divisible by 4) s (shelf)=0-1 cc (card)=0-15 BRSC can be removed only if the line cards are disabled and the BRSC is disabled. **Note:** After the BRSC is removed, the ISDN BRI line cards must be associated with the MISP again
REQ	...	

Print a BRSC configured for an MPH

Use LD 27 to print the following information for a BRSC which has been configured for an MPH:

- to print a BRSC, specify its superloop and shelf (card number is optional)
- to print a BRSC serving the superloop, specify its superloop number
- to print all BRSCs associated with a MISP, enter <cr> at BRSC prompt and specify the MISP loop number
- to print all BRSCs associated with a system, enter <cr> at both BRSC and MISP prompts.

Table 87: LD 27 - Print a BRSC configured for an MPH.

Prompt	Response	Description
REQ	PRT	Prints an ISDN BRI component.
TYPE	BRSC	Print BRSC data. **Note:** BRSC TNs associated with the MISP are also printed.
BRSC	lll s c lll s lll <cr>	Card location The values for this prompt are: lll (0-156) = superloop number s (0-1) = shelf number cc (0-15) = card number If input is: - lll s or lll s cc, data of the corresponding BRSC is printed. - lll, data of the BRSCs serving the superloop is printed. If <cr> is entered, the next prompt, MISP appears.
MISP	0-158 <cr>	MISP loop number (must be an even number that has already been configured). If <cr> is entered, all BRSCs configured in the system are printed; otherwise, all BRSCs associated with the MISP loop are printed.
REQ	...	

Add or remove BRSCs with a ISDN BRI system

To add or remove BRSCs with a system configured for ISDN BRI, follow these steps:

- Disable the MISP when the first BRSC is configured.
- If two or more line cards which are served by the MISP are in the IPE Module where the first BRSC is added, disable the MISP once only.

- Disable the MISP when changing from a configuration with three line cards and two BRSC to a configuration with two line cards and up to eight BRSCs.
- Conversely, disable the MISP when changing from a configuration with two line cards and up to eight BRSCs to a configuration with three line cards and one BRSC.

The following Table 88: Add or remove BRSCs with a configured ISDN BRI system on page 207 lists the possible scenarios for adding or removing BRSCs with a system configured for ISDN BRI.

Table 88: Add or remove BRSCs with a configured ISDN BRI system

Initial configuration			
Configured line cards installed	**BRSCs**	**Action**	**Same IPE Module?**
0 0	0 1	Add BRSC 1 Add BRSC 2-8	NA NA
1 1	0 1	Add BRSC 1 Add BRSC 2-8	Yes/No NA
2 2	0 1	Add BRSC 1 Add BRSC 2-8	Yes/No Yes/No
3	0	Add BRSC 1	Yes/No
3 2 0-2	0 1 2	Add BRSC 1 Add BRSC 2 Add BRSC 3-8	No Yes Yes/No
4 3 0-2	0 1 3-8	Add BRSC 1 Add BRSC 2 Add BRSC 3-8	Yes Yes Yes/No
2 2 3 3	2-8 1 1 0	Disable BRSC 1 Add line card 3 Delete BRSC Add line card 4	N/A N/A N/A N/A
2 2 3	2-8 0 0	Delete BRSCs Add line card 3 Add line card 4	N/A N/A N/A

Configure a SILC or UILC for an MPH

Add or change a new SILC or UILC for an MPH network interface by specifying its location, card type, and the MISP network loop that this card uses to transmit and receive signaling and D-channel packet data.

Note:

This step can be skipped and the card type specified when configuring the DSL in the procedure "Add a DSL."

The following procedure is used when configuring the SILC or UILC cards without configuring their DSLs.

Table 89: LD 27 - Add or change a SILC or UILC for an MPH.

Prompt	Response	Description
REQ	NEW CHG	Add or change a SILC or UILC line card.
TYPE	CARD	SILC or UILC line card
TN		Terminal number
	l s c	Format for Large System and CS 1000E system, where l = loop, s = shelf, c = card. The values for this prompt are: l = 0-156 (must be an even number, divisible by 4) s = 0-1 c = 0-15
MISP	0-158	Loop number (must be an even number that has already been configured) If there is a BRSC configured in the IPE module, the MISP prompt is skipped and the MISP lll and the BRSC lll s cc is displayed.
CTYP	SILC UILC	Card type to be added or changed. Remove any DSLs configured for this line card before changing the card type.
REQ	...	

Remove a SILC or UILC configured for an MPH

Remove a SILC or UILC configured for an MPH by specifying its card location. Before removing the SILC or UILC, all configured DSLs must first be removed from the card by using the procedure "Remove a DSL". When the last DSL is removed, the card is automatically deleted.

Note:

When removing the card, the database information is also deleted from the data block. Use LD 20 to list cards that have been removed.

Table 90: LD 27 - Remove a SILC or UILC configured for an MPH.

Prompt	Response	Description
REQ	OUT	Remove an ISDN BRI component
TYPE	CARD	ISDN BRI line card

Prompt	Response	Description
TN		Terminal number
	l s c	Format for Large System and CS 1000E system, where l = loop, s = shelf, c = card. The values for this prompt are: l = 0-156 (must be an even number, divisible by 4) s = 0-1 c = 0-15 Remove any DSLs that are configured for this card before removing the card.
...	...	

Print a SILC or UILC configured for an MPH

To print the configuration information for a SILC or UILC configured for an MPH, specify its card location.

Table 91: LD 27 - Print a SILC or UILC configured for an MPH.

Prompt	Response	Description
REQ	PRT	Print an ISDN BRI component.
TYPE	CARD	ISDN BRI line card
TN		Terminal number
	l s c	Format for Large System and CS 1000E system, where l = loop, s = shelf, c = card. The values for this prompt are: l = 0-156 (must be an even number, divisible by 4) s = 0-1 c = 0-15
...	...	

Configure a DSL for an MPH

To add or change a DSL for an MPH network interface, specify its port location and its DSL characteristics. DSL location specifies a SILC/UILC port connected to a DSL.

Table 92: LD 27 - Add or change a DSL for an MPH.

Prompt	Response	Description
REQ	NEW CHG	Add or change a DSL. **Note:** The defaults apply to adding, not changing, a DSL.
TYPE	DSL	DSL

Prompt	Response	Description
DSL	lll s cc dsl#	DSL location. 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
DES	x...x	Designator to assign to a DSL (ex. BUILD2) x...x = 1 to 6 alphanumeric DSL designator
CUST	xx	Customer number, as defined in LD 15
CTYP	SILC UILC	Card type. This prompt is displayed only if the SILC or UILC has not been previously configured.
OPT	(BRIL) <cr>	Defaults to ISDN BRI line application (BRIL). Enter <cr>.
MISP	0-158	Loop number (must be an even number of a MISP that has already been configured). This prompt is displayed only if the MISP has not been assigned to the specified SILC or UILC. If there is a BRSC configured in the IPE Module, the MISP prompt is skipped and the MISP lll and the BRSC lll s cc are displayed as shown in the following example: MISP 8 BRSC 24 0 15.
MODE	NTAS NTFS	Network terminal line sampling mode (this prompt is displayed only if you specified the card type as SILC). The values for this prompt are: NTAS = Adaptive sampling Extended passive bus, Branched passive bus, Point-to-point bus, U interface DSL. NTFS = Fixed sampling Short passive bus.
B1CT	(VCE) (DTA) PMD IPD XPMD XIPD	B-channel 1 call type. Enter IPD for B-channel packet data using the MPH. PMD = B-channel packet data with dedicated connection from DSL to a PRI channel using external packet handler For B1CT = IPD or PMD, B-channel packet data must have been specified at the PH prompt in LD 27. IPD and PMD cannot be combined with any other options. Do not select <cr> which defaults to VCE (circuit switched voice) and DTA (circuit switched data) which may not run concurrently with packet data. XPMD = Delete B-channel packet data using external packet handler. XIPD = Delete B-channel packet data using MPH.
MPH	0-158	MPH loop number associated with B-channel 1. The MPH loop must have been configured the MISP configuration, in LD 27. **Note:** If an MPH loop number is to be changed, the DSL has to be first disabled and any association of the MPH with any TSP has to be removed.

Prompt	Response	Description
B2CT	(VCE) (DTA) PMD IPD XPMD XIPD	B-channel 2 call type. Enter IPD for B-channel packet data using the MPH. PMD = B-channel packet data with dedicated connection from DSL to a PRI channel using external packet handler For B2CT = IPD or PMD, B-channel packet data must have been specified at the PH prompt in LD 27. IPD and PMD cannot be combined with any other options. Do not select <cr> which defaults to VCE (circuit switched voice) and DTA (circuit switched data) which may not run concurrently with packet data. XPMD = Delete B-channel packet data using external packet handler. XIPD = Delete B-channel packet data using MPH.
MPH	0-158	MPH loop number associated with B-channel 2. The MPH loop must have been configured the MISP configuration, in LD 27.
CH	1-24 1-30	**Note:** A response to this prompt is only required for packet data implementation. Enter the PRI2 (1-30) channel number that carries B-channel 2 packet data. The B-channel must be configured for packet data by using Trunk Route Administration Program LD 16 and Trunk Administration Program LD 14. **Note:** This prompt appears only if the call type for B2CT is set to PMD.
LDN	0-3 (NO)	Departmental listed directory number 0-3 = Departmental listed DN specified in LD 15 NO = No departmental listed DN associated with the DSL
XLST	(0)-254	Pretranslation group (if configured in customer data block).
MTEI	1-(8)-20	Maximum number of Terminal Endpoint Identifiers, both static and dynamic combined assigned to the logical terminals on this DSL.
LTEI	n1 n2 mm <cr> Xmm	The Logical Terminal Endpoint Identifier (LTEI) is used to address D-channel packet data terminals. LTEI consists of two components: n1 and n2 = Logical Terminal Identifier (LTID) m = Static Terminal Identifier (TEI) The maximum number of Logical Terminal Endpoint Identifiers (LTEIs) that can be configured is defined above by the prompt MTEI. The ranges for all entries are: n1 = Logical Terminal Group (LTG) = 1-15 n2 = Logical Terminal Number (LTN) = 1-1023 m = Static TEI = 0-63 **Note:** LTG=15 and LTN=1023 is an invalid combination.

Prompt	Response	Description
		Xm=Deletes LTID and TEI as a pair for the specified TEI. <cr> = Stops this prompt from being displayed again and skips to the next prompt. This prompt is displayed only if D-channel packet data was specified for the associated MISP.
MCAL	2-(16)-32	Maximum number of calls on the DSL at one time. This includes calls waiting and on hold. Warning is received if less than 8 is specified.
MTSP	1-(8)-16	Maximum number of TSPs allowed for a DSL
PGPN	0-15	Protocol group number The protocol group should be previously added as described in "Configure a protocol group."
PRID	1- 6	Defines the protocol to be used on the DSL Selection of the protocol ID is terminal dependent. The values for this prompt are: 1 = ANSI 2 = ETSI 3 = DMS 4 = NET64 5 = NUMERIS 6 = NI-1 **Note:** A response of 6 allows the ISDN BRI Conference feature to be configured in the TSP of the DSL.
FDN	n...n	Flexible CFNA directory number. Enter a 1-13 digit DN.
EFD	n...n	Flexible external call CFNA DN. Enter a 1-13 digit DN.
HUNT	n...n	Hunt directory number. Enter a 1-13 digit DN.
EHT	n...n	Hunt external call directory number. Enter a 1-13 digit DN.
TGAR	(0)-31	Trunk group access restriction
NCOS	(0)-99	Network Class Of Service
CLS		Class of Service access restrictions. More than one Class of Service can be entered by separating each entry with a space. Default features shown in parenthesis are selected by pressing <cr>.
	(ICDD) ICDA	Internal Call Detail Recording (Denied) Allowed
	(MRD) MRA	Message Restriction (Denied) Allowed
	(UDI) RDI	(Unrestricted) Restricted DID
	(UNR) CTD CUN FR1 FR2 FRE SRE TLD ICDA (ICDD) <cr>	(Unrestricted) Conditionally Toll Denied Conditionally Unrestricted Fully Restricted class 1 Fully Restricted class 2 Fully Restricted Semi-Restricted Toll Denied Internal Call Detail Recording allowed (Internal Call Detail Recording denied) Enter <cr> to select the defaults. More than one Class of Service may be selected by separating each entry with a space.

Prompt	Response	Description
REQ	...	

Remove a DSL configured for an MPH

Remove a DSL, which has been configured for an MPH, by specifying its location. To remove a DSL, first remove all the TSPs assigned to this DSL. When the last configured DSL on a card is removed, the card is removed automatically.

Table 93: LD 27 - Remove a DSL configured for an MPH.

Prompt	Response	Description
REQ	OUT	Remove an ISDN BRI component
TYPE	DSL	DSL
DSL	lll s cc dsl#	DSL location lll (superloop) = 0-156 (must be a number divisible by 4) s (shelf) = 0-1 cc (card) = 0-15 dsl# (DSL location) = 0-7
...	...	

Print a DSL configured for an MPH

Print the configuration information for a single DSL by specifying its location.

Table 94: LD 27 - Print a DSL configured for an MPH.

Prompt	Response	Description
REQ	PRT	Print an ISDN BRI component.
TYPE	DSL	DSL
DSL	lll s cc dsl# lll s cc lll s lll	DSL information lll s cc dsl# = Prints information for the specified dsl# lll s cc = Prints information for DSLs on the specified card lll s = Prints information for DSLs in the specified shelf lll = Prints information for DSLs on the specified loop
DATE	(<cr>) x y z	Print data and display the last active date, where x = day (1-31), y = month (Jan-Dec), and z = year (1979-9999) specifies the starting date of the data to be displayed or printed.
PAGE	YES (NO)	YES = prints one DSL per page NO = prints without paging
DES	<cr>	1-digit to 6-digit alphanumeric DSL designator No designator for DSLs
NACT	(NO)	Activity date is updated to current date.

Configure a TSP for an MPH

To configure a TSP to work with an MPH, define the service profiles for ISDN BRI terminals connected to a DSL, and the associated MPH packet data information.

Table 95: LD 27 - Add or change a TSP for an MPH.

Prompt	Response	Description
REQ	NEW CHG	Add or change a TSP
TYPE	TSP	Assign TSP to a DSL
DSL	lll s cc dsl#	DSL location lll (superloop) = 0-156 (must be a number divisible by 4)s (shelf) = 0-1 cc (card) = 0-15 dsl# (DSL location) = 0-7 The DSL must have been configured using the "DSL configuration procedures."
USID	0-15	User service identifier 0 is default TSP assigned to non-initializing terminals. The total number of TSPs defined for a DSL cannot exceed the maximum number of TSPs allowed for a DSL as specified by the MTSP prompt in the "DSL configuration procedures." A default TSP should be configured for non-initializing terminals. This is done by assigning USID=0 to the TSP.
MPHC	YES (NO)	Enter YES, to have the TSP work with an MPH. NO = The TSP is not used with an MPH **Note:** The status of MPHC cannot be changed. If CHG is entered in the REQ prompt, the MPHC and its stored response is the output, followed by the next applicable prompt.
TRMT	d b	d = Terminal type is for D-channel configuration b = Terminal type is for B-channel configuration If TRMT = d, then the LAPD protocol set group to be used by the MPH is the one selected in its associated DSL configuration. Before changing the terminal type from b to d, disable the terminal first with the DSIF command in LD 32. To change the terminal type, the associated MPH must have the same DNIC. If REQ = NEW/CHG and TRMT = b, then B1CT and/or B2CT of the associated DSL must = IPD. Response to TRMT largely determines the "associated MPH." For example, if TRMT = b and BCH = 1, the MPH associated with B1CT in the DSL configuration is used.

Prompt	Response	Description
		If TRMT = b and BCH = 2, the MPH associated with the B2CT is used. If TRMT = d, then the MPH selected by MISP or BRSC of associated ISDN BRI line card is used.
TEI	0-63	Static TEI for addressing terminal TEI is prompted only if TRMT = d. TEI is unique for the D-channel packet terminals within a DSL. An error message is printed is a TEI number if entered for more than one TSP of a DSL.
BCH	1-2	1 = TSP is associated with the B1 channel of the DSL. 2 = TSP is associated with the B2 channel of the DSL. The prompt BCH is given if TRMT = b If REQ = NEW/CHG and BCH = 1, then B1CT of associated DSL must = IPD. If REQ = NEW/CHG and BCH = 2, then B2CT of the associated DSL must = IPD. If a value to the BCH is to be changed (REQ = CHG), then the terminal has to be disabled using the DSIF command in LD 32.
LAPB	0-15	LAPB protocol set group number used on the MPH user interface. The prompt LAPB is given only if TRMT = b If REQ = NEW/CHG and TRMT = b, then B1CT and/or B2CT of the associated DSL must = IPD. LAPB must have been defined in LD 27.
X25P	0-15	X.25 packet protocol set group number used on the MPH user interface. If selected X.25 protocol set group does not exist, an error message is printed and the X25P prompt is repeated.
NTN	nn...nn	National Terminal Number of the TSP (1-10 digits.) **Note:** The Data Network Address (DNA) of the TSP is composed by combining the NTN entry with the DNIC of the associated MPH. "Associated MPH" is determined primarily by the response to the TRMT prompt. If TRMT = b, and BCH = 1, then the MPH associated with the B1CT in the DSL configuration will be used. If TRMT = b, and BCH = 2, then the MPH associated with the B2CT in the DSL configuration will be used. If TRMT = d, then the MPH selected by the Service Access Point Identifier (SAPI) separator (MISP or BRSC) of the associated ISDN BRI line card will be used.
PVC	n1 n2 <cr>	The range of the Permanent Virtual Circuit Logical Channel Number.

Prompt	Response	Description
		n1 = Lowest PVC LCN (1-4095) n2 = Highest PVC LCN (1-4095) <cr> = None. The lowest and highest Logical Channel Numbers selected define the range of logical channels to be used for the specified type of call connection. The Logical Channel Numbers values must follow this format: PVC lower than IC lower than TC lower than OC. **Note:** Before changing the value of PVC, the associated TSP must be first disabled.
IC	n1 n2 <cr>	The range of Incoming Logical Channel Number n1 = Lowest incoming LCN (1-4095) n2 = Highest incoming LCN (1-4095) <cr> None. **Note:** Before changing the value of IC, the associated TSP must be first disabled.
TC	n1 n2 <cr>	The range of Two-way Logical Channel Number. n1 = Lowest Two-way LCN (1-4095) n2 = Lowest Two-way LCN (1-4095) **Note:** Before changing the value of TC, the associated TSP must be first disabled.
OC		The range of Outgoing Logical Channel Number n1 = Lowest Outgoing LCN (1-4095) n2 = Highest Outgoing LCN (1-4095) <cr> = None **Note:** Before changing the value of OC, the associated TSP must be first disabled.
CDR	YES (NO)	YES = Turn on internal Call Detail Recording NO = Turn off internal CDR The local packet data CDR option is selected in the TSP, but it is overridden by the CDR option selection in LD15. If CDR = NO in LD15, the local CDR option is always considered OFF. But, if CDR = YES in LD15, the local CDR option is determined by the setting for the CDR prompt in the TSP configuration. For local packet data calls, the originator selection for CDR dominates.
FEAT	HTA (HTD) FNA (FND) SFA (SFD) CFTA (CFTD) MWA (MWD) FBA	Class of Service features HTA = Hunt allowed (always assign if terminal has CWT capability) HTD = Hunt denied FNA = Call forward no answer allowed FND = Call forward no answer denied SFA

Prompt	Response	Description
	(FBD) HBTA (HBTD)	= Second level call forward no answer allowed SFD = Second level call forward no answer denied CFTA = Call forward by call type allowed CFTD = Call forward by call type denied MWA = Message waiting allowed MWD = Message waiting denied FBA = Call forward busy allowed FBD = Call forward busy denied HBTA = Hunting by call type allowed HBTD = Hunting by call type denied
	DNO1 DNO2 (DNO3)	DNO1/DNO2/(DNO3) = QSIG Call Diversion Notification for calling party where: 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 (default). DNDN/(DNDY) = QSIG Call Diversion Notification for forwarded-to (diverted) party where:
	DNDN	DNDN = no notification of called party's number and name notification
	(DNDY)	(DNDY) = notification with called party's number and name when available (default). More than one Class of Service can be entered by separating each entry with a space. Press <cr> to select multiple default features shown in parenthesis.
DFDN	n...n	Default directory number Enter a 1-digit to 7-digit DN. This DN must be defined at the preceding DN prompt A DN can be associated with multiple TSPs. Only one default DN can be defined for a TSP. This DN is sent in the outgoing setup if the terminal does not send a calling line identification number with the outgoing call.
REQ	...	

Remove a TSP configured for an MPH

Before removing a TSP configured for an MPH:

- Disable the B-channel.
- Disable the TSPs associated terminal with the DSIF command in LD 32.
- If the TSP is associated with an existing PVC connection, remove the PVC first.

To remove a single TSP from a DSL, specify the DSL location and the user service identifier. Remove all TSPs from a DSL by entering ALL at the USID prompt.

Note:

Removal of the TSP disconnects all calls associated with the TSP's D-channel packet switched data and circuit switched voice data.

Table 96: LD 27 - Remove a TSP configured for an MPH.

Prompt	Response	Description
REQ	OUT	Remove an ISDN BRI component.
TYPE	TSP	TSP Before removing a TSP: • Disable the B-channel. • If MPHC = YES, disable the TSP's associated terminal with the DSIF command in LD 32. • If TSP is associated with an existing PVC connection, remove the PVC first. Removal of the TSP disconnects all calls associated with the TSP's D-channel packet switched data and circuit switched voice data.
DSL	lll s cc dsl#	DSL location lll (0-156) = superloop (must be an even number divisible by 4) (0 = invalid) s (0-1) = shelf cc (0-15) = card dsl# (0-7) = DSL location
USID	0-15 ALL	User service identifier 0-15 = Removes a specified TSP from 0 to 15 ALL = Removes all TSPs for the specified DSL
REQ	...	

Print a TSP configured for an MPH

Configuration information can be printed for a TSP configured for an MPH, based on characteristics such as user service identifier, service profile ID, and directory number.

Table 97: LD 27 - Print a TSP.

Prompt	Response	Description
REQ	PRT	Print an ISDN BRI component.
TYPE	TSP	TSP
DSL	lll s cc dsl#	DSL location lll (superloop) = 0-156 (even number divisible by 4) s (shelf) = 0-1 cc (card) = 0-15 dsl# (DSL location) = 0-7 The DSL must have been configured using "DSL configuration procedures."

Prompt	Response	Description
OPT	SPID USID SUID DN DNS NTN <cr>	Print option. SPID = Prints the TSPs with the specified service profile ID USID = Prints the TSP with the specified user service id SUID = Prints the profile ID and the user service id map DN = Prints the TSPs with the specified directory numbers DNS = Prints all the directory numbers defined for the DSL NTN = Prints the TSPs that contain the specific NTN <cr> = Prints all the TSPs defined for the DSL
SPID	x...x	Enter a 1-20 digit alphanumeric service profile ID. This prompt appears only if OPT = SPID.
USID	0-15	User service identifier. This prompt appears only if OPT = USID.
DN	x...x y...y	Directory Number associated with the TSP and CLID entry in LD 15. This prompt appears only if OPT = DN.
NTN	1-10	NTN value (1-10 digits). This prompt appears only if OPT = NTN.
REQ	...	

Configure a Permanent Virtual Circuit (PVC) connection for an MPH (optional)

The craftsperson may request the MPH to establish an internal PVC connection by entering two DNA numbers (NTN1 and NTN2) and their respective Logical Channel Numbers (LCN1 and LCN2).

An external PVC connection can be established with the entries of a DNA number (NTN1 and the unique DNIC used by the MPH), its associated Logical Channel Number (LCN1), the MPH network interface (NWIF) on which NTN1 resides, and the Logical Channel Number (LCN2) for the network interface.

The following procedures describe how to add or remove a PVC connection for an MPH, and how to print PVC information.

Note:

No changes to the PVC are allowed.

Table 98: LD 27 - Add or change PVC connection configuration.

Prompt	Response	Description
REQ	NEW	Add a new data set. PVC connections are not allowed to be changed spontaneously. The user has to remove the currently defined PVC connection

Prompt	Response	Description
		before using REQ = NEW to redefine it with a different configuration. No changes to PVC are allowed.
TYPE	PVC	To administer the PVC connection PVC is valid type only if package #248 for MPH has been activated. A warning message notifies the user at the end of the PVC configuration that the PVC connection is enabled only if the associated TSPs and MPH network interface are enabled.
MPH	0-158	Even loop number for the MPH MPH loop number is required to identify the MPH card on which these PVC connections are to be established.
PVCN	1-4	PVC connection number A maximum of 4 PVC connections is allowed for each MPH card
XPVC	(YES) NO	YES = external PVC connection is selected NO = internal PVC connection is selected For an external PVC connection, the user may establish a PVC connection between NTN1 and the physical MPH network interface on which the NTN1 resides.
NWIF	1-3	MPH network interface identifier. NWIF prompted only if XPVC = YES. The network interface must be disabled if it is chosen to work with the configured PVC. Otherwise, an error message is printed.
NTN1	nn..nn	nn..nn = 1-10 digits NTN1 = The first NTN of the PVC connection. NTN1 must have been associated with a configured TSP or an error message is printed. If XPVC = YES, then the NTN1 must also exist in the DNA tables associated with the selected MPH network interface. Otherwise, an error message is printed.
LCN1	1-4095	PVC Logical Channel Number associated with NTN1. LCN1 must be chosen from the range of logical channels defined in the parameters configuration of the applicable TSP. If the selected logical channel is occupied by another PVC connection, then an error message is printed and the logical channel prompt is repeated.
NTN2	nn...nn	nn..nn = 1-10 digits NTN2 = The second NTN of an internal PVC connection. NTN2 is prompted only if XPVC = NO. The user has chosen to establish an internal PVC connection.
- LCN2	1-4095	PVC Logical Channel Number associated with the MPH network interface or NTN2. The Logical Channel Number selected is associated with NTN2 for internal PVC and the MPH network interface for external PVC.

Prompt	Response	Description
		LCN2 must be chosen from the range of logical channels defined in the parameters configuration of the applicable TSP or the MPH network interface. If the selected logical channel is occupied by another PVC connection, then an error message is printed and the logical channel prompt is repeated.

Table 99: LD 27 - Remove PVC connection configuration data.

Prompt	Response	Description
REQ	OUT	Remove an existing data set.
TYPE	PVC	To remove the PVC connection number.
MPH	0-158	Even loop number for MPH.
PVCN	1-4 <cr>	PVC connection number to be removed. <cr> = none If the selected PVC connection number does not exist, an error message is printed and the prompt PVCN is repeated.

Table 100: LD 27 - Print PVC connection configuration data.

Prompt	Response	Description
REQ	PRT	Print configuration data.
TYPE	PVC	To print a PVC connection data.
MPH	0-158	Even loop number for MPH.
PVCN	1-4 <cr>	PVC connection data to be printed. <cr> = Print all PVC connection data. If the selected PVC connection number does not exist, an error message is printed and the PVCN prompt is repeated.

Configure a tandem connection for an MPH (optional)

Use LD 14 to configure trunk entries to allow an MPH to be used in a tandem connection between systems. The new type, TCON, provides a system access to PSDN through a dedicated connection in another system. The system establishes a dedicated connection between an incoming ISDN PRI channel (IPRI) and an outgoing ISDN PRI channel (OPRI) in the intervening system.

Up to 32 tandem connections can be configured in a system. The same loop and channel cannot be used for multiple TCON.

The PRI loop must have been configured in LD17. The PRI channel must have been configured for BRI packet data in LD 16 (TIE trunk routes) and LD 14 (TIE trunks).

The procedures which follow describe how to add or change, remove and print an MPH tandem connection configuration, respectively.

Table 101: LD 14 - Add or change a tandem connection configuration.

Prompt	Response	Description
REQ	NEW CHG	NEW = Adds a data set CHG = Drops existing connection and rebuilds it using the new PRI channels. No disabling of the previous PRI channels is required.
TYPE	TCON	To administer the tandem connection TCON is independent of the ISDN BRI line application and does not require the ISDN BRI line package to be activated.
T_TN	l ch	l (0-159) = PRI loop number ch (1-23) = PRI channel on which the dedicated connection from the other PRI channel is terminated. Prompted if REQ = CHG Enter either end of the tandem PRI connection to be changed. If the connection exists, the two PRI loop-channels are displayed.
IPRI	lll ch	Incoming PRI channel lll (0-159) = The PRI loop number ch (1-23) = The PRI channel on which the dedicated connection from the outgoing PRI channel is terminated. The PRI loop must have been configured in LD 17. The PRI channel must be configured for ISDN BRI packet data route using LD 16 and LD 14. If the channel is not free, an error message is displayed and the prompt IPRI or OPRI appears again. Enter another PRI loop number and channel number or the same loop number with a free channel number. The PRI channel for IPRI and OPRI must be different or an error message is printed.
OPRI	lll ch	Outgoing PRI channel lll (0-159) = The PRI loop number ch (1-23) = The PRI channel on which the dedicated connection from the incoming PRI channel is terminated.

Table 102: LD 14 - Remove a tandem connection configuration.

Prompt	Response	Description
REQ	OUT	Remove an existing configuration data set.
TYPE	TCON	To administer the tandem connection.
T_TN	lll ch	Enter either end of the tandem PRI connection to be changed. If the connection exists, the two PRI loop-channels are displayed. lll (0-159) = The PRI loop number ch (1-23) = The PRI channel on which the dedicated connection from the other PRI channel is terminated.

Table 103: LD 14 - Print a tandem connection configuration.

Prompt	Response	Description
REQ	PRT	Print an existing configuration data set.
TYPE	TCON	The tandem connection.
TN	l ch	Terminal number, where: l (0-159) = PRI loop number ch (1-23) = PRI channel on which the dedicated connection from the other PRI channel is terminated.
REQ	...	

Configure Call Detail Recording for the MPH (optional)

The following procedures describe how to add CDR information for the MPH, using LD 15 which configures the customer data block.

Table 104: LD 15 - Add Call Detail Recording for the MPH.

Prompt	Response	Description
REQ:	NEW CHG MOV	Add, change, or move a data set.
TYPE	CDR	Call Detail Recording data.
..	..	..
- IMPH	YES (NO)	CDR for incoming packet data call to MPH from PSDN
- OMPH	YES (NO)	CDR for outgoing packet data call by means of the MPH to PSDN
- AXID	YES (NO)	Auxiliary Identification Output in CDR records
..	..	..

Initialize ISDN BRI terminals for an MPH

After configuring the TSPs, initialize the ISDN BRI terminals by entering the required parameter values at the terminal key pad or keyboard. The user manual shipped with each terminal provides instructions for initializing the terminal for a specific application.

Program the M5317TDX for an external packet handler

To access the Install Menu

1. Disconnect the line cord from the telephone for a few seconds, then reconnect.
2. Press and hold the **Release key** and the **Hold key** simultaneously until the display shows:

   Select `ROM` to execute `BOOTROM MAINROM`
3. Press the **MAINROM softkey**. After a few seconds, the display shows:

   Select configuration menu `Install Network Lang Exit more...`

Configuration for packet data

1. From the above display, press **Network**. The display shows:

   `SPID TEI SPM X.25DN more...`
2. Press the **TEI softkey**. The display shows:

   `Phone Data X25 ok`

   Note:

   MPH requires static TEI assignment.
3. Press the **X.25 softkey**.
4. Enter your `2 digit TEI` and press **OK**.
5. Press the **X.25DN softkey**.
6. Press **Clear** to clear out any existing data.
7. Enter your `X.25 address` and press **OK**.
8. Press the **more... softkey**.
9. Press the **Exit softkey**.
10. Press the **Exit softkey** again.

How to use your M5317TDX for packet data

1. You will be in the Main Menu after completing step 10 above. From the Main Menu, press the **Setup softkey**.
2. Press the **Data softkey**.
3. Select your Baud Rate, Character Size and Parity.

   Note:

   You may use the defaults of 9600, 8 bits, no parity.
4. After setting these values, press the **more... softkey**.

5. Make sure the settings for DTR, CD HI, and RTS Hi are all set to YES. Press the **more... softkey**.
6. Press the **Data softkey** until the display shows `X25`.
7. Press **Pkt/Win** until the display shows `No Neg`. This setting results in a packet/window size of 128/2 which is the MPH's default setting. If you know your terminal is configured for a packet or window size other than this, then press **Pkt/Win** until the display shows the appropriate setting.

Program the M5209TDcp for packet data

1. Press the **Hold** and **Release** keys simultaneously until the Main Menu appears on the set display.

   `MAIN MENU CONFIG`

2. Press **#**.

   `ENTERPASSWORD`

3. Dial `ISDN` and press **#** to enter configuration mode.

   `CONFIGURATION MENU TEI`

4. Press **#**.

   `ENTER TEI VOICE AUTO`

5. Press **#**.

   `ENTER TEI PSD XXX`

6. Enter your `TEI` and press **#**.

   Note:

   MPH requires static TEI assignment.

7. Press **#**.

   `CONFIGURATION MENU SPID`

8. Press ***** until `EXIT` appears on the display. Press **#**.

   `MAIN MENU RING`

9. Press ***** until `DATA` appears on the display. Press **#**.

   `DATA MENU MODE`

10. Press **#**.

    `SELECT DATA MODE PACKET`

11. Press **#**.

DATA MENU PARAMS

12. Press * until the profile you want to use appears on the display.

    DATA MENU SELECT 0

    SELECT 0 selects PROF 90 profile.

    SELECT 1 selects PROF 91 profile.

    DEFAULT selects the factory default profile.

13. Press # to select the desired profile.

    DATA MENU EXIT

14. Press * until SAVE appears on the display.

    DATA MENU SAVE

15. Press # to save your changes.
16. Press #. Press # again.

Configure ISDN BRI trunk access

Note:

ISDN BRI trunk access is not supported in North America.

Configuration order for ISDN BRI trunk access

You must configure the following components in the order listed below to configure ISDN BRI trunk access.

1. Configure an ISDN customer, using overlay 15.
2. Configure trunk pad tables, using overlay 73 (optional).

   Note:

   The digital pad provides gain or attenuation values to condition the level of the digitized transmission signal according to the network loss plan. This determines transmission levels for the B-channel circuit-switched voice calls.

3. Configure the LAPD Protocol Group, using overlay 27.
4. Configure the ISDN BRI trunk route data block, using overlay 16.
5. Configure the MISP using overlay 27.
6. Configure the SILC and/or UILC card using overlay 27.

7. Configure trunk DSL, using overlay 27.
8. If the SILC clock is configured, enter the ISDN BRI trunk clock reference in overlay 73.

Define a customer for a trunk

Define or change an ISDN BRI trunk access customer using the Customer Data Block (LD 15).

Table 105: LD 15 - Define a customer for a trunk.

Prompt	Response	Description
REQ:	NEW CHG	Define a new customer, or change an existing customer.
TYPE:	NET	Networking Data
CUST		Customer number
	0-99	Range for Large System and CS 1000E system.
	0-31	Range for Media Gateway 1000B.
ISDN	YES	The customer is equipped with ISDN.

Configure pad tables for a trunk (optional)

This is an optional procedure. Define the pad settings if required.

The digital pad provides gain or attenuation values to condition the level of the digitized transmission signal according to the network loss plan. This determines transmission levels for the B-channel circuit-switched voice calls.

Table 106: LD 73 - Configure pad tables (optional).

Prompt	Response	Description
REQ	NEW	New settings.
TYPE	BRIT	Pad table type.
FEAT	PAD	Set the pad values used for ISDN BRIL
PDCA	1-16	Pad category table.
DFLT	(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
		The following prompts define the pad levels. The receiving pad code is r and the transmission pad code is t. These entries have the range 0-26. 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
OPX	r... t	Off-premises extension
DTT	r... t	Digital TIE trunks
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	2Mb DTI digital TOLL office trunks
ACO	r... t	Analog local exchange or WATS trunks
AFX	r... t	Analog FEX trunks
ADD	r... t	Analog DID trunks
SATT	r... t	Analog satellite TIE trunks
ATO	r... t	Analog TOLL office trunks
PRI2	r... t	2Mb PRI trunk (prompted only if the 1.5/2Mb Gateway feature is equipped and TYPE=2Mb PRI)
XUT	r... t	Analog local exchange trunk (prompted only if the 1.5/2Mb Gateway feature is equipped and TYPE=PRI2)
XEM	r... t	Analog TIE trunk (prompted only if the 1.5/2Mb Gateway feature is equipped and TYPE=PRI2)
BRIT	r... t	ISDN BRIT pad values. Valid inputs are 0-26. Refer to Table 107: ISDN BRI trunk pad codes and values on page 228.

Table 107: ISDN BRI trunk pad codes and values on page 228 shows ISDN BRI pad codes and their values. Positive dB represents loss and negative dB represents gain.

Table 107: ISDN BRI trunk pad codes and values

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					

Configure an LAPD a protocol group for a trunk

Add a Link Access Procedure on the D-channel (LAPD) protocol group, by using LD 27 and specifying its protocol group number. You can also change its LAPD parameters as needed or accept the default values. LAPD is a transmission protocol that specifies the transmission timers, the maximum number of retransmissions, the size of the data frame, and the number of negative acknowledgments allowed before the system issues an alarm.

Table 108: LD 27 - Add or change an LAPD protocol group for a trunk.

Prompt	Response	Description
REQ	NEW	Add an ISDN protocol group.
TYPE	LAPD	LAPD Protocol group
PGPN	0-15 <cr>	Protocol group number The values for this prompt are: 0-15 = Adds a specified protocol group <cr> = Stops this prompt from being displayed again
LAPD	YES (NO)	LAPD parameters —The values for this prompt are: YES = Define or modify the LAPD parameters. (NO) = Does not prompt the LAPD parameters and assigns the default values shown in ( ) to these parameters.
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.

Prompt	Response	Description
K	(1)-32	Maximum number of outstanding negative acknowledgment (NAKs) allowed before alarming the system.
N2X4	0-(10)-20	For 1TR6 connectivity - number of status inquiries when the remote station is in peer busy state.
PGPN	<cr>	Press <cr> to prevent repetition of all the parameters starting with LAPD.

Remove an LAPD protocol group for a trunk

You can remove a protocol group as long as it is not assigned to a DSL. If a protocol group is assigned to a DSL, delete the DSL before removing the protocol group.

Table 109: LD 27 - Remove an LAPD protocol group for a trunk.

Prompt	Response	Description
REQ	OUT	Remove an ISDN BRI component.
TYPE	LAPD	Protocol group
PGPN	0-15 ALL <cr>	Protocol group number 0-15 = Removes a specified protocol group from 0-15 ALL = Removes all protocol groups <cr> = No change the protocol group is not removed. A protocol group cannot be removed if it is assigned to a DSL.
...	...	...

Print a protocol group for a trunk

Configuration information for a specific protocol group or for all protocol groups can be printed.

Table 110: LD 27 - Print an LAPD protocol group for a trunk.

Prompt	Response	Description
REQ	PRT	Prints an ISDN BRI component.
TYPE	LAPD	Protocol group
PGPN	0-15 <cr>	Protocol group number 0-15 = Prints a specified protocol group from 0-15 <cr> = Prints all protocol groups and the number of DSLs in each group

Prompt	Response	Description
REQ	...	

Configure ISDN BRI trunk route data block parameters

ETS standards considerations

Route data block parameters for the ISDN BRI Trunk access capability are configured using LD 16.

Note:

In order to support countries that have not yet upgraded to the ETS 300 403 standard, the system 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, such as 7kHz telephony and Video telephony.

  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.

LD 16 - Configure ISDN BRI trunk route parameters

Prompt	Response	Description
REQ	NEW	Add ISDN BRI protocol group settings.
TYPE	RDB	Route data block.
CUST	xx	Customer number, as defined in LD 15
DMOD	<cr>	Default model number for this route
ROUT		Route number
	0-511	Range for Large System and CS 1000E system.
	0-127	Range for Media Gateway 1000B.
TKTP	TIE COT DID	TIE trunk type. COT Central Office Trunk type. Direct Inward Dialing trunk type.
RCLS	<cr>	Class marked route
DTRK	YES	Digital Trunk Route

Prompt	Response	Description
BRIP	NO	ISDN BRI packet handler route (NO is entered, since packet data is not required).
DGTP	BRI	Digital trunk type. The input to TKTP must be either TIE, COT, or DID.
- NASA	YES (NO)	Network Attendant Service Interface.
- MBGA	YES (NO)	Multi Business Group interface on the D Channel.
- IFC		DCH interface type.
	(SL1) EURO D100 D250 ESS4 ESS5 S100 SS12 AXEA AXES D70 ISIG ISGF ESIG ESGF 1TR6	SL1 = Meridian 1, Avaya Communication Server 1000M (Avaya CS 1000M) EURO = EuroISDN D100 = Meridian DMS-100 D250 = Meridian DMS-250 ESS4 = AT&T ESS#4 ESS5 = AT&T ESS#5 S100 = Meridian SL-100 SS12 = SYS-12 for Norway AXE = Ericsson AXE-10 for Australia AXS = Ericsson AXE-10 for Sweden D70 = Japan D70 ISIG = ISO QSIG ISIG with GF platform ESIG = ETSI QSIG ESIG with GF platform 1TR6 = Germany 1TR6
	E403	EuroISDN Interface conforming to the ETS 300 403, or the extended ETS 300 102 version for country-specific interfaces. Refer to the ETS standards considerations on page 231.
- - CNTY		Enter the country pertaining to EuroISDN or APAC. For EuroISDN:
	AUS DEN (ETSI) FIN GER ITA NOR POR SWE EIR DUT SWI BEL ESP EAUS	Austria Denmark ETS 300-102 basic protocol Finland Germany Italy Norway Portugal Sweden Ireland Holland Switzerland Belgium Spain United Kingdom Australia ETSI
		Enter country pertaining to the E403 interface. If either ETSI or NET are entered, the interface is fully compliant with ETS 300 403. See the ETS standards considerations on page 231.
	ETSI NET	ETS 300 403 for the user side. ETS 300 403 for the network side. If any of the countries listed for the EURO IFC are entered, the interface functions with the extended ETS 300 102 capabilities. See the ETS standards considerations on page 231.
	AUST CHNA HKNG INDI	Enter the CNTY pertaining to Asia Pacific: Australia. China. Hong Kong. India. Indonesia. Japan. Malaysia. Philippines. Singapore. Taiwan. New Zealand. Thailand.

Prompt	Response	Description
	INDO JAPN MSIA PHLP SING TAIW TCNZ THAI	
...		
- CDR	(NO) YES	YES = CDR on route (NO) = No CDR on trunk route If answer supervision is defined for the trunk, CDR records generates only on call completion.
...		
CLID	OPT4	OPT4 is the CLID default for the Asia Pacific interface.
PROG	NCHG MALE	Progress signal. No Change. This is the default for all Asia Pacific interfaces except Singapore, Japan, and Australia. Alert message.
	MCON	Connect message. This is the default for the Australia interface. **Note:** The PROG prompt should not be configured for APAC Japan and Singapore interfaces, since these countries do not support the Progress signal.
...	...	...
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 is applies to APAC only if CPFXS = NO. 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 is the case with PFX2, the HLCL prefix can be from one-to-four digits long.
...		
- DSEL		Data selection.

Prompt	Response	Description
	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.
...		
- OTL	(NO) YES	YES = CDR on outgoing toll calls (NO) = No CDR on outgoing toll calls.
...		
- OAN	(NO) YES	YES = CDR on all answered outgoing toll calls (NO) = No CDR on all answered outgoing toll calls
- MR	DURC ENDC STAC	Allow Advice of Charge for EuroISDN, Australia ETSI, Japan D70, or Australia (for Asia Pacific interface). DURC = Activation of the AOC-D subservice ENDC = Activation of the AOC-E subservice STAC = Activation of the AOC-S subservice.
- - RUCS	0 - 9999	Route unit cost. This prompt does not appear for Denmark or Sweden.
- - RURC	0 - 9999 (0) - 3	Route unit reference cost. Note that the formula for the route unit reference cost is: $X*10^{(-Y)}$. where X = 0 - 9999, Y = 0 - 3 The default value of X is identical to the previously entered RUCS value. This prompt does not appear for Denmark or Sweden.
- - RUCF	0 - (1) - 9999 (0) - 3	Route unit conversion factor. This prompt does not appear for Denmark or Sweden.
...	...	...
MCTS	YES	Enable Malicious Call Trace signaling for EAUS, AUST, or TCNZ.

Prompt	Response	Description
- 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 (EAUS, AUST, or TCNZ only).
- MTND	(NO) YES	Malicious Call Trace disconnect delay for tandem calls (EAUS, AUST, or TCNZ only).
...		
- SIDE	NET USR	CS 1000M node type (either network or user), prompted only if IFC = SL1. If IFC is not SL-1, it defaults to USR. If IFC is SL1, it defaults to NET.

Prompt	Response	Description
		Note: SIDE cannot be changed from NET to USR if NT mode members exist on the route; NT mode DSLs must be on NET side.
- CNEG	(1)	Channel negotiation option. Channel is indicated and no alternative acceptable, exclusive.
	2	Channel is indicated and any alternative acceptable, preferred. **Note:** for the APAC Singapore interface, CNEG must be set to 1.
...	...	...
- PGPN	0-15	Protocol Group, as defined previously in overlay 27. PGPN cannot be changed without disabling all ISDN BRI trunk members associated with this route.
- RCAP		Remote D-channel capabilities. Enter X followed by the option to remove the configured capability. This prompt is repeated until <cr> is entered.
	NCT RVQ ND1 ND2 NAS BRI COLP	NCT = Network Call Trace RVQ = Remote Virtual Queueing ND1 = Network Name Display 1 ND2 = Network Name Display 2 NAS = Network Attendant Service BRI = allows ISDN line/ trunk interworking) COLP = Connected Line ID supplementary service (for APAC Indonesia, India, Taiwan, and Philippines interfaces).
	DV1I DV1O DV2I DV2O DV3I DV3O	These are QSIG SS Call Diversion Notification remote capability responses, used to 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 Remote Capability Meanings for ISDN BRI routes on page 238 for more information.
	CCBO CCBI XCCB	Enter the Operation Coding method for the QSIG/ETSI Call Completion to a Busy Subscriber supplementary service. Coding by Object ID. Coding by Integer Value. Remove the CCBO or CCBI value. **Note:** CCBO and CCBI are mutually exclusive.

Prompt	Response	Description
	CCNO CCNI XCCN	Enter the Operation Coding method for the QSIG Call Completion on No Reply supplementary service. Coding by Object ID. Coding by Integer Value. Remove the CCBO or CCBI value. **Note:** CCNO and CCNI are mutually exclusive.
	CTI XCTI CTO XCTO	Enter the Operation Coding method for the QSIG Call Transfer Notification feature. 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.
	NDO NDI	Enter the Operation Coding method for the QSIG Name Display supplementary services. 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.
	MCID	Allow Malicious Call Trace for Australia ETSI.
	MQC	Add MCDN QSIG Conversion, for the MCDN End to End Transparency.
- - MQC_FEAT		MCDN QSIG feature type. Prompted if RCAP = MQC. Precede MQC feature type with an X to remove the value.
	NAS NACD NMS	Enable NAS on QSIG. Enable NACD on QSIG. Enable NMS-MC and NMS-MM on QSIG.
...	...	...
TIMR	YES	Set programmable 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

Prompt	Response	Description
		seconds. SING = 10 seconds. TCNZ = 10 seconds. THAI = 10 seconds.
- - INC_T306	0-(2)-T306	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 stops sending after T306 times out, so the maximum time is T306. The value is stored in two-second increments, which are rounded up. T306 can be defined up to 30 seconds for all APAC interfaces except Australia, which can have T306 defined up to 60 seconds.
- - OUT_T306	0-(30)-T306	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 stops sending after T306 times out, so the maximum time is T306. The value is stored in two-second increments, which are rounded up. T306 can be defined up to 30 seconds for all APAC interfaces except Australia, which can have T306 defined up to 60 seconds.
...	...	...
- OVLR	YES (NO)	Overlap Receiving Allow/Disallow OVLR is not prompted if IFC = NUME; it defaults to NO. For APAC, it is prompted for all CNTY interfaces except TAIW.
- DIDD	(0)-15	Number of leading digits that are ignored for DID calls during Overlap Receiving.
- OVLS	YES (NO)	Allow (disallow) Overlap Sending. OVLS is not prompted if IFC = NUME; it default to NO. For APAC, 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.
...	....	....

Remote Capability Meanings for ISDN BRI routes

Remote capability	Meaning for Operation Coding	Meaning for Notification Informations	Meaning for Rerouting request
None of the following remote capabilities.	Not applicable (nothing sent)	Not sent	Not processed when received
DV1O	Sent coded as Object Identifier	Sent	Not processed when received
DV1I	Sent coded as Integer Value		
DV2O	Sent coded as Object Identifier	Not Sent	Processed when received
DV2I	Sent coded as Integer Value		
DV3O	Sent coded as Object Identifier	Sent	Processed when received
DV3I	Sent coded as Integer Value		

When using the preceding table 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.

Only one remote capability allows the QSIG Diversion configuration on an ISDN BRI route. This remote capability gathers the three following possibilities for the route:

- coding of operations 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 as 'O', whereas for Integer Value, the remote capability ends with as 'I'. This means that remote capabilities explained below in 2 and 3 are defined twice.
- sending of QSIG Diversion Notification Information to the remote switch: these informations are 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.
- 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.

Configure a MISP for a trunk

To add or change a MISP for a trunk, specify its even loop number.

The MISP must be enabled by using the **ENLL l** command in Network and IPE Diagnostic Program LD 32.

Table 111: LD 27 - Add or change MISP for a trunk.

Prompt	Response	Description
REQ	NEW CHG	Add or change a MISP.
TYPE	MISP	MISP
LOOP	0-158	MISP loop number must be an even number, with the next odd loop number unequipped.
	1-9	MISP card slot number has to be 1-4, 7-10 for Media Gateway 1000 1-10 for Media Gateway 1010
APPL	BRIT BRIE	Application type for the DSL. Precede with a "X" to remove an entered value.
		Enter BRIT for the following:
		SL1 = Meridian 1, CS 1000M SS12 = SYS-12 for Norway AXE = Ericsson AXE-10 for Australia AXS = Ericsson AXE-10 for Sweden D70 = Japan D70 1TR6 = Germany 1TR6 NUME = France Numeris TCNZ = Telecom New Zealand (NEAX-61) interface.
		Enter BRIE for one of the following: EURO = EuroISDN ISIG = ISO QSIG ISGF = ISIG with GF platform ESIG = ETSI QSIG ESGF = ESIG with GF platform APAC = Asia Pacific APPL is prompted until <cr> is entered.
DSPD	YES (NO)	YES = D-channel Packet Switched Data (NO) = No D-channel Packet Switched Data. Use the default value (NO). Subsequent prompts is skipped.
...	...	

Remove a MISP configured for a trunk

Remove a MISP which has been configured for a trunk by specifying its loop number. Before removing the MISP, remove all DSLs connected to SILCs and UILCs associated with the MISP.

Disable the MISP loop with the **`DISL l`** command in LD 32.

Table 112: LD 27 - Remove a MISP configured for a trunk.

Prompt	Response	Description
REQ	OUT	Remove an ISDN BRI component.
TYPE	MISP	MISP
LOOP	0-158	Loop number must be an even number.
	1-9	MISP card slot number has to be 1-4, 7-10 for Media Gateway 1000 1-10 for Media Gateway 1010 All BRSCs and SILC and/or UILC DSLs associated with the MISP must be removed before removing the MISP. See the section "Remove a SILC or UILC configured for a trunk" found later in this chapter.
REQ	...	

Print a MISP configured for a trunk

Print the configuration information for a MISP which has been configured for a trunk by specifying its network loop number. If the MISP network loop number is not known, use LD 22 to print the system configuration.

Table 113: LD 22 - Print a MISP configured for a trunk.

Prompt	Response	Description
REQ	PRT	Prints an ISDN BRI component.
TYPE	MISP	Print the MISP data.
LOOP	0-158 1-9	Loop number for the MISP must be an even number.
	<cr>	Print all MISPs in system.

Configure a SILC or UILC for a trunk

Add or change a new SILC or UILC for trunk access by specifying its location, card type, and the MISP network loop that this card uses to transmit and receive signaling.

Note:

This step may be skipped and the card type specified when configuring the DSL in "Add a DSL for a trunk." The following procedure is used when configuring the SILC or UILC cards without configuring their DSLs.

Table 114: LD 27 - Add or change a SILC or UILC for a trunk.

Prompt	Response	Description
REQ	NEW CHG	Add or change a SILC or UILC line card.
TYPE	CARD	SILC or UILC line card
TN		Terminal number
	l s c	Format, where: l (Loop)= • 0-156 (must be an even number divisible by 4) s (Shelf)= • 0–1 c (Card)= • 0-15 for Large System, 1-4, 7-10 for chassis, 1-10 for cabinet and MG 1010
MISP	0-158	Loop number must be an even number that has already been configured.
CTYP	SILC UILC	Card type to be added or changed. Remove any DSLs configured for this line card before changing the card type.

Remove a SILC or UILC configured for a trunk

Remove a SILC or UILC which has been configured for a trunk by specifying its card location. Before removing the SILC or UILC, all configured DSLs must first be removed from the card by using the "Remove a DSL configured for a trunk" procedure. When the last DSL is removed, the card is automatically deleted.

When removing the card, the database information is also deleted from the data block. Use LD 20 to list cards that have been removed.

Table 115: LD 27 - Remove a SILC or UILC configured for a trunk.

Prompt	Response	Description
REQ	OUT	Remove an ISDN BRI component
TYPE	CARD	ISDN BRI line card
TN		Terminal number
	l s c	Format, where:

Prompt	Response	Description
		l (Loop)= • 0-156 (must be an even number divisible by 4) s (Shelf)= • 0–1 c (Card)= • 0-15 for Large System, 1-4, 7-10 for chassis, 1-10 for cabinet and MG 1010 Remove any DSLs that are configured for this card before removing the card.

Print a SILC or UILC configured for a trunk

To print the configuration information for a SILC or UILC which has been configured for a trunk, specify its card location.

Table 116: LD 27 - Print a SILC or UILC configured for a trunk.

Prompt	Response	Description
REQ	PRT	Print an ISDN BRI component
TYPE	CARD	ISDN BRI line card
TN		Terminal number
	l s c	Format, where: l (Loop)= • 0-156 (must be an even number divisible by 4) s (Shelf)= • 0–1 c (Card)= • 0-15 for Large System, 1-4, 7-10 for chassis, 1-10 for cabinet and MG 1010

Configure a DSL for a trunk

To add or change a DSL for trunk access, specify its port location and its DSL characteristics. DSL location specifies a SILC/UILC port connected to a DSL.

You can change the characteristics of a DSL by changing one or more parameters to adapt it to new transmission or feature requirements. If you wish to skip a parameter, press the Enter key and the next prompt appears. The DSL must be idle or disabled before making a change. Use the **STAT l s c dsl#** and **DISU l s c dsl#** commands in LD 32 to query the status of the DSL and to disable it.

Table 117: LD 27 - Add or change a DSL for a trunk.

Prompt	Response	Description
REQ	NEW CHG	Add or change a DSL.
TYPE	DSL	DSL
DSL	lll s cc dsl#	DSL location, where: • lll (superloop) = 0-156 (must be zero or a number divisible by 4) • s (shelf) = 0-1 • cc (card) = 0-15 and for CS 1000E 1–20 • dsl# (DSL location) = 0-7 Assign 31 DSLs for each MISP if DCH or BDCH were specified at the PH prompt in "MISP configuration procedures".
APPL	BRIT BRIE	Application type for the DSL. Precede with a "X" to remove an entered value. Enter BRIT for the following: SL1 = Meridian 1, CS 1000M, SS12 = SYS-12 for Norway AXE = Ericsson AXE-10 for Australia AXS = Ericsson AXE-10 for Sweden D70 = Japan D70 1TR6 = Germany 1TR6 NUME = France Numeris TCNZ = Telecom New Zealand (NEAX-61) interface. Enter BRIE for Enhanced ISDN BRI trunking. BRIE is entered for one of the following: EURO = EuroISDN ISIG = ISO QSIG ISGF = ISIG with GF platform ESIG = ETSI QSIG ESGF = ESIG with GF platform APAC = Asia Pacific APPL is prompted until <cr> is entered.
CUST	xx	Customer number, as defined in LD 15
CTYP	SILC UILC	The card type (enter SILC or UILC as appropriate).
MISP	0-158 1-9	Loop number for Options 51C - 81C (must be an even number of an already configured MISP).
MODE	(TE) NT	The mode for the trunk DSL. TE is entered for Terminal Equipment, NT is used for Network Termination. This prompt is displayed only if SILC was specified as the card type. For UILC, this entry defaults to NT mode. For SILC, the default is TE.

Prompt	Response	Description
		Note: The mode cannot be changed from TE to NT if the clock on the DSL is referenced in the Digital Data Block or the DTI2/PRI2 system data. The reference must be first removed. If the mode is changed to NT, CLOK is set to NO.
B1CT	VCE DTA	B-Channel 1 call type is voice and data, for the EuroISDN 7kHz/Videotelephony Teleservice. **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, for the EuroISDN 7kHz/Videotelephony Teleservice. **Note:** At least one of the B-Channels must be configured for voice and data.
...		
MTFM	YES (NO)	Enable/Disable multi-frame option, prompted only for TE mode DSLs. If enabled this prompt allows you to receive more diagnostic messages.
TKTP	TIE COT DID	Trunk type
CLOK	YES (NO)	Whether this trunk DSL is provisioned for clock source. This prompt appears if the following conditions are met: • the card type is SILC • the DSL# is 0 or 1 • the trunk DSL has been defined as TE mode **Note:** The clock prompt cannot be changed from YES to NO if the clock on the DSL is referenced in the Digital Data Block or the DTI2/PRI2 system data. The reference must be first removed. Also, you cannot out a trunk DSL if an active clock exists on it and is referenced in the Digital Data Block or the DTI2/PRI2 system data; this reference must be first removed.
PDCA	(1)-16	Pad table number (previously configured in LD 73) to be associated with this DSL
ROUT		Route number for the trunk DSL. The specified route must match the ISDN BRI route type as well as the trunk type specified at the TKPT prompt.

Prompt	Response	Description
	0-511	Range for Large System and CS 1000E system.
	0-127	Range for Media Gateway 1000B. If the DSL is in the NT mode (MODE = NT in LD 27), the entered route must be on the network side (SIDE = NET in LD 16).
B1	YES (NO)	Configure B Channel 1. If REQ = NEW, a response to this prompt is not required, because B1 parameters are mandatory. The system automatically displays the prompts that follow. If REQ = CHG and ROUT was changed, B1 automatically is YES, since a new member number must be entered.
- MEMB	1-254	Route member number to be associated with B-channel 1.
- TGAR	(0)-31	Trunk Group Access Restriction.
- NCOS	(0)-99	Network Class of Service Group Number
- CLS		Class of Service options.
	(APN) APY (UNR) CTD CUN FR1 FR2 FRE SRE TLD MRA	APN = ACD Priority not allowed APY = ACD Priority allowed UNR = Unrestricted (default) CTD = Conditionally Toll Denied (valid for TIE trunks only) CUN = Conditionally Unrestricted (valid for TIE trunks only) FR1 = Fully Restricted class 1 (valid for TIE trunks only) FR2 = Fully Restricted class 2 (valid for TIE trunks only) FRE = Fully Restricted (valid for TIE trunks only) SRE = Semi-Restricted (valid for TIE trunks only) TLD = Toll Denied (valid for TIE trunks only). MRA = Message Registration Allowed (assigning meters to ISDN BRI sets, for Advice of Charge for EuroISDN, APAC (Australia and Japan), or Australia-AXE and Japan D70.
	(MRD) PGNA (PGND)	MRD = Message registration denied. PGNA = Call Page Network Wide Allowed. (PGND) = Call Page Network Wide Denied.
B2	YES (NO)	Configure B Channel 2. If REQ = NEW, and the default of NO is entered to this prompt, all parameters entered for B1 will be applied to B2, except the route member number will be an unused value. The message "B2 will use Route # Member #" will be displayed. If REQ = CHG, if NO is entered and ROUT was changed, all parameters for B2 will remain the same except that the route member will be an unused member number. The message "B2 will use Route # Member #" will be displayed.
- MEMB	1-254	Route member number to be associated with B-channel 2.
- TGAR	(0)-31	Trunk Group Access Restriction.
- NCOS	(0)-99	Network Class of Service Group Number

Remove a DSL configured for a trunk

Remove a DSL which has been configured for a trunk by specifying its location. When the last configured DSL on a card is removed, all active calls are dropped.

The DSL must be idle or disabled before being removed. Use the **`STAT l s c dsl#`** and **`DISU l s c dsl#`** commands in LD 32 to query the status of the DSL and to disable it.

Table 118: LD 27 - Remove a DSL configured for a trunk.

Prompt	Response	Description
REQ	OUT	Remove an ISDN BRI component.
TYPE	DSL	DSL
DSL	lll s cc dsl#	DSL location, where: • lll (superloop) = 0-156 (must be zero or a number divisible by 4) • s (shelf) = 0-1 • cc (card) = 0-15 and for CS 1000E 1–20 • dsl# (DSL location) = 0-7 Assign 31 DSLs for each MISP if DCH or BDCH were specified at the PH prompt in "MISP configuration procedures".

Print a DSL configured for a trunk

Print the configuration information for a single DSL by specifying its location.

Table 119: LD 27 - Print a DSL configured for a trunk.

Prompt	Response	Description
REQ	PRT	Print an ISDN BRI component.
TYPE	DSL	DSL
DSL	lll s cc dsl# lll s cc lll s lll	DSL information for Large Systems. lll s cc dsl# = Prints information for the specified dsl# lll s cc = Prints information for DSLs on the specified card lll s = Prints information for DSLs in the specified shelf lll = Prints information for DSLs on the specified loop DSL location, where: • lll (superloop) = 0-156 (must be zero or a number divisible by 4) • s (shelf) = 0-1

Prompt	Response	Description
		• cc (card) = 0-15 and for CS 1000E 1–20 • dsl# (DSL location) = 0-7 Assign 31 DSLs for each MISP if DCH or BDCH were specified at the PH prompt in "MISP configuration procedures".

Configure a trunk clock reference source

In the case where an ISDN BRI trunk is providing a reference clock source to the system clock controller, the Digital Data Block (overlay 73) must be modified as follows.

Note:

Clock signaling is only supported on DSL1 and DSL2.

Table 120: LD 73 - Configure a trunk clock reference source, for 1.5 Mb PRI/DTI.

Prompt	Response	Description
REQ	NEW	New settings.
TYPE	DDB	Digital data block.
PREF	l s c	Location, where: • l (loop) = the primary clock source from a PRI/DTI loop pack for CS 1000M single and multi group systems. • s (slot) = number of the PRI/DTI/SILC card for CS 1000E. • c (card) = the primary clock source from a ISDN BRI SILC pack (DSL0) for CS 1000M single and multi group systems.
SREF	l s c	Location, where: • l (loop) = the secondary clock source from a PRI/DTI loop pack for Large Systems. • s (slot) = the secondary clock source from a ISDN BRI SILC pack (DSL1) for Large Systems. • c (card) = the PRI/DTI/SILC card for CS 1000E. PREF SILC may be different than SREF SILC when it is providing the reference clock source.

Note:

Clock signaling is only supported on DSL1 and DSL2.

Table 121: LD 73 - Configure trunk clock reference source, for 2.0 Mb PRI/DTI.

Prompt	Response	Description
REQ	NEW	New settings.
TYPE	DTI2 PRI2	Digital system data block.
FEAT	SYTI	Digital system timers and counter (only one set per system).
PERS	0-(100)-256	Persistence timer for group II problems.
MGCLK	MGCLK sl s c	Superloop, shelf and card number of Clock Controller for CS 1000E Systems.
PREF CK0	l	Loop number of the primary reference clock for Clock Controller 0, from a PRI2/ DTI2 loop pack, for CS 1000M single and multi-group systems.
	l s c	Location of the primary reference clock for Clock Controller 0, from a ISDN BRI SILC pack (DSL0), for CS 1000M single and multi-group systems.
	C	Card slot number of PRI2/DTI2/SILC card, for CS 1000E systems.
PREF CK1	l	Loop number of the primary reference clock for Clock Controller 1, from a PRI2/ DTI2 loop pack for Large Systems.
	l s c	Location of the primary reference clock for Clock Controller 1, from an ISDN BRI SILC pack (DSL0), for Large Systems.
SREF CK0	l	Loop number of the secondary reference clock for Clock Controller 0, from a PRI2/ DTI2 loop pack, for Large Systems.
	l s c	Location of the secondary reference clock for Clock Controller 0, from an ISDN BRI SILC pack (DSL1) for Large Systems.
SREF CK1	l	Loop number of the secondary reference clock for Clock Controller 1, from a PRI2/ DTI2 loop pack.
	l s c	Location of the secondary reference clock for Clock Controller 1, from an ISDN BRI SILC pack (DSL1).
		PREF SILC may be different than SREF SILC when it is providing the reference clock source.

Prompt	Response	Description
	C	PREF SILC may be different than SREF SILC when it is providing the reference clock source.

Chapter 7: Sample configurations

Contents

This section contains information on the following topics:

Introduction

This chapter provides typical examples of how to configure a line application, an external and integrated packet handler, and a trunk access application.

Note:

These examples pertain to Large Systems.

Example 1: Configure a line application

This example shows how to configure a typical ISDN BRI line application to an existing system. The task is to configure a DSL to support circuit-switched integrated voice and data terminals. Figure 28: DSL terminal configuration on page 252 illustrates these terminals and shows that each transmission mode requires a different TSP and that the same TSP can be assigned to multiple Directory Numbers on the same DSL.

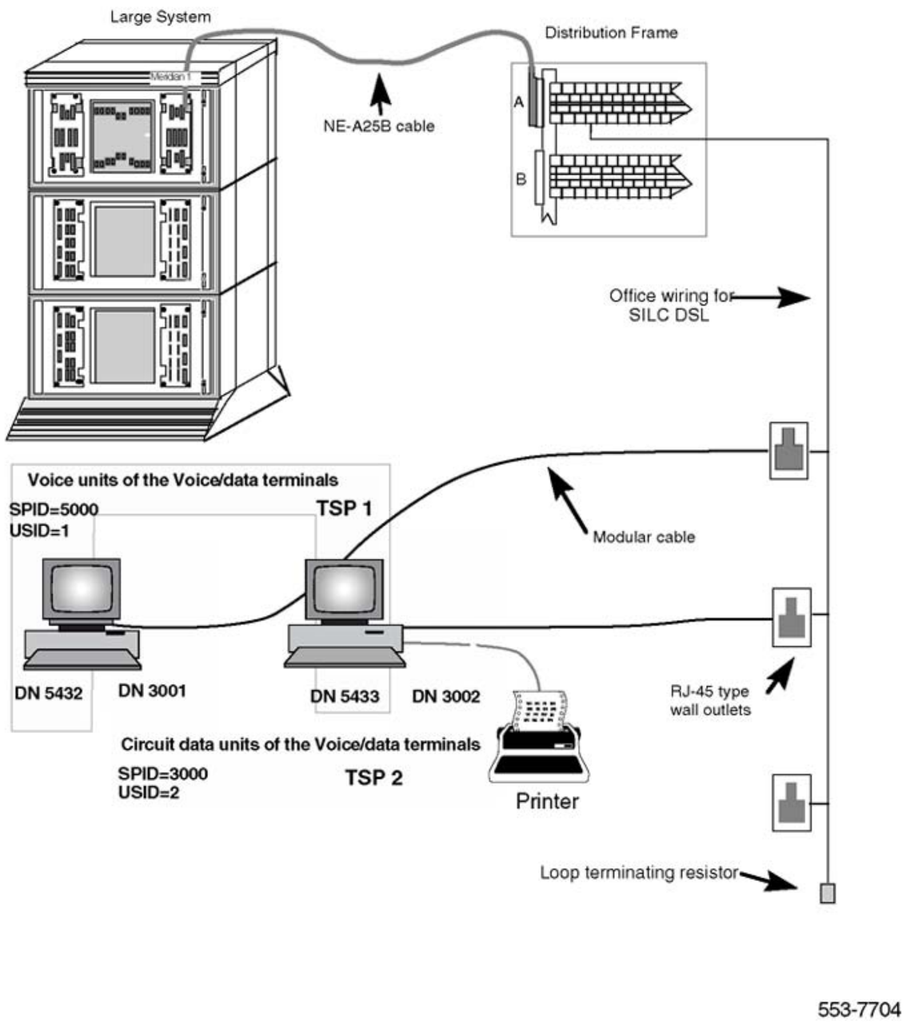


Figure 28: DSL terminal configuration

Configuration procedures

Configure the ISDN BRI line application

To configure ISDN BRI service shown in Figure 28: DSL terminal configuration on page 252, follow the steps below.

1. After logging on to access LD 27, enter LD 27 at the prompt.
2. Configure the LAPD protocol to use the ANSI standard of transmission with specific LAPD transmission characteristics as follows:

Table 122: LD 27 - Configure the LAPD protocol.

Prompt	Response	Description
REQ	NEW	Add an ISDN BRI component
TYPE	LAPD	To administer the LAPD protocol group.
PGPN	1 <cr>	Protocol group number
LAPD	YES	LAPD parameters
T200	4	Maximum retransmission timer is 2 seconds (in units of 0.5 second)
T203	60	Maximum time between frames is 60 seconds (in units of 0.5 second)
N200	6	Maximum number of retransmissions
N201	200	Maximum number of information octets
K	10	Maximum number of outstanding NAKs
		Displays number of DSLs defined.
PGPN	<cr>	
REQ	...	

3. Configure the MISP located on network loop number 8 as follows:

Table 123: LD 27 - Configure the MISP.

Prompt	Response	Description
REQ	NEW	Add an ISDN BRI component.
TYPE	MISP	MISP
LOOP	8	Loop number
APPL	BRIL <cr>	ISDN BRI line application
DPSD	NO	None of the DSL line cards directly connected to the MISP have D-channel Packet Switched Data
REQ		

4. Add a BRSC linked to superloop 24, located in IPE Module 0, in IPE card slot 15. Specify that the BRSC is connected to the MISP at loop 8.

Table 124: LD 27 - Add a BRSC.

Prompt	Response	Description
REQ	NEW	New data
TYPE	BRSC	Add a BRSC

Prompt	Response	Description
BRSC	24 0 15	TN of the BRSC being added
MISP	8	MISP loop number The MISP where layer 3 packets are sent
DPSD	NO	There is no D-channel Packet Switched Data
REQ		

5. Configure a SILC linked to superloop 24, located in IPE Module 0, in IPE card slot 0 as follows:

Table 125: LD 27 - Configure an SILC.

Prompt	Response	Description
REQ	NEW	Add an ISDN BRI component.
TYPE	CARD	ISDN BRI line card
TN	24 0 0	Card location
		Because there is a BRSC configured in the IPE Module, the MISP prompt is skipped and the MISP lll and the BRSC lll s cc are displayed: MISP 8 BRSC 24 0 15
CTYP	SILC	Card type
REQ		

6. Configure SILC port 1 to support the terminals in Figure 28: DSL terminal configuration on page 252. Specify DSL parameters for an extended passive loop that requires adaptive sampling.

Table 126: LD 27 - Configure the DSL for SILC port 1.

Prompt	Response	Description
REQ	NEW	Add an ISDN BRI component.
TYPE	DSL	DSL
DSL	24 0 0 1	DSL location
DES	BLDG1	DSL is in Building 1
CUST	1	Customer number, as defined in LD 15
OPT	<cr>	ISDN BRI line application
MODE	NTAS	Network terminal mode Adaptive sampling (extended passive loop)
B1CT	VCE DTA	B-channel 1 call type Supports integrated voice and data terminal

Prompt	Response	Description
B2CT	VCE DTA	B-channel 2 call type Supports integrated voice and data terminal
LDN	NO	Listed directory number
XLST	0	Pretranslation group
MTEI	6	Maximum number of Terminal Endpoint Identifiers, both static and dynamic
LTEI	<cr>	LTEI is skipped because there is no D-channel packet data
MCAL	16	Maximum number of calls
MTSP	8	Maximum number of TSPs
PGPN	1	Protocol group number
PRID	6	ISDN BRI Conference feature
FDN	<cr>	Flexible Call Forward No Answer (CFNA) directory number denied
EFD	<cr>	Flexible external call CFNA DN denied
HUNT	<cr>	Hunt directory number denied
EHT	<cr>	Hunt external call directory number denied
TGAR	0	Trunk group access restriction
NCOS	0	Network Class Of Service
CLS	UNR ICDD	Class of Service Unrestricted, Internal CDR denied

7. Configure the TSPs for circuit switched voice and circuit switched data for the terminals shown in Figure 28: DSL terminal configuration on page 252 as follows:

Table 127: LD 27 - Configure the TSPs.

Prompt	Response	Description
REQ	NEW	Add a TSP
TYPE	TSP	TSP
DSL	24 0 0 1	DSL location
USID	1	User service identifier
MPHC	NO	The TSP is not used with a Meridian 1 Packet Handler.
SPID	5000 <cr>	Service profile ID
FEATID	A06 15 <cr>	A06 = a 6-party ISDN BRI Conference

Prompt	Response	Description
		15 is the Feature Activation ID, the number associated with ISDN BRI Conference on this ISDN BRI terminal. 15 is assumed for the Feature Indication ID, the number associated with ISDN BRI Conference on this system. If no entry is made for it, the Feature Indication ID number uses the same number as entered for the Feature Activation ID. FEATID is prompted when PRID = 6 in the DSL. Recommended terminal assignments are: M5317TDX = 15 M5209TDcp = 9
DN	5432 1	Directory number to be associated with the TSP, and the CLID entry.
- CT	VCE	Directory number call type is circuit switched voice
MCAL	8	Maximum number of calls on TSP
CLIP	YES	Displays calling party DN on incoming calls
PRES	NO	Does not display DN to the called party on outgoing calls
FEAT	HTA FNA CFTA SFD MWA	Class of Service Hunt allowed Call Forward No Answer allowed Call Forward by Call Type allowed Second level Call Forward No Answer denied Message waiting allowed
DN	5433 2	Directory number associated with the TSP, and the CLID entry.
- CT	VCE	Directory number call type is circuit switched voice
MCAL	4	Maximum number of calls
CLIP	YES	Calling party number displayed on the called party terminal
PRES	YES	Displays DN to the called party on outgoing calls
FEAT	HTA FNA CFTA SFD MWA	Class of Service Hunt allowed Call Forward No Answer allowed Call Forward by Call Type allowed Second level Call Forward No Answer denied Message waiting allowed
DN	<cr>	
DFDN	5432	Default directory number
REQ	NEW	Add a TSP
TYPE	TSP	TSP
DSL	24 0 0 1	DSL location
USID	2	User service identifier

Prompt	Response	Description
MPHC	NO	The TSP is not used with a Meridian 1 Packet Handler.
SPID	3000 <cr>	Service profile ID
FEATID	A06 15 <cr>	A06 = a 6-party ISDN BRI Conference 15 is the Feature Activation ID, the number associated with ISDN BRI Conference on this ISDN BRI terminal. 15 is assumed for the Feature Indication ID, the number associated with ISDN BRI Conference on this system. If no entry is made for it, the Feature Indication ID number uses the same number as entered for the Feature Activation ID. FEATID is prompted when PRID = 6 in the DSL. Recommended terminal assignments are: M5317TDX = 15 M5209TDcp = 9
DN	3001 3	Directory number to be associated with the TSP, and the CLID entry.
- CT	DTA	Directory number call type is data
MCAL	1	Maximum number of calls
CLIP	YES	Displays calling party DN on incoming calls
PRES	YES	Displays DN to the called party on outgoing calls
FEAT	HTD FND CFTD SFD MWD	Hunt Denied Call Forward No Answer Denied Call Forward By Call Type Denied Second level Call Forward No Answer Denied Message Waiting Denied
DN	3002 4	Directory number to be associated with the TSP, and the CLID entry.
- CT	DTA	Directory number call type is data
MCAL	8	Maximum number of calls
CLIP	YES	Displays calling party DN on incoming calls
PRES	NO	Does not display DN to the called party on outgoing calls
FEAT	HTD FND CFTD SFD MWD	Hunt Denied Call Forward No Answer Denied Call Forward By Call Type Denied Second level Call Forward No Answer Denied Message Waiting Denied
DN	<cr>	
DFDN	3001	Default directory number
REQ	NEW	Add a TSP

Prompt	Response	Description
TYPE	TSP	TSP
DSL	24 0 0 1	DSL location
USID	0	User service identifier
MPHC	NO	The TSP is not used with a Meridian 1 Packet Handler.
SPID	5000 <cr>	Service profile ID
FEATID	A06 15 <cr>	A06 = a 6-party ISDN BRI Conference 15 is the Feature Activation ID, the number associated with ISDN BRI Conference on this ISDN BRI terminal. 15 is assumed for the Feature Indication ID, the number associated with ISDN BRI Conference on this system. If no entry is made for it, the Feature Indication ID number uses the same number as entered for the Feature Activation ID. FEATID is prompted when PRID = 6 in the DSL. Recommended terminal assignments are: M5317TDX = 15 M5209TDcp = 9
DN	2000	Directory number to be associated with TSP
- CT	VCE	Directory number call type is circuit switched voice
MCAL	1	Maximum number of calls
CLIP	NO	Does not display calling party DN on incoming calls
PRES	NO	Does not display DN to the called party on outgoing calls
FEAT	HTD FND CFTD SFD	Hunt Denied Call Forward No Answer Denied Call Forward By Call Type Denied Second level Call Forward No Answer Denied
DN	4000	Directory number associated with TSP
- CT	DTA	Directory number call type is data
MCAL	1	Maximum number of calls
CLIP	NO	Displays calling party DN on incoming calls
PRES	NO	Does not display DN to the called party on outgoing calls
FEAT	HTD FND CFTD SFD	Class of Service Hunt denied Call Forward No Answer denied Call Forward by Call Type denied Second level Call Forward No Answer denied
DFDN	2000	Default directory number
DN	<cr>	

Prompt	Response	Description
REQ	END	Terminates the program and saves the configuration

8. Enable the MISP using ENLL 8 in LD 32.
9. Enable the BRSC using ENLC 24 0 15 in LD 32.
10. Enable all the terminals on DSL 24 0 0 1 using **`ENLU 24 0 0 1`** in LD 32.
11. Follow the instructions in the User manual for the specific terminal and enter the appropriate service profile IDs (SPIDs) for the voice and data circuits of the integrated voice and data terminals as follows:

    For the voice circuit of the two terminals, enter at each terminal the SPID number **`5000`** at the SPID prompt to define TSP1

    For the circuit data of the two terminals, enter each terminal SPID number **`3000`** at the SPID prompt to define TSP2.

Example 2: Add an MPH

Configuration example 2 adds an ISDN BRI MPH application to a system with an existing ISDN BRI configuration. The task is to configure a DSL to support D-channel and B-channel packet data terminals using the MISP as an MPH. The MPH has a Meridian Communication Unit (MCU) connection to the Packet Switched Data Network (PSDN).

Figure 29: DSL terminal configuration with D-channel and B-channel packet data on page 259 illustrates these terminals.

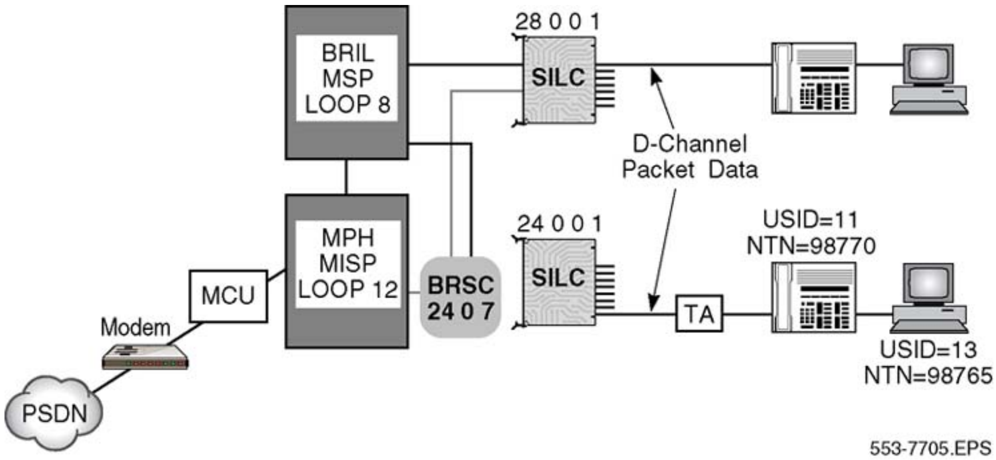


Figure 29: DSL terminal configuration with D-channel and B-channel packet data

Configuration procedures

Configure the ISDN BRI service

To configure ISDN BRI service shown in Figure 29: DSL terminal configuration with D-channel and B-channel packet data on page 259, follow the steps below. Use LD 27 unless noted otherwise.

1. Configure the LAPD protocol to set up layer 2 as follows:

Table 128: LD 27 - Configure the LAPD protocol.

Prompt	Response	Description
REQ	NEW	Add an ISDN BRI component.
TYPE	LAPD	To administer protocol group
PGPN	1	Protocol group number
LAPD	YES	
- T200	4	Maximum retransmission timer is 2 seconds (in units of 0.5 second)
- T203	60	Maximum time between frames is 60 seconds (in units of 0.5 second)
- N200	6	Maximum number of retransmissions
- N201	200	Maximum number of information octets
- K	10	Maximum number of outstanding NAKs
REQ	...	

2. Configure the LAPB protocol to set up Layer 2 parameters as follows:

Table 129: LD 27 - Configure the LAPB protocol.

Prompt	Response	Description
REQ	NEW	Add an ISDN BRI component.
TYPE	LAPB	To administer the LAPB protocol group
PGPN	2	LAPB protocol group number
LAPB	NO	Because LAPB = NO and REQ = NEW, the remaining prompts for the LAPB parameters are not given and defaults are assigned. T1 = 3 seconds, T2 = 2 seconds, and T3 = 6 seconds.

3. Configure the X.25 protocol to set up Layer 3 parameters as follows:

Table 130: LD 27 - Configure the X.25 protocol.

Prompt	Response	Description
REQ	NEW	Add an ISDN BRI component.
TYPE	X25P	To administer the X.25 protocol group
PGPN	3	X.25 protocol set group number
X25P	NO	Because X.25 = NO and REQ = NEW, the remaining prompts for theX.25 parameters are not given and defaults are assigned. T10/T20, T11/21, T12/T22, and T13/23 all equal 180 seconds.

4. Configure the DNA table with a range of 20 numbers, 98765 through 98784 as follows:

Table 131: LD 27 - Configure the DNA tables.

Prompt	Response	Description
REQ	NEW	Add DNA tables
TYPE	DNAT	To administer the DNA tables
DNAT	4	DNA table number 4 is associated with MPH network interface.
DNIC	4321	Data Network Identification Code (DNIC) for the table
NTN	98765 20	Network Terminal Number (NTN) for the selected table and the range of numbers.
REQ	...	

5. Configure the TIE trunk route for packet data in LD 16.

Table 132: LD 16 - Configure the TIE trunk route for packet data (LD 16).

Prompt	Response	Description
REQ	NEW	Add ISDN BRI protocol group settings.
TYPE	RDB	Route data block
CUST	xx	Customer number, as defined in LD 15
ROUT	1	Route number
TKTP	TIE	Trunk route type
...	...	
DTRK	YES	Digital trunk route
BRIP	YES	Packet handler route

Prompt	Response	Description
		Note: Prompted only if DTRK = YES.
ACOD	xxxxxx	Trunk route access code
TARG	<cr>	Access restriction group number
CNTL	<cr>	Changes to control timers
...	...	...

6. Configure the TIE trunk for packet data in LD 14.

Table 133: LD 14 - Configure the TIE trunk for packet data.

Prompt	Response	Description
REQ	NEW	Enter new trunk data
TYPE	TIE	Trunk type
TN	24 00	Loop, channel number
CUST	xx	Customer number, as defined in LD 15
NCOS	<cr>	Network Class Of Service group
RTMB	1	Route and route member
MNDN	<cr>	Manual directory number
TGAR	<cr>	Trunk group access restriction
CLS	<cr>	Unrestricted Class of Service
...	...	...

7. Use LD 11 to configure the Meridian Communication Unit (MCU).

Table 134: LD 11 - Add an MCU network interface.

Prompt	Response	Description
REQ:	NEW	Add a data set.
TYPE:	MCU	To administer the MCU
TN	24 00 16 30	Terminal Number
CDEN	4d	Quadruple density (not prompted for superloops)
DES	a	Designator
CUST	xx	Customer number, as defined in LD 15
MPHI	YES	MCU is used as an MPH network interface.
OPE	<cr>	No change to data port parameters

Prompt	Response	Description
		If OPE = NO, LD11 is completed.
...	...	...

8. Configure MISP loop number 12 for an MPH as follows:

Table 135: LD 27 - Configure the MISP for an MPH.

Prompt	Response	Description
REQ	NEW	Add a MISP.
TYPE	MISP	To administer the MISP card
LOOP	12	MISP loop number
APPL	MPH cr>	MPH application
PRFX	1	Prefix to be used by the DNA tables of the MPH
NTNO	YES	PSDN present NTN only
DNIC	4321	The Data Network ID Code for the DNA used with the MPH
NWIF	2	The MPH network interface identifier for configuration
TN	24 00 16 30	TN of the MCU on which the dedicated connection from the MPH is terminated
- LAPB	2	The LAPB protocol set group number to be used on the MPH network interface
- X25P	3	The X.25 protocol set group number to be used on the MPH network interface
- PVC	1 1	The range of Permanent Virtual Circuit Logical Channel Numbers
- IC	300 400	The range of Incoming Logical Channel Numbers
- TC	500 700	The range of Two-way Logical Channel Numbers
- OC	701 900	The range of Outgoing Logical Channel Numbers
- DNAT	4	The DNA table associated with the MPH network interface

9. Configure MISP on loop 8 with the MPH on loop 12 to add the ISDN BRI line application:

Table 136: LD 27 - Configure the MISP for an ISDN BRI line application.

Prompt	Response	Description
REQ	NEW	Add a MISP.

Prompt	Response	Description
TYPE	MISP	To administer the MISP card
LOOP	8	MISP loop number
APPL	BRIL	BRIL = ISDN BRI line application
DPSD	YES	There is D-channel Packet Switched Data.
MPHC	YES	DPSD are routed to the MPH card
MPH	12	The MPH loop on which the dedicated connection from the MISP is terminated

10. Add a BRSC at superloop 24, shelf number 0 and card number 7.

Table 137: LD 27 - Configure the BRSC.

Prompt	Response	Description
REQ	NEW	Add a BRSC.
TYPE	BRSC	To administer a BRSC
BRSC	24 0 7	BRSC superloop number, shelf number, card number.
MISP	8	BRSC is associated with MISP loop number 8
DPSD	YES	There is D-channel Packet Switched Data.
MPHC	YES	DPSD routed to MPH
MPH	12	MISP loop number 12 is the MISP with MPH application where D-channel packet data are being sent

11. Configure a SILC linked to superloop 24, located in IPE Module 0, in IPE card slot 0 as follows:

Table 138: LD 27 - Configure the SILC.

Prompt	Response	Description
REQ	NEW	Add an SILC.
TYPE	CARD	ISDN BRI line card
TN	24 0 0	Card location
		Because there is a BRSC configured in the IPE Module, the MISP prompt is skipped and the MISP lll and the BRSC lll s cc is displayed: MISP 8 BRSC 24 0 7
CTYP	SILC	Card type
REQ	...	...

12. Configure a SILC linked to superloop 28, located in IPE Module 0, in IPE card slot 0 as follows:

Table 139: LD 27 - Configure the second SILC.

Prompt	Response	Description
REQ	NEW	Add an SILC.
TYPE	CARD	ISDN BRI line card
TN	28 0 0	Card location
		Because there is a BRSC configured in the IPE Module, the MISP prompt is skipped and the MISP lll and the BRSC lll s cc is displayed: MISP 8 BRSC 24 0 7
CTYP	SILC	Card type
REQ	...	

13. Configure DSL parameters as follows to support the terminals in Figure 29: DSL terminal configuration with D-channel and B-channel packet data on page 259:

Table 140: LD 27 - Configure the DSL.

Prompt	Response	Description
REQ	NEW	Add a DSL.
TYPE	DSL	To administer the digital subscriber loop
DSL	24 0 0 1	
CTYP	SILC	Card type
OPT	(BRIL)	ISDN BRI line application
MISP	8	DSL is associated with MISP loop number 8
MODE	NTAS	NT Mode Adaptive Sampling
BICT	VCE DTA	B-channel 1 call type has circuit switched voice and circuit switched data
B2CT	IPD	B-channel 2 call type has packet data using an MPH
MPH	12	MPH loop number
LDN	NO	Not associated with listed directory number
XLST	0	Pretranslation group
MTEI	6	Maximum number of TEIs (static and dynamic combined) allowed
LTEI	<cr>	LTID and static TEI pair for D-channel packet data
MCAL	8	Maximum number of calls allowed per DSL

Prompt	Response	Description
MTSP	8	Maximum number of TSP allowed
PGPN	1	Protocol group number
PRID	6	Protocol ID for ISDN BRI Conference

14. Configure a second set of DSL parameters as follows to support the terminals in Figure 29: DSL terminal configuration with D-channel and B-channel packet data on page 259:

Table 141: LD 27 - Configure the DSL.

Prompt	Response	Description
REQ	NEW	Add a DSL.
TYPE	DSL	To administer the digital subscriber loop
DSL	28 0 0 1	
CTYP	SILC	Card type
OPT	BRIL	ISDN BRI line application
MISP	8	DSL is associated with MISP loop number 8
MODE	NTAS	NT Mode Adaptive Sampling
BICT	VCE DTA	B-channel 1 call type has circuit switched voice and circuit switched data
B2CT	VCE DTA	B-channel 2 call type has packet data using an MPH
MPH	12	MPH loop number
LDN	NO	Not associated with listed directory number
XLST	0	Pretranslation group
MTEI	6	Maximum number of TEIs (static and dynamic combined) allowed
LTEI	<cr>	LTID and static TEI pair for D-channel packet data
MCAL	8	Maximum number of calls allowed per DSL
MTSP	8	Maximum number of TSP allowed
PGPN	1	Protocol group number
PRID	6	Protocol ID for ISDN BRI Conference

15. Configure the TSP for USID 11 for the D-channel as follows:

Table 142: LD 27 - Configure the TSP for USID 11.

Prompt	Response	Description
REQ	NEW	New data

Prompt	Response	Description
TYPE	TSP	To administer the Terminal Service Profiles
DSL	28 0 0 1	DSL location
USID	11	D-channel terminal
CUST	1	Customer number, as defined in LD 15
MPHC	YES	Yes, the TSP is used with an MPH
TRMT	D	D-channel type
TEI	15	Static TEI for addressing terminal
X25P	3	X.25 protocol set group number used on the MPH user interface
NTN	98770	Network Terminal Number of the TSP is 98770
- PVC	1 1	The lowest Permanent Virtual Circuit Logical Channel Number is 1
- IC	<cr>	There is no range set for Incoming Logical Channel Number
- TC	30 60	The lowest Two-way Logical Channel Number is 30. The highest Two-way Logical Channel Number is 60.
- OC	<cr>	There is no range set for Outgoing Logical Channel Number
CDR	NO	No Call Detail Recording for packet data calls

16. Configure the TSP for USID 13 for the B-channel.

Table 143: LD 27 - Configure the TSP for USID 13.

Prompt	Response	Description
REQ	NEW	New data
TYPE	TSP	To administer the Terminal Service Profiles
DSL	24 0 0 1	DSL location
USID	13	B-channel terminal
CUST	1	Customer number, as defined in LD 15
MPHC	YES	Yes, the TSP is used with an MPH
- TRMT	B	Terminal type is set for B-channel configuration
- BCH	2	TSP is associated with the B2 channel of the DSL
LAPB	2	The LAPB protocol set group number to be used on the MPH network interface is 2.

Prompt	Response	Description
NTN	98765	Network Terminal Number of the TSP is 98765
- PVC	1 1	The lowest Permanent Virtual Circuit Logical Channel Number is 1
- IC	<cr>	There is no range set for Incoming Logical Channel Number
- TC	30 60	The lowest Two-way Logical Channel Number is 30. The highest Two-way Logical Channel Number is 60.
- OC	<cr>	There is no range set for Outgoing Logical Channel Number

17. Configure the PVC connection as follows:

Table 144: LD 27 - Configure the PVCs.

Prompt	Response	Description
REQ	NEW	Add a Permanent Virtual Circuit (PVC).
TYPE	PVC	To administer the PVC connection
MPH	12	MPH loop number
PVCN	1	PVC connection number
XPVC	NO	No external PVC connection
NTN1	98765	First Network Terminal Number of PVC internal connection
- LCN1	1	PVC Logical l Number associated with Network Terminal Number 1
NTN2	98770	Second Network Terminal Number of PVC internal connection
- LCN2	1	PVC Logical Channel Number associated with Network Terminal Number

18. Enable the MISP on network loop number in the Network and IPE Diagnostic Program, LD 32, using **`ENLL 8`**.
19. Enable the terminals connected to DSL 24 0 0 1 and DSL 28 0 0 1 in the Network and IPE Diagnostic Program, LD 32, using **`ENLU 24 0 0 1`** and **`ENLU 28 0 0 1`**.

    Follow the instructions in the User manual for the specific terminal and enter the appropriate service profile IDs (SPIDs) for the voice and data circuits of the integrated voice and data terminals.

Example 3: Configure an ISDN BRI TIE trunk

In the configuration, depicted in Figure 30: ISDN BRI TIE trunk configuration on page 269, a Meridian Customer Defined Networking (MCDN) TIE trunk connection can be implemented by connecting two systems to the ISDN BRI leased line through the Local Exchange via two SILC cards. The S/T interface is connected to the Local Exchange using the NT1 supplied by the PTT. There is no distance limitation on this configuration. System clock synchronization may be achieved by having the system slave to the Local Exchange; the clock source can be derived from the ISDN BRI Local Exchange connection.

Note:

The configuration in the example is being done on the user side.

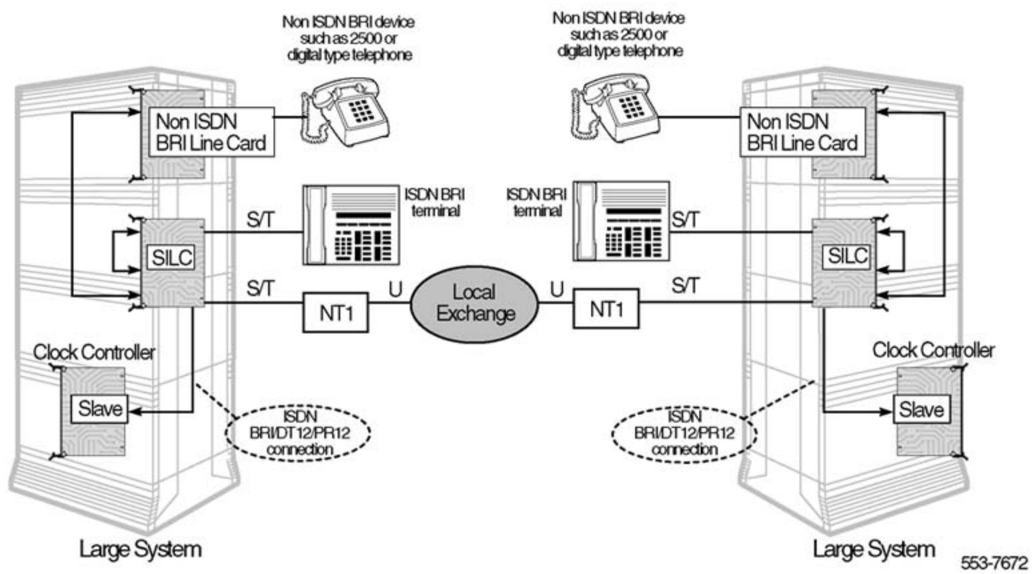


Figure 30: ISDN BRI TIE trunk configuration

Configuration procedures

To configure the ISDN BRI TIE trunk configuration depicted in Figure 30: ISDN BRI TIE trunk configuration on page 269, follow these procedures:

1. Define an ISDN BRI trunk access customer using the Customer Data Block (LD 15).

Table 145: LD 15 - Define a customer for a trunk.

Prompt	Response	Description
REQ:	NEW	Define a new customer.
TYPE:	NET	Networking data.
CUST	0	Customer number 0.
...	...	...
ISDN	YES	The customer is equipped with ISDN.
...	...	...

2. Configure the LAPD protocol to use the ANSI standard of transmission with specific LAPD transmission characteristics as follows:

Table 146: LD 27- Configure the LAPD protocol.

Prompt	Response	Description
REQ	NEW	To add an ISDN BRI component
TYPE	LAPD	To administer the LAPD protocol group.
PGPN	1 <cr>	Protocol group number.
LAPD	YES	To define LAPD parameters which follow.
- T200	4	Maximum retransmission timer is 2 seconds (in units of 0.5 second)
- T203	60	Maximum time between frames is 30 seconds (in units of 0.5 second)
- N200	6	Maximum number of retransmissions
- N201	200	Maximum number of information octets
- K	10	Maximum number of outstanding NAKs
		Displays number of DSLs defined.
N2X4	0-(10)-20	For 1TR6 connectivity - number of status inquiries when the remote station is in peer busy state. <cr> has been entered.
PGPN	<cr>	
REQ		

3. Configure the route data block parameters for the ISDN BRI TIE trunk.

Table 147: LD 16 - Configure ISDN BRI trunk route parameters.

Prompt	Response	Description
REQ	NEW	Add ISDN BRI protocol group settings.

Prompt	Response	Description
TYPE	RDB	Route data block.
CUST	0	Customer number, as defined in LD 15.
DMOD	<cr>	Default model number for this route. <cr> has been entered.
ROUT	99	Route number 99 has been used.
TKTP	TIE	Trunk route type.
RCLS	<cr>	Class marked route.
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.
- NASA	(NO)	No Network Attendant Service Interface is required.
- MBGA	(NO)	No Multi Business Group interface on the D Channel is required.
- IFC	(SL1)	DCH interface type.
- - CNTY	<cr>	Country pertaining to EuroISDN interface type.
...		
- CDR	(NO)	No CDR on the trunk route is required.
- OTL	(NO)	No CDR on outgoing toll calls is required.
...		
- OAN	(NO)	No CDR on answered outgoing toll calls is required.
- MR	(NO)	Advice of Charge for EuroISDN, AXE-Australia or Japan D70. The route does not support AOC.
- - RUCS	0	Route unit cost.
- - RURC	0	Route unit reference cost.
- - RUCF	0 - (1) - 9999 (0) - 3	Route unit conversion factor. This prompt does not appear for Denmark or Sweden.
- SIDE	USR	Meridian 1, CS Avaya Communication Server 1000M (Avaya CS 1000M), or node type (either network or user), prompted only if IFC = SL1.
- PGPN	1	Protocol Group (as defined previously in step 2. using LD 27).

Prompt	Response	Description
- RCAP	BRI <cr>	Remote D-channel capabilities. BRI allows ISDN line/trunk interworking). This prompt is repeated until <cr> is entered.
...	...	...
- INAC	(NO)	Do not insert NARS/BARS access code on incoming calls.
- DSEL	(vod)	Data selection. (VOD) = voice and data.
...	...	...
- OVLR	(NO)	Do not allow Overlap Receiving.
- OVLS	(NO)	Do not allow Overlap Sending.
...	....	....

4. Configure the MISP on loop 18.

Table 148: LD 27 - Add an MISP for the trunk.

Prompt	Response	Description
REQ	NEW	Add or a MISP
TYPE	MISP	MISP
LOOP	18	MISP lop number.
APPL	BRIT <cr>	Application type for the MISP. BRIT = ISDN BRI trunking. APPL is prompted until <cr> is entered.
DSPD	(NO)	(NO) = No D-channel Packet Switched Data.
...	...	

5. Enable the MISP using the **`ENLL l`** command in LD 32.
6. Configure the DSL on superloop 24, shelf module 1, card slot 4, connected to SILC port 0.

Table 149: LD 27 - Add a DSL for the trunk.

Prompt	Response	Description
REQ	NEW	Add a DSL.
TYPE	DSL	DSL
DSL	24 1 4 0	DSL location superloop 24 shelf 1 card slot 4 DSL location 0 (connected to SILC port 0)

Prompt	Response	Description
APPL	BRIT	Application type for the DSL. BRIT = ISDN BRI trunking.
CUST	0	Customer number, as defined in LD 15
CTYP	SILC	Connected to a SILC line card.
MISP	18	MISP loop number (as configured in step 4, using LD 27).
MODE	TE	The mode for the trunk DSL. TE is entered for Terminal Equipment.
MTFM	(NO)	Enter NO to disable the multi-frame option.
TKTP	TIE	Trunk type of TIE.
CLOK	(NO)	This trunk DSL is not provisioned for clock source.
PDCA	(1)	Use the default pad table number to be associated with this DSL.
ROUT	99	Route number for the trunk DSL.
B1	YES	Configure B Channel 1. If REQ = NEW, a response to this prompt is not required, because B1 parameters are mandatory. The system will automatically display the prompts that follow.
- MEMB	1	Route member number to be associated with B-channel 1.
- TGAR	(0)	Trunk Group Access Restriction.
- NCOS	(0)	Network Class Of Service Group Number
- CLS	<cr>	Class of Service options. The following options are selected as defaults: APN = ACD Priority not allowed UNR = Unrestricted (default).
B2	(NO)	Configure B Channel 2. All parameters entered for B1 will be applied to B2, except the route member number will be an unused value. The following message will be displayed. B2 will use Route # 99 Member # 2
REQ	...	

7. You can use the **`STAT l s c dsl#`** command in LD 32 to query the status of the DSL.

Chapter 8: Setting up ISDN BRI traffic reports

Contents

This section contains information on the following topics:

Introduction

The System generates and stores traffic statistics for ISDN BRI. These statistics can be displayed on the administration terminal or printed on the administration printer. To set up and print traffic reports use LD 02.

Refer to *Avaya Software Input Output Reference - Administration, NN43001-611* LD 02 for complete details concerning Traffic reporting prompts. Refer to *Avaya Traffic Measurement Formats and Output Reference, NN43001-750* for complete discussions of traffic and the reports generated.

The following ISDN BRI traffic reports can be generated.

- Network traffic report (001) shows ISDN BRI and non ISDN BRI traffic on the network loops;
- MISP and/or BRSC traffic report (011) shows ISDN BRI DSL traffic on the MISP/BRSC loops and BRSC cards;
- MISP and/or BRSC D-channel management messages report (012) shows the management messages handled by each D-channel on the MISP loops and BRSC cards;
- MISP and/or BRSC messages report (013) shows all the messages handled by the MISP loops and BRSC cards;
- Trunk DSL traffic system traffic report (014) shows system traffic on ISDN BRI trunks.
- MPH traffic report (015) shows the ISDN BRI data traffic over MCU, PRI, B-channel, DSL and BRSC links.

Select the report types

Generate system traffic reports by using the following command:

`SOPS (options)--(OPTIONS)`

The input and output parameters for these commands follow.

OPTIONS refers to the reports to be generated. This must be one or more of the following numbers.

001 Network traffic report

011 ISDN BRI MISP and/or BRSC traffic report

012 ISDN BRI MISP D-channel management messages report

013 ISDN BRI MISP messages report

014 ISDN BRI trunk DSL system traffic report

015 ISDN BRI MPH traffic report

The following commands can also be used to find out or clear the current report options that have been selected.

`TOPS (options)` Displays or prints the current report options selected for the system

`COPS (options) -- (OPTIONS)` Clears the report options selected for the system

Scheduled reporting

The reports selected using the SOPS command can be printed or displayed at a specified time, according to a schedule. The data is erased after it is displayed or printed.

To set up a schedule for the whole system, use the following command.

SSHS (sd) (sm) (ed) (em)--(SD) (SM) (ED) (EM) (sh) (eh) (so)--(SH) (EH) (SO) (d)--(D)

The input and output parameters for these commands follow.

sd Start day of the reporting period. This must be a 1-digit or 2-digit number from 1 to 31 signifying the 1st to the 31st day of the month.

sm Start month of the reporting period. This must be a 1-digit or 2-digit number from 1 to 12 signifying the 1st to the 12th month of the year.

ed End day of the reporting period. This must be a 1-digit or 2-digit number from 1 to 31 signifying the 1st to the 31st day of the month.

em End month of the reporting period. This must be a 1-digit or 2-digit number from 1 to 12 signifying the 1st to the 12th month of the year.

sh Start hour of the schedule. This must be a 1-digit or 2-digit number from 1 to 24 signifying the 1st to the 24th hour of the day.

eh End hour of the schedule. This must be a 1-digit or 2-digit number from 1 to 24 signifying the 1st to the 24th hour of the day.

so Frequency of the schedule. This must be one of the numbers shown in the following table.

Table 150: Option numbers for frequency of schedule

Number	Meaning
0	No traffic displayed or printed
1	Hourly on the hour
2	Hourly on the half-hour
3	Half-hourly, on the hour and the half-hour

d Day of the week. This must be one or more of the numbers shown in the following table.

Table 151: Option numbers for days of the week

Number	Meaning
1	Sunday

Number	Meaning
2	Monday
3	Tuesday
4	Wednesday
5	Thursday
6	Friday
7	Saturday

For example, to have the specified reports displayed or printed for the period starting on January 1, 1993 and ending on December 31, 1993 every Monday through Friday every hour on the hour, instead of for the period starting January 1, 1993 and ending on March 31, 1993, Saturday and Sunday, half-hourly, enter the information shown in bold type in the example below. The program displays the current parameter values of the command (shown in unbolded type in the example below).

SSHS 1 1 31 3--**1 1 31 12** 8 8 3--**1 24 1** 1 7--**2 3 4 5 6**

The following command can also be used to display the current schedule.

`TSHS (sd) (sm) (ed) (em) (sh) (eh) (so) (d)`

Note:

MISP/BRSC reports contain data collected in the previous period. For example, if the system traffic reports are scheduled to be printed every hour and on the hour, the data displayed at 3:00 will correspond to those collected from 1:00 to 2:00. This is also true for immediate reports. That is, if the user issues the command at 2:05, the data displayed will correspond to those collected from 1:00 to 2:00.

Immediate reporting

The reports selected with the SOPS command can be printed or displayed immediately. The data is not erased after the information is displayed or printed.

To print or display traffic reports immediately for the whole system, use the **INVS** command.

For example, to print or display the traffic reports specified by the SOPS command for the whole system, enter the information shown below.

INVS x x

where x x is the range of reports.

ISDN BRI traffic reports

Network traffic report (TFS001)

The network traffic report (TFS001)shows the traffic activities for the Meridian 1 lines and trunks including ISDN BRI DSLs. It is used to determine system peak traffic requirements and traffic load distribution. This information is used to optimize the available system resources or to add more resources to handle the existing requirements. It contains the following columns of information.

Table 152: Network traffic report column descriptions

Column	Description
Loop number	Subscriber loop number
Loop type	Subscriber loop type, which can be a terminal, conference, or TDS
Intraloop FTM	Number of call attempts that failed to match the channel call type on the subscriber loop (in CCS)
Intraloop CCS	Traffic load on the subscriber loop
Intraloop peg count	Number of call attempts over the subscriber loop (in CCS)
Total loop FTM	Total number of call attempts that failed to match for all terminals on the subscriber loop (in CCS)
Total loop CCS	Total traffic load over the subscriber loop
Total loop peg count	Total number of calls handled by all the terminals on the subscriber loop

The following is an example of the Network traffic report.

Table 153: Example of Network traffic report

Format							
System ID	TFS001						
Loop number	Loop type	Intraloo p FTM	Intraloop CCS	Intraloop peg count	Total loop FTM	Total loop CCS	Total loop peg count
Example							
200	TFS001						

004	TERM	00000	0000142	00161	00001	0002056	01652 S
008	TERM	00000	0000184	00180	00001	0002500	01725 S
012	TDMS	00000	0000000	00000	00013	0000031	01496
013	CONF	00000	0000000	00000	00000	0000010	00006
014	TERM	00000	0000085	00060	00006	0000544	00287
015	TERM	00003	0000064	00039	00014	0000372	00284

Note:

Superloops are identified by an "S" at the end of the line. Superloops exist in multiples of four (4, 8, 12, 16, etc.). For example, if superloop 4 exists, loops 5, 6, and 7 do not.

Note:

Determine the grade of service provided within the listed loop, and determine the total loop traffic by dividing the FTM by the peg count.

MISP/BRSC traffic report (TFS011)

The MISP/BRSC traffic report shows the call processing activities of all DSLs associated with each MISP in the system. It is used to determine the type of ISDN BRI traffic such as voice, data, or packet data.

If a MISP is serving BRSCs, the MISP/BRSC traffic report contains MISP and BRSC information. In TFS011, MISP information in the four Dchannel parameters shows totals collected for the line cards served directly by the MISP. BRSC information shows Dchannel traffic data collected for line cards at the BRSC in the IPE Module. The information collected for each MISP and BRSC in the system is described below.

Table 154: MISP/BRSC traffic report column descriptions

Column	Description
Attempted calls (MISP only)	Number of attempted calls, including all successfully completed and unsuccessfully completed calls.
Completed call (MISP only)	Number of successfully completed calls for the reported period.
Call length	Average length of a call of a successfully completed call (in seconds).
MISP/BRSC messages	Number of signaling messages sent by the MISP and/or the BRSC to the terminals, on the D-channel.
Terminal messages	Number of signaling messages sent by the terminals to the MISP and/or the BRSC, on the D-channel.

Column	Description
MISP/BRSC data packets	Number of data packets sent by the MISP and/or the BRSC to the terminals.
Terminal data packets	Number of D-channel data packets sent by the terminals to the MISP and/or the BRSC.

The following is an example of a MISP/BRSC traffic report, where MISP004 serves as a BRSC at 24 0 15.

Table 155: Example of a MISP/BRSC traffic report

Format						
System ID	TFS011					
MISP and BRSC ID						
Attempted calls	Completed calls	Call length	MISP/ BRSC messages	MISP/ BRSC terminal messages	MISP/ BRSC data packets	MISP/ BRSC for terminal data packets
Example						
0111	TFS011					
MISP002						
00020	00006	00019	00080	00040	00000	00006
MISP004						
00030	00001	00030	00125	00600	00180	00012
BRSC 24 0 15			005110	001020	003600	000200

MISP/BRSC D-channel management messages report (TFS012)

The MISP/BRSC D-channel management messages report contains the traffic management activity for each DSL based on the exchange of signaling messages between the MISP and the terminals over the D-channels. It is used to see if there are any communication problems between the MISP/BRSC and the terminals.

If a MISP is serving BRSCs, the MISP D-channel management messages report contains MISP and BRSC information. In TFS012, MISP information shows totals collected for the line cards served directly by the MISP. BRSC information shows Dchannel traffic data collected for line cards at the BRSC. The following example shows the information collected for each MISP and BRSC in the system.

Table 156: MISP/BRSC D-channel management report column descriptions

Column	Description
MISP/BRSC links	Number (quantity) of MISP/BRSC initiated link initializations
Terminal links	Number (quantity) of terminal initiated link initializations
MISP/BRSC messages	Number (quantity) of management messages sent from the MISP and the BRSC to terminals
Terminal messages	Number (quantity) of management messages sent from terminals to the MISP
Incomplete calls	Number of times the links associated with D-channels were not able to complete calls
Link errors	Number (quantity) of management data link errors

The following is an example of the MISP/BRSC D-channel management messages report where MISP004 serves as a BRSC at 24 0 16.

Table 157: Example of a MISP/BRSC D-channel management messages report

Format					
System ID TFS012					
MISP/BRSC ID					
MISP/BRSC links	Terminal links	MISP/ BRSC messages	Terminal messages	Incomplete calls	Link errors
Example					
0111 TFS012					
MISP001					
00010	00015	00010	00016	00011	00002
BRSC 24 0 16					
00000	00008	00016	00009	00017	00001

MISP/BRSC messages report (TFS013)

The MISP/BRSC messages report shows the total number of call processing, maintenance, and management messages sent through each MISP in the system. The totals are grouped according to the size of the message.

If a MISP is serving BRSCs, the MISP/BRSC messages report contains MISP and BRSC information. In TFS013, MISP information shows totals collected for the line cards served directly by the MISP. BRSC information shows D-channel traffic data collected right at the BRSC. The following information is collected for each MISP and BRSC in the system.

Table 158: MISP messages report column descriptions

Column	Description
1-10 bytes	Total number of messages that are from 1 to 10 bytes long
11-20 bytes	Total number of messages that are from 11 to 20 bytes long
Greater than 20	Total number of messages that are over 20 bytes long

The following is an example of the MISP/BRSC messages report, where MISP004 serves as a BRSC at 24 0 17.

Table 159: Example of MISP/BRSC messages report

Format System ID TFS013 MISP/BRSC ID 1-10 bytes	 11-20 bytes	 Greater than 20
Example		
0111 TFS013		
MISP001		
00060	00000	00000
BRSC 24 0 17		
00012	00004	00000

ISDN BRI trunk DSL system traffic report (TFS014)

The ISDN BRI trunk DSL system traffic report (TFS014), dedicated to ISDN BRI trunk DSLs, provides traffic measurement similar to the one provided by the ISDN PRI system traffic report; please refer to *Avaya Traffic Measurement Formats and Output Reference, NN43001-750*.

The report contains the following information for each MISP in the system.

Table 160: ISDN BRI trunk DSL system traffic report

System ID TFS014 MISP ID Total number of outgoing maintenance messages	 Total number of incoming maintenance messages

Total number of outgoing administration messages	Total number of incoming administration messages
Total number of outgoing protocol messages	
Total number of Layer 3 protocol errors	
Total number of Layer 2 protocol errors	
Total number of Layer 1 errors	
Total number of connected calls	

Meridian 1 Packet Handler traffic report (TFS015)

The Meridian 1 Packet Handler traffic report provides specific information about incoming and outgoing calls and data packets. This report is particularly useful for analyzing the flow of data over network links.

The MPH traffic report contains the following columns:

Table 161: MPH traffic report column descriptions

Column	Description
aa	MPH loop number.
bb	MPH link interface type, up to four characters (MCU, PRI, BCH, BRIL, or BRSC). **Note:** To determine which link interface type is listed in bb, perform a **`STIF`** command in LD 32; this command displays the interface type and the associated timeslot.
cc	The timeslot number of the MPH link interface type, up to five digits.
dd	The number of times the link was initialized, up to five digits.
ee	The number of incoming calls that were attempted, up to five digits.
ff	The number of incoming calls that were completed, up to five digits.
gg	The number of outgoing calls that were attempted, up to five digits.
hh	The number of outgoing calls that were completed, up to five digits.
ii	The average length, in seconds, of data calls, up to five digits.
jj	The number of incoming data packets, up to 10 digits.
kk	The number of outgoing data packets, up to 10 digits.

The following is an example of a Meridian 1 Packet Handler traffic report.

Note:

The example shown is for an MCU interface type; the format would be exactly the same for the other interface types (PRI, BRIL, BCH, or BRSC), with "bb" indicating the interface type.

Table 162: Example of a Meridian 1 Packet Handler traffic report

Format							
System ID TFS015							
MPH aa							
bb	cc	dd	ee	ff	gg	hh	ii
jj			kk				
Example							
0111 TFS015							
MPH002							
MCU	0006	0019	00040	00040	00006	00001	00360
	0000000780		0000000568				

Call Detail Recording for the Meridian 1 Packet Handler

CDR for the MPH has internal and external record types. When either or both originating and TSPs have CDR, the system generates an internal record type G. Internal CDR may be configured on the TSP whether or not the customer has CDR enabled CDR data does not print if the customer with the TSP does not have CDR.

A call that connects to the public data network, including calls between two different MPH applications on the same switch, generates an external record type H. External CDR configuration is based on customer block data. Incoming and/or outgoing packet data calls may generate external CDR records. The PVC and B-channel calls have no CDR because there is no call establishing process involved.

The MPH traffic report has ten columns as described below:

Table 163: CDR for MPH report column descriptions

Column	Description
aa	MPH record type, G for internal or H for external; one character.
bb	The record number field, identifying the current record in the CDR sequence. It is the CDR record number for the customer, and it increments for all CDR record types; three characters, right justified.

Column	Description
cc	The customer number field, identifying the customer associated with the call; two characters.
dd	The originating ID is the originating DNA number for internal or external outgoing calls. For external incoming calls, this value is the TN of the incoming link (MCU = ll ss cc uu, PRI = ll cc; filled to 14 characters, left justified.
ee	The terminating ID is the originating DNA number for internal or external outgoing calls. For external outgoing calls, this value is the TN of the outgoing link (MCU = ll ss cc uu, PRI = ll cc; filled to 14 characters, left justified.
ff	Date in Month/Day format; five characters.
gg	Time in Hour:Minute format; five characters.
hh	Call duration, shown in format Hour:Minute:Second; eight characters.
ii	The number of incoming data packets, up to 10 digits.
jj	The number of outgoing data packets, up to 10 digits.

The following is an example of a CDR for an MPH report.

Table 164: Example of CDR for MPH report

Format							
aa	bb	cc	dd	ee	ff	gg	hh
ii			jj				
Example							
G	123	11	ll cc	ll cc	02/26	08:59	00:20:06
	0000000780		0000000568				

Chapter 9: ISDN BRI features

Contents

This section contains information on the following topics:

Feature description

This section begins with a quick reference table (Table 165: Features on ISDN BRI lines on page 288) of features with a column that indicates how the feature is supported on ISDN BRI. The notes at the end of the table provide some detail about the support of the feature. Individual feature description chapters begin on Attendant Through Dialing Networkwide.

The following legend and information apply to Table 165: Features on ISDN BRI lines on page 288.

Legend

yes = fully supported

n/a = Not Applicable or transparent to ISDN BRI

no = not supported due to ISDN BRI Standards limitations. See ISDN PRI feature interactions with ISDN BRI on page 336.

P = partially supported with description of limitation. See Features partially supported by ISDN BRI on page 317.

Features labeled with "**" have a more detailed description later in this chapter.

Table 165: Features on ISDN BRI lines

Features and services	Supported	Comments
— A—		
Access Restrictions**	yes	See details later in this chapter.
Access to Paging**	yes	See details later in this chapter.
Access to Recorded Telephone Dictation**	yes	See details later in this chapter.
Automatic Call Distribution (ACD)**	P	See Footnote.. [1]
ACD Answering Time in Night Service	no	
ACD/CDR Q Record Option.	no	
ACD Call Priority	no	
ACD Call Waiting Threshold	no	
ACD Calls on Hold	no	
ACD Enhancements	no	
• Call Delays	no	
• Dynamic Queue Threshold	no	
• Threshold Visual Indication	no	
ACD Least Call Queuing	no	
ACD Night Call Forward without Disconnect Supervision	no	
Activity Codes for Not Ready State	no	
Advice of Charge for EuroISDN**	yes	See details later in this chapter.
Advice of Charge for AXE-10 Australia and Japan D70**	yes	See details later in this chapter.
Alternative Conference Pad Levels	yes	
Alternative Loss Plan	yes	
Application Module	n/a	
Asia Pacific CO Connectivity	yes	trunks only
ARIES Automatic Gain Control	n/a	

[1] ISDN BRI Terminal can ONLY initiate calls to the user equipped with this feature.

Features and services	Supported	Comments
ARIES Handsfree Download	n/a	
Attendant Administration	no	
Attendant Alternative Answering**	P	See Footnote.
Attendant Barge-in	no	
Attendant Blocking of DN**	P	See details later in this chapter.
Attendant Break-in	no	
Attendant Busy Verify	no	
Attendant Call Party Name Display.	yes	
Attendant Call Selection	n/a	
Attendant Calls Waiting Indication	n/a	
Attendant Consoles	no	
Attendant Display of Speed Call or Autodial	no	
Attendant End-to-End Signaling**	yes	See details later in this chapter.
Attendant Forward No Answer**	P	See details later in this chapter.
Attendant Incoming Call Indicators	n/a	
Attendant Interpositional Transfer	n/a	
Attendant Lockout	n/a	
Attendant Overflow Position**	P	See Footnote. and details later in this chapter.
Attendant Overflow Position Busy	no	
Attendant Position Busy. See also Night Service	n/a	
Attendant Recall**	P	See Note 2 and details later in this chapter.
Attendant Recall with Splitting Optional	no	
Attendant Release**	P	See details later in this chapter.
Attendant Secrecy	yes	
Attendant Splitting	yes	
Attendant Supervisory Console	n/a	
Attendant Trunk Group Busy Indication	n/a	

Features and services	Supported	Comments
Audible Message Waiting	yes	See Note 3
Audible Reminder of Held Call	no	
Attendant Through Dialing Network Wide	yes	trunks only
Australia ETSI	yes	
Autodial	yes	See Note 3
Autodial with Authorization Code	no	
Autodial Tandem Transfer	no	
Automatic Answerback	no	
Automatic Gain Control Inhibit and Handset Volume Reset	no	
Automatic Guard Detection	yes	
Automatic Hold	no	
Automatic Line Selection	yes	See Note 3
Automatic Number Identification	no	See Note 4
Automatic Number Identification on DTI	no	See Note 4
Automatic Preselection of Prime DN	yes	See Note 3
Automatic Redial	no	
Automatic Set Relocation	no	
Automatic Timed Reminders**	P	See Note 5 and details later in this chapter.
Automatic Trunk Maintenance	n/a	
Automatic Wake Up	no	
Automatic Wake Up Flexible Feature Code Delimiter	no	
Auxiliary Processor Link	n/a	

Features and services	Supported	Comments
Auxiliary Signaling	no	
— B —		
B-Channel Overload Control	no	
Background Terminal	no	
Bar Reciprocal Call Forward	no	
Barge-in	no	

Features and services	Supported	Comments
Basic Alternate Route Selection	yes	
Basic Authorization Codes	no	
Basic Call, North American ISDN PRI Connectivity**	yes	See details later in this chapter.
Bearer Capability in CDR	no	
Boss/Secretary Filtering Enhancement	no	
Break-in Features • Break-in to inquiry calls • Break-in to line lockout denied	no	
Bridging	no	
Business Network Express/EuroISDN Call Diversion	P	Supported over BRIT trunks, but not on BRI sets
Business Network Express/EuroISDN Explicit Call Transfer	yes	
Business Network Express/EuroISDN Name and Private Number Display	yes	
Busy Lamp Field Array	no	
Busy Tone Detection for APAC and CALA	no	
Busy Tone Detection for Japan	no	
Busy Tone to Night DN on Busy DN**	P	See details later in this chapter.
Busy Verify on Calling Party Control	no	
Buzz - See Manual Signaling (Buzz)	no	
— C —		
Call-Back Queuing	no	
Call-Back Queuing/Conventional Main	no	
Call Capacity Report Enhancement	no	
Call Connection Restriction	yes	
Call Detail Recording	yes	
Call Detail Recording Enhancement	no	
Call Detail Recording Expansion	yes	
Call Detail Recording 100 Hour Call	no	
Call Detail Recording on Busy Tone	yes	

Features and services	Supported	Comments
Call Detail Recording Outpulsed Digits	no	
Call Detail Recording with Optional Digit Suppression	yes	
Call Forward All Calls**	P	See details later in this chapter.
Call Forward and Busy Status	no	
Call Forward, Break-in and Hunt Internal or External Network Wide	no	
Call Forward Busy	yes	See ISDN BRI Special Call For Busy
Call Forward Destination Deactivation	yes	
Call Forward External Deny	no	
Call Forward/Hunt Override via FFC**	P	See details later in this chapter.
Call Forward and Hunt by Call Type**	yes	See details later in this chapter.
Call Forward No Answer**	yes	See details later in this chapter.
Call Forward No Answer/ Flexible	yes	See note 7.
Call Forward/Save on Dump	no	
CFNA, Second Level for MWA stations	yes	See note 7.
Call Forward No Answer, Second Level	yes	See note 7.
Call Forward to Trunk Restriction	yes	
Call Hold**	yes	See details later in this chapter.
Call Hold, Deluxe	no	
Call Hold, Individual Hold Enhancement	no	
Call Hold, Permanent	no	
Calling Line Identification**	P	See details later in this chapter.
Calling Line Identification on Analog Trunks for Singapore, Australia, and Hong Kong (A-CLID)	P	Supported over BRIT trunks
Calling Line Identification Presentation	yes	See details later in this chapter.

Features and services	Supported	Comments
Calling Line Identification Restriction	yes	See details later in this chapter.
Call Page Network Wide	yes	
Calling Party Privacy	no	
Calling Party Privacy Override	no	
Call Park**	P	See details later in this chapter.
Call Park Network Wide	no	
Call Park on Unsupervised Trunks	no	
Call Party Name Display**	P	See details later in this chapter.
Call Pickup	no	
Call Pickup, Directed	no	
Call Pickup Network Wide	no	
Call Processor Card NT5D10	n/a	
Call Redirection by Day	no	
Call Selection	n/a	
Call Splitting	yes	
Call Status Indication	n/a	

Features and services	Supported	Comments
Call Transfer**	P	See Note 2 and details later in this chapter
Call Waiting/Internal Call Waiting**	yes/yes	See details later in this chapter.
Called Party Control on Internal Calls	no	
Called Party Disconnect Control**	P	See details later in this chapter.
Calling Line Identification**	P	See details later in this chapter.
Calling Line Identification Presentation and Restriction**	yes	See details later in this chapter.
Calling Party Number	yes	
Calling Party Privacy Override	no	
Charge Account and Calling Party No	yes	See Note 1

Features and services	Supported	Comments
Calls Waiting Indication (Attendant)	n/a	
CAMA Trunks	no	
Camp-on	no	
Camp-on to Multiple Appearance DN	no	
Capacity Expansion	yes	
Card LED Status	no	
Centralized Multiple Line Emulation	no	
Centrex Switchhook Flash	no	
Code Restriction	yes	
Charge Account and Calling Party Number **	P	See details later in this chapter.
Charge Account, Forced	no	
Charge Display at End of Call	no	
China #1 Signaling Features**	P	See details later in this chapter.
CIS ANI Reception	yes	sets only
CIS ANI Digit Manipulation and Gateway Enhancement	yes	
CIS Multifrequency Shuttle	no	
CIS Toll Dial Tone Detection	no	
CLASS: Calling Number and Name Delivery	no	
CLASS: Visual Message Waiting Indicator	no	
Class of Service	yes	
CO Trunk Priority Option - Call Pickup	no	
Conference	no	See Note 2
Conference Control	n/a	
Console Digit Display	n/a	
Console Presentation Group Level Services	no	
Control of Trunk Group Access	n/a	
Controlled Class Of Service	no	
Controlled Class of Service, Enhanced	no	
Coordinated Dialing Plan	yes	

Features and services	Supported	Comments
Coordinated Call-Back Queuing	no	
Coordinated Call-Back Queuing - Main	no	
— D —		
Data access via Data Modules	P	See Note 2
Data Port Hunting	yes	
Data, Circuit Switched**	yes	See details later in this chapter.
Data, Packet**	yes	See details later in this chapter.
D-Channel Expansion	no	
Default Loss Plan	yes	
Deluxe Hold	no	
Departmental Listed Directory Number (LDN)**	yes	See details later in this chapter.
Dial Access to Group Call	no	
Dial Intercom	no	
Dial Pulse/Dual Tone Multifrequency Conversion	n/a	
Dial Tone Detection	no	
Dialed Number Identification Service	P	See note 7.
DID to Network Calling	yes	
Digit Display	yes	See note 7.
Digital Private Networking Signaling System 1 (DPNSS1)	no	
Digital Trunk Interface	yes	
Direct Inward System Access	yes	
Directed Call Pickup	no	
Directory Number	yes	
Directory Number Expansion**	yes	See details later in this chapter.
DISA on Unsupervised Trunks	no	
Display of Access Prefix on CLID	yes	trunks only
Display of Calling Party Denied	no	

Features and services	Supported	Comments
Distinctive/New Distinctive Ringing	no	
Distinctive Ringing by DN	no	
Distinctive Ringing for Dial Intercom	n/a	
Distinctive Ringing Network Wide	no	
Do Not Disturb**	yes	See details later in this chapter.
DNIS Length Flexibility	no	
DSN Station Loop Preemption	no	
Dual Value Added Server Identification	n/a	
— E—		
E.164/ESN Numbering Plan Expansion	yes	
Electronic Brandlining	no	

Features and services	Supported	Comments
Electronic Switched Network	yes	
Emergency Services Access	yes	
End Dialing on Direct Inward/Outward Dialing	yes	
End of Selection	yes	
End of Selection Busy	yes	
End-to-End Signaling	yes	
Enhanced Charge Display	no	
Enhanced Controlled Class of Service	no	
Enhanced End-to-End Signaling	yes	
Enhanced Flexible Hot Line**	P	See Note 2
Enhanced Malicious Call Trace	no	
Enhanced Music	yes	
Enhanced Night Service	no	
Enhanced 911 Interface	no	
Enhanced Secrecy	no	
Electronic Lock Network Wide/Electronic Lock on Private Lines	no	
Equal Access Compliance	yes	

Features and services	Supported	Comments
ESN Queuing	no	
EuroISDN**	yes	See details later in this chapter.
EuroISDN 7kHv/Videotelephony Teleservices	yes	
EuroISDN Continuation	yes	
EuroISDN Continuation Phase III	yes	trunks only
EuroISDN ETS 300 403 Compliance	yes	
EuroISDN Malicious Call Identification	yes	trunks only
EuroISDN Network Side	yes	
Exclusive Hold	no	
Executive Distinctive Ringing	no	
— F —		
510 Trunk Route Member Expansion	yes	
Fast Tone Digit Switch	n/a	
FCC Compliance for DID Answer Supervision	n/a	
FCC Compliance for Equal Access	yes	
FCC Compliance for Equal Access - CAC Expansion	no	
First-Second Degree Busy Indication	no	
Flexible Attendant DN	n/a	
Flexible Call Forward No Answer	yes	
Flexible Attendant Call Waiting Threshold	no	
Flexible Direct Inward Dialing	no	
Flexible Busy Tone Timer	no	
Flexible Dial Tone Detection	no	
Flexible Direct Inward Dialing	no	
Flexible ESN "0" Routing	yes	
Flexible Hot Line	P	See Note 2
Flexible Feature Codes	no	
Flexible Key Assignment	no	
Flexible Line Lockout	no	

Features and services	Supported	Comments
Flexible Numbering Plan	no	
Flexible Numbering Plan Enhancement	no	
Flexible Orbit Prevention Timer	yes	
Flexible Tone and Digit Switch Control	no	
Flexible Tones and Cadences	no	
Flexible Voice/Data Terminal Number	no	
Forced Camp-on**	P	See details later in this chapter.
Forced Charge Account	no	
Forward No Answer Call Waiting for DID	no	
— G —		
Global CDR Record	no	
Global Line Cards	no	
Group Call	no	
Group Hunt**	P	See details later in this chapter.
Guest Entry of Automatic Wake Up	no	
— H —		
History File	yes	
Hot Line	P	See Note 2
Hold	yes	Not supported by 1TR6 protocol
Hold Features • Automatic Hold • Held Call Clearing		
Hospitality Screen Enhancements	no	
Hotel/Motel features	no	
• Automatic Wake-Up	no	
• Background Terminal	no	
• Controlled Class of Service	no	
• Maid ID	no	

Features and services	Supported	Comments
• Property Mgmt. Sys. Interface	no	
• Room Service	no	
Hunting	yes	
Hunting by Call Type	yes	
Hunting and Speed Call Features**	P	See details later in this chapter.
— I, J—		
Idle Extension Notification	no	

Features and services	Supported	Comments
IMS/VMS**	P	See details later in this chapter.
In-Band ANI	n/a	
Incoming DID Digit Conversion**	P	See note 7 and details later in this chapter.
Incremental Software Management**	yes	See details later in this chapter.
Incoming Call Indicators	n/a	
Incoming Trunk Group Exclusion	n/a	
Incoming Trunk Programmable CLID	n/a	
India Phase II	no	
Individual Hold	no	
INIT ACD Queue Call Restore	no	
Inspect key	yes	See Note 3
Instant ISM	no	
Integrated Messaging System Link	P	See IMS/IVMS
Integrated Voice and Data	yes	
Intercept Computer Interface Dial From Directory	no	
Intercept Computer Enhancements • Answering Machine enhancements • Malicious Call Trace DN and TN print • Call Forward interaction control**	 yes yes	

Features and services	Supported	Comments
	P	See details later in this chapter.
Intercept Computer Interface**	P	See details later in this chapter.
Intercept Treatment**	yes	See details later in this chapter.
Intercept Treatment Enhancement	no	
Interchangeable NPA	no	
Internal Call Detail Recording	yes	
Internal Call Waiting	yes	
Interpositional Transfer	n/a	
Inventory Reporting	no	
IPE Completion	no	
IPE Loss Plan for China	no	
ISDN Application Protocol	no	
ISDN Call Connection Limitations	yes	
ISDN BRI Calls to MCA/MCU	yes	
ISDN BRI Circuit- Switched Data Call Accessing DTI Trunk	yes	
ISDN BRI Connected Line Presentation/ Restriction	yes	
ISDN BRI to ISDN BRI Circuit Switched Data Call	yes	
ISDN BRI Circuit-Switched Data Call over PRI	yes	
ISDN BRI Circuit- Switched Data Call for Tandem	yes	
ISDN BRI Network Ring Again	yes	
ISDN NI-1 BRI Compliance Enhancements	yes	
ISDN BRI Special Call Forward Busy	yes	
ISDN BRI Special Hunting	yes	
ISDN BRI Trunk Access for Japan	yes	
ISDN Calling Line Identification Enhancements	yes	
ISDN QSIG Alternate Routing	no	

Features and services	Supported	Comments
ISDN QSIG Basic Call	yes	
ISDN QSIG Call Completion	no	
ISDN QSIG Call Completion Enhancement	no	
ISDN QSIG Call Diversion Notification	yes	BRI sets are supported.
ISDN QSIG Call Diversion Notification Enhancement	yes	BRI sets are supported.
ISDN QSIG Call Transfer Notification	yes	BRI sets are supported.
ISDN QSIG Generic Functional Transport	no	
ISDN QSIG/ETSI Generic Functional Transport Enhancement	no	
ISDN QSIG/EuroISDN Call Completion	yes	
ISDN QSIG/EuroISDN Call Completion Enhancement	yes	
ISDN QSIG Name Display**	P	See details later in this chapter.
ISDN QSIG Path Replacement	yes	trunks only
ISDN QSIG Supplementary Services - Name Display Enhancements**	yes	
ISDN Semi-permanent Connection for Australia	no	
ISM Enhancements	no	
Italian Phase 2 Features for DTI2	no	
Japan (D70) PRI nB+D	no	
Japan TTC Common Channel Signaling	no	
— L —		
Last Number Redial	yes	See Note 3
Limited Access to Overlays	yes	
Line Load Control	no	
Line Lockout	no	
Listed Directory Numbers	n/a	

Features and services	Supported	Comments
Lockout	n/a	
Loop Start Supervisory Trunks	no	

Features and services	Supported	Comments
Loop Start Supervisory Trunks (incoming calls)	no	
— M —		
Maid Identification	no	
Maintenance Telephone	no	
Make Set Busy	no	
Make Set Busy Enhancement	no	
Make Set Busy Improvement	no	
Malicious Call Trace**	P	See Note 2 and details later in this chapter
Malicious Call Trace Enhancement	no	
Malicious Call Trace Idle	no	
Malicious Call Trace on DID	no	
Manual Line Service	n/a	
Manual Service Recall to Attendant	no	
Manual Signaling (Buzz)	no	
Manual Trunk Service	yes	
MCDN Alternate Routing	no	
MCDN End to End Transparency	no	
Meridian 1 Attendant Console Enhancement	n/a	
Meridian 1 to New Zealand NEAX-61 ISDN PRI connectivity**	yes	See details later in this chapter.
Meridian Communications Adapter (MCA) Data Module**	P	See details later in this chapter.
Meridian Companion Enhanced Capacity	N/A	
Meridian Hospitality Voice Services	no	
Meridian Mail Trunk Access Restriction	no	
Meridian MAX	n/a	
Meridian Mail Trunk Access Restriction**	P	See details later in this chapter.
Message Center**	P	See details later in this chapter.
Message Intercept	no	

Features and services	Supported	Comments
Message Registration	no	
Message Waiting Lamp Maintenance	no	
Message Waiting Indicator by Directory Number	no	
Message Waiting Unconditional	no	
Modem Trunk Hunting	no	
MSDL Idle Code Selection	N/A	
MSDL Port Overload Counter	N/A	
MSDL Status Enquiry Message Throttle	N/A	
Multifrequency Compelled Signaling Features**	P	See details later in this chapter.
Multiple Appearance Directory Number (DN)**	P	On the same DSL only. See details later in this chapter.
Multiple Appearance Redirection Prime	no	
Multi Language Messages	no	
Multi-Language Wake Up	no	
Multi-Party Operation**	p	See details later in this chapter.
Multiple-Customer Operation	yes	
Multiple-Console Operation	n/a	
Multiple DID Office Code Screening	n/a	
Multiple-Tenant Service	no	
Multi-Site Mobility Networking	no	
Music	yes	
Music Broadcast	no	
Music, Enhanced	yes	
Music on Hold**	yes	See details later in this chapter.
— N —		
N Digit DNIS	no	
Network Anti-tromboning	no	
Network Application Protocol Link Enhancement	no	

Features and services	Supported	Comments
Network Automatic Call Distribution	P	See Note 1
Network Alternate Route Selection**	yes	See details later in this chapter.
Network Attendant Service**	p	See details later in this chapter.
Network Authorization Codes	no	
Network Call Party Name Display	P	
Network Call Pick-up and TAFAS	no	
Network Call Transfer**	P	See Note 2 and details later in this chapter
Network Call Redirection Service**	P	See details later in this chapter.
Network Call Trace and Call Diagnostics	no	
Network Class of Service**	yes	See details later in this chapter.
Network Drop Back Busy and Off-hook Queuing	yes	
Network Individual Do Not Disturb	no	
Network Intercom	no	
Network Message Center	P	See Message Center
Network Ring Again	no	

Features and services	Supported	Comments
Network Signaling	yes	
Network Signaling on VNS	no	
Network Signaling for Network ACD**	P	See details later in this chapter.
Network Speed Call	P	See Note 2
Network Tenant Service	no	
Network Traffic	n/a	
Network Wide LDN**	P	See details later in this chapter.
Networking Features**	P	See details later in this chapter.
New Distinctive Ringing	no	

Features and services	Supported	Comments
New Flexible Code Restriction	yes	
NI-2 Call By Call Service Selection	no	
Night Key for DID Digit Manipulation	no	
Night Service/TAFAS**	P	See Note 1 and details later in this chapter
Night Service Improvements	P	See details later in this chapter.
Night Service by Time of Day	P	See Note 6
No Hold Conference	P	See Note 2
NPI and TON in CDR tickets	no	
— O —		
Off-Hook Alarm Security	no	
Off-Hook Queuing	no	
Off-Net Number Recognition	yes	
Off-Premise Extension	n/a	OPX - applicable to analog sets only
Office Data Administration System	no	
On Hold on Loudspeaker	no	
On-Hook Dialing	yes	See Note 3
Optional Outpulsing Delay	no	
Optional Privacy	no	
Outgoing Trunk Hunting	n/a	
Overflow Tone	yes	
Overlap Signaling	yes	
Overlay Cache Memory	yes	
Override	no	
Outpulsing of Asterisk and pound**	yes	See details later in this chapter.
— P —		
Packet Switched Data Service	yes	Supported for both B and D channels
Paging	yes	
Partial Dial Timing	no	

Features and services	Supported	Comments
Periodic Camp-on Tone	no	
Periodic Clearing	no	
Periodic Pulse Metering**	P	See details later in this chapter.
Permanent Hold	n/a	
Phantom TNs	no	
Position Busy	n/a	
Preference Trunk Usage	yes	
Pretranslation**	yes	See details later in this chapter.
Pretranslation and System Speed Call Enhancement	no	
Preventing Reciprocal Call Forward	no	
Prime DN	n/a	
Priority Override**	p	See details later in this chapter.
Privacy	no	
Privacy Override	no	
Privacy Release	no	
Private Line Service	no	
Private to Public CLID Conversion	no	
Process Notification for Networked Calls	no	
Property Management System Interface	no	
Pulsed E&M DTI2 Signaling	no	
— R —		
Radio Paging Improvements**	P	See details later in this chapter.
Radio Paging Product Improvement Continuation	no	
Recall (Attendant)**	P	See Note 2 and See details later in this chapter.
Recall After Parking	no	
Recall to Same Attendant**	P	See details later in this chapter.

Features and services	Supported	Comments
Recall with Priority During Night Service	yes	
Recorded Announcement**	yes	See details later in this chapter.
Recorded Announcement Broadcast	no	
Recorded Announcement for Calls Diverted to External Trunks**	P	See details later in this chapter.
Recorded Overflow Announcement	yes	
Recorded Telephone Dictation	yes	
Recovery on Misoperation at the Attendant Console	no	
Remote Call Forward	no	
Remote Peripheral Equipment	no	
Remote Virtual Queuing	no	

Features and services	Supported	Comments
Restricted Call Transfer	no	
Restricted DID Class of Service	yes	
Ring Again	no	
Ring Again on No Answer	no	
Ring and Hold Lamp Status	no	
Ringing Change Key	no	
Ringing Number Pickup	no	
RPE (2Mb) Alarm Handling	no	
R2 Multifrequency Compelled (MFC) Signaling	yes	
R2 MFC Selective Route to Attendant	yes	
R2 MFC CNI/CDR Enhancements	yes	
R2 MFC Timer Control	no	
Room Status	no	
Round Robin Trunk Hunting	n/a	
— S —		
7 Digit DNIS for MAX	n/a	
Schedule Access Restriction	no	

Features and services	Supported	Comments
Secrecy (Attendant)	yes	
Secretarial Filtering	no	
Selectable Conferee Display and Disconnect	no	
Semi-automatic Camp-on	no	
Semi-compelled MFC and Calling Number Identification Changes	no	
Series Call	no	
Seizure Acknowledgment	yes	
Selectable Directory Number Size	yes	
Short Buzz for digital sets	n/a	
Signal source and destination	no	
Single Appearance DN	yes	
Single Digit Access to Hotel Services	no	
Six Party Conference for 2500 Sets	yes	See Note 3
Slow Answer Recall Modification	yes	
Software Capacity Expansion	yes	
Source Included When Attendant Dials	no	
Spanish KD3 DID/DOD Interface	no	
Special Dial Tone	P	Message Waiting Indication
Special Dialtone after Dialed Numbers**	P	See details later in this chapter.
Special Service prefix (SPRE)	no	
Speed Call	P	See Note 2
Speed Call DN Access	no	
Speed Call Delimiter	no	
Speed Call on Private Lines	no	
Speed Call with Authorization Codes	no	
Speed Call, System	no	
Splitting** (See Attendant Splitting)	yes	See details later in this chapter.
Station Activity Record	yes	
Station Category Indication	no	

Features and services	Supported	Comments
Station Hunting--see ISDN BRI special hunting	yes	
Station Loop Preemption	no	
Station Camp-on	no	
Station-to-Station-Calling	yes	
Station-to-Station Call Waiting	yes	
Stored Number Redial	yes	See Note 3
Supervised CO Trunk Simplification		
Supervisory Attendant Console	n/a	
System Capacity Expansion	yes	
Switched 56 Kbps Service**	P	See details later in this chapter.
— T —		
10/20 Digit ANI on 911 Calls	no	
Taiwan R1 Modified Signaling	yes	sets only
Telelink Mobility Switch	no	
Telset Call Timer Enhancement	no	
Time and Date	yes	See Note 3
Time Forced Disconnect** (See Paging)	yes	See details later in this chapter.
Tones, Flexible Incoming	no	
Tone to Last Party	no	
Transfer**	P	See Note 2 and details for Call transfer later in this chapter
Transfer Tone	no	
Trunk Answer From Any Station	no	
Trunk Anti-Tromboning	no	
Trunk Barring	yes	
Trunk Direct Inward Dialing Digitone Receiver Acknowledgment	no	
Trunk Failure Monitor	no	
Trunk Group Access Restrictions (TGAR)	yes	

Features and services	Supported	Comments
Trunk Group Busy Keys/Indication**	yes	See details later in this chapter.
Trunk Hook Flash Enhancement	no	
Trunk Hunting	n/a	
Trunk Signaling Arrangements	n/a	
Trunk to Trunk Connection	no	
Trunk Verification from a Station	no	SPRE codes are not supported
— U —		
Uninterrupted Line Connections	no	
Uniform Dialing plan	yes	
Universal ISDN Protocol Engine	yes	
— V —		
Variable Flash Timing and Ground Button	no	
Variable Guard Timing	yes	
VIP Wake up	no	See Automatic Wake Up
Virtual Network Services	yes	
Virtual Network Services in the UK with DASS2/DPNSS1 Bearers	no	
Virtual Network Services Virtual DN Expansion	no	
Voice Call	yes	See Note 3
Voice Call Override	no	

Note:

- ISDN BRI Terminal CAN NOT invoke this feature; however, the non-ISDN BRI calling party (calling an ISDN BRI user) will have access to this feature.
- Feature is ISDN BRI Terminal dependent.
- Automatic Number Identification, as used here, refers to the Hotel/Motel ANI feature. Please refer to "Calling Party Number" feature listed above for CLID.
- Camp-on feature can not be invoked against an ISDN BRI Terminal.

- Feature is not available to ISDN BRI user, however, ISDN BRI Terminal calling other non-ISDN BRI set equipped with this feature will receive appropriate call handling treatment.
- ISDN BRI Terminal display, if equipped, will not be updated to reflect call modification.

The following descriptions provide additional information for the features tagged with "**" in Table 165: Features on ISDN BRI lines on page 288. These descriptions provide further details of the feature interaction with ISDN BRI.

Access Restrictions

Access Restrictions for an ISDN BRI terminal are based on assigned Class of Service, Trunk Group Access Restriction, and area and exchange codes dialed by the terminal.

Class of Service and Trunk Group Access Restriction for an ISDN BRI terminal are defined when configuring DSL using Service Change ISDN BRI Program LD 27.

The same access restrictions apply to all voice and data ISDN BRI DNs defined for a DSL.

Access to Paging

An ISDN BRI terminal can access paging equipment by dialing a paging trunk access code. End-to-end signaling must be provided by the ISDN BRI terminal.

Access to Recorded Telephone Dictation

An ISDN BRI terminal can access dictation equipment by dialing an equipment access code. End-to-end signaling must be provided by the ISDN BRI terminal.

Attendant End-to-End Signaling

This feature is supported for ISDN BRI terminals where the attendant can signal to an ISDN BRI terminal and an ISDN BRI terminal can dial the attendant and request access to services requiring end-to-end signaling.

Call Forward No Answer

This feature is supported for calls originated by an ISDN BRI terminal and calls terminated at an ISDN BRI terminal. Internal and external call forward no answer DNs are defined in the DSL, and call forward no answer enable and call forward by call type are defined for ISDN BRI DNs in the TSP using Service Change ISDN BRI Program LD 27.

An ISDN BRI DN can be defined as a Call Forward No Answer DN allowing the features available on the ISDN BRI terminal to control the call.

Call forward no answer and second level call forward no answer are activated if FNA and SFA are selected for a DN when configuring the TSP using Service Change ISDN BRI Program LD 27. If a call is not answered after a predetermined timer expires, the initial call setup is released from the DSL and call forward no answer routes the call to an alternate DN.

An ISDN BRI terminal originating the call is not updated to show on its display that the call was redirected to a different DN as a result of call forward no answer. This is due to lack of standard for layer 3 messages used to update the ISDN BRI terminal display.

Call Hold (not supported by 1TR6 protocol)

This feature is activated by pressing the Hold key on the terminal. This feature allows an ISDN BRI terminal to place an active call on hold to accept or originate another call. You can dial a new call and go back to the call on hold.

An ISDN BRI terminal can place an ISDN BRI or non-ISDN BRI terminal on hold.

Deluxe hold and permanent hold do not apply to an ISDN BRI terminal, however a non-ISDN BRI terminal can place an ISDN BRI terminal in deluxe hold or permanent hold.

Call Waiting

This feature alerts a busy ISDN BRI terminal that a call is waiting to be answered. To answer a waiting call, the ISDN BRI terminal must place the active call on hold or release the active call. If the ISDN BRI terminal user tries to answer a call waiting call when no B-channel is available, the call is released from the ISDN BRI terminal and extended to the attendant.

When both B-channels on a DSL are busy, an incoming call is presented to the DSL as call waiting as long as the number of calls on this DSL does not exceed the maximum number of calls specified, which include active calls, calls waiting, and calls on hold. This maximum number of simultaneous calls allowed is specified when configuring the DSL using Service Change ISDN BRI Program LD 27.

A call waiting from an ISDN BRI terminal is presented to a non-ISDN BRI terminal as a normal call waiting call.

Call waiting for a call terminating at a DSL will be activated only if do-not-disturb and hunting for this DSL are not enabled or fail to handle the call. ISDN BRI call waiting is subject to call forward no answer as defined in the system.

Departmental LDN

This feature allows specified terminals to share the same numbering plan and to access the attendant console or consoles dedicated to a departmental LDN.

Directory Number Expansion

ISDN BRI terminal DNs can be increased from a maximum length of four digits to a maximum length of seven digits when the DN Expansion option is equipped.

Do Not Disturb

The attendant can place an ISDN BRI DN in do-not-disturb mode, which allows the terminal to make outgoing calls but it makes it look busy to incoming calls.

Incremental Software Management

Software pricing is implemented based on the number of ISDN BRI DSLs (United States only).

Intercept Treatment

A call originated by an ISDN BRI terminal that is intercepted can be routed to the attendant, given a busy tone, given an overflow tone, or routed to a recorded announcement. There is no special treatment for ISDN BRI terminals.

Music on Hold

ISDN BRI terminals on hold and terminals placed on hold by an ISDN BRI terminal can receive music if configured to do so. An ISDN BRI terminal can also access a music trunk by dialing the route access code.

Network Wide Listed Directory Numbers

This feature 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.

This feature is supported over ISDN BRI trunk DSLs.

Pretranslation

An ISDN BRI DN can be assigned to a first digit pretranslation group.

The first digit dialed on an ISDN BRI terminal can be pretranslated into a DN while making a call.

Pretranslation is configured for a DSL therefore all DNs for a DSL are assigned to the same pre translation group.

An ISDN BRI DN can also be assigned as a translated DN in a pre translation group's speed call list.

Recorded Announcement

An ISDN BRI terminal can be connected to a recorded announcement machine by an intercept condition, by the attendant, by an ACD agent, or by direct access to a recorded announcement machine.

Departmental listed directory number groupings are supported for ISDN BRI and are configured for a DSL using Service Change ISDN BRI Program LD 27. This allows assignment of a DSL to one of the four listed directory number groups. All DNs for a DSL are assigned to the same departmental LDN.

Network Alternate Route Selection

ISDN BRI terminals can have the following networking capabilities: simple network access codes, uniform dialing plan, digit manipulation, automatic least-cost routing, time-of-day routing, automatic on-net to off-net overflow, network control through network Class of Service, traveling Class of Service and facility restriction level, routing control through network Class of Service based on time-of-day schedule, 11 digit translation, free calling area screening, expensive route warning tone, network call detail recording, and network speed dial.

Network Class of Service

An ISDN BRI DSL can be assigned a network Class of Service to control access to routes, ability to receive expensive route warning tone, and ability to access network speed dial. All DNs for a DSL are assigned to the same NCOS.

Outpulsing of Asterisk and Outpulsing of Pound

Routes can be dedicated for the outpulsing of asterisks (*) and pound (#) when an ISDN BRI terminal is making a trunk call.

Splitting

The attendant selectively talks to either party of an established call made by an ISDN BRI terminal by excluding one party from the conversation or the other using the Exclude Source or Exclude Destination key.

Timed Forced Disconnect

An ISDN BRI terminal is disconnected if it uses the paging trunk longer than a preset time.

Trunk Group Busy Keys/Indication

When all trunks in a trunk group or its hunting groups are busy, a user-selectable tone is returned to the calling ISDN BRI terminal.

Set-based ISDN BRI features on system ISDN BRI terminals

The following features are supported on M5317TDX terminals:

- AutoDial Keys (up to 9 keys each with 20 digit numbers)
- Inspect Key
- Data Port Configuration from the menu
- HandsFree
- Conference - Soft Key (recommended key #15)
- DiscData Soft key for Disconnecting Data Calls
- English/French Language Support
- Set-based Ringing Patterns
- Set-based Clock

The following features are supported on M5209TDcp terminals:

- Last Number Redial (invoked by ##)
- Set based Speed Call (Can store up to five 25 digit numbers)
- Store Number Redial (multiple keys)
- Conference (recommended key #9)
- DiscData Soft key for Disconnecting Data calls
- English/French Language Support
- Set-based Ringing Patterns

Features partially supported by ISDN BRI

The following descriptions provide additional information for the features tagged with "P" and/or "**" in Table 165: Features on ISDN BRI lines on page 288.

Attendant features

Attendant Alternative Answering

An ISDN BRI DN cannot be assigned as an attendant alternative answering DN. Calls originating from an ISDN BRI terminal can be forwarded to an attendant alternative answering DN defined for the attendant console when the attendant does not answer the call. The originating ISDN BRI terminal display is not updated.

Attendant Blocking of DN

This feature 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.

This feature is supported over ISDN BRI trunks in an MCDN environment, if NAS is equipped. It is not supported if a call is extended to an ISDN BRI set.

Attendant Forward No Answer

This feature allows calls that are not answered by an attendant within a defined period of time, to be routed to the night DN or to another attendant. It also allows DID calls that are not answered within a defined period of time, to be disconnected.

This feature is transparent to calling ISDN BRI terminals. However, originating a call routed by Attendant Forward No Answer cannot be updated. Also, an ISDN BRI terminal cannot be defined as a night terminal.

Attendant and Network Wide Remote Call Forward

This enhancement introduces the RCFW feature across the Meridian Customer Defined Network (MCDN), while also providing the attendant with RCFW capabilities.

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

This feature is supported over ISDN BRI trunks within an MCDN environment; it is not supported on ISDN BRI sets.

Attendant Overflow Position

An ISDN BRI DN cannot be assigned as an attendant overflow position DN. Calls originated by ISDN BRI terminals can be automatically routed to a predefined DN, however, the terminal display will not be updated to show the call modification.

Attendant Recall

An ISDN BRI terminal communicating with a non-ISDN BRI terminal is recalled to the attendant as the source party if the recall is initiated by the non-ISDN BRI terminal. An ISDN BRI terminal cannot initiate an attendant recall.

Attendant Release

The attendant has no control over disconnection of an ISDN BRI terminal if the call is with another ISDN BRI terminal. The ISDN BRI terminal can release the call even if the attendant has not pressed the Release key.

Automatic Timed Reminders

A call made by an ISDN BRI terminal can be extended by the attendant to an ISDN BRI or non-ISDN BRI terminal and timed for slow answer or call waiting recall. When the timer expires, the call is recalled to the attendant.

Camp-on recall is not supported by the ISDN BRI DN.

Busy Tone to Night DN on Busy DN

This feature allows busy tone, rather than ringback tone, to be provided to a night DN during night service.

This feature is supported transparently if the calling station is an ISDN BRI terminal; however, an ISDN BRI terminal cannot be supported as a night DN terminal.

Automatic Call Distribution

An ISDN BRI terminal cannot be an ACD agent terminal, however, an ISDN BRI terminal can place a call to an ACD agent.

Call Forward All Calls

An ISDN BRI terminal does not support Call Forward All Calls. Calls originating from an ISDN BRI terminal can be forwarded if call forwarding is activated by a non-ISDN BRI terminal. The calling ISDN BRI terminal display is not updated to show the call change. An ISDN BRI DN can also be defined as a call forward DN.

Call Forward/Hunt Override via FFC

Call Forward/Hunt 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. Sets without Call Forward/Hunt Override denied (CFHD) Class of Service will not be able to use the Call Forward/Hunt Override Via Flexible Feature Code (FFC) feature.

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

This feature is supported over ISDN BRI trunks in an MCDN environment, if NAS is equipped. It is not supported on ISDN BRI sets.

Call Park

An ISDN BRI terminal cannot be parked nor can it park a call, however it can retrieve a parked call through a system park DN. An ISDN BRI DN can be used as a station park DN.

Call Party Name Display

Names are not displayed on ISDN BRI terminals; however, a name for an ISDN BRI DN can be defined using Calling Party Name Display Program LD 95 to allow a non-ISDN BRI terminal with display to identify the ISDN BRI terminal that originated the call. This name can be transmitted across ISDN PRI network.

Called Party Disconnect Control

A disconnect signal from an incoming trunk call to an ISDN BRI terminal is ignored by the terminal.

A call originating from an ISDN BRI terminal and to an outgoing trunk can be disconnected from either end.

Call Transfer

An ISDN BRI terminal can be transferred by a non-ISDN BRI terminal and a non-ISDN BRI terminal can transfer a call to an ISDN BRI terminal. In either case, the ISDN BRI terminal display is not updated to indicate call modification.

Calling Line Identification

Calling line identification is sent across ISDN PRI to and/or from ISDN BRI terminals. Trunk access code is displayed at the called ISDN BRI terminal instead of the calling line identification if it has calling party number restricted.

If a network call is redirected the calling line identification is not updated.

Charge Account/Calling Party Number

Charge account cannot be activated by an ISDN BRI terminal; however, a non-ISDN BRI terminal communicating with an ISDN BRI terminal can activate charge account and enter an ISDN BRI DN as the calling party number to be included in the call detail recording report.

China 1 signaling features

This group of features is comprised of the following:

- External Operator features, comprised of:
  - Call Back
  - Calling Party Control (CGPC);
  - Called Party Control (CDPC);
  - Toll Operator Break-in (TOBI).
- Multiple Frequency Compelled (MFC) Direct Outward Dial (DOD) with outgoing Calling Number Identification (CNI);
- Outgoing Toll Call Identification

The External Operator Call Back feature is used by CGPC, CDPC, and TOBI. It allows a calling station connected to a trunk call to be put on hold while going on hook. When the special operator signal is received over the trunk the system will ring the station again. CGPC allows the call to be kept on hold for a defined period of time, and to reconnect the call when the called party goes off hook.

The call is released if the calling party goes on hook once more. CDPC allows an operator, involved in an operator-assisted call from a system, to automatically recall a station without having to re-dial it. TOBI allows a toll operator to break in to an established call.

If an ISDN BRI terminal is the controlling party (that is, the called party in CDPC or the calling party in CGPC), the features are supported transparently, since disconnect messages from the trunk are ignored. If an ISDN BRI terminal is not the controlling party, when it goes on hook, this operation is not supported on ISDN BRI.

TOBI to an ISDN BRI terminal is not supported.

MFC DOD with outgoing CNI allows outgoing Calling Number Identification information on outgoing MFC trunk calls. CNI information is customer-related and terminal-related, both of which are supported for ISDN BRI. Terminal-related information is a category code defined in overlay 10 or 11. For ISDN BRI terminals, the category code is supported on a DSL basis. Overlay 27 has been modified to configure and print category codes, using the "category code" (CAC) prompt.

Outgoing Toll Call Identification allows outgoing toll calls to be identified in a new manner. This feature is supported transparently for ISDN BRI.

Addendum to China 1 signaling features

This group of features is comprised of the following:

- Operator Call Back Feature Enhancement;
- Special Operator Call Back Ringing;
- Malicious Call Trace Enhancement;
- Tones and Announcements;
- Active Feature Dial Tone;
- Audible Alarm.

The Operator Call Back Enhancement allows Call Wait or Camp-On to Calling Party Control and Called Party Control calls, and Attendant Break-in to outgoing Calling Party Control calls and incoming Called Party Control calls. The Camp-on and Break-in capabilities are not supported on ISDN BRI, while the Call Wait capability is supported transparently for ISDN BRI.

The Special Operator Call Back Ringing Enhancement provides operator control of ringing cadences used in Toll Operator Call Back. This capability is supported transparently for ISDN BRI.

The Malicious Call Trace Enhancement allows a system to have Called Party Control on incoming calls when the Malicious Call Trace (MCT) feature is activated from a station, or when the Multifrequency Compelled (MFC) Idle Call Trace (IDCT) signal is sent. This enhancement is not provided on ISDN BRI.

The Tones and Announcement feature allows a howler tone to be given to indicate that an analog (500/2500-type) terminal is off-hook. This capability is not applicable to ISDN BRI terminals. This feature also provides new intercept treatments upon the reception of some MFC signals. This capability is supported transparently for ISDN BRI.

The Active Feature Dial Tone capability provides a distinctive dial tone to a station going off hook when it has the Do Not Disturb (DND) or Make Set Busy (MSB) feature active. This capability is not provided on ISDN BRI.

The Audible Alarm feature provides an alarm to be sounded when an emergency number has been dialed, or when the system is alerted of an incoming malicious call. This capability is supported transparently for ISDN BRI.

Part 4 - Ministry of Electricity and Industry (MOEI) features

This group of features is comprised of the following:

- KE Multifrequency Compelled Signaling
- Flexible Timers (dialtone, interdigit, and delayed answer)
- Calling Party DN Option
- Flexible Feature Codes (FFCs)

KE Multifrequency Compelled Signaling is used to inform the Central Office that the call is a tandem call. This capability is supported transparently for ISDN BRI.

Flexible Timers provides customer-defined parameters to control the following time-outs:

- dialtone time-out, after the terminal has been placed off-hook and no digits dialed. After time-out, the terminal is placed in line lockout
- interdigit pause time-out, between the first and second digits, and the interdigit pause time-out, after the second digit
- delayed answer timer provides a customer-defined time-out to control the period that a terminal remains ringing before it is answered. If time-out occurs, the ringing and ringback stop, and the call is disconnected.

Flexible timers is not supported on ISDN BRI terminals.

Calling Party DN allows the system to send to the Toll Office the calling-party customer DN and, if applicable, the calling-party DID DN for all outgoing calls. This capability is supported when the calling party is an ISDN BRI terminal.

The Flexible Feature Code enhancement answers China's Ministry of Electronic Industry's requirement to access the following features from an analog (500/2500-type) terminal, using Flexible Feature Codes:

- Autodial
- Call Waiting
- Make Terminal Busy
- Multiple Wake-up

This capability is not provided for ISDN BRI.

Enhanced Flexible Hotline

Hotline is not available on an ISDN BRI terminal, however, an ISDN BRI DN can be defined as a Hotline number.

Forced Camp-on

Forced Camp-on allows a call to be camped on to a busy station while providing a warning tone. This is typically followed by the activation of Priority Override, which allows break-in to the established connection.

Forced Camp-on is activated automatically (if Automatic Forced Camp-on is defined); or, it can be activated manually using the Enhanced Override (EOVR) key on M1000 series and Meridian digital terminals or the Enhanced Override Flexible Feature Code on analog

(500/2500-type terminals). If the EOVR key is pressed again or the Enhanced Override Flexible Feature Code dialed again, Priority Override is activated.

The terminal performing the override must have a priority level equal to or higher than the terminal being overridden. To activate Priority Override, the user of a 500/2500 terminal dials the Override Flexible Feature Code, while the user of a M1000 series or Meridian digital terminal presses the Override key (OVR). Priority Override can also be activated using the Enhanced Override Flexible Feature Code or the Enhanced Override key (EOVR).

Forced Camp-on is partially supported on ISDN BRI terminals, as explained below:

- Forced Camp-on to an ISDN BRI terminal is not possible, because Forced Camp-on requires Warning Tone Allowed Class of Service (ISDN BRI terminals are defaulted to WTD COS);
- Forced Camp on from an ISDN BRI terminal is not possible because feature keys and FFC are not supported for ISDN BRI;
- a BRI terminal transferred to a non-BRI terminal can be have forced camp-on applied to it, if no COS restrictions apply; however, the ISDN BRI terminal display is not updated.

Group Hunt

Group Hunt is partially supported on ISDN BRI terminals, as explained below:

- an ISDN BRI terminal cannot be a member of a group hunt list, due to lack of support of FFC (Group Hunt Deactivation is not possible);
- an ISDN BRI terminal cannot access a group hunt list, since FFCs are not supported;
- a call from an ISDN BRI terminal to another ISDN BRI terminal can be hunted, or call forwarded no answer, to a Group Hunt Pilot DN FFC.

Incoming DID Digit Conversion

An ISDN BRI DN can be defined as the converted DN, however, the dialed number identification service number will not be displayed on an ISDN BRI terminal.

Intercept Computer Interface

This feature allows the system to use an intercept (attendant assistance service) computer for storing and retrieving call messages. Calls to an absent tenant's DN using this feature are routed to a designated Intercept Position (ICP) DN. A terminal at the ICP displays a message stating why the tenant at the DN is absent. The person at the ICP can then store the caller's message for the tenant's DN and activate the message waiting LED at the tenant's telephone. The tenant at the DN retrieves the stored caller messages by calling the ICP, where the messages are displayed on the terminal (or optionally printed).

The feature can be activated or deactivated by the following:

- Flexible Feature Code (FFC) dialed from the tenant's telephone. This code specifies the reason for the tenant's absence and can be extended with a date and time as extra information. The FFC decodes into a text message;
- Pressing the Call Forward All Calls (CFW AC) key on a Meridian 1 proprietary telephone (deactivation);
- From the ICP terminal;
- Automatically when a TN is disabled or enabled by a maintenance overlay program.

The Intercept Computer Interface is partially supported on ISDN BRI terminals, as explained below:

- an ISDN BRI terminal cannot access the ICP feature, since FFCs are not supported for ISDN BRI; also, the ICP feature cannot be activated on an ISDN BRI terminal from the intercept computer or via a maintenance overlay program, since the ICP feature is activated on a terminal basis;
- an ISDN BRI terminal cannot act as an intercept position, or be allowed to be a default DN for intercept transfer; only Message Center ACD DNs and attendant DNs are allowed to be intercept positions;
- a call originating from an ISDN BRI terminal and terminating on a non-ISDN BRI terminal with ICP active is intercepted according to the ICP configuration; the ISDN BRI terminal display cannot be updated after the call redirection.

Intercept Computer enhancements

Call Forward interaction control

When an intercept transfer is activated from a customer's or tenant's extension, it can be configured that only external calls be forwarded to the external intercept DN (ECDN), while internal calls are forwarded to an answering machine, or the internal intercept DN (ICDN). Note that this capability applies only if the extension's flexible call forward no answer DN (FDN) is not configured as an intercept position.

This feature is partially supported on ISDN BRI terminals, as explained below:

- since an ISDN BRI terminal cannot access the ICP feature, it cannot support this ICP enhancement;
- a call originating from an ISDN BRI terminal that is considered as an internal calling party, and terminating on a non-ISDN BRI having the ICP feature active, is diverted to the intercept DN for internal calls.

IMS/IVMS

Calls originated from an ISDN BRI terminal can be redirected to IMS/IVMS and leave a text or voice message for the called party.

For voice messages, an ISDN BRI terminal that can generate end-to-end signaling can access other options available.

Calls terminated on an ISDN BRI terminal can be redirected to IMS/IVMS through call forward no answer or hunting, but the ISDN BRI DN will not have a mailbox defined and a caller can not leave a message for an ISDN BRI DN.

ISDN BRI Network Ring Again

The Network Ring Again feature cannot be activated from nor offered to an ISDN BRI terminal due to a lack of standardized functional protocol for supporting the feature. For non-ISDN terminals encountering a busy ISDN BRI terminal, the Network Ring Again feature is not offered to the non-ISDN terminals for the same reason.

ISDN QSIG Name Display

Calling Party Privacy (CPP) Flexible Feature Code is not supported on BRI sets. Therefore, Calling/Connected Name Identification Restriction on a per-call basis is not supported on BRI sets.

Malicious Call Trace

An ISDN BRI terminal cannot activate malicious call trace. A non-ISDN BRI terminal can activate malicious call trace on a call connected to an ISDN BRI terminal.

Meridian Mail Trunk Access Restriction

This feature does not support Call Transfer from ISDN BRI sets.

Message Center

A call originating by an ISDN BRI terminal to a non-ISDN BRI terminal can be redirected to Network Message Service - Message Center. The associated MIK/MCK and the lamp states are not supported for the corresponding ISDN BRI terminal.

Multifrequency Compelled Signaling features

R2 and MFC signaling on DID and TIE trunks

This feature allows line and register signaling on DID and TIE trunks, using an MF sender and receiver card. Each forward signal sent to the Central Office is acknowledged by a backward signal. This feature also allows the system to request Calling Number Identification (CNI) information, to be shown on the originating terminal's display.

The register signaling part of this feature is supported for ISDN BRI. The CNI display is not supported.

Multifrequency Signaling for Socotel (MFE)

Multifrequency Signaling for Socotel (MFE) is similar to R2 Multifrequency Compelled (MFC) Signaling, but is not compelled in the same way. Instead of each signal being answered by another signal of some meaning, each signal is answered by a control frequency which indicates to the other end of the call that the signal has been received and its transmission can cease. In this way, signals can originate at either end of the call.

R2 MFC Signaling operates by answering each forward signal from the originating end, with a backward signal from the terminating end. Each signal must be interpreted using the appropriate table. Unlike R2 MFC Signaling, backward and forward signals use the same frequency combinations. The need for the concept of an incoming or outgoing sender/receiver is replaced by an MFE sender/receiver, which can act in both directions during any single call.

Pulsed signals, which are used in R2 MFC Signaling to indicate a message being initiated from the CO which normally replies, are no longer necessary with the new MFE card.

The sequence of messages sent to and received from the CO is transparent to the ISDN BRI terminal.

Multiple Appearance Directory Number

Multiple ISDN BRI terminals on the same DSL can have the same DN. An ISDN BRI and a non-ISDN BRI terminal cannot have the same DN.

ISDN BRI multiple appearance DN is not allowed across different DSLs and is also not allowed for non-ISDN BRI terminals.

Multi-Party Operations

Call Join

The Call Join feature allows a controlling party to conference-in or transfer an active party to a held party. The controlling party can then hang up. The controlling party's terminal must be equipped with a Conference 3/6 key, and at least one secondary DN or Call Waiting key.

ISDN BRI terminals cannot be the controlling party. If the ISDN BRI terminal is the active party or held party, the Call Join feature is supported as described; however, the ISDN BRI terminal display is not updated.

Networking features

International ISDN PRI interworking

The interworking between ISDN BRI and International ISDN PRI Central Office Connectivity allows voice calls to always be completed to and from ISDN BRI terminals; however, some restrictions can apply to data calls.

Advice of Charge for NUMERIS connectivity

This feature provides the total cost for a call made from a system switch to the ISDN access designed for France (NUMERIS). The system to NUMERIS PSTN connectivity is implemented using ISDN packages PRI2 and IPRI.

This feature is supported for ISDN BRI terminals as the Periodic Pulse Metering feature is supported.

Advice of Charge for AXE-10 Australia and Japan D70 connectivity

The Advice of Charge (AOC) at End of Call for AXE-10 Australia and Japan D70 feature supports charge information being sent from an AXE-10 (Australian) or D70 (Japanese) Central Office to the PBX, over an ISDN BRI connection. The information is sent for outgoing calls from the PBX.

Information is received and displayed when the call is taken down. It is displayed on the caller's telephone display as supported 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. Charge Display applies to display-equipped M2317 and Meridian Modular telephones only. The information is also printed as part of the Call Detail Recording (CDR) record.

Australia introduces PBX control of the AOC facility, meaning that the information must be requested for each outgoing call as opposed to expecting it for every call once the feature has been configured. Japan does not support PBX control of the AOC facility.

Message Registration (MR) and Periodic Pulse Metering (PPM) are both packaged under software package 101. Prior to the introduction of this feature, the method used to differentiate which feature was equipped was to check if the International Supplementary Features software package 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 differentiation 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

The AOC feature does not support AOC being sent to ISDN BRI terminals. A meter can be assigned to a DSL. All chargeable calls made by an ISDN BRI terminal on this DSL are charged against the DSL's meter.

Advice of Charge EuroISDN connectivity

This feature provides Integrated Services Digital Network (ISDN) Primary and Basic Rate Interfaces to Central Offices/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 to be supported.

The supplementary services 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. In addition, Advice of Charge for EuroISDN is supported in some of these countries. For more information about Advice of Charge for EuroISDN, see the feature description contained in this document.

The EuroISDN feature also provides interworking with other ISDN or non-ISDN interfaces including Meridian Customer Defined Network (MCDN), QSIG, Digital Private Network Signaling System 1 (DPNSS1), R2 Multifrequency Compelled Signaling (R2 MFC), 2 Mb Digital Trunk Interface (DTI2), and analog trunk interfaces. This feature also provides interworking between NET3 S0 (ETSI compliant BRI sets) and DPNSS2/DASS2 (Digital Access Signaling System 2) links.

The system to New Zealand NEAX-61 ISDN PRI Connectivity

This feature provides an ISDN Primary Rate Interface (PRI) between a system and New Zealand NEAX-61. The design in based on the international 30B + D PRI configuration.

The NEAX-61 interface will support calls from ISDN BRI sets within the MCDN. These will be able to call out to and receive calls from the Public Switched Telephone Network (PSTN), also called the ISDN public network. Operation will be as though the call was a tandem trunk call.

Although ISDN BRI trunks to the NEAX-61 are not supported, any ISDN BRI trunks supported to other interfaces will be able to tandem to the PSTN transparently. Feature operation will appear to the user as though the call was from a normal PRI network trunk to the PSTN.

Night Service improvements

All calls remain queued for night service

This feature allows all calls that are queued for an attendant when night service is entered, to remain queued in order to be presented to the night DN. This feature is supported for ISDN BRI-originated calls; however, the displays on the ISDN BRI terminals are not updated to reflect the call modification.

Automatic timed reminder recalls to the night DN

This feature allows any recall that times out during night service, to be presented or queued to the night DN. This applies to slow answer recall, call waiting and camp-on calls extended by an attendant, and camp-on calls extended by a terminal.

This capability applies to external call only, since there is no internal recall to a night DN unless done by NAS routing.

This feature is partially supported for ISDN BRI. If an external call has been extended an attendant to an ISDN BRI terminal, and the slow answer recall timer or the call waiting recall timer expires before the call is answered, the call is presented or queued to a local or remote night DN. The ISDN BRI terminal is disconnected in the case of the slow answer recall, and the call waiting canceled in the case of the call waiting. The camp-on recall timer does not apply, because Camp-on is not supported for ISDN BRI.

If a call originating from an ISDN BRI terminal has been extended or camped on by the attendant to a non-ISDN BRI terminal, the call is timed for slow answer recall, call waiting recall, or camp-on recall. If the timer expires, the call is dropped if not controlled by NAS routing. The call is also dropped if it is controlled by NAS routing and the night DN is at a remote node. If the call is controlled by NAS routing and the night DN is at the local node, the call is presented or queued to the night DN. When the recall timer expires, the terminating party is disconnected, or the call waiting or camp-on is canceled.

A non-ISDN BRI terminal cannot extend a camp-on to an ISDN BRI terminal, since Camp-on is not supported for ISDN BRI.

Periodic Pulse Metering

This feature, based on the Message Registration feature, allows meters to be assigned to terminals, attendant consoles, trunk routes, and customers. This permits customers to maintain accurate records of Central Office and DOD calls, for billing or administrative purposes. The Call Detail Recording (CDR) feature has been enhanced to record the PPM metering information.

This feature has been adapted for ISDN BRI application as follows. Since meters cannot be assigned to ISDN BRI terminals, because there is no standard protocol to update an ISDN BRI terminal's meter, meters have been assigned to Digital Subscriber Loops (DSLs) instead. All charges due to ISDN BRI terminals located on the same DSL are accumulated on the assigned meter. The contents of this meter can be read or changed from any authorized non-ISDN BRI station.

The metering of conventional trunks is supported transparently. The metering of ISDN BRI trunks is not supported.

Meter recalls to the attendant and background terminal messaging uses the originating ISDN BRI CLID.

Meter charge transfers associated with the activation of Hunt or Call Forward No Answer from an ISDN BRI terminal are not supported; the calling party is charged for the entire call.

Radio Paging

This feature allows call paging through the use of a Flexible Feature Code. The paged party, who is notified of a call page by the buzzing of a special device, can later access the paging party by dialing another FFC. There are three methods of paging a call:

- pre-selection, whereby the paging is performed right away;
- post-selection, whereby a FFC is used in cases where the desired party is busy or does not answer;
- forwarding to a paging equipment.

Paging cannot be done from an ISDN BRI terminal, since FFCs are not supported for ISDN BRI. Paging to an ISDN BRI terminal is supported for the pre-selection and post-selection methods; it is not supported for the forwarding method.

Radio Paging Product Improvements

Attendant Recall over Network

This product improvement enables Radio Paging (RPA) to recall the attendant who originated the Radio Paging call only; the attendant can be located anywhere within an ISDN Meridian Customer Defined Network (MCDN) configured with Network Attendant Services (NAS).

Digit Display to Same Attendant

This improvement enables the attendant's display to be updated 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.

Since ISDN BRI sets do not support FFCs, they cannot be used to access or answer RPA calls if the ISDN BRI sets are local on the paging node. For network situations, ISDN BRI sets can access and answer remote RPA calls. This is possible because the RPAX/RPAN FFCs are dialed as DSC/TSC steering codes.

Attendant Recall over Network is supported over ISDN BRI trunks within an ISDN Meridian Customer Defined Network (MCDN) configured with Network Attendant Services (NAS).

Recorded Announcement for Calls Diverted to External Trunks

Recorded Announcement for Calls Diverted to External Trunks (RANX) provides an optional recorded announcement when the call is being forwarded to external Public Exchange/Central Office (over DTI, DTI2, PRI2, PRI, or analog trunks) or over ISDN BRI trunks connected to AXE-10 or EuroISDN routes. The announcement notifies the calling party that call forwarding is taking place and the call can take longer than usual to set up. The delay depends on the required signaling to reach the destination party.

This feature allows special dial tones to be provided after certain dialed digits.

This feature is generally supported for digits dialed from ISDN BRI terminals. For each TERMINALUP and INFO message, the software determines if tones are required; however, if enblock dialing is used, it cannot be confirmed whether or not some of the messages contain only one digit, or several digits. If they contain several digits, some tones can be skipped.

Station Activity Record

Station activity records are generated for sets with Class of Service Call Detail Monitoring Allowed (CDMA) for all incoming and outgoing calls if Call Detail Recording (CDR) is allowed for the customer. Their format is identical to regular CDR records, but they have a new type identifier (D). Other CDR records are not affected by this new functionality.

Station activity records are supported on ISDN BRI sets with a Class of Service Call Detail Monitoring Allowed (CDMA), for external calls.

Three-Party Service

The Three Party Service feature allows a user of an analog (500/2500-type) terminal with Three Party Service Allowed (TSA) COS to toggle between an active party and a held party on an

ISDN BRI terminal, through the use of the Call Hold feature. However, a three party conference cannot be formed.

AN ISDN BRI terminal can be placed on hold or placed in a three party conference by a controlling terminal that is a non-ISDN BRI terminal; however, the ISDN BRI terminal display is not updated.

Conference 6

The Conference 6 feature is an extension of Three-party service, allowing users of an analog (500/2500-type) terminal, with TSA and C6A COS, the added capability of establishing a conference of up to six parties.

AN ISDN BRI terminal can be placed in a six party conference by a controlling terminal that is a non-ISDN BRI terminal; however, the ISDN BRI terminal display is not updated.

Recovery of Misoperation on Call Transfer

The Recovery of Misoperation of Call Transfer feature prevents external calls from being dropped due to misoperation of the Call Transfer feature. Optional treatments of a misoperation can be configured on a customer basis.

Recovery of Misoperation of Call Transfer is not available on ISDN BRI terminals. If the controlling station misoperates on a transfer of a call originating from an ISDN BRI terminal, the call receives the configured misoperation treatment; however, the ISDN BRI terminal display is not updated.

Network Signaling for Network ACD

ISDN BRI terminals cannot be configured as network ACD DNs; however, ISDN BRI terminals can make calls to the network ACD DNs.

Network Call Redirection Service

ISDN PRI redirection for ISDN BRI terminals is limited as follows:

- ISDN BRI terminals can be redirected across the PRI network, however, the terminal will not be notified about the redirection.
- ISDN BRI terminals can redirect a PRI call using hunting and call forward no answer. It can also redirect a call across PRI the same way.

Network Attendant Service

Network Attendant Service (NAS) provides the capability of stand-alone attendant service, full or part time, across a system network.

NAS provides the following attendant features network wide:

- attendant routing, which allows calls to an attendant to be routed to any other destination within the network;
- night service;
- call extension, which allows calls to an attendant to be extended across a network;
- timed reminder recall, which allows unanswered calls to be recalled to the attendant. When the attendant extends a call to a destination at another node, the trunks are not released. This prevents call disconnection and gives the attendant control of the call, including the ability to exclude the source or destination;
- incoming call indication, which is received when a call reaches the attendant via NAS routing;
- Camp-on and Call Waiting;
- Break-in;
- attendant control, which gives the attendant control of the call, including control of disconnecting the call and the ability to exclude the source or destination;
- ring held party, which allows the attendant console to receive ringing if a terminal that originated a call to the attendant disconnects while attendant control is configured;
- attendant display of Calling Line Identification (CLID), which allows the CLID of the calling party to be displayed on the attendant console.

These features are supported transparently for ISDN BRI, with the following exceptions:

- Camp-on;
- Break-in;
- network wide attendant control; when an ISDN BRI terminal disconnects from an attendant extended call, the call is released on the ISDN BRI side;
- ring held party; when an ISDN BRI terminal disconnects from an attendant extended call, the call is released on the ISDN BRI side.

Note:

The displays on ISDN BRI terminals are never updated.

Network Call Transfer

An ISDN BRI terminal cannot transfer a call, but a call originating from an ISDN BRI terminal can be transferred to an ISDN BRI or a non-ISDN BRI terminal. The display on the ISDN BRI terminal will not be updated.

Night Service/TAFAS

A call originated by an ISDN BRI terminal to an attendant console in night service will be routed through a customer night DN. The display on the ISDN BRI terminal will not be updated to show call redirection.

Priority Override

Priority Override is partially supported on ISDN BRI terminals, as explained below:

- Priority Override to an ISDN BRI terminal is not possible, because Priority Override require Warning Tone Allowed Class of Service (ISDN BRI terminals are defaulted to WTD COS);
- Priority Override from an ISDN BRI terminal is not possible because feature keys and FFC are not supported for ISDN BRI;

Recall to Same Attendant

This feature is supported transparently; however, the displays on ISDN BRI terminals are not updated.

Switched 56 kbps Service

An ISDN BRI terminal can use switched 56 kbps service if a data route has been previously established or the ISDN BRI terminal can generate a 2100 Hz tone to the network. This 2100 Hz tone disables echo cancellation that is provided by the equal access carriers.

ISDN PRI feature interactions with ISDN BRI

All system networking features and services supported by ISDN PRI are also supported by ISDN BRI. The following exceptions apply:

ISDN PRI features not supported

The following ISDN PRI specific features are not supported by ISDN BRI:

- 1.5/2.0 Mb gateway
- 2 Mb gateway
- Automatic Trunk Maintenance
- Backup D-channel
- Channel negotiation - The channel negotiation capability applies at the DSL level. If channel negotiation fails on a route with several ISDN BRI trunks (several DSLs) there is no second call attempt on another DSL in the same route.
- Digital Private Network Signaling No. 1 (DPNSS 1) - ISDN BRI supports a gateway between IDA (DPNSS1 version of PRI) and MCDN protocols for basic call features. Ring again features are not supported.
- In-Service Messaging
- Integrated Service Access (ISA)

- ISDN Primary Rate Access to AXE-10 Sweden Connectivity
- ISDN Primary Rate Access to AXE-10 Australia Connectivity
- ISDN Primary Rate Access to SYS-12 Connectivity
- Integrated Service Digital Network Signaling Link (ISL)
- Non-Associated Signaling Channels (nB+D)
- Trunk Anti-tromboning - supported only over Virtual Network Services BRI trunks
- Trunk Optimization (SL-1 to SL-1 only)
- Virtual Network Services - VNS takes advantage of ISDN signaling through a private D-channel using the Public Exchange voice connections instead of TIE trunk connections. It is not possible to configure a D-channel of an ISDN BRI Trunk Access as a VNS D-channel. However, the voice connection through the Public Exchange of a VNS call can use an ISDN BRI Local Exchange trunk. Also, Trunk Anti-Tromboning is supported on VNS BRI trunks.

Other ISDN PRI features:

- The restrictions and limitations applicable to the remaining ISDN PRI features are similarly applicable to ISDN BRI trunks.

Chapter 10: Australia ETSI

Contents

This section contains information on the following topics:

Feature description

The Australia ETSI feature supports 2.0 Mbit 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 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 PRI and BRI Basic Call Service
- Circuit-mode bearer capabilities (speech, 3.1 kHz audio, 64 Kbit/s digital, and adapted 56 Kbit/s to 64 Kbit/s 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 the systems.

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 can 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 system network is not supported.

The display of charges is not supported on BRI phones 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, the Customer Data Block (LD 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 phone. 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 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 phone basis.

This means that the CLID that is sent from a phone 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 phone can now have DN keys that each has their own CLID. Or, the CLID of any one key on a phone could be programmed to use the CLID of any other key on the phone.

The construction of CLID is based on the CPFXS prompt in LD 16. If CPFXS = NO, then when constructing the Calling Number, the prefixes are retrieved from the Route Data Block through 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 (LD 15) through the entries in the CLID table (refer to the paragraph above for more details).

Also, the system now supports multiple office codes, location codes and steering codes in CLID. This means that any phone on one system can send a CLID that will have calls returned to another system. 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 LD 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 system 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 contain the following:

- 1 -6 digit national code for a home national number (HNTN), which is the equivalent of PFX1
- 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
- 1- 7 digit Home Location Code (HLOC)
- 1 -7 digit Local Steering Code (LSC)

Another capability 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).

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 can 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 system over an Australia ETSI PRI2/BRI interface to a Central Office:

For PRI2 connectivity:

- Integrated Services Digital Network (ISDN) package 145
- 2.0 Mbit/s 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

For ISDN Basic Rate Interface Trunking connectivity:

- Basic Rate Interface (BRI) package 216
- Basic Rate Interface Trunk (BRIT) package 233

Feature implementation

Task summary list

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

15. Table 180: LD 27 - Assign meters to a DSL. on page 355
16. Table 181: LD 16 - Allow Advice of Charge on the route configured for the Australia ETSI. on page 356

Primary Rate Configuration

Table 166: 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		PRI2 loop number
	0-159	For Large Systems

Table 167: LD 17 - Configure the D-channel for ETSI Australian ISDN connectivity.

Prompt	Response	Description
REQ	CHG	Change existing data.
TYPE	ADAN	Action Device and Number.
- ADAN	NEW DCH xx	Add a D-channel on logical port 0-63 Large System and Avaya Communication Server 1000E (Avaya CS 1000E) system.
	NEW DCH xx	Add a D-channel on logical port 0-15; Media Gateway 1000B (Avaya MG 1000B).
- CTYP	MSDL	Multi-purpose Serial Data Link card or Downloadable D-Channel Daughterboard.
- GRP	0-4	Network group number (Large Systems).
- DNUM	0-15	Device number for I/O ports Large Systems. 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.
- PORT		Port number on the MSDL card.
	0-7	For Large Systems
- USR	PRI	This D-channel is used for Primary Rate Interface only.
- IFC	EURO	EuroISDN interface.
- - CNTY	EAUS	Australia ETSI.

Prompt	Response	Description
- - 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	0-159	PRI2 loop number for D-channel.
	1-9	PRI2 loop number for D-channel for Small Systems .
- CNEG		Options for outgoing Channel Negotiation.
	(1)	Option 1: Channel is non-negotiable.
	2	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)	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.
- 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.

Table 168: LD 16 - Configure the ETSI Australian ISDN PRI2 Route Data Block.

Prompt	Response	Description
REQ	NEW	Add new data.
	CHG	Change existing data.
TYPE	RDB	Route Data Block.
CUST	xx	Customer number, as defined in LD 15.
ROUT		Route number
	0-511	Range for Large System and CS 1000E system.
	0-127	Range for Media Gateway 1000B.
TKTP	TIE	TIE trunk type.
	COT	Central Office Trunk type.
	DID	Direct Inward Dialing trunk type.
DTRK	YES	Digital trunk route.
- DGTP	PRI2	2.0 Mbit PRI digital trunk type.
ISDN	YES	Integrated Services Digital Network.
- MODE	PRA	ISDN PRI route.
- IFC	EURO	EuroISDN interface.
- - CNTY	EAUS	Australia ETSI.
ICOG		Incoming and/or Outgoing trunk.
	IAO	The trunk is Incoming and Outgoing.
	ICT	The trunk is Incoming only.
	OGT	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 phones with MCT Class of Service).
- MTND	(NO) YES	(Do not) apply a Malicious Call Trace disconnect delay for tandem calls.

Table 169: LD 14 - Configure the Australia ETSI ISDN PRI2 trunks.

Prompt	Response	Description
REQ	NEW	Add new data.

Prompt	Response	Description
TYPE	CHG	Change existing data. **Note:** Must match TKTP defined in LD 16.
	TIE	TIE trunk data block.
	COT	Central Office Trunk data block.
	DID	Direct Inward Dialing trunk data block.
TN		Terminal number
	l	Loop and channel for digital trunks Large Systems, where: Previously defined PRI2 loops.
	ch	Channel 1-30
	l s c u	Format for Media Gateway 1000B.
CUST	xx	Customer number, as defined in LD 15.
RTMB		Route number and Member Number
	0-511 1-4000	Range for Large System and CS 1000E system.
	0-127 1-4000	Range for Media Gateway 1000B.
TGAR	0 - (1) - 30	Trunk Group Access Restriction The default of 1 automatically blocks direct access.

Note:

The MR package 101 must be equipped on the system.

Table 170: LD 17 - Configure Advice of Charge for Australia ETSI.

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.

Table 171: LD 15 - Allow Charge Display and CDR Charge.

Prompt	Response	Description
REQ:	CHG	Change existing data.

Prompt	Response	Description
TYPE:	PPM	Periodic Pulse Data.
CUST		Customer number
	0-99	Range for Large System and CS 1000E system.
	0-31	Range for Media Gateway 1000B.
OPT	CHDA	Charge Display Allowed.
UCST	(0)-9999	Unit cost for PPM.

Table 172: LD 10 - Assign meters to analog (500/2500-type) phones.

Prompt	Response	Description
REQ:	NEW CHG	Add new data, or change existing data.
TYPE:	500	Analog phone.
TN		Terminal number
	l s c u	Format for Large System, CS 1000E system, Media Gateway 1000B where l = loop, s = shelf, c = card, u = unit.
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.

Table 173: LD 11 - Assign meters to the system proprietary phones.

Prompt	Response	Description
REQ:	NEW CHG	Add new data, or change existing data.
TYPE:	a...a	Telephone type. Type **?** for a list of possible responses.
TN		Terminal number
	l s c u	Format for Large System, CS 1000E system and Media Gateway 1000B, where l = loop, s = shelf, c = card, u = unit.
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.

Prompt	Response	Description
KEY	xx aaa yyy	Phone function key assignments.

Table 174: LD 16 - Allow Advice of Charge on the route configured for Australia ETSI.

Prompt	Response	Description
REQ	NEW	Add new data.
	CHG	Change existing data.
TYPE	RDB	Route Data Block.
CUST	xx	Customer number, as defined in LD 15.
ROUT		Route number
	0-511	Range for Large System and CS 1000E system.
	0-127	Range for Media Gateway 1000B.
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 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 $X*10^{(-Y)}$ 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

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.

Table 175: LD 27 - Define a Link Access Procedure on the D-channel (LAPD) protocol group.

Prompt	Response	Description
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		LAPD parameters.
	(NO) YES	(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.

Table 176: LD 16 - Configure Route Data Block parameters for the ISDN BRI Trunk access capability.

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.

Prompt	Response	Description
ROUT		Route number
	0-511	Range for Large System and CS 1000E system.
	0-127	Range for Small System, Media Gateway 1000B, and Media Gateway 1000T.
TKTP	TIE	TIE trunk type.
	COT	COT Central Office 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.
- 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.

Prompt	Response	Description
...		
- 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 phones with MCT Class of Service).
- MTND	(NO) YES	(Do not) apply a Malicious Call Trace disconnect delay for tandem calls.

Table 177: LD 27 - Configure for a Multi-purpose ISDN Signaling Processor (MISP) for an ISDN BRI trunk.

Prompt	Response	Description
REQ	NEW	Add new data.
	CHG	Change existing data.
TYPE	MISP	Multi-purpose ISDN Signaling Processor.
LOOP	0-158	MISP loop number for Large Systems.
	1-9	MISP loop number for Small Systems.
APPL		Application type.
	BRIE	Enter BRIE for Australia ETSI.
APPL	<CR>	To end configuration procedure.

Table 178: LD 27 - Configure an S/T Interface (SILC) or U-Interface (UILC) line card, for an ISDN BRI trunk.

Prompt	Response	Description
REQ	NEW	Add new data.

Prompt	Response	Description
	CHG	Change existing data.
TYPE	CARD	SILC or UILC configuration.
TN		Card location for Large Systems.
	lll	lll (superloop) = 0-156 (must be an even number, divisible by 4)
	s	s (shelf) = 0-1
	cc	cc (card) = 0-15
		Card location for Small Systems . c (card) = 1-9
MISP		Must be an even loop number that has already been configured.
	0-158	MISP loop number for Large Systems.
	1-9	MISP loop number for Small Systems .
CTYP		**Note:** Remove any DSLs configured for this line card before changing the card type.
	SILC	SILC line card is to be added or changed.
	UILC	UILC line card is to be added or changed.

Table 179: LD 27 - Configure a Digital Subscriber Loop (DSL) for an ISDN BRI trunk.

Prompt	Response	Description
REQ	NEW	Add new data.
	CHG	Change existing data.
TYPE	DSL	Digital Subscriber Loop data block.
DSL	lll s cc dsl#	DSL Terminal Number location Large Systems. 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
		DSL Terminal Number location for Small Systems.
	cc dsl	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.
CTYP	SILC	Interface card type is SILC.
MISP	0-158	MISP loop number for Large Systems
	0-9	MISP loop number for Small Systems .
MODE	TE	Enter TE (user side) as the mode for Australia ETSI.

Prompt	Response	Description
- MTFM	(NO) YES	BRI multiframe option.
TKTP		Must be the same entry as defined in LD 16.
	TIE	TIE trunk type.
	COT	Central Office Trunk type.
	DID	Direct Inward Dialing trunk type.
CLOK	(NO) YES	(Do not) use the DSL as the clock source.
PDCA	(1)-16	Pad table number.
ROUT		Route number **Note:** Both B-Channels must belong to the same route.
	0-511	Range for Large System and CS 1000E system.
	0-127	Range for Small System,Media Gateway 1000B, and Media Gateway 1000T.
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 Large Systems.

Assign meters to a DSL (this step is required for Advice of Charge). The MR/PPM package 101 must be equipped on the system.

Table 180: LD 27 - Assign meters to a DSL.

Prompt	Response	Description
REQ	NEW	Add an ISDN protocol group
TYPE	DSL	LAPD Protocol group
DSL	lll s cc dsl#	DSL Terminal Number location Large Systems. 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
		DSL Terminal Number location for Small Systems.
	cc dsl	cc (card) = 1-20 dsl# (DSL number) = 0-7
CLS	MRA	Allow Message Registration on the DSL.

Table 181: LD 16 - Allow Advice of Charge on the route configured for the Australia ETSI.

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.
ROUT		Route number
	0-511	Range for Large System and CS 1000E system.
	0-127	Range for Small System, Media Gateway 1000B, and Media Gateway 1000T.
TKTP	TIE	TIE trunk type.
	COT	Central Office 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.
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	Define AOC at call set-up.
	DURC	Define AOC during the call.
	ENDC	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 $X*10^{(-Y)}$ 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.

Feature operation

No specific operating procedures are required to use this feature.

Chapter 11: BRI/PRI Basic Call Interworking

Contents

This section contains information on the following topics:

Feature description

BRI/PRI Basic Call Interworking provides data connectivity between ISDN BRI and ISDN PRI.

Basic Call Interworking does the following:

- allows better high- and low-level compatibility checking between the calling and terminating equipment
- supports the V.120 protocol between BRI TEs over PRI
- supports a greater range of Bearer Capability, which is the network data transmission rate
- allows end users to support many terminals on the same BRI DSL, such as Group IV fax, data monitor
- propagates existing IEs with existing encodings over tandem PRIs between BRI TEs

The affected IEs are:

- Bearer Capability BRI and PRI propagate octet 4ab without modification.
- Called party subaddress The system decodes and saves the called party subaddress when it is received from PRI, passing it to the terminating BRI. The system also sends the subaddress to PRI when the originating BRI or PRI includes it.
- Calling party subaddress The system decodes and saves the calling party subaddress when it is received from PRI, passing it to the terminating BRI. The system also sends the subaddress to PRI when the originating BRI or PRI includes it.
- Cause BRI and PRI propagate octet 4 without modification.
- High layer compatibility The system decodes and saves high-layer compatibility information received from PRI and passes it to the terminating BRI. The system also sends the information element, without interpreting it, to PRI after receiving it from the originating BRI or PRI.
- Low layer compatibility The system decodes and saves low-layer compatibility information received from PRI and passes it to the terminating BRI. The system also sends the information element, without interpreting it, to PRI after receiving it from the originating BRI or PRI.

BRI/PRI supports these interfaces:

- Meridian 1 PRI
- Japan D70 PRI

Operating parameters

New IEs and IEs with new encodings are only supported when the RCAP is configured in LD 17. Existing IEs and encodings are supported end-to-end regardless of the RCAP value.

A call with new Bearer Capability encodings will only terminate to a BRI terminal. If the terminating terminal is not a BRI terminal, the call is blocked.

Voice calls are successful between BRI and PRI with no restrictions.

Feature interactions

The following feature interactions are unique to BRI/PRI basic data call interworking.

- ISDN BRI Data Call Added IEs (such as LLC and HLC) and the expanded set of supported data values for the Bearer Capability IE enhance BRI's ability to support a variety of circuit-switched data calls.
- ISDN PRI D-channel Error Reporting and Monitor The DCH table supports the new IEs in the PRI call messages. The DCH monitor displays the new IEs and a label for monitor level 2.
- Incoming Digit Conversion If an incoming SETUP message with the new Bearer Capability encodings goes through incoming digit conversion, it must be translated to a BRI DN. If the terminating DN is not a BRI DN, the call will be blocked.
- Network Alternate Route Selection If an outgoing call contains a SETUP with a new Bearer Capability encoding, it can only be terminated on PRA/ISL trunks with RCAP configured. This means that at least one entry in the RLB must be PRA/ISL and have the DCH RCAP configured for BRI. NARS will continue to search the RLB until it finds RCAP = BRI. An outgoing call is blocked if RCAP is not configured, or if all trunks with RCAP configured are busy.

Feature packaging

BRI/PRI, as a feature, has no packaging requirements. However, the requirements for ISDN BRI and ISDN PRI must be met.

Feature implementation

Task summary list

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

1. Table 182: LD 17 - Configure far-end BRI support. on page 362
2. Table 183: LD 22 - Print the configuration record. on page 362

Table 182: LD 17 - Configure far-end BRI support.

Prompt	Response	Description
REQ	CHG	Change existing data.
TYPE	CFN	Configuration record
ADAN	CHG DCH X	
..		
RCAP	BRI	Add far-end BRI support.
	XBRI	Remove far-end BRI support. Valid only for IFC=SL1, D70, ESS4, ESS5

Table 183: LD 22 - Print the configuration record.

Prompt	Response	Description
REQ	PRT	Print system data
RCAP	BRI	Far-end BRI support Valid only for IFC=SL1, D70, ESS4, ESS5

Feature operation

This feature operates in the background according to how the BRI TEs are configured and the RCAP value in LD 7.

Chapter 12: BRI Trunks with Night Service Enhancement

Contents

This section contains information on the following topics:

Feature description

The system supports ISDN Basic Rate Interface (BRI).

ISDN BRI consists of two 64Kb/s Bearer (B) channels, and one 16Kb/s Data (D) channel. The BRI interface is referred to as a 2B+D connection as well as a Digital Subscriber Loop (DSL).

B-channels transmit user voice and data information at high speeds, while D-channels are packet-switched links that carry call set-up, signaling and other user data across the network.

One single DSL can carry two simultaneous voice or data conversations, to the same or to different locations. In either case, the D-channel can also be used for packet communication to a third location simultaneously. The two B-channels can also be combined for transmitting data at non-compressed speeds of up to 128 Kb/s.

A wide range of devices and telephone numbers can be associated with a single DSL, to offer equipment flexibility and reductions in line, wiring and installation costs.

Night Service Enhancement for BRI Trunks

Night Service Enhancement for BRI Trunks enhances the functionality of the Night Service feature. Night Service allows calls that are normally directed to the attendant to be routed to another defined destination. With the Night Service Enhancement for BRI Trunks feature, a Night Service DN (NITE) or Night Service Group number (NGRP) can be defined for BRI trunks.

The Night Service Enhancement for BRI Trunks feature introduces the NITE and NGRP prompts for Overlay 27.

The NITE prompt appears in Overlay 27 when the following conditions apply:

- Enhanced Night Service is disabled (ENS = NO) in Overlay 15.
- Auto Terminate is disabled (AUTO = NO) in Overlay 16.
- The BRI trunk is defined as a Direct Inward Dialing (DID) or Central Office (COT) trunk in Overlay 27.

The NGRP prompt appears in Overlay 27 when the following conditions apply:

- Enhanced Night Service is enabled (ENS = YES) in Overlay 15.
- Auto Terminate is disabled (AUTO = NO) in Overlay 16.
- The BRI trunk is defined as a DID or CO trunk in Overlay 27.

The Auto Terminate DN (ATDN) prompt appears in Overlay 14 when the following conditions apply:

- Auto Terminate is enabled (AUTO = YES) in Overlay 16.
- The BRI trunk is defined as a DID, CO, or TIE trunk in Overlay 27.

Note:

In this case, neither the NITE prompt nor the NGRP prompt appears in Overlay 27.

Operating parameters

BRI connectivity supports two interface standards:

- S/T Interface Line Card (SILC)
- U Interface Line Card (UILC)

The S/T Interface Line Card (SILC) and the U Interface Line Card (UILC) can reside in Media Gateway Expander.

With the Night Service Enhancement feature, you can enter a Group Hunt Pilot DN as the Night Service DN.

You cannot assign a BRIL DN to the night station.

The Night Service DN defined in Overlay 27 takes precedence over the Customer Night DN defined in Overlay 15.

The NITE or NGRP prompt appears for each B-channel. You can enter different Night DNs for each B-channel.

If you enter B2 = NO for the configuration of the second B-channel, the Night DN or Night Service Group number that you entered for the first B-channel (BI) is used.

If the definition for the ENS prompt is changed from NO to YES while Night Service is in effect, the system verifies that the Night number defined is a group number or a DN. If a Night DN or 0000 is defined, the existing Night DN, defined in Overlay 15, is used.

Feature interactions

For line access, BRI is compliant with CCITT, ANSI, ETSI NET-3 (EuroISDN), INS NET-64 (Japan D70), National ISDN-1, 1TR6, Numeris VN2, and EuroISDN standards.

BRI trunks connect to local exchanges that support Numeris VN3, 1TR6, ETSI NET-3 (EuroISDN), INS NET-64 (Japan D70), and Asia-Pacific protocols.

For Night Service and Night Service Enhancements feature modules refer to *Avaya Features and Services Fundamentals* (NN43001-106) for feature interactions.

Feature packaging

This feature requires the following packages:

- Enhanced Night Service (ENS) package 133, in order for the NGRP prompt to appear
- Integrated Services Digital Network (ISDN) package 145
- Primary Rate Access (PRA) package 146
- 2.0 Mb/s Primary Rate Interface (PRI2) package 154
- International Primary Rate Access (IPRA) package 202
- Basic Rate Interface (BRI) package 216

- Integrated Services Digital Network Basic Rate Interface Trunk Access (BRIT) package 233
- Basic Rate Interface Line Application (BRIL) package 235

Feature implementation

Task summary list

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

1. Table 184: LD 27 - Define a Night Service Group number when ENS is disabled in Overlay 15. on page 366
2. Table 185: LD 27 - Define a Night Service Group number when ENS is enabled in Overlay 15. on page 367

Table 184: LD 27 - Define a Night Service Group number when ENS is disabled in Overlay 15.

Prompt	Response	Description
REQ	CHG	Change existing data.
TYPE	DSL	Digital Subscriber Loop Data Block
DSL	c dsl	Digital Subscriber Loop address
APPL		Application type for this DSL
	BRIE	Basic Rate Interface protocol engine BRIE supports the QSIG and EuroISDN interfaces, and requires BRIT package 233. Any changes in the DSL route must match the BRIE loadware application.
	BRIT	Basic Rate Interface Trunk BRIT supports SL-1, Numeris, and ITR6 interfaces. BRIT package 233 is required.
TKTP		Trunk Type
	DID COT	Direct Inward Dialing trunk type Central Office trunk type
PRID	1 - 4	Protocol ID
PDCA	1 - 16	Pad Category table, as defined in Overlay 73
ROUT		Route number
	0-511	Range for Large System and Avaya Communication Server 1000E (Avaya CS 1000E) system.

Prompt	Response	Description
	0-127	Range for Media Gateway 1000 B (Avaya MG 1000B).
B1	YES	Change B-channel 1 configuration
NITE	x...x	Night Service Directory Number This DN can have up to four digits, or up to seven digits with Directory Number Expansion (DNXP) package 150. You can enter a Group Hunt Pilot DN at this prompt. The DN that you enter here takes precedence over the NITE and NIT1-NIT4 prompts in LD 15. If you enter a DN at this prompt, calls will go to this DN. If you do not enter a DN at this prompt, calls will go to the DNs defined at the NITE prompts in LD 15.
TGAR	0-(1)-31	Trunk Group Access Restriction number
...		
B2	YES	Change B-channel 2 configuration
NITE	x...x	Night Service Directory Number This DN can have up to four digits, or up to seven digits with Directory Number Expansion (DNXP) package 150. You can enter a Group Hunt Pilot DN at this prompt. The DN that you enter here takes precedence over the NITE and NIT1-NIT4 prompts in LD 15. If you enter a DN at this prompt, calls will go to this DN. If you do not enter a DN at this prompt, calls will go to the DNs defined at the NITE prompts in LD 15.

Table 185: LD 27 - Define a Night Service Group number when ENS is enabled in Overlay 15.

Prompt	Response	Description
REQ	CHG	Change existing data.
TYPE	DSL	Digital Subscriber Loop Data Block
DSL	c dsl	Digital Subscriber Loop address
APPL		Application type for this DSL
	BRIE	Basic Rate Interface protocol engine BRIE supports the QSIG and EuroISDN interfaces, and requires BRIT package 233. Any changes in the DSL route must match the BRIE loadware application.
	BRIT	Basic Rate Interface Trunk BRIT supports SL-1, Numeris, and ITR6 interfaces. BRIT package 233 is required.
...		
TKTP		Trunk Type

Prompt	Response	Description
	DID COT	Direct Inward Dialing trunk type Central Office trunk type
...		
B1	YES	Change B-channel 1 configuration.
NGRP	0 - 9	Night Service Group number
		The NGRP prompt replaces the NITE prompt when ENS = YES in LD 15.
TGAR	0-(1)-31	Trunk Group Access Restriction number
...		
B2	YES	Change B-channel 2 configuration.
NGRP	0 - 9	Night Service Group number
		The NGRP prompt replaces the NITE prompt when ENS= YES in LD 15.

Feature operation

Refer to the Night Service and Night Service Enhancements feature modules in *Avaya Features and Services Fundamentals* (NN43001-106) for Night Service and Night Service Enhancements feature operation.

Chapter 13: Business Network Express/ EuroISDN Call Diversion

Contents

This section contains information on the following topics:

Applicable regions

ISDN BRI trunking is not available in North America. Contact your system supplier or your Avaya representative to verify support of this product in your area.

Feature description

Business Network Express (BNE) refers to a group of EuroISDN network functionalities. The BNE capabilities provide systems connected on a EuroISDN public network with the following functionalities:

- EuroISDN Call Completion
- EuroISDN Name and Private Number Display
- EuroISDN Call Diversion
- EuroISDN Explicit Call Transfer

BNE provides a Virtual Private Network (VPN) solution for the systems through the EuroISDN public network. BNE is appropriate for companies that require a network that operates like a private network, but has a lower initial cost. The Virtual Network Services (VNS) solution provides more features than BNE (VNS is a version of the ISL interface). However, VNS requires a leased line for the D-channel between the systems.

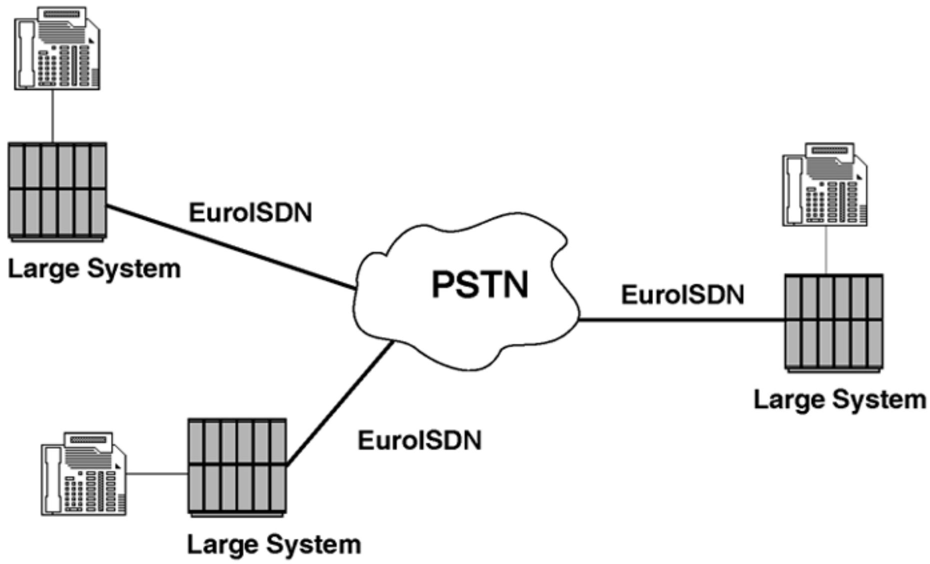


Figure 31: Example of a network where BNE is useful

This chapter provides information about the EuroISDN Call Diversion aspect of the BNE package.

For information about the other BNE features, refer to the *Business Network Express/ EuroISDN Name and Private Number Display and Business Network Express/EuroISDN Explicit Call Transfer* feature modules in this book.

The Call Diversion supplementary services that are compliant with EuroISDN standard EN 300 207-1 include the following:

- Call Forwarding Unconditional (CFU), known as Call Forward All Calls
- Call Forwarding Busy (CFB), known as Hunt
- Call Forwarding No Reply (CFNR), known as Call Forward No Answer

Refer to Table 189: Correspondence between the ETSI reason for diversion names and the system features on page 387 for a complete list of the equivalent features on the system supported by this feature.

This chapter uses the terms served user or served phone. These terms refer to the phone that is diverting calls to another phone in the network. Figure 32: Call Diversion environment on page 372 shows the component parts and terms used in the Call Diversion environment.

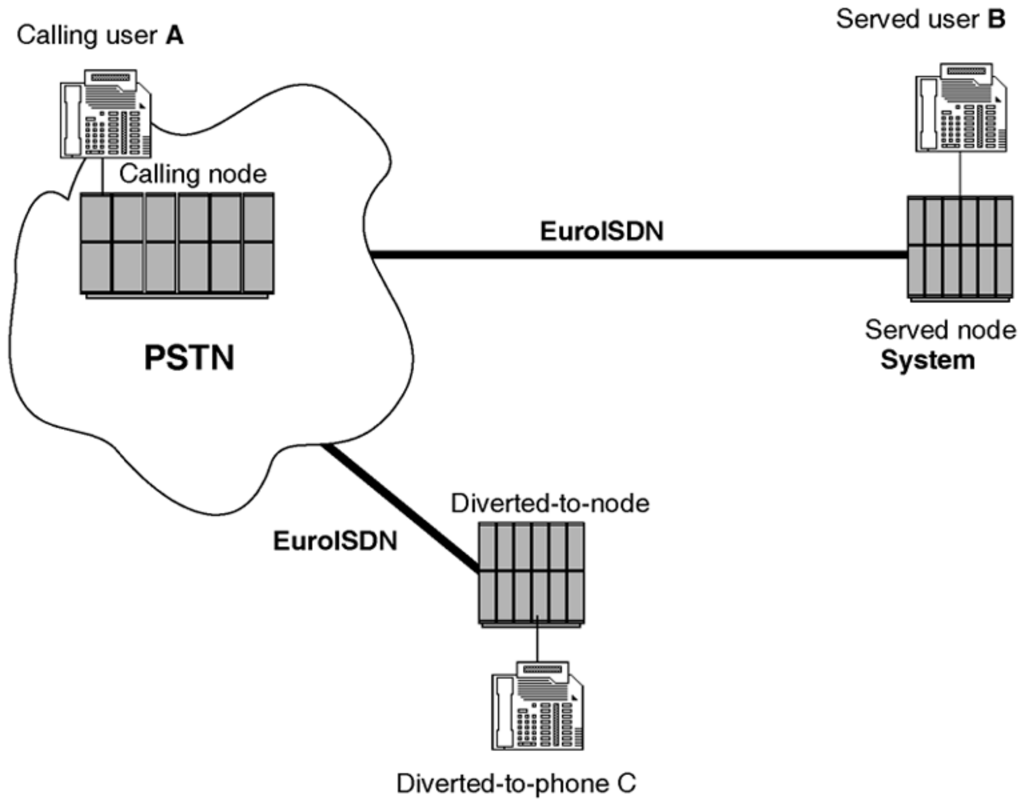


Figure 32: Call Diversion environment

Redirection services

The CFU service enables the network to redirect all calls, addressed to the ISDN number of a served phone, to another phone in the network. The CFU supplementary service does not affect the served user's ability to originate calls. The network forwards calls, independent of the status of the served phone, when the CFU supplementary service is active.

The CFB service enables the network to redirect calls, addressed to the busy ISDN number of a served phone, to another phone in the network. The CFB service does not affect the served user's ability to originate calls.

When a call is not answered (for a defined period of time) at an ISDN number of a served phone, the CFNR service enables the network to redirect calls to another phone in the network. The CFNR supplementary service does not affect the served user's ability to originate calls.

Rerouting

The public EuroISDN network performs Call Rerouting. Rerouting is a network routing algorithm that performs call diversion by replacing the connection from user A's node (located in the public ISDN), to user B's node (located in a private ISDN), with another connection from user A's node to user C's node (located in the public ISDN). The new connection is established in the public ISDN by joining together the original connection from user A's node to the public ISDN gateway node and a second, new connection from the public ISDN gateway node to user C's node.

The system sends a Rerouting request to tell the network that it must reroute the call. This feature controls only the EuroISDN user side (the system side).

The following EuroISDN interfaces are supported:

- ETS 300 102 compliant
- EN 300 403 compliant

Class of Service

The Class of Service of the served phone affects the notification that the calling phone and the final destination phone receive. Table 186: Relationship between the Class of Service of the served user and the notification of the calling user on page 373 and Table 188: Relationship between the served user Class of Service and the notification on the final destination (diverted-to) phone on page 374 shows a summary of the relationships.

Table 186: Relationship between the Class of Service of the served user and the notification of the calling user

Served user's Class of Service	Calling user's notification
DN01	No notification
DN02	Notification without diverted-to-number
DN03	Notification with diverted-to-number (default)

The diverted-to-number displays on the calling user's phone under the following conditions:

- the information received indicates to allow presentation
- the served user's Class of Service (received within the Diversion Notification Information from the served node) allows presentation

Table 187: Examples of calling user display related to the served user Class of Service on page 374 provides examples of the display on the calling user's phone under different conditions.

Table 187: Examples of calling user display related to the served user Class of Service

	Calling user display	
Class of service of served phone: "Calling user receives notification that call has been diverted"	**after receipt of served phone's diversion notification information**	**after receipt of diverted-to-phone's diversion notification information**
No	0164665000	0164665000
Yes without diverted-to-number	0164665000 F	0164665000 F
Yes with diverted-to-number	00164665000 F	0164666000 F

Table 188: Relationship between the served user Class of Service and the notification on the final destination (diverted-to) phone

Served user's Class of Service	Diverted-to-phone receives
DNDN	Served phone number not shown
DNDY	Served phone number shown (default)

Diversion reason codes appear on the calling user's phone and the diverted-to-phone, if:

- they are programmed at their correct nodes in LD 95, and
- the Class of Service of the served phone allows it.

The redirection code displays when the phone receives the Diversion Notification Information from the served node.

Multiple diversions

For the purpose of discussion, assume that the following events occur: Originating user A calls B1. B1 has activated CFU, CFB or CFNR to B2. B2 has activated diversion to B3. B3 has activated diversion to the next phone. Call diversion continues until phone Bn activates diversion to phone C. The user at phone C answers.

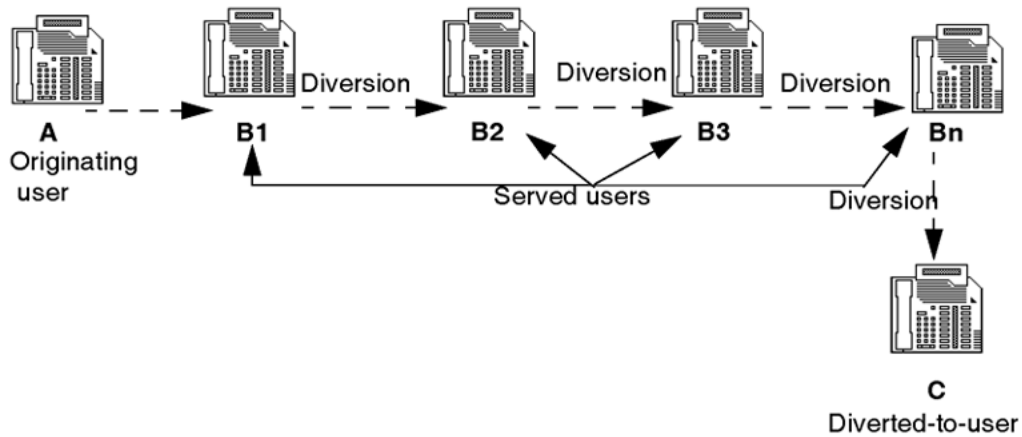


Figure 33: Example of multiple diversions

Identification of the diverted-to-user to the calling user

Diversion Reason Notification rules:

The last diversion reason replaces the previous one.

Diverted-to-Number Notification rules:

When diversion first occurs, and for each diversion following, the system receives the following information from the served node (public ISDN):

- the Class of Service setting related to if "notify the calling user of diversion"
- the reason for redirection code
- the diverted-to-number

The system presents the diverted-to-number to the calling user if both the following conditions exist:

- any Class of Service information related to "notify the calling user of diversion" received contains the value "Yes with diverted-to number"
- any diverted-to-number information (the presentation indicator of the Redirecting Number IE) received allows the display of the diverted-to-number

The last diverted-to-number replaces the previous one.

Notification at the diverted-to-user

Diversion Reason Notification rules:

The rules are identical to that of a single diversion case.

Served Number Notification rules:

No served user number displays because of the digital phone display limits.

Procedures for interworking with private ISDNs

A call from the public ISDN is diverted by rerouting

Figure 34: Rerouting takes place in the public network on page 376 shows an example of interworking. In the figure, calling user A in the public network makes a call through a EuroISDN link to the served user B on a system. The call is forwarded through a EuroISDN link to the diverted-to-user C in the public network.

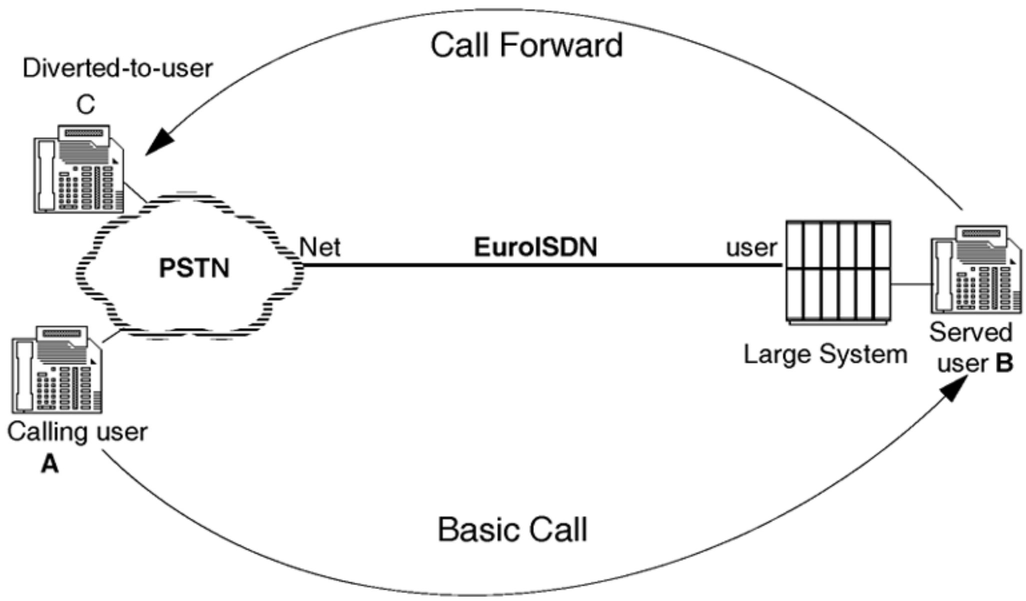


Figure 34: Rerouting takes place in the public network

The Rerouting request sent by the system contains the following information:

- the rerouting reason. For multiple diversions within the same node, the rerouting reason is the reason for the last diversion.
- the diverted-to-address
- the number of diversions
- the embedded Bearer Capability IE, and if available, the Low Layer Compatibility IE or User-to-User IE information

- the served user, or the last served user number when there are multiple diversions. The system also sends presentation information. The presentation information can be one of the following values:
  - "PresentationAllowedNumber" if the served user's CLS = DNDY
  - "PresentationRestricted" if the served user's CLS = DNDN
  - "NumberNotAvailableDueToInterworking" - if the gateways perform this service
- the calling party subaddress, if available
- the calling user notification information, depending on the served user's Class of Service (see Table 186: Relationship between the Class of Service of the served user and the notification of the calling user on page 373).

A call from the public ISDN is diverted within the system

Figure 35: Call from the public ISDN is diverted within the system on page 377 shows an example of interworking. Calling user A, in the public network, calls through a EuroISDN link to the served user B on a system. The call forwards to the diverted-to-user C on the same system.

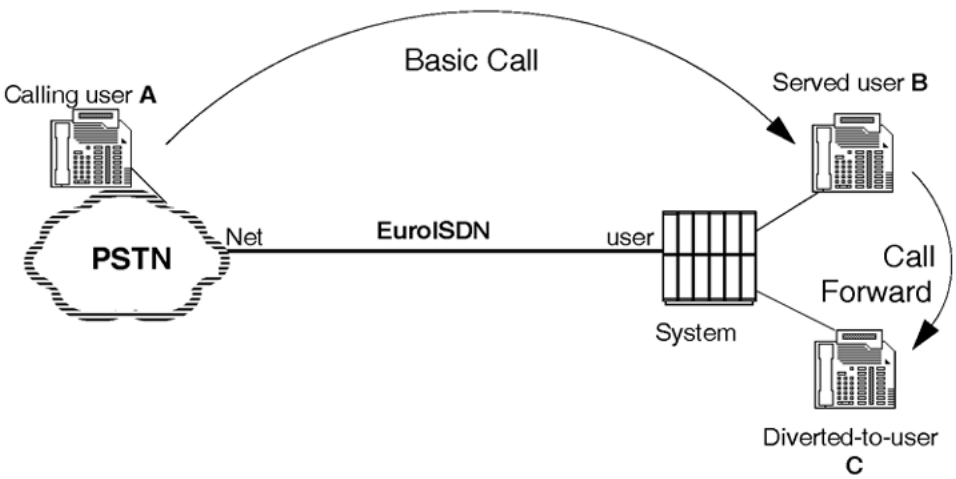


Figure 35: Call from the public ISDN is diverted within the system

When the call is diverted to a phone on the same system, a FACILITY message, containing a DivertingLegInformation1, is sent by the system to the public ISDN. This DLI1 component contains the following information:

- the diversion reason
- the calling user notification information that depends on the served user's Class of Service (see Table 186: Relationship between the Class of Service of the served user and the notification of the calling user on page 373)
- the diverted-to user's number

For CFU or CFB, when the diverted-to-phone starts ringing, an ALERT message that includes Diversion Notification Information DLI3 (diverted-to-number-presentation-indicator) is sent back from the diverted-to-node to the public ISDN.

For CFNR, a second FACILITY message including Diversion Notification Information DLI3 (diverted-to number-presentation-indicator) is sent back from the diverted-to-node to the public ISDN.

Presentation of a call that is diverted within the public ISDN to the system

Figure 36: The calling and served users are in the public network and the diverted-to-user is on the system on page 378 shows an example of interworking. In this example, the calling user A in the public network makes a call to the served user B who is also in the public network. The call forwards to the diverted-to-user C on a system through the EuroISDN link.

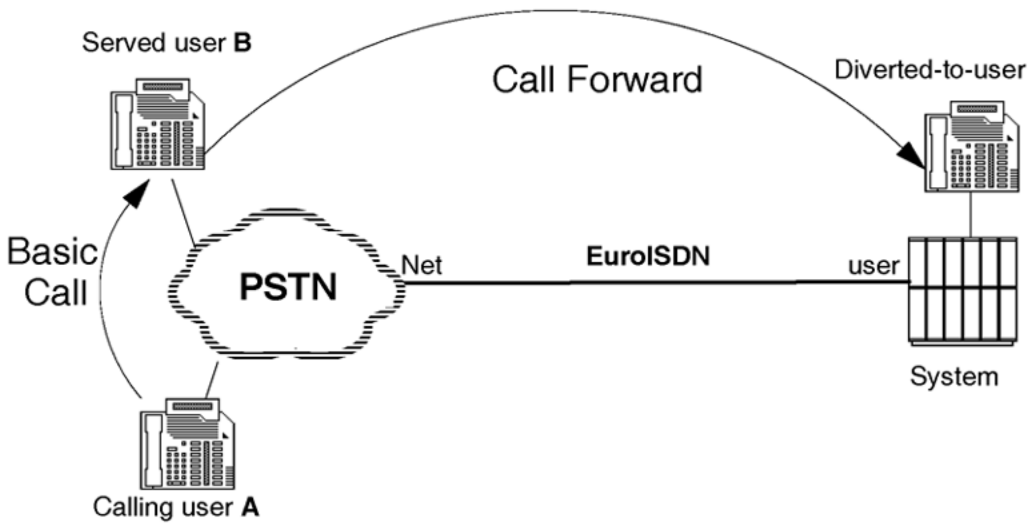


Figure 36: The calling and served users are in the public network and the diverted-to-user is on the system

When the system receives a DivertingLegInformation 2 (DLI2) APDU, the diverted-to-user's display shows the diversion reason given in the DLI2 APDU and the calling user's number (even if the last served user's number is present in the DLI2 APDU), if presentation is allowed. No served user number displays due to the digital phone display limitations. A DLI3, containing the Presentation Indictor of the diverted phone, is also sent to the CO in an ALERT message for CFU/CFB or in a FACILITY message for CFNR.

Presentation of a diverted call from the system to the public ISDN

Figure 37: The calling and served users are on the system and the diverted-to-user is in the public network on page 379 illustrates an example of interworking. In this illustration, the calling

user A on a system makes a call to the served user B on the same system. The call is then forwarded through a EuroISDN link to the diverted-to-user C in the public network.

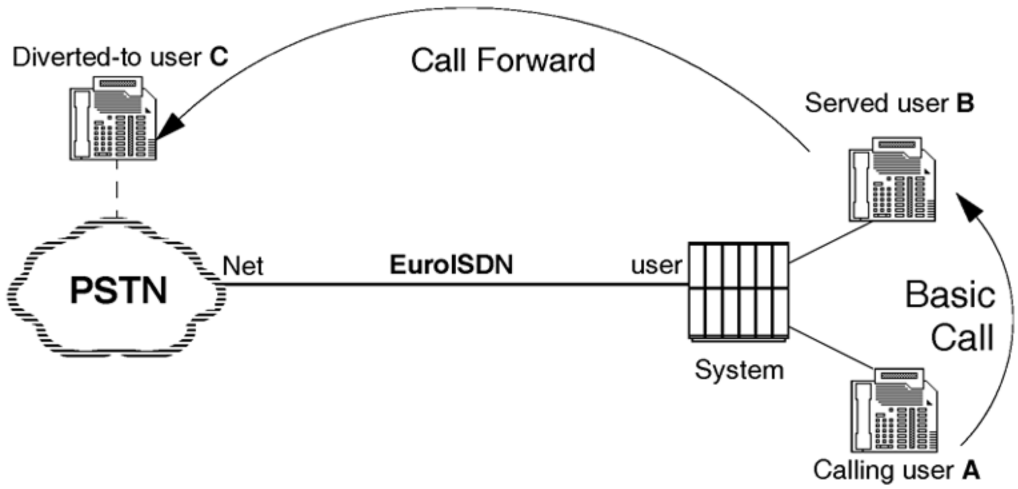


Figure 37: The calling and served users are on the system and the diverted-to-user is in the public network

When a call is forwarded by a served user B to the diverted-to-user C in the public network, a SETUP message including a DivertingLegInformation 2 (DLI2) APDU is sent to the public network. This APDU contains the following information:

- diversionCounter—the number of times the call has been diverted
- diversionReason—the reason associated with the last diversion
- diversionNr—the ISDN number of the last served user (depending on the served user's Class of Service; see Table 188: Relationship between the served user Class of Service and the notification on the final destination (diverted-to) phone on page 374)
- originalCalledNr—the ISDN number of the first served user (depending on the served user's Class of Service; see Table 188: Relationship between the served user Class of Service and the notification on the final destination (diverted-to) phone on page 374)

A DLI3 message can be received from the public ISDN in a FACILITY or an ALERTING or a CONNECT message. The presentationAllowedIndicator affects whether the phone displays the diverted-to-ISDN-user's number.

A call from the system is diverted within the public ISDN

Figure 38: The calling user is on the system and the served user and the diverted-to-users are in the public network on page 380 illustrates an example of interworking. In this illustration, the calling user A, on a system, makes a call through a EuroISDN link to the served user B in the public network. The call forwards to the diverted-to-user C who is also in the public network.

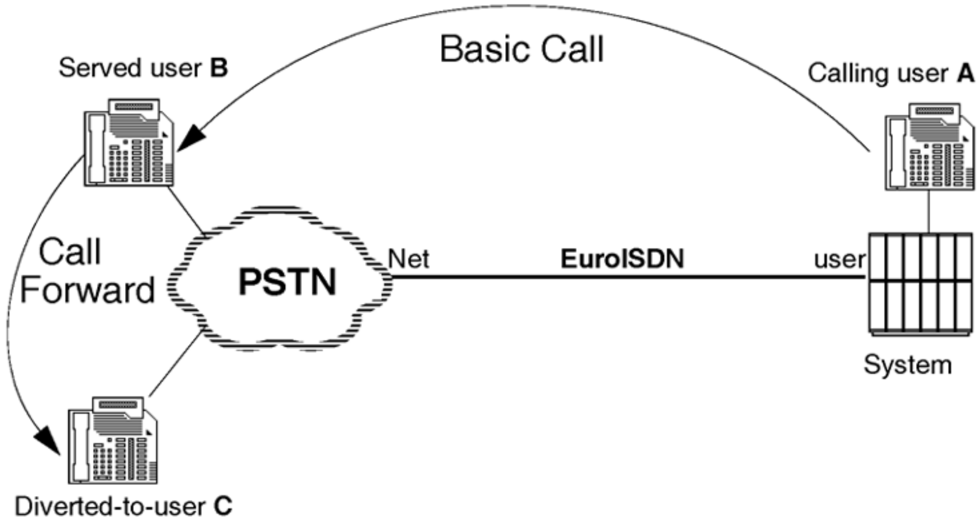


Figure 38: The calling user is on the system and the served user and the diverted-to-users are in the public network

The calling user displays the diversion reason and the diverted-to-number when both of the following events occur:

- after receiving a DLI1 message (in a FACILITY or a PROGESS or an ALERTING message)
- a DLI3 message (in a FACILITY or an ALERTING or a CONNECT message)

The calling user display depends on the received value of the Class of Service related to the "calling user is notified of diversion" and the presentationAllowedIndicator.

The PSTN can send a Notification Indicator IE and Redirection Number IE instead of DLI1 and DLI3. If this happens, the display on the originating phone is updated when the system receives these IEs, the same way it does when it receives DLI1 and DLI3 messages.

QSIG, MCDN and DPNSS Gateways

The BNE/EuroISDN Call Diversion feature allows notification to occur when the private network is a multi-node network using the following protocols (see figure Figure 39: Interworking with EuroISDN, QSIG, MCDN, or DPNSS on page 381):

- QSIG
- DPNSS
- MCDN

The notification of the originating, and diverted-to user, depends on the different protocols involved at various stages of the call establishment, and on the diversion specifications for the protocols.

For example, in case of a EuroISDN/DPNSS gateway, the presentation information is not mapped, since it is not supported by the DPNSS protocol.

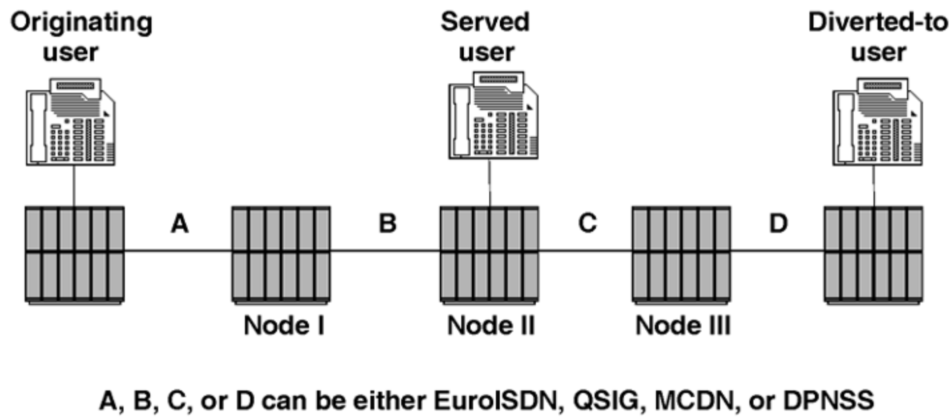


Figure 39: Interworking with EuroISDN, QSIG, MCDN, or DPNSS

Node I Gateways (A/B)

The following gateways can exist between the Originating user node and the Served user node (refer to Figure 39: Interworking with EuroISDN, QSIG, MCDN, or DPNSS on page 381):

- QSIG/EuroISDN and EuroISDN/QSIG

  Because the messaging is the same for both protocols, the DLI1 and DLI3 APDUs are sent in the same message format as they were received in.

- MCDN/EuroISDN

  The Originating user is notified as soon as the Diversion information (DLI1 and DLI3) have been received at the gateway. DLI3 can be received either in an ALERTING, a FACILITY, or a CONNECT message. With the CONNECT message, the originating user is notified only at connection.

- EuroISDN/MCDN

  All diversion information is received at the gateway node in one message (NOTIFY) on the MCDN link. This information is mapped into the two Diversion information elements: DLI1 and DLI3, which are sent in two facility messages.

- DPNSS/EuroISDN

  The redirection information, DLI1 and DLI3, are both received on the EuroISDN link, but only DLI1 is mapped since Presentation Indication is not supported on DPNSS.

In the case of Call Forward Unconditional or Call Forward Busy, the redirection information is mapped into a NAM message. Call forward No Reply is mapped into an EEM message.

If the received DLI3 on EuroISDN indicates that presentation is restricted, an empty (without digit) DVD is sent on DPNSS.

- EuroISDN/DPNSS

  The diversion information (if available) received at the gateway on the DPNSS link is contained in the NAM message (for CFU and CFB) or the EEM message (for CFNR).

  Since DPNSS does not support Presentation Indication, all diversion information received on the DPNSS link is mapped into DLI1 information and sent on the EuroISDN link within a FACILITY message. For DLI3, the default value (Presentation allowed) is sent within either a FACILITY message (for CFNR) or ALERTING message (for CFU or CFB).

Node II Gateways (B/C)

The following gateways can exist at the Served user node (refer to Figure 39: Interworking with EuroISDN, QSIG, MCDN, or DPNSS on page 381):

- QSIG/EuroISDN and EuroISDN/QSIG

  The call establishment message (SETUP) and the FACILITY message, with DLI1 APDU, are not impacted by this type of gateway and are sent with the diversion information.

  The sending of DLI3 APDU on the originating interface is done within the same message as it was received on the terminating interface (ALERTING, FACILITY, or CONNECT).

- MCDN/EuroISDN

  The call establishment message (SETUP) is not impacted by the gateway and is sent with the Diversion information.

  Once the gateway receives the DLI3 Diversion information, it propagates the redirection information by sending a NOTIFY message on the MCDN link. The information can be received in an ALERTING, FACILITY, or CONNECT message, depending on the network structure and on multiple redirection. Therefore, in a case where the DLI3 information is received in a CONNECT message, instead of an ALERTING message, the Originating user is notified at the connection.

- EuroISDN/MCDN

  The call establishment message (SETUP) is not impacted by the gateway and is sent with the Diversion information.

  DLI1 and DLI3 are both sent in a FACILITY message. The FACILITY message sending is triggered by the reception of the NOTIFY message on MCDN.

- DPNSS/EuroISDN

The call establishment message (SETUP) is not impacted by the gateway and is sent with the Diversion information.

In case of CFU and CFB, the DVD string is included into the NAM message. Otherwise, for CFNR, the DVD string is included into an EEM message.

If the received DLI3 on EuroISDN indicates that presentation is restricted, an empty DVD is sent on DPNSS.

- EuroISDN/DPNSS

  The call establishment message (ISRM) is not impacted by the gateway and is sent with the Diversion information.

  The FACILITY message carrying the DLI1 information is sent as soon as the diversion occurs.

  The DLI3 information is sent in the ALERTING message for CFU and CFB, and in a FACILITY message for CFNR.

  Since Presentation Indication is not supported on DPNSS, default DLI3 information (presentation allowed) is sent on the EuroISDN link.

Node III Gateways (C/D)

The following gateways can exist between the Served user node and the Diverted-to user node (refer to Figure 39: Interworking with EuroISDN, QSIG, MCDN, or DPNSS on page 381):

- QSIG/EuroISDN and EuroISDN/QSIG

  Because the messaging is the same for both protocols, the DLI2 and DLI3 APDUs are sent in the same message format as the one they were received in.

- MCDN/EuroISDN

  The call establishment message is mapped with the relevant information (DLI2 diversion information is included in the SETUP message on the EuroISDN link).

  Once the gateway receives the DLI3 diversion information, it propagates the redirection information by sending a NOTIFY message on the MCDN link. The information can be received in an ALERTING message (for CFU or CFB), in a Facility message (for CFNR), or in a CONNECT message (for CFU, CFB, or CFNR).

- EuroISDN/MCDN

  The call establishment message is mapped with the relevant information (diversion information is included in the SETUP message on the MCDN link).

  The redirection information is received on MCDN in a NOTIFY message which is sent:

  - after the ALERT message.
  - before the CONNECT message, if no ALERT message has been sent.

This information is sent on EuroISDN in a FACILITY message with DLI3 APDU.

- DPNSS/EuroISDN

  The call establishment message is mapped with the relevant information (DLI2 Diversion information is included in the SETUP message on the EuroISDN link).

  The DLI3 information can be sent on the EuroISDN link in an ALERTING message, in a Facility message, or in a CONNECT message (depending on the reason of the redirection). However, since the DLI3 information (presentation indication) is not supported by DPNSS, then it is not necessary to wait for them at the gateway before sending the NAM message. So the NAM message is sent as soon as the ALERTING message is received.

- EuroISDN/DPNSS

  The call establishment message is mapped with the relevant information (Diversion information is included in the ISRM message on the DPNSS link).

  As soon as a NAM is received on the DPNSS link, it is mapped into an ALERTING message with DLI3 Diversion information (presentation always allowed).

Message mapping for rerouting method

The rerouting method has different names, depending on which interface it applies. The diverting node can send a:

- Call Rerouting Request, on QSIG interfaces
- TRO FACILITY message, on MCDN interfaces with TRO configured
- Diverting-Immediate message, on DPNSS interfaces

This can occur at the originating node or on a gateway node between the originating node and the served node (refer to Figure 40: Rerouting request received at the originating node. on page 385, Figure 41: Call rerouting request sent to the CO. on page 386, and Figure 42: Call rerouting request processed at the gateway node. on page 386).

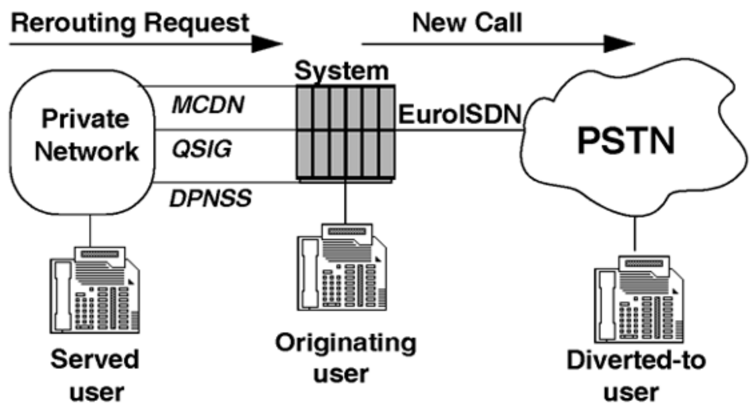


Figure 40: Rerouting request received at the originating node.

Rerouting request received at the originating node

When a valid rerouting request is received on either a QSIG, MCDN or DPNSS interface, and the generation of a new call on the EuroISDN is necessary (due to the rerouting), a DLI2 APDU is included within the SETUP message sent on EuroISDN.

Rerouting request received at the gateway node

When receiving a rerouting request on a EuroISDN/QSIG, EuroISDN/MCDN or EuroISDN/ DPNSS gateway node, the diverted-to number is analyzed to determine whether it is located within the private network, or in the public network.

If the diverted-to user is located within the public network, a Call Rerouting Request is sent to the CO.

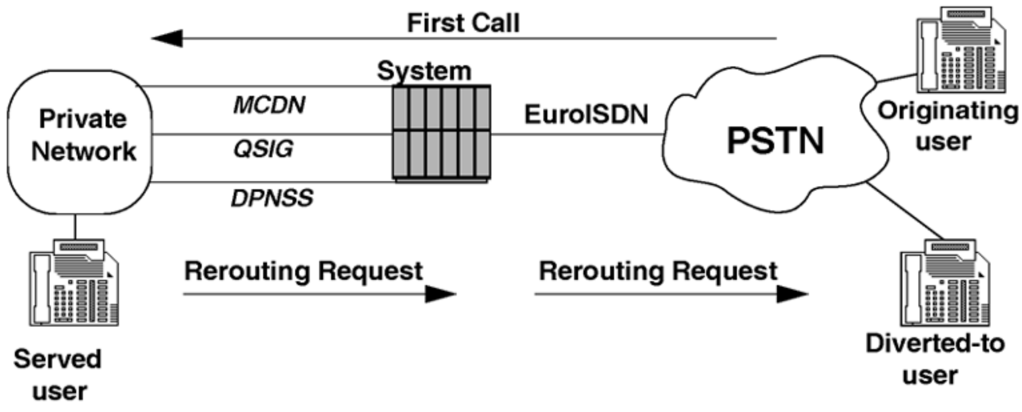


Figure 41: Call rerouting request sent to the CO.

If the diverted-to user is located within the private network, the request is processed by the gateway node.

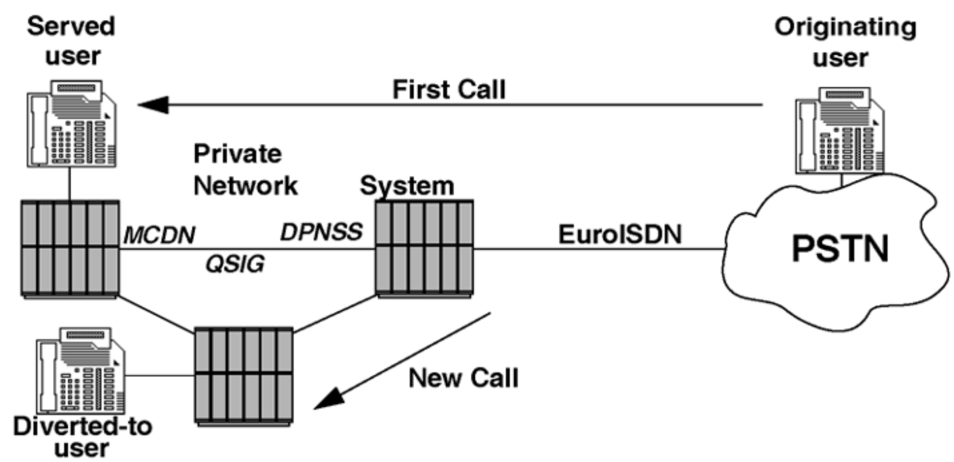


Figure 42: Call rerouting request processed at the gateway node.

If the redirection reason is CFU or CFB, then a FACILITY message with DLI1 information is sent to the originating user as soon as the gateway node processes the rerouting request. If the redirection reason is CFNR, and an ALERT notification is received for the new call, then a FACILITY message with DLI1 information is sent to the originating user.

For the rest of the call establishment messaging, the rerouting node behaves as a gateway node II. It can be either QSIG, MCDN or DPNSS.

Operating parameters

Table 189: Correspondence between the ETSI reason for diversion names and the system features on page 387 summarizes the correspondence between the ETSI reason for diversion supported by this feature, and the system equivalent features.

Table 189: Correspondence between the ETSI reason for diversion names and the system features

ETSI reason for diversion	System features
Call Forwarding Unconditional (CFU)	Call Forward All Calls Internal Call Forward BRI ETSI Call Forward Unconditional ICP Call Forward
Call Forwarding Busy (CFB)	Hunting Hunt by Call Type BRI Special Hunt Call Forward Busy
Call Forwarding No Reply	Call Forward No Answer Flexible Call Forward No Answer Second-Level Call Forward No Answer BRI Special Call Forward No Answer Call Forward No Answer by Call Type Attendant Forward No Answer Timed Reminder Recall (all types) Call Waiting Redirection

The following services included in EN 300 207-1 are not part of this feature:

- Call Deflection (CD)
- Selective Call Forwarding Busy (SCFB)
- Selective Call Forwarding No Reply (SCFNR)
- Selective Call Forwarding Unconditional (SCFU)

The BNE/EuroISDN Call Diversion feature has the following limitations:

- The served user does not receive an indication when a call is diverted.
- The calling user is notified each time a redirection occurs, if information is provided by the network. This means that:
    - the last received notification replaces the previous diversion notification
    - if a redirection occurs and no diversion information is provided by the network, the previous notification (if any) remains unchanged

- A user on the system cannot activate, deactivate, or interrogate EuroISDN Diversion on any switch remotely through the EuroISDN network. A user on another switch cannot activate, deactivate or interrogate EuroISDN Diversion on the system remotely through the EuroISDN network.
- Verification of the validity of the diverted-to-number is not supported.
- The EuroISDN Call Diversion supplementary service is not supported for BRI lines. This feature supports EuroISDN Call Diversion over PRI2 and BRI trunks. Any procedure that is specific to the BRI phone is beyond the scope of this feature.
- The calling user can receive notification that a call has been diverted. There are three possible values:
  - No
  - Yes without diverted-to-number
  - Yes with diverted-to-number (when available).

Due to the limitation of the number of digits that a digital phone can display, only the diverted-to number displays, when available, on the calling phone. The served user number does not display.

Table 190: Relationship between the display of the calling user and the Class of Service served user on page 388 summarizes the effect of these Class of Service settings on the calling user's display. These examples only show the information that is displayed on a phone which is related to this feature. Each terminal has its own way of presenting the information and this feature does not change that.

For the purposes of this discussion, consider only the information type in the examples, not the information location.

For this example assume:

Served user's ISDN number: 0164665000

Diverted-to user's ISDN number: 0164666000

The reason for redirection code for Call Forward All Calls is "F.

Table 190: Relationship between the display of the calling user and the Class of Service served user

Served user's Class of Service related to the calling user's display	Calling user's display once diverted-to-phone rings
No	0164665000
Yes without diverted-to number	0164665000 F
Yes with diverted-to number when available	0164666000 F

- The served user can release their number to the diverted-to-user. There are two possible values: No, Yes. Due to the limitation of the number of digits that a digital phone can display, only the calling number displays on the diverted-to-phone, independent of the served user's Class of Service.
- Table 191: Relationship between the display on the diverted-to-phone and the Class of Service of the served user. on page 389 summarizes the effect of these Class of Service settings on the display of the diverted-to-phone. These examples only show the information that is displayed on a phone which is related to this feature. Each terminal has its own way of presenting the information and this feature does not change that. For the purposes of this discussion, consider only the information type in the examples, not the information location.

  For this example assume:

  Calling user's ISDN number is: 0164664000

  The reason for redirection code for Call Forward All Calls is "F".

Table 191: Relationship between the display on the diverted-to-phone and the Class of Service of the served user.

Served user's Class of Service related to the diverted-to-phone display	Display of the diverted-to-phone
No	0164664000 F
Yes	0164664000 F

When both the calling and served phones are on the same node, there is no change introduced with this feature. In particular, the served phone's Class of Service has no impact on the notification to the calling user.

If the Central Office (CO) rejects the call rerouting request, the system does nothing. It remains in the same basic call state it was in before it sent the call rerouting request. The system waits for the CO to disconnect the call.

If a call from the public ISDN is diverted within the system and a reject component is received from the CO, the system accepts this information and continues to establish the call.

If a diverted call is presented from a public ISDN to the system and a reject component is received from the CO, the system accepts this information and continues to establish the call.

If a diverted call is presented from the system to the public ISDN and the system does not receive a DivertingLegInformation3 component after it receives a CONNECT message, it assumes that presentation of the diverted-to-number is not allowed and continues with the call establishment. If the system receives a reject component from the CO, it accepts this information and continues to establish the call.

If a call from the system is diverted within the public ISDN and the system does not receive a DivertingLegInformation3 component after it receives a CONNECT message, it assumes that presentation of the diverted-to-number is not allowed and continues with the call establishment. If the system receives a reject component from the CO, it accepts this information and continues to establish the call.

Feature interactions

Access Restrictions / Trunk Group Access Restrictions

EuroISDN Call Diversion is not performed if the served user is not able to access the route to the diverted-to-node.

Call Detail Recording (CDR)

When a call forwards by rerouting, no CDR ticket is generated because no established call takes place and the rerouting operation is done by the CO.

Call Forward by Call Type

This feature redirects internal and external calls differently with both the Call Forward No Answer and Hunting features. Different DNs are programmed for internal calls and external calls.

Call Forward by Call Type is supported by the EuroISDN Call Diversion service and the definition of an internal call is not modified by this feature. In particular, ISDN trunk calls using public numbering are considered external.

Note:

The system does not attempt to determine the real originating party with EuroISDN; it only looks at the type of numbering plan for the EuroISDN call.

Call Forward/Hunt Override

The feature allows the use of the Flexible Feature Code for Call Forward/HUNT Override to override Call Forward All Calls, ICP-Call Forward, Call Forward No Answer, Hunting or Make Set Busy at the phone level and by attendants, in both stand-alone and network (MCDN) applications.

This feature is not supported by EuroISDN Call Diversion. A system user can neither originate nor receive a call by using the FFC for CFHO through EuroISDN.

Call Forward Option

The active Class of Service is always the served user's Class of Service. The EuroISDN Call Diversion feature is not affected by the OPT configuration (CFO/CFF) in the served user's Customer Data Block.

Call Waiting Redirection

The Call Waiting Redirection feature allows unanswered calls in the Call Waiting state to be redirected using Call Forward No Answer (CFNA). The waiting call redirects to the active phone's CFNA DN, after the CFNA timer defined in the Customer Data Block expires. The CFNA DN (which can be a messaging service such as Avaya CallPilot®, Voice Mail, and Message Center) handles this redirected call as an unanswered call.

The EuroISDN Call Diversion service handles this type of call as a usual Call Forward No Answer call.

Flexible Orbiting Prevention Timer

The Flexible Orbiting Prevention Timer feature prevents a call from being diverted off-node by the Call Forward feature at a station for a period of FOPT seconds after a call has already been forwarded off-node by a station. FOPT is defined on a customer group basis.

EuroISDN Call Diversion supports the Flexible Orbiting Prevention Timer feature. Consider using it as a workaround to help prevent Reciprocal Call Forward network-wide. However, while this feature allows you to avoid infinite looping, it also limits the number of diversions that can be performed by a phone in a specified period of time. Therefore, if you expect frequent use of EuroISDN Call Diversion, consider using Total Redirection Count instead, which limits the number of diversions on a single call.

Phantom TN

When a Phantom TN is Call Forwarded, the EuroISDN Call Diversion feature treats the Phantom TN as a normal TN.

Networking feature interactions

User to User (UUS1) services

The system does not support the diversion of UUS1 messages.

Network Automatic Call Distribution (NACD)

If a DID call terminates on an ACD DN, the DID call is linked to the ACD queue. NACD takes precedence over EuroISDN Call Diversion.

BNE Name and Private number display

After an incoming EuroISDN call with BNE Name information and a private CLID forwards through a EuroISDN network, the BNE information disappears from the display and is replaced by the notification numbers provided by the Call Diversion feature.

Auxiliary product interactions

Symposium Call Center Server

The call type is updated in the SCC message if the EuroISDN call is diverted to a CDN.

Meridian Link

Present Call Indication (PCI)

This message contains an IE called "Call Type" which contains diversion information about the incoming call. This field is updated for an incoming diverted EuroISDN call.

Unsolicited Status Message (USM)

When a phone stops ringing because the Call Forward No Answer feature has sent the call to the EuroISDN network, a USM message is sent to Meridian Link.

Feature packaging

The following software packages are required for this feature to operate on EuroISDN BRI Trunks:

- Call Party Name Display (CPND) package 95
- ISDN Supplementary Features (ISDN INTL SUP) package 161
- Basic Rate Interface (BRI) 216
- Multipurpose Serial Data Link (MSDL) package 222
- ISDN BRI Trunk Access (BRIT) 233
- EuroISDN (EURO) package 261
- Business Network Express (BNE) package 367

The following software packages are required for this feature to operate on a EuroISDN PRI2 network:

- Call Party Name Display (CPND) package 95
- Primary Rate Access (PRA) package 146
- 2.0 Mbit Primary Rate Interface (PRI2) package 154
- ISDN Supplementary Features (ISDN INTL SUP) package 161
- International Primary Rate Access (IPRA) package 202
- Multipurpose Serial Data Link (MSDL) package 222
- EuroISDN (EURO) package 261
- Business Network Express (BNE) package 367

Feature implementation

Task summary list

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

Table 192: LD 10 - Configure analog (500/2500-type) phone for EuroISDN Call Division notification.

Prompt	Response	Description
REQ:	CHG	Change.
TYPE:	500	Analog (500/2500 type) phone.
TN		Terminal number
	l s c u	Format for Large System and Avaya Communication Server 1000E (Avaya CS 1000E) system, where l = loop, s = shelf, c = card, u = unit.
	c u	Format for Media Gateway 1000 B (Avaya MG 1000B), where c = card and u = unit.
DN	x..x	Directory Number.
HUNT	x..x	Hunt DN.
CLS		Class of Service.
	DN01	Call Diversion: No notification to the calling user.
	DN02	Call Diversion: Notification without diverted-to-number to the originating user.
	(DN03)	Call Diversion: Notification with diverted-to-number to the originating user.
	DNDN	Call Diversion: no served user's number notification to the diverted-to-user.
	(DNDY)	Call Diversion: served user's number notification (when available) to the diverted-to-user.

Prompt	Response	Description
	CFXA	Call Forward to external DN allowed.
	HTA	Hunting allowed.
	FNA	Call Forward No Answer allowed.
	CFTA	Call Forward by Call Type allowed.
	DDGA (DDGD)	DN Display on the other set allowed (denied).
...	...	
FTR		Features.
	CFW nn x..x	Call Forward All Calls, maximum number of digits, destination number.
	EFD x..x	External Flexible Call Forward No Answer DN.
	EHT x..x	External Hunt DN.
	FDN x..x	Flexible Call Forward No Answer DN.

Table 193: LD 11 - Configure digital phone for EuroISDN Call Diversion notification.

Prompt	Response	Description
REQ:	CHG	Change.
TYPE:	aaaa	Phone type.
TN		Terminal number
	l s c u	Format for Large System and CS 1000E system, where l = loop, s = shelf, c = card, u = unit.
	c u	Format for Media Gateway 1000B, where c = card and u = unit.
FDN	x..x	Call Forward No Answer DN (for internal calls if Call Forward by Call Type is active).
CLS	DN01	Call Diversion: No notification to the calling user.
	DN02	Call Diversion: Notification without diverted-to-number to the originating user.
	(DN03)	Call Diversion: Notification with diverted-to-number to the originating user.
	DNDN	Call Diversion: no served user's number notification to the diverted-to-user.
	(DNDY)	Call Diversion: served user's number notification (when available) to the diverted-to-user.
	CFXA	Call Forward to external DN Allowed.
	HTA	Hunting allowed.

Prompt	Response	Description
	FNA	Call Forward No Answer Allowed.
	CFTA	Call Forward by Call Type Allowed.
	DDGA (DDGD)	DN Display on the other set allowed (denied).
...	...	
EFD	x..x	External Flexible Call Forward No Answer DN.
HUNT	x..x	Hunt DN.
EHT	x..x	External Hunt DN.

Table 194: LD 27 - Configure the BRI Digital Subscriber Loop for EuroISDN Call Diversion notification.

Prompt	Response	Description
REQ	CHG	Change.
TYPE	DSL	Digital Subscriber Loop.
DSL		DSL location
	lll s cc dsl#	Format for Large System and CS 1000E system, where: lll (superloop) = 0-156 (must be a number divisible by 4) s (shelf) = 0-1 cc (card) = 0-15 dsl# (DSL location) = 0-7
	c dsl#	Format for Media Gateway 1000B, where: c (card) = 1-20 dsl# (DSL number) = 0-7
FDN	x..x	Flexible Call Forward No Answer DN.
EFD	x..x	External Flexible Call Forward No Answer DN.
HUNT	x..x	Hunt DN.
EHT	x..x	External Hunt DN.

Table 195: LD 27 - Configure BRI phone for EuroISDN Call Diversion notification.

Prompt	Response	Description
REQ	CHG	Change
TYPE	TSP	Terminal Service Profile.
DSL		DSL location
	lll s cc dsl#	Format for Large System and CS 1000E system, where: lll (superloop) = 0-156 (must be a number divisible by 4) s (shelf) = 0-1 cc (card) = 0-15 dsl# (DSL location) = 0-7

Prompt	Response	Description
	c dsl#	Format for Media Gateway 1000B, where: c (card) = 1-20 dsl# (DSL number) = 0-7
DN	x..x y..y	Directory Number and CLID entry.
- CT	aaa	Call Types for the DN (aaa = VCE or DTA). VCE = circuit switched voice, DTA = circuit switched data.
FEAT		Features.
	CFTA	Call Forward by Call Type allowed.
	CFXA	Call Forward to external DN allowed.
	DN01	Call Diversion: No notification to the calling user.
	DN02	Call Diversion: Notification without diverted-to-number to the originating user.
	(DN03)	Call Diversion: Notification with diverted-to-number to the originating user.
	DNDN	Call Diversion: no served user's number notification to the diverted-to-user.
	(DNDY)	Call Diversion: served user's number notification (when available) to the diverted-to-user.
	FNA	Call Forward No Answer allowed.
	HTA	Hunting allowed.
SSRV_ ETSI		ETSI Supplementary Service. Prompted if PRID = 2 (ETSI) in the DSL.
	VCFW	Voice Call Forward. VCFW is valid if CT = VCE or if CT = VCE and DTA.
	DCFW	Data Call Forward. DCFW is valid if CT = DTA or if CT = VCE and DTA.

Table 196: LD 95 - Configure the redirection reason codes.

Prompt	Response	Description
REQ	NEW	Add new data.
	CHG	Change existing data.
TYPE	CPND	Call Party Name Display.
...	...	
RESN	YES	Allow display of reason for redirection codes.
CFWD	aaaa (F)	Mnemonic for Call Forward All Calls display.
CFNA	aaaa (N)	Mnemonic for Call Forward No Answer display.

Prompt	Response	Description
HUNT	aaaa (B)	Mnemonic for Hunting/Call Forward Busy display.

Table 197: LD 16 - Configure EuroISDN Call Diversion on BRI Trunks.

Prompt	Response	Description
REQ	CHG	Change.
TYPE	RDB	Route Data Block.
...	...	
IFC	E403 EURO	ETS 300 403 compliant EuroISDN. ETS 300 102 compliant EuroISDN. **Note:** This feature is supported by either interface type.
CNTY	aaaa	All countries that are supported by the E403 interface: ETSI, AUS, DEN, FIN, GER, ITA, NOR, POR, SWE, DUT, EIR, SWI, ESP, UK, BEL, FRA, CIS.
...	...	
RCAP	DV3I	EuroISDN Call Diversion.

Table 198: LD 17 - EuroISDN Call Diversion configuration on PRI2 trunks.

Prompt	Response	Description
REQ	CHG	Change
TYPE	CFN	Configuration Record.
...	...	
IFC	E403 EURO	ETS 300 403 compliant EuroISDN. ETS 300 102 compliant EuroISDN. **Note:** This feature is supported by either interface type.
CNTY	aaaa	Countries that are supported by the E403 interface: ETSI, AUS, DEN, FIN, GER, ITA, NOR, POR, SWE, DUT, EIR, SWI, ESP, UK, BEL, FRA, CIS.
...	...	
RCAP	DV3I	EuroISDN Call Diversion.

Feature operation

No specific operating procedures are required to use this feature.

Chapter 14: Business Network Express/ EuroISDN Explicit Call Transfer

Contents

This section contains information on the following topics:

Applicable regions

ISDN BRI trunking is not available in North America. Contact your system supplier or your Avaya representative to verify support of this product in your area.

Feature description

Business Network Express

Business Network Express (BNE) is a Virtual Private Network (VPN) solution for connecting several systems through a EuroISDN interface. The BNE solution is a mix of EuroISDN public services and select proprietary features. Refer to Figure 43: Example of a BNE solution on page 402 for an example.

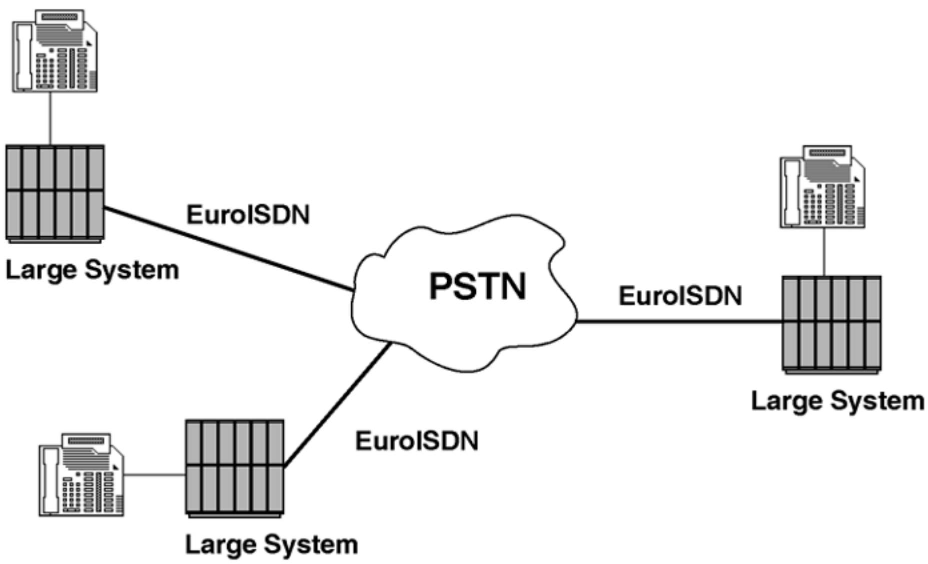


Figure 43: Example of a BNE solution

The BNE solution provides the following functionality between system sites:

- EuroISDN Call Completion
- EuroISDN Name and Private Number Display
- EuroISDN Call Diversion
- EuroISDN Explicit Call Transfer

BNE/EuroISDN Explicit Call Transfer

While Call Transfer functionality exists on EuroISDN with the software feature "Call Transfer", this feature extends the functionality of the private network to:

- notify the public network that a transfer has occurred within the private network
- optimize the call, by requesting the public network to perform the transfer

This service is supported by either EURO (compliant with ETS 300-102) or E403 (compliant with ETS 300-403) interfaces.

BNE/EuroISDN Explicit Call Transfer example

Phone A connects to phone B. Phone A transfers phone B to phone C. Phone A is on the served node, phones B and C are on the remote nodes.

At least one call, phone A to B, or phone A to C, is over a EuroISDN link. The system supports the functionality Explicit Call Transfer at the served node (the node receiving the original call) and at the remote nodes.

With this feature, and depending on network configuration, the system can:

- send transfer notifications on a EuroISDN link
- receive transfer notifications on a EuroISDN link
- activate Call Transfer within the public network on a EuroISDN link

Three types of network configuration are:

- Only the Served User (A) is in the system private network (or stand alone system). The system sends transfer notifications to, or activates Call Transfer within PSTN. Refer to Figure 44: Served User A is the only one in the system private network (or stand alone system) on page 404 and Figure 45: Served user A is the only one in the system private network on page 404.

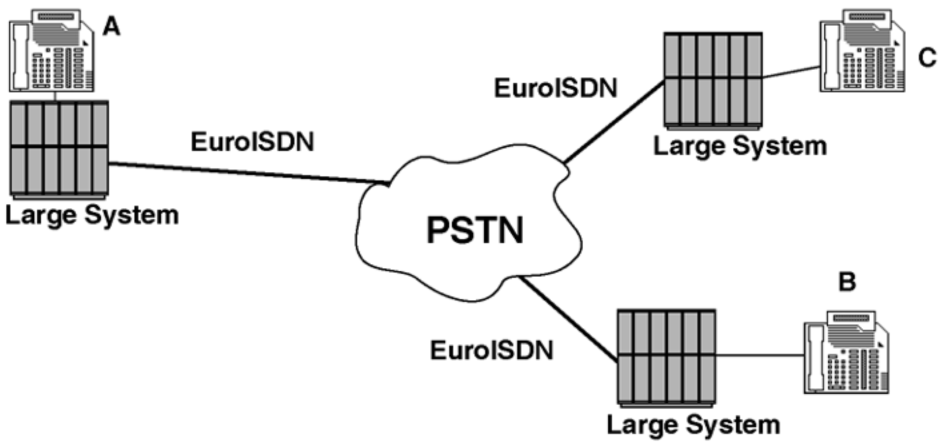


Figure 44: Served User A is the only one in the system private network (or stand alone system)

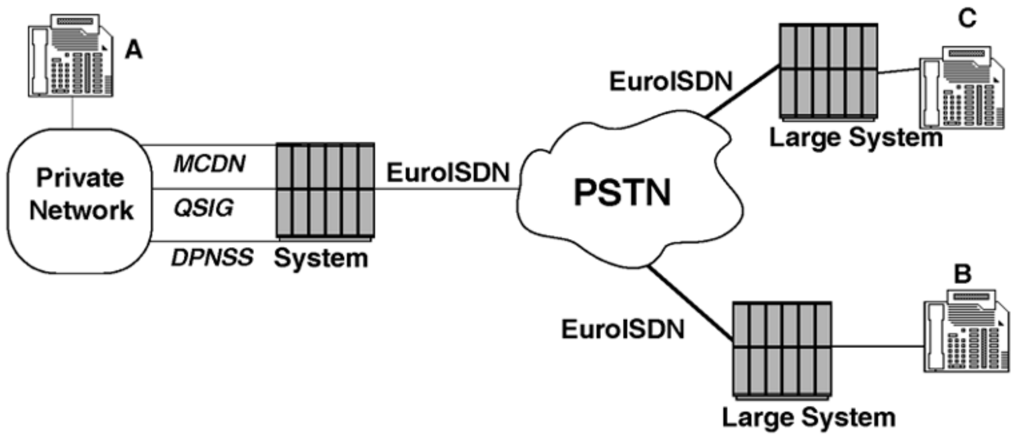


Figure 45: Served user A is the only one in the system private network

- The served user (A) and one of the remote users (B or C) are in the system private network. The system sends transfer notifications. Refer to Figure 46: User A and user B or C are in the system private network on page 405 to Figure 48: User A and user B or C are in the system private network. on page 405. In a gateway connection, the system can be the gateway node (Figure 47: User A and user B or C are in the system private network on page 405), or the gateway node and the served node together (Figure 48: User A and user B or C are in the system private network. on page 405).

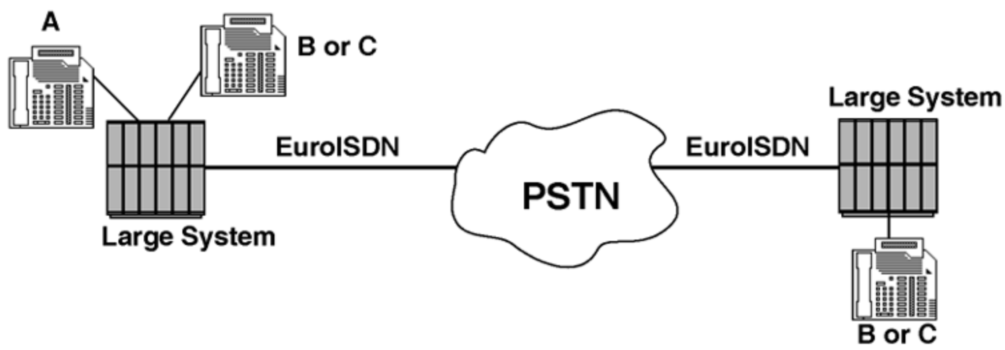


Figure 46: User A and user B or C are in the system private network

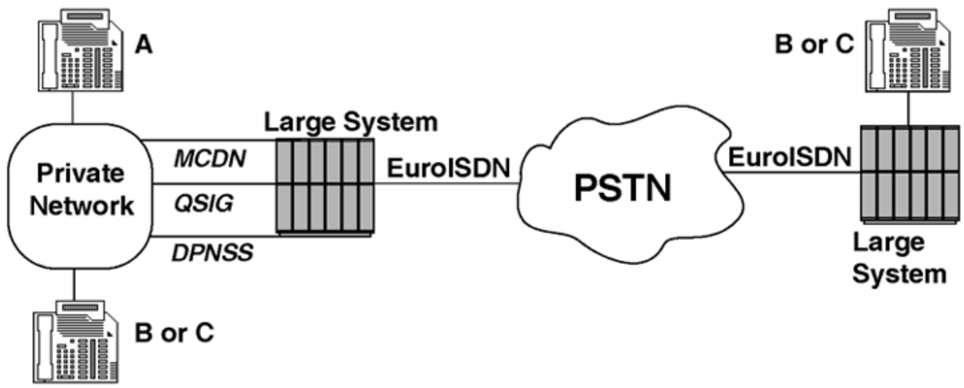


Figure 47: User A and user B or C are in the system private network

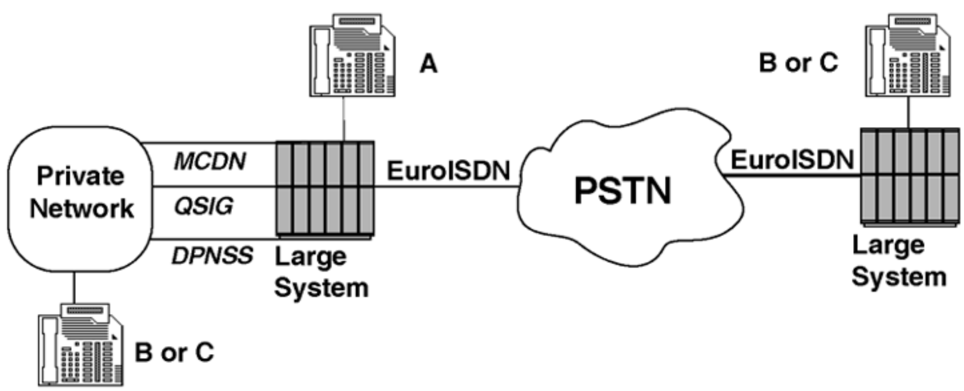


Figure 48: User A and user B or C are in the system private network.

- Only remote users B or C, or B and C, are in the system private network. The system receives transfer notifications. Refer to Figures Figure 49: Remote user B or C is in a system private network on page 406 and Figure 50: Remote user B or C is in the system private network on page 406.

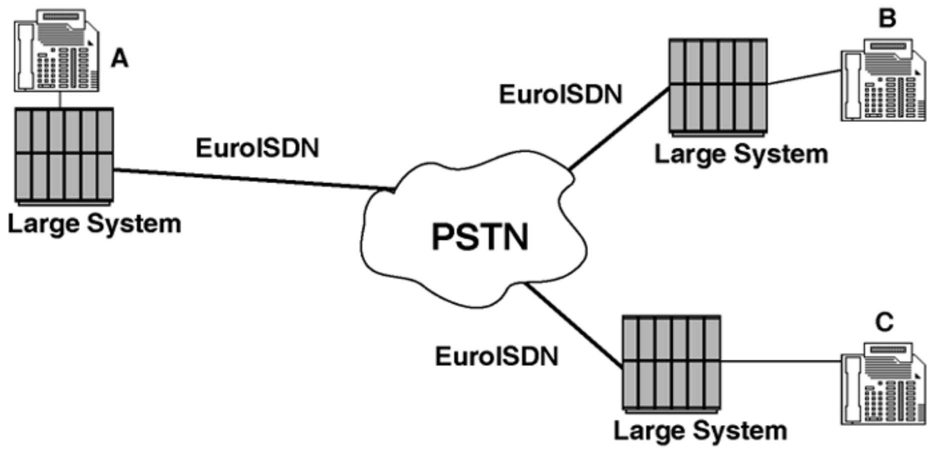


Figure 49: Remote user B or C is in a system private network

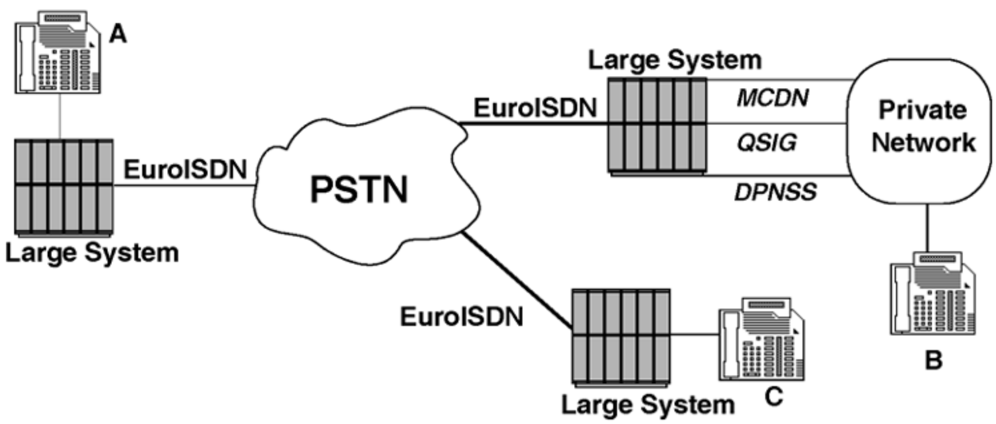


Figure 50: Remote user B or C is in the system private network

Call Transfer through the PSTN

When the following conditions are met, Call Transfer through the PSTN is possible:

- Only the served user, A, is in the system private network (refer to Figure 44: Served User A is the only one in the system private network (or stand alone system) on page 404).
- Both D-channels required in the transfer have the Remote Capability for Call Transfer notification and invocation (ECTO) configured.

Note:

In most situations, D-channels 1 and 2 are the same.

- Both calls are in the established call state.

The system, working as the served node, invokes the Call Transfer in the public network. This can optimize trunk usage by suppressing tromboning between the system and the PSTN. Trunk optimization occurs only when both calls involved in the transfer are in the established call state.

With a supervised transfer, the transfer (by join) is first completed on the system, and notifications are sent to the PSTN. If the conditions are met, the system invokes Explicit Call Transfer to the PSTN.

With an Unsupervised transfer, the transfer (by join) is first completed on the system, and notifications are sent to the PSTN. The system waits until both calls are in an established call state. When both calls are established, and all the conditions are met, the system invokes Explicit Call Transfer in the PSTN.

The Explicit Call Transfer invocation takes place in three steps:

- Request of LinkId to the Public Network
- Request of Call Transfer, using the received LinkID
- Reception of Call Transfer Confirmation

The served node (the system) requests a LinkId for Call Transfer to the Public Network, for the call between user A and user B. The served node stores the LinkId received. This LinkId is used by the PSTN to link the two calls involved in the transfer.

Then the served node requests an Explicit Call Transfer for the call between user A and user C, sending the Linkid previously received.

Upon receipt of the Call Transfer request, the public network releases the LinkId value, by:

- connecting user B to user C in the public network
- disconnecting the call between user A and user C
- sending the result of the Call Transfer request
- disconnecting the call between user A and user B

If the public network does not reply to the LinkId or Call Transfer requests, or reply with an error or rejection component, no action is taken by the system. If this occurs, transferred and transferred-to nodes are not informed of the transfer. It will not have any impact on the served and remote users, because the call was already transferred by the system.

Call Transfer Notification Display

If the network provides the information, the originating caller is notified, on the display of the phone, when a transfer occurs. This means that:

- if a previous Call Transfer notification was provided, it is replaced by the last received notification.
- if a transfer occurs, with no Call Transfer information provided by the network, and a previous notification was provided, the notification remains unchanged.

The following scenario is considered to be a standard Call Transfer situation (refer to Figure 51: User B calls user A. User A transfers call to user C. on page 409, Figure 52: Display of

established call between user A and user B on page 409, Figure 53: User A presses Call Transfer key and calls user C on page 409, and Figure 54: User A presses Call Transfer key a second time to transfer the call to user C on page 410):

1. User B calls user A.
2. User A answers the call.
3. User A presses the transfer key, and calls user C.
4. User A presses the Call Transfer key again to complete the transfer (the transfer can be completed when the secondary call is alerting or established).

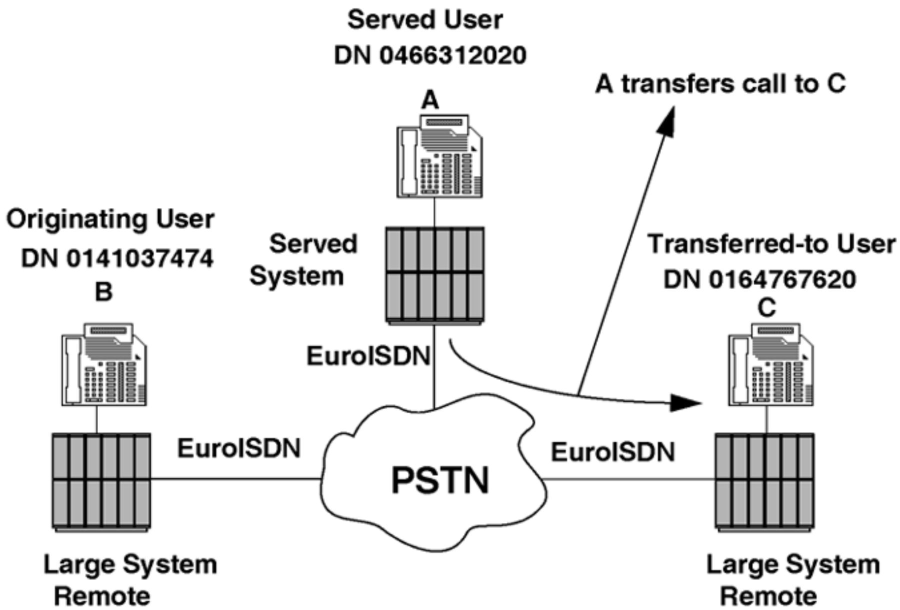


Figure 51: User B calls user A. User A transfers call to user C.

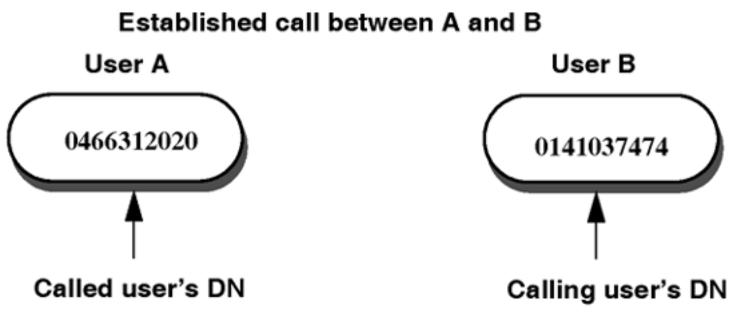


Figure 52: Display of established call between user A and user B

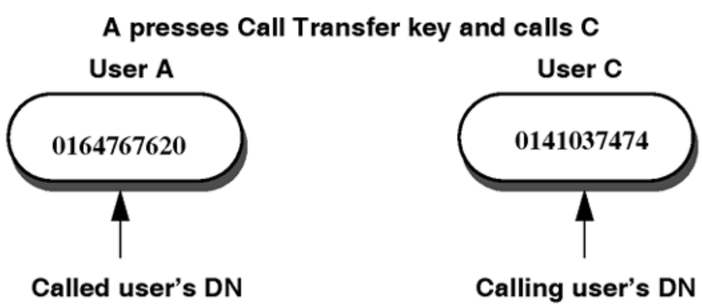


Figure 53: User A presses Call Transfer key and calls user C

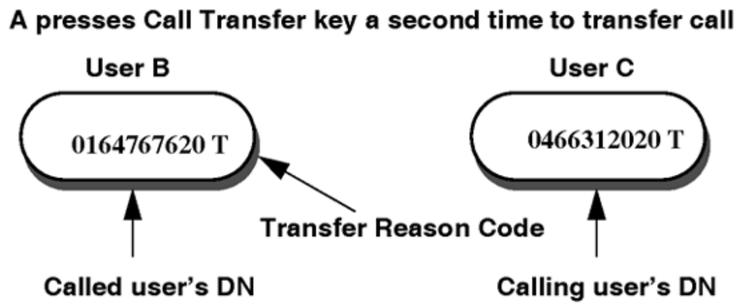


Figure 54: User A presses Call Transfer key a second time to transfer the call to user C

Call Transfer Notifications Display Rules

The Call Transfer Notifications Display Rules are similar to Transfer Notifications rules on a system private link, and depend on the:

- Class of Service configured on the phones
- Presentation Indicator of the Redirection Number received

Class of service definitions:

- DDGA - DN Display on other phone Allowed
- DDGD - DN Display on other phone Denied
- CNDA - Calling Name Display Allowed
- CNDD - Calling Name Display Denied

Note:

The reason is displayed on a phone only when the CNDA Class of Service is configured.

Transferred and Transferred-to user notification rules

Transfer Reason Notification rules:

The reason displayed on the Transferred or Transferred-to user's phone (when the transfer notification information is received from the served node) is configured in LD 95. No redirection reason is displayed on a phone if the CNDD Class of Service is configured

Redirection Number Notification rules:

The redirection number displays on the user's phone if the received presentation information indicates that presentation is allowed. Otherwise, the phone displays the trunk route access code and trunk route member number (instead of the redirection number).

If a remote user has the DDGD Class of Service defined, the phone sends its number in a redirection number with the Presentation Indicator set to Presentation Restricted.

If a remote user has the DDGA Class of Service defined, the phone sends its number in a redirection number with Presentation Indicator set to Presentation Allowed.

Table 199: Originating and Transferred-to users notification in a system environment on page 411 identifies the originating user's notification according to the originating and transferred-to users' configuration options.

Table 199: Originating and Transferred-to users notification in a system environment

Class of Service Originating user phone B DN 0141037474	Class of Service Transferred-to user phone C DN 0164767620	Originating user's display after receipt of Transferred-to user's transfer notification information	Transferred-to user's display after receipt of originating user's transfer notification information
CNDA DDGA	CNDA DDGA	0164767620 T	0141037474 T
CNDD DDGA	CNDD DDGA	0164767620	0141037474
CNDA DDGD	CNDA DDGA	0164767620 T	211-4 T
CNDD DDGD	CNDD DDGA	0164767620	211-4
CNDA DDGD	CNDA DDGD	312-6 T	211-4 T
CNDD DDGD	CNDD DDGD	312-6	211-4

Operating parameters

If the LinkId Request, or the Call Transfer Request to the PSTN is rejected, the system does not take any action.

The EuroISDN Call Transfer supplementary Service is not supported on the EuroISDN master mode interface.

This feature depends on the following system hardware:

- ISDN Primary Rate Interface
    - 2 Mbit Primary Rate Access card (NT8D72BA) for layer 1 interface on Large Systems
    - 2 Mbit Primary Rate Access card (NTBK50) for layer 1 interface for Small Systems .
    - The downloadable DCH daughterboard (NTBK51) card is required for Small Systems .
    - The Dual PRI pack
    - The clock controller card (NTAK20BA/BB) for Small Systems
    - MSDL card (NT6D80AA) on Large Systems
    - Clock Controller NTRB53 for Large Systems
- ISDN Basic Rate Interface
- SILC card (NT6D70BA) for layer 1 interface
- MISP card (NT6D73AA) for Large Systems
- MISP card (NTBK22XX) for Small Systems.

Feature interactions

Call Detail Recording (CDR)

For invocation of Explicit Call Transfer within the public network, CDR tickets issued do not reflect the complete duration of the call to the transferred-to phone.

When Call Transfer is completed on an established call, an S (Start) record is generated for each calling party involved at the time Call Transfer was activated. After the call is terminated, an E (End) record is generated showing its final disposition. Start and End records are generated at the Transferring node.

If more than one transfer occurs, an X (Transfer) record is generated for each transfer when the primary call involved a CDR-X call. If N transfers occurs, (N-1) records are generated in addition to the Start and End records.

When a EuroISDN gateway is used, the BLID field is updated with the Call Transfer Notification information received at the Transferring node.

In a stand-alone situation, when only the served user A is on the system, no notification is received. There is always one incoming call, and one outgoing call, because it is not possible to transfer an incoming DID call over an outgoing DID call. When a transferred call is released, the BLID field of the E record is filled with the Redirection number sent on the outgoing side of the transfer.

Networking feature interactions

BNE/EuroISDN Name and Private Number display

BNE Name and Private number information cannot be carried out in EuroISDN Explicit Call Transfer Notifications. If an incoming EuroISDN call with the BNE name information and the private CLID is being forwarded through EuroISDN, after Call Transfer occurs, the BNE information name and number are replaced on the display and the notification numbers provided by the Explicit Call Transfer feature.

Auxiliary product interactions

Meridian Link

Unsolicited Status Message (USM)

When an ACD agent is transferred over a EuroISDN link, a USM message is sent to the Meridian link.

Feature packaging

The Business Network Express/EuroISDN Explicit Call Transfer and Gateways feature requires the following package Business Network Express (BNE) package 367.

The Business Network Express/EuroISDN Explicit Call Transfer and Gateways feature is dependent on the following packages:

- Call Party Name Display (CPND) package 95
- Primary Rate Access (PRA) package 146
- 2.0 Mbit Primary Rate Interface (PRI2) package 154
- ISDN Supplementary Features (ISDN INTL SUP) package 161
- International Primary Rate Access (IPRA) package 202
- Basic Rate Interface (BRI) 216

- Multipurpose Serial Data Link (MSDL) package 222
- ISDN BRI Trunk Access (BRIT) 233
- EuroISDN (EURO) package 261
- Business Network Express (BNE) package 367

Feature implementation

Task summary list

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

1. Table 200: LD 10 - Configure an analog (500/2500-type) phone for EuroISDN Call Transfer. on page 414
2. Table 201: LD 11 - Configure a Meridian 1 proprietary phone for EuroISDN Call Transfer. on page 415
3. Table 202: LD 95 - Configure the Call Transfer Reason for Redirection Code. on page 415
4. Table 203: LD 17 - Configure EuroISDN Call Transfer on PRI2. on page 416
5. Table 204: LD 16 - Configure EuroISDN Call Transfer on BRI trunk. on page 416

Table 200: LD 10 - Configure an analog (500/2500-type) phone for EuroISDN Call Transfer.

Prompt	Response	Description
REQ:	CHG	Change
TYPE:	500	analog (500/2500-type) phone.
TN		Terminal number
	l s c u	Format for Large System and Avaya Communication Server 1000E (Avaya CS 1000E) system, where l = loop, s = shelf, c = card, u = unit.
	c u	Format for Media Gateway 1000 B (Avaya MG 1000B), where c = card and u = unit.
DN	x..x	Directory Number.
CLS		Class of Service.
	XFA	Call Transfer Allowed

Prompt	Response	Description
	DDGA	DN Display on the other phone allowed (default) DDGD = DN Display on the other phone denied

Table 201: LD 11 - Configure a Meridian 1 proprietary phone for EuroISDN Call Transfer.

Prompt	Response	Description
REQ:	CHG	Change.
TYPE:	aaaa	Phone type
TN		Terminal number
	l s c u	Format for Large System and CS 1000E system, where l = loop, s = shelf, c = card, u = unit.
	c u	Format for Media Gateway 1000B, where c = card and u = unit.
CLS		Class of Service.
	XFA	Call Transfer Allowed
	DDGA	DN Display on the other phone allowed (default) DDGD = DN Display on the other phone denied
	CNDA	Calling Name Display Allowed CNDD = Calling Name Display Denied (default) **Note:** There is no name sent on EuroISDN, but this must be configured to display the Reason for Redirection Code.

Table 202: LD 95 - Configure the Call Transfer Reason for Redirection Code.

Prompt	Response	Description
REQ	NEW	Add new data.
	CHG	Change existing data.
TYPE	CPND	Call Party Name Display.
...	...	
RESN	YES	Allow display of Reason for Redirection Codes.
XFER	aaaa (T)	Mnemonic for Call Transfer display.

Table 203: LD 17 - Configure EuroISDN Call Transfer on PRI2.

Prompt	Response	Description
REQ	CHG	Change.
TYPE	ADAN	ADAN Data Block.
ADAN	NEW aaa x CHG aaa x	Action Device and Number. Add I/O device. Where: aaa = type, x = port Change I/O device. Where: aaa = type, x = port
...	...	
IFC		Interface type for D-channel.
	E403 EURO	EuroISDN interface for ETS 300 403. EuroISDN interface.
...	...	
CNTY	xxxx	Country. **Note:** Countries that support the E403 interface.
...	...	...
RCAP	CTO ECTO	Remote Capabilities. Add Call Transfer notification. Add Call Transfer notification and invocation. XCTO = Remove Call Transfer notification (CTO) or Call Transfer notification and invocation (ECTO) **Note:** CTO and ECTO can not be configured together.

Table 204: LD 16 - Configure EuroISDN Call Transfer on BRI trunk.

Prompt	Response	Description
REQ	NEW	Add new data block.
	CHG	Change existing data block.
TYPE	RDB	Route Data Block.
DTRK	YES	Digital Trunk Route. No = default
DGTP	BRI	Basic Rate Interface.
...	...	
IFC	E403 EURO	Interface type for D-channel. EuroISDN interface for ETS 300 403. EuroISDN interface.
...	...	

Prompt	Response	Description
CNTY	xxxx	Country. **Note:** Countries that support the E403 interface.
...	...	...
RCAP	CTO ECTO	Remote Capabilities. Add Call Transfer notification. Add Call Transfer notification and invocation. XCTO = Remove Call Transfer notification XECTO = Remove Call Transfer notification and invocation **Note:** CTO and ECTO can not be configured together.

Feature operation

Refer to the Call Transfer feature described in *Avaya Features and Services Fundamentals, NN43001-106*.

Chapter 15: Business Network Express/ Name and Private Number Display

Contents

This section contains information on the following topics:

Feature description

Business Network Express (BNE) is a term that refers to a group of different EuroISDN network functionalities. The BNE capabilities provide the systems that are connected on a EuroISDN public network with the following functionalities:

- EuroISDN Call Completion
- EuroISDN Name and Private Number Display

- EuroISDN Call Diversion
- EuroISDN Explicit Call Transfer

BNE provides a Virtual Private Network (VPN) solution for the systems through the EuroISDN public network. BNE is appropriate for companies that require a network that operates as if it is a private network, but has an affordable start-up cost. The pre-existing Virtual Network Services (VNS) solution provides more features than BNE (VNS is a version of the ISL interface); however, VNS requires a leased line for the D-channel between the systems.

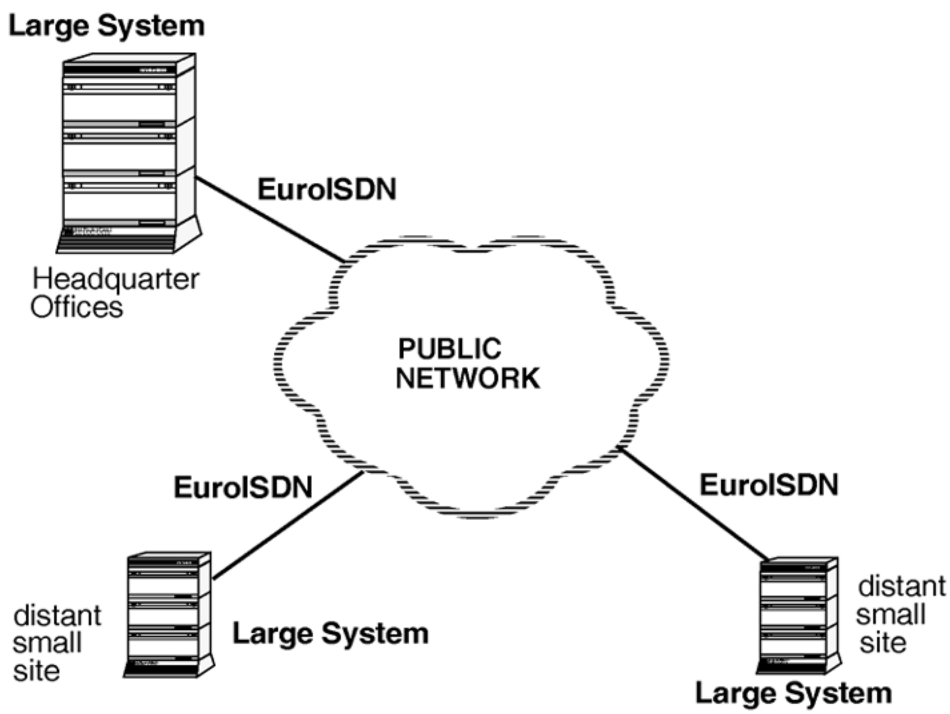


Figure 55: Example of a network where BNE is useful

With BNE implemented, when a user dials a private network number to reach a user at another system site through the public network, the ESN software causes the dialed number to be outpulsed as a public number. The BNE software inserts the Calling Name and the Private CLID in the User-to-User Information Element (IE) carried by the SETUP message.

At the destination switch, the Private CLID is displayed, along with the Calling Name, on the alerted phone. The name associated with the alerted phone is delivered to the calling user in a User-to-User IE carried in the ALERT message and displayed on the calling phone.

When the call is answered, the Connected Name and the private Connected Number is provided to the calling user in a User-to-User IE carried in the CONNECT message.

Consistent with MCDN and QSIG networking, the letter H is displayed in front of the private number.

You can implement restrictions on displaying the name and number of the calling, called, or connected party.

The information presented here deals with the Name and Private Number Display parts of the BNE package.

The Name and Private Number Display parts use the User-to-User Information Element (IE) defined in the EuroISDN basic call standards (ETS 300 102-2 and EN 300 403-1) and the implicit User-to-User service 1, (defined in ETS 300 284 and ETS 300 286-1), to carry user-defined signaling.

For information about the other parts of the BNE package, refer to the *Business Network Express/EuroISDN Call Diversion and Business Network Express/EuroISDN Explicit Call Transfer* feature modules in this book.

Name Display on EuroISDN

This functionality is based on the existing MCDN and QSIG Call Party Name Display (CPND) and Display of Calling Party Denied (DPD) features. The following three services are supported.

1. Calling Name Identification Presentation (CNIP)

CNIP is a supplementary service which provides the called user with the calling user's name. This service is permanent and based on the Class of Service of the phone originating the call.

- For BRI phones, set PRES to YES in LD 27.
- For all other phones, configure NAMA in the Class of Service in LD 10 and LD 11.
- For attendants, the CNIP service is always provided; it is not configurable.

CNIP does not deliver the calling user's name to the called user if:

- The Calling Party Name is not available. This occurs when a name is not configured in the CPND data block for the calling party DN, or in the case of interworking.
- Presentation is restricted for the phone originating the call, as controlled by the CNIR service.

CNIP - EuroISDN/MCDN Gateway

On reception of a call coming from an MCDN network, with the calling user's name information routed to the PSTN network, the calling name is sent through EuroISDN to the destination node (if the EuroISDN route list block supports BNE).

On reception of a EuroISDN call with the calling user's name information routed to the MCDN network, the gateway node delivers the calling user's name information to the MCDN network.

CNIP - EuroISDN/QSIG Gateway

On reception of a call coming from a QSIG network with the calling user's name information, and routed to the PSTN network, the calling name is sent through EuroISDN to the destination node (if the EuroISDN route list block supports BNE).

On reception of a EuroISDN call, with calling user's name information routed to the QSIG network, the gateway node delivers the calling user's name information to the QSIG network.

CNIP - EuroISDN/DPNSS Gateway

DPNSS does not support name display.

2. Connected Name Identification Presentation (CONP)

This is a service offered to the calling user. CONP provides the calling user with the alerted/ connected user's name. CONP service also delivers:

- the name of the alerted user to the calling user whenever the called user's phone starts ringing
- the name associated with the phone that answers the call

  The Alerting/Connected Name information is included in the User -to-User IE and carried in the ALERTING/CONNECT message.

  When an Alerting/Connected Name is received with a "presentation allowed" setting, it is displayed on phones or attendant consoles equipped with displays.

  This service is permanent and based on the Class of Service on the phone receiving the call:
- For BRI phones, set PRES to YES in LD 27.
- For all other phones, set NAMA in the Class of Service in LD 10 and LD 11.
- For attendants, the CONP service is always provided; it is not configurable.

CONP does not deliver the called user's name to the calling user if:

- The Called Party Name is not available. This occurs when a name is not configured in the CPND data block for the called party DN, or in the case of interworking.
- Presentation is restricted for the terminating phone as controlled by the CNIR service.

CONP - EuroISDN/QSIG Gateway

The QSIG network receives the connected (or alerting) user's name from the BNE feature. The connected (or alerting) user's name provided by the QSIG network is sent over EuroISDN to the originator of the call.

CONP - EuroISDN/MCDN Gateway

The connected (or alerting) user's name provided by the MCDN network is sent over EuroISDN to the originator of the call.

Note:

The alerted name carried in the NOTIFY message (RCAP = ND2) is not provided to the originator because UUS1 doesn't define any message to tandem this information.

The connected (or alerting) user's name delivered by the BNE feature is sent to the MCDN network.

CONP - EuroISDN/DPNSS Gateway

DPNSS does not support name display.

3. Calling/Connected Name Identification Restriction (CNIR)

This service prevents the user's name from being presented to another user. This service is activated in two ways:

- For all calls. It is based on the Display of Calling Party Denied feature. The Calling/Connected/Called/Alerting Name is denied or allowed using the Class of Service.
    - For BRI phones, set PRES to NO in LD 27. Do not enter a name for the default DN in LD 27.
    - For all other phones, set NAMD in the Class of Service in LDs 10 and 11.
    - For attendants, the CNIR service is not supported.
- For each call (Class of Service NAMA and the user dials the Calling Party Privacy Flexible Feature Code when initiating a call). The Calling Number and Name is restricted when the user dials the CPP code. Attendants can dial the CPP code for CNIR.

Display of restricted name

If the Calling Name information is received with a "presentation restricted" setting, then Xs are displayed on the called user's display, if it is able and authorized to receive the Calling Name information. If the called user's name information is received in the ALERTING message and its presentation is restricted, then Xs are displayed on the calling user's display, if it is able and authorized to receive name information. If the connected user's name information is received in the CONNECT message and its presentation is restricted, then Xs are displayed on the calling user's display, if it is able and authorized to receive name information.

CNIR - EuroISDN/QSIG Gateway

When a user invokes the CNIR service, the calling, alerting, and connected names are marked as "presentation is restricted", and this indication is passed to the other network.

CNIR - EuroISDN/MCDN Gateway

When a user invokes the CNIR service, the calling, alerting, and connected names are marked as "presentation is restricted", and this indication is passed to the other network.

CNIP - EuroISDN/DPNSS Gateway

DPNSS does not support name display.

Private Calling Number on EuroISDN

EuroISDN public networks can support the same private Calling Number capabilities as QSIG and MCDN networks, with the BNE/Name and Private Number Display feature implemented on the systems.

This functionality delivers a Calling Party Number in a private format (based on a Coordinated Dialing Plan or Uniform Dialing Plan numbering plan) in addition to the public-format Calling Party Number. The public-format number is delivered in the Calling Number IE. The BNE software is responsible for delivering the private number in the User-to-User IE. The Connected Number IE is provided by the Central Office in a public format but the private Connected Number is displayed on the calling user's phone.

The private format depends on the numbering plan the caller used to dial the call.

The private calling number is constructed based on the CLID Enhancement feature. It contains the following information:

- numbering plan field (private)
- type of number field (CDP or LOC or unknown)
- the DN digits of the calling phone prefixed by an LSC (CDP) or HLOC (UDP), if configured
- presentation flag to allow or deny the display on the called user's phone

The following two services are supported:

1. Calling Line Identification Presentation (CLIP)

CLIP provides the called party with the identification of the calling party in a form that allows the called party to return the call, if desired, using the VPN network built on the public EuroISDN connections. The CLIP option is configured in the phone programming as follows:

- BRI phones: use PRES, CLIP and TRANS in LD 27
- other phones: Class of Service DDGA in LD 10 and LD 11
- attendant: CLIP is always provided; it is not configurable

CLIP - EuroISDN/MCDN gateway

On reception of a call coming from MCDN network with a private calling number and routed to the PSTN network, the private calling number is sent through EuroISDN to the destination node by the BNE feature.

On reception of a EuroISDN call with a BNE private calling number routed to the MCDN network, the gateway node uses the calling number delivered by the BNE feature to build the CLID IE sent over MCDN.

CLIP - EuroISDN/QSIG Gateway

On reception of a call coming from a QSIG network, with a private calling number and routed to the PSTN network, the private calling number is sent through EuroISDN to the destination node by the BNE feature.

On reception of a EuroISDN call, with a BNE private calling number routed to the QSIG network, the gateway node uses the calling number delivered by the BNE feature to build the CLID IE sent over QSIG.

CLIP - EuroISDN/DPNSS Gateway

On reception of a call coming from a DPNSS network, with a private calling number (OLI) and routed to the PSTN network, the private calling number is sent through EuroISDN to the destination node by the BNE feature.

On reception of a EuroISDN call, with a BNE private calling number routed to the DPNSS network, the gateway node uses the calling number delivered by the BNE feature to build the OLI sent over DPNSS.

'H' is not displayed in the private number on the DPNSS side, according to the existing DPNSS gateway.

2. Calling Line Identification Restriction (CLIR)

This service enables the calling party to prevent presentation of the calling number on the called user's phone. There are two options for implementation:

- Presentation restricted for all calls. Define DDGD in the Class of Service of the phone. CLIR is not supported for attendant consoles. For BRI phones use the CLIP, PRES and TRANS prompts in LD 27.

Table 205: Reception of CLID on BRI phone

CLIP	TRANS	Presentation of the calling number IE	CLID IE transmitted to the called BRI phone
YES	YES	allowed	transparent
YES	YES	restricted	transparent
YES	NO	allowed	transparent
YES	NO	restricted	calling number digits are removed from the IE, but the "empty" CLID field is still sent
NO			CLID IE is not sent

- Presentation restricted for individual calls. The user dials the Calling Party Privacy (CPP) Flexible Feature Code. Define DDGA in the Class of Service of the phone.

Class of Service CLBA/CLBD (Calling Party Number and Name per-line blocking allowed or denied): On a permanent basis, the Calling Number and Name can be restricted using the CLBA Class of Service in LD 10 and LD 11 (not applicable to BRI phones). If you program CLBD, the user can dial the CPP code for blocking of name and number for individual calls.

Users of BRI phones cannot dial the CPP code to block name and number; they must use a presentation soft key.

CLIR - EuroISDN/QSIG Gateway

When the CLIR service is invoked, the calling number is marked as "presentation is restricted", and this indication is passed to the other network.

CLIR - EuroISDN/MCDN Gateway

When the CLIR service is invoked, the calling number is marked as "presentation is restricted", and this indication is passed to the other network.

CLIR - EuroISDN/DPNSS Gateway

The CLIR service is not supported on DPNSS. Upon receiving the calling number from DPNSS, it is marked as "presentation is unrestricted" and then passed to the EuroISDN side.

If a calling number marked as "presentation restricted" is received from the EuroISDN side, it is passed to the DPNSS side without the possibility of indicating "presentation restriction". Therefore, the calling number will display.

Private Connected Number on EuroISDN

EuroISDN public networks can support the same private Connected Number capabilities as QSIG and MCDN networks, with the BNE/Name and Private Number Display feature equipped on the systems.

This functionality delivers a Connected Number in a private format (CDP or UDP numbering plan) in addition to the public-format Connected Number. The public-format number is delivered in the Connected Number IE. The BNE software is responsible for delivering the private Connected Number to the calling party in the User-to-User IE. The Connected Number IE is provided by the Central Office in a public format but the private Connected Number is displayed on the calling user's phone.

The private Connected Number is delivered to the calling user only if a private Calling Number was provided from the calling user. The format depends on the numbering plan of the received private CLID.

The private Connected Number contains the following information:

- numbering plan field (private)—depends on the NPI of the received CLID
- type of number (TON) field (CDP or LOC or unknown)—depends on the TON of the received CLID
- the DN digits of the connected phone prefixed by an LSC (CDP) or HLOC (UDP), if configured
- presentation flag to allow or deny the display on the calling user's phone

Two services are supported:

1. Connected Line Identification Presentation (COLP)

This service allows the calling party to receive identification of the connected party. The Connected Number replaces the dialed number on the display of the calling phone. If the called party has presentation restriction, using the COLR supplementary service, the private Connected Number field is empty or presented with the presentation restriction flag on (to users with an override category). The attendant DN is sent when the call is answered by the attendant.

Note:

BRI phones and attendant consoles can have an override key.

The COLP option is configured for phones as follows:

- BRI phones: use COLP and TRANS in LD 27 for each DN
- other phones: Class of Service DDGA in LD 10 and LD 11
- attendant: COLP is always provided; it is not configurable

COLP - EuroISDN/QSIG Gateway

The connected number, delivered by the BNE feature, is sent to the QSIG network. The connected number, provided by the QSIG network, is sent over EuroISDN to the originator of the call.

COLP - EuroISDN/MCDN Gateway

The connected number, delivered by the BNE feature, is sent to the MCDN network. The connected number, provided by the MCDN network, is sent over EuroISDN to the originator of the call.

Note:

The connected number is provided by the MCDN network only in the case of call diversion.

COLP - EuroISDN/DPNSS Gateway

The connected number, delivered by the BNE feature, is sent to the DPNSS network. The connected number, provided by the DPNSS network, is sent over EuroISDN to the originator of the call.

'H' is not displayed in the private number on the DPNSS side, in accordance with the existing DPNSS gateway.

2. Connected Line Identification Restriction (COLR)

This service enables the connected party to prevent presentation of its number on the calling user's phone. There are two options for implementation:

- presentation allowed: the allowed option is set in the CONNECT message. The calling user is presented with the Connected Number.
- presentation restricted: the restricted option is set in the CONNECT message. The Connected Number is always provided to the network. If the calling user has an "override" category, the network passes this Connected Number to it. If not, the Connected Number is not available to the calling user.

The COLR option is configured for phones as follows:

- other phones: Class of Service DDGD in LD 10 and LD 11
- attendant: COLR is not provided
- BRI phones: Table 206: Reception of COLP on BRI phones on page 428 summarizes the possibilities:

Table 206: Reception of COLP on BRI phones

COLP	TRANS	presentation of the Connected Number IE	COLP IE transmitted to the calling BRI phone
YES	YES	allowed	transparent
YES	YES	restricted	transparent
YES	NO	allowed	transparent
YES	NO	restricted	connected number digits are removed from the IE, but the "empty" COLP field is still sent to the phone
NO			connected number IE is not passed to the phone

Note:

The same rules are used for the public Connected Number, if no private Connected Number is received.

COLR - EuroISDN/QSIG Gateway

When the COLR service is invoked, the connected number is marked as "presentation is restricted", and this indication is passed to the other network.

COLR - EuroISDN/MCDN Gateway

When the COLR service is invoked, the connected number is marked as "presentation is restricted", and this indication is passed to the other network.

COLR - EuroISDN/DPNSS Gateway

The COLR service is not supported on DPNSS. Upon receiving the connected number from DPNSS, it is marked as "presentation is unrestricted" and then passed to the EuroISDN side.

If a connected number marked as "presentation restricted" is received from the EuroISDN side, it is passed to the DPNSS side without the possibility of indicating "presentation restriction". Therefore, the connected number will display.

Operating parameters

The hardware requirements for BRI Trunk (BRIT) access are:

- NT6D73AA MISP
- NT6D70BA SILC
- NT6D71AA UILC

The hardware requirements for PRI2 are as follows:

- 2.0 Mbit NT8D72 PRI card with either one of the following cards for handling the D-channel:
    - NT6D11 DCHI card
    - NT6D80 MSDL card
- NT5D97AD Dual DTI/PRI 2.0 Mbit/s card with one of the following cards for handling the D-channel:
    - NT6D11 DCHI card
    - NT6D80 MSDL card
    - NTBK51 Downloadable D-channel daughter board

For BNE functionality to work, the public network must support User-to-User service 1 implicit procedures. The node at the terminating end must support the BNE/Name and Private Number Display feature. Configure PSTN routes in Route List Indexes to these destinations with the BNE feature activated (BNE = YES). For calls to nodes that do not support the feature, use the default setting (BNE = NO) on PSTN routes in the Route List Indexes.

CDP or UDP numbering plans must be used. Trunk route access codes are not supported. CDP or BARS or NARS software must be equipped.

The maximum length of names carried by the BNE feature is 27 characters (maximum length allowed by the CPND feature). Other factors that can affect the number of characters displayed are the size of the display on the phone and the display of the charges. Names are truncated if their length exceeds 18 characters.

Basic Rate Interface (BRI) phones cannot have names displayed but they can send a name to a called phone.

If the called phone is busy, BNE/Name and Private Number Display does not operate.

When the Call Transfer and Conference features are used, the BNE feature does not provide to the caller the name and number associated with the remote phone. This happens because the User-to-User service 1 only uses SETUP, ALERT and CONNECT messages to convey user signaling.

Most of the options for BRI phones are configured on the phone and not the system. The BNE/ Name and Private Number Display feature does not introduce any new Classes of Service or configurable data related to the phone. The Classes of Service are used by the BNE feature in the same way they are used on EuroISDN or QSIG networks for Calling Number IE or Name display information. Some BRI phones cannot handle the presentation flag in the Calling Number IE. With the prompt TRANS, you can remove the digits in the CLID sent to the BRI phone when the presentation is restricted. For Calling Line Identification Restriction (CLIR), if the BRI phone provides a presentation indication in the CLID information, the PRES option is not used in LD 27. In all other cases, the presentation flag is set based on the PRES configuration. If no CLID is provided by the BRI phone, the default DN of the Terminal Service Profile (TSP) is used. The same rules are used for the public Calling Number, if no private Calling Number is received.

When you configure the D-channel in LD 17 for PRI, LD 16 for BRI (respond UUS1 to the RCAP prompt), it means:

- the system decodes incoming User-to-User IEs for calls terminating locally or originating from this node, if the BNE package is equipped
- the system is allowed to send User-to-User IEs to the public network

Feature interactions

CNIR and CNIP/CONP

The CNIR supplementary service takes precedence over the CNIP supplementary service.

The CNIR supplementary service takes precedence over the CONP supplementary service.

COLR and COLP

The COLR supplementary service takes precedence over the COLP supplementary service.

CLIR and COLR

The same Class of Service controls the CLIR and COLR services. If a user has presentation restricted configured, the number is sent to the other party with the presentation flag set to restricted for incoming and outgoing calls.

Call Pickup

Refer to Figure 56: Call Pickup in a EuroISDN network on page 431. If phone A at one node calls phone B at another node but phone C activates Call Pickup, the name and private number associated with phone A are displayed on phone C, according to the presentation programming of phone A. The display of phone A shows the name and private number associated with phone B while phone B is ringing, if the presentation is allowed. phone A is updated with name and Connected Number information when a user at phone C answers.

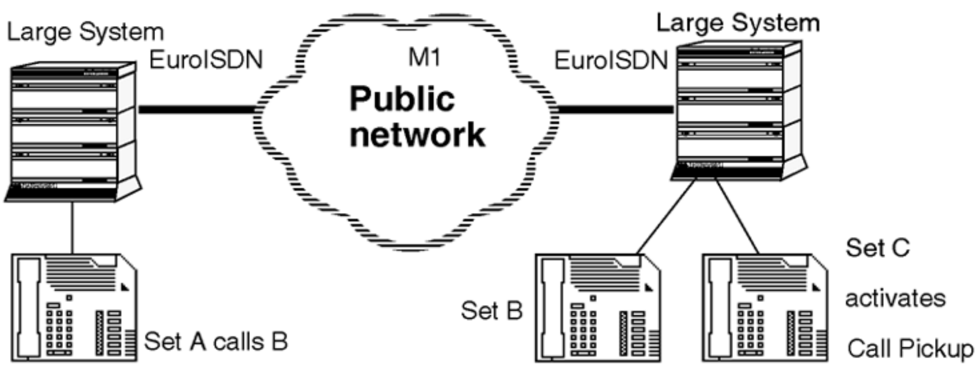


Figure 56: Call Pickup in a EuroISDN network

Call Transfer

Refer to Figure 57: Local Call Transfer on page 432 for an illustration of a local Call Transfer. Refer to Figure 58: External Call Transfer on page 433 for an illustration of an external Call Transfer. Note that in these illustrations, Explicit Call Transfer is not activated.

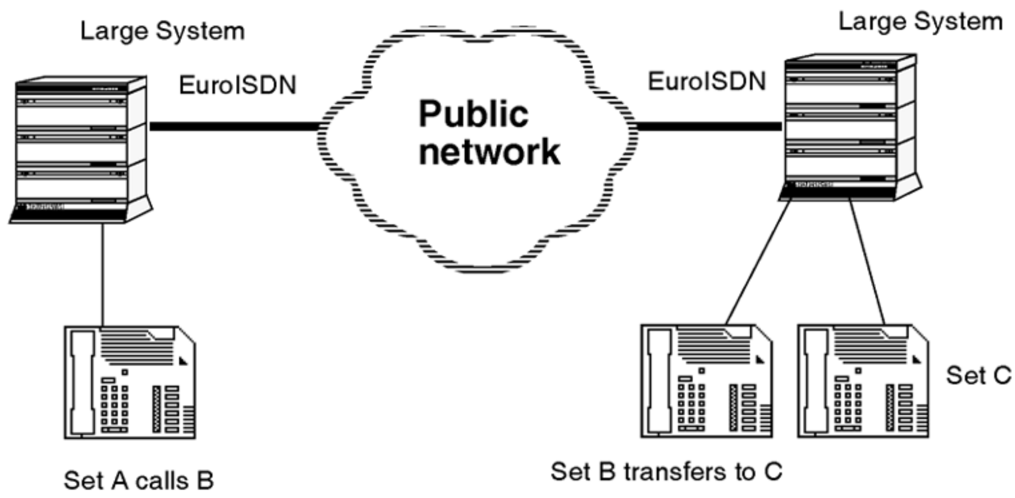


Figure 57: Local Call Transfer

Transfer on ringing (internal)

Figure 57: Local Call Transfer on page 432 illustrates a local transfer of an incoming EuroISDN call that has BNE Name information and a private CLID. For this discussion, assume the user transfers the call while phone C is ringing. When the Call Transfer is completed, the name and private number associated with phone A, display on phone C, according to the presentation programming of phone A. Phone A shows the name and number associated with phone B.

Transfer after answer (internal)

Figure 57: Local Call Transfer on page 432 illustrates a local transfer of an incoming EuroISDN call that has BNE Name information and a private CLID. For this discussion, assume the user transfers the call after a user at phone C answers. When the Call Transfer is completed, the name and private number associated with phone A display on phone C, according to the presentation programming of phone A. Phone A shows the name and number associated with phone B.

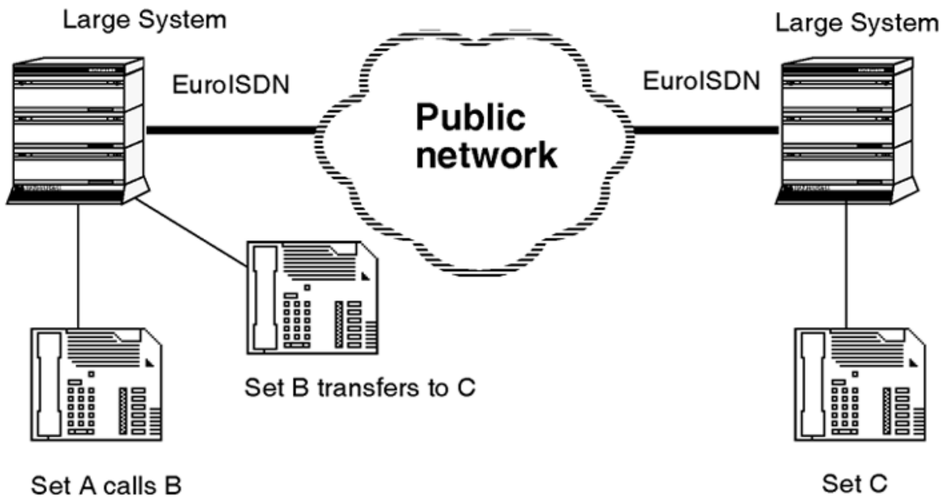


Figure 58: External Call Transfer

Transfer on ringing (external)

Figure 58: External Call Transfer on page 433 illustrates the transfer of a local call over the EuroISDN network to phone C. For this discussion, assume the user transfers the call while phone C is ringing. When the Call Transfer is completed, and while phone C is ringing, the displays do not change. When the user at phone C answers, the user's name and number associated with phone C display on phone A, according to the presentation programming of phone C. Phone C shows the name and private number of phone B.

Transfer after answer (external)

Figure 58: External Call Transfer on page 433 illustrates the transfer of a local call over the EuroISDN network to phone C. For this discussion, assume the user transfers the call after the user at phone C answers. When the Call Transfer is completed, the displays do not change; the user's name and number associated with phone B display on phone A. Phone C shows the name and private number associated with phone B.

Conference

Figure 59: Local Conference on page 434 illustrates a conference call involving parties connected through a EuroISDN network.

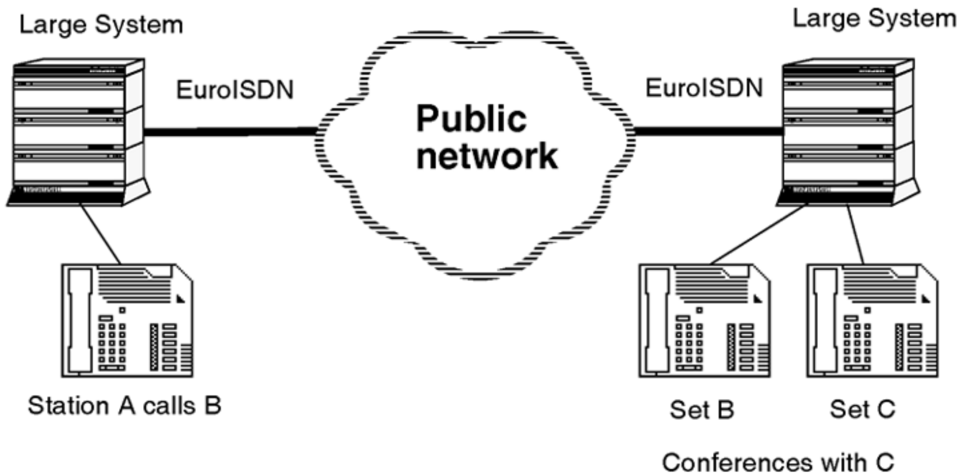


Figure 59: Local Conference

Figure 59: Local Conference on page 434 illustrates an incoming EuroISDN call that has BNE Name information and a private CLID which is conferenced locally. If phone B drops out of the conference call, the user's name and private number associated with phone A display on phone C, if the presentation is allowed, but the display on phone A does not change.

Call Forward

Figure 60: Local Call Forward on page 435 illustrates a local Call Forward situation involving parties connected through a EuroISDN network. Figure 61: External Call Forward on page 436 illustrates an external Call Forward situation involving parties connected through a EuroISDN network. Note that in this illustration, Explicit Call Transfer is not activated.

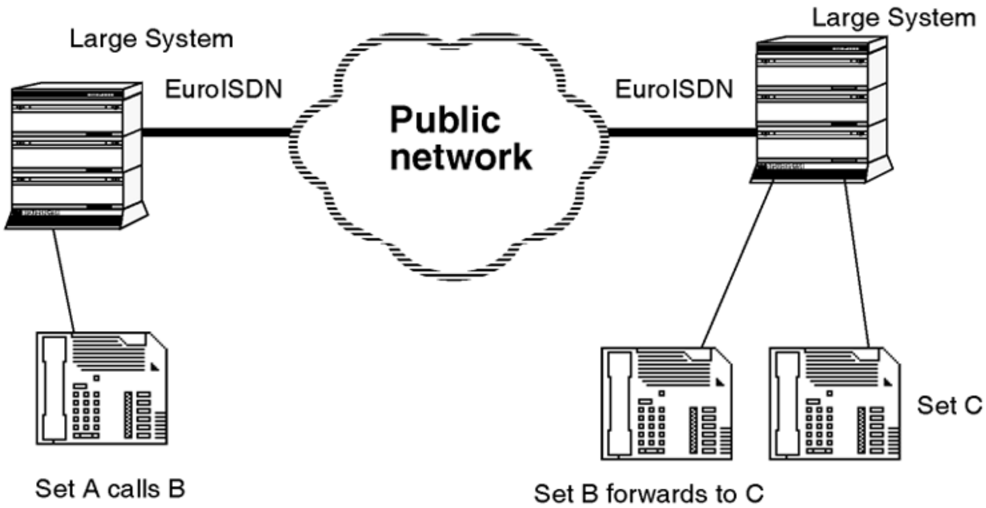


Figure 60: Local Call Forward

Call Forward All Calls (internal)

Figure 60: Local Call Forward on page 435 illustrates an incoming EuroISDN call that has BNE Name information and a private CLID which forwards all calls to a local phone. While phone C is ringing, phone A shows the name and number associated with phone C. The name and number associated with phone A display on phone C, according to the presentation programming of phone A.

Call Forward No Answer (internal)

Figure 60: Local Call Forward on page 435 illustrates an incoming EuroISDN call that has BNE Name information and a private CLID which forwards calls on a no answer condition to a local phone.

After the call forwards, and while phone C is ringing, the display on phone A shows the name and private number associated with phone B. The name and number associated with phone A display on phone C, according to the presentation programming of phone A. When the user at phone C answers, phone A shows the name and number associated with phone C. The display on phone C does not change.

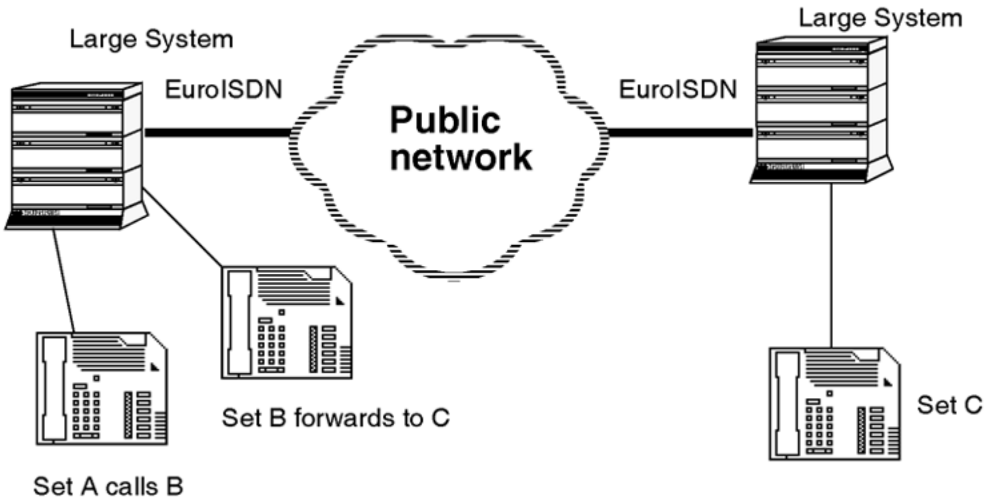


Figure 61: External Call Forward

Call Forward All Calls (external)

A local call can forward over a EuroISDN network, as shown in Figure 61: External Call Forward on page 436. While phone C is ringing, phone A shows the name and number associated with phone C. The name and number associated with phone A are displayed on phone C, according to the presentation programming of phone A.

Call Forward No Answer (external)

A local call can forward unanswered calls over a EuroISDN network as shown in Figure 61: External Call Forward on page 436. After the forwarding occurs, and while phone C is ringing, phone A shows the name and private number associated with phone B. The name and number associated with phone A are displayed on phone C, according to the presentation programming of phone A. When the user at phone C answers, phone A shows the name and number associated with phone C. The display on phone C does not change.

Hunting/Group Hunt

Figure 62: Local Hunting on page 437 illustrates a local Hunting situation.

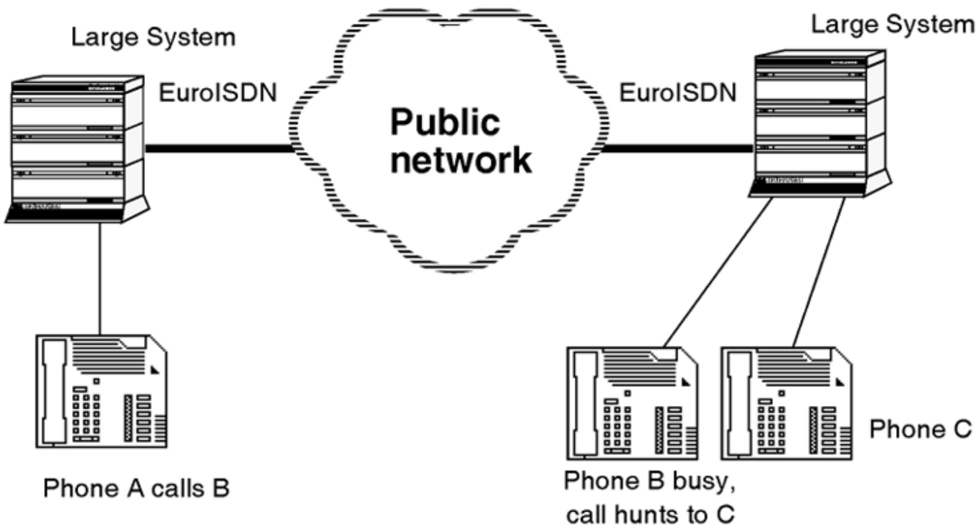


Figure 62: Local Hunting

Figure 62: Local Hunting on page 437 illustrates an incoming EuroISDN call with BNE Name information and a private CLID that redirects to phone C when phone B is busy. As soon as phone C rings, the name and private number associated with phone A display on phone C, according to the presentation programming of phone A. The name and private number associated with phone C are delivered to phone A.

Advice of Charge (AOC)

Figure 63: AOC and Charge Display on page 437 illustrates an example of a situation involving the Advice of Charge feature.

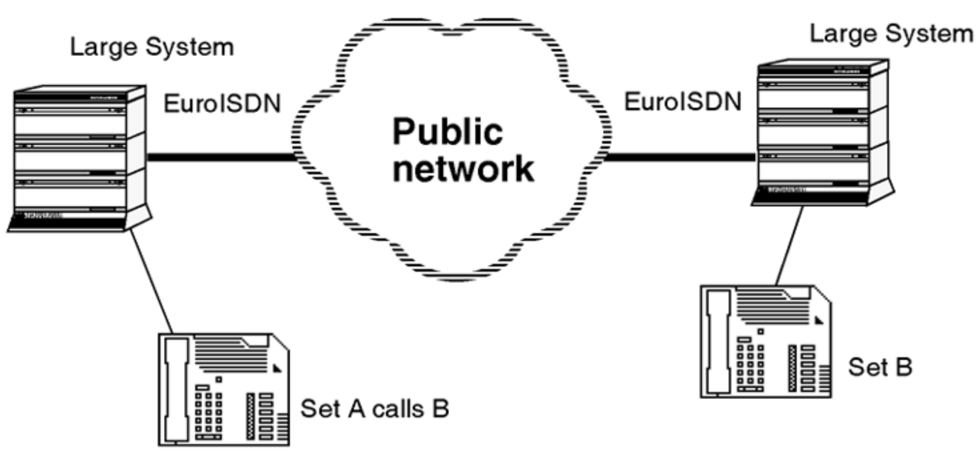


Figure 63: AOC and Charge Display

An outgoing EuroISDN call, carrying BNE signaling, as shown in Figure 63: AOC and Charge Display on page 437, is charged by the Central Office. If phone A is a Meridian Modular Digital phone and AOC Real Time Display is configured, the charge is displayed in the right corner of the first line, when it is received. The display of charge takes precedence over the display of name. The name displayed on phone A is truncated if there is not enough space to display both the name and the charge.

Display of Access Prefix

The private numbers provided by the BNE feature are displayed with the prefixes configured by the Display of Access Prefix on the CLID feature for a private numbering plan.

Networking feature interactions

Call Diversion (diversion notification sent):

Call Forward All Calls (Call Forward Unconditional)

Figure 64: Call Diversion in networking on page 439 illustrates an incoming EuroISDN call that has BNE Name information and a private CLID forwarding all calls to phone C over the EuroISDN network. After the call forwards, the BNE information name and number are replaced by the notification numbers provided by the Call Diversion feature. While phone C is ringing, phone A shows the name associated with phone C.

Call Forward No Answer

Figure 64: Call Diversion in networking on page 439 illustrates an incoming EuroISDN call that has BNE Name information and a private CLID forwarding unanswered calls to phone C over the EuroISDN network. After the call forwards, the BNE information name and number are replaced by the notification numbers provided by the Call Diversion feature. Phone A shows the name associated with phone C when the call is established.

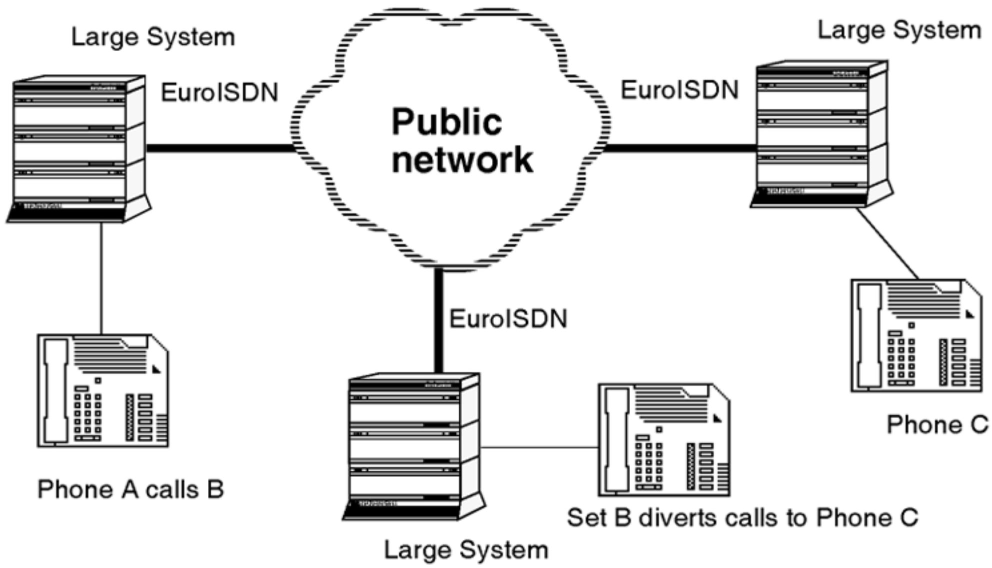


Figure 64: Call Diversion in networking

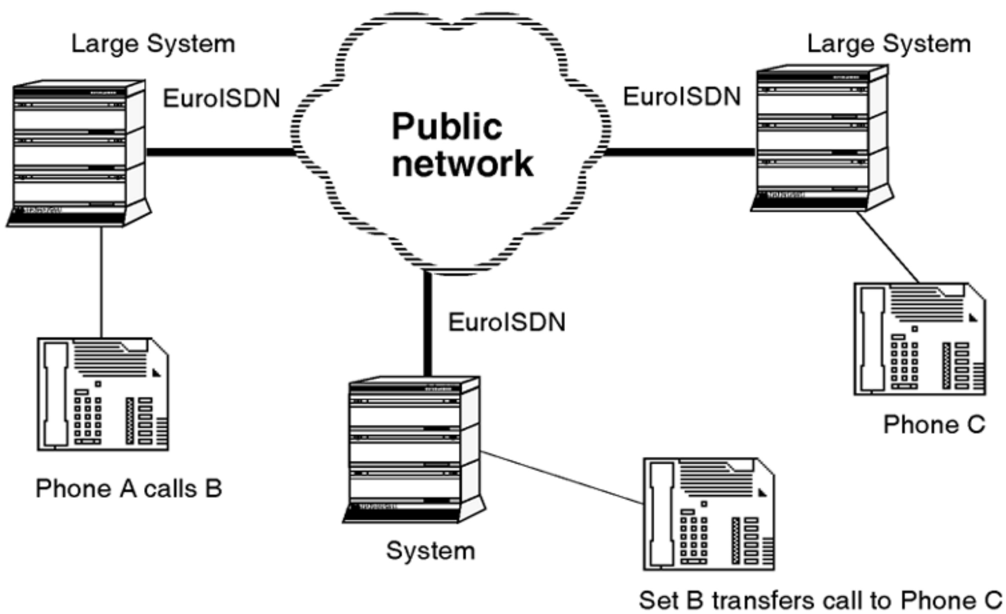


Figure 65: Call Transfer across a network

Explicit Call Transfer (Call Transfer notification sent)

Figure 65: Call Transfer across a network on page 439 illustrates a Call Transfer across a network. Before the transfer is completed, phone C shows the name and number associated with phone B. Phone A shows the name and number associated with phone B. After the transfer, the BNE information name and number are replaced by the notification numbers provided by the Call Transfer feature.

Feature packaging

Business Network Express (BNE) package 367 is introduced with this feature.

The following software packages are required for Business Network Express and BRIT:

- Integrated Services Digital Network (ISDN) package 145
- International (INTL) package 161
- Basic Rate Interface (BRI) package 216
- Basic Rate Trunk Application (BRIT) package 233
- EuroISDN (EURO) package 261
- Business Network Express (BNE) package 367
- and at least one of the following three packages:
    - Coordinated Dialing Plan (CDP) package 57
    - Basic Automatic Route Selection (BARS) package 58
    - Network Alternate Route Selection (NARS) package 59

The following software packages are required for Business Network Express and PRI2:

- Integrated Services Digital Network (ISDN) package 145
- 2.0 Mbit Primary Rate interface (PRI2) package 154
- International (INTL) package 161
- International Primary Rate Access (IPRA) package 202
- Multipurpose Serial Data Link (MSDL) package 222
- EuroISDN (EURO) package 261
- Business Network Express (BNE) package 367
- and at least one of the following three packages:
    - Coordinated Dialing Plan (CDP) package 57
    - Basic Automatic Route Selection (BARS) package 58
- Network Alternate Route Selection (NARS) package 59

The following software packages are required for Business Network Express Name Display:

- Call Party Name Display (CPND) package 95
- Flexible Feature Code (FFC) package 139
- Calling Party Privacy (CPP) package 301

The following software packages are required for Business Network Express Private CLID and COLP:

- Digit Display (DDSP) package 19
- Flexible Feature Code (FFC) package 139
- Calling Party Privacy (CPP) package 301

Feature implementation

Task summary list

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

Table 207: LD 95 - Create a CPND data block.

Prompt	Response	Description
REQ	NEW	New.
TYPE	CPND	CPND data block.
CUST	xx	Customer number, as defined in LD 15.

Table 208: LD 95 - Create a new name string.

Prompt	Response	Description
REQ	NEW	Add new data.
TYPE	NAME	Create a new name string.
CUST	xx	Customer number, as defined in LD 15.
...	...	
DN	xxxx	DN linked with the name string.

Table 209: LD 10 - Allow or deny name and digit display on analog (500/2500- type) phone.

Prompt	Response	Description
REQ:	NEW CHG	Add new data, or change existing data.
TYPE:	500	Type of phone.
CUST	xx	Customer number, as defined in LD 15.
...	...	
FTR	CPND	Allow CPND name assignment on this phone (not required if CPND is programmed in LD 95).
CLS	CNDA (CNDD)	Allow (deny) user names to be displayed on this phone.
	NAMA (NAMD)	Allow (deny) name display on the far end.
	DDGA (DDGD)	Allow (deny) digit display on the far end.
	CLBA (CLBD)	Allow (deny) calling number and name per-call blocking.

Table 210: LD 11 - Allow or deny name and digit display on Meridian 1 proprietary phones.

Prompt	Response	Description
REQ:	NEW	Add new data.
	CHG	Change existing data.
TYPE:	xxxx	Phone type
CUST	xx	Customer number, as defined in LD 15.
...	...	

Prompt	Response	Description
CLS	CNDA (CNDD) DNDA (DNDD)	Allow (deny) user names to be displayed on this phone. Allow (deny) display on this phone of the originally dialed phone's name on redirected calls.
	NAMA (NAMD) DDGA (DDGD) CLBA (CLBD)	Allow (deny) digit display on the far end. Allow (deny) calling number and name per-call blocking. Allow (deny) calling number and name per-call blocking.

Table 211: LD 12 - Allow or deny name display on 2250 consoles.

Prompt	Response	Description
REQ:	NEW	Add new data.
	CHG	Change existing data.
TYPE:	2250	Attendant Console type.
CUST	xx	Customer number, as defined in LD 15.
...	...	
CPND	CNDA (CNDD)	Allow (deny) user names to be displayed on this console.

Table 212: LD 27 - Configure the BRI Digital Subscriber Loop.

Prompt	Response	Description
REQ	NEW	Add new data.
TYPE	TSP	Terminal Service Profile.
DSL		DSL location
	lll s cc dsl#	Format for Large System and Avaya Communication Server 1000E (Avaya CS 1000E) system, where: lll (superloop) = 0-156 (must be a number divisible by 4) s (shelf) = 0-1 cc (card) = 0-15 dsl# (DSL location) = 0-7
	c dsl#	Format for Media Gateway 1000 B (Avaya MG 1000B), where: c (card) = 1-20 dsl# (DSL number) = 0-7
DN	xxxx	DN associated with the TSP.
CLIP	(YES) NO	Calling Line ID presentation service (allowed), denied.
PRES	(YES) NO	Display of party number on far end (allowed), denied.
TRANS	(YES) NO	Party number digits from far end transmitted (not transmitted), if the presentation is restricted.

Table 213: LD 57 - Assign Flexible Feature Code for Name Display.

Prompt	Response	Description
REQ	NEW	Add new data.
TYPE	FFC	Flexible Feature Code.
CUST	xx	Customer number, as defined in LD 15.
...	...	
CPP	xxxx	Calling Party Privacy feature access code. Four digit maximum. Prompted only if CPP software package is equipped.

Table 214: LD 16 - Configure D-channel for User-to-User service 1 (BRI).

Prompt	Response	Description
REQ	CHG	Change existing data.
TYPE	RDB	Route Data Block.
...	...	
IFC	EURO E403	Interface type. EuroISDN interface - complies with ETS 300 102 ETSI standard. EuroISDN interface - complies with ETS 300 403-1 ETSI standard.
...	...	
RCAP	UUS1	User-to-User implicit service 1.

Table 215: LD 17 - Configure D-channel for User-to-User service 1 (PRI).

Prompt	Response	Description
REQ	CHG	Change existing data.
TYPE	RDB	Route Data Block
...	...	
IFC	EURO E403	Interface type. EuroISDN interface - complies with ETS 300 102 ETSI standard. EuroISDN interface - complies with ETS 300 403-1 ETSI standard.
...	...	
RCAP	UUS1	User-to-User implicit service 1.

Table 216: LD 86 - Configure Route List Index for BNE feature.

Prompt	Response	Description
REQ	NEW	Add a new Route List Index.
	CHG	Change an existing Route List Index.

Prompt	Response	Description
CUST	xx	Customer number, as defined in LD 15.
FEAT	RLB	Route List Data Block.
RLI	xxx	Route List Index.
ENTR	xx	Entry number.
...	...	
FSNI		
BNE	YES	Business Network Express/Name Display, Private CLID and COLP allowed.
	(NO)	Business Network Express/Name Display, Private CLID and COLP denied. BNE prompt appears for EuroISDN routes only.

Feature operation

Refer to the Calling Party Privacy feature in *Avaya Features and Services Fundamentals, NN43001-106* for information on the use of the CPP FFC.

Chapter 16: Call Charge Keeping

Contents

This section contains information on the following topics:

Feature description

On trunk calls between the system and 1TR6 Central Office connection, call charge information can be taken from the ISDN network and used by the system in its call charge records. Accumulated charging information for the call is interfaced with the system Periodic Pulse Metering function, to provide a transparent call-charging feature to the customer.

On the node with a 1TR6 connection, call charge information is received by the system from the network as part of the connect data. This information is temporarily stored by the system. 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) phone, a Meridian 1 proprietary telephone, an Attendant Console or a trunk in a tandem call.)

Call charging under 1TR6 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
- Meridian 1 proprietary telephone access to MR data (on a phone with digit display and an MRK key)
- CDR on Multiple Call Transfer for outgoing calls

Operating parameters

Call charge keeping is only supported between the system and 1TR6 Central Office connectivity.

Feature interactions

There are no feature interactions associated with this feature.

Feature packaging

This feature requires the following packages:

- Call Detail Recording (CDR) package 4
- Periodic Pulse Metering/Message Registration (MR) package 101
- Integrated Services Digital Network (ISDN) package 145

Feature implementation

There are no specific implementation procedures for this feature

Feature operation

No specific operating procedures are required to use this feature.

Chapter 17: Call Connection Restriction

Contents

This section contains information on the following topics:

Feature description

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

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

Operating parameters

There are no operating parameters associated with this feature.

Feature interactions

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 can 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.

Feature packaging

This feature is included in base System Software.

Feature implementation

Table 217: LD 15 - Configure the Customer Data Block to allow Call Connection Restriction.

Prompt	Response	Description
REQ:	CHG	Change existing data block.
TYPE:	NET	Networking data.
CUST		Customer number
	0-99	Range for Large System and Avaya Communication Server 1000E (Avaya CS 1000E) system.
	0-31	Range for Media Gateway 1000 B (Avaya MG 1000B).
ISDN	YES	Integrated Services Digital Network allowed for customer.
CNTP	LDN	Listed Directory Number.
RCNT	0-(5)	Redirection Count for ISDN calls.
PSTN	(NO) YES	Public Service Telephone Networks.
TNDM	0-(15)-31	Tandem Threshold/Loop Avoidance Limit.
PCMC	0-(15)-31	Pulse Code Modulation Conversions permitted.
SATD	0-(1)-5	Satellite Delays.

Feature operation

No specific operating procedures are required to use this feature.

Chapter 18: Call Forward/Hunt Override Via Flexible Feature Code

Contents

This section contains information on the following topics:

Feature description

Call Forward/Hunt Override Via Flexible Feature Code (FFC) allows phone users (with a specific Class of Service) and attendants, to override Intercept Computer Call Forward (ICP-CFW), Call Forward All Calls, Call Forward No Answer, Hunt and Make Set Busy by entering an FFC. In order to use this feature, the originating phone 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 phone is rung regardless of any diversion. If the dialed phone is busy and has Hunt active, the calling party will terminate on the wanted phone and receive a busy indication. Phones without Call Forward/Hunt Override denied (CFHD) Class of Service will not be able to use the Call Forward/Hunt Override Via FFC feature.

Call Forward/Hunt Override Via FFC works in network environments with system 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 phone (that is, it has to be dialed before dialing the DN that has Call Forward All Calls, Intercept Call Forward, Call Forward No Answer, Hunt, or Make Set Busy active).

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

The Call Forward/Hunt Override FFC is normally not allowed to be defined as a Flexible DN, External Flexible DN, Hunt DN, or External Hunt DN.

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

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

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 stand-alone 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.

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 phone are overridden by Call Forward/Hunt Override Via FFC with the same limitations as for other phones.

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.

BRI

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

BRIT

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

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 phone, if Break-in is enabled.

Call Redirection by Time of Day

Call Forward/Hunt Override Via FFC takes precedence over Call Redirection by Time of Day (CRTOD).

Call Transfer

A phone 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 phone 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 phone is encountered, it is possible to Camp-on to the phone, even if Call Forward/Hunt Override Via FFC has been activated.

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.

DPNSS1

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

Do Not Disturb (DND)

This feature is not overridden by the Call Forward/Hunt Override Via FFC feature. Trying to override DND results in DND treatment.

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.

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 FFC feature. If a phone activating Call Forward/Hunt Override FFC encounters a busy phone, no hunt steps are performed; the call terminates on the DN and a busy tone is 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.

Last Number Redial

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

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 old ICP Override FFC. To get the functionality of ICP override, the ACD agent ICP position must have the new Class of Service CFHA.

Idle Extension Notification (IEN)

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

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 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

Using the feature Priority Override is possible after using the Call Forward/Hunt Override FFC and encountering a busy phone.

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 phone for which the Call Forward/Hunt Override FFC was used (that is, the phone 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 phone that does not answer. Ring Again No Answer can be placed against the phone for which the Call Forward/Hunt Override FFC was used (that is, the phone 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 phone, it is possible to activate SACP, if it is applicable.

Speed Call

The Call Forward/Hunt Override FFC can be stored in a speed call list.

Feature packaging

For stand-alone environments, the following package is required:

- Flexible Feature Codes (FFC) software package 139

For network environments, the following software package must also be provided:

- Network Attendant Service (NAS) package 159

Note:

Attendant Overflow Position (AOP) package 56 must be restricted, as it is mutually exclusive with Network Attendant Service.

Feature implementation

Task summary list

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

1. Table 218: LD 57 - Define FFC for Call Forward/Hunt Override analog (500/2500-type). on page 460
2. Table 219: LD 10 - Set Class of Service for the Forward/Hunt Override Via FFC. on page 461
3. Table 220: LD 11 - Set Class of Service for the Forward/Hunt Override through FFC for Meridian 1 Propriety Phones. on page 461
4. Table 221: LD 18 - Configure ABCD key for the Forward/Hunt Override Via FCC. on page 461

Table 218: LD 57 - Define FFC for Call Forward/Hunt Override analog (500/2500-type).

Prompt	Response	Description
REQ	NEW	Add new data.
	CHG	Change existing data.
TYPE	FFC	Flexible Feature Code.
...		
CODE	CFHO	Call Forward/Hunt Override Via FFC.

Prompt	Response	Description
CFHO	nnnn	Call Forward/Hunt FFC.

Table 219: LD 10 - Set Class of Service for the Forward/Hunt Override Via FFC.

Prompt	Response	Description
REQ:	NEW	Add new data.
	CHG	Change existing data.
TYPE:	500	Type of phone.
...		
CLS	(CFHD) CFHA	Call Forward/Hunt Override Via FFC is (denied) or allowed.

Table 220: LD 11 - Set Class of Service for the Forward/Hunt Override through FFC for Meridian 1 Propriety Phones.

Prompt	Response	Description
REQ:	NEW CHG	Add new data, or change existing data.
TYPE:	a...a	Telephone type. Type **?** for a list of possible responses.
...		
CLS	(CFHD) CFHA	Call Forward/Hunt Override Via FFC is (denied) or allowed.

Table 221: LD 18 - Configure ABCD key for the Forward/Hunt Override Via FCC.

Prompt	Response	Description
REQ	NEW	Add new data.
	CHG	Change existing data.
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.

Prompt	Response	Description
D	CFHO*FFC *	CFHO is assigned to key D.

Feature operation

There are no operating procedures specified for this feature.

The user receives 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 phone is idle, the phone is rung regardless of any diversion (for example, Call Forward All Calls, Intercept Call Forward, Call Forward No Answer, or Hunt) or Make Set Busy on the phone.

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

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

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

Chapter 19: Calling Line Identification in CDR

Contents

This section contains information on the following topics:

Feature description

Call Detail Recording (CDR) records information about selected calls for accounting purposes. For each call, CDR identifies the calling and called parties and notes the time and duration of the call. A record describing the complete call is output by the system when the call is terminated. The following five recording options are available and can be specified by the customer in any combination for each trunk route:

- all outgoing calls
- all outgoing toll calls
- outgoing answered calls
- outgoing answered toll calls
- all incoming calls

For detailed descriptions of the Call Detail Recording feature, please refer to *Avaya Call Detail Recording Fundamentals, NN43001-550*.

The Calling Line ID in CDR feature is an enhancement to the CDR feature. The description which follows applies to a stand-alone as well as an ISDN PRI environment.

If the Calling Line ID in CDR feature is enabled, the CLID number is included in Call Detail Recording (CDR) records. This gives the customer the calling station's ID, even from a tandem node. This information allows the customer to charge the calling party for services rendered in connection with an incoming call. For example, calls to an attorney can be accurately charged to the calling client.

The CLID information in the call SETUP message is added to all applicable CDR message types, in both TTY message format and the compressed binary formats for downstream processing. If the CLID information is not included in the SETUP message, it cannot be printed.

In the TTY output, the CLID information is printed on the second line, as shown in Table 3. The field is always 16 characters: the actual CLID digits, followed by X's to total 16.

Table 222: CLID number in the TTY output

Rec Type	Rec No	CustNo	OrigID	TerID	AuxID lll.s.cc.uu	Date mm/dd	Time hh:mm	Dur - ation hh:mm:s	Digits
N	001	00	DN499	A001000	027.1.02.1	06/28	10:14	00:00:20	95559124
N	002	00	T002001	DN5000		06/28	10:15	00:00:40	
95552222xxxxxxxx									

Note:

The CLID field always displays 16 characters. The feature inserts an "x" for each missing character.

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 can be charged 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 66: CDR in Multi-site Configuration on page 465 illustrates a network with three system 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.

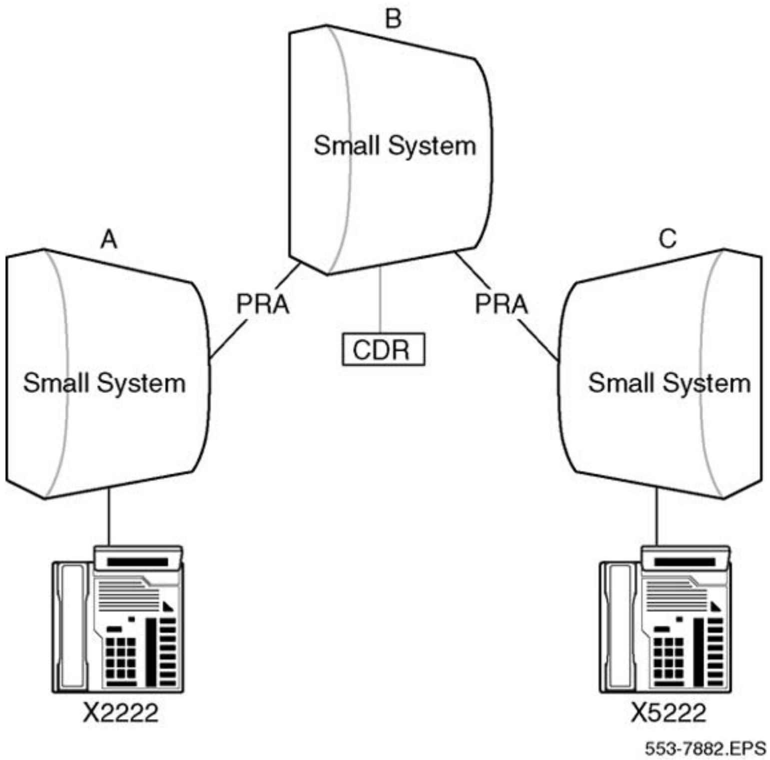


Figure 66: CDR in Multi-site Configuration

Operating parameters

There are no operating parameters associated with this feature.

Feature interactions

INIT ACD Queue Call Restore

Call information associated with Calling Line Identification (CLID) is lost after system initialization and call restoration.

Feature packaging

This feature requires the Calling Line Identification in Call Detail Recording (CCDR) package 118. The following packages are also required:

- Call Detail Recording (CDR) package 4
- Call Detail Recording on Teletype Terminal (CTY) package 5
- Network Alternate Route Selection (NARS) package 58
- Integrated Services Digital Network (ISDN) package 14
- ISDN signaling Link (ISL) package 147 or
- 2.0 Mbit Primary Rate Interface (PRI2) package 154

Feature implementation

Task summary list

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

1. Table 223: LD 17 - Change the Configuration Record to enable CLID. on page 466

   It is assumed that ISDN PRI is configured for the customer.
2. Table 224: LD 15 - Change the Customer Data Block to configure CLID. on page 467Table 226: LD 16 - Allow CDR records in the Trunk Route Data Block. on page 468
3. Table 225: LD 17 - Allow CDR to be printed on the TTY terminal. on page 468
4. Table 226: LD 16 - Allow CDR records in the Trunk Route Data Block. on page 468

Table 223: LD 17 - Change the Configuration Record to enable CLID.

Prompt	Response	Description
REQ	CHG	Change existing data
TYPE	PARM	Change parameters.
...		

Prompt	Response	Description
CLID	YES	Enable CLID.

Table 224: LD 15 - Change the Customer Data Block to configure CLID.

Prompt	Response	Description
REQ:	NEW	Add new customer data.
	CHG	Change existing customer data.
TYPE:	NET	Networking Data.
CUST		Customer number
	0-99	Range for Large System and Avaya Communication Server 1000E (Avaya CS 1000E) system.
	0-31	Range for Media Gateway 1000 B (Avaya MG 1000B).
ISDN	YES	Change ISDN options.
- PNI	1-32700	Private Network Identifier.
CLID	YES	Calling Line Identification.
- SIZE	0-(256)-1000	Number of CLID entries required
-INTL	xx	Country code (1-4 digit). X to remove.
- ENTRY	xx	CLID entry to be configured.
- PFX1	xxxx	Prefix (area) code for International PRI.
- PFX2	xxxx	Central Office Prefix for IPRA.
- HNPA	100-999	Home Number Plan Area code.
- HNXX	100-999	Prefix for Central Office.
- HLOC	100-9999	Home Location Code (ESN).
- LSC	xxxx	Local steering code.
- CNTP	(PDN)	CLID feature displays the phone's Prime DN.
	LDN	CLID feature displays the customer's Listed Directory Number (LDN).
- RCNT	0-(5)	Maximum inter-node hops in a network redirection call.

Note:

Attendant Consoles have a Listed Directory Number (LDN) only.

Table 225: LD 17 - Allow CDR to be printed on the TTY terminal.

Prompt	Response	Description
REQ	CHG	Change.
TYPE	ADAN	All input/output devices.
ADAN	CHG TTY XX	Change I/O device where xx = port.
...		
USER	CTY	Use the TTY for CDR records.

Table 226: LD 16 - Allow CDR records in the Trunk Route Data Block.

Prompt	Response	Description
REQ	NEW	Add new data.
	CHG	Change existing data.
TYPE	RDB	Route Data Block.
CUST	xx	Customer number, as defined in LD 15.
...		
CDR	YES	Allow CDR.
- INC	YES	Print CDR information for CLID on incoming trunks.

Feature operation

No specific operating procedures are required to use this feature.

Chapter 20: Attendant Through Dialing Networkwide

Contents

This section contains information on the following topics:

Feature description

Attendant Through Dialing Networkwide extends the functionality of through dialing through 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.

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.

Attendant Through Dialing Networkwide on page 469 illustrates Attendant Through Dialing Networkwide.

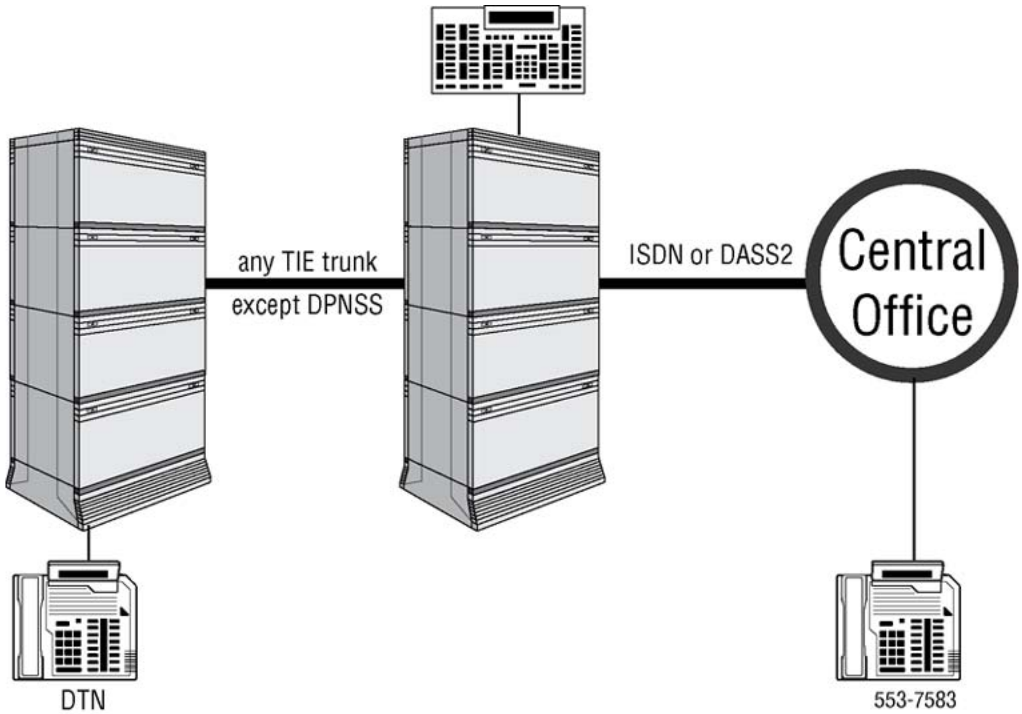


Figure 67: Attendant Through Dialing Networkwide

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 227: Numbering Plans and Attendant Release of external trunk on page 470 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 227: 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

Numbering Plan used to seize external trunk	Destination (DEST) becomes lit
Coordinated Dialing Plan	After Trunk Steering Code

Operating parameters

This feature supports 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 DTI2 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, Scheduled Access Restrictions and Trunk Barring are not affected by Attendant Through Dialing Networkwide.

This feature is not supported on phones configured with Dial Pulse (CLS = DIP). Attendant Through Dialing Networkwide is only supported on phone 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 phone 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 phone 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 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 LD 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

This feature 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 plan packages:

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

Feature implementation

Task summary list

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

1. Table 228: LD 15 - Allow Attendant Through Dialing Networkwide. on page 474
2. Table 229: LD 15 - Configure Improved End-to-End Signaling. on page 475
3. Table 230: LD 17 - Allow Calling Line Identification (CLID) field in Call Detail Recording (CDR) records. on page 475

Table 228: LD 15 - Allow Attendant Through Dialing Networkwide.

Prompt	Response	Description
REQ:	CHG	Change existing data.
TYPE:	ATT	Attendant Console data block.
CUST		Customer number
	0-99	Range for Large System and Avaya Communication Server 1000E (Avaya CS 1000E) system.
	0-31	Range for Media Gateway 1000 B (Avaya MG 1000B).
OPT	(ATDA)	Attendant Through Dialing Allowed (default). ATDD = Attendant Through Dialing Denied.

Note:

The configuration of Improved End-to-End Signaling in LD 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.

Note:

Improved End-to-End Signaling is provided when EEST = YES and DTMF = NO.

Table 229: LD 15 - Configure Improved End-to-End Signaling.

Prompt	Response	Description
REQ:	CHG	Change existing data.
TYPE:	FTR	Customer Features and options.
CUST		Customer number
	0-99	Range for Large System and CS 1000E system.
	0-31	Range for Media Gateway 1000B.
EEST	YES	Send feedback tone to the originator of End-to-End Signaling.
- DTMF	NO	Improved End-to-End Signaling for single tone feedback.

Table 230: 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.

Feature operation

1. The Calling party dials an attendant that is located either on the same node as the caller or another node.
2. The 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, if 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 an external number.

Chapter 21: Calling Party Privacy

Contents

This section contains information on the following topics:

Feature description

The Calling Party Privacy (CPP) feature enables the system to support the blocking of a Calling Party's Number and Name from being displayed at the terminating phone on a per-call basis. Users can dial a Calling Party Privacy code (for example, *67 from a Meridian 1 Proprietary Phone or 1167 from an analog [500/2500-type] phone) to prevent their phone number and name from being displayed on a receiving phone across the Public Switched Telephone Network (PSTN). Internal calls within the system 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 phones 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 system 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, can 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 (for example, *67). No Privacy Indicator is expected for an incoming non-ISDN trunk call. For an ISDN call between two system 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 (that is, 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 Calling Party Privacy Enhancement (CPPE) feature provides a route option to ignore the Calling Party Privacy Indicator on incoming calls received on BRI trunks.

This feature is supported on the following BRI trunk interfaces:

- MCDN Enterprise networking variants (including the peer to peer variant- SL1 and the enterprise UNI variant- SL100).
- Euro ISDN (All variants)
- APAC (All variants)
- QSIG (ISO and ETSI)

Note:

This feature is not supported on BRI lines.

When the Privacy Indicator Ignore (PII) prompt is set to YES in LD 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 phone, 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 LD 16, the Calling Party Privacy Indicator is honored, and the existing functionality is maintained.

A route option, Auxiliary processor applications (AUXP), in LD 16, enhances the system's ability to honor or ignore the Privacy Indicator for a Calling Party Privacy call, based on the incoming route, for auxiliary applications such as Contact Center Manager (CCM).

The configurable AUXP prompt is dependent on the PII prompt. If the PII prompt is checked, then the AUXP prompt is automatically checked, and remains so until the PII prompt remains checked. AUXP cannot be modified by the user unless PII is unchecked.

In LD 16, when PII is set to YES, the AUXP prompt is automatically set to YES. When PII is set to NO, AUXP remains set to YES and can be modified by the administrator. When PII is set to YES and the presentation is enabled, the AUXP prompt has no effect.

When the AUXP prompt is set to YES in LD 16, the CLID Presentation Indicator and the CPND Indicator (if it exists) are changed from restricted/denied to allowed for auxiliary applications. When AUXP is set to NO (default) in LD 16, there is no change to the CLID Presentation Indicator.

Example: PII and AUXP settings

Consider a scenario where an originating node, with blocked CLID, calls Node A (Set A and Aux A) and Node B (Set B and Aux B). Table 231: PII and AUXP settings on page 479 displays the working of PII and AUXP settings.

Table 231: PII and AUXP settings

<table>
<tr><th>Settings at Node A</th><th colspan="3">Settings at Node B</th></tr>
<tr><td></td><td>PII - NO AUXP - NO</td><td>PII - NO AUXP - YES</td><td>PII - YES AUXP - YES</td></tr>
<tr><td>PII - NO AUXP - NO Call to Node A</td><td colspan="3">Set A: No presentation Aux A: No presentation Set B: No impact</td></tr>
<tr><td>PII - NO AUXP - NO Call to Node B</td><td>Set A: No impact Set B: No presentation Aux B: No presentation</td><td>Set A: No impact Set B: No presentation Aux B: Presented</td><td>Set A: No impact Set B: Presented Aux B: Presented</td></tr>
<tr><td>PII - NO AUXP - YES Call to Node A</td><td colspan="3">Set A: No presentation Aux A: Presented Set B: No impact</td></tr>
<tr><td>PII - NO AUXP - YES Call to Node B</td><td>Set A: No impact Set B: No presentation Aux B: No presentation</td><td>Set A: No impact Set B: No presentation Aux B: Presented</td><td>Set A: No impact Set B: Presented Aux B: Presented</td></tr>
<tr><td>PII - YES AUXP - YES Call to Node A</td><td colspan="3">Set A: Presented Aux A: Presented Set B: No impact</td></tr>
<tr><td>PII - YES AUXP - YES Call to Node B</td><td colspan="3">Set A: No impact Set B: Presented Aux B: Presented

Note:

For all calls tandem through Node A to Node B, the setting in Node A to ignore the presentation indicator, converts the call to 'unrestricted'. Hence, when PII is YES at Node A, all calls are treated as 'unrestricted' at Node B.</td></tr>
</table>

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 (for example, 1167) to be used for rotary phones, across the Public Switched Telephone Network (PSTN) or Meridian Customer Defined Network (MCDN).

The code which is outpulsed on non-ISDN analog or digital trunks can also be flexibly defined on a per-route basis, for station phones for public and private network calls. 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 phones, 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 DMS-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 system 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 phone.

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 outpulsing the Privacy Indicator will include the Privacy Indicator in the outpulsed digits field when generating the CDR records if the outgoing non-ISDN trunk route has Outpulsed 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 tandemed to the far end to inhibit the display of the Calling Party Name or Number provided that the outgoing trunk route on the tandem node also has 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 phone (i.e, at the tandem node) from being displayed on the terminating phone. 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 phone.

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 phone.

Call Pickup Network Wide

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

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 phone. 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 phone of the final terminating party.

Calling Line Identification Restriction (CLIR)

The Flexible Feature Code is not supported on BRI phones. 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 phone 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 phone, 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.

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.

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 system switch locally, the received 10-digit Automatic Number Identification (ANI) will be displayed on the terminating phone 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 through 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 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 phone 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.

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.

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 can 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 Phone, 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

This feature requires the following packages:

- 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 implementation

Task summary list

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

1. Table 232: LD 57 - Define the FFC for CPP feature. on page 489
2. Table 233: LD 16 - Define Privacy Indicators. on page 490
3. Table 234: LD 10/11 - Activate Calling Party per-line blocking. on page 491

Table 232: LD 57 - Define the FFC for CPP feature.

Prompt	Response	Description
REQ	NEW	Add new data.
	CHG	Change existing data.
TYPE	FFC	Flexible Feature Code.
CUST	xx	Customer number, as defined in LD 15.
FFCT	(NO) YES	Flexible Feature Confirmation Tone.
...		
CODE	CPP	FFC type to be altered. <CR> means that no FFC types are prompted.
...		
CPP	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 Phone, an "*" can be entered as the first digit. A suggested value is *67. CPP will be prompted until a <CR> is entered.

Note:

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.

Table 233: LD 16 - Define Privacy Indicators.

Prompt	Response	Description
REQ	NEW	Add new data.
	CHG	Change existing data.
TYPE	RDB	Route Data Block.
CUST	xx	Customer number, as defined in LD 15.
TKTP	COT	Central Office Trunk data block.
	DID	Direct Inward Dialing trunk data block.
	FEX	Foreign Exchange trunk data block.
	WAT	Wide Area Telephone Service trunk data block.
...		
DTRK	YES	Digital trunk route.
DGTP	xx	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	(*67) 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.

Prompt	Response	Description
- DPPI	(1167) 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. **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). For further details on the PII route option, refer to Calling Party Privacy Enhancement on page 478.

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.

Table 234: LD 10/11 - Activate Calling Party per-line blocking.

Prompt	Response	Description
REQ:	NEW CHG	Add new data, or change existing data.
TYPE:	nnnn	Type of phone.
TN		Terminal number
	l s c u	Format for Large System and Avaya Communication Server 1000E (Avaya CS 1000E) system, where l = loop, s = shelf, c = card, u = unit.
	c u	Format for Media Gateway 1000 B (Avaya MG 1000B), where c = card and u = unit.
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.

Feature operation

Any outgoing call initiated from a phone 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 phone types, except BRI phones.
2. Request CPP on BRI phones by setting the softkey "ID PRES" (if it exists) to "Denied" state or the "PRES" prompt to "NO" in LD 27. Flexible Feature Codes are not supported on BRI phones.

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.

Chapter 22: Calling Party Privacy Override

Contents

This section contains information on the following topics:

Applicable regions

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

Feature description

Calling Party Privacy Override (CPPO) enhances the functionality of the Calling Party Privacy (CPP) feature. With Calling Party Privacy Override, calling party information can be selectively unblocked on a per-call basis.

The Calling Party Privacy (CPP) feature enables the system to permanently block the Calling Party Number and Name from being displayed on the terminating phone across the Public Switched Telephone Network (PSTN). This permanent blocking occurs when Class of Service is set to Calling Party Number and Name per-line blocking allowed (CLBA).

When Class of Service is set to Calling Party Number and Name per-line blocking denied (CLBD), the user can block the Calling Party Number and Name on a per-call basis. To block the calling party information on a per call basis, the user dials a Calling Party Privacy Flexible Feature Code (FFC) prior to dialing the destination number.

With the Calling Party Privacy Override feature, a Private Branch Exchange (PBX) user can selectively unblock calling party information on a per-call basis when Class of Service is set to CLBA. The user unblocks the calling party information by dialing a Calling Party Privacy Override Flexible Feature Code prior to dialing the destination number.

When the CPPO Flexible Feature Code is dialed before the destination number, the user's calling party information is displayed on the terminating phone. The default for the Calling Party Privacy Override Flexible Feature Code is "*82" for Meridian 1 Proprietary Phones and "1182" for analog (500/2500-type) phones. The Calling Party Privacy Override Flexible Feature Code is defined in LD 57.

CPPO is provisioned on a trunk route basis. Any trunk type that can support an outgoing call can request the CPPO feature.

Note:

For non-ISDN trunks, only Central Office Trunk (COT), Direct Inward Dial (DID), Foreign Exchange (FEX), and Wide Area Telephone (WATS) trunks are supported. However, all ISDN trunk routes support the CPPO feature.

When the CPPO Flexible Feature Code is dialed prior to the normal dialing sequence, the call is marked as a CPPO call. The CPPO Flexible Feature Code is then removed from the dialed digits stored in the call register. If the outgoing trunk route provisions CPPO, then the Privacy Override Indicator is sent to the far end, and the Calling Party Number and Name information is displayed on the receiving phone. If the outgoing trunk route does not provision CPPO, the call does not carry the Privacy Override Indicator.

The following example illustrates Calling Party Privacy Override functionality:

1. Phone A, a Meridian 1 Proprietary Phone with Class of Service set to CLBA, goes off-hook.
2. Phone A dials the Calling Party Privacy Override Flexible Feature Code, defined in LD 57. Calling Party Privacy Override is initiated.
3. Phone A dials the destination number for Phone B.
4. Phone B rings because of the call.
5. Phone B presents the calling party information of Phone A on the display screen.

Outgoing calls

For an outgoing non-ISDN trunk call, the Privacy Override Indicator is defined on the outgoing trunk route. The CPPO Flexible Feature Code is outpulsed to the far end provided that the outgoing trunk route provisions CPPO. If CPPO is not provisioned on the trunk route, then the call does not carry the Privacy Override Indicator.

For an outgoing ISDN call from one system to another, the Privacy Override Indicator is represented when the Presentation Indicator field is set to "Presentation Allowed" in the Calling

Party Number Information Element (IE) and the Call Party Name Display (CPND) Indicator field is set to "Presentation Allowed" in the Display IE.

For an outgoing ISDN call to the Central Office, the Privacy Override Indicator is represented when the Presentation Indicator field is set to "Presentation Allowed" in the Calling Party Number IE and when the CPND information is included in the Display IE.

Incoming calls

An incoming ISDN call is recognized as a CPPO call (that is, it carries the Privacy Override Indicator) if the Presentation Indicator field is set to "Presentation Allowed" in the Calling Party Number IE and if the CPND Indicator is set to "Presentation Allowed" in the Display IE (if it exists).

When an incoming call is on a non-ISDN route, the system does not receive the Privacy Override Indicator.

Tandem Calls

Incoming ISDN calls

ISDN to ISDN tandem

For an incoming call tandeming through the system, any incoming Privacy Override Indicator is only repeated to the outgoing trunk route that also has CPPO provisioned.

When an incoming ISDN trunk call is tandemed through an ISDN trunk to a system switch, the Presentation Indicator or the CPND Indicator, received from the incoming ISDN trunk, is tandemed to the outgoing ISDN trunk.

When an incoming ISDN trunk call is tandemed through an ISDN trunk to a CO, the Presentation Indicator received from the incoming ISDN trunk is tandemed to the outgoing ISDN trunk. If the Display IE with the CPND Indicator set to "Presentation Allowed" is received from an incoming ISDN trunk, the Display IE, containing the Call Party Name, is sent across in the SETUP message tandemed to the outgoing ISDN trunk.

ISDN to non-ISDN tandem

When an incoming ISDN trunk call is tandemed to a non-ISDN trunk, the incoming call is treated as a CPPO call only if both the CLID and CPND Indicators are set to "Allowed". Otherwise, the call is treated as a CPP call.

Incoming non-ISDN calls

For incoming non-ISDN calls, the system does not receive the Privacy Override Indicator.

When a call on an incoming non-ISDN route is tandemed on the system, the call is tandemed based on how the CPP flag (TCPP) prompt is defined in the Route Data Block for the outgoing route.

When TCPP is set to YES, an incoming non-ISDN call tandemed to this route is treated as a CPP call.

When TCPP is set to NO, an incoming non-ISDN call tandemed to this route is treated as a CPPO call.

Non-ISDN to ISDN tandem

Even though a Privacy Override Indicator is not provided for an incoming non-ISDN trunk call, if the outgoing route has TCPP set to NO, the Presentation Indicator field in the Calling Party IE is set to "Presentation Allowed".

Non-ISDN to non-ISDN tandem

A Privacy Override Indicator is not provided for an incoming non-ISDN trunk call. If the outgoing route has TCPP set to NO, the Privacy Override Indicator defined for that route is outpulsed, provided that the outgoing route provisions CPPO.

Note:

Call Party Name Display (CPND) information is optional for ISDN calls.

Operating parameters

Central Office Trunks (COT), Foreign Exchange (FEX), Wide Area Telephone Service (WATS), and Direct Inward Dial (DID) are the only trunk route types (including ISA service routes) that can outpulse the Privacy Override Indicator for an outgoing non-ISDN call. All ISDN trunk routes provision the CPPO feature.

A non-ISDN trunk route does not provision the CPPO feature if the Outpulse Asterisk and Octothorpe (OPAO) package (package 104) is configured. During SYSLOAD, the CPPO database is removed from the non-ISDN trunk routes if the OPAO package is configured.

The Privacy Override Indicator, defined for a non-ISDN trunk route (dial-pulse or digitone), consists of any four arbitrary digits from 0-9. The asterisk (*) or octothorpe (#) cannot be part of the Privacy Override Indicator for dial-pulse trunks. For digitone trunks, the asterisk (*) can only be the first digit of the Privacy Override Indicator Flexible Feature Code.

The asterisk and octothorpe are not outpulsed if the OPAO package is configured. The asterisk signals a 3-second pause and the octothorpe indicates end-of-dialing. The octothorpe cannot be used in a Privacy Override Indicator.

Privacy Override Indicators are not received from the CO or non-ISDN DID trunks.

The CPPO Flexible Feature Code cannot conflict with any internal DN, including the Special Prefix (SPRE) code.

When a user dials the Flexible Feature Code defined for the CPPO feature and if CPPO is not provisioned on the outgoing trunk route, the call proceeds without carrying the Privacy Override Indicator.

The CPPO feature does not affect whether or not the Calling Party Number and Name information is displayed for internal calls within the system, even if the originator requests CPPO.

All incoming non-ISDN calls with the Privacy Override Indicator terminate on the system. If the Privacy Override Indicator is not defined in the Flexible Feature Code for CPPO, an overflow tone (unrecognized digits) is provided to the user.

If the Stored Number Redial (SNR)/Last Number Redial (LNR) feature is used by the originator of a CPPO call to store the dialed digits, the CPPO Flexible Feature Code is stored against the SNR/LNR database. If the user removes that CPPO Flexible Feature Code and then the SNR/LNR feature is used to re-initiate the call, overflow tone is returned to the user.

ISDN implementation for this feature includes DMS100/250, SL-100, AT&T4, AT&T5, TR-1268 (NI-2), Meridian Customer Defined Network (MCDN) Private Networks, EuroISDN, QSIG, and BRI trunks.

The CPPO feature is supported on the following International PRI (IPRI) connectivities: Ericsson AXE-10 CO Connectivity (Australia), Ericsson AXE10-CO Connectivity (Sweden), French Numeris CO Connectivity, Japan D70 CO Connectivity, Swissnet 2 CO Connectivity, SYS-12 CO Connectivity, 1TR6 CO Connectivity (Germany), and Asia Pacific ISDN Phase 2.

The CPPO feature supports the following North American connectivities: DMS100/250, S1100, Lucent #4 ESS (ESS4), Lucent #5 EES (ESS5), and TR-1268 (NI-2).

CPPO does not support R2MFC signaling.

Feature interactions

Attendant Consoles

A CPPO call can be originated from any system Attendant Console. Attendant Consoles request CPPO by preceding the normal dialing sequence with the Flexible Feature Code for CPPO.

Attendant Consoles can also initiate a CPPO call using the Autoline key. An outgoing trunk call, initiated by pressing the Autoline key, carries the Privacy Override Indicator if the CPPO Flexible Feature Code, followed by the normal dialing sequence, is stored against the Autoline key. The CPPO Flexible Feature Code is counted against the maximum number of digits (currently 31) stored against the Autoline key.

The CPPO Flexible Feature Code can also be stored against the Autoline key. An outgoing CPPO call can then be initiated by pressing the Autoline key followed by manually dialing the destination number.

An outgoing CPPO call can also be initiated by dialing the CPPO Flexible Feature Code followed by pressing the Autoline key, on which the normal dialing sequence of digits for the destination number is stored.

Autodial

An outgoing trunk call, initiated by pressing the Autodial key, carries the Privacy Override Indicator if the CPPO Flexible Feature Code followed by the normal dialing sequence is stored against the Autodial key. The CPPO Flexible Feature Code is counted against the maximum number of digits (currently 31) stored against the Autodial key.

The CPPO Flexible Feature Code can be stored against the Autodial key. In this case, an outgoing CPPO call can be initiated by pressing the Autodial key followed by manually dialing the normal sequence of digits for the destination number.

An outgoing CPPO call can also be initiated by dialing the CPPO Flexible Feature Code followed by pressing the Autodial key on which the normal dialing sequence of digits for the destination number is stored.

Automatic Call Distribution

Calls placed by means of Enhanced Automatic Call Distribution (ACD) Routing, Enhanced Interflow, Enhanced Night Call Forward, Enhanced Network Routing, and Network ACD recognize the originator's CPPO request.

Automatic Call Distribution MAX

If the CPP package is equipped, ACD MAX reports include the Calling Line Identification (CLID) for incoming ISDN calls that have the CLID Presentation Indicator set to "Allowed".

Basic Rate Interface

Although Basic Rate Interface (BRI) networking is not supported in North America, CPPO treats BRI trunk calls in the same manner as an ISDN trunk call.

Call Detail Recording

Call Detail Recording (CDR) records continue to include the Calling Party Number even if the caller has requested CPPO. When the CDR record is generated, the CPPO Flexible Feature Code dialed by the originator is included in the DIGIT field (if it displays the dialed digits).

The CPPO Flexible Feature Code dialed by the originator is not included in the DIGIT field if it displays the outpulsed digits. The Privacy Override Indicator, outpulsed by an outgoing non-ISDN trunk route that provisions CPPO, is included in the outpulsed digits.

Call Pickup Network Wide

When an incoming trunk call with the Privacy Override Indicator is picked up by a remote phone (the requesting party), the Calling Party Number and Name is displayed on the requesting phone.

Call Hold

When an incoming trunk call with the Privacy Override Indicator is taken off hold, the Calling Party Number and Name information is displayed on the phone.

Call Forward All Types

Hunt

Network Hunt

The existing call redirection functionality is not changed by this feature.

When an incoming ISDN trunk call with the Privacy Override Indicator is forwarded into the public or private networks, the Privacy Override Indicator is tandemed to the far end to allow the display of the Calling Party Number and Name, provided that the outgoing trunk route on the tandem node has CPPO provisioned.

When an incoming ISDN call with Calling Party Number and Name set to "Presentation Allowed" is forwarded to a phone within the same node, the Calling Party Number and Name is displayed on the terminating phone.

When an incoming non-ISDN trunk call is forwarded onto a trunk (where the Privacy Override Indicator is not expected), the outgoing trunk call from the tandem node carries the Privacy Override Indicator, provided that the outgoing trunk route on the tandem node has CPPO provisioned. Also, the TCPP prompt in the Route Data Block must be set to NO.

The CPPO Flexible Feature Code can be stored on the forwarding Directory Number (DN), including the forwarding DN for Call Forward All Calls, Hunt DN and Flexible Call Forward No Answer DN (FDN).

If CPPO is requested on the forwarding DN and the call is forwarded across an ISDN link, the outgoing SETUP message includes the Redirecting Number IE (if it exists) with the Presentation Indicator set to "Presentation Allowed".

If CPPO is requested on the forwarding DN and the call is forwarded across a non-ISDN link, no Privacy Override Indicator is outpulsed to the terminating node if the originating phone did not request CPPO. This is because no Redirecting Number information is sent across a non-ISDN link.

When an internal call is forwarded into the public or private networks, if the originator requests CPPO and the outgoing trunk route provisions CPPO, the Privacy Override Indicator is sent to the far end to allow the display of the Calling Party Number and Name.

Call Pickup

With CPPO activated, when an incoming trunk call with the Privacy Override Indicator is picked up locally, the Calling Party Number and Name information is displayed on the terminating phone.

Call Transfer

As per existing operation, if an incoming non-ISDN call is transferred or an incoming ISDN call is transferred to a non-ISDN trunk, the Connect Party Number and Name information is not passed to the terminating node. The CPPO feature does not change this operation.

When an incoming call with the Privacy Override Indicator is transferred across the MCDN network or to a local phone, the originator's calling party information is displayed on the final terminating phone.

Calling Line Identification Restriction

Basic Rate Interface (BRI) phones do not support the Flexible Feature Code (FFC) feature. CPPO can only be requested by applying the existing Calling Line Identification Restriction (CLIR) Service option. This is done by setting the soft key "ID PRES" (if it exists) to "Allowed" or the Presentation of CLID to far end on outgoing calls (PRES) prompt to YES in LD 27. Then an outgoing ISDN/non-ISDN trunk call carries the Privacy Override Indicator if the outgoing trunk route provisions CPPO. However, if the Calling Party Number Information Element (IE) with the Presentation Indicator set to "Presentation Denied" is included in the SETUP message generated by the Basic Rate Interface (BRI) terminal, then the BRI terminal does not allow CPPO. This is because the Presentation Indicator, generated by the BRI terminal, always overwrites the Calling Line Identification Restriction (CLIR) service option.

Calling Party Privacy

If the user requests both Calling Party Privacy and Calling Party Privacy Override, then the feature last requested takes precedence. The Flexible Feature Code dialed last determines the type of call.

If a phone with Class of Service set to CLBA requests CPPO by dialing the CPPO Flexible Feature Code, then the call is treated as a CPPO call. If a phone with Class of Service set to CLBD requests CPP by dialing the CPP Flexible Feature Code, then the call is treated as a CPP call.

If a user dials the Flexible Feature Code for CPPO followed by the Flexible Feature Code for CPP, then the call is treated as a CPP call. If a user dials the Flexible Feature Code for CPP followed by the Flexible Feature Code for CPPO, then the call is treated as a CPPO call.

Calling Party Privacy and Call Forward

Phone A, requesting CPPO, calls Phone B. Phone B Call Forwards All Calls to Phone C. The CPP Flexible Feature Code is part of the forwarding DN. Phone A's number and name is displayed on Phone C as the Calling Party Number and Name; although, no redirecting number is displayed on Phone C. The tandem node sends the Display IE with the Presentation Indicator set to "Allowed" and the Redirecting Number IE with the Presentation Indicator set to "Restricted".

Phone A, requesting CPP, calls Phone B. Phone B Call Forwards All Calls to Phone C. The CPPO Flexible Feature Code is part of the forwarding DN. Phone B's number is displayed on Phone C as the Redirecting Number; although, no Calling Party Number and Name is displayed on Phone C. The tandem node sends the display IE with the Presentation Indicator set to "Restricted" and the Redirecting Number IE with the Presentation Indicator set to "Allowed".

Calling Party Privacy and Call Transfer

Phone A, requesting CPPO, calls Phone B. Phone B answers the call, requests CPP, and initiates a transfer to Phone D. After the transfer is complete, Phone A's Calling Party Number and Name is displayed on Phone D. The request made by the connected party takes precedence over the transferring party while displaying the Connect Party Number and Name.

Phone A, requesting CPP, calls Phone B. Phone B answers the call, requests CPPO, and initiates a transfer to Phone D. After the transfer is complete, Phone A's Calling Party Number and Name is not displayed on Phone D. The request made by the connected party takes precedence over the transferring party while displaying the Connect Party Number and Name.

Conference

The CPPO feature passes the Privacy Override Indicator to the terminating phone in order to display the Calling Party Number and Name, if the Conference feature is used for the purpose of performing a transfer.

Display of Calling Party Denied

When the CPP package is equipped, the CPPO feature takes precedence over the Display of Calling Party Denied (DPD) feature for allowing the Calling Party Number and Name to be displayed. For example, when an outgoing ISDN call is marked as a CPPO call, then the outgoing SETUP message includes the Calling Party Number IE with the Presentation Indicator set to "Presentation Allowed" and the Display IE with the CPND Indicator set to "Presentation Allowed". This enables both the Calling Party Number and Name to be displayed on the terminating phone, regardless of whether the DPD feature allows or denies the display of the Calling Party Number and/or Name.

E.164 ESN Numbering Plan Enhancement

CPPO can be requested for ESN calls by preceding the dialing sequence with the Flexible Feature Code defined for the CPPO feature. The CPPO Flexible Feature Code is counted against the maximum number of digits (currently 31) allowed for the destination DN.

Feature Group D

When an incoming Feature Group D (FGD) call terminates at a system switch locally, the received 10-digit Automatic Number Identification (ANI) is displayed on the terminating phone if the Show ANI Digits on Terminal Displays (SHAN) field is set to YES in the FGD data block that is associated with the incoming trunk route. If the originator requests CPPO, the end office sends the 10-digit ANI to the PBX.

If an incoming FGD call is routed to another switch through ISDN Primary Rate Interface (PRI) or ISDN Signaling Link (ISL), the outgoing SETUP message includes the 10-digit ANI (if it exists) as the Calling Party Number (CLID) with the Presentation Indicator set to "Presentation Allowed". This occurs if the incoming call requests CPPO. CPPO takes precedence over the SHAN field that is defined in the FGD data block and is associated with the incoming trunk route to allow the 10-digit ANI display.

Hot Line

Hot Line calls carry the Privacy Override Indicator if the CPPO Flexible Feature Code followed by the normal dialing sequence is stored in the Hot Line DN. The CPPO Flexible Feature Code is counted against the maximum number of digits (currently 31) allowed for the Hot Line DN.

Last Number Redial

The Last Number Redial (LNR) feature stores the CPPO Flexible Feature Code in the LNR database if the CPPO Flexible Feature Code was included in the last number dialed by the user. The outgoing redialed calls also send the Privacy Override Indicator to the far end.

Incoming Trunk Programmable Calling Line Identification

When the incoming trunk route is a non-ISDN route, the billing number (CLID) assigned by the incoming trunk route is passed to the CO with the Presentation Indicator field set to "Presentation Allowed", if the outgoing ISDN trunk route has the TCPP prompt set to NO.

When the incoming trunk route is an ISDN route, the "Allowed" Presentation Indicator is tandemed to the outgoing trunk route. If the Presentation Indicator is set to "Presentation Allowed" or no Calling Party Number IE is received on the incoming trunk route, the billing number assigned by the incoming trunk route is passed to the CO with the Presentation Indicator field set to "Presentation Allowed", if the incoming trunk route has the Billing Number Display (BDSP) prompt set to YES or NO.

ISDN Signaling Link

CPPO treats an ISDN Signaling Link (ISL) call in the same manner as an ISDN trunk call.

Malicious Call Trace

An incoming call to a phone with the Malicious Call Trace (MCT) feature activated includes the Terminal Number (TN) of the calling party in the MCT record, whether or not the caller has requested CPPO.

Meridian 911

An incoming 911 call with Automatic Number Identification (ANI) information always displays the ANI digits on the terminating phone or passes the ANI information to the Meridian 911.

Meridian Interactive Voice Response

An incoming ISDN call with the CLID Presentation Indicator set to "Allowed" sends the CLID to the Meridian Interactive Voice Response (IVR) if the CPP package is equipped.

Meridian Link

If the CPP package is equipped, an incoming ISDN call with the CLID Presentation Indicator set to "Allowed" includes the CLID in the Application Module Link (AML) messages sent to the Meridian Link module.

Meridian MAX

If the CPP package is equipped, an incoming ISDN call with the CLID Presentation Indicator set to "Allowed" sends the CLID to Meridian MAX.

Network Call Redirection

If a phone receives a call and is then redirected to the public network on an ISDN trunk that supports call redirection, then the redirecting IE in the outgoing SETUP message has the Presentation Indicator set accordingly. For instance, if the call that had requested CPPO is redirected, the outgoing SETUP message has the Presentation Indicator set to "Allowed".

Network Message Center

An incoming trunk call with the Privacy Override Indicator displays the Calling Party Number and Name on the Message Center operator's terminal.

Network Ring Again

A call placed by means of the Network Ring Again feature recognizes the CPPO request from when the call was originally dialed.

Symposium Call Center

As per existing operation, an incoming CPPO call routed to Symposium Call Center contains the CLID.

Private Line Service

The Private Line Service feature outpulses the Privacy Override Indicator only if it is dialed by the originator. The asterisk (*) is outpulsed to the far end only if it is an Outpulse Asterisk and Octothorpe (OPAO) call. Otherwise, the asterisk (*) signals a three-second pause.

Remote Virtual Queuing

The Remote Virtual Queuing feature has automatic re-try capabilities that are used when congestion is encountered within the network. The same Calling Party Privacy Override considerations are provided to the "re-tries" as were provided to the originally dialed call.

Ring Again - Busy Trunk

A call that is automatically redialed by the Ring Again - Busy Trunk feature recognizes the CPPO requested when the call is originally dialed.

Speed Call

System Speed Call

When an outgoing trunk call is initiated by dialing a Speed Call code, the Speed Call code carries the Privacy Override Indicator if the CPPO Flexible Feature Code followed by the normal dialing sequence is stored in the Speed Call Entry represented by the Speed Call code.

The CPPO Flexible Feature Code is counted against the maximum number of digits (currently 31) allowed per Speed Call list entry.

The user can also store the CPPO Flexible Feature Code in the Speed Call Entry (or Speed Call key). An outgoing CPPO call can be initiated by dialing the Speed Call code (or pressing the Speed Call key), followed by manually dialing the digits.

Stored Number Redial

In the Stored Number Redial (SNR) programming mode, the user can store the CPPO Flexible Feature Code, followed by the normal dialing sequence in the SNR database. The outgoing calls originated by the Stored Number Redial feature send the Privacy Override Indicator to the far end. The CPPO Flexible Feature Code is counted against the maximum number of digits (currently 31) allowed by the SNR feature.

During an active call on a Meridian 1 Proprietary Phone, the Stored Number Redial feature stores the CPPO Flexible Feature Code in the SNR database if the CPPO Flexible Feature Code is included in the number dialed by the originator. The outgoing redialed calls also send the Privacy Override Indicator to the far end.

Trunk Anti-Tromboning

When trunks are removed, due to the Trunk Anti-Tromboning (TAT) operation, an ISDN call recognizes the CPPO/CPP requested by the originator.

Trunk Optimization Before Answer

An optimized call, due to Trunk Optimization Before Answer (TRO) operation, recognizes the CPPO/CPP requested by the originator.

Virtual Network Services

CPPO treats Virtual Network Services (VNS) trunk calls in the same manner as ISDN trunk calls. For instance, CPPO does not affect the existing VNS operation. If CPPO was requested when originating a call, the Presentation Indicator field of CLID is set to "Presentation Allowed".

VISIT

The VISIT which connects to a phone receives the Calling Party Number or Name, since an incoming CPPO call sends the Calling Party Number or Name to the phone for display.

Feature packaging

This feature requires the following packages:

- Calling Party Privacy (CPP) package 301, which has the following dependency:
  - Flexible Feature Codes (FFC) package 139.

For Calling Party Name Display, Calling Party Name Display (CPND) package 95 is required. ISDN package 145 is required for ISDN routes.

Note:

Non-ISDN trunks must restrict the Outpulse Asterisk and Octothorpe (OPAO) package 104 to provision the Calling Party Privacy Override feature.

Feature implementation

Task summary list

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

1. Table 235: LD 16 - Configure Privacy Override Indicators for a Non-ISDN route. on page 508
2. Table 236: LD 16 - Set the TCPP flag in RDB to tandem non-ISDN calls on an ISDN trunk route. on page 509
3. Table 237: LD 57 - Define the Flexible Feature Code for the Calling Party Privacy Override feature. on page 509
4. Table 238: LD 10/11 - Activate Calling Party Number and Name per-line blocking. on page 510

Configuration procedures require that the following conditions are met:

- CPPO is configurable on COT, DID, FEX, WAT and ISA routes.
- OAPO package 104 is restricted or unequipped.
- Route is either OGT (outgoing) or IAO (incoming and outgoing).

Table 235: LD 16 - Configure Privacy Override Indicators for a Non-ISDN route.

Prompt	Response	Description
REQ	CHG	Change existing data.
TYPE	RDB	Route Data Block.
CUST	xx	Customer number, as defined in LD 15.
ROUT		Route number
	0-511	Range for Large System and Avaya Communication Server 1000E (Avaya CS 1000E) system.
	0-127	Range forMedia Gateway 1000 B (Avaya MG 1000B).
CPP	YES	Calling Party Privacy/Privacy Override (CPP/CPPO) flag. Enable CPP/CPPO feature and configure parameters. (NO) = CPP/CPPO feature is disabled is the default.
- TCPP	(NO) YES	CPP/CPPO flag treatment for an incoming non-ISDN trunk call tandemed to this trunk route. Outgoing call will carry the Privacy Override Indicator (default). Outgoing call will carry the Privacy Indicator.
- DTPI	(*67) nnnn	Digitone Trunk Privacy Indicator nnnn = 0-9999, an asterisk (*) can be entered as the first digit.
- DPPI	0-(1167)-9999	Dial-pulse Trunk Privacy Indicator
- DTPO	(*82) nnnn	Digitone Trunk Privacy Indicator nnnn = 0-9999, an asterisk (*) can be entered as the first digit.
- DPPO	0-(1182)-9999	Dial-pulse Trunk Privacy Indicator

Configuration procedures require that the following conditions are met:

- The CPP package 301 is equipped.
- Route is either OGT (outgoing) or IAO (incoming and outgoing).

Table 236: LD 16 - Set the TCPP flag in RDB to tandem non-ISDN calls on an ISDN trunk route.

Prompt	Response	Description
REQ	CHG	Change existing data.
TYPE	RDB	Route Data Block.
CUST	xx	Customer number, as defined in LD 15.
ROUT		Route number **Note:** All ISDN trunk routes are CPPO configurable.
	0-511	Range for Large System and CS 1000E system.
	0-127	Range forMedia Gateway 1000B.
CPP	YES	Calling Party Privacy/Privacy Override (CPP/CPPO) flag. Enable CPP/CPPO feature and configure parameters. (NO) = CPP/CPPO feature is disabled is the default.
- TCPP	(NO) YES	CPP/CPPO flag treatment for an incoming non-ISDN trunk call tandemed to this trunk route. Outgoing call will carry the Privacy Override Indicator (default). Outgoing call will carry the Privacy Indicator.

Table 237: LD 57 - Define the Flexible Feature Code for the Calling Party Privacy Override feature.

Prompt	Response	Description
REQ	CHG	Change existing data.
TYPE	FFC	Flexible Feature Code.
CUST	xx	Customer number, as defined in LD 15.
FFCT	(NO)	Flexible Feature Confirmation Tone denied.
	YES	Flexible Feature Confirmation Tone allowed.
...		
CODE	CPP	CPP Flexible Feature Code
- CPP	xxxx	Calling Party Privacy code xxxx = 0-9999, an asterisk (*) can be entered as the first digit. The Flexible Feature Code can be up to 4 digits, or up to 7 digits with the Directory Number Expansion (DNXP) package (150).
- CPP	xxxx	Change the CPP code or enter a <CR> to accept.
CODE	CPPO	CPPO Flexible Feature Code

Prompt	Response	Description
- CPPO	xxxx	Calling Party Privacy Override code xxxx = 0-9999, an asterisk (*) can be entered as the first digit. The Flexible Feature Code can be up to 4 digits, or up to 7 digits with the Directory Number Expansion (DNXP) package (150).
- CPPO	xxxx	Change the CPPO code or enter a <CR> to accept.

Table 238: LD 10/11 - Activate Calling Party Number and Name per-line blocking.

Prompt	Response	Description
REQ:	NEW CHG	Add new data, or change existing data.
TYPE:	aaaa	Type of phone.
TN		Terminal number
	l s c u	Format for Large System and CS 1000E system, where l = loop, s = shelf, c = card, u = unit.
	c u	Format for Media Gateway 1000B, where c = card and 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 defined in LD 15.
CLS	CLBA	Activate Calling Party Number and Name per-line blocking. CLBD = Deactivate Calling Party Number and Name per-line blocking (default).

Feature operation

For a user to override the Calling Party Number and Name per-line blocking allowed (CLBA) Class of Service, the following steps must be performed.

1. The user goes off hook.
2. The user initiates a call by dialing the Calling Party Privacy Override Flexible Feature Code, defined in LD 57.
3. The user dials the destination number.

Chapter 23: Call Page Network Wide

Contents

This section contains information on the following topics:

Feature description

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 phones.

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:

- Attendant Console or phone is programmed with PAGENET allowed in the Class of Service
- point-to-point D-channel is configured with remote capability (for example, 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 through 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

A user can 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 will 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 phone and the point-to-point D-channel connection between the nodes has remote capability of network access (for example, 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 phone 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 phone 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 implementation

Task summary list

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

Table 239: LD 16 - Configure Paging Route.

Prompt	Response	Description
REQ	NEW	New.
TYPE	RDB	Route data block.
CUST	xx	Customer number, as defined in LD 15.
...		
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.

Table 240: LD 16 - Configure BRI Trunk Route.

Prompt	Response	Description
REQ	NEW	New.
TYPE	RDB	Route data block.
CUST	xx	Customer number, as defined in LD 15.
...		
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).

Table 241: LD 14 - Configure Page Trunk.

Prompt	Response	Description
REQ	NEW	New.
TYPE	PAG	Trunk type.
TN		Terminal number
	l s c u	Format for Large System and Avaya Communication Server 1000E (Avaya CS 1000E) system, where l = loop, s = shelf, c = card, u = unit.
	c u	Format for Media Gateway 1000 B (Avaya MG 1000B), where c = card and u = unit.
RTMB		Route number and Member Number
	0-511 1-4000	Range for Large System and CS 1000E system.
	0-127 1-4000	Range for Media Gateway 1000B.

Table 242: LD 10 - Assign Class of Service to analog (500/2500-type) phones.

Prompt	Response	Description
REQ:	NEW CHG	Add new data, or change existing data.
TYPE:	500	Type of phone.
...		
CLS	PGNA	Call Page Network Wide Allowed. (PGND) = Call Page Network Wide Denied (default).

Table 243: LD 11 - Assign Class of Service to Meridian 1 Proprietary Phones.

Prompt	Response	Description
REQ:	NEW CHG	Add new data, or change existing data.
TYPE:	a...a	Telephone type. Type **?** for a list of possible responses.
...		
CLS	PGNA	Call Page Network Wide Allowed. (PGND) = Call Page Network Wide Denied (default).

Table 244: LD 27 - Assign Class of Service to ISDN BRI phones.

Prompt	Response	Description
REQ	NEW	Add new data.
	CHG	Change existing data.

Prompt	Response	Description
TYPE	DSL	Type of data block.
DSL	l s c dsl c dsl	Digital Subscriber Loop For Large Systems.
CLS	PGNA	Call Page Network Wide Allowed. (PGND) = Call Page Network Wide Denied (default).

Table 245: 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	DCHI	D-channel interface card.
RCAP	NAC	Remote capability where: NAC = Class of Service data. XNAC = removes Class of Service as a remote capability (default).

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 through dial access, one of the following dialing plans must be used:

- Route Access Code (ACOD) that connects user to Paging node and Paging route ACOD
- BARS/NARS, or
- Coordinated Dialing Plan.

Chapter 24: Call Park Network Wide

Contents

This section contains information on the following topics:

Feature description

Previous to the introduction of Call Park Network Wide, a call in a parked state on a System Park Directory Number (DN) or station phone park DN could only be retrieved by an attendant console or a station phone 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 phone 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 phone 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 recall timer and number of System Park DNs expansions are included in base system software. However, the 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) prompt of the associated call park block. This recall to an attendant occurs even if the call was parked by a station phone.

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 Phone 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 through 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 phone 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. Expansions to the recall timer and the number of System Park DNs are included in Call Park (CPRK) package 33.

Feature implementation

Task summary list

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

- Table 246: LD 15 - Enable Call Park Network Wide. on page 523
- Table 247: LD 17 - Set the remote capability at both ends of the ISDN link to Call Park. on page 523Table 251: LD 12 - Add/Change Call Park Key on Attendant Consoles. on page 524
- Table 248: LD 50 - Add/Change Customer Call Park Data. on page 523

- Table 249: LD 10 - Enable Call Park for analog (500/2500-type) phones. on page 524
- Table 250: LD 11 - Add/Change Call Park Key on Meridian 1 proprietary phones. on page 524
- Table 251: LD 12 - Add/Change Call Park Key on Attendant Consoles. on page 524

Table 246: LD 15 - Enable Call Park Network Wide.

Prompt	Response	Description
REQ:	CHG	Change.
TYPE:	FTR	Features and options.
CUST		Customer number
	0-99	Range for Large System and Avaya Communication Server 1000E (Avaya CS 1000E) system.
	0-31	Range for Media Gateway 1000 B (Avaya MG 1000B).
- OPT	CPN	Enable Call Park Network Wide. CPA = Enables Call Park. CPD = Disables Call Park (default).

Table 247: LD 17 - Set the remote capability at both ends of the ISDN link to Call Park.

Prompt	Response	Description
REQ	CHG	Change.
TYPE	ADAN	Action Device and Number options.
ADAN	CHG DCH xx	Define changes to D-channel xx.
...		
- RCAP	CPK	Define Call Park as a remote capability.
...		
- NASA	YES	Allow Network Attendant Service.

Table 248: LD 50 - Add/Change Customer Call Park Data.

Prompt	Response	Description
REQ	NEW	Add new data.
	CHG	Change existing data.
	PRT	Print existing data.
TYPE	CPK	Call Park data block.
CUST	xx	Customer number, as defined in LD 15.

Prompt	Response	Description
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	xx	Music Route number for parked call.

Table 249: LD 10 - Enable Call Park for analog (500/2500-type) phones.

Prompt	Response	Description
REQ:	NEW CHG	Add new data, or change existing data.
TYPE:	500	Type of phone.
TN		Terminal number
	l s c u	Format for Large System and CS 1000E system, where l = loop, s = shelf, c = card, u = unit.
	c u	Format for Media Gateway 1000B, where c = card and u = unit.
CLS	XFA	Call Transfer Allowed. XFD = Call Transfer Denied.

Table 250: LD 11 - Add/Change Call Park Key on Meridian 1 proprietary phones.

Prompt	Response	Description
REQ:	NEW CHG	Add new data, or change existing data.
TYPE:	a...a	Telephone type. Type **?** for a list of possible responses.
TN		Terminal number
	l s c u	Format for Large System and CS 1000E system, where l = loop, s = shelf, c = card, u = unit.
	c u	Format for Media Gateway 1000B, where c = card and u = unit.
KEY	xx PRK xx TRN xx AO3 xxAO6	Key assignment for Call Park. Key number for Transfer. Three Party Conference. Six Party Conference.

Table 251: LD 12 - Add/Change Call Park Key on Attendant Consoles.

Prompt	Response	Description
REQ:	NEW CHG	Add new data, or change existing data.

Prompt	Response	Description
TYPE:	2250	Attendant Console type.
TN		Terminal number
	l s c u	Format for Large System and CS 1000E system, where l = loop, s = shelf, c = card, u = unit.
	c u	Format for Media Gateway 1000B, where c = card and u = unit.
KEY	xx PRK	Key number, Call Park.

Feature operation

Call Park Expansion

Attendant Console with Park Key

Attendant Console through Dial Access

Meridian 1 Proprietary Phone 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 the 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 the number dialed is an invalid or unavailable station Park DN, an overflow tone is heard.
    - If the 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 the PRK key, or if using SPRE or FFC, press the Release RLS key to complete Call Park.

Meridian 1 Proprietary Phone with Display through 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 the number dialed is available, the digit display is cleared and the dialed Park Access ID is displayed (without SPRE + 71 or CPRK FFC).
    - If the number dialed is an invalid System Park DN or an unavailable station park DN, overflow tone is heard.
    - If the 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 Phone without Display

analog (500/2500-type) phones

The Call Park operation on these types of phones 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 phone's MCDN network.

Attendant Console

Meridian 1 Proprietary Phone with Display with Park Key

Attendant Console through 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 phone 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 a Meridian 1 Proprietary Phone with Display, press the PRK key to complete Call Park.

Attendant Parking an Extended Call

1. Press the PRK key.
2. The 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.
3. At the remote node, if numbers dialed are an invalid or unavailable Park DN, overflow tone is heard.
4. Press the RLS key to complete call park.

Meridian 1 Proprietary Phone with Display through 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 Phone without Display with PRK key

Meridian 1 Proprietary Phone without Display through Dial

Access analog (500/2500-type) phone

1. Press the PRK, CONF, TRN or perform switchhook flash depending on type of phone 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 the 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 phone 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 Phone

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

Analog (500/2500-type) phone

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

External Call Park Access

External Call Park Access allows a parked call on a System Park DN to be retrieved by an external user through 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.

Chapter 25: CLID on Analog Trunks for Hong Kong (A-CLID)

Contents

This section contains information on the following topics:

Feature description

With the Calling Line Identification on Analog Trunks (A-CLID) feature and the DXUT-A card (NTRB37AA), on an incoming Central Office (CO) call, the system can extract information such as:

- Calling Party Number
- Calling Party Name
- Reason for absence of Calling Party Number or Name (if necessary)

The A-CLID information is treated similar to ISDN CLID for delivery to other modules and applications in the system, including the display on digital phones and consoles at the local node and other network nodes (if any).

You can enable or disable A-CLID on an individual trunk port basis.

The A-CLID information passes to the terminating party, which includes:

- Trunks: ISDN (PRI/BRI/QSIG), R2MFC (DTI/DTI2, Analog)
    - Calling Party Number information can be tandemed over all ISDN and R2MFC interfaces
    - Calling Party Name information can be tandemed only on SL1 and QSIG ISDN interfaces. R2MFC does not support name information.
- Terminals: Attendant Consoles and phones (CLASS, 2208 with display, 2216, 2616, 2317, 5317, Avaya 3902/3903/3904/3905 Digital Deskphones).
- Applications: Avaya CallPilot, Customer Controlled routing, Meridian Link, and Symposium Call Center Server (calling party number only).

More detailed information on A-CLID is found in *Avaya Features and Services Fundamentals, NN43001-106*.

Operating parameters

There are no operating parameters associated with this feature.

Feature interactions

There are no feature interactions associated with this feature.

Feature packaging

This feature is included in base System Software.

Feature implementation

There are no specific implementation procedures for this feature.

Feature operation

No specific operating procedures are required to use this feature.

Chapter 26: Channel Negotiation

Contents

This section contains information on the following topics:

Feature description

The Channel Negotiation feature operates on connections between the system and Central Offices conforming to the following protocols:

- AXE-10
- SYS-12
- 1TR6
- QSIG
- Japan D70 (INS NET-64)
- NEAX-61
- Numeris
- EuroISDN (in some countries)

Channel Negotiation allows call setup to continue even where a chosen bearer channel is unacceptable to the receiving switch. When this occurs, a search for an alternative channel acceptable to both ends of the call can take place.

On an incoming or an outgoing call, the SETUP message sent by the Central Office or the system respectively contains the number of the requested B-channel. The receiving side then sends a response to this SETUP message, also containing a B-channel number. Where the requested B-channel was acceptable to the receiving side, this number will be the same as the one sent in the SETUP message. If the requested channel was unavailable or unacceptable, a different, alternate B-channel number is given.

If Channel Negotiation is not enabled and the requested B-channel is either unavailable or unacceptable to the receiving switch, call clearing will take place. On an outgoing call, reorder tone will be presented to the system caller. (The exception to this occurs where the channel requested by the system does not exist at the Central Office; the system will search for another B-channel to use.)

Note:

If channel negotiation is used on a PRI interface, the B-channels must not be shared between customers.

Outgoing calls

If Channel Negotiation has been enabled (by way of the CNEG prompt in LD 17) and an alternate B-channel is received on an outgoing call, the system checks that B-channel's state. If the alternate B-channel is idle, the call proceeds on that channel. Should the alternate be unacceptable to the system, a RELEASE signal is sent to the CO. The system searches for another idle B-channel and re-attempts the call.

Incoming calls

With Channel Negotiation enabled, the system responds to an unacceptable B-channel request on an incoming call by looking for an alternative, acceptable B-channel (one also controlled by the D-channel controlling the channel requested by the CO). If it finds one, it sends the alternative B-channel number in its response to the CO's SETUP message. If the system cannot find another acceptable B-channel under the same D-channel, a RELEASE COMPLETE message is sent back to the CO, clearing the call.

Operating parameters

Channel negotiation cannot take place over ISDN PRI connections between system nodes.

Feature interactions

There are no feature interactions associated with this feature.

Feature packaging

This feature is included in BASE System Software.

Feature implementation

There are no specific implementation procedures for this feature.

Feature operation

No specific operating procedures are required to use this feature.

Chapter 27: DID-to-network Calling

Contents

This section contains information on the following topics:

Feature description

This feature facilitates Direct Inward Dialing into the private ISDN. The Direct Inward Dial (DID) call will be treated as though the entire ISDN is a large PBX.

DID calls entering the network at the local node (that is, the node on which the destination phone 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 phone's node.

Routing of the DID call across the ISDN will be 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.)

Operating parameters

There are no operating parameters associated with this feature.

Feature interactions

There are no feature interactions associated with this feature.

Feature packaging

This feature is included in base System Software.

Feature implementation

There are no specific implementation procedures for this feature.

Feature operation

No specific operating procedures are required to use this feature.

Chapter 28: Digit Key Signaling at Console

Contents

This section contains information on the following topics:

Feature description

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

These functions allow attendants to help callers operate the features offered by CallPilot (for instance, playing voice messages from an external rotary dial phone).

The attendant can send keypad digits (0-9, * and #) under the following conditions:

- while extending source calls to CallPilot
- during direct calls to CallPilot

The digits are sent to CallPilot by way of ISDN/AP KEY messages (and not by way of End to End signaling). Dialed digits are not saved by the system and are not displayed at the Attendant Console.

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

Extending source calls to Avaya CallPilot

When extending a source call to CallPilot, 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 CallPilot agent.
- The DEST call to CallPilot is not a conference call.

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

Direct calls to Avaya CallPilot

Digit Key signaling also operates when the attendant dials CallPilot 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 CallPilot agent (Class of Service of VMA).
- The SRC call to CallPilot 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.)

Operating parameters

There are no operating parameters associated with this feature.

Feature interactions

There are no feature interactions associated with this feature.

Feature packaging

This feature requires the following packages:

- Integrated Services Digital Network (ISDN) package 145
- Digit Key Signaling (DKS) package 180

Feature implementation

There are no specific implementation procedures for this feature.

Feature operation

No specific operating procedures are required to use this feature.

Chapter 29: E.164/ESN Numbering Plan Expansion

Contents

This section contains information on the following topics:

Applicable regions

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

Feature description

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.

The following enhancements are offered by this feature:

- Support of the numbering plan for ISDN and PSTN dialing has been increased to a 15-digit maximum, from a 12-digit maximum.
- 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, an entry of either 55 or 5551212 can be entered.
- Addition of "allow" (ALOW) as a 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 ALOW entry allows a call to go through, if the dialed digits did not match any entry within the Supplemental Digit Restriction and Recognition table.
- Special Number Translation (SPN) screening digits scheme is changed.
- Capability of the base features to store digits is increased to 31 digits.
- 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 were 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 was 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 expanded ESN functionalities

Table 252: Summary of ESN changes on page 547 summarizes the changes made to the ESN functionality.

Note:

The components that are not affected are not listed.

Table 252: Summary of ESN changes

Component	Old	Expanded
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 for each 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 for each 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 ('ALOW' is added). ALOW is an 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.

Component	Old	Expanded
		The leftwise-unique restriction, imposed on SDRR entry codes, is removed. For example, if 555 is an existing entry, an entry of either 55 or 5551212 can be entered.

Summary of base features enhancements

Table 253: Summary of base features changes on page 548 summarizes the changes made to the base features functionality.

Note:

The components that are not affected are not listed.

Table 253: Summary of base features changes

Component	Old	Expanded
Maximum number of digits for an Autodial entry.	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 phones.	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 254: Summary of system ACD changes on page 549 summarizes the changes made to the ACD functionality.

Note:

The components that are not affected are not listed.

Table 254: Summary of system ACD changes

Component	Old	Expanded
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 255: Summary of CCR changes on page 549 summarizes the changes made to the Customer Controlled Routing functionality.

Note:

The components that are not affected are not listed.

Table 255: Summary of CCR changes

Component	Old	Expanded
Maximum number of Termination DN digits in the CCR ITR To-Route-A-Call message.	24	31

Summary of ISDN features enhancements

Table 256: Summary of ISDN changes on page 550 summarizes the changes made to the ISDN functionality.

Note:

The components that are not affected are not listed.

Table 256: Summary of ISDN changes

Component	Old	Expanded
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 can dial before a SETUP message is sent over the D-channel. **Note:** Overlap Signaling is not supported on North American interfaces.	16	24

Operating parameters

All existing operating parameters that apply to the system 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 the M2317 telephone, since its firmware limits entering a CFAC DN from the phone to a maximum of 23 digits.

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 are processed.

Flexible Feature Codes (FFC) DNs have a length of 1-7 digits. Call Forward All Calls (CFAC), Internal Call Forward All Calls (IFC) 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 IFC 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 have to be stored for automatic dialing. These digits are stored in sequence, and subsequent digits are 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, can be pushed out when the CPP digits are inserted, since only 31 digits can automatically be processed. This means that up to four digits can 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 base system features and ISDN PRI and BRI networking features, other than as described in Table 252:

Summary of ESN changes on page 547 to Table 256: Summary of ISDN changes on page 550 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
- Flexible Numbering Plan (FNP) package 160

For ISDN expansions:

- Digit Display (DDSP) package 19
- Integrated Services Digital Network (ISDN) package 145
- Primary Rate Access (PRI) package 146
- Integrated Service Digital Network Signaling Link (ISL) package 147
- 2.0 Mbit Primary Rate Interface (PRI2) package 154
- 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
- QSIG Interface (QSIG) package 263

For base system features expansions:

- Optional Features (OPTF) package 1
- 2500 Phone Features (SS25) package 18
- Basic Automatic Call Distribution (BACD) package 40
- Automatic Call Distribution Basic, Package B (ACDB) package 41

- Automatic Call Distribution Advanced, Package A (ACDA) package 45
- 500 Phone Features (SS5) package 73
- Controlled Class of Service (CCOS) package 81
- Background Terminal (BGD) package 99
- Flexible Feature Codes (FFC) package 139
- Phantom Terminal Numbers (PHTN) package 254

Feature implementation

Task summary list

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

Note:

For analog (500/2500-type) phones, configure the expansion pertaining to the Autodial, Call Forward All Calls, Default Call Forward (for Phantom TNs) and Internal Call Forward DNs.

Table 257: LD 10 - Configure expansion for analog (500/2500-type) phones.

Prompt	Response	Description
REQ:	NEW CHG	Add new data, or change existing data.
TYPE:	500	Type of phone.
TN		Terminal number
	l s c u	Format for Large System and Avaya Communication Server 1000E (Avaya CS 1000E) system, where l = loop, s = shelf, c = card, u = unit.
	c u	Format for Media Gateway 1000 B (Avaya MG 1000B), where c = card and u = unit.
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.

Note:

LD 11 for Meridian 1 Proprietary Phones, configure the expansion pertaining to the Autodial, Call Forward All Calls, Default Call Forward (for Phantom TNs) and Internal Call Forward DNs.

Table 258: LD 11 - Configure expansion for Meridian 1 Proprietary Phones.

Prompt	Response	Description
REQ:	NEW CHG	Add new data, or change existing data.
TYPE:	a...a	Telephone type. Type **?** for a list of possible responses.
TN		Terminal number
	l s c u	Format for Large System and CS 1000E system, where l = loop, s = shelf, c = card, u = unit.
	c u	Format for Media Gateway 1000B, where c = card and u = unit.

Prompt	Response	Description
KEY	nn ADL ll xxx...x	Phone function key assignments. Enter the maximum number of Autodial digits that can be stored. nn = Key number ll = 4,8,12,(16),20,24,28,31,NUL 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 can 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 can be stored for the Call Forward All Calls DN. nn = Key number ll = (4)-31 (for M2317 phones, 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 can 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.)

Table 259: LD 12 - Configure expansion for Attendant Consoles pertaining to the maximum length of the Autodial DN.

Prompt	Response	Description
REQ:	NEW CHG	Add new data, or change existing data.
TYPE:	2250	Attendant Console type.
TN		Terminal number
	l s c u	Format for Large System and CS 1000E system, where l = loop, s = shelf, c = card, u = unit.
	c u	Format for Media Gateway 1000B, where c = card and u = unit.
KEY		Attendant console function key assignments.
	nn ADL xxx...x	Enter the maximum number of Autodial digits that can be stored. nn = Key number xxx...x = the Autodial DN, up to 31 digits (this is an optional entry.)

Table 260: 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		Customer number

Prompt	Response	Description
	0-99	Range for Large System and CS 1000E system.
	0-31	Range for Media Gateway 1000B.
- 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.

Table 261: LD 23 - Configure the expansion pertaining to the ACD Night Call Forward DN and the ACD Interflow DN.

Prompt	Response	Description
REQ	NEW	Add new data.
	CHG	Change existing data.
TYPE	ACD	Automatic Call Distribution data block.
CUST	xx	Customer number, as defined in LD 15.
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.)

Note:

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.

Table 262: LD 86 - Configure the expansion for the ESN data block.

Prompt	Response	Description
REQ	NEW	Add new data.
	CHG	Change existing data.
CUST	xx	Customer number, as defined in LD 15.
FEAT	ESN	ESN data block feature.
...		

Prompt	Response	Description
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)-2000 (0)-256 (0)-32	Enter the maximum number of Digit Manipulation tables. (0)-2000 = 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.

Note:

LD 86 - For the Digit Manipulation data block, configure the expansion pertaining to the maximum number of leading digits to be deleted and inserted.

Table 263: LD 86 - Configure the expansion for the Digit Manipulation data block.

Prompt	Response	Description
REQ	NEW	Add new data.
	CHG	Change existing data.
CUST	xx	Customer number, as defined in LD 15.
FEAT	DGT	Digit Manipulation data block feature.
...		
DMI	1-1999 1-255 1-31	Enter the Digit Manipulation table. 1-1999 = 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.

Note:

LD 86 for the Route List data block, configure the expansion pertaining to the minimum Overlap Digit Length.

Table 264: LD 86 - Configure the expansion for the Route List data block.

Prompt	Response	Description
REQ	NEW	Add new data.
	CHG	Change existing data.
CUST	xx	Customer number, as defined in LD 15.
FEAT	RLB	Route List data block feature.
RLI	0-1999	Route List Index.
DMI	1-1999 1-255 1-31	Enter the Digit Manipulation table. 1-1999 = 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.

Note:

LD 87 for a Free Special Number Screening Index, configure the expansion pertaining to the maximum number of Special Numbers to be screened.

Table 265: LD 87 - Configure the expansion for a Free Special Number Screening Index.

Prompt	Response	Description
REQ	NEW	Add new data.
	CHG	Change existing data.
CUST	xx	Customer number, as defined in LD 15.
FEAT	FSNS	Free Special Number Screening.
FSNI	1-255	Free Special Number Screening Index.
SPN	1-19	Special Number to be screened.

Note:

LD 87 for a Trunk Steering Code, configure the expansion pertaining to the maximum number of Flexible Numbers.

Table 266: LD 87 - Configure the expansion for a Trunk Steering Code.

Prompt	Response	Description
REQ	NEW	Add new data.
	CHG	Change existing data.
CUST	xx	Customer number, as defined in LD 15.
FEAT	CDP	Coordinated Dialing Plan.
TYPE	TSC	Trunk Steering Code.
TSC	1-7	Trunk Steering Code.
- FLEN	0-24	Flexible Number.

Note:

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 for each entry, and the addition of allow (ALOW) as a new Supplemental Digit Restriction and Recognition type.

Table 267: LD 90 - Configure the expansion for Network Translation tables.

Prompt	Response	Description
REQ	NEW	Add new data.
	CHG	Change existing data.
CUST	xx	Customer number, as defined in LD 15.
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 can 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 of allowing one less digit if the first digit of a digit string is not "1", is removed.
- 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	Type of Supplemental Digit Restriction and Recognition. Restricted codes. Recognized local DID codes. Recognized local

Prompt	Response	Description
	DDD ITED ARRN STRK ALOW	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 ALOW 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.
- - ALOW	x...x	Code to be allowed.

Feature operation

There are no operating procedures specified for this feature.

Chapter 30: ESN interworking

Contents

This section contains information on the following topics:

Feature description

The Electronic Switched Network (ESN) operates within ISDN, supporting Network Class of Service (NCOS) capability. The tables in this module describe which ESN features are available over ISDN and illustrate ESN capabilities over ISDN.

Network Class Of Service (NCOS) information is embedded in SETUP messages, and provides the means to control a user's eligibility for the following:

- access routes
- access queuing
- receive Expensive Route Warning Tone (ERWT)
- access network speed call

When NCOS information, which includes Traveling Class of Service (TCOS) information, is sent to a DMS-100, the DMS switch can provide access to AT&Ts Electronic Tandem Network (ETN).

ESN translation

Currently, a system switch with a BARS or NARS package can have two (or one) separate ESN translators for handling BARS/NARS calls. NARS can have two translators and BARS can have only one translator (AC1). This is in addition to its standard translator, which handles all other call types. Each NARS/BARS translator has its access code (ESN AC1/AC2), which is defined on the standard translator. The remaining digits, for example, NPA, NXX, LOC, and SPN, are defined on the NARS/BARS translator.

Therefore, a receiving switch must determine which translator to use and insert the needed NARS/BARS access code (unless the receiving number is complete, that is, it includes the proper NARS/BARS access code in its digit stream already). This can be done by configuring an ESN digit manipulation to insert the ESN access code in the sending switch.

To insert the needed BARS/NARS access code in the receiving switch, two route options in LD 16 can be used:

- INST: An incoming route option that inserts the specified digits to the incoming digit stream for all the calls received on that route.
- INAC: An incoming ISDN route option that inserts the needed NARS/BARS access code to the incoming digit stream. The insertion is based on the Type of Number of the received Called Number Information Element, bypassing the digit insertion (INST) of the route. The AC2 option in LD 15 provides a mapping between the incoming Type of Number and the desired translator. This is used to determine which NARS/BARS access code is inserted.

ESN access code configuration with package 148

For system configurations that utilize ISDN networking features, ESN access code insertion must be performed on the receiving system. The originating system must not use digit manipulation to insert and send the ESN access code to the terminating system.

Therefore, in order to accomplish networking feature transparency (requiring package 148), the ESN access code insertion must be performed in the receiving system, by means of the INAC prompt in LD 16. Although the INST or the INAC prompt can be used, the INAC prompt is the recommended method.

When the PBX is upgraded, the above database conversion is required to allow networking applications (such as NRAG) to function properly.

Operating parameters

There are no operating parameters associated with this feature.

Feature interactions

There are no feature interactions associated with this feature.

Feature packaging

This feature is included in base System Software.

Feature implementation

Task summary list

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

1. Table 269: LD 86 - Configure digit manipulation. on page 564
2. Table 270: LD 16 - Enable the INAC option. on page 565
3. Table 271: LD 15 - Map the NARS/BARS access code to the incoming call types. on page 565 ESN access code insertion is performed in the receiving switch either by the INST or INAC option in

ESN default call types

When an ISDN call is made, a default Electronic Switched Network (ESN) call type is set initially. The call type is set depending on the dialing method used (that is, how the call is made). Table 268: Default Call Types on page 564 shows the ISDN default call types. LD 86 describes how to define the default call types.

Note:

When a trunk access code is used to dial on an ISDN route, the caller dials all digits without waiting for subsequent dial tones.

Table 268: Default Call Types

Dial method	Default call type	Comment
Direct dial	UKWN (unknown)	Direct dial to trunks through access code
CDP (DSC/ TSC)	CDP (Coordinated dialing plan)	Dial CDP number through DSC/TSC code
CDP (LSC)	UKWN	Dial CDP number through LSC code
ESN (LOC)	LOC (location code)	Dial NARS location code
ESN (HLOC)	UKWN	Dial ESN HLOC number
ESN (NPA)	NPA (national)	Dial ESN NPA number
ESN (NXX)	NXX (local)	Dial ESN NXX number
ESN (SPN)	SPN (special number)	Dial ESN SPN number

Table 269: LD 86 - Configure digit manipulation.

Prompt	Response	Description
REQ	NEW	Add new data.
	CHG	Change existing data.
CUST	xx	Customer number, as defined in LD 15.
FEAT	DGT	Digit manipulation data block
DMI	1–255	Digit manipulation table index numbers for NARS/BARS
CTYP		Call type. Enter the call type to the manipulated digits. This call type must be recognized by the far end switch.
	(NCHG)	The call type will not be changed
	INTL	Special number in international format
	NPA	NPA
	NXX	NXX
	LOC	Location Code
	CDP	Coordinated Dialing Plan
	SPN	Special number
	UKWN	Unknown call type

ESN access code insertion

LD 16 and LD 15, starting on Table 270: LD 16 - Enable the INAC option. on page 565, describe how to insert the needed NARS/BARS access code to the incoming digit stream. This is required for ISDN networking features such as Network Ring Again (NRAG) and Network Message Services (NMS).

ESN access code insertion is performed in the receiving switch either by the INST or INAC option in LD 16. The INAC option is recommended. The called number can go through ESN digit manipulation, which must not include the ESN access code in the sending switch.

Table 270: LD 16 - Enable the INAC option.

Prompt	Response	Description
REQ	NEW	Add new data.
	CHG	Change existing data.
TYPE	RDB	Route data block
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.

Table 271: LD 15 - Map the NARS/BARS access code to the incoming call types.

Prompt	Response	Description
REQ:	NEW CHG	Add new data, or change existing data.
TYPE:	NET	Networking data.
AC2		Access code 2. Enter call types (type of number) that use access code 2. Multiple responses are permitted. 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
ISDN	YES	Integrated Services Digital Network.
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). For different

Prompt	Response	Description
		customers, the PNI in LD 15 should match the PNI in LD 16 at the far end switch (M1/SL-1 only).

Private Network Hopoff

Private Network Hopoff is provided by the ESN feature. Private Network Hopoff allows ISDN callers to use the private network to complete public calls. The call is routed through the private network as far as possible before "hopping off" onto the public lines (see Figure 68: Private Network Hopoff on page 566).

Users select this service, on a single call basis, by dialing the private network facilities prefix digit, followed by the public network number. For example, the number 9 is often used as a private network facilities prefix digit before dialing a local number.

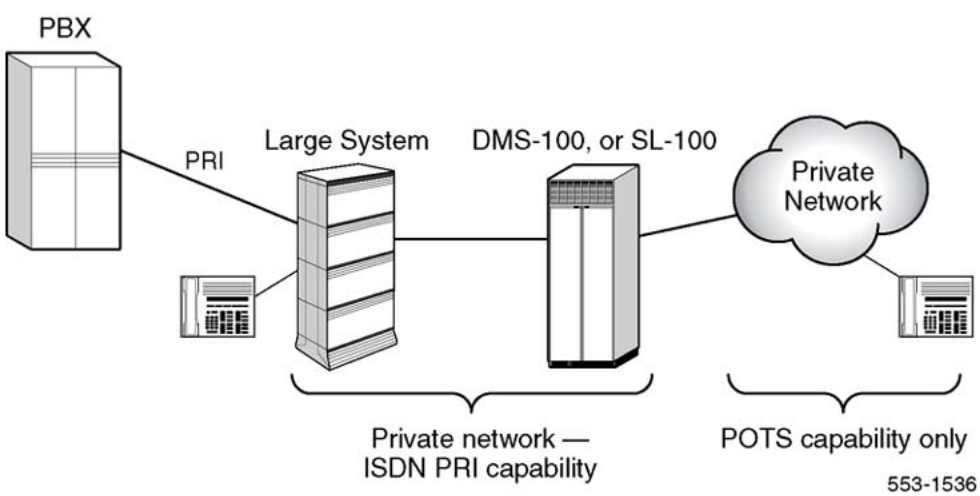


Figure 68: Private Network Hopoff

Private Network Overflow

This service allows overflow from a private network onto public network facilities (see Figure 69: Private Network Overflow on page 567). On outgoing calls over a DMS-100 PRI connection, the DMS provides network overflow when all of a customer's private trunks are busy.

For outgoing calls that encounter blocking (all B-channels are busy), the network provides the capability to overflow to alternate trunks. This is provided by the Electronic Switched Network (ESN) feature. This access can be restricted using a trunk or set class of service.

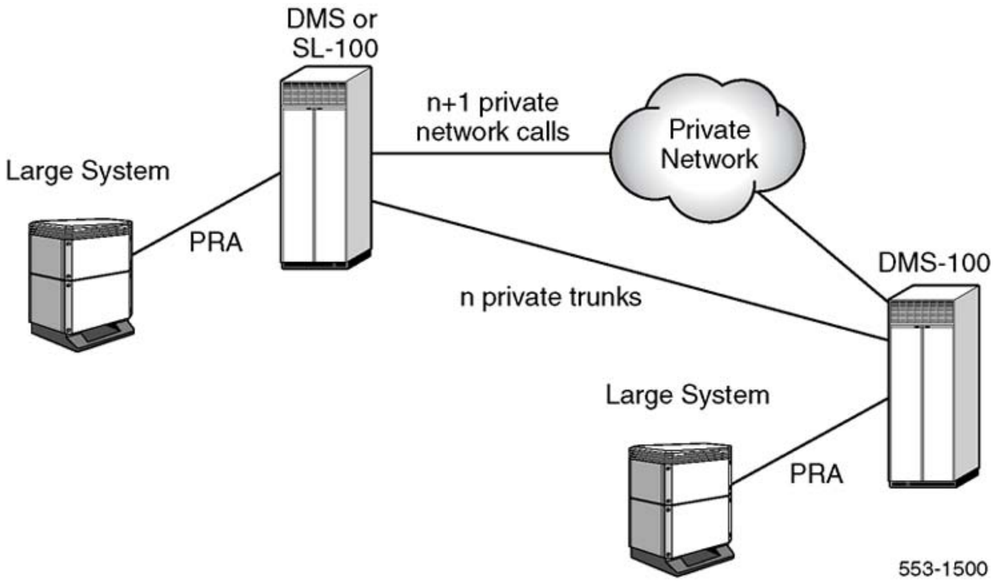


Figure 69: Private Network Overflow

Feature operation

No specific operating procedures are required to use this feature.

Chapter 31: Emergency Services Access

The Emergency Services Access (ESA) feature places a customer in compliance with new federal legislation that requires the Private 911 type of functionality provided by ESA. Please note, however, that the ESA feature is also generally useful for users who are not subject to legislation, and is broad enough to be used in different countries.

For example, it will be appreciated by any customer who wants to route emergency calls in a special manner, or who wants to be notified when a phone user makes an emergency call. It would also appeal to a customer who wishes to have ESA calls answered on-site, on the business premises, rather than being forwarded to the Public Services Answering Point (PSAP).

For complete information on ESA, please refer to *Avaya Emergency Services Access Fundamentals, NN43001-613*.

Chapter 32: Equi-distribution Routing Network Attendant Service

Contents

This section contains information on the following topics:

Feature description

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 systems using Meridian Customer Defined Network (MCDN) signaling over Integrated Services Digital Network Signaling Link (ISL)/ISDN TIE links.

All nodes in the system network should have the Equi-distribution NAS Routing feature.

This feature is not 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 for each 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

The following packages are required for Equi-distribution Routing and Network Attendant Service:

- Network Alternate Route Selection (NARS) package 58
- Integrated Services Digital Network Signaling (ISDN) package 145, or ISDN Signaling Link (ISL) package 147
- Network Attendant Service (NAS) package 159

Feature implementation

Task summary list

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

Table 272: LD 15 - Configure the EFLL.

Prompt	Response	Description
REQ:	CHG	Change existing data.
TYPE:	ATT	Attendant Console options.
CUST		Customer number
	0-99	Range for Large System and Avaya Communication Server 1000E (Avaya CS 1000E) system.
	0-31	Range for Media Gateway 1000 B (Avaya MG 1000B).
EFLL	(0)-8064	Efficiency Factor Loading Level. 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.

Table 273: LD 86 - Define the attendant alternatives for NAS routing in the Electronic Switched Network.

Prompt	Response	Description
REQ	CHG	Change.
CUST	xx	Customer number, as defined in LD 15.
FEAT	NAS	Network Attendant Service
...		
ALST	1-7	Alternatives List. Attendant alternatives for NAS routing (up to 4 alternatives are defined for each schedule period).
DBK	(N) Y	Drop Back busy option.

Prompt	Response	Description
		YES = Drop Back Busy allowed, NO = Drop Back Busy denied.

Table 274: LD 93 - Configure the EFLL.

Prompt	Response	Description
REQ	CHG	Change.
TYPE	aaa	Type of data block.
CUST	xx	Customer number, as defined in LD 15.
...		
EFLL	(0)-8064	Efficiency Factor Loading Level. 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.

Feature operation

No specific operating procedures are required to use this feature.

Chapter 33: Error Handling on ISDN Signaling Link Analog TIE Trunks

Contents

This section contains information on the following topics:

Feature description

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 (2Mbit).

Feature interactions

There are no feature interactions associated with this feature.

Feature packaging

The following package is required for Error Handling on ISL Analog E&M TIE Trunks:

- ISDN Signaling Link (ISL) package 147, which is dependent on ISDN Signaling (ISDN) package 145.

Feature implementation

Table 275: LD 16 - Configure ISL on Analog Trunks.

Prompt	Response	Description
REQ	NEW	Add new data.

Prompt	Response	Description
	CHG	Change existing data.
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.

Feature operation

Overlay Changes

The enabling/disabling commands in the maintenance overlay is changed so that if the trunk unit is in an 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 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.

Table 276: System messages

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.

Chapter 34: EuroISDN 7kHz/ Videotelephony Teleservices

Contents

This section contains information on the following topics:

Feature description

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 types of communication:

- 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 requests can be made either from a local ETSI BRI phone supporting the teleservice, or from a remote phone whose request is received over a EuroISDN link. The terminating party can accept the call, if the teleservice is supported end to end, or reject the call, if the requested teleservice is not supported.

Note:

A videotelephone terminal does 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 is established with the requested teleservice or rejected.

If the call request reaches a non EuroISDN link, or if the requested phone is a non-ETSI BRI phone, the call is rejected. For analog (500/2500-type) phones and Meridian Proprietary Phones, the call is rejected. The call is also rejected for calls over a EuroISDN link and for ETSI BRI phones 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 phones).

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 phone 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 is applied if Call Forward All Calls was activated. If fall-back is not allowed, the call is treated as a normal operation.

If fall-back occurred, a call request for the alternate (3.1 kHz) teleservice is sent to the forwarded-to phone or trunk. If fall-back has not already occurred, the call is treated as a normal operation for the forwarded-to phone or trunk.

Call forward Busy

If Call Forward Busy feature has been activated on a busy phone, 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 phone, the call is treated as a normal operation for the forwarded-to phone 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 phone or trunk. If fall-back is allowed and has not already occurred, the call is treated as a normal operation for the forwarded-to phone or trunk.

Hunt

If Hunt has been applied to a busy destination, destination phone is busy, and if the Hunt feature is configured, the call is treated as a normal operation for the hunted-to phone or trunk.

DISA call

For a DISA call requesting 7 kHz or videotelephony teleservice, fall-back occurs, if allowed, or the call is rejected.

ACD DN

For a request for 7 kHz or videotelephony teleservice terminating on an ACD DN, fall-back occurs, if allowed, or the call is rejected.

Advice Of 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 (non-BRI) phones is presented to all phones. For an incoming call to a Multiple Appearance DN requesting the 7 kHz or videotelephony teleservice, fall-back occurs if allowed, or 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 occurs further in the network, a voice call can 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.

Feature packaging

This feature requires the following packages:

For ISDN PRI2:

- Integrated Services Digital Network (ISDN) package 145
- 2.0 Mbit/s Primary Rate Interface (PRI2) package 154
- ISDN Supplementary Features (ISDN_SUPP) package 161
- International Primary Rate Access (IPRA) package 202
- Multi-Purpose Serial Data Link (MSDL) package 222
- EuroISDN (EURO) package 261

For ISDN Basic Rate trunking:

- Integrated Services Digital Network (ISDN) package 145
- ISDN Supplementary Features (ISDN_SUPP) package 161
- International Primary Rate Access (IPRA) package 202
- Basic Rate Interface (BRI) package 216
- Basic Rate Interface Trunking (BRIT) package 233
- EuroISDN (EURO) package 261

For ISDN Basic Rate line application:

- Basic Rate Interface Line (BRIL) package 235

Feature implementation

Task summary list

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

1. Table 277: LD 17 - Configure D-Channel for the ETS 300 403 interface. on page 585
2. Table 278: LD 16 - Configure the 7 kHz/Videotelephony teleservices for a PRI2 EuroISDN E403 route. on page 585
3. Table 279: LD 14 - Configure the 7 kHz/Videotelephony teleservices PRI2 EuroISDN E403 trunks. on page 586
4. Table 280: LD 16 - Configure the 7 kHz/Videotelephony teleservices for an ISDN BRI EuroISDN E403 route, in the Route Data Block. on page 587

5. Table 281: LD 27 - Configure the ETSI ISDN BRI phones, on the Digital Subscriber Loop (DSL), for voice and data. on page 588
6. Table 282: LD 27 - Configure the Terminal Service Profile (TSP) for voice and data, and the 7 kHz/Videotelephony teleservices on the ETSI ISDN BRI phones. on page 589

ISDN PRI2 implementation

Table 277: 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.

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.

Table 278: LD 16 - Configure the 7 kHz/Videotelephony teleservices for a PRI2 EuroISDN E403 route.

Prompt	Response	Description
REQ	CHG	Change existing data.
TYPE	RDB	Route Data Block.
CUST	xx	Customer number, as defined in LD 15.
ROUT		Route number
	0-511	Range for Large System and Avaya Communication Server 1000E (Avaya CS 1000E) system.
	0-127	Range for Media Gateway 1000 B (Avaya MG 1000B).
TKTP	aaaa	Trunk type.
DTRK	YES	EuroISDN route is digital.

Prompt	Response	Description
- DGTP		Digital trunk type.
	PRI2	PRI2 = 2.0 Mbit/s 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 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.
	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.

Table 279: LD 14 - Configure the 7 kHz/Videotelephony teleservices PRI2 EuroISDN E403 trunks.

Prompt	Response	Description
REQ	NEW	Add new data.
	CHG	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.
RTMB		Route number and Member Number
	0-511 1-4000	Range for Large System and CS 1000E system.

Prompt	Response	Description
	0-127 1-4000	Range for Media Gateway 1000B.

ISDN Basic Rate Interface implementation

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.

Table 280: LD 16 - Configure the 7 kHz/Videotelephony teleservices for an ISDN BRI EuroISDN E403 route, in the Route Data Block.

Prompt	Response	Description
REQ	CHG	Change existing data.
TYPE	RDB	Route Data Block.
CUST	xx	Customer number, as defined in LD 15.
ROUT		Route number
	0-511	Range for Large System and CS 1000E system.
	0-127	Range for Media Gateway 1000B.
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 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.
	AUS DEN FIN GER ITA NOR POR SWE	Austria. Denmark. Finland. Germany. Italy. Norway. Portugal. Sweden. Ireland. Holland. Switzerland. Belgium. Spain. United Kingdom. France. Commonwealth of Independent States (Russia and the Ukraine).

Prompt	Response	Description
	EIR DUT SWI BEL ESP UK FRA CIS	
...		
- 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.

Table 281: LD 27 - Configure the ETSI ISDN BRI phones, on the Digital Subscriber Loop (DSL), for voice and data.

Prompt	Response	Comment
REQ	NEW	Add new data.
	CHG	Change existing data.
TYPE	DSL	Digital Subscriber Loop data block.
DSL	l s c dsl#	DSL location, where: • lll (superloop) = 0-156 (must be zero or a number divisible by 4) • s (shelf) = 0-1 • cc (card) = 0-15 and for CS 1000E 1–20 • dsl# (DSL location) = 0-7
CUST	xx	Customer number, as defined in LD 15.
CTYP	SILC UILC	Interface card type is SILC or UILC.
APPL	BRIL	Basic Rate Interface.
MISP	0-158	MISP loop number.
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.

Table 282: LD 27 - Configure the Terminal Service Profile (TSP) for voice and data, and the 7 kHz/Videotelephony teleservices on the ETSI ISDN BRI phones.

Prompt	Response	Comment
REQ	NEW	Add new data.
	CHG	Change existing data.
TYPE	TSP	Terminal Service Profile data block.
DSL	lll s cc dsl#	DSL location, where: • lll (superloop) = 0-156 (must be zero or a number divisible by 4) • s (shelf) = 0-1 • cc (card) = 0-15 and for CS 1000E 1–20 • dsl# (DSL location) = 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	The ETSI ISDN BRI phone supports the 7kHz/Videotelephony teleservices.
	XVID7	Precede with an X to remove the configured 7kHz/Videotelephony teleservices.

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 an 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 specific operating procedures are required to use this feature.

Chapter 35: EuroISDN Continuation

Contents

This section contains information on the following topics:

Applicable regions

ISDN BRI trunking is not available in North America. Contact your system supplier or your Avaya representative to verify support of this product in your area.

Feature description

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) phone
- the configuration of Connected Line Presentation (COLP) on each 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) phone
- 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 single configuration basis
- capability of using different options to build the Calling Line Identification (CLID)
- called party number size increase to 31 octets

Note:

The older ESTI standards support a length of 23 octets only.

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 in the 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.

If more than one Channel Identification Information Element (IE) is received in a SETUP message, only the first one is used by the system. If the Information Element is not available, the call is processed as each channel negotiation configuration. If no negotiation is allowed, the call is released.

The transfer of an unanswered EuroISDN call to a remote ringing phone 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 phones on an individual 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 system.

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 phone 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 phone.

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 permits the Public Switched Telephone Network (PSTN) to start charging the caller. However, an unsupervised trunk can also be called. In this case, the CONNECT message is sent up on the expiry of the pseudo answer supervision timer.

The destination non-ISDN trunk can 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 system 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 system. These features are listed inTable 283: Networking features supported by more than one ISDN interface on page 596. 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 283: Networking features supported by more than one ISDN interface

	Euro - ISDN	MCDN	QSIG	BRI phones	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 phone dialing an external number, before the external phone answers, to both local and remote phones 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
- 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
- Digit Display (DDSP) package 19
- Network Call of Service (NCOS) package 32
- Basic Alternate Route Selection (BARS) package 57
- 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
- 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

Feature implementation

Task summary list

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

- Table 284: LD 10 - Configure Fax Allowed Class of Service for analog (500-type) phone. on page 599
- Table 285: LD 17 - Create a new D-channel. on page 599

- Table 286: LD 16 - Configure new DID routes (enter the same responses to the IFC and CNTY prompts that were used in LD 17). on page 600
- Table 287: LD 14 - Configure new DID trunks. on page 602

Table 284: LD 10 - Configure Fax Allowed Class of Service for analog (500-type) phone.

Prompt	Response	Description
REQ:	NEW	Add new data.
TYPE:	500	Type of phone.
...		
CLS	FAXA	Fax Class of Service allowed for phone or modem. ISDN call is generated with 3.1 KHz bearer capability. FAXD = Fax Class of Service denied.

Table 285: 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	Card type, where: MSDL = Multipurpose Serial Data Link.
- GRP	0-4	Network Group Number.
- DNUM	0-15	Device number for I/O ports.
- PORT	0-3	Port number on MSDL card.
- 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.
...		

Prompt	Response	Description
- - 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_C UST	xx	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.

Table 286: LD 16 - Configure new DID routes (enter the same responses to the IFC and CNTY prompts that were used in LD 17).

Prompt	Response	Description
REQ	NEW	Add.
TYPE	RDB	Route Data Block.
CUST	xx	Customer number, as defined in LD 15.
ROUT		Route number
	0-511	Range for Large System and Avaya Communication Server 1000E (Avaya CS 1000E) system.
	0-127	Range for Media Gateway 1000 B (Avaya MG 1000B).
TKTP	DID	Trunk type, where: DID = Direct Inward Dialing.
DTRK	YES	Digital trunk route.
- DGTP	PRI2 BRI	Digital trunk type for route. 2.0 Mbit Primary Rate Access. ISDN Basic Rate Access.
- IFC	EURO SWIS	Interface type for route EuroISDN 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.

Prompt	Response	Description
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.

Prompt	Response	Description
		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 = YES, the prefixes 0 (national) or 00 (international) are added to the Calling Party Number if the Type of Number (TON) is public. If ADDP = NO, the Calling or Connected Party Number displayed is not modified.

Table 287: 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		Terminal number
	l s c u	Format for Large System and CS 1000E system, where l = loop, s = shelf, c = card, u = unit.
	c u	Format for Media Gateway 1000B, where c = card and u = unit.
RTMB		Route number and Member Number
	0-511 1-4000	Range for Large System and CS 1000E system.
	0-127 1-4000	Range for Media Gateway 1000B.

Feature operation

No specific operating procedures are required to use this feature.

Chapter 36: EuroISDN Continuation Phase III

Contents

This section contains information on the following topics:

Applicable regions

ISDN BRI trunking is not available in North America. Contact your system supplier or your Avaya representative to verify support of this product in your area.

Feature description

The EuroISDN feature provides ISDN PRI and ISDN BRI connectivity between the system and Public Exchanges that comply with 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 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), 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 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

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 LD 15 and which has three possible responses (NO, EXT, or ALL).

This functionality applies to the following types of calls being redirected at a gateway node.

Note:

There is a limitation on the functionary 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 connectivity, are forwarded back to the Public Network.

  Note:

  Standard ISDN connectivity means all supported ISDN connectivities including, EuroISDN, Asia Pacific, and NI-2.

  - 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 70: Incoming EuroISDN call redirected back to the Public Network/PBX on page 605.

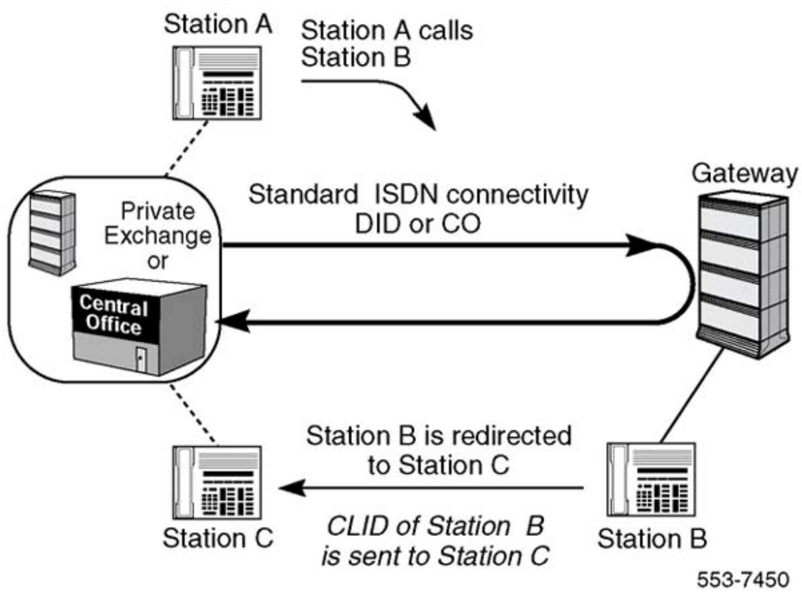


Figure 70: Incoming EuroISDN call redirected back to the Public Network/PBX

- 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 71: MCDN/QSIG call redirected to the Public Network/PBX on page 605.

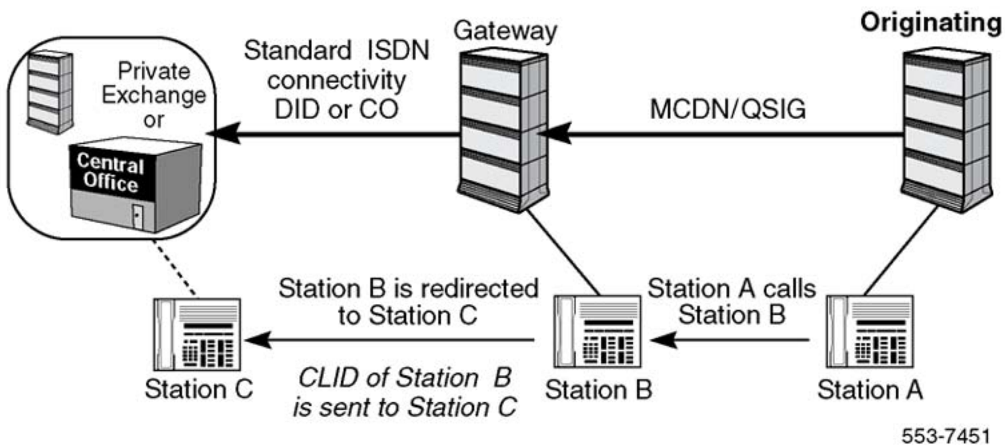


Figure 71: MCDN/QSIG call redirected to the Public Network/PBX

- 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 72: Internal call (stand-alone case) redirected to the Public Network/PBX on page 606.

Note:

If Station B is redirected to an ACD queue through 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.

There is a limitation on the functionary 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.

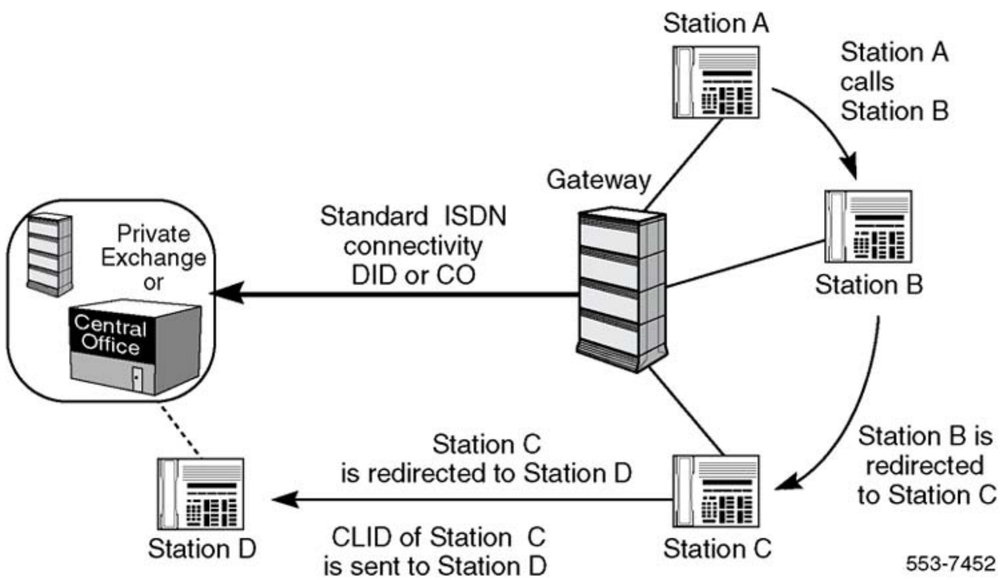


Figure 72: Internal call (stand-alone case) redirected to the Public Network/PBX

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 73: MCDN call redirected to the Public Network/PBX from a non-gateway node on page 607.

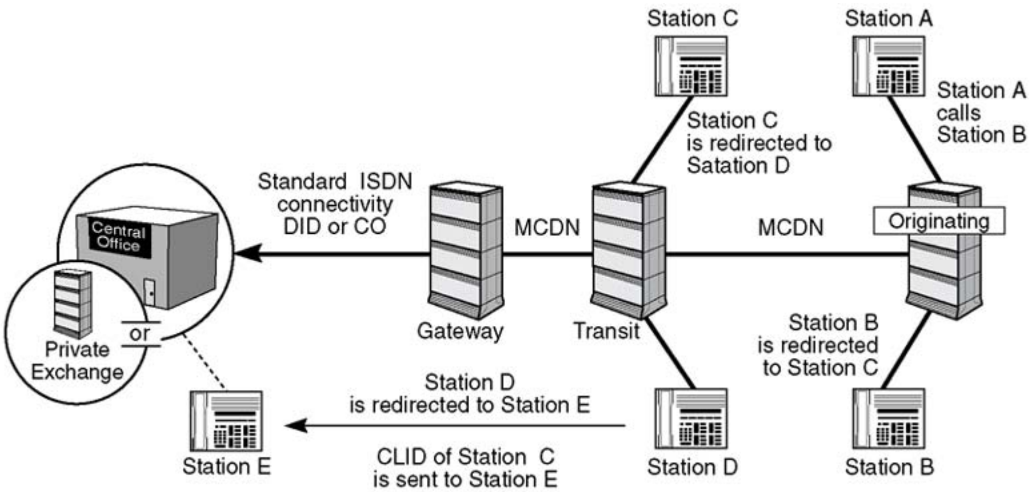


Figure 73: MCDN call redirected to the Public Network/PBX from a non-gateway node

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 74: EuroISDN calling scenario before Trunk Optimization is applied on page 607).

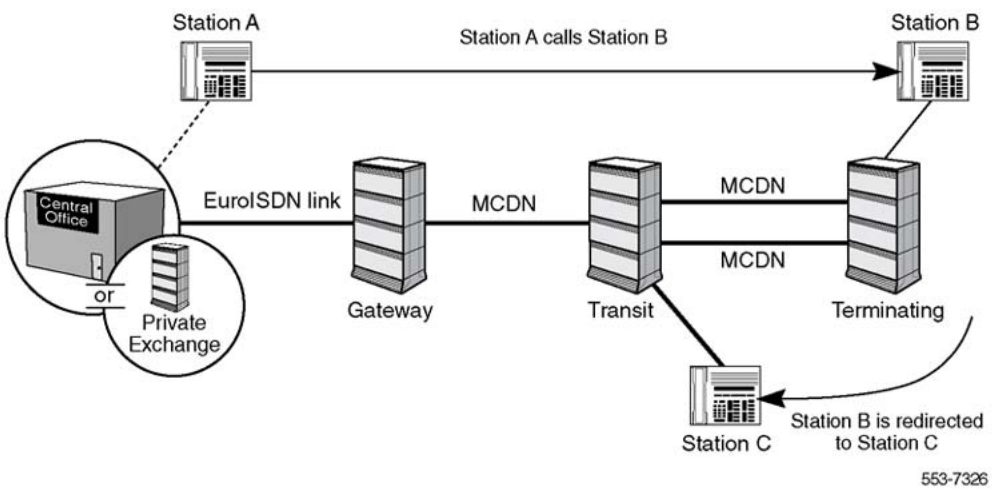


Figure 74: EuroISDN calling scenario before Trunk Optimization is applied

- 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 75: EuroISDN calling scenario after Trunk Optimization has been applied on page 608.

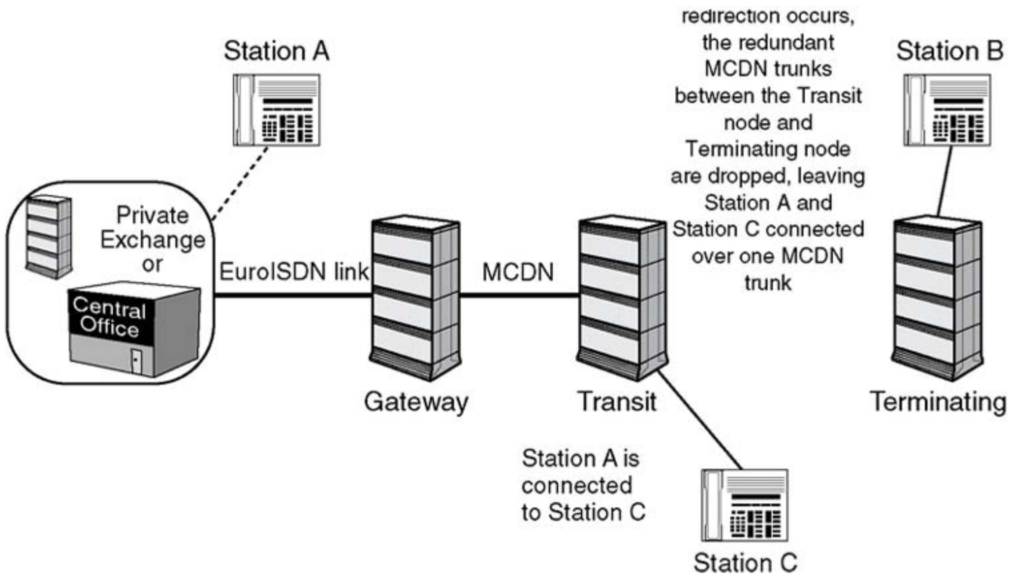


Figure 75: EuroISDN calling scenario after Trunk Optimization has been applied

Numbering Plan Identification and Type of Number in CDR tickets for EuroISDN calls

The Numbering Plan Identification (NPI) and Type of Number (TON) fields can 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 LD 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.

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 single call basis on non-ISDN BRI phones.

For the Optional Sending of Last Forwarding DN as CLID functionality, if more than one ACD redirection takes place when an ACD phone is involved, the DN of the called phone is sent as the CLID rather than the DN of the last forwarding phone. Refer to Optional sending of last forwarding DN as CLID on page 604 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 604.

CLID Enhancements

The Optional Sending of Last Forwarding DN as CLID functionality interacts with the ISDN CLID Enhancements feature, in that the CLID entry programmed against the prime DN of the forwarding phone 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 system.

Incoming Trunk Programmable CLID

The Optional Sending of Last Forwarding DN as CLID functionality interacts with the Incoming Trunk Programmable CLID feature. This feature provides the capability of assigning a billing number to incoming trunks.

In the scenario described on Figure 73: MCDN call redirected to the Public Network/PBX from a non-gateway node on page 607, 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 system.

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 can 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 can 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 system 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 system. These features are listed in Table 288: Networking features supported by more than one ISDN interface on page 612. 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 288: Networking features supported by more than one ISDN interface

	Euro-ISDN	MCDN	QSIG	ETSI BRI phones	DPNSS 1	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 phone) to transfer an unanswered outgoing EuroISDN call, after dialing has been completed and before the transferred-to phone answers (while the phone is ringing). The transferred-to phone can be local, or can 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 phone 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 phone answers, a CDR Start (S) is generated when the transferred-to phone 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 phone is ringing, an S ticket will be generated only when the transferred-to phone 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
- Digit Display (DDSP) package 19
- Network Call of Service (NCOS) package 32
- 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 implementation

Task summary list

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

1. Table 289: LD 15 - Configure the Optional Sending of Forwarding CLID functionality (in response to the OCLI prompt). on page 615
2. Table 290: LD 17 - Create a new D-channel for the EuroISDN interface. on page 616
3. Table 291: LD 16 - Configure new DID routes (enter the same responses to the IFC and CNTY prompts that were used in LD 17). on page 617
4. Table 292: LD 17 - Configure the NPI and TON fields to be included in the CDR tickets. on page 619

Table 289: 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		Customer number
	0-99	Range for Large System and Avaya Communication Server 1000E (Avaya CS 1000E) system.
	0-31	Range for Media Gateway 1000 B (Avaya MG 1000B).
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	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 system node, and redirected back over the standard ISDN (EuroISDN, APAC, NI-2) DID or CO link.
	ALL	ALL = The same as for EXT. Moreover, the last forwarding DN is sent as CLID information for internal calls from a local phone or

Prompt	Response	Description
		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.

Table 290: 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	Multipurpose Serial Data Link or Downloadable D-Channel card.
- GRP	0-7	Network Group Number.
- DNUM	0-15	Device number for I/O ports.
- PORT	0-3	Port number on MSDL card.
- 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 DEN (ETSI) FIN GER ITA NOR POR SWE EIR DUT SWI BEL ESP UK FRA	Austria Denmark ETS 300-102 basic protocol Finland Germany Italy Norway Portugal Sweden Ireland Holland Switzerland Belgium Spain United Kingdom France
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 =

Prompt	Response	Description
		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.

Table 291: LD 16 - Configure new DID routes (enter the same responses to the IFC and CNTY prompts that were used in LD 17).

Prompt	Response	Description
REQ	NEW	Add new data.
	CHG	Change existing data.
TYPE	RDB	Route Data Block.
CUST	xx	Customer number, as defined in LD 15.
ROUT		Route number
	0-511	Range for Large System and CS 1000E system.
	0-127	Range for Media Gateway 1000B.
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. 2.0 Mbit Primary Rate Access. ISDN Basic Rate Access.
- IFC		Interface type.
	EURO	EURO = EuroISDN.
- - CNTY		Enter country pertaining to EuroISDN and Asia Pacific interface type.
	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
ICOG	IAO OGT ICT	Incoming and Outgoing trunk. Outgoing trunk only. Incoming trunk only.
ACOD	x..x	Access code for this trunk route.

Prompt	Response	Description
- 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 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.
...		
- 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.

Prompt	Response	Description
- - 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)	If ADDP = NO, the Calling or Connected Party Number displayed is not modified.
	YES	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.

The following table describes 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).

Table 292: LD 17 - Configure the NPI and TON fields to be included in the CDR tickets.

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.

Feature operation

No specific operating procedures are required to use this feature.

Chapter 37: EuroISDN ETS 300 403 Compliance Update

Contents

This section contains information on the following topics:

Feature description

The EuroISDN feature was initially introduced to provide ISDN PRI and ISDN BRI connectivity between the system 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 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), and the United Kingdom.

EuroISDN Continuation Phase III, added France, Russia and the Ukraine to the list of countries having country-specific Application Documents comply with ETS 300 102 for EuroISDN.

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 upgraded to the ETS 300 403 standard, the system still interworks with Central Offices conforming to the ETS 300 102 standard.

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 teleservices offered, such as 7kHz telephony and Videotelephony.

Note:

A user can 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 signaling 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.

Existing EuroISDN functionalities supported by the EuroISDN ETS 300 403 Compliance Update feature

This section summarizes the existing EuroISDN functionalities that are supported by the EuroISDN ETS 300 403 Compliance Update feature.

The ETSI GF subset provides the following 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 system on the EuroISDN connectivities:

- Basic call service (3.1 kHz, speech, unrestricted digital information)
- 64 Kbit/s 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 293: Networking features supported by more than one ISDN interface on page 624 summarizes 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 293: Networking features supported by more than one ISDN interface

	Euro-ISDN	MCDN	QSIG	ETSI BRI phones	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:

This is supported using the ICCL tandem count feature existing over the NAS feature.

Note:

This is only supported for ETSI version of QSIG. For the ISO version, it is discarded.

Note:

This is supported using the Loop Avoidance supplementary service.

Note:

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 phone
- 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 phone
- Connected Line Presentation service configurable on a 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 phone

- 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 single configuration basis
- capability of using different options to build the CLID
- capability of sending a RELEASE message to a BRI phone, 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 phone originating call to busy or invalid DN.
  - when receiving a disconnect message from the Central Office through 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 single 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 through EuroISDN
  - incoming MCDN/QSIG calls forwarded to the public network through EuroISDN
  - internal calls (stand-alone case) being forwarded to the public network through EuroISDN
- trunk optimization before call establishment (TRO) is enhanced for incoming EuroISDN calls
- Type Of Number (TON) and Numbering Plan Identification (NPI) 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.

- 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 Mbit/s), 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 /NTBK51CA Downloadable D-Channel card is required onLarge Systems, and the NTBK51BA Downloadable D-Channel card is required on theSmall Systems).

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 system. If it is not available, the call is processed according to the channel negotiation configuration. The call is released if no negotiation is allowed. The transfer of a unanswered EuroISDN call to a remote ringing phone 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 system, but this information is not used to perform any checking on a system 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 system. As a result, this IE is never sent by the system.
- 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 system. No LLC is generated by the system, but this information is tandemed if received (from an ISDN BRI phone for example).

- Low layer compatibility negotiation procedures are not supported.
- The USER INFORMATION message is not implemented in the software.

Operating parameters pertaining to ETSI GF:

- ETSI GF gateways to and from other signaling 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 LD 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 system ETSI GF platform supports up to a maximum of eight ROSE components in one message. The eight components can 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 system. 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 system.

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 dialing 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, for each customer.

Each 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:

- Enbloc receiving is configured and an incomplete or invalid number has been received.
- Overlap receiving is configured, and:
    - 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 timers 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 phone 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 phone (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 phone and the Ter ID is the outgoing trunk.

The Recovery of Misoperation during Call Transfer function is applicable as a 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 phone 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 signaling

End to End signaling 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 signaling.

Flexible Hotline

This feature is not supported with Overlap signaling.

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 system is able, on a 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 through 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 phone 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 through 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 product improvement "Optional Sending of Forwarding CLID". The CLID entry programed against the prime DN key of the forwarding phone 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 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 system, 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 phones, 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 Phone, 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.

Virtual Network Service

VNS provides ISDN features when no MCDN link is available between two system 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 signaling 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 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 system 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 phone 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 Mbit 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 signaling, the following additional packages are required:

- Overlap signaling (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 implementation

Task summary list

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

1. Table 294: LD 16 - Configure a PRI2 loop for the ETS 300 403 interface. on page 637
2. Table 295: LD 17 - Configure a D-Channel for the ETS 300 403 interface. on page 637
3. Table 296: 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). on page 639
4. Table 297: LD 14 - Configure the PRI2 trunks for the E403 interface. on page 641

Table 294: 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.

Table 295: 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.

Prompt	Response	Description
- ADAN	NEW DCH xx	Add a D-Channel on port 0-63 for Large Systems.
	MSDL DCHI	Multipurpose Serial Data Link or Downloadable D-Channel card.
- GRP	0-4	Network Group Number.
- DNUM	0-15	Device number for I/O ports.
- PORT	0-3	Port number on MSDL card.
- 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. Refer to the Feature description on page 621.
- - 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 the Feature description on page 621.
	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 the Feature description on page 621.
	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.

Prompt	Response	Description
- - 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.

Table 296: 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 defined in LD 15.
ROUT		Route number
	0-511	Range for Large System and Avaya Communication Server 1000E (Avaya CS 1000E) system.
	0-127	Range for Media Gateway 1000 B (Avaya MG 1000B).
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. Refer to the Feature description on page 621.
- - 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 the Feature description on page 621.
	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 the Feature description on page 621.

Prompt	Response	Description
	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		Progress Signal.
	NCHG	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.
...		

Prompt	Response	Description
- 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 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 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.

Table 297: LD 14 - Configure the PRI2 trunks for the E403 interface.

Prompt	Response	Description
REQ	NEW	Add new data.
	CHG	Change existing data.
TYPE	TIE COT DID	TIE trunk data block. Central Office Trunk data block. Direct Inward Dialing trunk data block.

Prompt	Response	Description
		Note: Must match TKTP defined in LD 16.
TN		Terminal number
	l ch	Loop and channel for digital trunks, where: l = Previously defined PRI2 loop. ch = channel 1-24 for 1.5 Mb/s DTI/PRI or 1-30 for 2.0 Mb/s DTI/PRI
CUST	xx	Customer number, as defined in LD 15.
RTMB		Route number and Member Number
	0-511 1-4000	Range for Large System and CS 1000E system.
	0-127 1-4000	Range for Media Gateway 1000B.
TGAR	0 - (1) - 31	Trunk Group Access Restriction The default of 1 automatically blocks direct access.

Feature operation

No specific operating procedures are required to use this feature.

Chapter 38: EuroISDN Malicious Call Identification

Contents

This section contains information on the following topics:

Applicable regions

ISDN BRI trunking is not available in North America. Contact your system supplier or your Avaya representative to verify support of this product in your area.

Feature description

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) phones using a SPRE code and a two-digit Flexible Feature MCT access code or from Meridian 1 Proprietary Phones 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 76: EuroISDN MCID invoked during the active phase of a malicious call on page 644 illustrates a situation where EuroISDN MCID is initiated during the active phase of a malicious call.

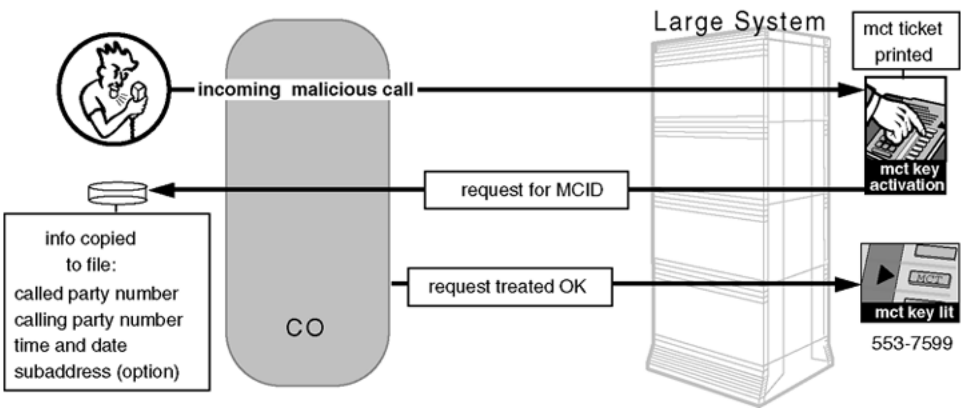


Figure 76: EuroISDN MCID invoked during the active phase of a malicious call

Figure 77: EuroISDN MCID invoked after the active phase of a malicious call on page 644 illustrates a situation where EuroISDN MCID is initiated after the active phase of a malicious call.

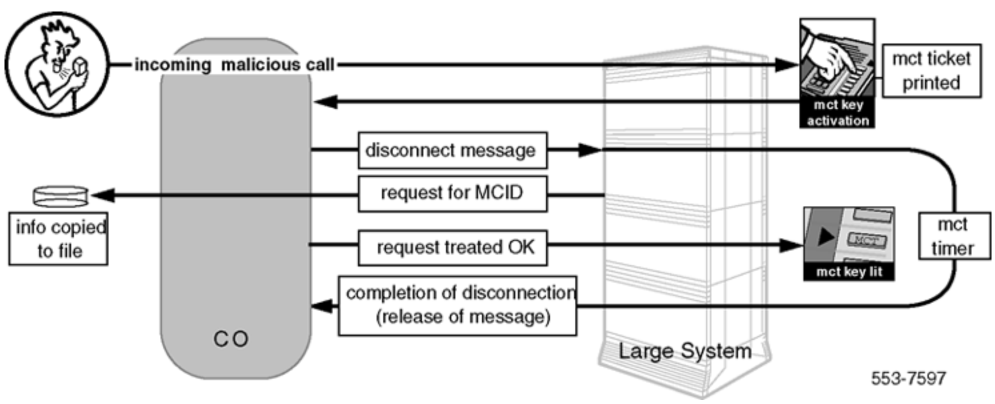


Figure 77: EuroISDN MCID invoked after the active phase of a malicious call

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 system switch supporting EuroISDN PRI/BRI connectivity.
- For tandem applications, another system 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 through 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 LD 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

The Malicious Call Trace (MCT) feature is only activated against one call at one time from either a phone or an attendant. If a phone 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) phones. However, it is supported over ISDN BRI trunks.

Soft keys cannot be defined as a TRC key on M2317 or M5317 digital phones.

For Large Systems, the EuroISDN Malicious Call Identification requires the following hardware:

- for Primary Rate Access, the NT6D72 PRI card or the NT5D97AD Dual DTI/PRI 2.0 Mbit/s 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 NT5D97AD Dual DTI/PRI 2.0 Mbit/s card instead of an MSDL card
- for Basic Rate Access, the NT6D73AA MISP card and the NT6D70BA SILC card for trunk access

When EuroISDN Malicious Call Identification is configured on a phone, 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 phone 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

This feature requires the following packages:

- Malicious Call Trace (MCT) package 107
- International Supplementary Features (SUPP) package 131
- Flexible Feature Code (FFC) package 139 for analog (500/2500-type) phones
- Integrated Services Digital Network (ISDN) package 145
- Multipurpose Serial Data Link (MSDL) package 222
- Euro ISDN (EURO) package 261
- Universal ISDN Gateway (UIGW) package 283
- Euro Supplementary Service (ETSI_SS) package 323

Primary Rate Interface (PRI) connectivity requires the following packages:

- International Primary Rate Access CO (PRA) package 146
- 2.0 Mbit Primary Rate Interface (PRI2) package 154
- Integrated Services Digital Network Supplementary Features (ISDN INTL SUP) package 161
- 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 implementation

Task summary list

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

1. Table 298: LD 17 - Configure remote D-channel capability for EuroISDN interface for PRI and BRI. on page 648
2. Table 299: LD 16 - Configure remote capability for EuroISDN interface for PRI. on page 648

3. Table 300: LD 16 - Configure remote capability for EuroISDN interface for BRI. on page 649
4. Table 301: LD 15 - Modify system and software parameters. on page 649
5. Table 302: LD 10 - Configure analog (500/2500-type) phones. on page 649
6. Table 303: LD 11 - Configure Meridian 1 Proprietary Phones. on page 650
7. Table 304: LD 12 - Configure attendant consoles. on page 650
8. Table 305: LD 57 - Configure Flexible Feature Codes for analog (500/2500-type) phones. on page 650
9. Table 306: LD 16 - Configure Malicious Call Trace Timer and Tandem Delay. on page 651

Table 298: 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.

Table 299: 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.

Table 300: 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.

Table 301: 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.

Table 302: LD 10 - Configure analog (500/2500-type) phones.

Prompt	Response	Description
REQ:	CHG	Change existing data.
TYPE:	500	Type of phone.
TN		Terminal number
	l s c u	Format for Large System and Avaya Communication Server 1000E (Avaya CS 1000E) system, where l = loop, s = shelf, c = card, u = unit.
	c u	Format for Media Gateway 1000 B (Avaya MG 1000B, where c = card and u = unit.
...		
CLS	MCTA	Malicious Call Trace allowed. (MCTD) = Malicious Call Trace denied.

Table 303: LD 11 - Configure Meridian 1 Proprietary Phones.

Prompt	Response	Description
REQ:	CHG	Change existing data.
TYPE:	a...a	Telephone type. Type **?** for a list of possible responses.
TN		Terminal number
	l s c u	Format for Large System and CS 1000E system, where l = loop, s = shelf, c = card, u = unit.
	c u	Format for Media Gateway 1000B, where c = card and u = unit.
CLS	MCTA	Malicious Call Trace allowed. MCTD = Malicious Call Trace denied (default).
KEY	xx TRC	Trace key number.

Table 304: LD 12 - Configure attendant consoles.

Prompt	Response	Description
REQ:	CHG	Change existing data.
TYPE:	2250	Type of Attendant Console.
TN		Terminal number
	l s c u	Format for Large System and CS 1000E system, where l = loop, s = shelf, c = card, u = unit.
	c u	Format for Media Gateway 1000B, where c = card and u = unit.
CLS	MCTA	Malicious Call Trace allowed. MCTD = Malicious Call Trace denied (default).
KEY	xx TRC	Trace key number.

Table 305: LD 57 - Configure Flexible Feature Codes for analog (500/2500-type) phones.

Prompt	Response	Description
REQ	NEW	Add new data.
	CHG	Change existing data.
TYPE	FFC	Flexible Feature Code data block.
CUST	xx	Customer number, as defined in LD 15.
...		
CODE	MTRC	Malicious Call Trace code
- MTRC	xxxx	Enter Flexible Feature Code for Malicious Call Trace.

Note:

This configuration is only required to handle Malicious Call Trace during call disconnection.

Table 306: LD 16 - Configure Malicious Call Trace Timer and Tandem Delay.

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.

Feature operation

No specific operating procedures are required to use 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) phone, 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 Phones, 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 Phones 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 phone 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 can be activated to initiate the call identification request again.

Chapter 39: EuroISDN Trunk - Network Side

Contents

This section contains information on the following topics:

Applicable regions

ISDN BRI trunking is not available in North America. Contact your system supplier or your Avaya representative to verify support of this product in your area.

Feature description

EuroISDN Trunk - Network Side refers to the behaviour of EuroISDN as a network interface (generally referred to as "network side"), where the EuroISDN interface 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. 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 system. Another arrangement could be the connection of key systems to a system, 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 system to achieve a greater public network connection efficiency. Such access to the system can be through 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)
- 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 its Bearer Capability. Any ISDN PRI or ISDN BRI route can be configured (in LD 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 with 3.1 KHz.

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 (system to system, system to SL-100, system to DMS-100/250,system to AT&T 4ESS, system 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 Large Systems.

For ISDN PRI access:

- NT8D72BA PRI2 card or the NT5D97AD Dual DTI/PRI 2.0Mbs card
- for D-Channel handling, the NT6D80 MSDL card or the NTBK51BA Downloadable D-Channel Daughterboard
- QPC414 Network Interface card, if the NT8D72BA PRI2 card
- NTRB53 Clock Controller card

For ISDN BRI access:

- 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)
- NTRB53 Clock Controller card

The following specific hardware is required for Small Systems .

For ISDN PRI access:

- NTBK50AA PRI2 card
- NTBK51BA Downloadable D-Channel Daughterboard
- NTAK20 (vintage BB or later) Clock Controller daughterboard

For ISDN BRI access:

- 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 307: Basic Call Gateways on page 656 lists these gateways and the associated support for basic call functionality.

Table 307: 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
MCDN	Basic Voice Call Data Call (64K unrestricted) Calling Line ID transmission
CIS	Basic Voice Call Calling Line ID transmission

Gateway	Basic Call Support
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.

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 Automatic 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) are 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) are 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 phone 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 can 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
- EuroISDN (EURO) package 261

Feature implementation

Task summary list

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

For ISDN PRI access:

1. Table 308: LD 17 - Configure the D-Channel for the network side EuroISDN ETSI protocol. on page 662
2. Table 309: LD 16 - Configure the ISDN BRI link for the network side EuroISDN ETSI protocol. on page 662

For ISDN BRI access:

3. For ISDN BRI access: on page 662 Table 310: LD 16 - For an ISDN or DPNSS1/DASS2 trunk, configure the route for the Bearer Capability-Based Routing call type. on page 663

For ISDN PRI access:

Note:

LD17 for the network side EuroISDN Trunk - Network Side connectivity, configure the D-Channel for the network side EuroISDN ETSI protocol.

Table 308: LD 17 - 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. x = 0-63.
- CTYP	MSDL	The card type to be used is the MSDL. Card type for Option 11, 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:

Table 309: 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, as defined in LD 15.
ROUT		Route number
	0-511	Range for Large System and Avaya Communication Server 1000E (Avaya CS 1000E) system.
	0-127	Range for Media Gateway 1000 B (Avaya MG 1000B).
DTRK	YES	Digital trunk.
...		

Prompt	Response	Description
- IFC	EURO	Interface type. Enter EURO for EuroISDN.
- - CNTY	NET	Country pertaining to EuroISDN interface. Enter NET for network side.

Table 310: 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, as defined in LD 15.
ROUT		Route number
	0-511	Range for Large System and CS 1000E system.
	0-127	Range for Media Gateway 1000B.
TKTP		Supported trunk type.
	CO DID TIE IDA	Central Office trunk. Direct Inward Dial trunk. TIE trunk. Integrated Digital Access trunk.
DTRK	YES	Digital trunk.
- DSEL		Data Selection.
	(VOD) VCE DTA 3VCE 3DTA TDN	Route is for both voice and data Route is voice only Route is data only Route is voice and 3.1 KHz Route is data and 3.1 KHz Transparent Data Network

Feature operation

No specific operating procedures are required to use this feature.

Chapter 40: Idle Extension Notification

Contents

This section contains information on the following topics:

Feature description

The Idle Extension Notification feature provides the attendant with the ability to be notified when a busy extension becomes idle. There are several applications for this feature, but the primary one is for an attendant who wants to leave a person an urgent message, but still does not want to break in on an existing call.

Operating parameters

The following limitations apply for the Idle Extension Notification feature:

- This feature is part of the SACP feature and is dependent on configuring the SACP key on the attendant console.
- The attendant can use Idle Extension Notification for extensions with ordinary HOT/MCN/ MCR/SCN/ SCR DNs. It does not work for ACD DNs or PLDNs.
- Only one Idle Extension Notification at a time can be requested for an extension.

- If a call is already camped on (or SACP camped on) to an extension, no Idle Extension Notification can be requested for that extension at the same time.
- The SACP customer option SACP = ALL or SNGL in the customer data block is not applicable for Idle Extension Notification. The Idle Extension Notification feature is always performed on a call basis.
- The Idle Extension Notification feature can only be activated on a first degree busy extension. Generally if Camp-On is not allowed on an extension Idle Extension Notification is not allowed for that extension.
- Before Idle Extension Notification is requested no indication is given to the attendant of the extension being first or second degree busy. The feature first/second degree busy indication (ABDA) does not give an indication for calls on the source side of the attendant.
- When an Idle Extension Notification request has recalled and is answered by the attendant, a destination DN cannot be dialed until there is an established call on the SRC side.
- No Camp-On/Periodic Camp-On tone is given to an extension that is supervised for Idle Extension Notification even if the customer's OPT is CTA and the extension's Class of Service is WTA.
- The feature Idle Extension Notification is implemented in MCDN (Meridian Customer Defined Network) ISDN networks. Wherever ISDN network environments are mentioned, MCDN ISDN networks are considered. The feature is not supported on DPNSS networks or through MCDN to DPNSS Gateway nodes.
- The Idle Extension Notification feature is not valid for the 2616 phone.
- If in an ISDN network environment there are nodes with the Idle Extension Notification feature active and nodes without the feature active, the feature will not work across the network. However it will work locally in the nodes where it is active. If an attempt to request for Idle Extension Notification on an extension situated in a node that does not have this feature active is made, no response from the node of the extension will be received and the SACP lamp stays dark. The attempt is canceled when the attendant presses the Release/Release SRC key. An error message (ERR5334) is printed in the node of the extension.

Feature interactions

The Idle Extension Notification feature is not supported on ACD DNs.

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 can 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.

Feature packaging

This feature requires the following packages:

- Attendant Break-in/Trunk Offer (BKI) package 127
- International Supplementary Features (SUPP) package 131
- Integrated Services Digital Network (ISDN) package 145
- Advanced ISDN Network Services (NTWK) package 148
- Network Attendant Service (NAS) package 159
- Semi-automatic Camp-on (SACP) package 181

Feature implementation

Task summary list

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

- Table 311: LD 12 - Designate the Semi-automatic Camp-on key. on page 669
- Table 312: LD 15 - Configure a new Incoming Call Indicator (ICI) key for customer data. on page 669
- Table 313: LD 16 - Configure the Idle Extension Notification block timer. on page 670

Table 311: LD 12 - Designate the Semi-automatic Camp-on key.

Prompt	Response	Description
REQ:	CHG	Change existing data.
TYPE:	aaaa	Type of console, where aaaa = 2250 for M2250 console, or PWR if the TN is used for power or Attendant Supervisory Module (ASM).
TN		Terminal number
	l s c u	Format for Large System and Avaya Communication Server 1000E (Avaya CS 1000E) system, where l = loop, s = shelf, c = card, u = unit.
	c u	Format for Media Gateway 1000 B (Avaya MG 1000B), where c = card and u = unit.
CUST	xx	Customer number, as defined in LD 15.
KEY	xx SACP	Key number, Semi-automatic Camp-on/Idle Extension Notification.

Table 312: LD 15 - Configure a new Incoming Call Indicator (ICI) key for customer data.

Prompt	Response	Description
REQ:	NEW	Add new data.
	CHG	Change existing data.
TYPE:	ATT	Attendant Console options.
CUST		Customer number

Prompt	Response	Description
	0-99	Range for Large System and CS 1000E system.
	0-31	Range for Media Gateway 1000B.
ICI	xx IEN	ICI number, Idle Extension Notification.

Table 313: LD 16 - Configure the Idle Extension Notification block timer.

Prompt	Response	Description
REQ	NEW	Add new data.
	CHG	Change existing data.
TYPE	RDB	Route Data Block.
CUST	xx	Customer number, as defined in LD 15.
...		
CNTL	YES	Changes to Controls or Timers.
- 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.

Feature operation

No specific operating procedures are required to use this feature.

Chapter 41: Incoming Trunk Programmable Calling Line Identification

Contents

This section contains information on the following topics:

Feature description

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 phone 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
- 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 phones 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 phone off the Central Office the billing number will be displayed at that phone if the option is set (BDSP = YES).

Call redirection is not supported (for example, if the incoming call terminates on the system and is transferred, or call forwarded to the Central Office, the billing number is not passed on to the Central Office).

Feature interactions

R2MFC CNI/CDR Enhancements

Incoming Trunk Programmable CLID takes precedence over the R2MFC CNI/CDR Enhancements feature. If the outgoing ISDN trunk is allowed to send a billing number, the billing number is sent out as the CLID, not the CNI from the incoming trunk.

Feature packaging

The following package is required for Incoming Trunk Programmable Calling Line Identification:

- Integrated Services Digital Network (ISDN) package 145

For ISDN and networking, the following packages are required:

- Basic Routing (BRTE) package 14
- Network Class of Service (NCOS) package 32
- Basic Alternate Route Selection (BARS) package 57 or Network Alternate Route Selection (NARS) package 58 or Coordinated Dialing Plan (CDP) package 59

Feature implementation

Task summary list

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

1. Table 314: LD 16 - Configure the incoming route. on page 673
2. Table 315: LD 16 - Configure the outgoing route. on page 674

Table 314: LD 16 - Configure the incoming route.

Prompt	Response	Description
REQ	NEW	Add new data.
	CHG	Change existing data.
TYPE	RDB	Route data block.
...		
BILN	(NO) YES	Billing number (is not) is required.
BLEN	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.

Table 315: LD 16 - Configure the outgoing route.

Prompt	Response	Description
REQ	CHG	Change.
TYPE	RDB	Route data block.
...		
ISDN	YES	ISDN route must be YES.
...		
IFC	xxxx	Select the appropriate interface type for route (cannot be SL1 or IDA).
SBN	YES	(Do not) send billing number. Must be set to YES.

Feature operation

No specific operating procedures are required to use this feature.

Chapter 42: ISDN BRI Calling Line Identification Presentation and Restriction

Contents

The following are topics in this section:

Feature description

This feature allows or restricts the display of a Calling Line ID on the called party terminal. Calling Line ID Presentation (CLIP) controls the called party ISDN BRI terminal's choice to display the calling line identification for incoming calls.

Presentation (PRES) controls the calling terminal's choice to send or PRES restricts sending the Calling Line ID when placing an outgoing call. The calling terminal can choose to send or restrict sending the calling line identification on a call-by-call basis. The PRES button on the ISDN BRI terminal can be set to YES to allow sending the calling line identification or set to NO to restrict sending the calling line identification to the called terminal. The PRES button on the terminal overrides the PRES parameter. CLIP and PRES parameters are configured for an ISDN BRI DN when configuring the TSP using LD 27.

The calling party number is contained in the calling party information element in the setup message. The information element is optional and need not be contained in the setup message. If the information element is not presented, the default DN selected in the TSP is used as the calling party DN.

If the call is originated from a non-ISDN BRI to an ISDN BRI terminal, the calling party number is based on the Class of Service of the non-ISDN BRI terminal. The Class of Service can be set for PDN which uses the primary DN of the non-ISDN BRI terminal as the calling party number or LDN which uses the listed DN of the non-ISDN BRI terminal customer as the calling party number.

If the calling party is using an ISDN BRI terminal and has called party number restricted, the called party's display shows the trunk access code rather than the ISDN BRI DN.

Operating parameters

There are no operating parameters associated with this feature.

Feature interactions

There are no feature interactions associated with this feature.

Feature packaging

There is no new feature package introduced with this feature. However, the following feature packages are required for ISDN BRI:

- Basic Rate Interface (BRI) package 216
- Basic Rate Interface line application (235)

Feature implementation

There are no specific implementation procedures for this feature. The PRES parameter CLIP, and PRES parameters are configured for an ISDN BRI DN when configuring the TSP using LD 27.

Feature operation

No specific operating procedures are required to use this feature.

Chapter 43: ISDN BRI circuit-switched data calls

Contents

Feature description

ISDN BRI to ISDN BRI circuit-switched data calls

Circuit-switched data calls between two ISDN BRI terminals are supported for all bearer capability encodings. The system screens for the validity of its bearer capability codepoints only; it does not screen for protocol compatibility between two ISDN BRI devices. The system presents the call to the terminating party; the two ISDN BRI devices determine whether they can communicate with the specified bearer capability. If the ISDN BRI terminals decide that the protocol is incompatible, either terminal can drop the call.

ISDN BRI accessing system data modules

The system offers data access through special data adapters (for example, ASIM). These adapters provide Avaya proprietary DM-DM protocol for data transmission. The ISDN BRI terminal cannot communicate with these adapters because of protocol incompatibility. Calls from an ISDN BRI terminal to these data modules are blocked; similarly, calls from these data modules to an ISDN BRI terminal are also blocked.

ISDN BRI terminals can communicate with these data modules under one condition: ISDN BRI terminals that support the TLINK protocol can communicate with these system data adapters if the circuit-switched data call is placed across an ISDN PRI or DTI link. This access is possible because the ISDN PRI or DTI pack provides protocol conversion from TLINK to DM-DM and vice versa.

ISDN BRI accessing Meridian Communication Adapters

The MCA operates in three modes: DM-DM, TLINK and PSDS. An MCA operating in TLINK mode communicates with ISDN BRI terminals that support TLINK protocol. Calls from an ISDN BRI terminal (with the appropriate bearer capability encodings) to an MCA and vice versa are allowed to terminate; however, if the protocol exchange fails between the ISDN BRI and MCA devices, if the MCA is not operating in TLINK mode, for example, then the call can be dropped by either device.

The following describe the bearer capability encodings used by ISDN BRI terminals that support TLINK protocol. Only calls originating from ISDN BRI with these bearer capability encodings are allowed to terminate to an MCA:

- 64k clear:
  - Octet 3 = unrestricted digital information, octet 4 = 64 kbits/s, no octet 5s are included.
- 64k restricted:
  - Octet 3 = restricted digital information,
  - octet 4 = 64 kbits/s, no octet 5s are included.
- 56k:
  - Octet 3 = restricted or unrestricted digital information,
  - octet 4 = 64kbits/s, octet 5 = v.110 and octet 5a = 56kbits/s.

ISDN BRI accessing ADM trunks

ADM trunk is not a trunk; it groups ADM devices in a route to allow an idle ADM device defined in the given route to be searched or hunted easily when the route access code is dialed. The Meridian Communication Unit (MCU) can be defined as an ADM trunk with the MCU trunk as a subtype. Because MCU can support TLINK protocol, an ISDN BRI terminal accessing this type of ADM route is allowed to terminate to an idle device found in the route and the conditions described for accessing MCA apply here. However, if the ADM route is not an MCU trunk subtype, then ISDN BRI is blocked from accessing the ADM routes because of incompatible protocols.

ISDN BRI terminals can access ADM trunks that are not MCU subtypes under one condition: ISDN BRI terminals that support TLINK protocol can access an ADM trunk if the circuit-switched data call is placed across an ISDN PRI or DTI link. This access is possible because the ISDN PRI or DTI pack provides protocol conversion from the TLINK to the DM-DM and vice versa.

ISDN BRI accessing ISDN PRI

The ISDN PRI interface supports three bearer capability encodings:

- 64k clear:
  - Octet 3 = unrestricted digital information, octet 4 = 64 kbits/s, no octet 5s are included.
- 64k restricted:
  - Octet 3 = restricted digital information, octet 4 = 64 kbits/s, no octet 5s are included.
- 56k:
  - Octet 3 = restricted or unrestricted digital information, octet 4 = 64kbits/s, octet 5 = v.110 and octet 5a = 56kbits/s.

ISDN BRI originated circuit-switched data calls are allowed access to the ISDN PRI trunk if these encodings are used.

ISDN BRI devices using other bearer capability encodings can communicate with another ISDN BRI device across ISDN PRI under these conditions:

- the ISDN PRI interface is the system, Japan D70, or 4ESS/5ESS
- the remote capability for ISDN BRI interworking is turned on (in LD 17) for all ISDN PRI interfaces involved in the call.

ISDN BRI accessing DTI trunks

ISDN BRI circuit-switched data call can access a DTI trunk only if the bearer capability is:

- 56k:
  - octet 3 = restriction or unrestricted digital information, octet 4 = 64 kbits/s, octet 5 = v.110 and octet 5a = 56 kbits/s

ISDN BRI terminals using V.120 protocol at 56 kbps are not supported over DTI trunks.

ISDN BRI circuit-switched data call tandem across ISDN PRI

In the ISDN PRI and DTI pack, a protocol convertor is inserted by default to convert DM-DM protocol to TLINK at the tandem node and vice versa. Because of this, an ISDN BRI circuit-switched data call tandem across ISDN PRI, DTI trunks or a combination of ISDN PRI and DTI trunks is supported only for ISDN BRI terminals that use TLINK protocol. ISDN BRI terminals using other protocols, such as V.110 and V.120, require the Transparent Data Networking feature to provide transparent data channel through the tandem nodes.

Public Switched Data Service

Public switched data service provides a pure 56 kbps data transmission. The data module on both ends must establish identical parameters manually before the data call is made.

ISDN BRI terminals can access Public Switched Data Service provided a dedicated data route is used; when a voice/data shared route is used, the ISDN BRI terminal must generate a burst of tone to the network provider to turn off the echo cancellation.

ISDN BRI accessing analog line by means of a modem trunk

ISDN BRI circuit-switched data calls accessing analog lines through modem trunks are supported only if the data module connected to the modem uses TLINK, V.110 or V.120 protocol.

An ISDN BRI device with the TLINK protocol can access analog lines through modem trunk configuration using a DTE type MCU, because the MCU supports TLINK protocol. ISDN BRI devices cannot communicate with other system DTE type data modules that connect to modems because they use the DM-DM protocol.

ISDN BRI terminals can access analog lines through modem trunk configurations using DM-DM type data modules under one condition: ISDN BRI terminals that support TLINK protocol can access these modem trunks if the circuit-switched data call is placed across an ISDN PRI or DTI link. This access is possible because the ISDN PRI or DTI pack provides protocol conversion from TLINK to DM-DM and vice versa.

Operating parameters

There are no operating parameters associated with this feature.

Feature interactions

There are no feature interactions associated with this feature.

Feature packaging

There is no new feature package introduced by this feature. However, the following feature packages are required for ISDN BRI:

- Basic Rate Interface (BRI) package 216
- Basic Rate Interface line application package 235

Feature implementation

There are no specific implementation procedures for this feature.

Feature operation

No specific operating procedures are required to use this feature.

Chapter 44: ISDN BRI ETSI Call Forwarding Unconditional

Contents

This section contains information on the following topics:

Feature description

The European Telecommunication Standards Institute (ETSI) Call Forwarding Unconditional (CFU) supplementary service allows an incoming call to an ISDN BRI terminal to be forwarded to a predetermined destination, within or outside the system. The call is forwarded regardless of whether the user is busy or idle.

Voice and data basic services can be individually forwarded to the same DN. Also, calls can be forwarded to an ISDN BRI terminal or a non-ISDN BRI terminal. When the CFU feature is activated, outgoing calls can still be made from the ISDN BRI terminal.

The Call Forwarding capabilities can be provided to all users on the access (i.e., all the DNs defined under a DSL), or an individual user (i.e., a DN).

The following functionalities are currently supported:

- When a call is forwarded, if the caller is using an ISDN BRI set they will be notified by the system that the call has been forwarded. In addition, the caller is provided with the forwarded-to number.
- When a call is forwarded from an ISDN BRI terminal that has CFU activated, the terminating terminal will be notified by the system that the ISDN BRI terminal forwarded the call.

Operating parameters

The ETSI CFU supplementary service is based on the system Call Forwarding All Calls feature. Therefore, system Call Forwarding All Calls feature limitations also apply to this feature.

ETSI CFU basic service is limited to speech, 3.1 kHz audio, unrestricted digital information and restricted digital information.

Interrogation of all served users in TSP 0 will be supported. However, if the number of DNs is too large to fit into one message, interrogation of all served users in TSP 0 will not be supported.

ETSI public ISDN partial rerouting will not be supported. According to ETSI standards, when the system receives a local exchange call and the call is forwarded within or outside of the system, the system should send a message to the local exchange to make the call directly to the forwarded-to user.

During a data dump, the forwarded-to user's address will not be saved because of protected data storage space. In LD 17, the CFWS prompt, which enables users to save Call Forwarding feature activation during data dump so that during SYSLOAD the feature will be activated, will be ignored for users of ISDN BRI.

ISDN BRI terminals that support ETSI Call Forward Unconditional supplementary service are required.

Feature interactions

Call Waiting

User requesting all calls to be forwarded - ETSI CFU takes precedence over Call Waiting.

Forward-to user - A forwarded call can invoke Call Waiting.

Conference Call, Add-on

Calling user - If a conference controller calls a person who has CFU activated to establish a conference, the forwarded-to user will be alerted and added to the conference call after the call is answered.

Forwarded-to user - A call that has been forwarded to the conference controller can be added to an existing conference. A forwarded-to user can establish a conference using the existing forwarded call provided that the call is in the active state.

Call Forward All Calls

The system Call Forward All Calls feature allows only calls to the Prime DN or a single appearance DN to be forwarded. For an ISDN BRI set, calls are forwarded based on this basic service. Since ETSI CFU supplementary service is developed based on the Call Forward All Calls feature, all the existing Feature interactions of Call Forward All Calls apply to ISDN BRI terminals.

BRI Special Call Forward Busy

Call Forward No Answer

The ETSI CFU supplementary service takes precedence over BRI Call Forward Busy, and Call Forward No Answer.

Call Forward and Busy Status

Call Forward and Busy Status allows a user to forward calls and monitor the Call Forward and Busy Status from the forwarded-to user. This requires a Busy/Forward Status key. An ISDN BRI terminal cannot monitor other set's Call Forward Busy Status and in turn its Call Forward Busy Status cannot be monitored by other sets.

Remote Call Forward

Remote Call Forward allows a user to program Call Forwarding from a remote set. Call Forwarding remotely to an ISDN BRI terminal is supported. However, using Remote Call Forward from an ISDN BRI terminal is not supported.

Attendant and Network-wide Remote Call Forward

This feature allows the configuration of Call Forward from an Attendant Console and a remote set across a Meridian Customer Defined Network (MCDN). For an ISDN BRI terminal, Remote Call Forward is allowed from a set, or from an Attendant Console, but not from a BRI set either locally or network wide. When Call Forward is activated, it is assumed that the Call Forward for the BRI set is voice, not data.

Call Forward/Hunt Override via Flexible Feature Code

This feature allows all attendants and sets with Call Forward/Hunt Override Allowed (CFHA) Class of Service to override Call Forward All Calls (CFAC). Since an ISDN BRI terminal cannot access Flexible Feature Codes, it cannot override CFAC on other sets. However, an ISDN BRI set's Call Forward Unconditional can be overridden by a set having CFHA Class of Service or by an attendant in both standalone and network environments.

Internal Call Forward

An ISDN BRI terminal cannot activate the internal Call Forward feature.

Call Forward Reminder Tone

The Call Forward Reminder Tone is supported on ISDN BRI sets that have the Call Forward Unconditional feature active.

Call Forward Confirmation Tone

During activation or deactivation of Call Forward from a BRI terminal, a Call Forward confirmation tone is not provided to the BRI terminal. Instead, confirmation is done by sending a BRI message to the terminal.

Feature packaging

There is no new feature package introduced by this feature. However, the following feature packages are required for ISDN BRI:

- Meridian 1 Extended Peripheral Equipment (XPE) package 203
- Basic Rate Interface (BRI) package 216
- Basic Rate Interface line application package 235

Feature implementation

Task summary list

The following is a summary of the tasks to configure ISDN BRI Call Forwarding Unconditional:

1. Table 316: LD 27 - Configure ETSI protocol for a DSL. on page 689
2. Table 317: LD 27 - Configure an ISDN BRI terminal. on page 689

Table 316: LD 27 - Configure ETSI protocol for a DSL.

Prompt	Response	Description
REQ	NEW	Add new data.
TYPE	DSL	Digital subscriber loop.
DSL	lll s c dsl#	DSL address for system.
PRID	2	Protocol Identification. Enter 2 for ETSI.

Table 317: LD 27 - Configure an ISDN BRI terminal.

Prompt	Response	Description
REQ	NEW	Add new data.
TYPE	TSP	Terminal Service Profile.
DSL	lll s c dsl#	DSL address for system (as already entered in step 1).
USID	(0)-15	User service identifier. 0 = default TSP.

Prompt	Response	Description
MPHC	NO	This TSP is not configured for a Meridian 1 Packet Handler.
SPID	aaa...a <cr> Xaaa...a	Service profile ID aaa...a = any combination of 1-20 alphanumeric characters. <cr> = Stops this prompt from being displayed again. A maximum of 8 valid SPIDs per TSP are allowed. Xaaa...a removes the specified SPID. This prompt appears only if USID = 1-15. It repeats until <cr> is entered, but only up to 8 SPIDs can be entered. This SPID must be entered in the initializing terminal to associate the terminal with a USID.
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).
CT	VCE DTA	Call type (enter VCE for voice or DTA for data).
...		
FEAT	CFXA	Allow Call Forward External for the DN.
SSRV_ ETSI	VCFW DCFW	For voice call type, enter VCFW for Voice Call Forward. For data call type, enter DCFW for Data Call Forward.
...		

Feature operation

No specific operating procedures are required to use this feature.

Chapter 45: ISDN BRI ETSI Conference

Contents

This section contains information on the following topics:

Feature description

The ISDN BRI Supplementary Services European Telecommunication Standards Institute (ETSI) Conference feature provides a subset of the Conference capabilities to ISDN BRI users connected to a system using ETSI protocols.

The following capabilities are supported:

- Conference invocation (beginning the conference from the active call state)
- Adding a conferee
- Disconnection by a conferee
- Termination of the conference (this subfeature will not be implemented as ETSI specifies. ETSI specifies that when a conference controller disconnects, all conferees will be disconnected; however, this subfeature follows the current system Conference feature in that when a conference controller disconnects the conference continues for the remaining parties).

An ISDN BRI user can conference a maximum of three or six parties depending on the Terminal Service Profile (TSP) configuration administered through LD 27. The ETSI Conference feature can be invoked by a user while the user is on an established call. Once the conference is invoked, the user can add another party (active or held) to the conference call.

Operating parameters

ISDN BRI terminals must support both ETSI basic call and ETSI Conference Supplementary service signaling protocols.

ETSI Conference only supports speech or 3.1 kHz; other bearer services such as 7 kHz, unrestricted digital information, and restricted digital information are not supported.

In a conference call, a maximum of two parties (including conference controller and conferees) can be from the same Digital Subscriber Loop (DSL), due to the availability of only two B-channels per ISDN BRI interface.

The system only sends notification messages to conferees or the conference controller when conference operations are requested and performed if they are ISDN BRI users. If a user is on another node across ISDN or non-ISDN trunks, the Notify message will not be sent.

Feature interactions

Conference

ISDN BRI terminals can conference in both ISDN BRI and non-ISDN BRI terminal users; similarly, an ISDN BRI terminal can also be conferenced into a call by a non-ISDN terminal. However, in either case, the ISDN BRI terminal display will not be updated due to lack of protocol to support it.

ISDN PRI Network Call Redirection

When an ISDN BRI conference call is dropped to a simple call, if a party in the simple call is an ISDN BRI user, the corresponding BRI name and number will be updated on the other party's non-ISDN set display. However, the ISDN BRI display will not be updated due to lack of standard protocol to support this function.

Held Call Clearing

Held Call Clearing allows a set to clear both active calls and held calls by going on-hook. This will not apply to an ISDN BRI set.

ISDN BRI Trunk Access

ISDN BRI ETSI Conference is supported across ISDN BRI trunks.

Feature packaging

There is no new feature package introduced by this feature. However, the following feature packages are required for ISDN BRI:

- Meridian 1 Extended Peripheral Equipment (XPE) package 203
- Basic Rate Interface (BRI) package 216
- Basic Rate Interface line application package 235

Feature implementation

Task summary list

The following is a summary of the tasks to configure ISDN BRI ETSI Conference:

1. Table 318: LD 27 - Configure ETSI protocol for a DSL. on page 693
2. Table 319: LD 27 - Configure Conference in the TSP. on page 694

Table 318: LD 27 - Configure ETSI protocol for a DSL.

Prompt	Response	Description
REQ	NEW CHG	Add new data. Change existing data.
TYPE	DSL	Digital Subscriber Loop.
DSL	lll s c dsl#	DSL address.
PRID	2	Protocol Identification. Enter 2 for ETSI.
	(WTA) WTD	Enter WTD for Warning Tone Denied (Warning Tone Allowed has to be overridden).

Table 319: LD 27 - Configure Conference in the TSP.

Prompt	Response	Description
REQ	NEW CHG	Add new data. Change existing data.
TYPE	TSP	Terminal Service Profile.
DSL	lll s c dsl#	DSL address (as already entered in step 1).
SUPL_ SVC	AO6 AO3	AO6 = six-party conference (up to 30 parties can be conferenced, but it is not recommended). AO3 = three-party conference.

Feature operation

No specific operating procedures are required to use this feature.

Chapter 46: ISDN BRI National ISDN-1 Call Forward All Calls

Contents

This section contains information on the following topics:

Feature description

The ISDN BRI National ISDN-1 Call Forward All Calls feature enables a user to have calls redirected from the user's directory number to another directory number. Calls are redirected regardless of the busy or idle status of the interface to the user. Call Forward is assigned on the basis of the directory number and call type (i.e., the user can have voice calls forwarded, while data calls terminate normally).

This feature is invoked using the National ISDN-1 Feature Key Management interface. When Call Forward is activated the Feature Activation Information Element (IE) is sent from the base DN to the system for either feature activation or deactivation.

The Feature Activation IE consists of the feature identifier which represents the combination of the Call Forward feature, the DN, and the call type. There are two methods of feature activation: Call associated; and Non-call associated. In both cases the Feature Activation Information Element is used to activate the feature for the DN and Call Type.

Operating parameters

This feature is based on the system Call Forward All Calls feature; therefore, it is subject to the same limitation as the system Call Forward All Calls feature.

ISDN BRI terminals must support the Feature Key Management protocol.

When a DN from a Terminal Service Profile (TSP) activates or deactivates the Call Forward feature, it applies to all appearances of the same DN from different TSPs on the same Digital Subscriber Loop (DSL). For example, if a boss and a secretary each has their own BRI terminal and the secretary has their boss' DN on their set, when the boss activates Call Forward the call will not be terminated on the secretary's set. If the secretary deactivates the feature, the call will be terminated on the boss' and the secretary's sets.

Data dump does not store the Call Forward numbers and Call Forward status for BRI terminals. Therefore, after SYSLOAD the Call Forward feature is no longer activated and the Call Forward numbers are not saved in the system.

There is no confirmation tone provided from the system when the Call Forward feature is invoked from an ISDN BRI terminal.

Only one BRI terminal from a Digital Subscriber Loop (DSL) is allowed to activate or deactivate the Call Forwarding feature at a time. If two BRI terminals try to activate or deactivate at the same time only the first action is allowed. The second terminal is denied the service.

Feature interactions

Call Forward All Calls

The system Call Forward All Calls only allows calls to a prime DN or single appearance DN to be forwarded. For the BRI interface, the terminology of prime DN and single appearance DN does not apply. When Call Forward has been activated for a DN and Call Type, calls to the DN/Call Type are forwarded regardless of which Terminal Service Profile (TSP) that the DN/Call Type is assigned. In addition, redirecting information about the BRI terminal is delivered to the remote DN. If the remote DN is an ISDN BRI terminal, the redirecting information is passed in the Redirecting Number IE of the SETUP message to the BRI terminal.

When an ISDN BRI terminal has activated the Call Forward feature, it is treated as if it is a set from the system activating the feature. The system features that are normally applicable to the set will also be applicable to the BRI terminal.

Call Forward No Answer

Busy

As in the case of Call Forward All Calls, the National ISDN 1 Basic Rate Interface (BRI) Call Forward All Calls feature takes precedence over Call Forward No Answer, and Call Forward Busy.

Network Call Redirection

When a call to the BRI interface is forwarded, the redirecting DN and the reason for call redirection are passed to the remote user if the remote user is connected by ISDN. The redirecting information is encoded in the ISDN SETUP message for display purposes. In addition, Call Forward to another node from an ISDN BRI terminal is counted as one for the ISDN network Call Redirection counter which is defined in the customer data block.

Call Forward and Busy Status

This feature is not supported for ISDN BRI terminals.

Remote Call Forward

Remote Call Forward allows a user to program Call Forwarding from a remote set. Call Forwarding remotely to an ISDN BRI terminal is supported. However, using Remote Call Forward from an ISDN BRI terminal is not supported.

Attendant and Network-wide Remote Call Forward

This feature allows the configuration of Call Forward from an Attendant Console and a remote set across a Meridian Customer Defined Network (MCDN). For an ISDN BRI terminal, Remote Call Forward is allowed from a set, or from an Attendant Console, but not from a BRI set either locally or network wide. When Call Forward is activated, it is assumed that the Call Forward for the BRI set is voice, not data.

Internal Call Forward

An ISDN BRI terminal cannot activate the internal Call Forward feature.

Call Forward/Hunt Override via Flexible Feature Code

This feature allows all attendants and sets with Call Forward/Hunt Override Allowed (CFHA) Class of Service to override Call Forward All Calls (CFAC). Since an ISDN BRI terminal cannot access Flexible Feature Codes, it cannot override CFAC on other sets. However, an ISDN BRI set's Call Forward All Calls can be overridden by a set having CFHA Class of Service or by an attendant in both standalone and network environments.

Call Forward Reminder Tone

The Call Forward Reminder Tone is supported on ISDN BRI sets that have the Call Forward All Calls feature active.

Call Forward Confirmation Tone

During activation or deactivation of Call Forward from a BRI terminal, a Call Forward confirmation tone is not provided to the BRI terminal. Instead, confirmation is done by sending an ISDN BRI message to the terminal.

Call Forward Save on SYSLOAD

Call Forward Save on SYSLOAD is not supported for ISDN BRI terminals.

Feature packaging

There is no new feature package introduced by this feature. However, the following feature packages are required for ISDN BRI:

- Meridian 1 Extended Peripheral Equipment (XPE) package 203
- Basic Rate Interface (BRI) package 216
- Basic Rate Interface line application (235)

Feature implementation

Task summary list

The following is a summary of the tasks to configure ISDN BRI National ISDN-1 Call Forward All Calls:

1. Table 320: LD 27 - Configure National ISDN-1 protocol for a DSL. on page 699
2. Table 321: LD 27 - Configure Call Forward All Calls in the TSP. on page 699

Table 320: LD 27 - Configure National ISDN-1 protocol for a DSL.

Prompt	Response	Description
REQ	NEW	Add new data.
TYPE	DSL	Digital Subscriber Loop.
DSL	lll s c dsl#	DSL address.
PGPN	1	Protocol set group number.
PRID	6	Protocol identification. Enter 6 for National ISDN-1.

Table 321: LD 27 - Configure Call Forward All Calls in the TSP.

Prompt	Response	Description
REQ	NEW CHG	Add new data. Change existing data.
TYPE	TSP	Administer the Terminal Service Profile on the DSL.
DSL	lll s c dsl#	DSL address (as already entered in step 1).
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).
FEAT	CFXA	Allow Call Forward External for the DN.
SSRV_NI	VCFA 15 16	Feature identifiers for Call Forward voice activation.
	VCFD 15 16	Feature identifiers for Call Forward voice deactivation.
	DCFA 16 26	Feature identifiers for Call Forward data activation.
	DCFD 17 27	Feature identifiers for Call Forward data deactivation.

Feature operation

No specific operating procedures are required to use this feature.

Chapter 47: ISDN BRI National ISDN-1 Conference

Contents

This section contains information on the following topics:

Feature description

This version of ISDN BRI Conference is based on National ISDN-1.

Terminals which are supported are:

- M5317TDX (version 2.3a and above)
- M5209TDcp (version 2.28 and above), and
- third-party vendor manufactured terminals which are deemed compatible

ISDN BRI terminals can conference in both ISDN BRI and non-ISDN BRI terminal users. An ISDN BRI terminal can also be conferenced into a call by a non-ISDN BRI terminal.

System ISDN BRI supports two versions of Conference: A03, a 3-party Conference and A06, a 6-party Conference.

ISDN BRI Conference operates under the following conditions:

- the user employs the National ISDN-1 protocol (PRID=6) on the DSL
- the user is subscribed to Conference in its Terminal Service Profile.

- the Feature Activation Identifier and the Feature Indication Identifier are configured both in LD 27 and on the ISDN BRI terminal.
- the logical terminal is the controller of only one conference call.
- the bearer capability of the call associated with the conference request is speech or 3.1 kHz audio.

Invoke Conference while making a call

The ISDN BRI user can make a conference request by means of a pre-defined softkey or programmable key while dialing the outgoing digits. The provision of this capability is dependent on the implementation of the ISDN BRI terminal.

Invoke Conference during an established call

An ISDN BRI user can invoke Conference after the network successfully translates the dialed digits, or while the called party is being rung, or while the user is on an answered call.

Add a conferee

The ISDN BRI user can add a conferee with Consultation Hold, add a held call to an active conference, or add an active call to a held conference providing:

- the Conference Controller (an ISDN BRI user who is subscribed to Conference) is connected to only one active speech or 3.1 kHz audio call
- both the active call and the held call have been answered
- neither the active call nor the held call is undergoing call clearing
- the controller's conference size is not exceeded
- the B-channel to which the Conference Controller is connected can be used to complete the conference
- both calls are not conference calls

Conference disconnect

If the controller requests a disconnect signal, the network disconnects the entire conference if:

- only one party will remain after the controller disconnects
- none of the calls that will remain has been answered

- only two parties remain and both are outgoing trunk calls
- more than two parties remain and all are trunk calls

Otherwise, only the controller is disconnected and the rest of the conferees remain.

Call Transfer during Conference

Using the Conference feature, a Call Transfer can be achieved in the following way:

Ann makes a call to Bob. Bob answers the call and conferences Carl. After Carl answers the call, Bob completes the conference. Then Bob can disconnect himself from the conference, thus transferring Ann to Carl.

The same conditions that determine whether the remaining parties in a conference should be dropped when one party disconnects from the conference apply here. For example, Bob makes a call to Ann. Then Bob conferences in Carl. If both Ann and Carl are on a remote node (that is, both calls made by Bob to Ann and Carl are outgoing trunk calls), then as Bob disconnects from the conference, both Ann and Carl will be dropped as well. Call Transfer cannot be achieved in this scenario.

Operating parameters

There are no operating parameters associated with this feature.

Feature interactions

There are no feature interactions associated with this feature.

Feature packaging

There is no new feature package introduced by this feature. However, the following feature packages are required for ISDN BRI:

- Basic Rate Interface (BRI) package 216
- Basic Rate Interface line application package 235

Feature implementation

Task summary list

The following is a summary of the tasks to configure ISDN BRI National ISDN-1 Conference:

- Table 322: LD 27 - Configure National ISDN-1 for a DSL. on page 704
- Table 323: LD 27 - Configure Conference in the TSP. on page 704

Table 322: LD 27 - Configure National ISDN-1 for a DSL.

Prompt	Response	Description
REQ	NEW CHG	Add new data. Change existing data.
TYPE	DSL	Digital Subscriber Loop.
DSL	lll s c dsl#	DSL address.
PRID	6	Protocol Identification. Enter 6 for NI-1.
FDN	nnnnnnn	Flexible Call Forward No Answer DN.

Table 323: LD 27 - Configure Conference in the TSP.

Prompt	Response	Description
REQ	NEW CHG	Add new data. Change existing data.
TYPE	TSP	Terminal Service Profile.
DSL	lll s c dsl#	DSL address (as already entered in step 1).
FEATID	aaa mmm nnn Xmmm <cr>	Feature ID (must be unique for a TSP), where: aaa = feature; enter AO6 for six-party conference (up to 30 parties can be conferenced, but it is not recommended), or AO3 for three-party conference. mmm = feature activation ID (1-127). nnn = feature indication ID (1-127). Xmmm = delete feature. <cr> = skip feature ID definition (aaa mmm nnn is re-prompted until <cr> is entered.

Feature operation

No specific operating procedures are required to use this feature.

Chapter 48: ISDN BRI Special Call Forward Busy

Contents

This section contains information on the following topics:

Feature description

This feature is activated for a call terminated at a DSL. The call is forwarded to the attendant when a call encounters the following busy conditions:

- the maximum number of calls on a DSL is exceeded, and
- an ISDN BRI DN has Call Forward busy enabled, as configured in the TSP using LD 27
- ISDN Hunting is not allowed or the call fails to find an idle hunt DN.

The calling party receives a busy tone in all other cases.

With Call Forward Busy, Special Hunting, if both B-channels are defined and there is not a second DN, the result is Busy.

Operating parameters

There are no operating parameters associated with this feature.

Feature interactions

There are no feature interactions associated with this feature.

Feature packaging

There is no new feature package introduced by this feature. However, the following feature packages are required for ISDN BRI:

- Basic Rate Interface (BRI) package 216
- Basic Rate Interface line application (235)

Feature implementation

There are no specific implementation procedures for this feature.

Feature operation

No specific operating procedures are required to use this feature.

Chapter 49: ISDN BRI Special Hunting

Contents

This section contains information on the following topics:

Feature description

This feature is activated when a call terminating at a DSL encounters the following busy conditions:

- the maximum number of calls on a DSL is exceeded, or the total number of calls including active, on hold, waiting, and in progress exceeds the number of B-channels provided for the incoming call type

Hunting routes the call through predetermined steps until an idle DN is found for that call. Internal and external hunt DNs are configured in the DSL and Hunting Allowed and Call Forward by Call type are configured in the TSP using LD 27.

If Hunting is not allowed or the call fails to find an idle DN, the following occurs:

- a busy tone is given if the number of calls exceeds the maximum calls specified for that DSL and call forward busy does not succeed.

Note:

The call is forwarded only if it is a DID call

- the call is presented as Call Waiting if the maximum number of calls for that DSL is not exceeded.

An ISDN BRI terminal originating the call is not updated to show on its display that the call was redirected to a different DN as a result of Hunting. This is due to a lack of standard for layer 3 messages used to update the terminal display.

Operating parameters

There are no operating parameters associated with this feature.

Feature interactions

There are no feature interactions associated with this feature.

Feature packaging

There is no new feature package introduced by this feature. However, the following feature packages are required for ISDN BRI:

- Basic Rate Interface (BRI) package 216
- Basic Rate Interface line application package 235

Feature implementation

There are no specific implementation procedures for this feature.

Feature operation

No specific operating procedures are required to use this feature.

Chapter 50: ISDN BRI Connected Line Presentation/Restriction

Contents

This section contains information on the following topics:

Feature description

ISDN Basic Rate Interface Connected Line Presentation/Restriction is a supplementary service that enables the system to either allow or restrict the presentation of a connected party's ISDN number and sub-address on the display of an ISDN BRI terminal. The presentation of an ISDN number and sub-address to the calling party occurs when the connected party's ISDN BRI terminal answers the call. In addition to the BRI terminal, this service can be applied to Meridian 1 proprietary and analog (500/2500-type) sets.

The Connected Line Presentation (COLP) service applies to the calling party. COLP allows the presentation of the connected party's ISDN number and sub-address to the calling party.

The Connected Line Restriction (COLR) service applies to the connected party. COLR restricts the presentation of the connected party's ISDN number and sub-address to the calling party. COLR is activated on the ISDN BRI terminal of the connected party.

The connected party's identification is only provided to the calling party, if the ISDN BRI terminal of the connected party supports this feature.

When COLP/COLR feature is activated, the system monitors the configuration of the terminal service profiles on the calling and connected parties ISDN BRI terminals. Depending on this configuration, the system determines whether or not information is allowed or restricted to the calling party.

Operating parameters

Through configuration, the system BRI terminal interface controls the provisioning of the connected party's information on the interface (COLP prompt). This avoids sending COLP when the terminal does not support this supplementary information.

With the presentation restricted option activated, some BRI terminals might not have the capability to restrict the presentation of digits from the connected party. The system BRI terminal interface has the capability to restrict sending the connected party information the calling party if COLR is activated (COLP prompt).

ISDN BRI COLP/COLR coding is based on ETSI protocol. COLP/COLR from or to the public network is available to all countries, using the EuroISDN DSS1 protocol on the BRI interface.

The construction of the ISDN BRI connected number follows the same rules that apply to Calling Line Identification when sending an extension number over the same interface. National and local prefixes can be added. The type of numbering plan can be modified the same way they are modified for other types of terminals.

Feature interactions

Digital Terminal Display

When a calling party's digital terminal with display receives the connected party's information element, the display is updated with the received number depending on the presentation status. The presentation status applies to calls originating from non-ISDN extensions.

ISDN BRI terminals

Depending on the configuration of the Terminal Service Profiles, there is or is not full transparency for connected number and connected sub-address exchange between ISDN BRI terminals.

ISDN Central Office Trunks

When a call from an ISDN Central Office (CO) is answered by BRI terminal, the ISDN BRI terminal's connected number is sent over the ISDN CO trunk interface, provided that COLP/COLR service is supported. The existing rules to generate the connected number on the ISDN CO interfaces are not modified by COLP/COLR. With the exception of EuroISDN trunks, the connected number is passed only if a redirection occurs.

ISDN Private Trunks

When a call from a QSIG interface is answered by BRI terminal, the ISDN BRI terminal's connected number is sent over the QSIG interface. This connected message is given, even though no prior redirection occurred.

When a call from a Meridian Customer Defined Network (MCDN) interface is answered by BRI terminal, the ISDN BRI terminal's connected number is sent over the MCDN interface. A connect message is sent no matter how the D-channel is configured and only after a redirection has occurred.

ISDN Calling Line Identification Enhancement

The ISDN Calling Line Identification Enhancement allows the choice of National and Local prefixes. This is applicable to connected numbers received from BRI terminal and sent over trunk.

EuroISDN Continuation

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 BRI terminal and sent over trunk.

DPNSS/DASS2

The mapping of the connected number information element in a connect message from or to BRI terminal and the connected line identification in a CRM message from or to a DPNSS or DASS2 interface is supported.

Feature packaging

There is no new feature package introduced by this feature. However, the following feature packages are required for ISDN BRI:

- Extended Peripheral Equipment (XPE) package 203
- Basic Rate Interface (BRI) package 216
- Basic Rate Interface line application package 235

Feature implementation

Task summary list

The following is a summary of the tasks to configure ISDN BRI Connected Line Presentation/ Restriction:

1. Table 324: LD 27 - Configure COLP/COLR for ISDN BRI terminals. on page 714
2. Table 325: LD 27 - Configure Connected Line Presentation/Restriction for a DSL. on page 715

Table 324: LD 27 - Configure COLP/COLR for ISDN BRI terminals.

Prompt	Response	Description
REQ	CHG	Change.
TYPE	TSP	Terminal Service Profile Data.
...		
DN	xxx...x	Directory Number (1 to 7 digits) of ISDN BRI terminal.
- CLIP	(YES) NO	Calling Line Identification Presentation for Incoming Calls.
- PRES	(YES) NO	Presentation of Calling Line Identification to far end on outgoing calls (allowed) restricted.
- COLP	(NO) YES	YES = Connected, number is passed from the system to ISDN BRI terminal. NO = Connected, number is not passed from the system to ISDN BRI terminal.

Prompt	Response	Description
- TRANS	(NO) YES	YES = Connected, number is transmitted to the ISDN BRI terminal as received with no modification. NO = Connected, number is transmitted to the BRI terminal after erasing digits within information element. TRANS is only applicable if the connected number received has presentation restricted.

Table 325: LD 27 - Configure Connected Line Presentation/Restriction for a DSL.

Prompt	Response	Description
REQ	CHG	Create new data block.
TYPE	DSL	Digital Subscriber Loop.
DSL	lll s c dsl#	DSL address.
USID	x	User Service Identifier.
PRID	2	Protocol Identification. Where: 2 = ETSI is only supported.

Feature operation

No specific operating procedures are required to use this feature.

Chapter 51: ISDN Calling Line Identification

Contents

This section contains information on the following topics:

Feature description

The Calling Line Identification (CLID) feature provides the flexibility to build and send CLID on ISDN public and private interfaces based on various numbering plans.

CLID is 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 be programmed against any DN or DN key for each phone. This means that the CLID that is sent from a phone is predicated on what is in the CLID table.

A CLID for any key is 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 phone can have DN keys that each has its own CLID. Or, as explained in How CLID entries are assigned to a phone on page 719, the CLID of any one key could, on a phone, be programmed to use the CLID of any other key on the phone.

Also, the system can 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 phone on one system can send a CLID that will have calls returned to another system. 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

LD 15 is used to 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 system 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)
- a one-seven digit Local Steering Code (LSC)

International Numbering Plan

For users of an International Numbering Plan, the system 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 contains 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)

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 on page 721" section starting on Examples of CLID generation on page 721.

How CLID entries are assigned to a phone

Once the CLID table has been built for a customer, any CLID entry can be assigned to any phone, for each DN or DN key (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 phone, DN key 0 can send one CLID (the information contained in one CLID entry), DN key 1 can send a different CLID (the information contained in another CLID entry), and DN key 2 can send yet another CLID (the information contained in a third CLID entry). A customer can now send different CLIDs from the same phone.

Digital phones

When configuring the CLID for a DN key in LD 11, the value entered against the KEY prompt can 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 system initiates a search on the phone 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 721.

Information display on phones and terminals

The ISDN features listed in this section can be used on all Avaya phones. However, not all phones or terminals provide information displays.

Displayphone - The Meridian 1 Displayphone allows the full display of CLID information.

Other phones and terminals - The following devices, if equipped with a digit display, can receive and display a CLID name:

- Attendant Consoles
- Meridian 1 phones with digit display
- Digital phones (M2317)
- ASCII terminals with an Add-on Data Module (ADM)
- M2008, M2016S, M2216ACD, M2616 when equipped with displays
- Virtual phones that have PCA feature

Digit display format

As shown in Figure 78: Digit display format for Calling Line Identification on page 721, the digit display at the phone receiving the call will display:

- If the CLID display is unavailable because the call was not routed on ISDN routes for the entire call, the trunk route access code and trunk route member number are displayed.
- For public networks, CLID displays the standard North American Numbering Plan 7- or 10-digit number, depending on the number dialed.
- For a private network over ESN, the CLID displays an "H" followed by xxxx yyyyy, where xxxx = a four digit Home Location Code (HLOC) and yyyy = a four digit DN.
- For a private network over CDP, the CLID displays a four digit LSC (Local Steering Code) followed by one of the following:
    - the extension's trailing digits (forming the CDP DN) when CDP is equipped
    - the calling phone's extension

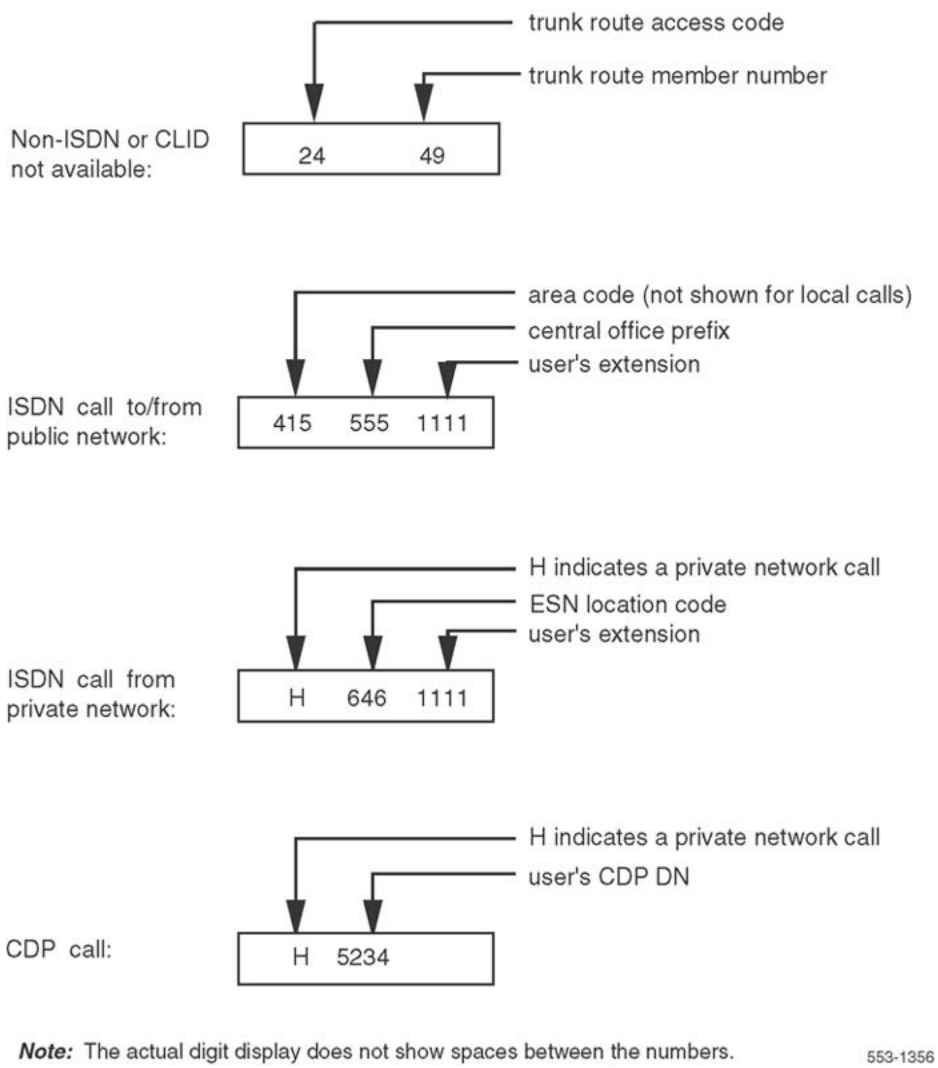


Figure 78: Digit display format for Calling Line Identification

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

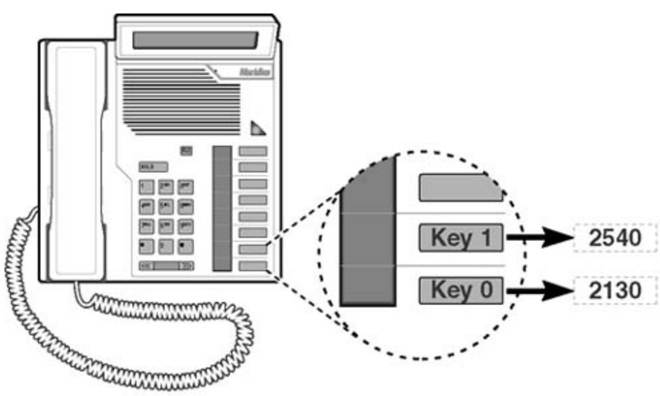
Please note that for calls made using the International Numbering Plan, the CLID for a Public Number is a National Number. For the examples of using the International Numbering Plan, the construction of a Public Local Number is for reference purposes only.

Example 1 - Phone 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 326: 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



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1. For a private ESN UDP call made from Key 0 (DN 2130) on the phone, 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.

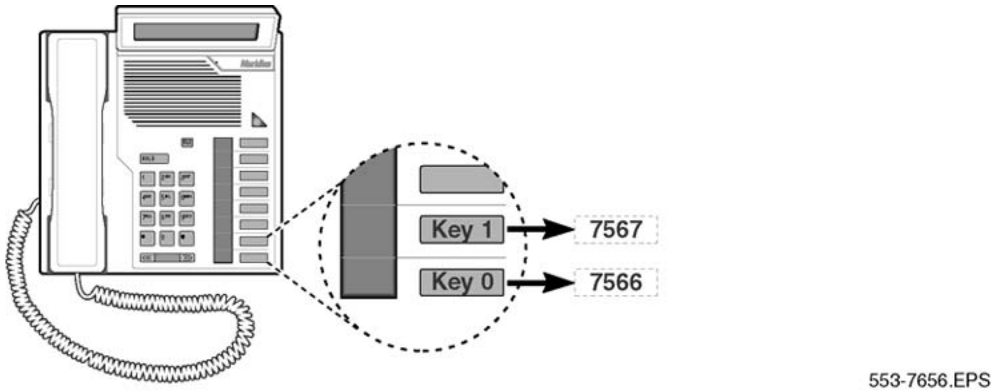
2. For a private ESN CDP call made from Key 0 (DN 2130) on the phone, the CLID would be built as follows:
    - LSC from CLID entry 0 (5) plus the DN of the active key (2130).
    - That is, the CLID that is sent would be 5 2130.
3. For a public local call made from Key 0 (DN 2130) on the phone, the 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 phone, 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 - Phone 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 327: 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



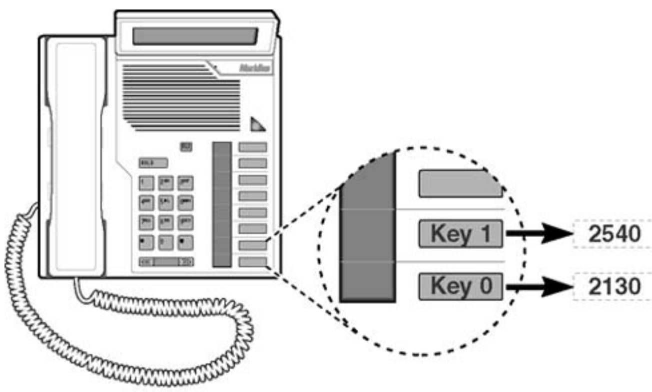
1. For a private ESN UDP call made from Key 0 (DN 7566) on the phone, the CLID would be built as follows:
    - HLOC from CLID entry 0 (578) plus the DN of the active key (7566).
    - That is, the CLID that is sent would be 578 7566.
2. For a private ESN CDP call made from Key 0 (DN 7566) on the phone, 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 phone, 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 phone, 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 - Phone 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 328: 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

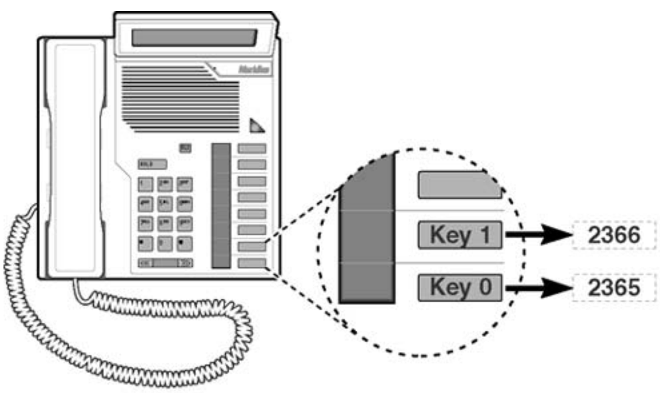
1. For a private ESN UDP call made from Key 1 (DN 2540) on the phone, the CLID would be built as follows:
    - HLOC from CLID entry 1 (646) plus the DN of the active key (2540).
    - That is, the CLID that is sent would be 646 2540.
2. For a private ESN CDP call made from Key 1 (DN 2540) on the phone, 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 phone, 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 phone, 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 - Phone 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 329: 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



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1. For a private ESN UDP call made from Key 1 (DN 2366) on the phone, 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 phone, 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 phone, 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 phone, 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.

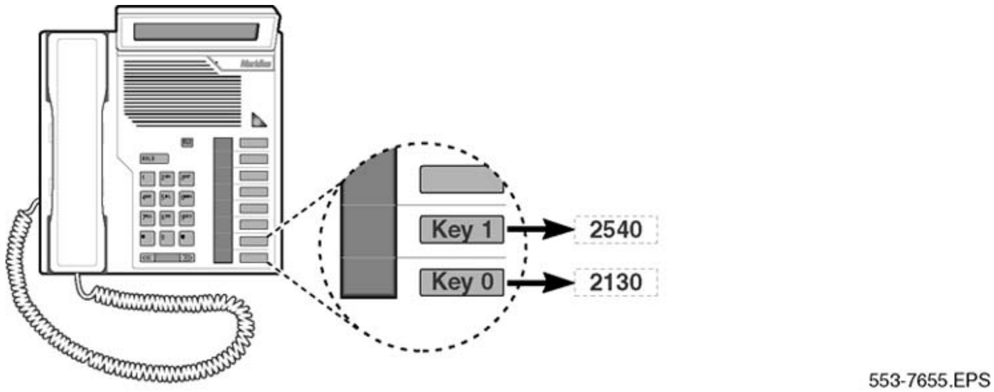
Example 5 - Phone does not have DID number on DN key 1, SRCH option is used (DIDN is set to SRCH)

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 0.

Table 330: 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



1. For a private ESN UDP call made from Key 1 (DN 2540) on the phone, 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 phone, 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 phone, 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 phone, 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

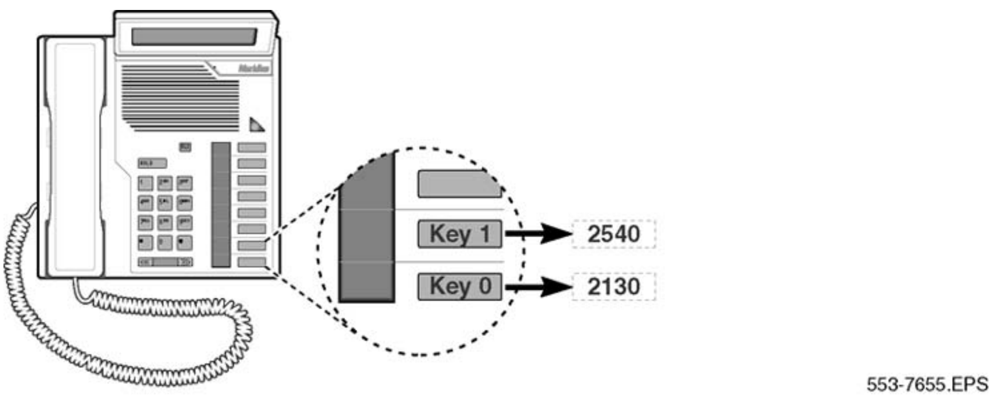
In this example, the calls are made on DN key 1, which does not have a DID number. Key 0 is configured against CLID table entry 1, and key 1 is configured against CLID table entry 2, which has the DIDN option set to SRCH. This means that the CLID assigned for key 1 is constructed based on a search starting 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 331: 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



1. For a private ESN UDP call made from Key 1 (DN 2540) on the phone, 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 phone, 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 phone, 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 phone, 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.

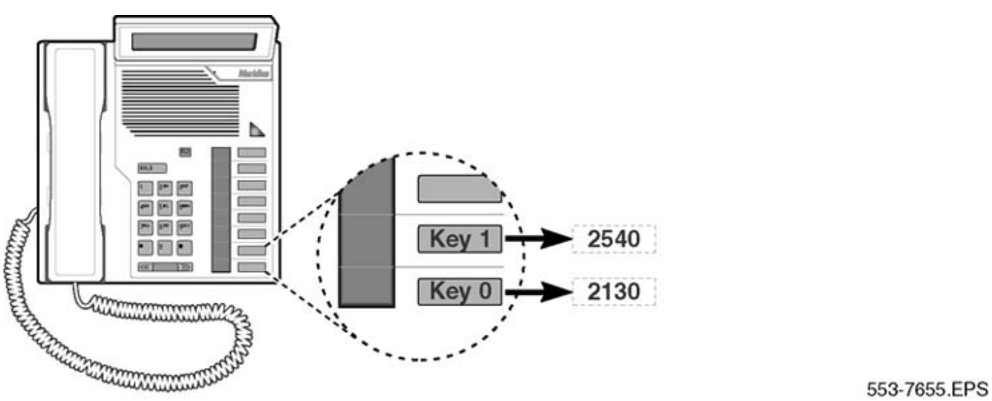
Example 7 - Phone with DN KEY prompt of D on key 1

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 phone 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 332: 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



1. For a private ESN UDP call made from Key 1 (DN 2540) on the phone, 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 phone, 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 phone, 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 phone, 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.

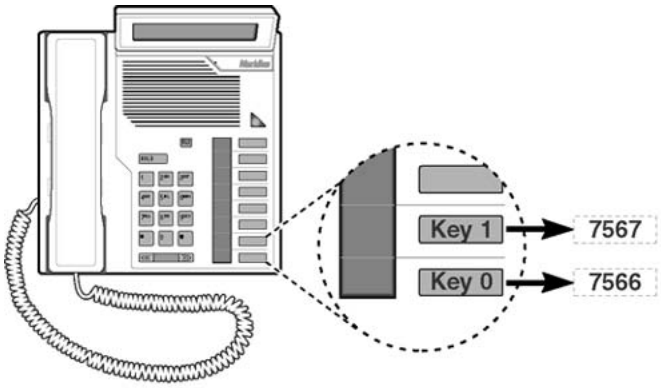
Example 8 - Phone with DN KEY prompt of D on key 1, International Numbering Plan

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 phone 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 333: 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



1. For a private ESN UDP call made from Key 1 (DN 7567) on the phone, 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 phone, 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 phone, 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 phone, 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.

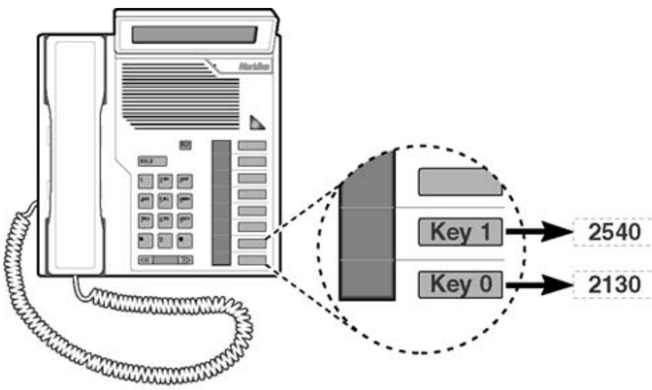
Example 9 - Phone with DN KEY prompt of D on key 0

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 phone 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 334: 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 phone, 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 phone, 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 phone, 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 phone, 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 system can be subject to any restrictions which can 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 through 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 phone 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 phone 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) phone.

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 phone

For an internal call terminating on an ISDN BRI phone, the calling phone'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 phone 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 phone to a EuroISDN interface, the CLID constructed by EuroISDN, based on the outgoing route, takes precedence over the CLID constructed for the calling station phone.

EuroISDN Trunk - Network Side

The EuroISDN Trunk - Network Side connectivity supports all of the user side ISDN CLID enhancements.

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 phones 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, by 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 phone 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

This feature requires the Integrated Services Digital Network (ISDN) package 145.

Feature implementation

Task summary list

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

1. Table 335: LD 15 - Construct the CLID entry for a customer. on page 738
2. Table 336: LD 21 - Print the CLID table for a customer. on page 740
3. Table 337: LD 10 - Define the CLID entry for analog (500/2500-type) phones. on page 741
4. Table 338: LD 11 - Define the CLID entry for DN keys for Meridian 1 Proprietary Phones. on page 741
5. Table 339: LD 27 - Define the CLID entry for ISDN BRI phones. on page 742

Table 335: LD 15 - Construct the CLID entry for a customer.

Prompt	Response	Description
REQ:	NEW CHG	Add new data, or change existing data.

Prompt	Response	Description
TYPE:	NET	Networking data (if REQ = CHG only.)
CUST		Customer number
	0-99	Range for Large System and Avaya Communication Server 1000E (Avaya CS 1000E) system.
	0-31	Range for Media Gateway 1000 B (Avaya MG 1000B).
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 can 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 can 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 can 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 can 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.

Prompt	Response	Description
- - - DIDN		How to use the DN as a DID when constructing a CLID national or local number.
	(YES)	YES = The default. The CLID is constructed using the digits defined in HLCL followed by the DN of the active key.
	NO SRCH	NO = Construct the CLID using the digits defined in HLCL. SRCH = Search on the phone, 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. Refer to Note 3 for the ENTRY prompt.

It is advised that after the CLID table is built in LD 15, generate a printout using LD 21 to verify the table entries defined in LD 15.

Table 336: 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, as defined in LD 15.
SIZE	0-(256)-1000	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.

Prompt	Response	Description
- 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.

Table 337: LD 10 - Define the CLID entry for analog (500/2500-type) phones.

Prompt	Response	Description
REQ:	NEW CHG	Add new data, or change existing data.
TYPE:	500	500 phone.
TN		Terminal number
	l s c u	Format for Large System and CS 1000E system, where l = loop, s = shelf, c = card, u = unit.
	c u	Format for Media Gateway 1000B, where c = card and u = unit.
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).

Table 338: LD 11 - Define the CLID entry for DN keys for Meridian 1 Proprietary Phones.

Note:

When assigning a CLID entry to an ACD phone, you cannot use the same position ID already on the phone. The phone 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:	a...a	Telephone type. Type **?** for a list of possible responses.
TN		Terminal number
	l s c u	Format for Large System and CS 1000E system, where l = loop, s = shelf, c = card, u = unit.
	c u	Format for Media Gateway 1000B, where c = card and u = unit.
KEY		Phone function key assignments.

Prompt	Response	Description
	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	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 (0-1999). 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 this table, that pertains to assigning a CLID entry to an ACD set.

Table 339: LD 27 - Define the CLID entry for ISDN BRI phones.

Prompt	Response	Description
REQ	NEW	Add new data.
	CHG	Change existing data.
TYPE	TSP	Administer the Terminal Service Profile on the Digital Subscriber Loop.

Prompt	Response	Description
...		
- 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).

Feature operation

No specific operating procedures are required to use this feature.

Chapter 52: ISDN QSIG

Contents

This section contains information on the following topics:

Applicable regions

ISDN BRI trunking is not available in North America. Contact your system supplier or your Avaya representative to verify support of this product in your area.

Feature description

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 Telecommunication Network Exchanges (PTNXs) connected within a Private Telecommunication Network (PTN). This protocol has been adopted by the European Telecommunication Standards Institute (ETSI) and the International Standards Organization (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 supports the following services:

- Call Establishment and Tear Down
- ETSI or ISO version of basic call
- 64 Kbit/s 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
- Transit Count information transmitted when ISDN Call Connection Limitation (ICCL) is present

The QSIG interface is supported on PRI, BRI, and ISDN Signaling Links (ISLs). The QSIG interface does not have any transit capability for supplementary services.

QSIG development underscores Avaya'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 through end-to-end ISDN networks.

Operating parameters

There are no operating parameters specified for this feature.

Feature interactions

Networking Features

Some networking features currently exist on more than one ISDN interface implemented on the system. These features are listed in Table 340: Networking features that exist on more than one ISDN interface implemented on the system on page 747. 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.

Table 340: Networking features that exist on more than one ISDN interface implemented on the system

	Euro-ISDN	MCDN	QSIG	BRI pho-nes	DPNSS	MFC	MFE	DTI2	Ana-log	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 system 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 does not 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 system 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 though 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.

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

Call Forward, Break-In and Hunt Internal or External Network Wide uses the Network Attendant Service equivalent information that is transported on protocols such as Party Category and Progress Indicator for QSIG.

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 for QSIG:

- Digit Display (DDSP) package 19
- Integrated Services Digital Network (ISDN) package 145
- Primary Rate Access (PRA) package 146
- 2.0 Mbit Primary Rate Access (PRI2) package 154
- Overlap Signaling (OVLP) package 184
- International Primary Rate Access (IPRA) package 202

Feature implementation

Task summary list

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

1. Table 341: LD 17 - Assign the configuration record. on page 749
2. Table 342: LD 16 - Define a Route Data Block. on page 749
3. Table 343: LD 97 - Configure the Extended Peripheral Equipment (XPE). on page 750 This overlay defines the Extended Peripheral Equipment (XPE) configuration. Data relating to the software downloading for the MSDL card remains in the overlay. Hence, this overlay must be modified to accept the parameters required for downloading the PRI application data files. The parameters specify conditional or forced downloading.

Table 341: LD 17 - Assign the configuration record.

Prompt	Response	Description
REQ	CHG	Change.
TYPE	CFN	Configuration Record.
...		
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.

Table 342: LD 16 - Define a Route Data Block.

Prompt	Response	Description
REQ	NEW	Add new data.
	CHG	Change existing data.
TYPE	RDB	Route Data Block.
...		
DGTP	PRI BRI	Digital route type.
IFC	ISIG ESIG	New DCH interface ID.

Table 343: LD 97 - Configure the Extended Peripheral Equipment (XPE).

Prompt	Response	Description
REQ	CHG	Change.
TYPE	SYSM	System parameters for MSDL/MISP cards.
...		
FDLC		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).

Feature operation

There are no operating procedures specified for this feature.

Chapter 53: ISDN QSIG Alternate Routing

Contents

This section contains information on the following topics:

Applicable regions

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

Feature description

The ISDN QSIG Alternate Routing feature provides a solution to calls encountering congestion due to high traffic situations within a QSIG network. The QSIG interface protocols, European Telecommunication 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 system 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 can 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 can 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 are 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 (see - COPT).

Transit node operation

Consider the following calling scenario for a transit node operation (refer to Figure 79: Transit node operation on page 753). An attempt is being made to establish a call over a QSIG link, from originating node A to terminating node C, through 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 an 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.

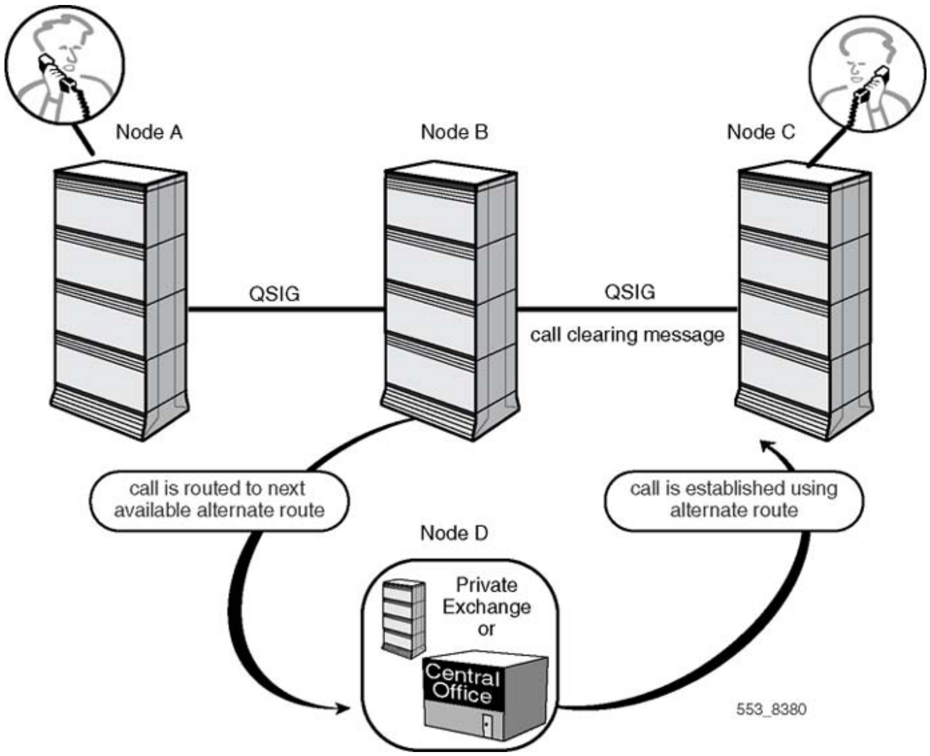


Figure 79: Transit node operation

An attempt is made to find a free alternate route until all of the alternate routes, as defined in LD 86 (refer to the ROUT prompt on ROUT), 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 80: Originating node operation on page 754). 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.

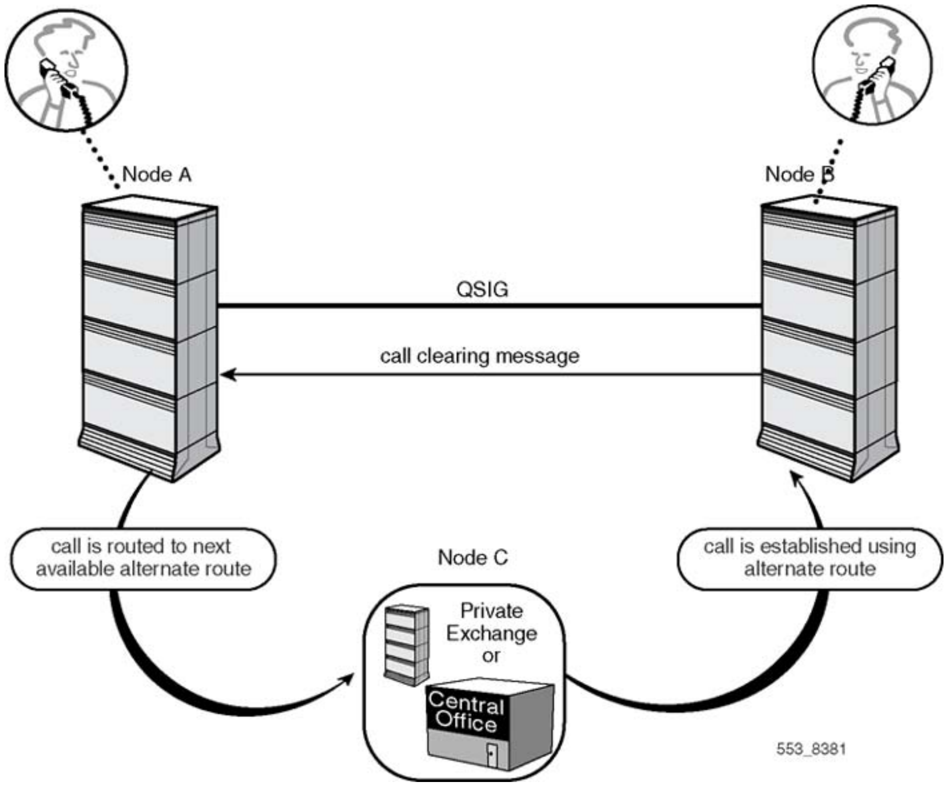


Figure 80: Originating node operation

An attempt to find a free alternate is made until all of the alternate routes, as defined in LD 86 (refer to the ROUT prompt on ROUT), 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) can 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 can create a situation where a call fails involving an ISDN BRI phone. This can occur when a Meridian proprietary phone calls an ISDN BRI phone 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

This feature requires the following packages:

- Basic Routing (BRTE) package 14
- Network Class of Service (NCOS) package 32

One of the following:

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

Feature implementation

The only task in this section is Table 344: LD 86 - Configure the QSIG Alternate Routing options. on page 757

Note:

QSIG Alternate Routing can be configured for each of the 512 different routes.

Table 344: LD 86 - Configure the QSIG Alternate Routing options.

Prompt	Response	Description
REQ	CHG	Change existing data.
CUST	xx	Customer number, as defined in LD 15.
FEAT	RLB	Configure the Route List as a feature.
MXLC	0-1999 0-16000	Maximum number of NARS Location Codes. Maximum number of Location Codes (with the ESN Location Code Expansion feature enabled and with the FNP package 160)

Prompt	Response	Description
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 = 0-1999 if the Flexible Numbering Plan is configured.
ENTR	0-63	Entry number for the NARS route list.
ROUT		Route number
	0-511	Range for Large System and Avaya Communication Server 1000E (Avaya CS 1000E) system.
	0-127	Range for Media Gateway 1000 B (Avaya MG 1000B).
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 can 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		ISDN Drop Back Busy options.
	(DBD)	Deny Drop Back Busy.
	DBA	Allow Drop Back Busy on, if all route phones are busy.
	DBI	Allow Drop Back Busy, if all Initial route phones are busy.

Feature operation

No specific operating procedures are required to use this feature.

Chapter 54: ISDN QSIG Basic Call

Contents

This section contains information on the following topics:

Applicable regions

ISDN BRI trunking is not available in North America. Contact your system supplier or your Avaya representative to verify support of this product in your area.

Feature description

Integrated Services Digital Network (ISDN) Q Reference Signaling Point (QSIG) Basic Call complies with the ISDN protocol that specifies the Layer 3 signaling requirement for support of circuit switched call control at the "Q" reference point between Private Telecommunication Network Exchanges (PTNXs) connected within a Private Telecommunication Network (PTN).

This protocol has been adopted by the European Telecommunication 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. This applies to both ISDN PRI and ISDN BRIT (trunk application.)

QSIG is oriented toward signaling and services that occur between two "peer-to-peer" switches across a "Q" reference point. For example, two PBXs, a PBX and a Centrex switch or two Centrex switches could exchange signaling and services across a "Q" reference point.

The ISDN QSIG Basic Call feature introduces the following basic call services on ISDN 1.5 Mbit PRI on QSIG connectivities (these services were previously available only on ISDN 2.0 Mbit PRI connectivities):

- Basic Call Service
- 64 K clear bearer capability
- TIE call types
- Connected number delivery (COLP/COLR)
- Channel Negotiation
- Flexible Numbering Plan
- Enbloc dialing
- Overlap sending
- Overlap receiving
- Calling Line Identification Presentation and Restriction (CLIP/CLIR)
- Loop avoidance (transit count) on ETSI QISG, not on ISO QSIG
- Party Category on ETSI QISG, not on ISO QSIG.

Operating parameters

Meridian Customer Defined Network (MCDN) features cannot be interworked between an MCDN and QSIG interface, with the exception of basic call, Calling Line Identification Presentation/Restriction and Calling and Connected Line Presentation/Restriction.

The system accepts up to three Progress Information Elements (IE) in one message on the QSIG interface.

The maximum length of an ISDN message is limited to 260 bytes.

Only circuit mode connection is supported.

nB+D is not supported for QSIG.

Feature interactions

Backup D-Channel

Backup D-Channel is not supported on the QSIG interface.

ISDN Signaling Link

ISL on QSIG is not supported on ISDN BRI trunking for Large Systems.

Network Attendant Services

ISDN QSIG Basic Call interacts with Network Attendant Services (NAS) as if the call is going to a route without NAS being equipped.

Network Automatic Call Distribution

Network Automatic Call Distribution signaling is not supported on the QSIG interface. Network ACD operation on QSIG is the same as the existing Network ACD treatment for unsupported interfaces.

Network Call Redirection

When a call is terminated on the system and Network Call Redirection (NCR) is active, the QSIG Basic Call can still operate; however, the original called number and redirection number IE that are used by NCR will not be sent on the QSIG interface.

Network Call Trace

Network Call Trace is not supported on the QSIG interface.

Network Calling Party Name Display

Network Calling Party Name Display (NCPND) is supported within the Meridian Customer Defined Network only. When QSIG is involved in the call setup, the existing NCPND operation on unsupported interfaces applies on QSIG.

Network Message Service

Network Message Service (NMS) is only supported on the Meridian Customer Defined Network. NMS operation on QSIG is the same as the existing treatment for unsupported interfaces.

Network Ring Again

Network Ring Again signaling is supported within the Meridian Customer Defined Network only. Network Ring Again requests which go through the QSIG interface will not be supported.

Remote Virtual Queuing

Remote Virtual Queuing (RVQ) does not operate on the QSIG interface. The existing RVQ operation on unsupported interfaces applies on the QSIG interface.

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 will go through the QSIG interface just as a normal basic call. However, TRO signaling will not be able to operate on the QSIG interface.

Virtual Network Service

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.

Feature packaging

This feature requires the following packages:

- Digit Display (DDSP) package 19
- Integrated Services Digital Network (ISDN) package 145
- Primary Rate Access (PRI) package 146
- 2.0 Mbit Primary Rate Access (PRI2) package 154
- Overlap Signaling (OVLP) package 184
- International Primary Rate Access (IPRA) package 202
- ISDN QSIG Basic Call (QSIG) is package 263

Feature implementation

Task summary list

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

1. Table 345: LD 17 - Assign Configuration Record. on page 765
2. Table 346: LD 16 - Define the Route Data Block. on page 766
3. Table 347: LD 97 - System Configuration. on page 766

Table 345: LD 17 - Assign Configuration Record.

Prompt	Response	Description
REQ	CHG	Change existing data.
TYPE	ADAN	Action Device and Number.
- ADAN	CHG DCH x	Change input/output device. x = 0-63
...		
- - IFC	ISIG ESIG	Interface ID for ISO QSIG download table. Interface ID for ETSI QSIG download table.
...		

Prompt	Response	Description
TIMR	(NO) YES	NO = skip timer prompt. YES = change timer value.
T310	110 - (120)	The timer is used to define how long the system can wait for the response message when the QSIG outgoing call is in an outgoing call proceeding state.

Table 346: LD 16 - Define the Route Data Block.

Prompt	Response	Description
REQ	NEW	Add new data.
	CHG	Change existing data.
TYPE	RDB	Route Data Block.
DTRK	YES	Digital Trunk Route.
...		
DGTP	PRI PRI2 BRI	1.5 Mbit PRI. 2.0 Mbit PRI. Basic Rate Interface.
IFC	ISIG ESIG	New DCH interface identification as entered in LD 17.

Table 347: LD 97 - System Configuration.

Prompt	Response	Description
REQ	CHG	Change existing data.
TYPE	SYSM	System parameters for Multi-purpose Serial Data Link (MSDL) or Multi-purpose ISDN Signaling Processor (MISP).
...		
FDLC	p1 p2 p3 p4	Fast download control parameters. p1 = Application will be set to PRIE or BRIE. p2 = Specifies conditional (C) or forced downloading (F). p3 = Specifies current (C), latest (L) and specified (S) version. p4 = xx (version number if p3 = S).

Feature operation

No specific operating procedures are required to use this feature.

Chapter 55: ISDN QSIG-BC and QSIG-GF Compliance Update

Contents

This section contains information on the following topics:

Applicable regions

ISDN BRI trunking is not available in North America. Contact your system supplier or your Avaya representative to verify support of this product in your area.

Feature description

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 versions of the standards.

The ISDN QSIG-BC and QSIG-GF Compliance Update feature is compliant with the following European standards documents for QSIG:

- ETS 300-172, fourth edition (1997)
- ETS 300-239, second edition

Initially, 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.

Later, 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 348: Unrecognized comprehension required IE treatment on page 768.

Table 348: 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: 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

Progress indicators 15, 16, 17 and 18, defined in the European Computer Manufacturer's Association (ECMA) coding are not defined for this interface.

QSIG GF Compliance Update

Facility IE coding

Figure 81: Facility IE layout on page 769 illustrates the layout of the Facility IE.

Bit	8	7	6	5	4	3	2	1	Octet
		Facility Information Element Identifier							
	0	0	0	1	1	1	0	0	1
	Length of Information Element Contents								2
	ext	spare		Protocol Profile					
	1	0	0						3
	NetworkFacilityExtension								3.1
	NetworkProtocolProfile								3.2
	InterpretationAPDU								3.3
	Service APDU								4...

Figure 81: Facility IE layout

Protocol profile coding

Table 349: Protocol Profile Coding on page 770 illustrates three different examples of coding in the Protocol Profile.

Table 349: 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 system codes the Protocol Profile as "Networking Extension."

On reception, the system 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 system 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 earlier 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 Mbit 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

There are two choices of PRI2 hardware forLarge Systems:

- PRI2 card (NT8D72) and MSDL card (NT6D80)
- Dual port DTI/PRI2 card and either the Downloadable D-channel Daughterboard card (NTBK51) or the MSDL card (NT6D80)

These systems require the NTRB53 or the QPC471 Clock Controller card.

The hardware required for ISDN BRI is as follows:

- SILC card (NT6D70)
- MISP card (NT6D73) forLarge Systems

Feature interactions

There are no feature interactions associated with this feature.

Feature packaging

This feature requires the following packages:

- ISDN Signaling (ISDN) package 145
- Primary Rate Access (PRA) package 146
- 2 Mbit/s 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
- 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 implementation

Task summary list

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

1. Table 350: LD 17 - Configure the updated interface type for the D-channel. on page 772
2. Table 351: LD 16 - Configure the updated interface type for the trunk route. on page 772

Table 350: LD 17 - Configure the updated interface type for the D-channel.

Prompt	Response	Description
REQ	CHG	Change existing data.
TYPE	CFN	Configuration Record.
ADAN	CHG DCH x	Change D-Channel number.
...		
- IFC	ESGF ISGF	ETSI QSIG interface with GF platform. ISO QSIG interface with GF platform.

Table 351: LD 16 - Configure the updated interface type for the trunk route.

Prompt	Response	Description
REQ	NEW	Add new data.
	CHG	Change existing data.
TYPE	RDB	Route Data Block.

Prompt	Response	Description
CUST	xx	Customer number, as defined in LD 15.
ROUT		Route number
	0-511	Range for Large System and Avaya Communication Server 1000E (Avaya CS 1000E) system.
	0-127	Range for Media Gateway 1000 B (Avaya MG 1000B).
DTRK	YES	Digital Trunk Type.
- DGTP	PRI BRI	PRI trunk. BRI trunk.
- IFC	ESGF ISGF	ETSI QSIG interface with GF platform. ISO QSIG interface with GF platform.

Feature operation

No specific operating procedures are required to use this feature.

Chapter 56: ISDN QSIG Call Diversion Notification

Contents

This section contains information on the following topics:

Applicable regions

ISDN BRI trunking is not available in North America. Contact your system supplier or your Avaya representative to verify support of this product in your area.

Feature description

ISDN QSIG Call Diversion Notification allows the system 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 phone is in use
- Call Forwarding No Reply (CFNR)—the called phone does not answer after a given number of rings
- Call Forwarding Unconditional (CFU)—the called phone is forwarded as soon as the phone 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 phone and on the forwarded-to (diverted) party's phone 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 phone. 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 82: Call Diversion Environment on page 777 shows an example of a Call Diversion environment.

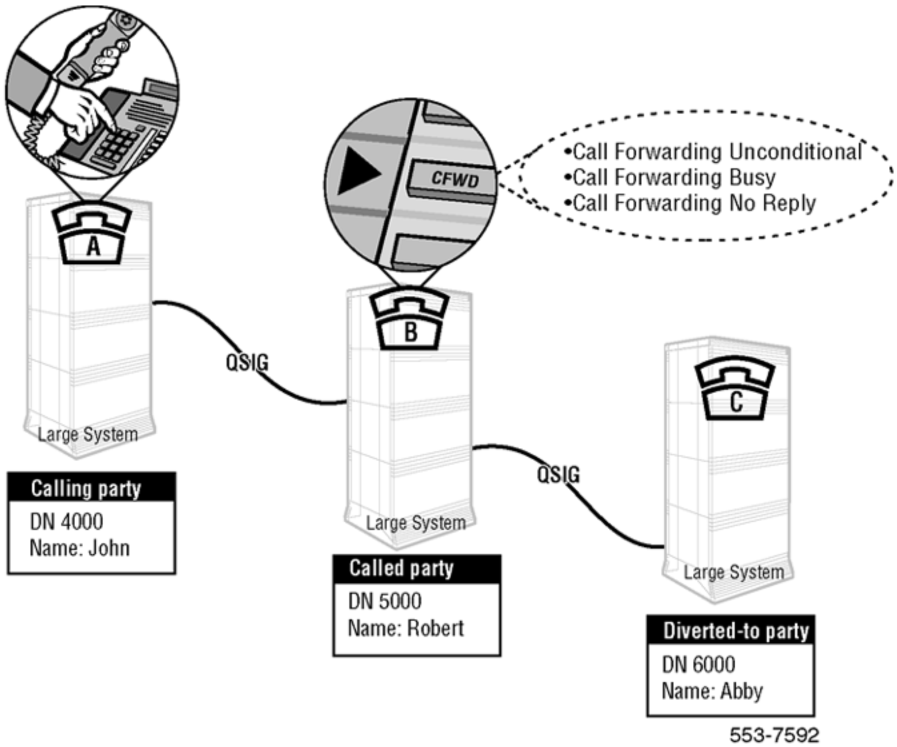
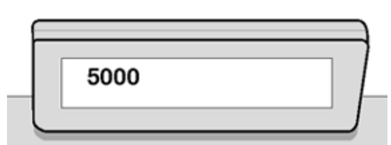
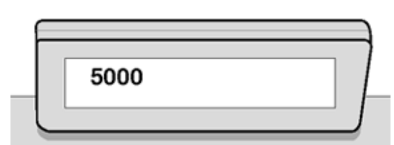


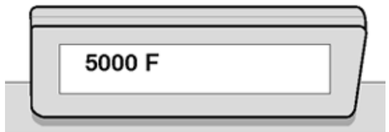
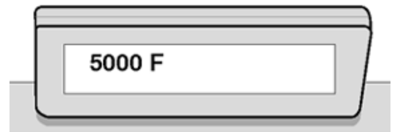
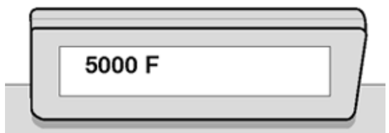
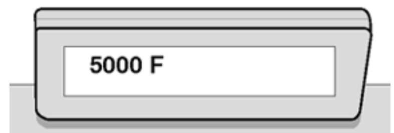
Figure 82: Call Diversion Environment

Calling Party Notification

The calling party that originates the call receives notification of call diversion depending on the configuration of the called party's phone. As shown in Table 352: Calling Party Notification on page 777, the called party configures one of the three possible Classes of Service.

Table 352: Calling Party Notification

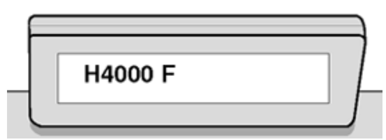
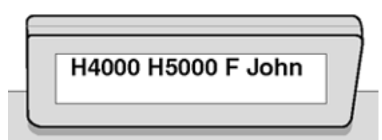
Called Party's Diversion Subscription Options (Class of Service)	Calling Party's Display when call is forwarded.	Forwarded-to Party's Display
CLS = DNO1 No diversion notification provided to calling party.		

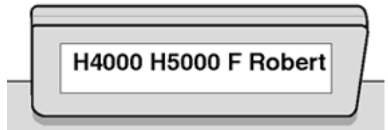
Called Party's Diversion Subscription Options (Class of Service)	Calling Party's Display when call is forwarded.	Forwarded-to Party's Display
CLS = DNO2 Diversion notification provided without forwarded-to (diverted) party's number and name.		
CLS = DNO3 (default) Diversion notification provided with forwarded-to (diverted) party's number and name when available.		

Called Party Notification

The forwarded-to (diverted) party receives notification of call diversion depending on the configuration of the called party's phone. As shown in Table 353: Forwarded-to Party Notification on page 778, the called party configures one of the two possible Classes of Service.

Table 353: Forwarded-to Party 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)

Called Party's Diversion Subscription Options (Class of Service)	Forwarded-to Party's Display.	Forwarded-to Party's Class of Service
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 phone, 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 system user cannot remotely activate or deactivate QSIG Call Diversion Notification from either a system or on another third party Private Branch Exchange (PBX).

According to the current operation of the Call Forward features, the system does not provide any indication, such an LED, on the called party's phone 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 a 27-character string length is supported. Accordingly, names processed by the system can 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 interactions

There are no feature interactions associated with this feature.

Feature packaging

QSIG Call Diversion Notification is ISDN QSIG Supplementary Service - Call Completion (QSIG-SS) package 316, which has the following dependencies:

- Digit Display (DDSP) package 19
- Network Alternate Route Selection (NARS) package 58 (for Uniform Dialing Plan), or Coordinated Dialing Plan (CDP) package 59
- Calling Party Name Display (CPND) package 95
- QSIG Interface (QSIG) package 263
- ISDN QSIG-GF Transport (QSIG-GF) package 305

ISDN PRI QSIG Interface requires:

- Primary Rate Access (PRI) package 146
- 2.0 Mbit 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 implementation

Task summary list

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

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 354: Remote Capability Meanings on page 782 indicates the remote capabilities meanings for both D-channel and BRI routes. When using Table 354: Remote Capability Meanings on page 782 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 usually coded with Integer Values. For ETSI interfaces (IFC ESGF), the operation coding depends on the standard version. The latest ones are likely to use Integer Values, whereas some older ones can 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 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. Treating 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 354: 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 sent	Not processed when received
DV1O	Sent coded as Object Identifier	Sent	Not processed when received
DV1I	Sent coded as Integer Value		

Remote capability	Meaning for Operation Coding	Meaning for Notification Information	Meaning for Rerouting request
DV2O	Sent coded as Object Identifier	Not Sent	Processed when received
DV2I	Sent coded as Integer Value		
DV3O	Sent coded as Object Identifier	Sent	Processed when received
DV3I	Sent coded as Integer Value		

Table 355: 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	xx	Software release.
...		
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.

Table 356: 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, as defined in LD 15.
ROUT		Route number
	0-511	Range for Large System and Avaya Communication Server 1000E (Avaya CS 1000E) system.
	0-127	Range for Media Gateway 1000 B (Avaya MG 1000B).
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 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.

Table 357: LD 10 - Configure QSIG Diversion Notification on analog (500/2500-type) phones.

Prompt	Response	Description
REQ:	NEW CHG	Add new data, or change existing data.
TYPE:	500	Type of phone.
TN		Terminal number
	l s c u	Format for Large System and CS 1000E system, where l = loop, s = shelf, c = card, u = unit.
	c u	Format for Media Gateway 1000B, where c = card and u = unit.
DN	x...x yyy	Directory Number and Calling Line Identification entry (Range is (0) - value entered for SIZE prompt in LD 15 minus one).

Prompt	Response	Description
- 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 phone Allowed.
	FNA	Call Forward No Answer Allowed.
	HTA	Hunting Allowed.
	NAMA	Name Display on other phone Allowed or (Denied).

Table 358: LD 11 - Configure QSIG Diversion Notification on Meridian 1 proprietary phones.

Prompt	Response	Description
REQ:	NEW	Add new data.
	CHG	Change existing data.
TYPE:	xxxx	Type of phone.
TN		Terminal number
	l s c u	Format for Large System and CS 1000E system, where l = loop, s = shelf, c = card, u = unit.
	c u	Format for Media Gateway 1000B, where c = card and u = unit.
FDN	x...x	Flexible Call Forward No Answer DN for an Internal Directory Number.

Prompt	Response	Description
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 phone Allowed.
	FNA	Call Forward No Answer Allowed.
	HTA	Hunting Allowed.
	NAMA	Name Display on other phone 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.

Table 359: LD 27 - Configure Basic Rate Interface phones.

Prompt	Response	Description
REQ	NEW	Add new data.
	CHG	Change existing data.
TYPE	TSP	Terminal Service Profile.
DSL	l s c dsl c dsl	Digital Subscriber Loop.
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.

Prompt	Response	Description
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_E TSI	aaa mmm nnn	ETSI Supplementary Service where: VCFW = Voice Call Forward and DCFW = Data Call Forward.

Table 360: 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, as defined in LD 15.
...		
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

Table 361: LD 95 - Define Name for phones.

Prompt	Response	Description
REQ	NEW	Add new data.
TYPE	NAME	Name.
CUST	xx	Customer number, as defined in LD 15.
...		
DN	x..x	Directory Number.
- NAME	a..a	Calling Party Name Display in ASCII characters.

Table 362: LD 11 - Configure the name that is to be displayed on the Forwarded-to (diverted) phone.

Prompt	Response	Description
REQ:	CHG	Change existing data.
TYPE:	a...a	Telephone type. Type **?** for a list of possible responses.
TN		Terminal number
	l s c u	Format for Large System and CS 1000E system, (DNDD)= Dialed Name Display Denied (default).
	c u	Format for Media Gateway 1000B, where c = card and u = unit.
CLS	DNDA	Dialed Name Display Allowed. (DNDD)= Dialed Name Display Denied (default).

Note:

LD 81 can 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 can 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 Class of Service prompt.

Feature operation

No specific operating procedures are required to use this feature.

Chapter 57: ISDN QSIG Call Diversion Notification Enhancements

Contents

This section contains information on the following topics:

Applicable regions

ISDN BRI trunking is not available in North America. Contact your system supplier or your Avaya representative to verify support of this product in your area.

Feature description

The ISDN QSIG Call Diversion Notification feature allows the system 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 phone is in use
- Call Forwarding No Reply (CFNR)—the called phone does not answer after a given number of rings
- Call Forwarding Unconditional (CFU)—the called phone is forwarded as soon as the phone 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 phone and on the forwarded-to (diverted) party's phone 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 phone. 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 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.

The QSIG Call Diversion Notification service is provided across an ISDN QSIG network using the QSIG Generic Functional (GF) protocol transport platform.

With this enhancement, the ISDN QSIG Call Diversion Notification Enhancements feature allows the QSIG Call Diversion Notification functionality to be also supported over a QSIG/Meridian Customer Defined Network (MCDN) and a QSIG/Digital Private Signaling System No.1 (DPNSS1) network, as well as a QSIG network. According to the existing implementation, the QSIG Call Diversion Notification Enhancements feature is supported on ISDN Primary Rate Interface and ISDN Basic Rate Interface Trunking (BRIT) interfaces.

The notification of the originating phone and the diverted-to phone now depends on the type of signaling protocol that is in play (QSIG, DPNSS1, or MCDN), at what stage of the call establishment that the diversion takes place, and the user's subscription option.

Operating parameters

There are no operating parameters associated with this feature.

Determination of the rerouting node

The determination of the rerouting node depends on the type of diversion that is configured for the node, either Call Diversion by Rerouting or Call Diversion by Forward Switching.

If the Forward Switching method is implemented (call diversion is generated by the system or another third-party PBX also using the Forward Switching method), the rerouting node is the served node. If the Call Diversion by Rerouting method is implemented (a third-party switch using the Rerouting method generates a diversion), the rerouting node is the originating node.

The Forward Switching method is the one presently implemented on the system. It performs the diversion by joining together the first connection from the originating user's node to the served user's node, and the second new connection from the served user's node to the diverted-to user's node. In fact, the rerouting node is the served user's node.

Notifications rules

This section describes Originating and Diverted-to user's notification rules, for single as well as multiple diversions in a pure QSIG environment.

Single Diversion

Originating and Diverted-to user's Notifications are detailed for the following scenario: "Station A calls Station B. Station B has activated CFU, CFB or CFNR to Station C. Station C answers".

Originating user notification rules

Diversion Reason Notification rules:

- This reason is displayed or not on the originating user's phone (as soon as Diversion Notification information is received from the served node) according to the served user's Subscription option (see Table 363: Notification to Originating user versus Subscription option of served user on page 792).

Diverted-to Number Notification rules:

- The diverted-to number (received in the Diversion Notification information delivered by the served node) is displayed on the originating user's phone:
  - if the received presentation information (received in the Diversion Notification information issued from the diverted-to node) indicates that presentation is allowed.

- if the served user's subscription option (received within Diversion Notification information from the served node) allows it (see Table 363: Notification to Originating user versus Subscription option of served user on page 792).

Diverted-to Name Notification rules:

- The diverted-to name, when available (as optionally received as part of the Diversion Notification information delivered by the diverted-to node), is displayed on the originating user's phone:
  - if the intrinsic name presentation (received in the Diversion Notification information issued from the diverted-to node) indicates that presentation is allowed.
  - if the served user's subscription option (received within Diversion Notification information from the served node) allows it (see Table 363: Notification to Originating user versus Subscription option of served user on page 792).

Note that DNDA/DNDD functionality has no impact on the originating user's notification.

When both originating and served users are on the same node, the existing system treatment is still applicable, which is that the served user's subscription options have no impact on the originating user notification.

When both originating and diverted-to users are on the same system node, and diversion is performed by the Rerouting method, then the served user's subscription options are effective for the originating user notification only before the diverted-to user answers, but no name is provided. As soon as the diverted-to user answers, then the originating user receives full notification (reason, diverted-to user's number and name), according to the served user's subscription option.

Table 363: Notification to Originating user versus Subscription option of served user on page 792 sums up originating user's Notification according rules, according to the served user's Subscription options.

Table 363: Notification to Originating user versus Subscription option of served user

	Originating user's display	
Served user's Subscription option: "Calling user receives notification that the call has been diverted"	after receipt of served user's diversion notification information	after receipt of diverted-to user's diversion notification information
No	5000	5000
Yes without diverted-to number and name	5000 F	5000 F
Yes with diverted-to number and name when available	5000 F	5000 H6000 F Cathy

Diverted-to user notification rules

Diversion Reason Notification rules:

- The reason is displayed on the diverted-to user's phone (as soon as Diversion Notification Information is received from the served node).

Served Number Notification rules:

- The served user's number (the originally-called number) is displayed or not on the diverted-to user's phone, according to the served user's Subscription option (see Table 364: Diverted-to user notification versus served user Subscription option and diverted-to user class of service for single diversion on page 794). If the served user's Subscription option is to not release the calling party name/number to the diverted-to user, the served user's number can be displayed with dashes (if received with a restricted presentation) or not displayed (if no served user's number is received).

Served Name Notification rules:

- The functionality DNDA/DNDD allows the diverted-to user to choose the name to be displayed after diversion has taken place:
    - if the diverted-to phone has a DNDA Class of Service, then the diverted-to user's phone displays one of the following: - the original called name when available (as optionally received from the served user's node, depending on the previous served users's Subscription Option), or - if the original called name is not available, the redirecting name when available (as optionally received from the served user's node, depending on the served user's Subscription Option), or - nothing if none of the two previous names is available.
    - if the diverted-to phone has a DNDD class of service, then the calling user's name is displayed if available, otherwise nothing is displayed.

When both served and diverted-to users are on the same node, the present system treatment applies, that is, the served user's subscription options have no impact on the diverted-to user notification.

When both the originating and diverted-to users are on the same system node and diversion is performed by the Rerouting method, then served user's subscription options have no impact on the diverted-to user notification. In this case, DNDA functionality is not supported.

Table 364: Diverted-to user notification versus served user Subscription option and diverted-to user class of service for single diversion on page 794 sums up the diverted-to user's Notification rules, according to the served user's Subscription options and the diverted-to user's class of service DNDA/DNDD.

Note that, in the case presented in the example, the Originating user name is Abby, with a DN of 4000. The Served user name is Bobby, with a DN of 5000. The Diverted-to user name is Cathy, with a DN of 6000.

Table 364: Diverted-to user notification versus served user Subscription option and diverted-to user class of service for single diversion

Served user's Subscription option: "Served user releases the number/name to the diverted-to user"	Diverted-to user's display (after receipt of served user's diversion notification information) depending on the diverted-to user class of service:	
	DNDD	DNDA
No	H4000 H---- F Abby OR H4000 H---- F	H4000 F Abby OR H4000 F
Yes	H4000 H5000 F Abby	H4000 H5000 F Bobby

Multiple diversions

Originating and Diverted-to user's Notifications are detailed for the following scenario: "Station A calls Station B. Station B has activated CFU, CFB or CFNR to Station C. Station C has activated diversion to Station D, which itself has activated diversion to Station E. Station E has activated diversion to Station F. Station F answers".

Originating user notification rules

Diversion Reason Notification rules:

- The same rules apply according to the single diversion case (refer to Single Diversion on page 791). The last diversion reason will replace the previous one.

Diverted-to Number Notification rules

The diverted-to number (as received in the last Diversion Notification information message issued from the last served node) is presented to the calling user if all of the following conditions apply:

- all previously Diversion Notifications information messages received at the originating node contain a subscription option with a value of "Yes with diverted-to number/name", and
- any previously Diversion Notification information message issued from the diverted-to node contain a presentation indicator that allows presentation.

The last diverted-to number will replace the previous one.

Diverted-to Name Notification rules

The diverted-to name, when available (as optionally received within the Diversion Notification information message delivered by the diverted-to node), is presented to the calling user if all of the following conditions apply:

- all previously Diversion Notifications information messages received at the originating node include a subscription option with a value of "Yes with diverted-to number/name", and
- the intrinsic name presentation is allowed.

The last diverted-to name will replace the previous one. Note that DNDA/DNDD functionality has no impact on the originating user's notification.

Diverted-to user notification rules

Diversion Reason Notification rules:

- The same rules apply according to the single diversion case (refer to Single Diversion on page 791).

Served Number Notification rules:

- Either the originally-called number (as soon as a Diversion Notification information message is received from the last served node) is displayed or not on the diverted-to user's phone, according to the first served user's Subscription option.

Served Name Notification rules:

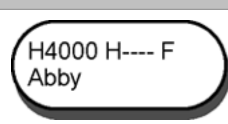
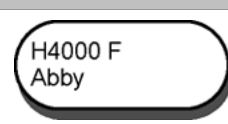
- The same rules, including the DNDA/DNDD functionality, apply as for the single diversion case (refer to Single Diversion on page 791).

Table 365: Diverted-to user notification versus served user Subscription option and diverted-to user class of service for multiple diversions on page 795 sums up a multiple diversion case considering the different parties' Subscription option, and the diverted-to user class of service.

Note:

In the example, the Originating user name is Abby, with a DN of 4000. The first Served user name is Bobby, with a DN of 5001. The second Served user name is Billy, with a DN of 5002. The Diverted-to user name is Cathy, with a DN of 6000.

Table 365: Diverted-to user notification versus served user Subscription option and diverted-to user class of service for multiple diversions

Served users' Subscription option: "Served user releases the number/name to the diverted-to user"		Diverted-to user's display (after receipt of diversion notification information) depending on the diverted-to user Class of Service:	
Bobby	**Billy**	**DNDD**	**DNDA**
No	No		

Served users' Subscription option: "Served user releases the number/name to the diverted-to user"		Diverted-to user's display (after receipt of diversion notification information) depending on the diverted-to user Class of Service:	
Bobby	**Billy**	**DNDD**	**DNDA**
Yes	No	OR H4000 H---- F H4000 H5001 F Abby	OR H4000 F H4000 H5001 F Bobby
No	Yes	H4000 H---- F Abby	H4000 F Abby
		OR	OR
		H4000 H---- F Billy	H4000 F Billy
Yes	Yes	H4000 H5001 F Abby	H4000 H5001 F Bobby

Feature interactions

Call Forward by Call Type

This feature allows to perform CFNA or Hunt to a different DN (FDN/EFD or HUNT/EHT) whether the call is internal or external.

This feature is supported by QSIG Diversion. The definition of an 'internal call' is not modified by the introduction of the QSIG Call Diversion Enhancements feature. An ISDN trunk call using private numbering is considered internal (note that no attempt is made to determine the real originating party with QSIG, only the type of numbering plan for the QSIG call is used).

Call Waiting Redirection

The Call Waiting Redirection feature allows unanswered calls given Call Waiting treatment to receive call redirection through Call Forward No Answer (CFNA) for the active phone. An unanswered call receiving Call Waiting treatment is redirected to the active phone's CFNA DN after the expiration of the customer data block defined CFNA timer selected for that phone. This redirected call is treated as an unanswered call by the CFNA DN (the call receives messaging services such as Voice Mail and Message Centers).

The QSIG Call Diversion Notification feature handles this feature as a usual CFNA.

Call Forward Subscription Option

The behavior of the Call Forward Subscription Option is not changed. The QSIG Call Diversion Notification Subscription Options are defined as Classes of Service, but the active one will always be the Served User's Subscription Option, independently of the OPT overlay configuration (CFO/CFF).

Phantom TN

A Phantom TN cannot be remotely activated, deactivated or interrogated through QSIG.

When a Phantom TN is Call Forwarded, the QSIG Diversion Notification is handled as if the Phantom TN was a normal DN.

Call Forward to Trunk Access Code

The CFTA prompt in LD 15, the Customer Data Block, is used to allow or disallow a Call Forward Trunk Access Code to be assigned to Call Forward All Calls DID calls.

Since QSIG Call Diversion is only supported on CDP or UDP dialing plans, it cannot be activated by CFTA.

Flexible Orbiting Prevention Timer

The Flexible Orbiting Prevention Time is supported on QSIG Diversion. However, since it limits the number of diversions that can be performed by a phone in a given time, it is recommended that the Total Redirection Count capability be used instead.

Call Forward/Hunt Override

This feature is not supported on QSIG Diversion. A system user can neither originate nor receive a call over a QSIG link through Call Forward/Hunt Override. At gateways between QSIG and MCDN, this information is lost.

Access Restrictions/Trunk Group Access Restrictions

Access restrictions are checked on the served node. This means that QSIG Diversion is not performed if the served user is not able to access the route to the diverted-to node.

Dialing plans

ISDN QSIG GF transport platform supports both the ISO QSIG GF specification and the ETSI QSIG GF specification. The same numbering plans and types of numbers supported by QSIG basic call are also supported by ISDN QSIG GF transport for any DN address translation required. They are summarized below:

E.164 Numbering Plan:

- Unknown
- International Number
- National Number
- Subscriber Number

Private Numbering Plan:

- Unknown
- ESN LOC (level 1 regional number)
- CDP (local number)

Unknown Numbering Plan:

- Same as for QSIG basic call. An Unknown Numbering Plan is treated as a Private numbering plan, with an 'unknown' type of number.

Meridian Link

Present Call Indication

A Present Call Indication message contents an IE called "Call Type", which contains Diversion information about the incoming call. In case of a QSIG Diverted incoming call, this field is updated.

Unsolicited Status Message

When a phone stops ringing because Call Forward No Reply has been applied to a QSIG call, an Unsolicited Status Message is sent to Meridian Link.

Call Completion to a Busy Subscriber

Call Forward Unconditional and Call Forward Busy

If Call Completion to a Busy Subscriber (CCBS) is to be invoked against a busy diverted-to user, the originating PINX stores the diverted-to number issued by the served PINX in order to use it for further CCBS treatments, without any manipulation.

Call Forward Unconditional (CFU) and Call Forward Busy (CFB) take precedence over CCBS.

Call Completion on No Reply

Call Forward Unconditional, Call Forward Busy, and Call Forward No Reply

If Call Completion on No Reply (CCNR) is to be invoked against a diverted-to user who does not answer, the originating PINX stores the diverted-to number issued by the served PINX in order to use it for further CCNR treatments, without any manipulation.

Call Forward No Reply (CFNR) takes precedence over CCNR.

QSIG Name Display Supplementary Services

Calling Name

After a QSIG Diversion takes place, and Call Diversion Notification information is received at the diverted-to node, the calling name is displayed on the diverted-to user's phone only if this phone has a DNDD class of service. If the diverted-to user's phone has a DNDA class of service, another name or nothing is displayed.

Alerting/Connected Name

When both names from a QSIG Name Display Service and a QSIG Diversion Notification are included in a same message, the name from the QSIG Name Display Service will be ignored. Note that these names should be the same.

If a name, either from QSIG Name Display Service and QSIG Diversion Notification, is received, it will be displayed only if no subsequent subscription option forbids it.

QSIG Path Replacement

QSIG Path Replacement can be triggered after QSIG Diversion has been applied. As QSIG Diversion has its own optimization method (diversion by Rerouting), an IE indicating the method used is sent by the system during QSIG Diversion towards the diverted-to node. At the diverted-to system node, upon receipt of this IE, QSIG Path Replacement should only be started if diversion has not already optimized the call.

QSIG Call Transfer

There is no interaction when Station A and Station B are connected after a diversion, and then Station B transfers the call to Station C.

Station A and Station B are connected. Station B transfers the call to Station C, who then diverts the call to Station D. The notification depends on when Station B completes the transfer. If the transfer is completed before the diversion occurs, then the two services do not interact. The final notification is given by the Call Diversion Notification. If the diversion is initiated before the transfer has been completed, then the final notification depends on the transfer notification rules.

QSIG Alternate Routing

When a QSIG Alternate Routing is performed, the Facility IE sent with the setup message are not re-sent. Therefore, the diverted-to user does not receive the DLI2 information and is not notified that a diversion occurred.

Preventing Reciprocal Call Forward

Configuring the Preventing Reciprocal Call Forward Allowed (PVCA) option in LD 15 prevents Call Forward looping. A phone that has been call forwarded to another phone, will not have the same call-forwarded back to it.

This functionality is not supported across a QSIG network.

ISDN BRI Supplementary Services - ETSI Call Forward Unconditional

This feature implements the ETSI BRI Call Forward Unconditional feature, which allows a BRI phone to activate Call Forward Unconditional. Two Call Forward DNs can be configured, one for voice and one for data.

A BRI phone, as a served user, has the subscription option choices to notify both originating and diverted-to parties.

Note:

If the originating or diverted-to phone is a BRI phone, notification information is provided to this phone, but the treatment by the phone is subject to the manufacturer's implementation.

Attendant and Network Wide - Remote Call Forward

The Attendant and Network Wide- Remote Call Forward feature expands the phone based Remote Call Forward functionality to apply across an MCDN ISDN network, and to allow Remote Call Forward for a phone to be activated from an attendant.

This functionality cannot be used to activate QSIG Call Diversion.

Total Redirection Count

The Total Redirection Count (TRCL) prompt in LD 15 allows the maximum number of redirections for a customer to be defined between 1 and 7 (0 by default). The counter is incremented each time the call is diverted by Call Forward All Calls, Hunt, Call Forward No Answer, or Call Forward Busy, in an MCDN environment, whether on-node or off-node. The counter is reset when the call is answered or enters the attendant queue.

This feature is supported by QSIG, which transports the equivalent information.

Note that this feature is an available solution for avoiding infinite diversion loops, since Preventing Reciprocal Call Forward is not supported network-wide.

MCDN Trunk Route Optimization

Consider the case presented in Figure 83: Call from MCDN TRO Diverted to QSIG on page 802.

For Call Forward No Answer (CFNA), call (1) is cleared before call (2) starts, which is not the usual mechanism for full QSIG CFNA.

Diversion Notification Information messages are sent from originating node to the diverted-to node. As the Served User's Subscription Option is not known (as well as the name in certain conditions), it is assumed to be "Served user releases the number/name to the diverted-to user: no". As a result, Diversion Notification Information messages provided in this case only contain the diversion counter, diversion reason and eventually the original reason for diversion.

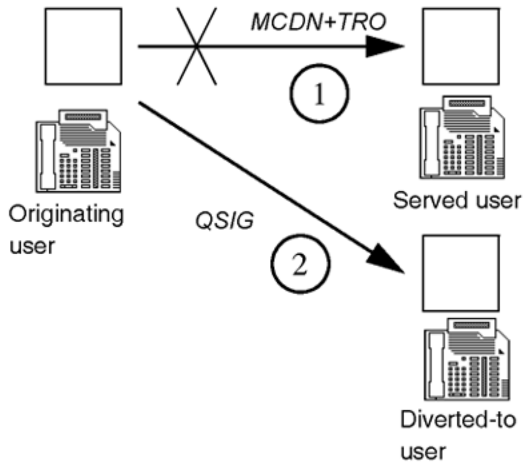


Figure 83: Call from MCDN TRO Diverted to QSIG

MCDN Trunk Route Optimization is blocked on MCDN calls due to the Rerouting method used by QSIG Diversion on No Reply, as far as the original call being kept by the originating PINX to eventually be restored for further treatment. This case arises if the diverted-to user activates TRO before alerting, that is, if Call Forwarded All Calls has been activated against the diverted-to user.

MCDN Trunk Anti Tromboning (TAT)

As the MCDN Trunk Anti Tromboning feature is activated once the call is established, there is no interaction between this feature and QSIG Call Diversion Notification.

ISDN Call Connection Limitations

QSIG Diversion is included in the Call Redirection Threshold, as part of the ISDN Call Connection Limitations feature.

Timed Reminder Recall

In case of QSIG Diversion on No Reply by the Rerouting method, if the original call is under Slow Answer Recall Timing, then the new call resulting from QSIG Diversion is also timed for Slow Answer Recall. This applies if the new diversion call is over an ETSI QSIG link, or to a local analog (500/2500-type) phone or Meridian proprietary phone, but not to a BRI phone.

The timer applied to the new diversion call is the remaining timer of the original call at the time of diversion.

Feature packaging

This feature requires the following packages:

- 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
- Integrated Services Digital Network (ISDN) package 145
- Primary Rate Access (PRI) package 146
- 2.0 Mbit/s Primary Rate Access (PRI2) package 154
- International Primary Rate Access (IPRA) package 202
- Multi-purpose Serial Data Link (MSDL) package 222
- Q-reference Signaling Point Interface (QSIG) package 263
- QSIG Generic Functional Protocol Transport (QSIG-GF) package 305
- Q-reference Signaling Point Interface Supplementary Services (QSIG-SS) package 316

For ISDN Basic Rate Interface Trunking (BRIT), the following additional packages are required:

- ISDN Basic rate Access (BRI) package 216
- ISDN Basic Rate Trunk Access (BRIT) package 233

For a QSIG/DPNSS1 Gateway, the following additional packages are required:

- Integrated Digital Access (IDA) package 122
- Digital Private Networking Signaling System No. 1 (DPNSS) package 123
- DPNSS Network Services (DNWK) package 231

For notification display, the following additional packages are required:

- Digit Display (DDSP) package 19
- Call Party Name Display (CPND) package 95

Feature implementation

Task summary list

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

Remote Capability Meanings, for PRI D-Channels and BRI routes

Table 366: Remote Capability Meanings on page 805 indicates the remote capabilities meanings for both PRI2 D-Channels and BRI routes. When using Table 366: Remote Capability Meanings on page 805 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 generally use Integer Values, whereas some older ones can 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. Treating 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 366: 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 sent	Not processed when received
DV1O	Sent coded as Object Identifier	Sent	Not processed when received
DV1I	Sent coded as Integer Value		
DV2O	Sent coded as Object Identifier	Not Sent	Processed when received
DV2I	Sent coded as Integer Value		
DV3O	Sent coded as Object Identifier	Sent	Processed when received
DV3I	Sent coded as Integer Value		

ISDN PRI implementation procedures

The following are ISDN PRI implementation procedures:

- Configure the QSIG Call Diversion Notification emote Capability for the associated D-Channel, using LD 17.
- Configure QSIG Diversion Notification Class of Service, on analog (500/2500 type) phones, using LD 10.
- Configure QSIG Diversion Notification Class of Service, on Meridian Proprietary Phones, using LD 11.
- Configure the QSIF Call Diversion Notification mnemonics, for Call Forward All Calls, Call Forward No Answer, and Call Forward Busy, using LD 95.
- Define a Calling Party Name Display name string for a phone, using LD 95.
- On a Meridian Proprietary Phone, configure whether or not the name of the originally-called party is to be displayed on the phone, if is a diverted-to phone, using LD 11.

ISDN BRIT implementation procedures

The following are the ISDN BRIT implementation procedures:

- Configure the QSIG Call Diversion Notification Remote Capability for a BRI route with QSIG Generic Functional Interface, using LD 16.
- Configure the Terminal Service Profile (TSP), to implement the QSIG Call Diversion options on BRI phones, using LD 27.

ISDN PRI implementation

Table 367: 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 for Large Systems.
	NEW DCH xx	Add a D-Channel on port 0-14 (for Small Systems).
- IFC	ESGF ISGF	ETSI QSIG interface with GF platform. ISO QSIG interface with GF platform.

Prompt	Response	Description
RLS	xx	Software release.
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).

Table 368: LD 10 - Configure QSIG Diversion Notification Class of Service on analog (500/2500-type) phones.

Prompt	Response	Description
REQ:	NEW CHG	Add new data, or change existing data.
TYPE:	500	Phone type.
...		
TN		Terminal number
	l s c u	Format for Large System and Avaya Communication Server 1000E (Avaya CS 1000E) system, where l = loop, s = shelf, c = card, u = unit.
	c u	Format for Small System, Avaya Communication Server 1000 Media Gateway 1000 B (Avaya MG 1000B), and Avaya CS 1000 Media Gateway 1000 T (Avaya MG 1000T), where c = card and 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.

Prompt	Response	Description
- - 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.

Table 369: LD 11 - Configure QSIG Diversion Notification Class of Service, on Meridian 1 Proprietary Phones.

Prompt	Response	Description
REQ	NEW	Add new data.
	CHG	Change existing data.
TYPE	xxxx	Type of phone.
TN		Terminal number
	l s c u	Format for Large System and CS 1000E system, where l = loop, s = shelf, c = card, u = unit.
	c u	Format for Small System, Media Gateway 1000B, and Media Gateway 1000T, where c = card and u = unit.
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 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.

Prompt	Response	Description
- -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.

Table 370: 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.
...		
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

Table 371: LD 95 - Define a Calling Party Name to be displayed on a receiving phone.

Prompt	Response	Description
REQ	NEW	Add new data.
TYPE	NAME	Name
CUST	xx	Customer number, as defined in LD 15.
...		
DN	x..x	Directory Number.
- NAME	a..a	Calling Party Name Display in ASCII characters.

Table 372: LD 11 - Configure Dialed Name Display Denied or Dialed Name Display Allowed of the originally-called party on the diverted-to phone.

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

Prompt	Response	Description
TN		Terminal number
	l s c u	Format for Large System and CS 1000E system, where l = loop, s = shelf, c = card, u = unit.
	c u	Format for Small System, Media Gateway 1000B, and Media Gateway 1000T, where c = card and u = unit.
CLS	(DNDD)	DNDD = Dialed Name Display Denied (the name of the originally-dialed party is not displayed on the phone).
	DNDA	DNDA = Dialed Name Display Allowed (the name of the originally-dialed party is displayed on the phone).

Note:

LD 81 can 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 can 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 Class of Service prompt.

ISDN BRIT implementation

Table 373: 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.
ROUT		Route number
	0-511	Range for Large System and CS 1000E system.
	0-127	Range for Media Gateway 1000B.
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.

Prompt	Response	Description
ISDN	YES	Integrated Services Digital Network.
- IFC	ESGF ISGF	QSIG interface type where xxxx is: Interface ID for ETSI QSIG with GF capability. 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).

Table 374: LD 27 - Configure the Terminal Service Profile (TSP) to implement the QSIG Call Diversion options on BRI phones.

Prompt	Response	Description
REQ	NEW	Add new data.
	CHG	Change existing data.
TYPE	TSP	Terminal Service Profile.
DSL	lll	DSL Terminal Number location for Large Systems: lll (superloop) = 0-156 (must be zero or a number divisible by 4)
	s cc dsl#	s (shelf) = 0-1 cc (card) = 0-15 dsl# (DSL location) = 0-7
		DSL Terminal Number location
	cc dsl	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

Prompt	Response	Description
		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).

Feature operation

No specific operating procedures are required to use this feature.

Chapter 58: ISDN QSIG Call Transfer Notification

Contents

This section contains information on the following topics:

Applicable regions

ISDN BRI trunking is not available in North America. Contact your system supplier or your Avaya representative to verify support of this product in your area.

Feature description

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 Mbit/s Primary Rate Interface (PRI2), and ISDN Basic Rate Trunking (BRIT) applications.

The QSIG Call Transfer supplementary service, 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, through 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, 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 phones 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 Telecommunication 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 system 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 375: Originating phone and transferred-to phone display possibilities on page 814 summarizes the display options pertaining to the originating phone and transferred-to phone.

Note:

The underlined Class of Service values are in effect.

Table 375: Originating phone and transferred-to phone display possibilities

Originating phone Class of Service	Originating phone display Originating phone display after receipt of transfer notification from transferred-to phone	Transferred-to phone Class of Service	Transferred-to phone display Transferred-to phone display after receipt of transfer notification from originating phone
CNDA NAMA DDGA	H6000 T Cindy	CNDA NAMA DDGA	H5000 T Betty
CNDA NAMA DDGD	H6000 T Cindy	CNDA NAMA DDGA	211-4 T Betty

CNDD NAMA DDGA	H6000	CNDA NAMA DDGA	H5000 T Betty
CNDA NAMD DDGA	H6000 T Cindy	CNDA NAMA DDGA	H5000 T XXXX
CNDA NAMA DDGA	_ _ _ _ _ T Cindy	CNDA NAMA DDGD	H5000 T Betty
CNDA NAMA DDGA	H6000 T Cindy	CNDD NAMA DDGA	H5000
CNDA NAMA DDGA	H6000 T XXXX	CNDA NAMD DDGA	H5000 T Betty

Note:

No reason is displayed if the Class Of Service CNDD is configured on the phone.

Note:

When presentation of the remote party number is not allowed (Class of Service 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 subAdressTransfer invoke APDU is not generated by the system.

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 phone display.

Connected Line Identification Presentation/Restriction

COLP/COLR applies to the originating and to the transferred-to phone 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 is 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.

QSIG Name Display Supplementary Services (QNDS)

Calling, Alerting, and 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 phone 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

This feature requires the following packages:

- Integrated Services Digital Network (ISDN) package 145
- Primary Rate Access (PRI) package 146
- 2.0 Mbit Primary Rate Access (PRI2) package 154
- International Primary Rate Access (IPRA) package 202
- Multi-purpose Serial Data Link (MSDL) package 222
- 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

For ISDN Basic Rate Interface Trunking (BRIT), the following packages are required:

- ISDN Basic rate Access (BRI) package 216
- ISDN Basic Rate Trunk Access (BRIT) package 233

For a QSIG/DPNSS1 Gateway, the following packages are required:

- Integrated Digital Access (IDA) package 122
- Digital Private Networking Signaling System No. 1 (DPNSS) package 123
- DPNSS Network Services (DNWK) package 231

For notification display, the following packages are required:

- Digit Display (DDSP) package 19
- Call Party Name Display (CPND) package 95

Feature implementation

Task summary list

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

1. Table 376: LD 15 - Configure a Private Integrated Services Network Exchange (PINX) DN for a customer. on page 819
2. Table 377: LD 17 - Configure the method of Operation Coding for the QSIG Call Transfer Notification feature for the associated QSIG D-Channel. on page 820

   The value is configured as a remote capability. It can be either by Object Identifier, used for ETSI, or by Integer Value, used for ISO.

3. Table 378: LD 95 - Configure the Reason for Redirection, and the Call Transfer mnemonic, for a customer. on page 820
4. Undefined Resource
5. Table 380: LD 10 - Configure an analog (500/2500-type) transferring phone. on page 821

   Use LD 10, so that after Call Transfer, the originating phone display and the transferred-to phone display show the name and number associated with the transferred-to phone and originating phone.

6. Table 381: LD 11 - Configure a Meridian 1 proprietary transferring phone. on page 822

   Use LD 11, so that after Call Transfer, the originating phone display and the transferred-to phone display show the name and number associated with the transferred-to phone and originating phone.

7. Table 382: LD 11 - Configure a Meridian 1 Proprietary Phone. on page 823

   Use 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.

8. Table 383: LD 16 - Configure the method of Operation Coding for the QSIG Call Transfer Notification for the associated QSIG ISDN BRI trunk route. on page 823

This section contains the procedures required to configure the QSIG Supplementary Services - Name Display Enhancement feature.

ISDN PRI implementation

Table 376: 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		Customer number

Prompt	Response	Description
	0-99	Range for Large System and Avaya Communication Server 1000E (Avaya CS 1000E) system.
	0-31	Range for Media Gateway 1000 B (Avaya MG 1000B).
- PINX_DN	xx...x	Private Integrated Services Network Exchange DN (1-7 digits).

Table 377: LD 17 - Configure the method of Operation Coding for the QSIG Call Transfer Notification feature for the associated QSIG D-Channel.

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 Large Systems.
- 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.

Table 378: LD 95 - Configure the Reason for Redirection, and the Call Transfer mnemonic, for a customer.

Prompt	Response	Description
REQ	NEW	Add new data.
	CHG	Change existing data.
TYPE	CPND	Configure the Calling Party Name Display data block.
CUST	xx	Customer number, as defined in LD 15.
...		
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.

Table 379: LD 95 - Define a name string for a customer.

Prompt	Response	Description
REQ	NEW	Add new data.
	CHG	Change existing data.
TYPE	NAME	Create a new Calling Party Name string.
CUST	xx	Customer number, as defined in LD 15.
...		
DN	xxx...x	Directory Number (1 to 7 digits) to which the name string is associated.
- NAME	a...a	CPND name, in ASCII characters.

Table 380: LD 10 - Configure an analog (500/2500-type) transferring phone.

Prompt	Response	Description
REQ:	NEW CHG	Add new data, or change existing data.
TYPE:	aaa	Type of analog phone aaa = 500 or 2500.
TN		Terminal number
	l s c u	Format for Large System and CS 1000E system, where l = loop, s = shelf, c = card, u = unit.
	c u	Format for Media Gateway 1000B, where c = card and u = unit.
CUST	xx	Customer number, as defined in LD 15.
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 phone.
	(CNDD) CNDA	Name Display (denied) allowed on this phone.

Prompt	Response	Description
	(NAMD) NAMA	Name Display (denied) allowed on other phones.
	(DDGD) DDGA	DN display (denied) allowed on other phones.
CLS	<CR>	The CLS prompt is generated until <cr> is entered.

Table 381: LD 11 - Configure a Meridian 1 proprietary transferring phone.

Prompt	Response	Description
REQ:	NEW CHG	Add new data, or change existing data.
TYPE:	a...a	Telephone type. Type **?** for a list of possible responses.
TN		Terminal number
	l s c u	Format for Large System and CS 1000E system, where l = loop, s = shelf, c = card, u = unit.
	c u	Format for Media Gateway 1000B, where c = card and u = unit.
CUST	xx	Customer number, as defined in LD 15.
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. First name. Last name.
...		
KEY	xx TRAN	Assign the Transfer key.
...		
CLS		Class of Service.
	(CNDD) CNDA	Name Display (denied) allowed on this phone.
	(NAMD) NAMA	Name Display (denied) allowed on other phones.
	(DDGD) DDGA	DN display (denied) allowed on other phones.
CLS	<CR>	The CLS prompt is generated until <cr> is entered.

Table 382: LD 11 - Configure a Meridian 1 Proprietary Phone.

Prompt	Response	Description
REQ:	NEW CHG	Add new data, or change existing data.
TYPE:	a...a	Telephone type. Type **?** for a list of possible responses.
TN		Terminal number
	l s c u	Format for Large System and CS 1000E system, where l = loop, s = shelf, c = card, u = unit.
	c u	Format for ,Media Gateway 1000B, where c = card and u = unit.
CUST	xx	Customer number, as defined in LD 15.
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 phone.
	(NAMD) NAMA	Name Display (denied) allowed on other phones.
	(DDGD) DDGA	DN display (denied) allowed on other phones.
CLS	<CR>	The CLS prompt is generated until <cr> is entered.

ISDN BRIT implementation

Table 383: 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).

Prompt	Response	Description
TYPE	RDB	Route Data Block.
CUST	xx	Customer number, as defined in LD 15.
ROUT		Route number
	0-511	Range for Large System and CS 1000E system.
	0-127	Range for Media Gateway 1000B.
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.

Feature operation

No specific operating procedures are required to use this feature.

Chapter 59: ISDN QSIG/ETSI GF Enhancement

Contents

This section contains information on the following topics:

Applicable regions

ISDN BRI trunking is not available in North America. Contact your system supplier or your Avaya representative to verify support of this product in your area.

Feature description

The Integrated Services Digital Network ISDN (ISDN) Q-reference Signaling Point (QSIG)/ European Telecommunication Standards Institute (ETSI) Generic Functional Transport (GFT) was introduced 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 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 to provide improvements in call control, to allow the supported QSIG signaling protocols to work together seamlessly. 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) can 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.

  The method of calculating the available length for an APDU in basic QSIG call control messages is now 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.

  The method of calculating the available length for an APDU in basic ETSI call control messages is now 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.

The following operating parameters apply to the ISDN QSIG 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 system wants 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 ETSI 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 can discard it. The MSI can be passed to the next node only if the transit PINX is a system.
- 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 system wants 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 can be rejected by the far end due to the unavailability of system resources, and due to overdo control. A notification message is sent to the supplementary service, so that it can 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 can 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 can 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 system. 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 associated with this feature.

Feature packaging

This feature requires the following packages:

- Integrated Services Digital Network (ISDN) package 145
- Primary Rate Access (PRA) package 146 for 1.5 Mbit signaling
- International Primary Rate Access (IPRA) package 202
- 2.0 Mbit Primary Rate Access (PRI2) package 154 for 2.0 Mbit signaling
- Multi-purpose Serial Data Link (MSDL) package 222
- Q-reference Signaling Point Interface (QSIG) package 263
- QSIG Generic Functional Protocol Transport (QSIG-GF) package 305

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) package 216
- ISDN Basic Rate Trunk Access (BRIT) package 233

Feature implementation

There are no specific implementation procedures for this feature.

Feature operation

No specific operating procedures are required to use this feature.

Chapter 60: ISDN QSIG/EuroISDN Call Completion

Contents

This section contains information on the following topics:

Applicable regions

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

Feature description

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 system 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 system 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 signaling 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 phone level basis, either by Class of Service or Ring Again key, a phone 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 phones.

Gateways between EuroISDN CCBS, MCDN Network Ring Again on Busy and DPNSS1 Call Back When Free are supported.

A Private Integrated Services Network Exchange (PINX) 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 phone 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 phone. 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) phone, Call Completion notification waits until the phone has finished an active call. If Call Waiting is configured on a phone, notification is presented after the Call Waiting call. If an additional call is queued while Ring Again free notification is waiting on a phone, the waiting call takes precedence over the Call Completion notification. An established Call Completion call is also queued if a phone 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 phone with Do Not Disturb (DND) activated. Call Completion requests can be applied to phones with the DND feature activated. However, this request does not advance until the DND feature is deactivated.

EuroISDN Trunk - Network Side

Call Completion to Busy Subscriber is supported on a EuroISDN Trunk - Network Side connectivity interface.

Call Completion on No Reply 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.

Flexible Feature Codes

Analog (500/2500-type) phone 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 phone 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 phone in Maintenance Busy state.

Make Set Busy

Phones 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 phone is considered busy while MSB is active. A CCBS request is registered against a busy phone, but only advances when the MSB feature is deactivated and the phone remains free.

Permanent Hold

Analog (500/2500 type) phones with Permanent Hold cannot use the Ring Again functionalities.

Ring Again

Ring Again No Answer

Analog (500/2500 type) phones can have only one Call Completion to Busy Subscriber request at a given time. Meridian 1 Proprietary Phones can make Ring Again requests based on the number of Ring Again keys programmed on a phone.

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
- Primary Rate Interface (PRI) package 146, or
- 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
- 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
- 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
- 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
- EuroISDN (EURO) package 261

Feature implementation

Task summary list

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

1. Table 384: LD 17 - Configure Remote Capabilities to Call Completion for QSIG or EuroISDN on Primary Rate Interface. on page 837
2. Table 385: LD 16 - Configure Remote Capability to Call Completion QSIG or EuroISDN on Basic Rate Interface. on page 837
3. Table 386: LD 10 - Add or Change Ring Again on analog (500/2500-type) phones. on page 838
4. Table 387: LD 11 - Add or Change Ring Again on Meridian 1 proprietary phones. on page 838
5. Table 388: LD 57 - Define Ring Again Flexible Feature Codes (FFCs) and FFC Confirmation Tone. on page 839
6. Table 389: LD 15 - Configure a Private Integrated Services Network Exchange (PINX) DN. on page 839

Table 384: 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.

Table 385: 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

Prompt	Response	Description
...		
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.

Table 386: LD 10 - Add or Change Ring Again on analog (500/2500-type) phones.

Prompt	Response	Description
REQ:	NEW CHG	Add new data, or change existing data.
TYPE:	500	Type of phone.
TN		Terminal number
	l s c u	Format for Large System and Avaya Communication Server 1000E (Avaya CS 1000E) system, where l = loop, s = shelf, c = card, u = unit.
	c u	Format for Media Gateway 1000 B (Avaya MG 1000B), where c = card and u = unit.
CLS	XRA	Ring Again allowed. XRD = Ring Again denied (default).

Table 387: LD 11 - Add or Change Ring Again on Meridian 1 proprietary phones.

Prompt	Response	Description
REQ:	NEW CHG	Add new data, or change existing data.

Prompt	Response	Description
TYPE:	a...a	Telephone type. Type **?** for a list of possible responses.
TN		Terminal number
	l s c u	Format for Large System and CS 1000E system, where l = loop, s = shelf, c = card, u = unit.
	c u	Format for Media Gateway 1000B, where c = card and u = unit.
KEY	xx RGA	Assign Ring Again key where xx = key number. On M2317 phones, xx = 27.

Table 388: 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, as defined in LD 15.
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.

Table 389: LD 15 - Configure a Private Integrated Services Network Exchange (PINX) DN.

Prompt	Response	Description
REQ:	CHG	Change
TYPE:	NET	Networking
CUST		Customer number
	0-99	Range for Large System and CS 1000E system.
	0-31	Range for Media Gateway 1000B.
- PINX_DN	xx...x	Node DN to a maximum of seven digits with DN extension.

Feature operation

Call Completion on Busy Subscriber

Meridian 1 Proprietary Phones

1. First, Phone A calls Phone B on an outgoing QSIG/EuroISDN route and receives a busy indication.
2. Before releasing this call, Phone 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, Phone A's RGA lamp key is steadily lit, and the DN lamp darkens.
4. Then Phone A awaits notification that Phone B is free. Phone A can make or receive calls while waiting for notification.
5. When Phone B goes on-hook, Phone 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. Phone A has thirty seconds to accept the recall before it is cancelled.
7. If Phone A accepts the recall and Phone B is free, the call is rung. However, if Phone B has either received or originated another call, Phone A receives a busy indication. Phone A has the option of requesting CCBS again.

Note:

If Phone A presses the RGA key before receiving notification, the request is cancelled.

Analog (500/2500-type) phones without RGA key

1. First, Phone A calls Phone B on an outgoing QSIG/EuroISDN route and receives a busy indication.
2. Then Phone A performs a switch-hook flash and a special dial tone is heard.
3. Next, Phone 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. Finally, Phone A goes on-hook and is free to make or receive other calls.
6. When Phone B goes on-hook, Phone A receives a burst of tone and has thirty seconds to accept the recall before it is cancelled.
7. If Phone B has originated or received another call, Phone A receives a busy indication and has the option of requesting the CCBS again.

Note:

If Phone A presses the RGA key before receiving notification then the request is cancelled.

Call Completion No Response

Meridian 1 Proprietary Phones

1. First, Phone A calls Phone B on an outgoing QSIG route. The call rings but is not answered. Phone A receives a ringback tone.
2. Then Phone A presses RGA key to activate Ring Again on No Answer prior to releasing the call. The DN key lamp darkens.
3. Next, Phone A goes on-hook and is free to make or receive calls.
4. When Phone B makes a call and goes on-hook. Phone A is notified by a one second burst of tone and the RGA lamp fast flashes.
5. Finally, Phone A can accept the call by selecting a DN key and pressing the RGA key. Phone A has thirty seconds to accept this call.
6. If Phone A accepts the call and Phone B is free, the call goes through. However, if Phone B receives or originates another call, Phone A receives a busy indication. Phone A has the option of requesting CCBS.

Note:

If Phone A presses the RGA key before receiving notification then the request is cancelled.

Chapter 61: ISDN QSIG/EuroISDN Call Completion Enhancement

Contents

This section contains information on the following topics:

Applicable regions

ISDN BRI trunking is not available in North America. Contact your system supplier or your Avaya representative to verify support of this product in your area.

Feature description

The ISDN QSIG/EuroISDN Call Completion was introduced, 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 system notices subsequent activity on the called extension, that has again become idle.

With enhancement, 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 Telecommunication 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 phones.

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 for each phone, using Class of Service for analog (500/2500-type) phones or defining a Ring Again key on Meridian Proprietary Phones, it is not possible to configure a phone 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 system. At the originating side, although both methods are possible, the system 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 system. At the originating side, although both methods are possible, the system 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 busy again after the originating party responds to a call completion recall, there are two methods of behavior that the system takes. 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 system does not support The Request Retention method.

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, the same as a normal call.

Call Forward

When a CCBS or CCNR request is received on a phone which has been call forwarded, the CCBS or CCNR request is registered against the DN to which the phone 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-type) phone after an active call has been completed. If there is a call waiting on the phone, the Call Completion free notification is presented to the phone after the call waiting call has been completed. If another call waiting occurs while a Ring Again free notification is waiting on the phone, the new call waiting presentation takes precedence.

A Call Completion call can wait to be established on a phone 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 can 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 can 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 phone 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 system, all Call Completion requests at the system are deleted. If the requesting party is not on the system, 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 phone in maintenance busy state.

Make Set Busy

If a phone with Make Set Busy active makes a Call Completion request to another station, then the Call Completion free notification overrides the Make Set Buy 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-type) phones with a call on permanent hold cannot use the Ring Again feature.

Pretranslation

Pretranslation can be used in conjunction with a Call Completion request.

Ring Again/Ring Again No Answer

Analog (500/2500-type) phones can only have one Call Completion request active at a time. Meridian Proprietary Phones can 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 Operating parameters on page 844, 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 on how to connect to the originating party.

This only applies 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.

Hunt and Group Hunt

Hunt and Group Hunt is not supported by the QSIG/EuroISDN Call Completion feature.

ISDN Basic Rate Interface

QSIG/EuroISDN Call Completion is not supported on ISDN BRI phones. CCBS and CCNR cannot neither be activated or received on a ISDN BRI phone.

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 PRI:

- Ring Again (RGA) package 1
- Digit Display (DDSP) package 19
- Integrated Services Digital Network (ISDN) package 145
- Primary rate Access (PRI) package 146
- 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
- Multi-purpose Serial Data Link (MSDL) package 222
- Q-reference Signaling Point Interface (QSIG) package 263
- QSIG Generic Functional Protocol Transport (QSIG-GF) package 305
- QSIG Supplementary Services (QSIG-SS) package 316
- Coordinated Dialing Plan (CDP) package 59, and one of the following:
    - Network Alternate Route Selection (NARS) package 58
    - Basic Alternate Route Selection (BARS) package 57

For ISDN BRI trunking:

- Digit Display (DDSP) package 19
- Integrated Services Digital Network (ISDN) package 145
- Overlap Signaling (OVLP) package 184 (optional)
- ISDN Basic rate Access (BRI) package 216
- ISDN Basic Rate Trunk Access (BRIT) package 233
- Q-reference Signaling Point Interface (QSIG) package 263

The following packages are required for the EuroISDN component of the QSIG/EuroISDN Call Completion Enhancement feature:

For ISDN PRI2:

- Ring Again (RGA) package 1
- Digit Display (DDSP) package 19
- Coordinated Dialing Plan (CDP) package 59 or Network Alternate Route Selection (NARS) package 58, or Basic Alternate Route Selection (BARS) package 57
- Integrated Services Digital Network (ISDN) package 145
- 2.0 Mbit Primary Rate Access (PRI2) package 154

- Network Attendant Service (NAS) package 159 (for MCDN to EuroISDN gateways)
- Flexible Numbering Plan (FNP) package 160
- ISDN International Supplementary Features (ISDN SUPP) package 161
- Overlap Signaling (OVLP) package 184 (optional)
- International Primary Rate Access (IPRA) package 202
- Multi-purpose Serial Data Link (MSDL) package 222
- ETSI Supplementary Services (ETSI-SS) package 323

For ISDN BRI trunking:

- Digit Display (DDSP) package 19
- Integrated Services Digital Network (ISDN) package 145
- ISDN International Supplementary Features (ISDN SUPP) package 161
- Overlap Signaling (OVLP) package 184 (optional)
- International Primary Rate Access (IPRA) package 202
- ISDN Basic rate Access (BRI) package 216
- ISDN Basic Rate Trunk Access (BRIT) package 233
- ETSI Supplementary Services (ETSI-SS) package 323

Feature implementation

Task summary list

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

This section contains the procedures required to configure the Call Completion to a Busy Subscriber (CCBS) over QSIG and EuroISDN networks, and Call Completion on No Response (CCNR) over a QSIG network, on a system.

ISDN PRI implementation

ETS standards considerations

When configuring the interface type, consider the following:

- In order to support countries that have not yet upgraded to the ETS 300 403 standard (as introduced by the EuroISDN ETS 300 403 Compliance Update feature), the system 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, such as 7kHz telephony and Videotelephony.

A user can still configure an interface fully compliant with the ETS 300 102 standard if IFC = EURO and CNTY = any of the supported country values.

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 generally use Integer Values (RCAP = CCBI or CCNI), whereas some older versions can use Object Identifies (RCAP = CCBO or CCNO).

Table 390: 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.

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
...		
- 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 ETS standards considerations on page 853.
- - 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 ETS standards considerations on page 853.
	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 ETS standards considerations on page 853.
		Enter country pertaining to EuroISDN interface (this prompt is generated if IFC = EURO).
	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).
...		
- RCAP		Enter the Operation Coding method for the QSIG/ETSI CCBS supplementary service.

Prompt	Response	Description
	CCBO CCBI XCCB	Coding by Object ID. Coding by Integer Value. Remove the CCBO or CCBI value. **Note:** CCBO and CCBI are mutually exclusive.

Note:

The value is configured as a remote capability. It can be either by Object Identifier, used for ETSI, or by Integer Value, used for ISO.

Table 391: LD 17 - Configure the method of Operation Coding for the Call Completion on No Reply supplementary service for the associated QSIG D-Channel.

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
...		
- 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.

Table 392: LD 10 - Configure Ring Again Class of Service on analog (500/2500-type) phones.

Prompt	Response	Description
REQ:	NEW CHG	Add new data, or change existing data.
TYPE:	500	Type of phone.
TN		Terminal number
	l s c u	Format for Large System and Avaya Communication Server 1000E (Avaya CS 1000E) system, where l = loop, s = shelf, c = card, u = unit.

Prompt	Response	Description
	c u	Format for Media Gateway 1000 B (Avaya MG 1000B), where c = card and 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.
CLS	(XRD) XRA	Deny or allow Ring Again Class of Service.

Table 393: LD 15 - Configure a Special Prefix Number (SPRE) for a customer.

Prompt	Response	Description
REQ:	CHG	Change existing data.
TYPE:	FTR	Features and options data.
CUST		Customer number
	0-99	Range for Large System and CS 1000E system.
	0-31	Range for Media Gateway 1000B.
SPRE	xxxx	Special Prefix Number (xxxx = 1-4digits).

Table 394: 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		Customer number
	0-99	Range for Large System and CS 1000E system.
	0-31	Range for Media Gateway 1000B.
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.

Table 395: 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.
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.

Table 396: 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 Large Systems.
- IFC	SL1	MCDN interface.
- NASA	YES	Allow Network Attendant Service.

ISDN BRI implementation

Table 397: 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, as defined in LD 15.

Prompt	Response	Description
ROUT		Route number
	0-511	Range for Large System and CS 1000E system.
	0-127	Range for Small System, Media Gateway 1000B, and Media Gateway 1000T.
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 ETS standards considerations on page 853.
- - 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 ETS standards considerations on page 853.
	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 ETS standards considerations on page 853.
		Enter country pertaining to EuroISDN interface (this prompt is generated if IFC = EURO).
	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).
...		
RCAP		Enter the Operation Coding method for the QSIG/ETSI CCBS supplementary service.
	CCBO CCBI XCCB	Coding by Object ID. Coding by Integer Value. Remove the CCBO or CCBI value.

Prompt	Response	Description
		Note: CCBO and CCBI are mutually exclusive.

Table 398: 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.
ROUT		Route number
	0-511	Range for Large System and CS 1000E system.
	0-127	Range for Small System, Media Gateway 1000B, and Media Gateway 1000T.
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.

Feature operation

No specific operating procedures are required to use this feature.

Chapter 62: ISDN QSIG Generic Functional Transport

Contents

This section contains information on the following topics:

Applicable regions

ISDN BRI trunking is not available in North America. Contact your system supplier or your Avaya representative to verify support of this product in your area.

Feature description

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 Telecommunication Standard Institute (ETSI QSIG):

- ISDN 1.5 Mbit/s Primary Rate Interface (PRI)
- ISDN 2 Mbit/s Primary Rate Interface (PRI2)
- ISDN Basic Rate Interface (BRI) Trunk
- 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 associated with this feature.

Feature packaging

The following packages are required for ISDN QSIG GF Transport:

- ISDN QSIG GF Transport (QSIGGF) is package 305

The following packages are required for QSIG on the ISDN PRI interface:

- Integrated Services Digital Network (ISDN) package 145
- Primary Rate Access (PRI) package 146
- 2.0 Mbit Primary Rate Interface (PRI2) package 154
- International Primary Rate Access (IPRA) package 202
- Multi-purpose Serial Data Link (MSDL) package 222
- QSIG Interface (QISG) package 263
- QSIG GF Transport (QSIGGF) package 305

The following packages are required for ISDN Signaling Link (ISL) interface:

- Integrated Service Digital Network Signaling Link (ISL) package 147
- Multi-purpose Serial Data Link (MSDL) package 222
- QSIG Interface (QSIG) package 263
- QSIG GF Transport (QSGIGF) package 305

The following packages are required for ISL shared mode:

- Integrated Services Digital Network (ISDN) package 145
- Primary Rate Access (PRI) package 146
- 2.0 Mbit Primary Rate Interface (PRI2) package 154

Feature implementation

Task summary list

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

1. Table 399: LD 16 - Configure the Route Data Block. on page 863
2. Table 400: LD 17 - Configuration Record. on page 864

Table 399: LD 16 - Configure the Route Data Block.

Prompt	Response	Description
REQ	NEW	Add new data.
	CHG	Change existing data.
TYPE	RDB	Route Data Block.
...		
DTRK	YES	Digital Trunk Route.
...		
DGTP	PRI PRI2 BRI	1.5 Mbit PRI. 2.0 Mbit PRI. 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.

Table 400: LD 17 - Configuration Record.

Prompt	Response	Description
REQ	CHG	Change
TYPE	ADAN	Action Device and Number
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.

Feature operation

No specific operating procedures are required to use this feature.

Chapter 63: ISDN QSIG Name Display

Contents

This section contains information on the following topics:

Applicable regions

ISDN BRI trunking is not available in North America. Contact your system supplier or your Avaya representative to verify support of this product in your area.

Feature description

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 phone. 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 phone. 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

The 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 phone.

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 phone.

For ISDN BRI phones 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 phones. Therefore, CNIR on a per-call basis is not supported on BRI phones.

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 phone. Name display information is not updated on the calling party's phone. 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 phone that picks up the call. If the incoming QSIG call has presentation denied, the calling party's name is not displayed on the phone 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 phone 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 phone. If the incoming QSIG call has presentation denied, name information is not displayed to the destination phone.

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 phone 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 phone 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 phone. With presentation restriction, the calling party's name information is not displayed to the destination phone.

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 phone. If presentation restricted is defined, then name information is not displayed on the destination phone.

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.

Hunt

When an incoming QSIG call with name display presentation allowed is hunted locally, the calling party's name information is displayed on the destination phone. With presentation restriction, the calling party's name information is not displayed.

Hunt - 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 phone. If presentation restricted is defined, then name information is not displayed on the forwarded phone.

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

This feature requires the following packages:

- Digit Display (DDSP) package 19
- Calling Party Name Display (CPND) package 95
- International Supplementary Features (SUPP) package 131
- Flexible Feature Codes (FFC) package 139
- Calling Party Privacy (CPP) package 301;
- Integrated Services Digital Network (ISDN) package 145
- Primary Rate Access (PRI) package 146
- 2.0 Mbit Primary Rate Interface (PRI2) package 154
- International Primary Rate Access (IPRA) package 202
- Multipurpose Serial Data Link (MSDL) package 222
- ISDN Basic Call QSIG (QSIG) package 263
- QSIG GF Transport (QSIGGF) package 305

The following packages are required for a Basic Rate Interface QSIG Interface:

- Integrated Services Digital Network (ISDN) package 145
- Basic Rate Interface (BRI) package 216
- Multipurpose Serial Data Link (MSDL) package 222
- ISDN BRI Trunk Access (BRIT) package 233

- ISDN Basic Call QSIG (QSIG) package 263
- QSIG GF Transport (QSIGGF) package 305

Feature implementation

Task summary list

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

Note:

Call Party Name Display is also defined at the phone level in either LD 10 or LD 11 depending on type of phone.

Table 401: LD 95 - Modify the Call Party Name Display (CPND) data block.

Prompt	Response	Description
REQ	NEW	Create Call Party Name Display data block.

Prompt	Response	Description
TYPE	CPND	Call Party Name Display data block.
CUST	xx	Customer number, as defined in LD 15.
....		
REQ	NEW	Create new name strings.
TYPE	NAME	Name Strings.
CUST	xx	Customer number, as defined in LD 15.
...		
DN	xxxx	Directory Number.

Table 402: LD 10 - Define Class of Service Name Display Allowed on analog (500/2500-type) phones.

Prompt	Response	Description
REQ:	NEW	Add new data.
	CHG	Change existing data.
TYPE:	500	Type of phone.
CUST	xx	Customer number, as defined in LD 15.
...		
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.

Table 403: LD 11 - Define Class of Service Name Display Allowed on Meridian 1 Proprietary Phones.

Prompt	Response	Description
REQ:	NEW CHG	Add new data, or change existing data.
TYPE:	a...a	Telephone type. Type **?** for a list of possible responses.
CUST	xx	Customer number, as defined in LD 15.
...		
CLS	CNDA	Call Party Name Display Allowed. Allows display of original DN name on redirection. (CNDD) = Call Party Name Display Denied (default).

Prompt	Response	Description
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.

Table 404: LD 12 - Define Class of Service Name Display Allowed on Attendant Consoles.

Prompt	Response	Description
REQ:	NEW CHG	Add new data, or change existing data.
TYPE:	2250	Attendant Console type.
TN		Terminal number
	l s c u	Format for Large System and Avaya Communication Server 1000E (Avaya CS 1000E) system, where l = loop, s = shelf, c = card, u = unit.
	c u	Format for Media Gateway 1000 B (Avaya MG 1000B), where c = card and u = unit.
CPND	CNDA	Call Party Name Display Allowed. (CNDD) = Call Party Name Display Denied (default).

Table 405: LD 27 - Define data for ISDN BRI phones.

Prompt	Response	Description
REQ	NEW	Add new data.
	CHG	Change existing data.
TYPE	TSP	Terminal Service Profile.
DSL	l s c dsl c dsl	Digital Subscriber Loop. For Large Systems .
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.

Table 406: LD 16 - QSIG BRI route configuration.

Prompt	Response	Description
REQ	CHG	Change.
TYPE	RDB	Route Data Block.
CUST	xx	Customer number, as defined in LD 15.
...		
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.

Table 407: 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, as defined in LD 15.
...		
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.

Table 408: 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.
...		

Prompt	Response	Description
- RCAP	NDS	Allow Name Display Services (NDS) as new remote capability.

Table 409: 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.

Table 410: LD 57 - Define the Calling Party Privacy Flexible Feature Code for Meridian 1 Proprietary and analog (500/2500-type) phones.

Prompt	Response	Description
REQ	NEW	Add new data.
	CHG	Change existing data.
TYPE	FFC	Flexible Feature Code data block.
CUST	xx	Customer number, as defined in LD 15.
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.

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.

Chapter 64: ISDN QSIG Name Display Enhancement

Contents

This section contains information on the following topics:

Applicable regions

ISDN BRI trunking is not available in North America. Contact your system supplier or your Avaya representative to verify support of this product in your area.

Feature description

The ISDN QSIG Supplementary Service - Name Display feature, 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 phone, as configured in LD 10 or LD 11. If the Class of Service prompt is configured as Name Presentation Allowed (NAMA), the name of the calling party is displayed on the phone of the terminating party. If the Class of Service 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 phones, 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.

  CNOP is based on the Class of Service of the terminating phone, as configured in LD 10 or LD 11. If the Class of Service prompt is configured as Name Presentation Allowed (NAMA), the name of the called party is displayed on the phone of the calling party. If the Class of Service 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 can be activated for all calls by setting the Class of Service prompt in LD 10 or LD 11 to Name Presentation Denied (NAMD). If the Class of Service has been set to Name Presentation Allowed (NAMA), CNIR can be activated for each call, by dialing the Calling Party Privacy Flexible Code before making a call.

Note:

The capability of activating CNIR on ISDN BRI phones is not supported.

These Name Display supplementary services are provided across an ISDN QSIG network using the QSIG Generic Functional (GF) protocol transport platform.

The QSIG Supplementary Service - Name Display Enhancement feature, provides the option of choosing the method of Operation Coding for the Name Display supplementary services described above. The choice can be by Object Identifier, used for European Telecommunication 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 can 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 OVLR prompt is set to YES in LD 17).

Feature interactions

Call Pickup (over a QSIG link)

Referring to Figure 84: Call Pickup over a QSIG link on page 880, 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.

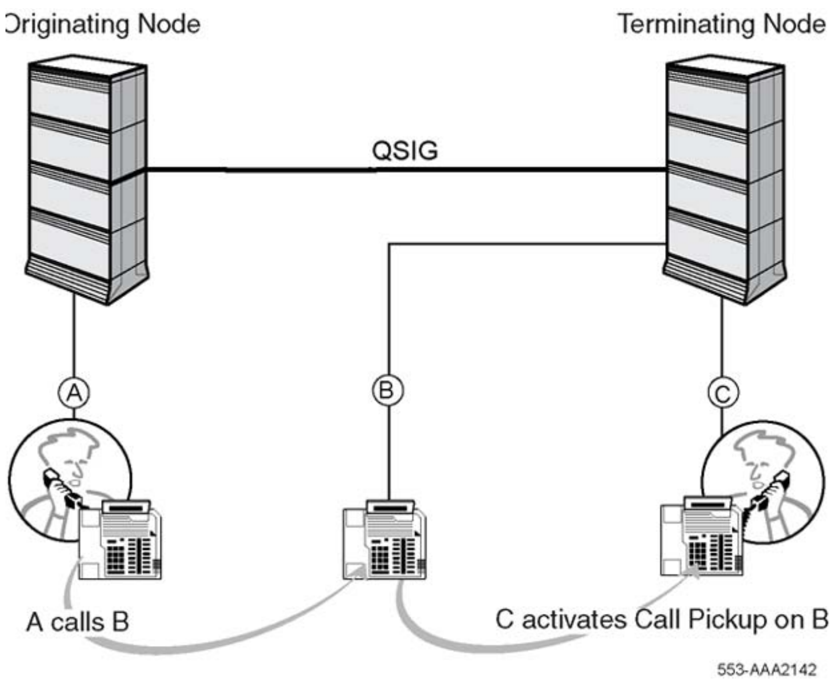


Figure 84: Call Pickup over a QSIG link

Call Transfer (over a QSIG link)

Referring to Figure 85: Call Transfer over a QSIG link on page 881, 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.

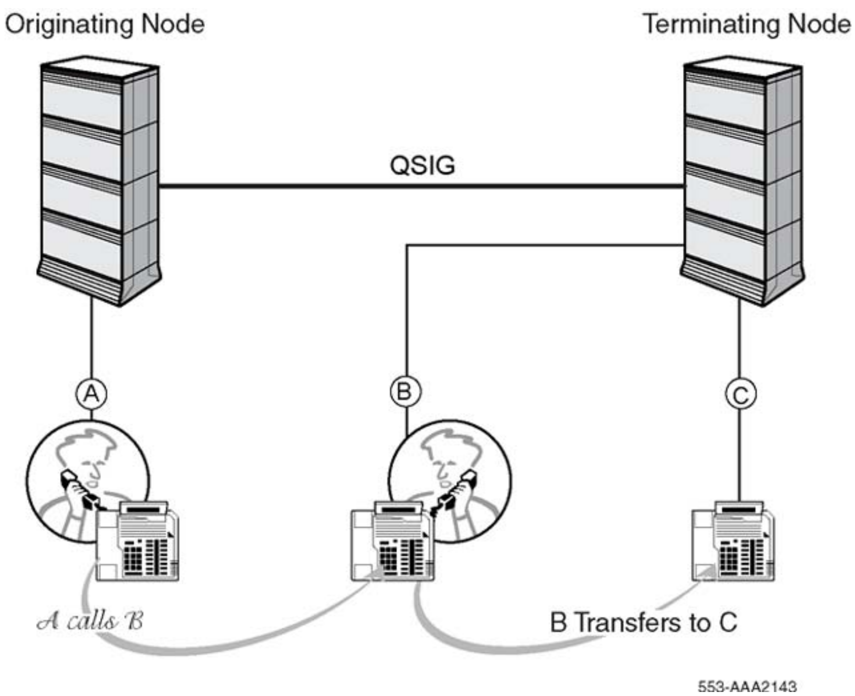


Figure 85: Call Transfer over a QSIG link

Conference (over a QSIG link)

Referring to Figure 86: Conference over a QSIG link on page 882, 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.

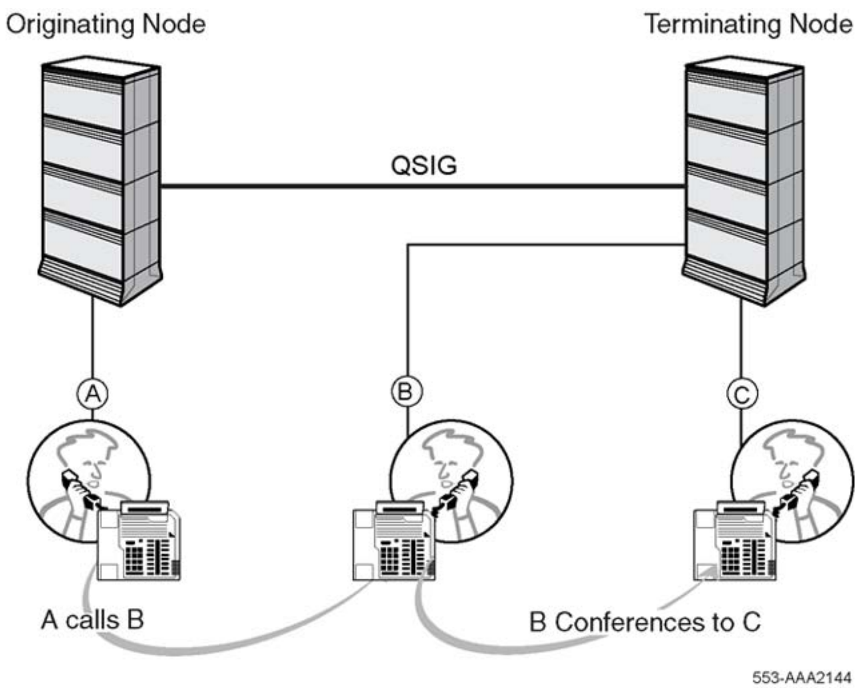


Figure 86: Conference over a QSIG link

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 87: Call Forward All Types over a QSIG link on page 883, 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 phone rings before Station C's phone (during Call Forward No Answer), then Station A's display will be updated with Station C's name information, once

Station C's phone rings. In the case where Station C's phone 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.

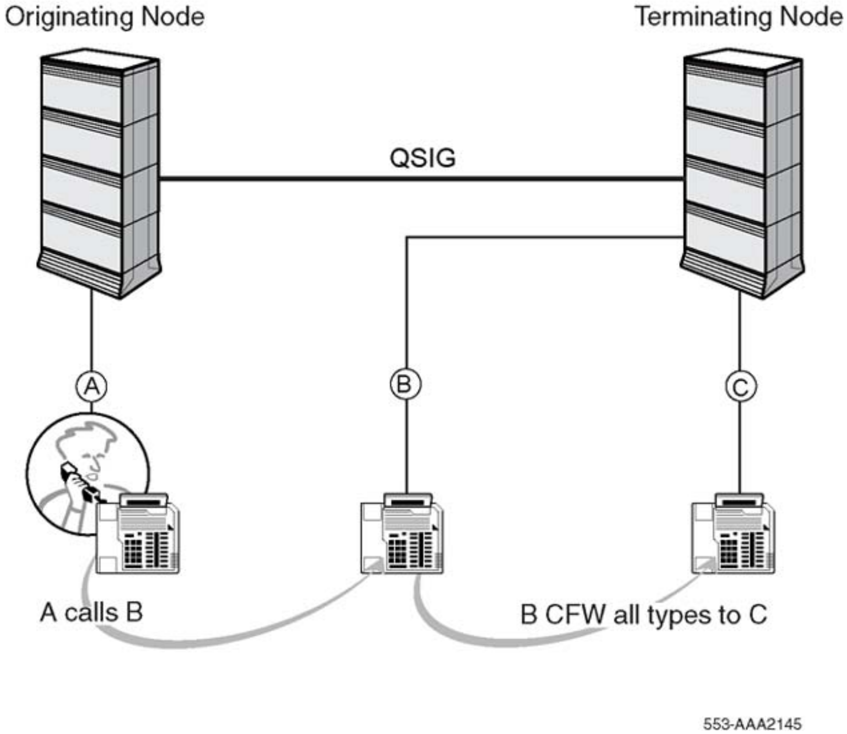


Figure 87: Call Forward All Types over a QSIG link

Hunt (over a QSIG link)

Referring to Figure 88: Hunt over a QSIG link on page 884, 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 phone 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 phone is ringing).

Station A's display will be updated with the Station C's name information, while Station C's phone is ringing. The reason for the call modification (Hunt) will not be presented to Station A.

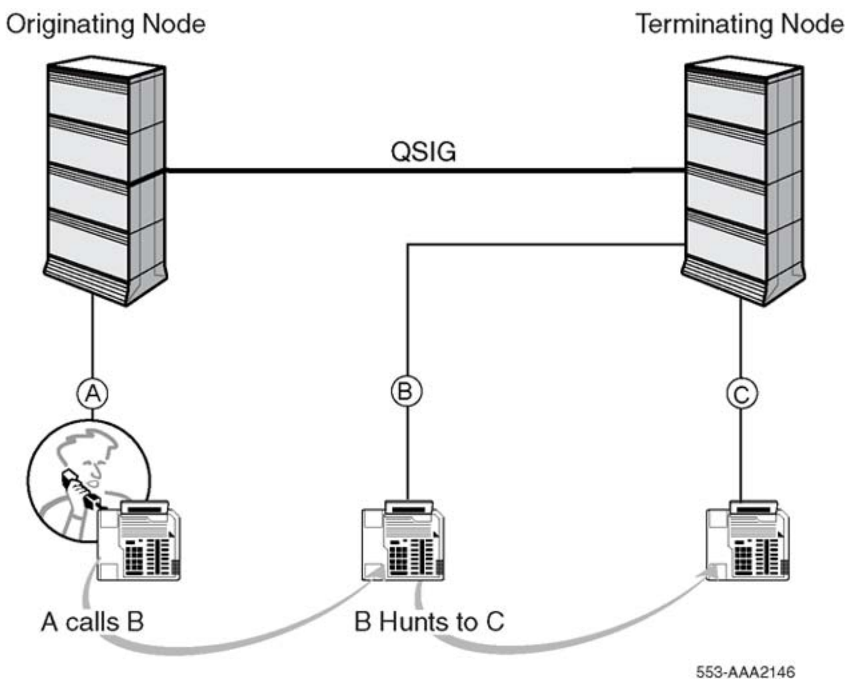


Figure 88: Hunt over a QSIG link

Incoming Digit Conversion (over a QSIG link)

Referring to Figure 89: Incoming Digit Conversion over a QSIG link on page 885, 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.

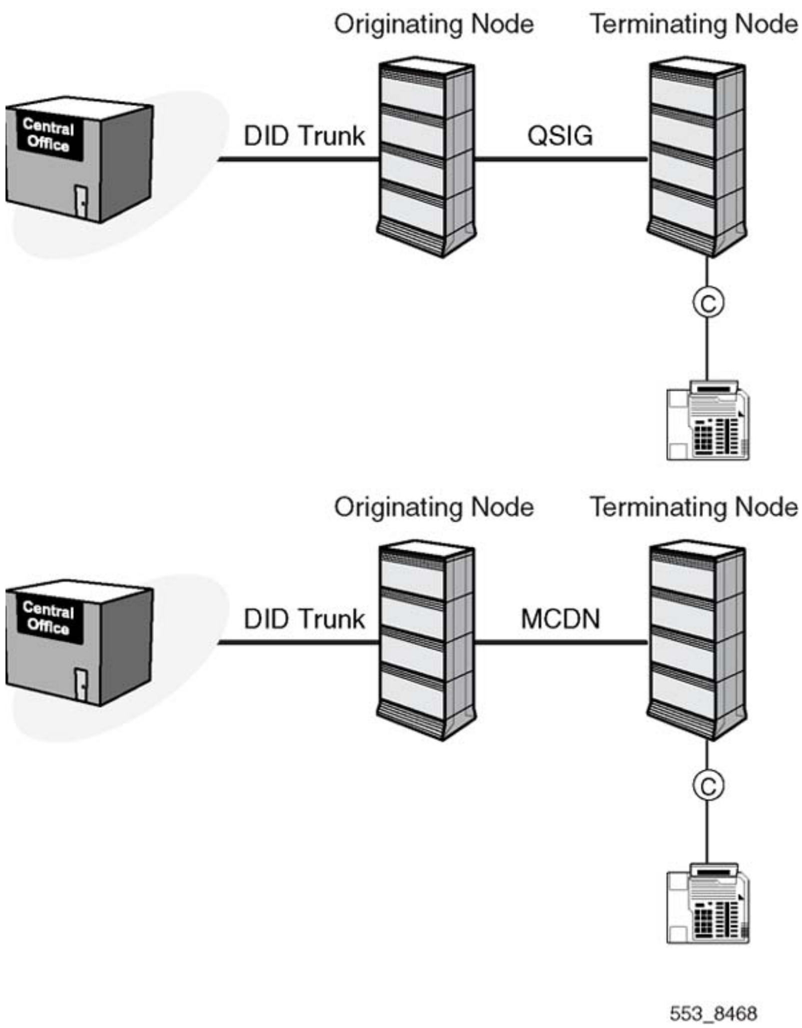


Figure 89: Incoming Digit Conversion over a QSIG link

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 90: Call Forward All Calls over a QSIG/MCDN link on page 886, 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 phone rings before Station C's phone (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 phone rings. In the case where Station C's phone 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.

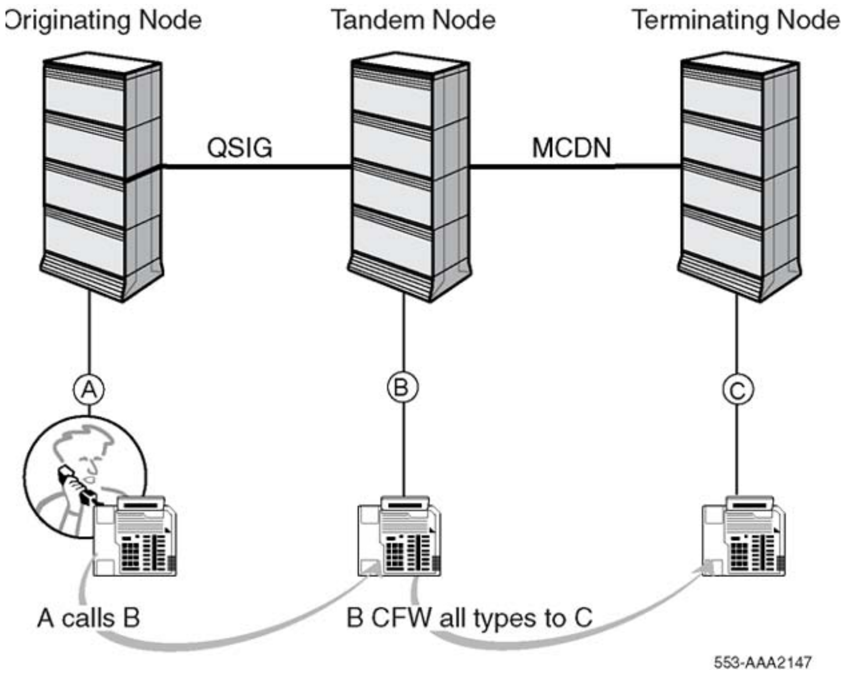


Figure 90: Call Forward All Calls over a QSIG/MCDN link

Hunt (QSIG/MCDN link)

Referring to Figure 91: Hunt over a QSIG/MCDN link on page 887, 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 phone 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 phone is ringing). Station A's display will be updated with the Station C's name information, while Station C's phone is ringing. The reason for the call modification (Hunt) will not be presented to Station A.

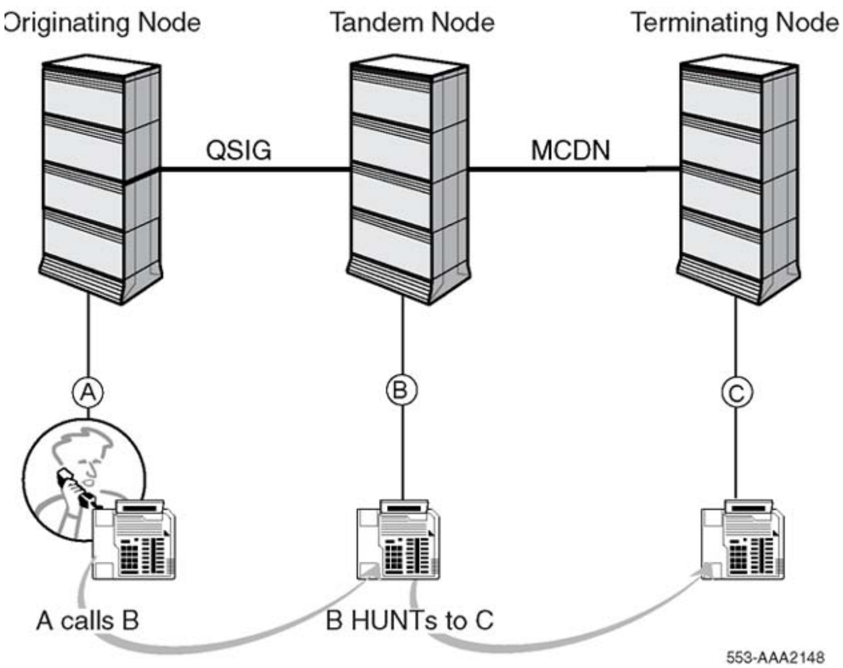


Figure 91: Hunt over a QSIG/MCDN link

Call Pickup Network Wide (QSIG/MCDN link)

Referring to Figure 92: Call Pickup Network Wide over a QSIG/MCDN link on page 888, 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.

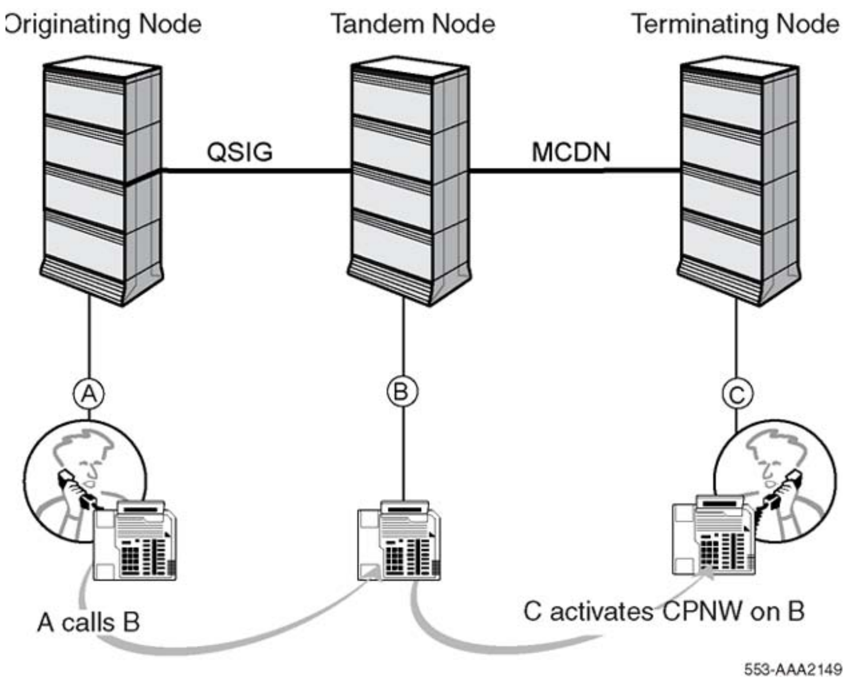


Figure 92: Call Pickup Network Wide over a QSIG/MCDN link

Call Transfer (QSIG/MCDN link)

Referring to Figure 93: Call Transfer over a QSIG/MCDN link on page 889, 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.

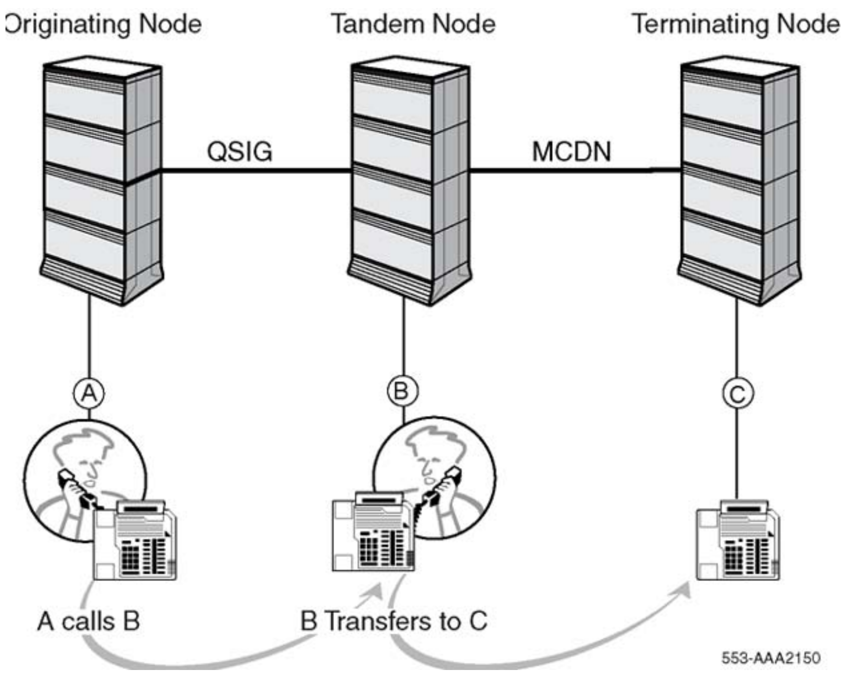


Figure 93: Call Transfer over a QSIG/MCDN link

Conference (QSIG/MCDN link)

Referring to Figure 94: Conference over a QSIG/MCDN link on page 890, 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.

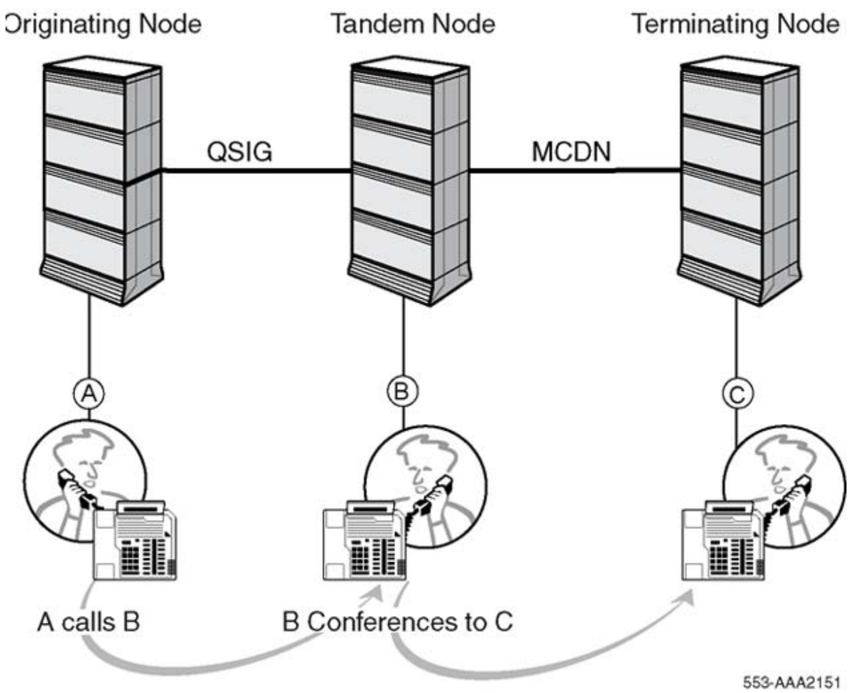


Figure 94: Conference over a QSIG/MCDN link

Incoming Digit Conversion (QSIG/MCDN or MCDN/QSIG link)

Referring to Figure 95: Incoming DID Digit Conversion over a QSIG/MCDN, or MCDN/QSIG link on page 891, 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.

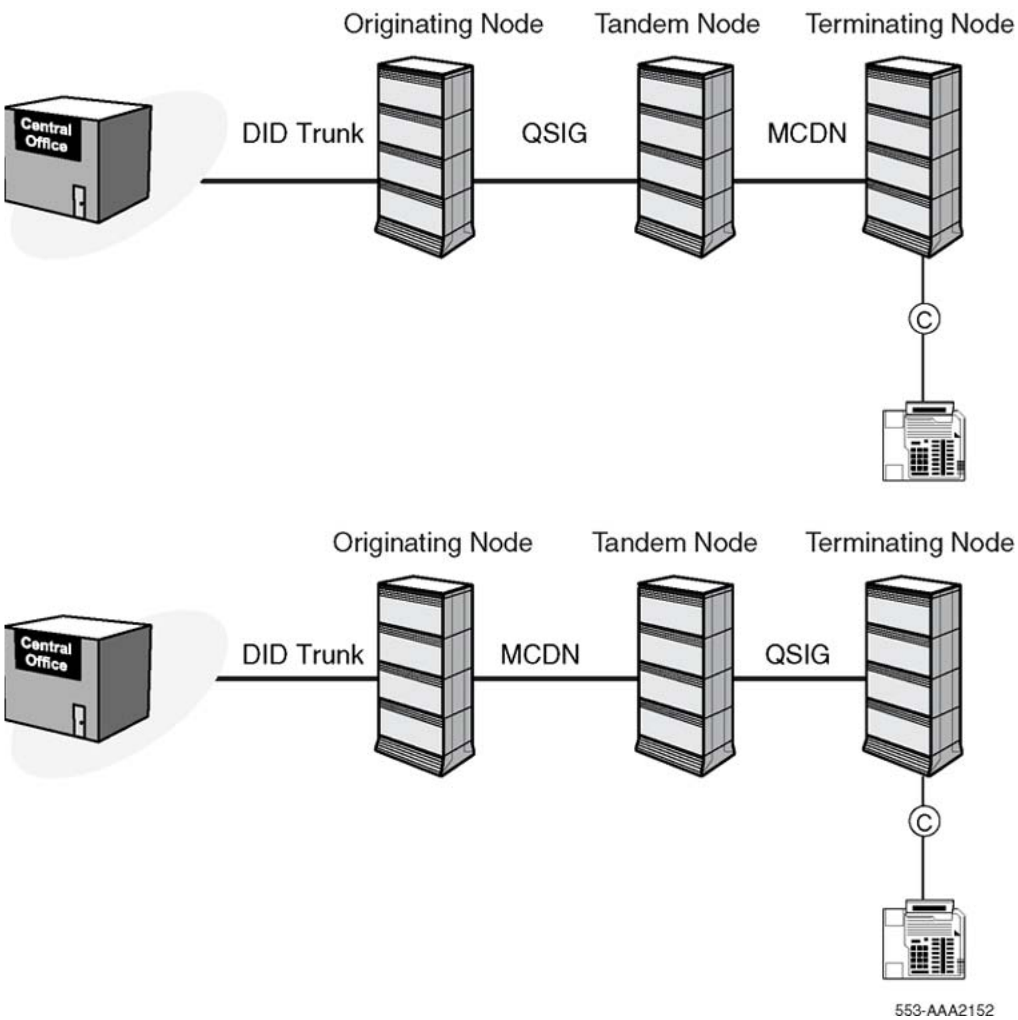


Figure 95: Incoming DID Digit Conversion over a QSIG/MCDN, or MCDN/QSIG link

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 96: MCDN Name Display features over a QSIG/MCDN link on page 892.

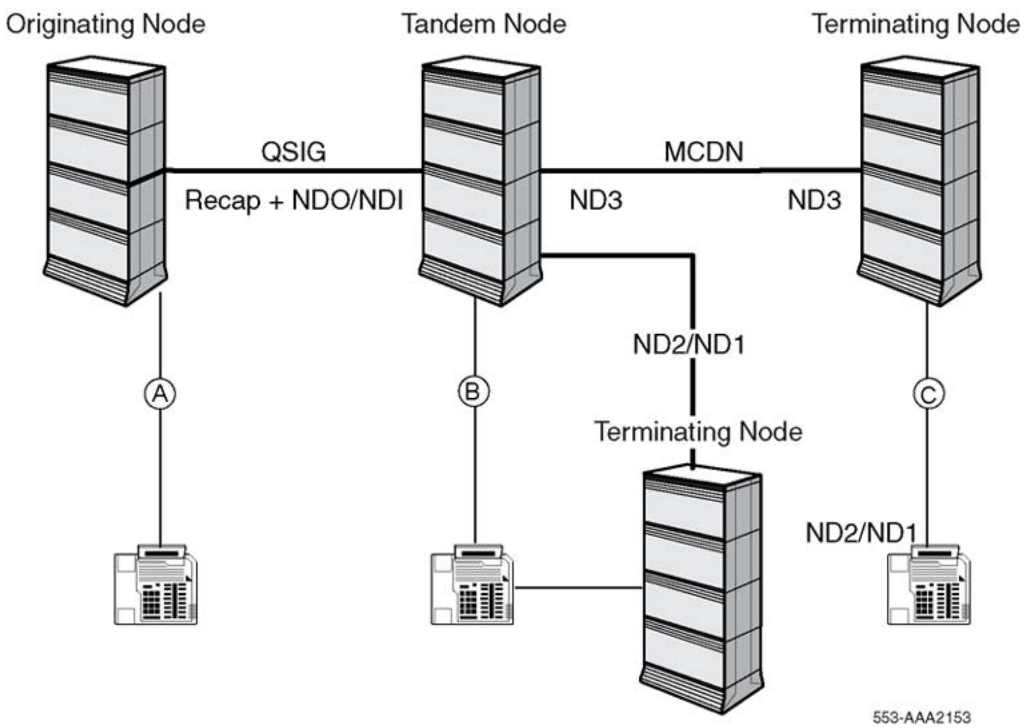


Figure 96: MCDN Name Display features over a QSIG/MCDN link

QSIG Call Diversion Notification

Calling Name

A QGIG 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 phone 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 phone.

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 phone.

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 can prevent it from being displayed.

Feature packaging

This feature requires the following packages:

- 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
- Primary Rate Access (PRI) package 146
- 2.0 Mbit 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

  This allows the Digit and Name Party restriction capability (without package 131, Digit and Name presentation is always 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 implementation

Task summary list

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

This section contains the procedures required to configure the QSIG Supplementary Services - Name Display Enhancement feature on a system.

ISDN PRI implementation

Table 411: LD 95 - Configure Calling Party Name Display data block for a customer.

Prompt	Response	Description
REQ	NEW	Add new data.
	CHG	Change existing data.
TYPE	CPND	Configure the Calling Party Name Display data block.

Prompt	Response	Description
CUST	xx	Customer number, as defined in LD 15.

Table 412: LD 95 - Define a name string for a customer.

Prompt	Response	Description
REQ	NEW	Add new data.
	CHG	Change existing data.
TYPE	NAME	Create a new Calling Party Name string.
CUST	xx	Customer number, as defined in LD 15.
...		
DN	xxx...x	Directory Number (1 to 7 digits) to which the name string is associated.

Table 413: LD 10 - Define the Calling Name Class of Service for analog (500/2500-type) phones.

Prompt	Response	Description
REQ:	NEW CHG	Add new data, or change existing data.
TYPE:	500	Type of analog phone.
TN		Terminal number
	l s c u	Format for Large System and Avaya Communication Server 1000E (Avaya CS 1000E) system, where l = loop, s = shelf, c = card, u = unit.
	c u	Format for Media Gateway 1000 B (Avaya MG 1000B), where c = card and 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 defined in LD 15.
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		Calling Party Name Display name.
	aaaa yyyy	aaaa = first name, yyyy = last name.
- XPLN	xx	Expected name length.
...		

Prompt	Response	Description
CLS	NAMA	Allow calling name display at the far end. (NAMD) = Name Display Denied is the default.

Table 414: LD 11 - Define the Calling Name Class of Service for Meridian 1 Proprietary Phones.

Prompt	Response	Description
REQ:	NEW CHG	Add new data, or change existing data.
TYPE:	a...a	Telephone type. Type **?** for a list of possible responses.
TN		Terminal number
	l s c u	Format for Large System and CS 1000E system, where l = loop, s = shelf, c = card, u = unit.
	c u	Format for Media Gateway 1000B, where c = card and 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 defined in LD 15.
CLS		Enter each non-default Class of Service entry separated by a space.
	CNDA	Allow the calling name to be displayed on this phone. (CNDD) = Deny calling name display is default.
	DNDA	Allow the name of the original caller to be displayed on this phone, for redirected calls. (DNDD) = Dialed Name Display Denied is default.
	NAMA	Allow calling name display at the far end. (NAMD) = Name Display on other phone Denied is default.
KEY	xx aaa yyyy	Phone function key and assignments.
- CPND	NEW CHG OUT	Calling Party Name Display Add data block. Change existing data block. Remove existing data block.
- NAME		Calling Party Name Display name.
	aaaa yyyy	aaaa = first name, yyyy = last name.

Table 415: LD 12 - Define the Calling Name Class of Service for Attendant Consoles.

Prompt	Response	Description
REQ:	NEW CHG	Add new data, or change existing data.

Prompt	Response	Description
TYPE:	2250	Type of Attendant Console.
TN		Terminal number
	l s c u	Format for Large System and CS 1000E system, where l = loop, s = shelf, c = card, u = unit.
	c u	Format for Media Gateway 1000B, where c = card and u = unit.
SETN		Second Terminal Number
	l s c u	Format for Large System and CS 1000E system, where l = loop, s = shelf, c = card, u = unit.
	c u	Format for Media Gateway 1000B, where c = card and u = unit.
CUST	xx	Customer number, as defined in LD 15.
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.

Note:

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 can be either by Object Identifier, used for ETSI, or by Integer Value, used for ISO.

Table 416: LD 17 - Configure the method of Operation Coding for the Name Display supplementary services for the associated QSIG 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 port 0-63 for Large Systems.
- 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.

Table 417: 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 defined in LD 15.
...		
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

Table 418: 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.
ROUT		Route number
	0-511	Range for Large System and CS 1000E system.
	0-127	Range for Small System, Media Gateway 1000B, and Media Gateway 1000T.
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.

Prompt	Response	Description
		Note: NDO and NDI are mutually exclusive.

Table 419: ISDN BRI implementation, task 2

Prompt	Response	Description
REQ	NEW	Add new data.
	CHG	Change existing data.
TYPE	TSP	Terminal Service Profile.
DSL	l s c dsl	Digital Subscriber Loop For Large Systems. 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	For Small Systems 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.

Feature operation

No specific operating procedures are required to use this feature.

Chapter 65: ISDN QSIG Path Replacement

Contents

This section contains information on the following topics:

Applicable regions

ISDN BRI trunking is not available in North America. Contact your system supplier or your Avaya representative to verify support of this product in your area.

Feature description

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 phone over a QSIG network).
- Network congestion on the first Private Integrated Service Network Exchange (PINX) node.
- Private Integrated Service Network Exchange (PINX) can 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 phone answer the call.

ISDN QSIG Path Replacement uses the following replacement mechanisms: triangulation and anti-tromboning. These are described in more detail beginning on Triangulation Path Replacement on page 903.

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 for each 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 97: Triangulation Path Replacement on page 904, Phone A on PINX 1 calls Phone B on PINX 2. Phone B has activated Call Forward All Calls to Phone 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 Phone C.

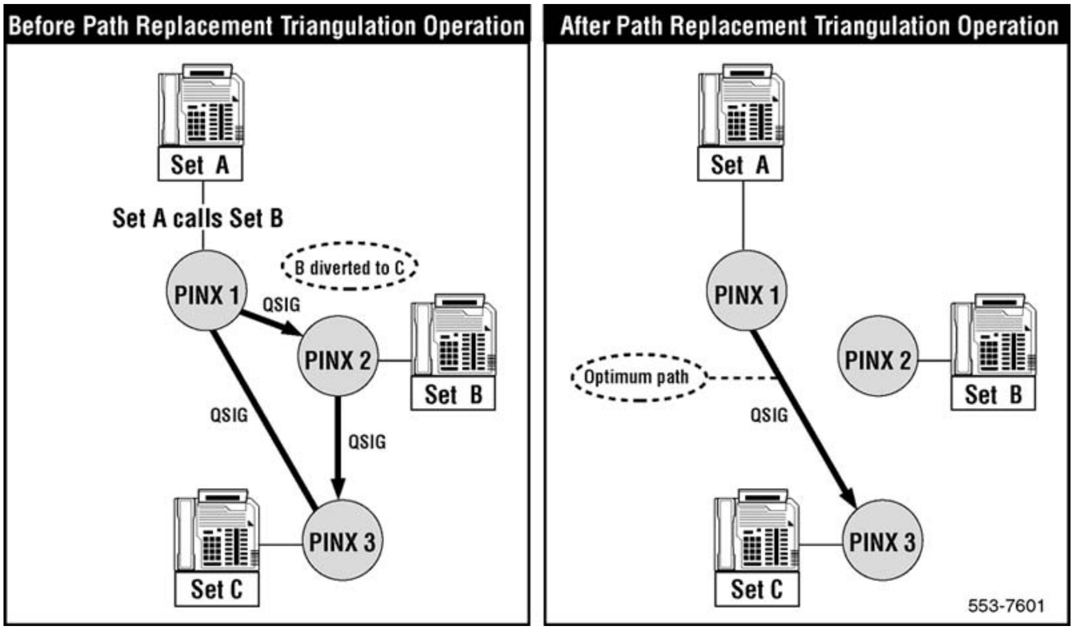


Figure 97: Triangulation Path Replacement

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 98: Tromboning Path Replacement on page 905, Phone A on PINX 1 calls Phone B on PINX 2. Phone B has activated Call Forward All Calls to Phone C on PINX 1. Antitromboning occurs between PINX 1 and PINX 2 and these trunks are torn down after Phone C answers the call.

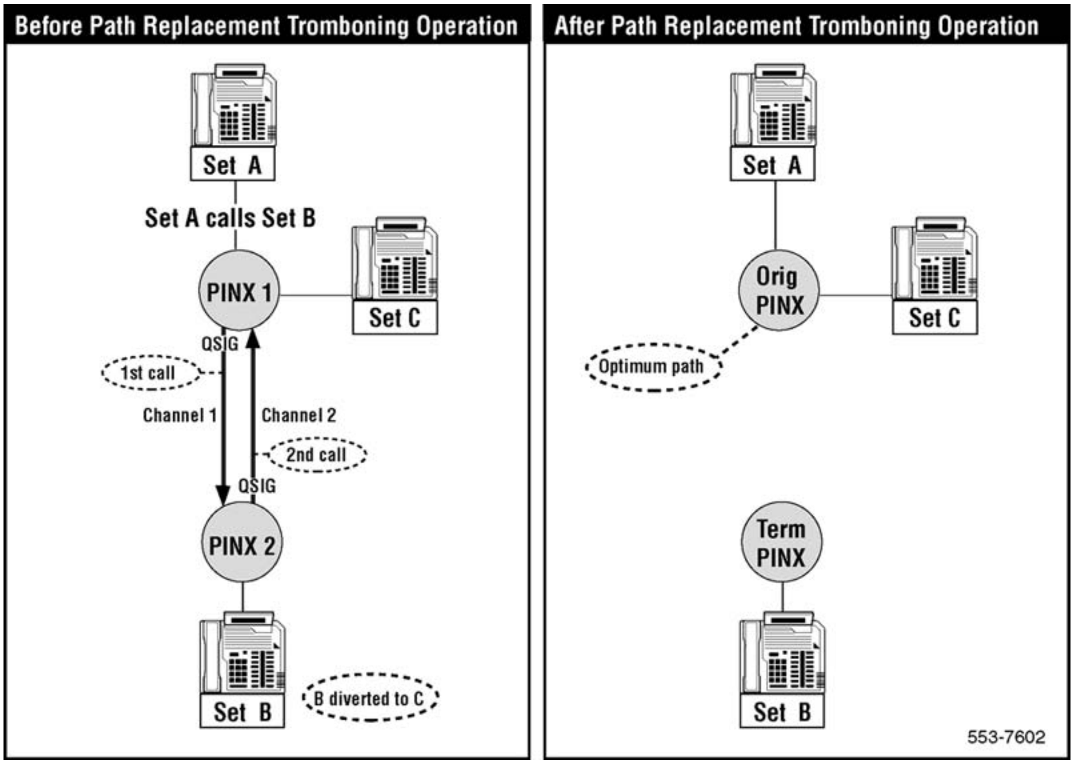


Figure 98: Tromboning Path Replacement

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 rerouted number. As shown in Figure 99: Path Replacement Retaining Part of Old Connection on page 906, the connection between the cooperating and transit PINX is retained after path replacement has occurred.

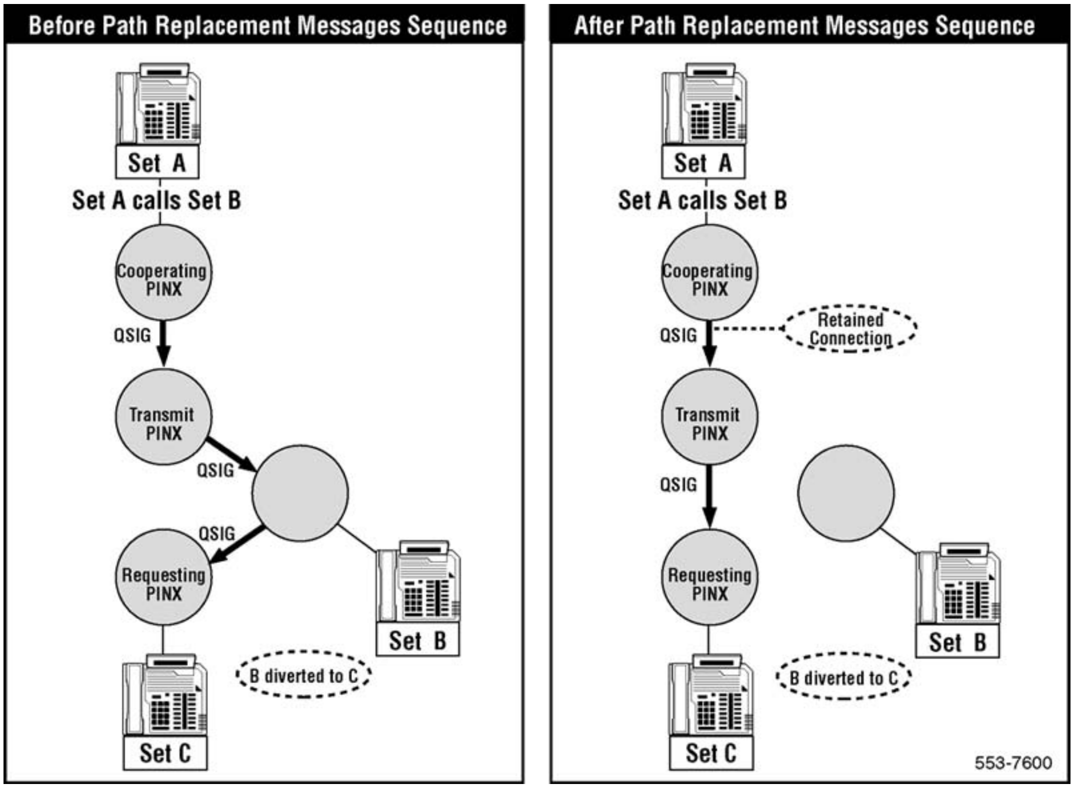


Figure 99: Path Replacement Retaining Part of Old Connection

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. Diversion of QSIG Call
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 system during an ISDN QSIG Path Replacement request, then the replacement request is terminated. After Call Transfer, the system can accept path replacement from another PINX.

Call Hold

Path Replacement cannot be invoked on a held call. Path replacement is only accepted on an active call.

Conference Calls

The system 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 system 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) phones, replacement is not terminated because the Digit Tone Multi-Frequency (DTMF) generated by the phone is detected.

Music

Radio Paging

Recorded Announcement

The system does not invoke the Path Replacement feature on Music, Paged or Recorded Announcement calls.

Feature packaging

This feature requires the following packages:

- Network Alternate Route Selection (NARS) package 58 (for Uniform Dialing Plan), or Coordinated Dialing Plan (CDP) package 59
- Flexible Numbering Plan (FNP) package 160
- QSIG Interface (QSIG) package 263
- QSIG-GF Transport (QSIGGF) package 305
- QSIG Supplementary Services (QSIG-SS) package 316
- Primary Rate Access (PRI) package 146
- 2.0 Mbit 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 implementation

Task summary list

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

1. Table 420: LD 15 - Configure Private Integrated Services Network Exchange Directory Number. on page 910
2. Table 421: LD 16 - Configure New Path Replacement parameters for an ISDN PRI or BRI trunk. on page 910
3. Table 422: LD 17 - Configure QSIG Path Replacement for a QSIG ETSI or ISO interface. on page 912

Table 420: LD 15 - Configure Private Integrated Services Network Exchange Directory Number.

Prompt	Response	Description
REQ:	CHG	Change existing data.
TYPE:	NET	Networking data.
CUST		Customer number
	0-99	Range for Large System and Avaya Communication Server 1000E (Avaya CS 1000E) system.
	0-31	Range for Media Gateway 1000 B (Avaya MG 1000B).
ISDN	YES	Integrated Services Digital Network.
- PINX_DN	xx...x	Private Integrated Services Network Exchange Directory Number, up to seven digits.

Table 421: LD 16 - Configure New Path Replacement parameters for an ISDN PRI or BRI trunk.

Prompt	Response	Description
REQ	NEW	Add new data.
	CHG	Change existing data.
TYPE	RDB	Route Data Block.
CUST	xx	Customer number, as defined in LD 15.

Prompt	Response	Description
ROUT		Route number
	0-511	Range for Large System and CS 1000E system.
	0-127	Range for Media Gateway 1000B.
DTRK	YES	Digital Trunk Type.
- DGTP	PRI BRI	PRI trunk. BRI trunk.
- IFC	ESGF ISGF	ETSI QSIG interface with GF platform. ISO QSIG interface with GF platform.
RCAP		Add Path Replacement as a remote capability.
	PRI PRO	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		Path Replacement Triggers.
	<CR>	<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.

Table 422: 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 xx	Change D-Channel number.
...		
- IFC	ESGF ISGF	ETSI QSIG interface with GF platform. 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.
	DV3I DV3O	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. 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).
...		
PR_TRIGS		Path Replacement Triggers.
	<CR>	<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. 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). Precede DIV, CNG, and CON with X to remove the Path Replacement trigger.

Prompt	Response	Description
PR_RTN	YES (NO)	Retain option is (is not) supported by the far end PINX.

Feature operation

No specific operating procedures are required to use this feature.

Chapter 66: MCDN End to End Transparency

Contents

This section contains information on the following topics:

Feature description

Meridian Customer Defined Network (MCDN) services are based on propriety specific Integrated Services Digital Network (ISDN) signaling. The MCDN End to End Transparency (MEET) 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). MEET 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) 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 100: MCDN End-to-End Transparency on page 916 illustrates the MEET feature.

Note:

ISDN BRI trunk access is not supported in North America.

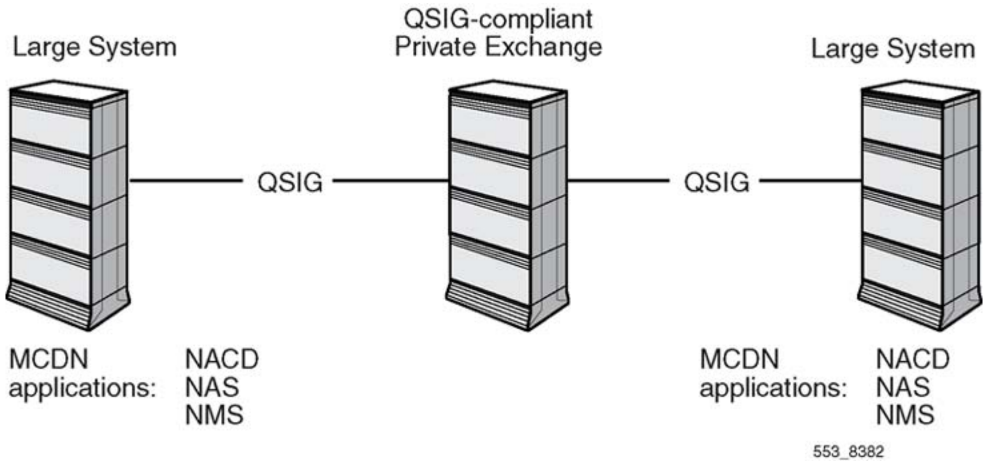


Figure 100: MCDN End-to-End Transparency

NAS using the QSIG transport

The MEET feature supports NAS on the ISDN Primary Rate Interface (PRI) or the ISDN Basic Rate Interface (BRI) using the QSIG transport.The following NAS functions 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 the NAS feature description in this document.

NACD using the QSIG transport

The MEET 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 cannot be mixed together. The following NACD functions 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

For information about these NACD features, see *Avaya Automatic Call Distribution Fundamentals, NN43001-551*.

NMS-MC using the QSIG transport

The MEET feature supports NMS-MC 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.

For further information about these NMS-MC features, see the NAS feature description in this document.

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 can 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 system 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 MEET feature, NACD continues to work fully with Symposium in an MCDN network. The MEET 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 using the QSIG transport

NMS-MC 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 system nodes.

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 the NAS feature description in this document 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 *Avaya Automatic Call Distribution Fundamentals, NN43001-551* 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 phone and the name of the target agent's ACD DN is displayed on the originator's phone 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*Avaya Automatic Call Distribution Fundamentals, NN43001-551* 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 can 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 phone. If the attendant NMS-MC has a specific ICI key defined, when a message Waiting call terminates to the attendant NMC-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

This feature requires the following packages:

- Multi-purpose Serial Data Link (MSDL) package 222
- QM reference signaling point Interface (QSIG) package 263
- QSIG Generic Functional protocol (QSIGGF) package 305
- MCDN End to End Transparency (MEET) package 348

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
- 1.5 Mbit Primary Rate Access (PRA) package 146, or
- 2.0 Mbit Primary Rate Interface (PRI2) package 154, or
- International Primary Rate Interface (IPRA) package 202

The following software packages are required for this feature to operate over QSIG ISDN BRI Trunks:

Note:

ISDN BRI trunk access is not supported in North America.

- Basic Rate Interface (BRI) 216
- ISDN BRI Trunk Access (BRIT) 233

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:

NAS packages and Attendant Overflow Position (AOP) are mutually exclusive and cannot be equipped on the same system.

Note:

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 (Class of Service) package 77
- Network Message Services (NMS) package 175

Feature implementation

Task summary list

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

1. Table 423: LD 17 - Configure a D-Channel for an ISDN PRI interface using the QSIG transport and add MCDN features. on page 925
2. Table 424: LD 16 - Configure route for an ISDN BRI interface using the QSIG transport and add MCDN features. on page 925
3. Table 425: LD 12 - Configure Attendant Consoles with NAS keys. (optional) on page 926
4. Table 426: LD 15 - Enable or disable network attendant control, NAS routing, and define a trunk ICI key. on page 927
5. Table 427: LD 86 - Define NAS routing tables. on page 927
6. Table 428: LD 15 - Configure a CLID table. on page 929
7. Table 429: LD 23 - Configure ACD Directory Number queues at source and target nodes. on page 929
8. Table 430: LD 23 - Configure an NACD Routing Table. on page 929
9. Table 431: LD 11 - Define a Meridian 1 proprietary ACD phone at source and/or target node. on page 930
10. Table 432: LD 10 - Define an analog (500/2500-type) phone at source and/or target node. on page 930

11. Table 433: LD 15 - Allow Message Waiting Center access. on page 931.
12. Table 434: LD 11 - Define a Meridian 1 proprietary ACD agent phone at source and/or target node. on page 932
13. Table 435: LD 11 - Define a Meridian 1 proprietary NMS-MC user phone. on page 932
14. Table 436: LD 23 - Create primary voice messaging ACD queue. on page 933
15. Table 437: LD 11 - Add agents to the primary agent queue. on page 933
16. Table 438: LD 23 - Configure ACD parameters for all voice service queue. on page 934

The following is a summary of the steps required to implement MEET:

1. In LD 17, configure a D-Channel for a PRI interface using the QSIG transport and select the NAS, NACD and/or NMS MCDN features or in LD 16, configure a route for an ISDN BRI interface using the QSIG transport and add the NAS, NACD and/or NMS MCDN features.
2. After selecting NAS in LD 17 or LD 16, configure NAS for a QSIG link by following these steps:
    a. LD 12 - Configure attendant consoles with NAS key (optional).
    b. LD 15 - Enable or disable network attendant control, NAS control and define trunk ICI keys and NAS routing thresholds in LD 15.
    c. LD 86 - Define the NAS routing table.
3. After selecting NACD in LD 17 or LD 16, configure NACD for a QSIG link by following these steps:
    a. Define a CDP or UDP between the two nodes in LD 87 and LD 90.
    b. LD 15 - Configure a CLID table.
    c. LD 23 - Configure the ACD DN queue at source and target nodes.
    d. LD 23 - Configure the NACD routing table in LD 23.
    e. LD 11 - Define a Meridian 1 proprietary ACD phone at source and/or target node.
4. After selecting NMS in LD 17 or LD 16, configure NMS-MC for a QSIG link by following these steps:
    a. Define a CDP (DSC or LSC) or UDP between the two nodes in LD 87 and LD 90.
    b. LD 15 - Allow Message Waiting Center access.
    c. LD 11 - Define a Meridian 1 proprietary ACD agent phone at source and/or target node.
    d. LD 11 - Define a Meridian 1 proprietary NMS-MC user phone.

ISDN PRI implementation using the QSIG transport

Table 423: 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	ETSI QSIG interface with GF platform. 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. XACD disables NACD on QSIG
	NMS	Enable NMS on QSIG. XNMS disables NMS on QSIG.

ISDN BRI implementation using QSIG transport

Table 424: LD 16 - Configure route for an ISDN BRI interface using the QSIG transport and add MCDN features.

Prompt	Response	Description
REQ	NEW	Add new data.
	CHG	Change existing data.
TYPE	RDB	Route Data Block.
TKTP	a...a	Trunk Type.
...		

Prompt	Response	Description
DTRK	(NO) YES	Digital Trunk Route.
- BRIP	(NO) YES	ISDN BRI Packet handler route
- DGTP	a...a BRI	Digital Trunk Type for route. Basic Rate Interface (Allowed if TKTP = TIE, COT or DID and BRIP = NO
...		
- IFC	ESGF ISGF	ETSI QSIG interface with GF platform. 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.
	NACD	Enable NACD on QSIG.
	NMS	Enable 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 or 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.

Table 425: LD 12 - Configure Attendant Consoles with NAS keys. (optional)

Prompt	Response	Description
REQ:	NEW CHG	Add new data, or change existing data.
TYPE:	2250	Attendant Console type.
TN		Terminal number

Prompt	Response	Description
	l s c u	Format for Large System and Avaya Communication Server 1000E (Avaya CS 1000E) system, where l = loop, s = shelf, c = card, u = unit.
	c u	Format for Media Gateway 1000 B (Avaya MG 1000B), where c = card and u = unit.
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.

Table 426: 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, or change existing data.
TYPE:	ATT	Attendant data.
TN		Terminal number
	l s c u	Format for Large System and CS 1000E system, where l = loop, s = shelf, c = card, u = unit.
	c u	Format for Media Gateway 1000B, where c = card and u = unit.
- ISDN	YES	Integrated Services Digital Network.
- ICI	0-19	Key number. Followed with a space and the trunk type.
	NCO NDID NTIE NFEX NWAT	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.

Table 427: LD 86 - Define NAS routing tables.

Prompt	Response	Description
REQ	NEW	Add new data.

Prompt	Response	Description
	CHG	Change existing data.
CUST	xx	Customer number, as defined in LD 15.
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 can 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 or and select NACD as an MCDN feature in LD 17, or configure an ISDN BRI interface using the QSIG transport or 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 phone at source and/or target node in LD 11.

Table 428: LD 15 - Configure a CLID table.

Prompt	Response	Description
REQ:	NEW CHG	Add new data, or change existing data.
TYPE:	NET	Networking data (if REQ = CHG only).
CUST		Customer number
	0-99	Range for Large System and CS 1000E system.
	0-31	Range for Media Gateway 1000B.
- CLID		CLID option.
	YES (NO)	Configure a CLID table for the customer. (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.

Table 429: LD 23 - Configure ACD Directory Number queues at source and target nodes.

Prompt	Response	Description
REQ	NEW	Add new data.
	CHG	Change existing data.
TYPE	ACD	Automatic Call Distribution data block.
CUST	xx	Customer number, as defined in LD 15.
...		
ACDN	xxxx	ACD Directory Number.

Table 430: LD 23 - Configure an NACD Routing Table.

Prompt	Response	Description
REQ	NEW	Add new data.
	CHG	Change existing data.
TYPE	aaaa	Type of data block. Enter NACD for Network ACD.
CUST	xx	Customer number, as defined in LD 15.
ACDN	xxxx	ACD Directory Number.

Prompt	Response	Description
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.

Note:

When assigning a CLID entry to an ACD phone, you cannot use the position ID already on the phone. You must out the set first or null the ACD key and then rebuild with the table entry number.

Table 431: LD 11 - Define a Meridian 1 proprietary ACD phone at source and/or target node.

Prompt	Response	Description
REQ:	NEW CHG	Add new data, or change existing data.
TYPE:	a...a	Telephone type. Type **?** for a list of possible responses.
TN		Terminal number
	l s c u	Format for Large System and CS 1000E system, where l = loop, s = shelf, c = card, u = unit.
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 (must be key 0). 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. Refer to the note at the top of LD 11 on assigning a CLID entry to an ACD phone.
KEY		Phone function key assignments.
	xx MSB yyyy (0)-N/D) xx NRD yyy (0)-N/D	xx = key number. MSB = Make Set Busy key. NRD = Not Ready key.

Table 432: LD 10 - Define an analog (500/2500-type) phone at source and/or target node.

Prompt	Response	Description
REQ:	NEW CHG	Add new data, or change existing data.

Prompt	Response	Description
TYPE:	500	Analog (500/2500-type) phone.
TN		Terminal number
	l s c u	Format for Large System and CS 1000E system, where l = loop, s = shelf, c = card, u = unit.
CLS	AGTA	ACD services for analog (500/2500-type) phones 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 or and select NMS as an MCDN feature in LD 17, or configure an ISDN BRI interface using the QSIG transport or 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 an ACD agent phone at source and/or target node in LD 11.
    - Define an NMS-MC user phone in LD 11.

Table 433: LD 15 - Allow Message Waiting Center access.

Prompt	Response	Description
REQ:	NEW CHG	Add new data, or change existing data.
TYPE:	FTR	Feature data.
CUST		Customer number
	0-99	Range for Large System and CS 1000E system.
	0-31	Range for Media Gateway 1000B.
OPT	(MCX) MCI	Message Center excluded. Message Center included.
	(MWUD) MWUA	Message Waiting Unconditional Denied. Message Waiting Unconditional Allowed.

Table 434: LD 11 - Define a Meridian 1 proprietary ACD agent phone at source and/or target node.

Prompt	Response	Description
REQ:	NEW CHG	Add new data, or change existing data.
TYPE:	a...a	Telephone type. Type **?** for a list of possible responses.
TN		Terminal number
	l s c u	Format for Large System and CS 1000E system, where l = loop, s = shelf, c = card, u = unit.
CLS	(MWD) MWA	Class of Service options. Message Waiting Denied Message Waiting Allowed.
KEY	xx MIK	MIK = Message Indication Key. xx = key number.
	xx MCK	MCK = Message Cancellation Key. xx = key number.

Table 435: LD 11 - Define a Meridian 1 proprietary NMS-MC user phone.

Prompt	Response	Description
REQ:	NEW CHG	Add new data, or change existing data.
TYPE:	a...a	Telephone type. Type **?** for a list of possible responses.
TN		Terminal number
	l s c u	Format for Large System and CS 1000E system, where l = loop, s = shelf, c = card, u = unit.
CLS	AGTA	ACD services for analog (500/2500-type) phones allowed.
KEY	xx MWK x...x	Phone 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 or and select NMS as an MCDN feature in LD 17, or configure an ISDN BRI interface using the QSIG transport or 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 Avaya CallPilot 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.

Table 436: 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, as defined in LD 15.
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).

Note:

When assigning a CLID entry to an ACD phone, you cannot use the position ID already on the phone. You must out the phone first or null the ACD key and then rebuild with the table entry number.

Table 437: LD 11 - Add agents to the primary agent queue.

Prompt	Response	Description
REQ:	NEW	Add new data.
TYPE:	xx	ACD data block.
TN		Terminal number
	l s c u	Format for Large System and CS 1000E system, where l = loop, s = shelf, c = card, u = unit.
KEY		xx = key number
	xx ACD yyyy 0-N/D zzzz	ACD key, where: xx = key number (must be key 0). 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 phone.
	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 can 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.

Prompt	Response	Description
		The DN can be up to 4 digits, up to 7 digits with Directory Number Expansion (DNXP) package 150. 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.

Table 438: 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, as defined in LD 15.
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.

Feature operation

Feature operation for direct Message Center calls and Indirect Message Center calls are presented in the following sections.

For NAS and NMS-MC and NMS-MM feature operation, see the NAS feature description in this document.

For NACD feature operation, see *Avaya Automatic Call Distribution Fundamentals, NN43001-551*.

Direct Message Center call

1. User A dials their MC DN or presses their MWK key.
2. The MC rings. For ACD phones, the MCK lamp state shows the MC user's message indication state: lit if there is no message; slow flash if a message is waiting; fast flash if there is bad data; dark if message waiting indication class of service is denied.
3. The MC operator answers the call. For DN phones, the MCK lamp state reflects MC user's message indication state.
4. User B is connected to MC and is given their message by the MC operator.
5. The MC operator presses MCK key (if MCK lamp is flashing) to turn off message waiting indication at station B. The MCK lamp changes to dark.
6. User A disconnects from the MC.

Indirect Message Center call

1. User A calls User B.
2. User B has calls redirected to the MC (Call Forward all Calls or No Answer).
3. The call is presented to the MC; MC rings; MC operator answers the call.
4. If QSIG Diversion is configured, the MIK lamp has the lamp state that reflects the MC user's message indication state: lit if there is no message; flash if a message is waiting; fast flash if there is bad data; dark if message waiting indication class of service is denied.User A is connected to MC and leaves a message.
5. If QSIG diversion is not configured, the MC operator activates the Message Waiting indication in the following sequence:
    - MC operator puts the call On Hold, presses the MIK key and MIK lamp lights.
    - MC operator dials User B's DN. The MIK lamp changes to the lamp state that reflects the MC user's message indication state.
    - If the MIK lamp is lit, the MC operator presses the MIK key to turn on the message waiting indication at station B, The MIK lamp changes to dark.
    - The MC operator returns to the Held call.
6. User A disconnects from the call.

Chapter 67: Meridian Hospitality Voice Services

Contents

This section contains information on the following topics:

Feature description

This feature provides an enhanced form of Meridian Mail tailored to Hospitality Services. It simplifies the use of mailboxes and 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 of the command and status information between the system 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 system and the VAS stops. Calls could be left in various ringing states indefinitely (until the ISDN/AP link is 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 Center.
- 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, 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 switch as the attendant and room phones.

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.

Operating parameters

No operating parameters are specified for this feature.

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, 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 can 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 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 can then access voice messaging. DKS is only supported from the attendant consoles local to Meridian Mail. The attendant can also place direct calls to Meridian Mail.

Digit Key Signaling at Console (DKS)

With DKS equipped, attendants assist callers in Meridian Mail activities. The attendant extends 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, callers can be redirected to Meridian Mail for Voice Mail Services. A called phone must have Hunting Allowed (HTA) Class of Service and the Customer Route Data Block must be set to "YES" in LD 15.

M2317 and Meridian Modular soft key menus

M2317 or Meridian Modular soft key menus are not supported by MHVS. These three phones with CCSA Class of Service are not presented with the Meridian Mail softkey 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 phones are located on the same switch.

Pretranslation

Prior to MHVS, the setup of calls using the ISDN/AP was not supported from phones 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 can 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) requires software 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
- Auxiliary Processor Link (APL) package 109

The Property Management System Interface (PMSI) package requires the following packages:

- Controlled Class of Service (CCOS) package 81
- Background Terminal (BGD) package 99
- Room Status (RMS) package 100
- 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 can 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.

Integrated Services Digital Network (ISDN) package 145 is also required.

Feature implementation

Task summary list

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

1. Table 439: LD 15 - Allow call redirection to Meridian Mail for voice messaging. on page 942
2. Table 440: LD 23 - Define the Attendant Overflow Position (AOP) Directory Number. on page 942

Table 439: LD 15 - Allow call redirection to Meridian Mail for voice messaging.

Prompt	Response	Description
REQ:	CHG	Change existing data.
TYPE:	RDR	Call Redirection data
CUST		Customer number
	0-99	Range for Large System and Avaya Communication Server 1000E (Avaya CS 1000E) system.
	0-31	Range for Media Gateway 1000 B (Avaya MG 1000B).
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.

Table 440: LD 23 - Define the Attendant Overflow Position (AOP) Directory Number.

Prompt	Response	Description
REQ	NEW	Add new data.
	CHG	Change existing data.
TYPE	ACD	Automatic Call Distribution Data Block.
CUST	xx	Customer number, as defined in LD 15.
ACDN	xxxx	ACD Directory Number.

Prompt	Response	Description
MWC	YES	ACD DN Message Center DN.
CMS	YES	Command and status link.

Feature operation

No specific operating procedures are required to use this feature.

Chapter 68: MSDL Port Overload Counter

Contents

This section contains information on the following topics:

Feature description

The MSDL Port Overload Counter feature provides the capability of locking out individual MSDL ports when incoming messages through a port exceed or equal the port overload threshold of 200 messages in a two-second time period.

When any of the four ports on the MSDL card reaches the port overload threshold it is locked-out, but the operation of other ports on the MSDL card will not be affected.

The MSDL card is still subject to a card overload threshold, which will cause a lock-out if it receives 300 or more messages for every two seconds. However, when an individual MSDL port becomes overloaded, the messages to that port are subtracted from the total card messages used in performing the card overload check. Furthermore, port overload checks have priority over card overload checks.

The overloaded port on the MSDL card is locked-out so that the card stops responding to the incoming messages from the overloaded port. This prevents the CPU from servicing a very high incoming message rate from the port, which could cause system degradation.

Figure 101: MSDL card lock-out on page 947 shows the existing MSDL card lockout, while Figure 102: MSDL port overload check preceding card overload check on page 947 shows the port lockout functionality introduced by the Port Overload Counter feature. Figure 103: MSDL port overload check preceding card overload check on page 948 demonstrates a port overload check being performed before a card overload check.

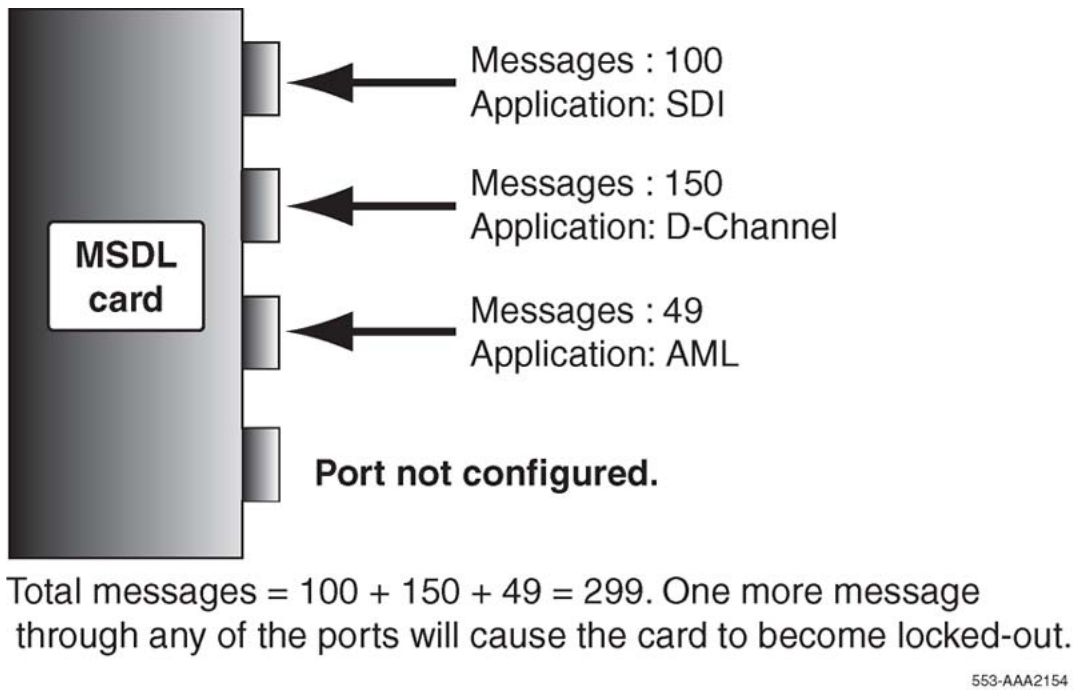


Figure 101: MSDL card lock-out

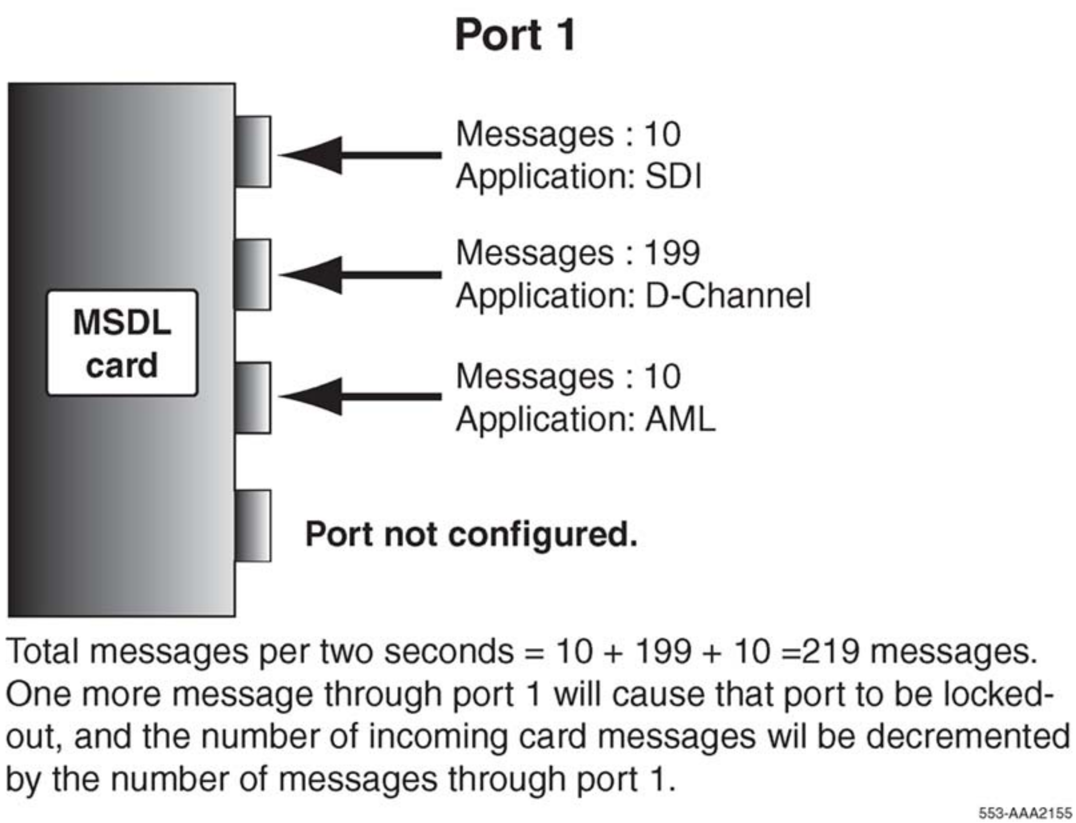


Figure 102: MSDL port overload check preceding card overload check

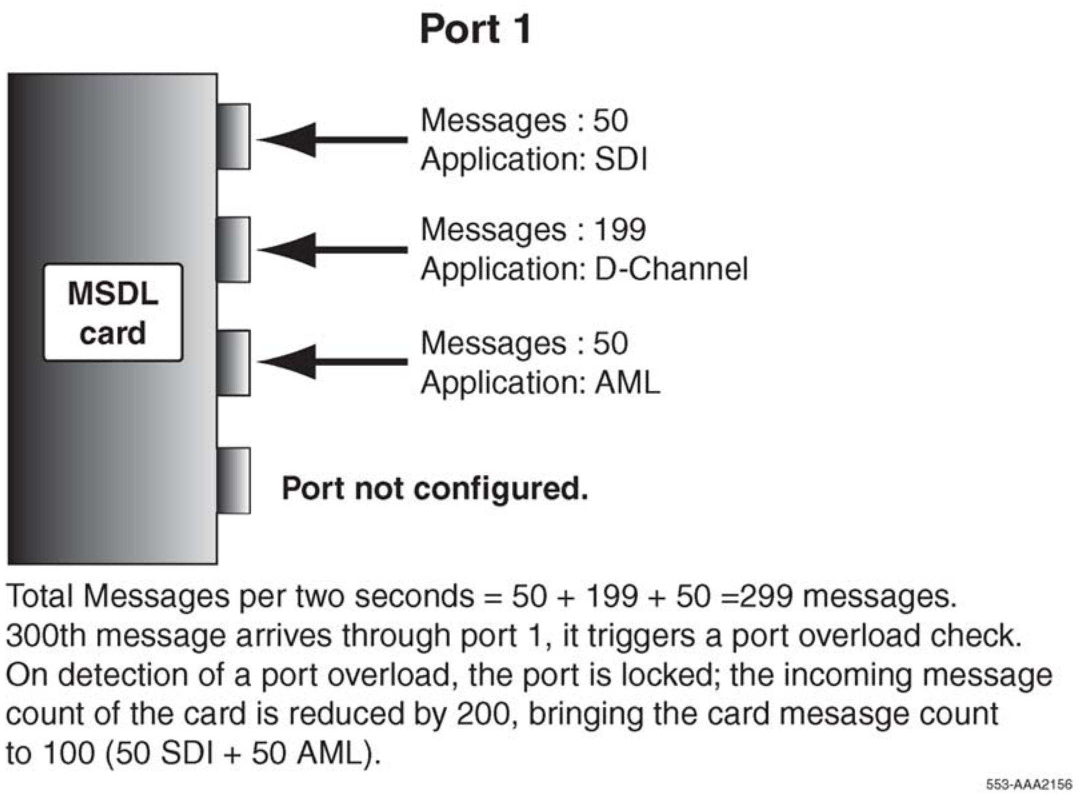


Figure 103: MSDL port overload check preceding card overload check

Operating parameters

There are no operating parameter specified for this feature.

Feature interactions

MSDL/MISP Interface Handler (MMIH)

The MMIH provides a software interface between the MSDL/MISP card and the system software, and allows applications on the MSDL/MISP card to access each other through the interface.

Previously, the interface software maintained a counter of incoming messages on an MSDL application basis. When the sum of all messages received indicated an excessive incoming message rate, the MSDL card was locked-out.

The Port Overload Counter feature modifies the MMIH so that incoming message counters are maintained for individual MSDL ports

This feature changes the functionality of the MMIH for an MSDL card, but the MISP remains unchanged.

DCH/SDI/AML Application

The Port Overload Counter feature changes the DCH/SDI/AML Application to cause the MSDL port to lockout when the incoming messages exceed the Port Overload threshold value.

B-Channel Overload Control (BCOC)

This feature has no interaction with the B-Channel Overload Control Feature.

MISP

This feature does not change the functionality of the MISP card interface.

Feature packaging

This feature requires package MSDL Card Package, package 222.

Feature implementation

There are no specific implementation procedures for this feature. However, as described in Maintenance and Diagnostics on page 950, modifications have been made to overlay maintenance and diagnostic commands.

The feature is activated whenever a configured application on an MSDL card is active and is receiving messages through the active port.

Feature operation

No specific operating procedures are required to use this feature.

Maintenance and Diagnostics

The Port Overload Counter feature has made the following modifications to overlay maintenance and diagnostic commands:

Table 441: LD 37 - I/O Diagnostic.

Commands	System Responses	Description
STAT MSDL xx (Where xx = physical MSDL number)	MSDL xx: ENBL SDI 7 OVLD PORT 0 DCH 11 OVLD PORT 1 AML 12 OVLD PORT 2	Display the status of ports on a card. The status includes overloaded ports.
STAT TTY xx (Where xx = logical SDI number)	TTY xx: OVLD	Display the status of an overloaded MSDL port running an SDI application.
ENL TTY xx Where xx = logical SDI number)	IOD_xxx	Failed to enable the SDI on MSDL port xx. Refer to Overloaded MSDL port running an SDI Application on page 952.

Table 442: LD 48 - Link Diagnostic.

Commands	System Responses	Description
STAT MSDL xx (Where xx = physical MSDL number)	MSDL xx: ENBL SDI 7 OVLD PORT 0 DCH 11 OVLD PORT 1 AML 12 OVLD PORT 2 (Where xx = physical MSDL number)	Display the status of ports on a card. The status includes overloaded ports.
STAT AML aa (Where xx = logical AML number)	AML: xx MSDL: yy PORT: zz DES: LYR2: OVLD (Where xx = logical AML number, yy = physical MSDL number, and zz = MSDL port)	Display the status of an overloaded MSDL port.

Commands	System Responses	Description
ENL AML xx (Where xx = logical AML number)	CSA_xxx	Failed to enable AML link. Refer to Overloaded MSDL port running an AML Application on page 951.

Table 443: LD 96 - PRI D Channel Diagnostic

Commands	System Responses	Description
STAT MSDL xx (Where xx = physical MSDL number)	SDI 7 OVLD PORT 0 DCH 11 OVLD PORT 1 AML 12 OVLD PORT 2	Display the status of ports on a card. The status includes overloaded ports.
STAT DCH xx (Where xx = logical D Channel number)	DCH xx: OVLD DES:	Display the status of an overloaded MSDL port running a D Channel Application.
ENL DCH xx (Where xx = logical D Channel number)	DCH_XXX	Failed to enable the D Channel. Refer to Overloaded MSDL port running a DCH Application on page 951.

Fault Clearance Procedures

Overloaded MSDL port running a DCH Application

Cause: The system received an excessive number of messages (equal to or greater than 200 messages in a two second time period) from the MSDL port.

To clear the fault: Manually disable the D-Channel in LD 96 using the command DIS DCH xx, where xx is the D-Channel number. Enable the D-Channel using LD 96 using the command ENL DCH xx.

Overloaded MSDL port running an AML Application

Cause: The system received an excessive number of messages (equal to or greater than 200 messages in a two second time period) from the MSDL port.

To clear the fault: Manually disable the AML Link in LD 48 using the command DIS AML xx, where xx is the AML Link number. Enable the AML Link using LD 48 using the command ENL AML xx.

Overloaded MSDL port running an SDI Application

Cause: The system received an excessive number of messages (equal to or greater than 200 messages in a two second time period) from the MSDL port. One scenario of the overloaded SDI port is the case of a smart-modem setup for "smart" vs. "dumb" mode. The system and the modem can be bouncing the same messages back and forth until the 200 port message threshold is reached.

To clear the fault: Manually disable the link in LD 37 using the command DIS TTY xx, where xx is the SDI number. Enable the AML Link using LD 37 using the command ENL TTY xx.

Chapter 69: Network and Executive Distinctive Ringing

Contents

This section contains information on the following topics:

Feature description

Network Distinctive Ringing (NDRG) allows a distinctive ringing cadence to be configured throughout a system network. Distinctive ringing is defined on a route basis. There are four NDRG distinctive ringing cadence indices that can be defined for a route. These indices are contained in the Flexible Tone and Cadences (FTC) table. If one of these indices has been defined for a route and an incoming trunk call over that route terminates on the local node, the terminating phone receives distinctive ringing. If the incoming call tandems to another node through an Integrated Services Digital Network (ISDN) TIE trunk, the terminating phone at the terminating node receives distinctive ringing. This occurs if the TIE trunk has marked as distinctive and if the NDRG feature is equipped at the terminating node; otherwise, normal ringing is given.

Executive Distinctive Ringing applies to both network and stand-alone environments. This feature allows a Class of Service to be entered for a phone, marking the phone as "executive". When a call is made from an executive phone, the called phone is rung distinctively. This feature uses the distinctive cadences introduced by the Network Distinctive Ringing (NDRG) feature.

One of five Classes of Service can be entered - EXR1, EXR2, EXR3, EXR4, or EXR0. EXR is the Class of Service mnemonic that marks the phone as executive, and the digits one to four indicate which of the four distinctive ringing cadences is to be applied. EXR0 is the default; it marks a phone as normal.

Operating parameters

Both Network Distinctive Ringing and Executive Distinctive Ringing can be equipped for a phone. In this case, a cadence that is selected for NDRG can also be selected for EDRG.

Within a network, if there are five routes marked as distinctive, and if an incoming call tandems between two nodes that are connected by a single TIE trunk, the terminating node can provide unique distinctive ringing for only four of the five routes. The originating node can provide unique distinctive ringing to all five routes since each route can use a different Flexible Tone and Cadence (FTC) table.

Feature interactions

Incoming trunk

An incoming trunk call that is redirected or attendant-extended will ring distinctively at the terminating phone, according to the cadence index of the originating trunk route. If the terminating phone is located at another node, it will ring distinctively according to the cadence index of the originating trunk route (if the NDRG feature is equipped at the terminating node).

Buzz

Network Distinctive Ringing and Executive Distinctive Ringing do not affect the buzzing of a phone.

Conference

If a new party is to be included in an established conference, the ringing that is applied to the phone of the new party depends on the phones of the established parties. The system scans the trunks and phones of the conferees for a trunk marked as distinctive or a phone designated as executive. The ringing cadence of the new phone depends on the highest index found by the scan.

Feature packaging

This feature requires the following packages:

- Distinctive Ringing (DRNG) package 74
- Flexible Tones and Cadences (FTC) package 125
- Integrated Services Digital Network (ISDN) package 145
- Integrated Services Digital Network Supplementary Features (ISDNS) package 161
- Executive Distinctive Ringing (EDRG) package 185

Feature implementation

Task summary list

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

1. Table 444: LD 10 - Define the distinctive ringing cadence for analog (500/2500-type). on page 956
2. Table 445: LD 11 - Define the distinctive ringing cadence/tone to be used for Meridian digital phones. on page 956
3. Table 446: LD 16 - Deny or allow Distinctive Ringing and define Network Ring Index. on page 956
4. Table 447: LD 56 - Define Flexible Tones and Cadences. on page 957

Table 444: LD 10 - Define the distinctive ringing cadence for analog (500/2500-type).

Prompt	Response	Description
REQ:	CHG	Change existing data.
TYPE:	500	Type of phone
TN		Terminal number
	l s c u	Format for Large System and Avaya Communication Server 1000E (Avaya CS 1000E) system, where l = loop, s = shelf, c = card, u = unit.
CLS	(EXR0) EXR1 EXR2 EXR3 EXR4	Executive Distinctive Ringing Off 0. Executive Distinctive Ringing On 1. Executive Distinctive Ringing On 2. Executive Distinctive Ringing On 3. Executive Distinctive Ringing On 4.

Table 445: LD 11 - Define the distinctive ringing cadence/tone to be used for Meridian digital phones.

Prompt	Response	Description
REQ:	CHG	Change existing data.
TYPE:	a...a	Telephone type. Type **?** for a list of possible responses.
TN		Terminal number
	l s c u	Format for Large System and CS 1000E system, where l = loop, s = shelf, c = card, u = unit.
CLS	(EXR0) EXR1 EXR2 EXR3 EXR4	Executive Distinctive Ringing Off (0). Executive Distinctive Ringing Tone 1. Executive Distinctive Ringing Tone 2. Executive Distinctive Ringing Tone 3. Executive Distinctive Ringing Tone 4.

Table 446: LD 16 - Deny or allow Distinctive Ringing and define Network Ring Index.

Prompt	Response	Description
REQ	CHG	Change existing data.
TYPE	RDB	Route Data Block
CUST	xx	Customer number, as defined in LD 15.
ROUT		Route number
	0-511	Range for Large System and CS 1000E system.
TKTP	aaa	Trunk Type.
DRNG	(NO) YES	Deny or allow Distinctive Ringing.
NDRI	(0)-4	Define the Network Distinctive Ringing Index.

Table 447: LD 56 - Define Flexible Tones and Cadences.

Prompt	Response	Description
REQ	NEW	Add new data.
	CHG	Change existing data.
TYPE	FTC	Flexible Tones and Cadences data block.
TABL	0-31	FTC Table number.
USER	(NO) YES	Print users of this table and tone table values (tone table value only).
DFLT	0-31	Default to existing FTC tone table.
RING	(NO) YES	Change the ringing feature definitions.
...		
- NDR1 PBX	0-255	Network Distinctive Ring 1 cadence for analog (500/2500-type) phones.
- NDR1 BCS		Network Distinctive Ring 1 for BCS Meridian 1 digital phones.
- NDR2 PBX	0-255	Network Distinctive Ring 2 cadence for analog (500/2500-type) phones.
- NDR2 BCS		Network Distinctive Ring 2 for BCS Meridian 1 digital phones.
- NDR3 PBX	0-255	Network Distinctive Ring 3 cadence for analog (500/2500-type) phones.
- NDR3 BCS		Network Distinctive Ring 3 for BCS Meridian 1 digital phones.
- NDR4 PBX	0-255	Network Distinctive Ring 4 cadence for analog (500/2500-type) phones.
- NDR4 BCS		Network Distinctive Ring 4 for BCS Meridian 1 digital phones.

Feature operation

No specific operating procedures are required to use this feature.

Chapter 70: Network Application Protocol Link Enhancement

Contents

This section contains information on the following topics:

Feature 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 will result in uninterrupted service.

The signaling from the caller's phone is ignored during the link failure. However, calls which were already established will remain established during the recovery period. Should the link be declared 'out of service', only those established calls which use trunks without disconnect supervision will not be maintained. (Such calls will be requeued 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 requeued. Requeued calls and incoming calls on the out-of-service link receive night-service treatment.

Normal operations resume when the link is restored.

Operating parameters

This feature applies only to the ISDN environment.

Feature interactions

There are no feature interactions associated with this feature.

Feature packaging

This feature requires the following packages:

- ISDN Application Protocol Link for Third-party Vendors (IAP3P) package 153, which is dependent on
  - Basic Automatic Call Distribution (BACD) package 40

Feature implementation

There are no specific implementation procedures for this feature.

Feature operation

No specific operating procedures are required to use this feature.

Chapter 71: Network Attendant Service

Contents

This section contains information on the following topics:

Feature description

Network Attendant Service (NAS) provides stand-alone attendant capabilities throughout a network. Any node in the network can have its attendant services located part-time or full-time at any other node in the network.

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 can 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 448: Determination of Night Service on page 962 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 448: 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 can 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 can 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 449: NAS Routing Possibilities on page 963.

Table 449: 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 can 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 can 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 phone is ringing or receiving an indication tone
    - the required party's phone 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 phone 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 system 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 phone, 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 phone a further opportunity to reach that phone. If the call is extended across the network to a busy phone, Camp-on or Call Waiting can 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 system.

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 phone 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 phone, the RLT trunk is released.
- If the phone 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 phone 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 phone answers the call. This can result in two tromboned TIE trunks or channels being held up while a phone is ringing.
- If anti-tromboning has been programmed, the tromboning is corrected after the phone is answered.
- If the phone 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 can be programmed as an option 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 for each 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 system can not have an actual attendant console. Instead, the system can be configured to use a Centralized Answering Position (using a Meridian 2616 digital phone.) 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 phone.

NAS and Centralized Attendant Service packages are mutually exclusive and can not be packaged together.

NAS and Attendant Overflow Position packages are mutually exclusive and can 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 phones 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
- 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 phone 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 Optimization/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 optimized on the MCDN side, but DPNSS1 trunks are not optimised on the DPNSS1 side of the RO/TAT Interworking gateway scenario.

EuroISDN Trunk - Network Side

NAS 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).

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 on each system. 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.

Virtual Network Services

As stated earlier, NAS and CAS cannot be configured together on the same system. Since VNS requires NAS, this means that VNS and CAS cannot be configured together on the same system, either.

Feature packaging

This feature requires the following packages:

- Basic Routing (BRTE) package 14
- Basic Queuing (BQUE) package 28
- Network Class of Service (NCOS) package 32
- Network Automatic Routing System (NARS) package 58 or
- Coordinated Dialing Plan (CDP) package 59
- Flexible Call Back Queuing (FCBQ) package 61
- Attendant Break-In/Trunk Offer (BKI) package 127
- Integrated Services Digital Network (ISDN) package 145
- Advanced ISDN Network Services (NTWK) package 148
- International Primary Access (PRA) package 146 or
- 2.0 Mbit Primary Rate Interface (PRI2) package 154 or
- ISDN Signaling Link (ISL) package 147
- Network Attendant Services (NAS) package 159
- ISDN Supplementary Features (ISDNS) package 161
- Originating Routing Control/Remote Virtual Queuing (ORC-RVQ) package 192 for Drop Back Busy (please see note under "Drop Back Busy" description.)

Example of NAS implementation

Figure 104: A simple application of NAS functionality in a private ISDN PRI network on page 972 illustrates a simple application of the NAS functionality in a private ISDN PRI network connecting three 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 system 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.

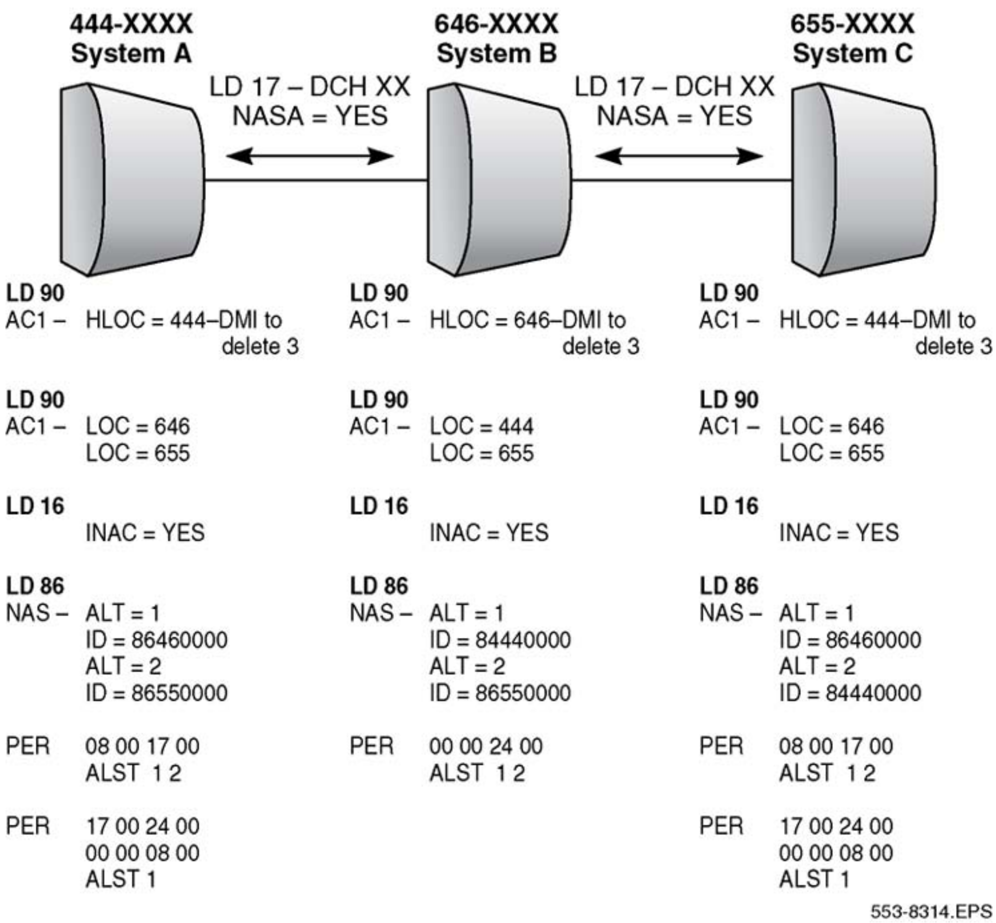


Figure 104: A simple application of NAS functionality in a private ISDN PRI network

Feature implementation

Task summary list

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

1. Table 450: LD 12 - Configure Attendant Consoles with NAS keys. on page 973
2. Table 451: LD 15 - Enable or disable network attendant control, Recall to Same Attendant, NAS routing, and define a TRK ICI key. on page 973
3. Table 452: LD 17 - Configure the D-channel interface (DCHI) for NAS. on page 974
4. Table 453: Task 4 on page 974
5. Table 454: LD 93 - Assign a NAS routing table to Console Presentation Groups. on page 976
6. Table 455: LD 16 - Allow or deny pre-answer tromboning (allow or deny an incoming call to be directly routed back on the same route). on page 976

Table 450: LD 12 - Configure Attendant Consoles with NAS keys.

Prompt	Response	Description
REQ:	NEW CHG	Add new data, or change existing data.
TYPE:	2250	Attendant Console type.
TN		Terminal number
	l s c u	Format for Large System and Avaya Communication Server 1000E (Avaya CS 1000E) system, where l = loop, s = shelf, c = card, u = unit.
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.

Table 451: 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:	NET	Networking Data.

Prompt	Response	Description
CUST		Customer number
	0-99	Range for Large System and CS 1000E system.
	0-31	Range for Media Gateway 1000B.
- 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.
- ICI	xx	Respond with the key number, from 0 to 19, followed by a space, followed by the trunk type.
	NCO NDID NTIE NFEX NWAT	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.

Table 452: LD 17 - Configure the D-channel interface (DCHI) for NAS.

Prompt	Response	Description
REQ	NEW	Add new data.
	CHG	Change existing data.
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.

Table 453: Task 4

Prompt	Response	Description
REQ	NEW	Add new data.
	CHG	Change existing data.
CUST	xx	Customer number, as defined in LD 15.
...		
FEAT	NAS	Specifies type of data to be changed.

Prompt	Response	Description
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 can 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	xxxxxxxxxxx xxxxx	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.

Table 454: LD 93 - Assign a NAS routing table to Console Presentation Groups.

Prompt	Response	Description
REQ	NEW	Add new data.
	CHG	Change existing data.
TYPE	CPG	Console Presentation Group.
CUST	xx	Customer number, as defined in LD 15.
CPG	1-63	Console Presentation Group number.
CPGS	(NO) YES	Customer Presentation Group Services.
ROUT		Route number
	0-511	Range for Large System and CS 1000E system.
	0-127	Range for Media Gateway 1000B.
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).

Table 455: 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	Add new data.
	CHG	Change existing data.
TYPE	RDB	Route Data Block.
CUST	xx	Customer number, as defined in LD 15.
ROUT		Route number
	0-511	Range for Large System and CS 1000E system.
	0-127	Range for Media Gateway 1000B.
TKTP	aaa	Trunk Type requires response when REQ=NEW.
TRMB	(YES) NO	Tromboning. NO = Tromboning denied. Incoming trunk call on route can not be routed back on same route. YES = Tromboning allowed. Incoming trunk call on route can 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 phone is answered, this prompt must be set to NO across the network where this might occur.

Feature operation

No specific operating procedures are required to use this feature.

Chapter 72: Network Call Party Name Display/ Network Name Delivery

Contents

This section contains information on the following topics:

Feature description

Network Call Party Name Display

Network Call Party Name Display (NCPND) provides a network-wide visual display of names and phone numbers to both parties of a call. For phones equipped with an alphanumeric display, NCPND provides the display of the calling party's name on the terminating phone and the called party's name on the calling phone. The name and number display lasts for the duration of the call.

The sending of the names over the private network is an option set up on a per route basis. The name is provided by Call Party Name Display (CPND) configured in each switch. For call redirections, a phone option provides a display of the redirecting name instead of the calling name.

The following phones and attendant consoles are supported:

- M2317 digital phones
- all Meridian Modular Telephones with digit display
- M2250 attendant consoles

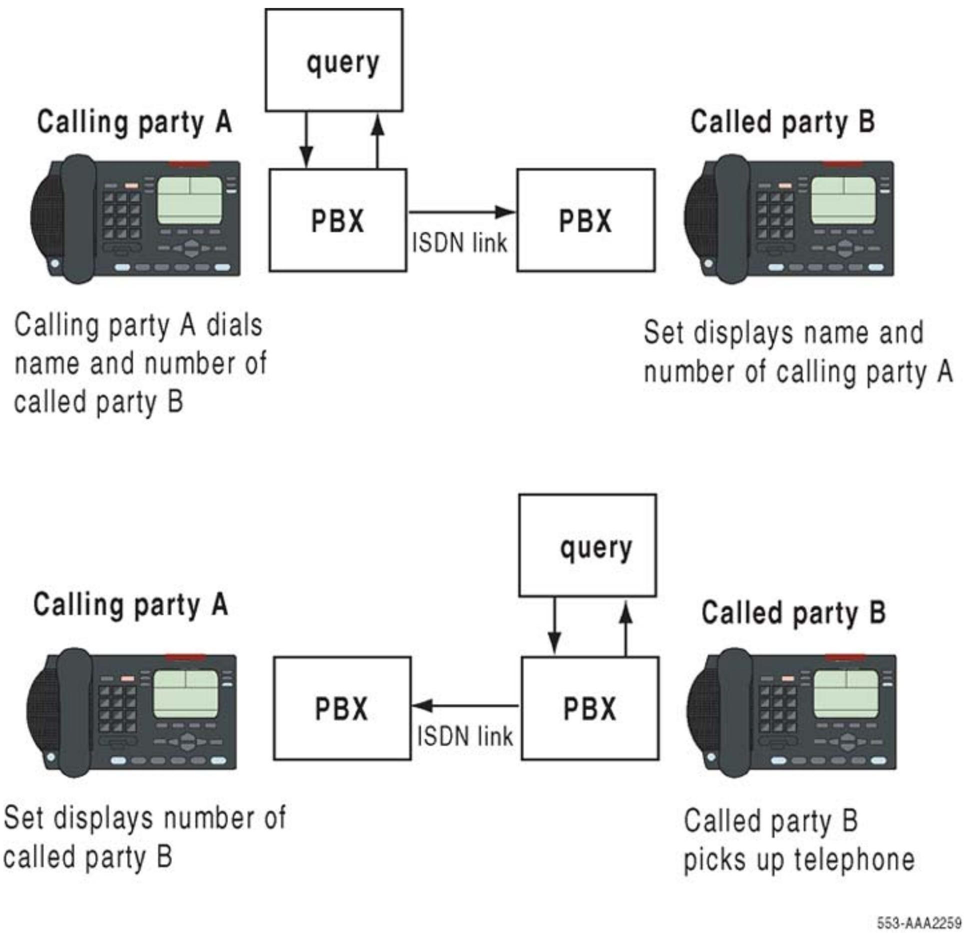


Figure 105: NCPND displays

Network Name Delivery

Network Name Delivery (NND) is the method used to send the names and numbers across the network. Network Name Delivery provides network-wide name display in compliance with the Meridian Customer Defined Networking (MCDN) protocol. It allows interworking among systems and a DMS-100/250 Central Office.

Operating parameters

The following list describes the Network Name Delivery operating parameters.

- For system to system, 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.
- Name Delivery is supported 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 phone 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.

Feature interactions

The same feature interactions apply as those for Call Party Name Display; refer to the Call Party Name Display description in *Avaya Features and Services Fundamentals, NN43001-106*

Feature packaging

This feature requires the following packages:

- Calling Party Name Display (CPND) package 95
- Integrated Services Digital Network (ISDN) package 145
- ISDN Primary Rate Interface (PRI) package 146 or
- 2.0 Mbit Primary Rate Access (PRI2)

Feature implementation

Task summary list

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

1. Table 456: LD 95 - Configure the Calling Party Name Display. on page 982
2. Table 457: LD 16 - Enable NCPND for each required trunk route. on page 982
3. Table 458: LD 17 - Indicate the remote capability (which Network Name Delivery protocol is supported by the remote node/switch on this DCH interface). on page 983

Table 456: LD 95 - Configure the Calling Party Name Display.

Prompt	Response	Description
REQ	NEW	Add new data.
	CHG	Change existing data.
TYPE	CPND	Calling Party Name Display.
CUST	xx	Customer number, as defined in LD 15.
...		
- NITC	aaaa (NI)	Non-Hot Line call. Indicates that the Hot Line call terminated as a normal call.

Table 457: LD 16 - Enable NCPND for each required trunk route.

Prompt	Response	Description
REQ	CHG	Change.
TYPE	RDB	Route Data Block.
CUST	xx	Customer number, as defined in LD 15.
ROUT		Route number
	0-511	Range for Large System and Avaya Communication Server 1000E (Avaya CS 1000E) system.
	0-127	Range for Media Gateway 1000 B (Avaya MG 1000B).
- ISDN	YES	ISDN option.

Prompt	Response	Description
- - NCNA	(YES) NO	Allow Network Call Name Display.

Table 458: LD 17 - Indicate the remote capability (which Network Name Delivery protocol is supported by the remote node/switch on this DCH interface).

Prompt	Response	Description
REQ	CHG	Change
TYPE	ADAN	Type of data block
ADAN	CHG DCH xx	Change D-channel.
...		
- RCAP	ND1 ND2	Network Name Delivery method 1 (ND1). Network Name Delivery method 2 (ND2).

Feature operation

No specific operating procedures are required to use this feature.

Chapter 73: Network Call Redirection

Contents

This section contains information on the following topics:

Feature description

The Network Call Redirection feature provides Network Call Forward No Answer (NCFNA) and Network Call Forward All Calls (NCFAC) over an ISDN PRA or ISL network. Calls can also be transferred over the network, but the CLID display will not reflect the transfer.

The Network Call Redirection (NCRD) feature is based on the stand-alone Call Redirection feature. Stand-alone Call Redirection permits redirection within a single system. Using Network Call Redirection, calls are redirected over more than one system. The user cannot tell the difference between a call-redirected and a network call-redirected call. The CLID digit and the name displays are the same for both call scenarios.

NCRD supports system to system connections. The private numbering plans, the Uniform Dialing Plan (UDP), and the Coordinated Dialing Plan (CDP) are supported by both network configurations.

For NCFNA, the call can pass through (hop) only one switch. However, as shown in Figure 106: Network configurations for NCFAC and NCFNA on page 986, a multiple-hop configuration is permitted for NCFAC.

The number of times a call is redirected depends on the Class of Service of the redirecting phone. If the redirecting phone has a Class of Service with second-level NCFNA denied, only one NCFNA is allowed. If a second NCFNA attempt is made, it is rejected and the call continues to ring at the first redirected station. If the redirecting phone has a Class of Service with second-level NCFNA allowed, the call can have an additional CFNA redirection.

Note:

A redirected call over PRA that terminates on a busy phone cannot activate the Network Ring Again (NRAG) feature. Also, CLID name display is not supported in call transfer, call forward busy, hunt, or call pickup.

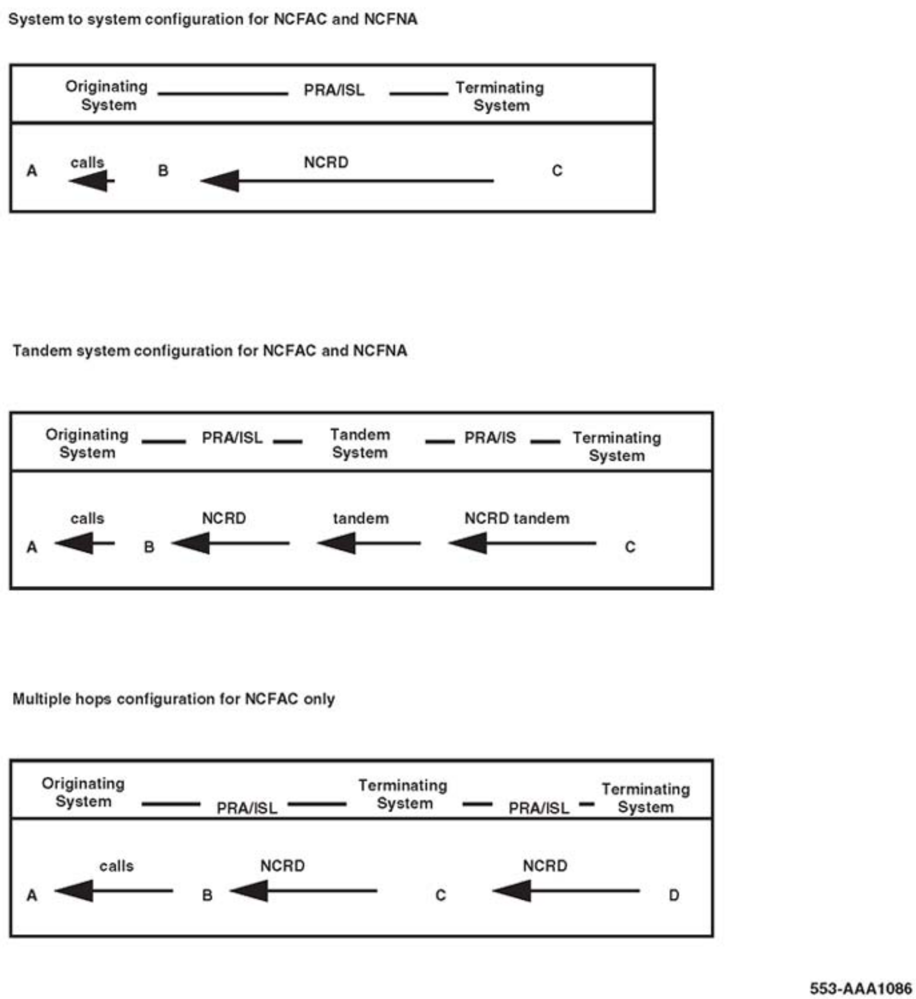


Figure 106: Network configurations for NCFAC and NCFNA

Call redirection terminology

There are four parties involved in a call redirection: the originating party, the originally called party, the redirecting party, and the terminating party.

If A calls B and B redirects the call to C, then:

A is the originating party

B originally called party and the redirecting party

C is the terminating party

With additional redirections, the terminology changes. For example, if A calls B and B redirects the call to C and C redirects the call to D and D redirects the call to E, then:

A is the originating party

B is the originally called party

C s the no name party

D is the redirecting party

E is the terminating or redirection party

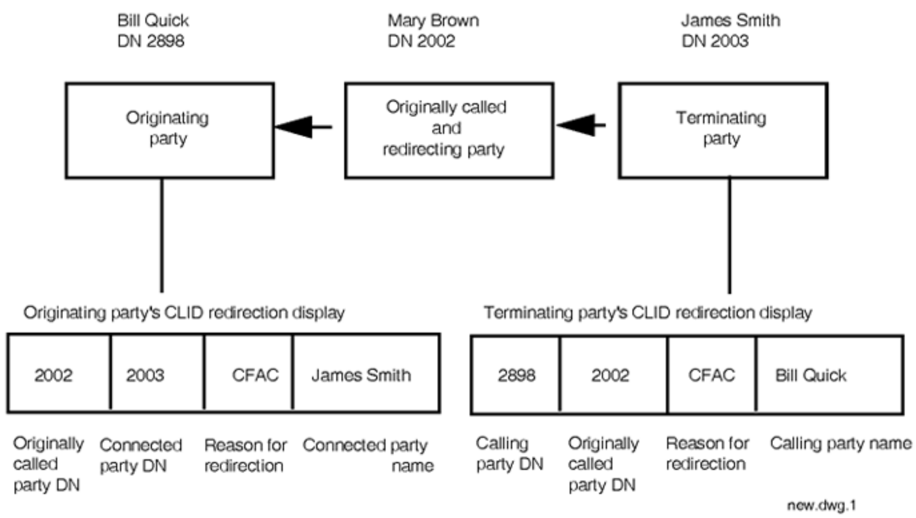


Figure 107: Simple redirection

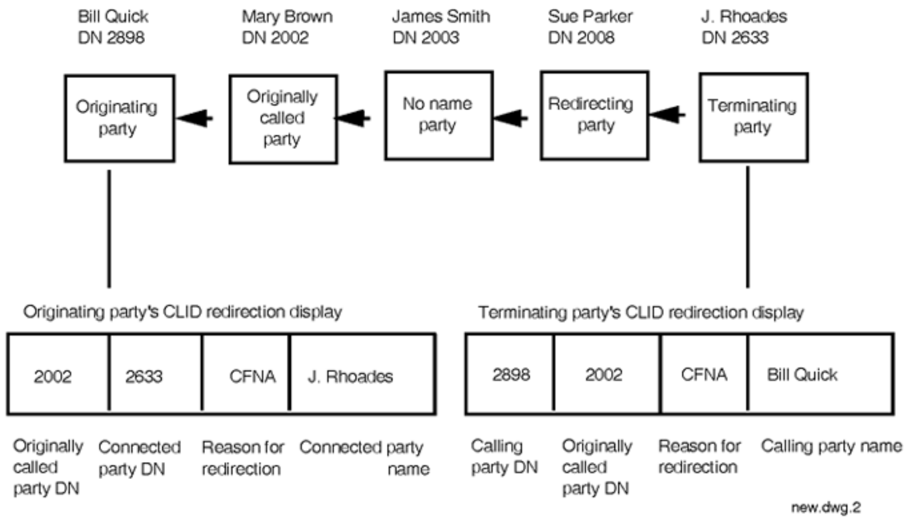


Figure 108: Multiple redirections

Notifying originating party of redirection

As shown in Figure 109: Originating party CLID redirection display on page 988, if the originating party is a subscriber to the Calling Line Identification (CLID) service, their phone displays a Call Redirection reason. This format is like that of internal call redirection CLID.

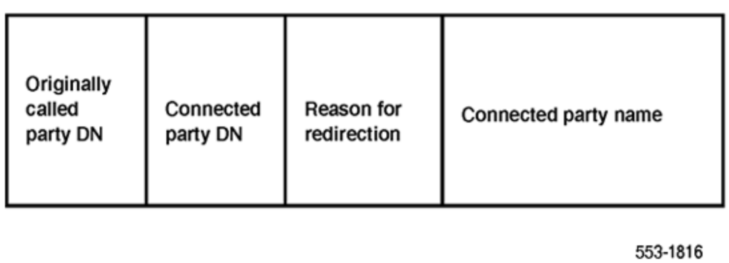


Figure 109: Originating party CLID redirection display

The reason field of the CLID display indicates why the call is redirected out of the original called party, that is either NCFAC or NCFNA. The reason mnemonic displayed is assigned by the customer in LD 95. See the PRA Administration document for a procedure about how to program the Network Call Redirection (NCRD) feature.

If the originally called party information is not available, the redirecting party DN is displayed in place of the originally called DN. For single call redirection, the originally called party is also the redirecting party.

Notifying terminating party of redirection

As shown in Figure 110: Terminating party CLID redirection display on page 989, if the terminating party is a subscriber to the CLID service, their phone displays a call redirection reason. The display format for the terminating party is the same as that for an internal Call Redirection.

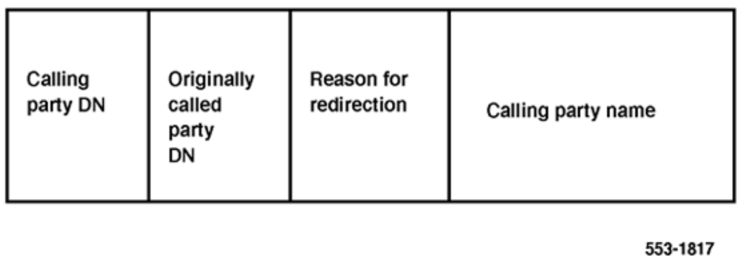


Figure 110: Terminating party CLID redirection display

If the originating party information is not available, the redirecting party DN is displayed instead.

Redirection tones

The tone the originating party receives is determined by the cause of the redirection. When a call cannot terminate because the forward-to DN is busy and none of the redirections are due to Network Call Forward No Answer (NCFNA), the originator receives a busy tone. However, if one redirection is due to NCFNA, an attempt is made to re-ring the phone that initiated the NCFNA.

When the call cannot terminate for any reason other than the forward-to DN is busy and no redirections are due to NCFNA, the originator receives an overflow tone. Again, if one redirection is due to NCFNA, an attempt is made to re-ring the phone that initiated the NCFNA.

A redirection counter value is passed with the call-forwarding information. When the redirection counter maximum is exceeded, there are two scenarios. If all call redirections are due to Network Call Forward All Calls (NCFAC), the calling party receives an overflow tone. If one of the redirections is due to NCFNA, an attempt is made to re-ring the phone that initiated the NCFNA.

Network Call Redirection configurations

The following figures are examples of typical Call Redirection configurations. The text associated with each figure explains the scenarios. These scenarios are shown:

- Intranode NCFAC
- Internode Tandem NCFAC
- Tandem NCFAC and Intranode redirection
- Tandem NCFAC
- NCFNA
- Tandem NCFNA

Note:

Call Redirection is supported only in system to system call connections.

Intranode NCFAC redirection

As shown in Figure 111: NCFAC intranode redirection on page 991, the following occurs in a NCFAC scenario:

- Station A calls Station B on another node. Station B has the Call Forwarding All Calls feature and forwards to Station C.
- The call is then forwarded to Station C. Station C resides in the same terminating node as Station B.
- The terminating node, Station C, sends the CLID display information to Station A. When Station C answers the call, a message is sent to the originating node, Station A.

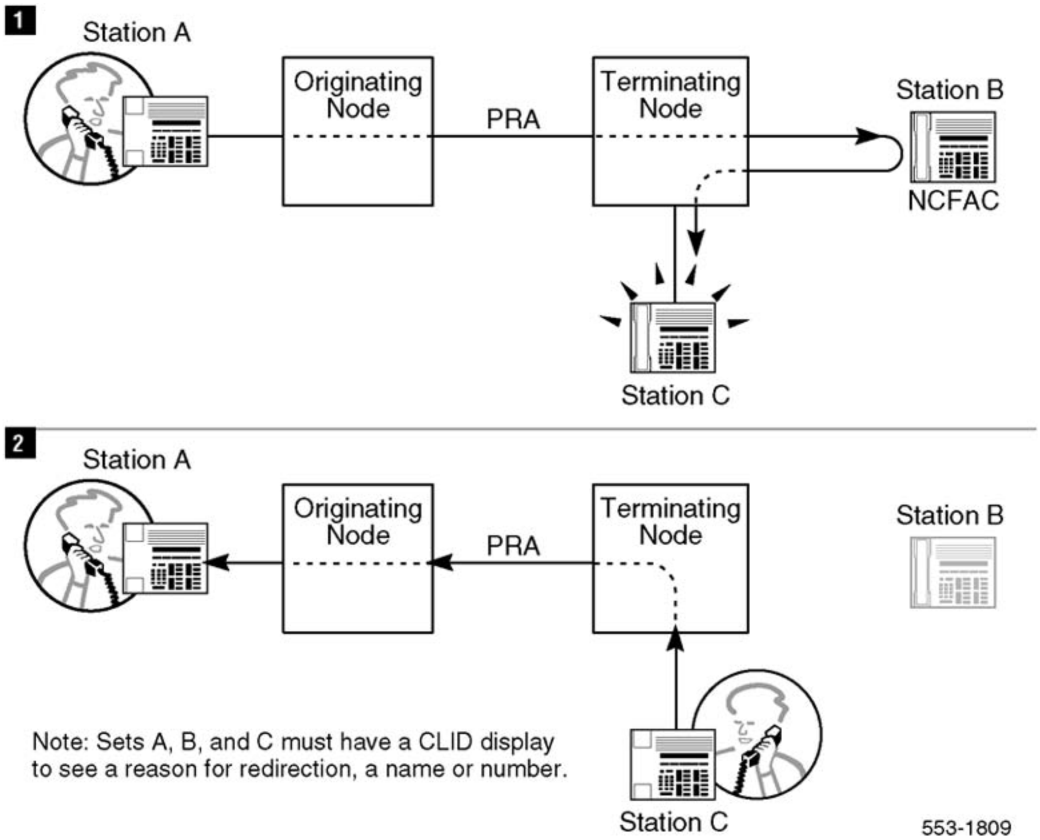


Figure 111: NCFAC intranode redirection

Internode NCFAC redirection

As shown in Figure 112: Internode NCFAC on page 992, the following occurs in an internode NCFAC redirection scenario:

- Station A generates a call to Station B. Station B has the Call Forwarding All Calls feature and forwards to Station C. Station C is located on another switch, making this an internode call.
- A message that contains the called number, calling number, original called number, original redirection reason, and the redirection counter is sent with the call.
- When Station C answers the call, a message is sent to the originating node indicating this response.

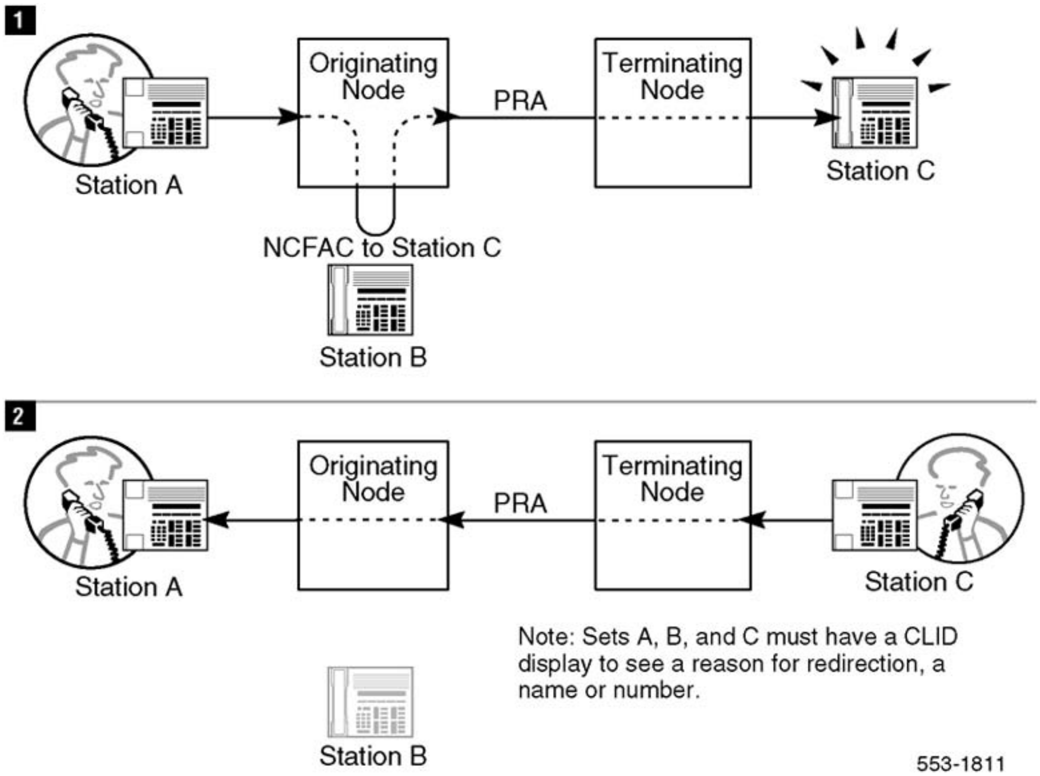


Figure 112: Internode NCFAC

Tandem NCFAC redirection

As shown in Figure 113: Tandem NCFAC on page 993, the following scenario occurs in tandem NCFAC redirection.

- Station A generates an internode call to Station B. Station B has the Call Forwarding All Calls feature and forwards to Station C on another switch.
- The call is translated at the tandem node and then forwarded to Station C at the terminating node.
- The message that accompanies the call from the tandem to the terminating node contains the called number (Station C), the calling number (Station A), original called number (Station B), the original reason for redirection (NCFAC), and the redirection counter that has a value of 1.
- The terminating node sends a message to the originating node that contains the call forwarding information along with an indication of ringing Station C.
- When Station C answers the call, a message indicating this response is sent first to the tandem node and then relayed to the originating node.

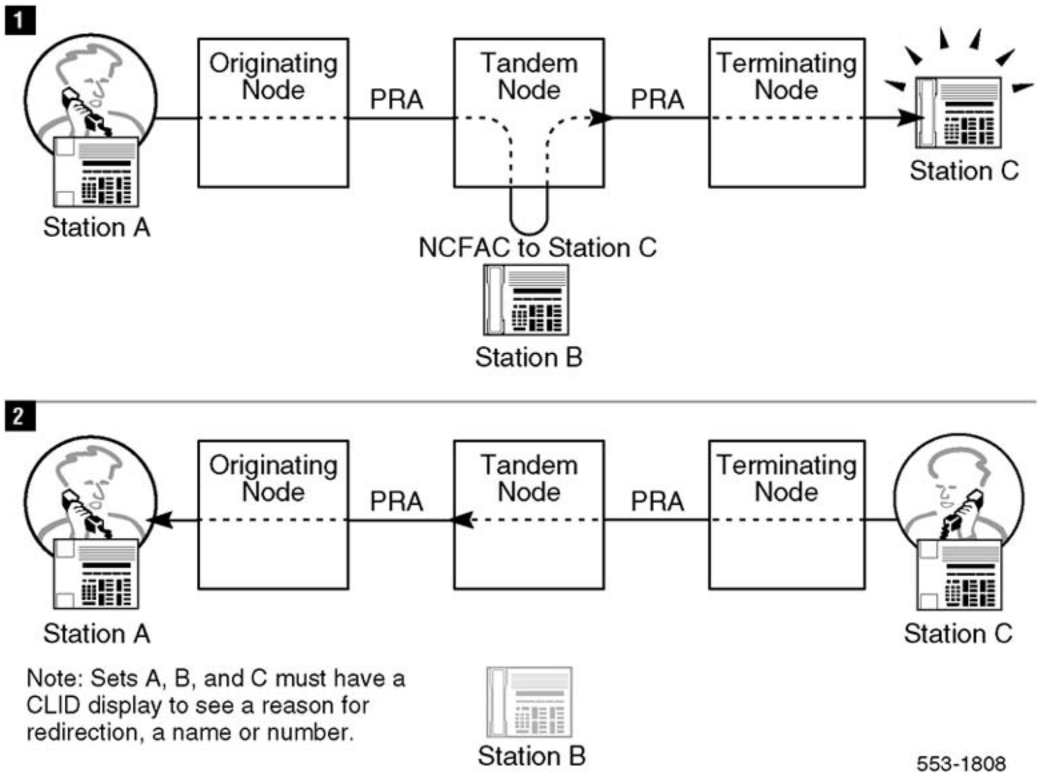


Figure 113: Tandem NCFAC

Tandem NCFAC and intranode redirection

The following occurs in a tandem NCFAC. This is followed by an intranode redirection scenario, as shown in Figure 114: Tandem NCFAC intranode on page 994.

- Station A generates an internode call to Station B. This Station B has the Call Forwarding All Calls feature and forwards to Station C on another node. In this scenario, Station C has the Call Forwarding All Calls feature and forwards the call to station D on the same node.
- The call is forwarded from the tandem node to the terminating node for Station C. The CLID information is also sent as a message along with the call. This information includes the called number (Station C), the calling number (Station A), the original called number (Station B), the original redirection reason (CFAC or NCFAC), and the redirection counter (1).
- The call is then forwarded to station D within the terminating node. The terminating node sends the call forwarding information to the originating node (Station A).
- The redirection counter is still 1 since there was a single network redirection.
- Station D answers and a message is sent notifying the originating node of this response.

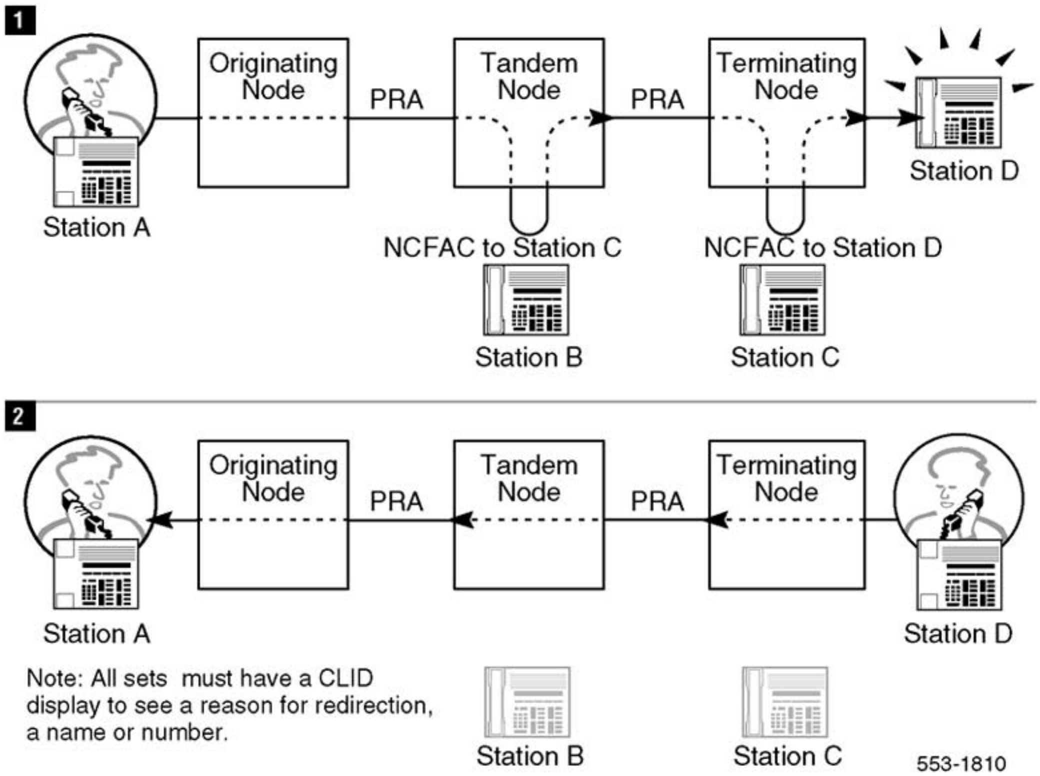


Figure 114: Tandem NCFAC intranode

NCFNA redirection

As shown in Figure 115: NCFNA on page 995, the following occurs in an NCFNA redirection scenario.

- Station A generates an internode call to Station B which has the Call Forwarding No Answer feature and forwards to Station C.
- The call is transferred to Station C when the ringing (or alerting) phase times out. The terminating node sends a message to the originating node that contains the redirection number (Station C) and the redirection reason (NCFNA).
- When Station C answers the call, the terminating node generates a message to the originating node indicating this response.

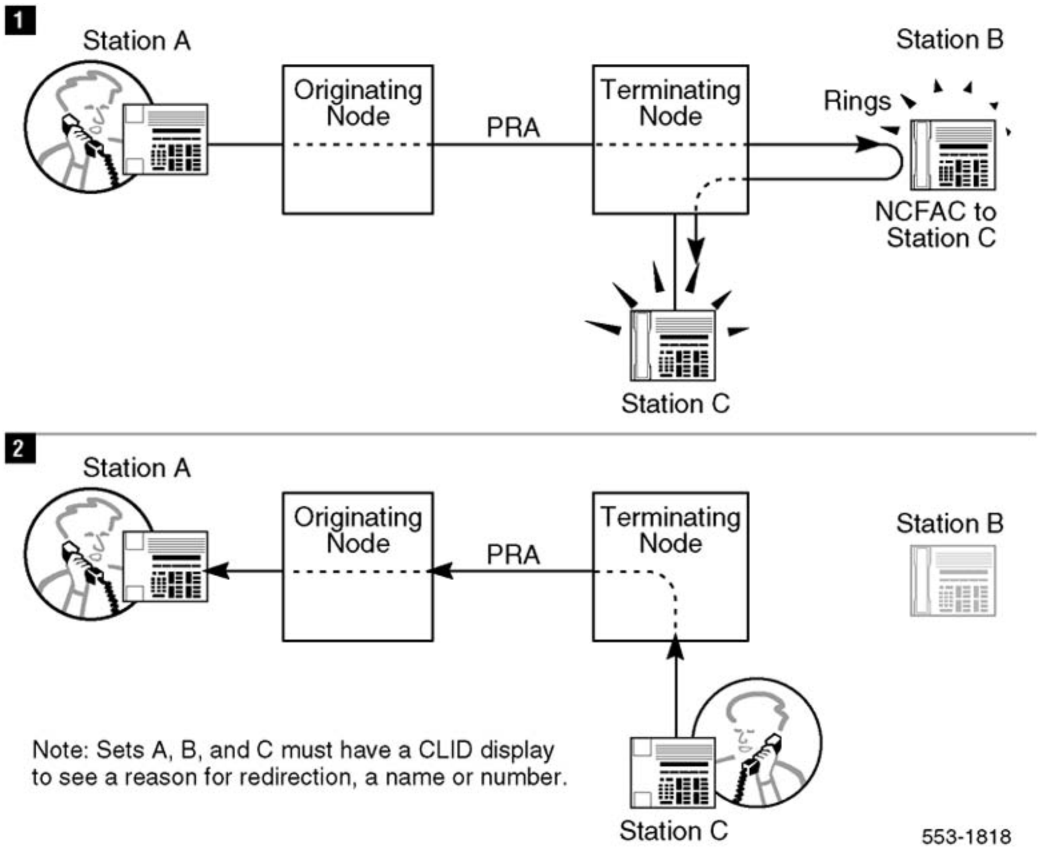


Figure 115: NCFNA

Tandem NCFNA redirection

As shown in Figure 116: Tandem NCFNA on page 996, the following occurs in a tandem NCFNA redirection scenario.

- Station A generates an internode call to Station B which has the Call Forwarding No Answer feature and forwards to Station C.
- The call is transferred to Station C when the ringing (alerting) phase times out.
- The message sent to the terminating node contains the called number (Station C), calling number (Station A), original called number (Station B), original reason for redirection (NCFNA), and the redirection counter with a value of 1.
- The terminating node sends a message to the tandem node which relays the message to the originating node with the redirection number (Station C) and the reason for redirection.

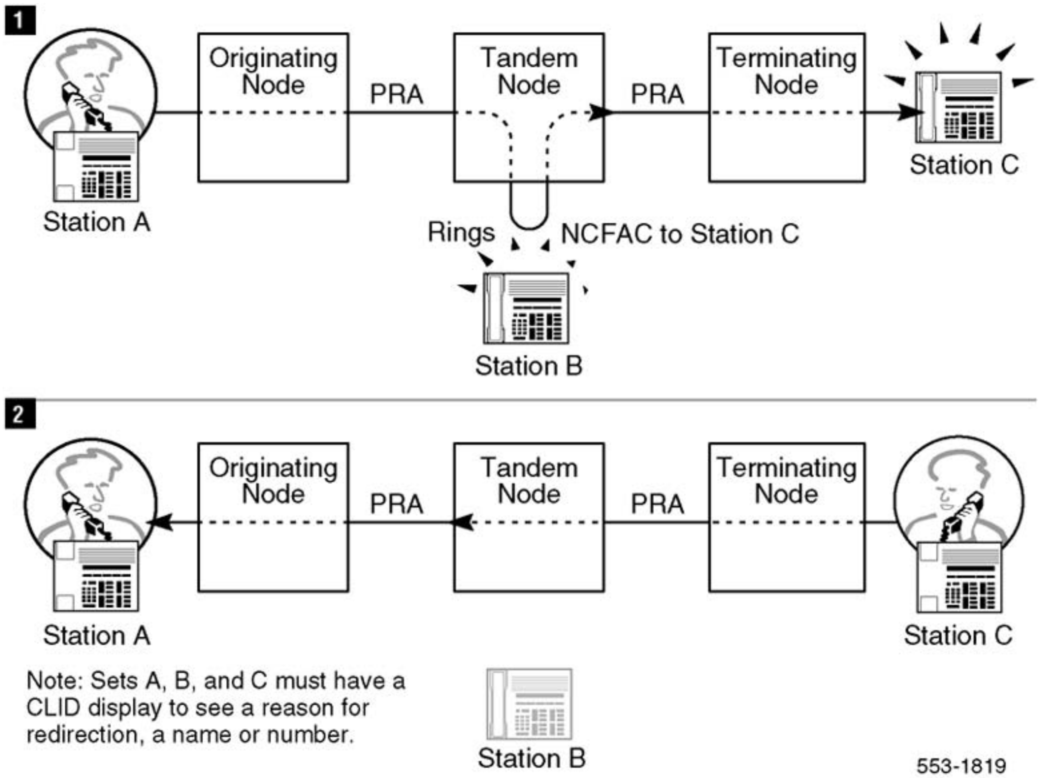


Figure 116: Tandem NCFNA

Operating parameters

There are no operating parameters associated with this feature.

Feature interactions

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

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

DPNSS1 Route Optimisation/MCDN Trunk Anti-Tromboning Interworking

If Network Call Redirection is not configured in a 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 phones on the MCDN side. However, if the bridged phones are on the same node, the displays are updated, even though NCRD is not configured.

EuroISDN Trunk - Network Side

It is possible to have a phone 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 can block some transfers from being completed.

Notices of call redirection or call modification are not transmitted over a EuroISDN Trunk - Network Side connectivity.

INIT ACD Queue Call Restore

Call information associated with Network Call Redirection is lost after system initialization and call restoration.

ISDN QSIG Basic Call

When a call is terminated on the system and Network Call Redirection (NCR) is active, the QSIG Basic Call can still operate; however, the original called number and redirection number IE that are used by NCR will not be sent on the QSIG interface.

Feature packaging

There are no packaging requirements specified for this feature.

Feature implementation

Task summary list

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

Table 459: LD 15 - Forward calls to a forwarding DN.

Prompt	Response	Description
REQ:	CHG	Change existing data block.
TYPE:	CDB RDR	Customer Data Block. Call Redirection data (Gate opener).
CUST		Customer number
	0-99	Range for Large System and Avaya Communication Server 1000E (Avaya CS 1000E) system.
	0-31	Range for Media Gateway 1000 B (Avaya MG 1000B).
- FNAD	FDN	Call forward no answer DID calls—Flexible CFNA DN.
- FNAT	FDN	Treatment for External CFNA calls (non-DID—when FDN is selected, CFCT handles the call.
- FNAL	FDN	Requests treatment for CFNA—when FDN is selected, DID calls are forwarded.

Table 460: LD 16 - Allow Network Call Redirection.

Prompt	Response	Description
REQ	CHG	Change
TYPE	RDB	Route Data Block

Prompt	Response	Description
CUST	xx	Customer number, as defined in LD 15.
ROUT		Route number
	0-511	Range for Large System and CS 1000E system.
	0-127	Range for Media Gateway 1000B.
NCNA	(NO) YES	Network Call Name is (is not) allowed.
NCRD	(NO) YES	Network Call Redirection. Allows network call redirection messages to be sent (or blocks messages if NCRD= NO)
		Network Call Redirection can occur without answering YES to the NCRD prompt. This prompt only controls the sending of Network Call Redirection messages, 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	(NO) YES	Trunk Optimization
		TRO economizes trunk use throughout the network as part of the NCRD feature.

Table 461: LD 95 - Display the reason calls are redirected.

Prompt	Response	Description
REQ	CHG	Change
TYPE	CPND	Call Party Name Display data block
CUST	xx	Customer number, as defined in LD 15.
ROUT		Route number
	0-511	Range for Large System and CS 1000E system.
	0-127	Range for Media Gateway 1000B.
DES	(NO) YES	Designator for Multiple Appearance DNs allowed.
RESN	YES	Allow display of reason for redirecting calls
CFWD	(F) xxxx	Display mnemonic for (Network) Call Forward All Calls. Default is "F." Enter the mnemonic that represents NCFAC on a phone's CLID display.
CFNA	(N) xxxx	Mnemonic for (Network) Call Forward No Answer display. Enter the mnemonic that represents NCFNA on a phone's CLID display. Default is "N."
HUNT	(B) xxxx	Mnemonic for Network Hunting display
PKUP	(P) xxxx	Mnemonic to allow Call Pickup display
XFER	(T) xxxx	Mnemonic for Call Transfer display

Table 462: LD 95 - Give each DN a name.

Prompt	Response	Description
REQ	CHG	Change
TYPE	NAME	Call Party Name Display name entry
CUST	xx	Customer number, as defined in LD 15.
DIG	xxx xx	An existing Dial Intercom Group number (0-253) and member number (0-99)
NAME	aaa...a	CPND name using ASCII characters. The DIG prompt is reprompted. Enter <CR> to get the DN prompt.
DN	xxxx	DN of eligible type.

Table 463: LD 10 - Enable the appropriate feature in the data block.

Prompt	Response	Description
REQ:	CHG	Change
TYPE:	500	Analog (500/2500-type) phone
HUNT	xxxx	Hunt DN for internal calls.
FTR	EFD xxx	External Flexible call forward DN.
		Only allowed if LD15 is properly configured: FNAD = FDN FNAL = FDN FNAT = FDN
		If the DNXP package is equipped, up to 7 digits are allowed; otherwise, only 4 digits can be entered. Accepted only if CLS is MWA or FNA.
	EHT xxxx	External Hunt DN.
		Only allowed if CLS = CFTA.
		Same digits defined as above.
	FDN xxxxxxx	Flexible Call Forward No Answer DN (cannot be an LDN).
		Same digits defined as above.

Table 464: LD 11 - Enable the appropriate feature in the data block.

Prompt	Response	Description
REQ:	CHG	Change
TYPE:	xxxx	Enter phone type
FDN	x..x	Flexible CFNA DN where xx is the MCDN. The FDN value should include AC1/AC2 when applicable (up to 13 digits).

Prompt	Response	Description
EFD	xxxx	Network CFNA DN for External calls.
HUNT	xxxx	Network Hunt DN for calls with CLS = CFTD.
EHT	xxxx	Network Hunt DN for External calls.

Feature operation

No specific operating procedures are required to use this feature.

Chapter 74: Network Call Transfer and Network Extended Calls

Contents

This section contains information on the following topics:

Feature 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, if NCPND is optioned, the calling party's display is updated to show the connected party's name and number.

Note:

Network Call Transfer over PRI does not provide the ESN Network Transfer feature. This feature eliminates tandem trunk connections that double back over the same route. PRI Network Transfer allows calls to be blindly transferred across the ISDN network.

Operating parameters

There are no operating parameters associated with this feature.

Feature interactions

There are no feature interactions associated with this feature.

Feature packaging

This feature is included in base System Software.

Feature implementation

There are no specific implementation procedures for this feature.

Feature operation

No specific operating procedures are required to use this feature.

Chapter 75: Network Drop Back Busy and Off-hook Queuing

Contents

This section contains information on the following topics:

Feature description

The Network Drop Back Busy (DBB) feature allows network calls that are blocked at a tandem node to be rerouted (dropped back) to the originating node. The calls are then directed over an alternate route. The Network Off-Hook Queuing (OHQ) feature allows Off-Hook Queuing to be configured at a tandem node, thereby allowing Off-Hook Queuing at that node.

Both DBB and OHQ give the originating node control over the routing of all outgoing network calls.

The DBB and OHQ capabilities are only supported over an ISDN network. When DBB and/or OHQ is configured, an ISDN call to a tandem node might encounter one of the following conditions.

Table 465: Conditions and treatments of an ISDN call to a tandem node, with DBB and/or OHQ configured

Configuration	Condition	Treatment
IDBB	All routes in initial (I) set busy	Drop back to originating node.
IDBB	All routes in "I" set and extended (E) set busy	If all routes in "I" set busy, attempt routing over "E" set. If all routes in "E" set are busy, drop back to originating node. This is the default configuration.
OHQ, IDBB	All routes in "I" set busy	Off-hook queue. If OHQ timer times out, drop back to originating node.
OHQ, IDBB	All routes in "I" set and "E" set busy	Off-hook queue to "I" set. When OHQ timer times out, attempt routing over "E" set. If all routes in "E" set are busy, drop back to originating node.

An Initial set of routes (I set) are those routes in a route list which have been customer-defined for a node as being inexpensive. The system attempts to complete a call over these routes before testing for queue eligibility. An Extended set of routes (E set) are those routes in a route list that are not part of the initial set. These routes are usually designated as expensive. The system attempts to complete a call over the E Set routes only when the I set queuing times out.

Operating parameters

There are no operating parameters associated with this feature.

Feature interactions

Off-Hook Queuing

Message Intercept

If the Message Intercept feature is equipped, a caller in an off-hook queue can receive the message intercept voice response rather than the Off-Hook Queuing tone.

Drop Back Busy

Remote Virtual Queuing

Drop Back Busy (DBB) and Remote Virtual Queuing (RVQ) are both packaged under Originating Routing Control/Remote Virtual Queuing (ORC-RVQ) package 192. If DBB and RVQ 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 "Feature administration" section in this feature module for more information.

Network Attendant Service

Network Attendant Service (NAS) routing takes precedence over DBB.

Feature packaging

This feature requires the following packages:

- Basic Routing (BRTE) package 14
- Network Class of Service (NCOS) package 32
- Network Alternate Route Selection (NARS) package 58
- Flexible Call Back Queuing (FCBQ) package 61

- Off-hook Queuing (OHQ) package 62 for Off-Hook Queuing
- Integrated Services Digital Network (ISDN) package 145
- 2.0 Mbit Primary Rate Access (PRI2) package 154
- Network Attendant Service (NAS) package 159
- ISDN Supplementary (ISDNS) package 161
- Originating Routing Control/Remote Virtual Queuing (ORC-RVQ) package 192 for Drop Back Busy

Feature implementation

Task summary list

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

1. Table 466: LD 87 - Allow OHQ Network Class of Service for the customer. on page 1008
2. Table 467: LD 86 - Configure the originating node for Remote Virtual Queuing. on page 1009

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 and RVQ 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.

Table 466: LD 87 - Allow OHQ Network Class of Service for the customer.

Prompt	Response	Description
REQ	CHG	Change existing data
CUST	xx	Customer number, as defined in LD 15.
FEAT	NCTL	Network Control
SOHQ	YES	Allow system (customer) Off-hook queuing
- OHTL	nn	Off-hook queue time limit
SCBQ	(NO) YES	Allow (disallow) system (customer) Call Back queueing
...		

Prompt	Response	Description
NCOS	nn	Network Class of Service number. The originating phone must have the same value.
- OHQ	(YES) NO	Off Hook Queuing (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	(NO) YES	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)

Table 467: LD 86 - Configure the originating node for Remote Virtual Queuing.

Prompt	Response	Description
REQ	CHG	Change
CUST	xx	Customer number, as defined in LD 15.
FEAT	RLB	Route List Data Block.
RLI	nn	Route List Index.
ENTR	nn	Route List entry number.
ROUT		Route number
	0-511	Range for Large System and Avaya Communication Server 1000E (Avaya CS 1000E) system.
	0-127	Range for Media Gateway 1000 B (Avaya MG 1000B).
IDBB	(DBD)	Enter DBB (Drop Back Busy Disabled).
	DBI DBA	This will disable Drop Back Busy, and enable Remote Virtual Queuing for the customer. Drop Back if Initial set is busy. Drop Back if all routes are busy.

Feature operation

No specific operating procedures are required to use this feature.

Chapter 76: Network Intercom (Hot Type D and Hot Type I Enhancements)

Contents

This section contains information on the following topics:

Feature description

Hot Line enables a designated phone to place calls to a predetermined destination that can be internal or external to the system. The call does not require attendant intervention. When the handset is lifted or when a preprogrammed key is activated, the system automatically dials 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.

There were two types of Hot Line keys (DN-based Hot Type D and Speed Call List-based Hot Type L). This enhancement 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 stand-alone and network environments.

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 a Hot Line call cannot be completed on the Hot Line key, the calling party is informed through the phone'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 provides the ability for Meridian digital phones to have two-way intercom calls on specially designated keys (not on the DN keys) with other Meridian digital phones connected to PBXs across a Meridian Customer Defined Network (MCDN) Integrated Services Digital Network (ISDN). A Hot Type D call can 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, but the Hot Line key flashes to indicate the call.

Hot Type D allows more than one phone to have the same target DN defined for the Hot Type D key. This enhancement includes Voice, Ringing, and Non-ringing termination modes, as well as 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 it is 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 phone that has a target DN that matches the originating DN, a No Answer Indication is left on that key.

Note:

When configuring two-way Hot Type D keys in voice mode, the fact that the CLID does not transmit the originator's Hot Type D DN between ISDN locations must be taken into account; it contains the prime DN of the originating phone, and not the originator's Hot Type D DN. Therefore, ringing can 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 called phone 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.

Operating parameters

Hot Type I calls are not allowed on analog (500/2500-type) phones.

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 Hot Type D (when the No Answer Indication applies) must be either a Coordinated Dialing Plan or a Universal Dialing Plan number that must terminate on a Prime DN of a Meridian digital phone; otherwise, the call is completed 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.

If the termination mode is voice, the called party is idle, and the handsfree voice call (HVA 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.

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 a 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 an overflow tone. Hot Line calls involving ACD phones 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's phone 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 way 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 can be transferred to another Hot Line key or to a normal DN key; likewise calls on a normal DN key can be transferred to a Hot Line key.

Conference

A Conference call can involve a mixture of intercom and regular DN keys.

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 override this feature (i.e., Hot Type I calls are not filtered by FFC Boss Secretarial filtering). The call terminates on the Boss' phone 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 phone.

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 phone is allocated the same prime DN, the Hot Type I call will terminate on the phone designated as the Multiple Appearance Redirection Prime (MARP). If the MARP DN is not the prime DN on the phone, or if the phone designated as the MARP DN is not a Meridian digital phone, the first Meridian digital phone 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 can 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 though 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 Type I keys and enhanced Hot Type D keys support Vacant Number Routing.

Feature packaging

The following packages are required for Network Intercom (Hot Type D and Hot Type I Enhancements):

- 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:

- Integrated Services Digital Network (ISDN) package 145
- Advanced ISDN Network Services (NTWK) package 148
- at least one of Integrated Services Digital Network Signaling Link (ISL) package 147; ISDN Primary Rate Access (PRA) package 146; 2.0 Mbit Primary Rate Interface (PRI2) package 154; Virtual Network Services (VNS) package 183; ISDN BRI Trunk Access (BRIT) package 233

DNPSSS1 connectivity for Hot Type D requires:

- Integrated Digital Access (IDA) package 122
- Digital Private Signaling System 1 (DPNSS) package 123

R2MFC connectivity for Hot Type D requires Multifrequency Compelled Signaling (MFC) package 128.

Feature implementation

Task summary list

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

1. Table 468: LD 11 - Define Hot Type D and I keys and Classes of Service. on page 1017
2. Table 469: LD 95- Configure the Calling Party Name Display. on page 1018

Table 468: LD 11 - Define Hot Type D and I keys and Classes of Service.

Prompt	Response	Description
REQ:	NEW CHG	Add new data, or change existing data.

Prompt	Response	Description
TYPE:	a...a	Telephone type. Type **?** for a list of possible responses.
TN		Terminal number
	l s c u	Format for Large System and Avaya Communication Server 1000E (Avaya CS 1000E) system, where l = loop, s = shelf, c = card, u = unit.
CLS	FICA (FICD) NAIA (NAID)	Forward Hot Type I allowed. Forward Hot Type I denied. No Answer indication allowed. No Answer indication denied.
...		
KEY	nn HOT D dd target_num hot_dn	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.
	R V N (H)	Termination mode: Ringing Voice Non-ringing Hot Line
KEY	nn HOT I dd target_num ber	Hot Line type I key. nn = key number. dd = number of digits dialed. target_number = terminating DN (31 digits maximum).
	(V) N R	Termination mode: Voice Non-ringing Ringing

Table 469: LD 95- Configure the Calling Party Name Display.

Prompt	Response	Description
REQ	NEW	Add new data.
	CHG	Change existing data.
TYPE	CPND	Calling Party Name Display.
CUST		Customer number
	0-99	Range for Large System and CS 1000E system.
	0-31	Range for Media Gateway 1000B.
- NITC	aaaa (NI)	Non-Hot Line call. Indicates that the Hot Line call terminated as a normal call.

Feature operation

Press the Hot Type I or D key to initiate a Hot Line call to a target DN (the DN can be an external DN in an MCDN ISDN network). The called party answers the call by pressing their Hot Type

I or D key if configured. If the called party has no Hot Type I or D key configured, the call will behave as a normal call and is answered accordingly.

If the called party does not answer, and has No Answer Indication Allowed Class of Service, the Hot Type I or Hot Type D key winks as a form of No Answer Indication.

Chapter 77: Network Time Synchronization

Contents

This section contains information on the following topics:

Description

The Network Time Synchronization feature is designed to ensure that all time stamps in a network are synchronized from one source. One switch becomes the master for this purpose.

In a private network environment, each switch in the network has an individual system clock. These system clocks can, under certain conditions, lose or gain time, causing inaccurate time stamps for different features.

Also, in a private network, several switches can be located in different time zones. As features become more centralized in a network environment, it is useful to have time stamps based on one time zone.

To provide Time Synchronization on a network-wide basis, Meridian Customer Defined Integrated Services Digital Network (ISDN) nodes can request Time Synchronization from another node, using D-channel messages. Therefore, Slave switches can request the time from the Master switch, while the Master switch can do the same to a Backup node.

A time difference (or delta) is provided for every node, in order to distinguish time zones for time usage by local features (e.g., Automatic Wake-up) and centralized ones (e.g., Centralized Call Detail Recording).

The Time Synchronization request messages are composed of:

- the message identifier
- the requester's ID
- the time
- the date
- the time-adjust factor

IDs are virtual DNs, and are used to route the messages.

On the Slaves, Time Synchronization requests are sent automatically under the background routines (default setup) or with the daily routines (optional setup), every time a time change is performed (to accurately set the seconds), and on every SYSLOAD and initialization.

On the Master, Time Synchronization requests are sent to a Backup node upon initialization and, therefore, after SYSLOAD. The Master will be forbidden to synchronize Slave switches during these backup periods. In the rare event where Master and Backup nodes would start requesting synchronization at the same time, the real time will be considered to be on the Slave node, if it is not initializing. If both nodes are only initializing, the real time would be considered to be carried by the Master switch. If a SYSLOAD occurs at the Master and the Slave is initializing, the real time would be considered to be carried by the Slave switch. A warning message will be printed if both switches SYSLOAD, and all Time Synchronization would be put to a halt until the Master's clock is reset.

If no answer is received on the first time synchronization request (by the end of the request time-out), extra Time Synchronization requests will be sent. If there is still no answer on the third time synchronization request, a warning message will be issued.

Note:

Upon SYSLOAD, the clock starts on time zero, while upon initialization, only the seconds are lost.

Note:

Through service change (LD 2) or through an Attendant Console, the clock can be reset to the correct time if desired. If the Network Time Synchronization feature is on, then the Master will be requested for synchronization upon these service changes (to permit fine synchronization).

Operating parameters

This feature uses D-channel messaging over a Meridian Customer Defined Integrated Services Digital Network (ISDN).

Feature interactions

Time-of-day Adjustment

Every time LD 2 is used to change the system time, a request for synchronization will be made of the Master to accurately set the seconds.

Time and Date (TAD) Attendant key

As done with LD 2, every time the TAD key is used to change the system time, a request for synchronization will be made to the Master to accurately set the seconds.

Call Detail Recording (CDR)

Upon receipt of synchronization messages, Slave switches will issue CDR records (if so equipped) for monitoring the feature. These CDR records will be identical to those issued by a time change performed in LD 2 or by an attendant's TAD key.

Feature packaging

This feature is included with the Integrated Services Digital Network Supplementary Features (ISDNS) package 161.

Feature implementation

The following parameters are used to configure Network Time Synchronization:

- Node Status: Can either be Master, Slave, or standard stand-alone node (MAST, SLAV or STDA).
- Customer Number: Customer that will issue and receive the Network Time Synchronization messages. The default value is "0". For a change (only possible on switches with the multi-customer package) to be accepted, the customer should already exist. Furthermore, the Local DN has to be reentered.
- Local DN: Virtual DN (access code included) dedicated for synchronization services on the Local node; up to 16 digits. A call with these routing digits should terminate on the previously designated customer.
- Master or Backup DN: Virtual DN (access code included) dedicated for synchronization services on the Master or Backup node; up to 16 digits.
- Time Delta: Time difference added to the local time in order to get the Master or Backup time. The entry is prefixed with the digit 1 for positive and 0 for negative.
- Requesting mode: Operating mode used to request the time synchronization messages (i.e., with the background routines (default setup) or with the daily services (BKGD or DSVC)).

Note:

The Node Status, Customer Number, Local DN, and Master or Backup DN parameters must be configured for the Network Time Synchronization feature to be operational.

Task summary list

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

1. LD 2 - Define entries for the Network Time Synchronization feature. on page 1024
2. LD 22 - Print the DN type for the Network Time Synchronization Virtual DN. The DN type is TIME. on page 1027

LD 2 - Define entries for the Network Time Synchronization feature.

The following commands are Network Time Synchronization feature specific:

Query Node Status (Type Time Synchronization Status).

The command format is:

INPUTOUTPUT

.TTSS.TTSS (STATUS)

Example:

.TTSS.TTSS MAST

Set Node Status (Set Time Synchronization Status).

The command format is:

.STSS (status) where status can be: STDA - stand-alone (default) MAST - Master SLAV - Slave

Example:

.STSS SLAV

Query Customer in charge (Type Time Synchronization Customer).

The command format is:

INPUTOUTPUT

.TTSC.TTSC (CUSTOMER NUMBER)

Example:

.TTSC.TTSC 5

Set Customer in charge (Set Time Synchronization Customer).

The command format is:

.STSC (customer number) where customer can be: 0 - 99 - 0 is default.

Example:

.STSC 5

Query Local Virtual DN (Type Local DN).

The command format is:

INPUTOUTPUT

.TLDN.TLDN (DN)

Example (for 6 = ESN access code, 613 = ESN location code, 5999 = DN):

.TLDN.TLDN 66135999

Set Local Virtual DN (Set Local DN).

The command format is:

.SLDN (dn)

Example:

.SLDN 66135999

Query Master or Backup Time Synchronization Number (Type Master DN).

The command format is:

INPUTOUTPUT

.TMDN.TMDN (DN)

Example (for 6 = Outside line, 514 = ESN code, 3999 = DN):

.TMDN.TMDN 65143999

Set Master or Backup Time Synchronization Number (Set Master DN).

The command format is:

.SMDN (dn)

Example:

.SMDN 65143999

Query Time Delta.

The command format is:

INPUT OUTPUT

.TDEL.TDEL (SIGN) (HR) (MIN)

Example:

.TDEL.TDEL 0 01 30

Set Time Delta.

The command format is:

.SDEL (sign) (hr) (min)

- sign - is the time-adjust factor direction indicator which can be: 0 - to indicate the Master switch is behind in time. or 1 - to indicate the Master switch is ahead in time.
- hr - is the number of hours the time must be adjusted by and can be any number from 0 to 23, and
- min - is the number of minutes the time must be adjusted by and can be any number from 0 to 59.

0 is the default for the SDEL parameters.

Example:

.SDEL 1 23 00

Note:

The hour and minute entries are two digits. The minute entry is defaulted to zero if not entered. "1" identifies a positive delta (Master is ahead in time), "0" identifies a negative delta.

Query Requesting Mode (Type MODe).

The command format is:

INPUT OUTPUT

.TMOD.TMOD (MODE)

Example:

.TMOD.TMOD BKGD

Set Requesting Mode (Set MODe).

The command format is:

.SMOD (mode) where mode can be: BKGD - Background (default) or DSVC - Daily Service (midnight)

Example:

.SMOD DSVC

LD 22 - Print the DN type for the Network Time Synchronization Virtual DN. The DN type is TIME.

Feature operation

Each node of the network that is to be synchronized sets its status (i.e., Master, Slave, or Standard stand-alone (the feature is not used)). The craftsperson also sets the node's customer in charge of synchronizing the switch (that customer will request Time Synchronization and receive the time from the Master or Backup switch). The customer must already exist, prior to referencing it, and then sets the local access codes with the virtual DN, the Master (or Backup) routing digits, the time difference between local and Master nodes, and the requesting mode (for example, performed under the background routines (default) or the daily services. The time delta and the requesting mode are optional entries).

The time synchronization feature is designed to work in a Meridian Customer Defined Integrated Services Digital Network (ISDN) environment, using D-channel messages. The synchronization messages are carried by TCAP facility messages on a Meridian Customer Defined ISDN, and routed according to the configuration defined in LD 2.

Once the configuration is defined in LD 2, the Slaves will automatically start requesting "time-stamps" from the Master periodically, upon initialize, and when the time and date is changed in LD 2. The message to be sent is identified as being a time synchronization request. The stored Master's routing digits (access codes + virtual DN) are used to route the synchronization requests. As part of the request, the requester's access codes + virtual DN is sent to provide the Master with a way back to the requesting node.

Upon receipt at the Master node, the terminating DN is recognized as being a time synchronization virtual DN, and the request is then processed. The processing consists of constructing a message, identified as being a time synchronization response and originating from that virtual DN, which includes:

- time
- date, and
- time-adjust factor (if used).

The message is then routed to the Slave using the access codes and virtual DN provided with the request. The time-adjust factor is sent to all Slaves in order to have the whole network correct any inaccurate clock settings in unison, if any slippage correction is necessary.

Up to three requests will be sent at one minute intervals, to allow the system to overcome possible temporary malfunctions. On receipt of the time message, the requester verifies the originator's ID (its virtual DN), and updates its clock accordingly (i.e., equal to the time sent minus the time delta between the two switches). If equipped with the CDR package, a time change record is provided with every time synchronization occurrence.

Chapter 78: Network-wide Listed Directory Number

Contents

This section contains information on the following topics:

Feature description

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 DNs are dialed. Without this feature, it was only possible to define four LDNs on a system. This feature makes it possible to define six LDNs on a system.

Without this feature, when an LDN call was routed from one node to another, the call was presented according to trunk type (for example, NDID, NTIE, NCO, NFEX, or NWAT). The call was presented on LDN key zero if none of the trunk route ICIs were configured. With the Network-wide Listed Directory Number feature, if the dialed DN is an LDN and an LDN key exists that corresponds to the dialed LDN, the call is 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.

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 through 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 for each 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 can 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

Network Wide LDN requires Network Attendant Service routing. The following packages are required for Network-wide LDN:

- Basic Routing (BRTE) package 14
- Network Class of Service (NCOS) package 32
- Network Alternate Route Selection (NARS) package 58
- Network Attendant Service (NAS) package 159
- 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.

Feature implementation

Task summary list

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

1. Table 470: LD 15 - Configure LDN data. on page 1032
2. Table 471: LD 15 - Enable the network recognition of the LDNs. The ICI keys can also be assigned to the new LDN values. on page 1033
3. LDs 10, 11 - The LDN prompt has been changed to accept a value of 0-5 in these overlays.

Table 470: LD 15 - Configure LDN data.

Prompt	Response	Description
REQ:	NEW CHG	Add new data, or change existing data.
TYPE:	LDN	Departmental Listed Directory Numbers data.
CUST		Customer number
	0-99	Range for Large System and Avaya Communication Server 1000E (Avaya CS 1000E) system.
	0-31	Range for Media Gateway 1000 B (Avaya MG 1000B).
- OPT	aaa	Options.
- DLDN	YES	Departmental Listed Directory Numbers.
- LDN3	...	Listed DN 3.
- 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.

Prompt	Response	Description
- LDN5	x...x	Listed Directory Number. If the DNXP package is equipped, up to seven digits are allowed; otherwise, only four digits are allowed.

Table 471: LD 15 - Enable the network recognition of the LDNs. The ICI keys can also be assigned to the new LDN values.

Prompt	Response	Description
REQ:	NEW CHG	Add new data, or change existing data.
TYPE:	LDN	Departmental Listed Directory Number data.
CUST		Customer number
	0-99	Range for Large System and CS 1000E system.
	0-31	Range for Media Gateway 1000B.
- 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.

Feature operation

No specific operating procedures are required to use this feature.

Chapter 79: NI-1 BRI Compliance Enhancements

Contents

This section contains information on the following topics:

Feature description

This feature enhances the system functional protocol to be compatible with the National ISDN-1 (NI-1) ISDN Basic Rate Interface (BRI) voice and data standard for line application. The NI-1 protocol is configured in Overlay 27, which is used for ISDN BRI administration, by entering "6" against the PRID prompt. This protocol information is downloaded to the Multi-purpose ISDN Signaling Processor's (MISP) network layer for handling ISDN BRI line application interface with NI-1 compatible terminals.

Another enhancement introduced by this feature is that the alphanumeric value for the Service Profile Identification (SPID) has been expanded from 9 to 20 characters (although the NI-1 standard only requires that the SPID range be from 9-20, the system implementation supports 1-20). The SPID is configured in Overlay 27 (by administering the Terminal Service Profiles) for the network side, and programmed on NI-1 compatible ISDN BRI terminals for the user side (both entries for the SPID must be the same).

Operating parameters

There are no operating parameters associated with this feature.

Feature interactions

There are no feature interactions associated with this feature.

Feature packaging

There is no new feature package introduced for this feature. However, the following feature packages are required for ISDN BRI:

- Basic Rate Interface (BRI) package 216
- Basic Rate Interface line application (235)
- Multi-purpose Serial Data Link (MSDL) package 222

Feature implementation

Task summary list

The following is a summary of the tasks to configure NI-1 BRI Compliance Enhancements:

- Table 472: LD 27 - Administer the Terminal Service Profiles (TSPs) on a Digital Subscriber Loop (DSL). on page 1037
- Table 473: LD 27 - Configure the NI-1 protocol for a DSL. Refer to the note above pertaining to selecting the NI-1 protocol on the ISDN BRI terminals. on page 1037

Table 472: LD 27 - Administer the Terminal Service Profiles (TSPs) on a Digital Subscriber Loop (DSL).

Prompt	Response	Description
REQ	NEW CHG	Add a TSP. Change a TSP.
TYPE	TSP	Administer the TSP on a DSL.
DSL	lll s c dsl#	Digital Subscriber Loop.
USID		User service identifier Set USID = 0 to configure a default TSP for non-initializing terminals.
	0-15	Set USID = 1-15 for initializing terminals, for example, the M5317TDX.
SPID	aaa...a <cr> Xaaa...a	Service profile ID aaa...a = any combination of 1-20 alphanumeric characters. <cr> = Stops this prompt from being displayed again. A maximum of 8 valid SPIDs per TSP are allowed. This prompt appears only if USID = 1-15 (if 0 was entered for the USID, a SPID is not required). This SPID must be programmed in the initializing terminal to associate the terminal with a USID.[2] Xaaa...a = removes the specified SPID.
...		

Table 473: LD 27 - Configure the NI-1 protocol for a DSL. Refer to the note above pertaining to selecting the NI-1 protocol on the ISDN BRI terminals.

Prompt	Response	Description
REQ	NEW CHG	Add a TSP. Change a TSP.
TYPE	DSL	Administer the DSL.
DSL	lll s c dsl#	Digital Subscriber Loop.
PRID	6	Defines the protocol to be used on the DSL. Enter 6 for NI-1. The values for this prompt are: 1=ANSI 2=ETSI 3=DMS 4=NET64 5=NUMERIS 6=NI-1

Feature operation

No specific operating procedures are required to use this feature.

[2] The ISDN BRI terminal requires that Layer 2 and Layer 3 parameters be programmed at the terminal. When programming the SPID, the voice SPID and data SPID should each match the one entered in LD 27 when administering the TSP which contains the terminal's DN. Also, if the terminal has an option to select the standard protocol, select the NI-1 standard. Refer to your terminal documentation for complete instructions.

Chapter 80: NI-2/QSIG Compliance Update

Contents

This section contains information on the following topics:

Feature description

The NI-2/QSIG Compliance Update feature makes the NI-2 and QSIG interfaces compliant with the latest standards documents. This feature applies to both Basic Rate Interface and Primary Rate Interface connections.

Note:

ISDN BRI trunking is not available in North America.

Compliancy 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. 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. 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 system handles a recoverable error. Instead of sending incorrect call state information, the system 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 system 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 system 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 system 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

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.

Feature interactions

There are no feature interactions associated with this feature.

Feature packaging

The following packages are required for the ISDN PRI QSIG interface:

- ISDN Signaling (ISDN) package 145
- 2 Mbit 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
- ISDN Signaling Link (ISL) package 147
- 2 Mbit 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 implementation

Task summary list

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

1. Table 474: LD 17 - Assign the configuration record. on page 1042
2. Table 475: LD 16 - Use this overlay to define a route data block. on page 1042

Table 474: LD 17 - Assign the configuration record.

Prompt	Response	Description
REQ	NEW	Add new data.
	CHG	Change existing data.
TYPE	ADAN	Action Device And Number
...		
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.

Table 475: LD 16 - Use this overlay to define a route data block.

Prompt	Response	Description
REQ	NEW	Add new data.
	CHG	Change existing data.
TYPE	RDB	Route data block.
...		
DGTP	PRI PRI2 BRI	Digital route type.
IFC	ISIG ESIG	New DCH interface ID.

Feature operation

No specific operating procedures are required to use this feature.

Chapter 81: NPI and TON in CDR

Contents

This section contains information on the following topics:

Feature description

The Numbering Plan Identification (NPI) and Type Of Number (TON) in Call Detail Recording (CDR) feature allows NPI and TON information to be optionally displayed on the third line of CDRs. 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 LD 17. Figure 117: Third line format for CDR ticket on page 1046 illustrates the format of the third line of a CDR ticket.

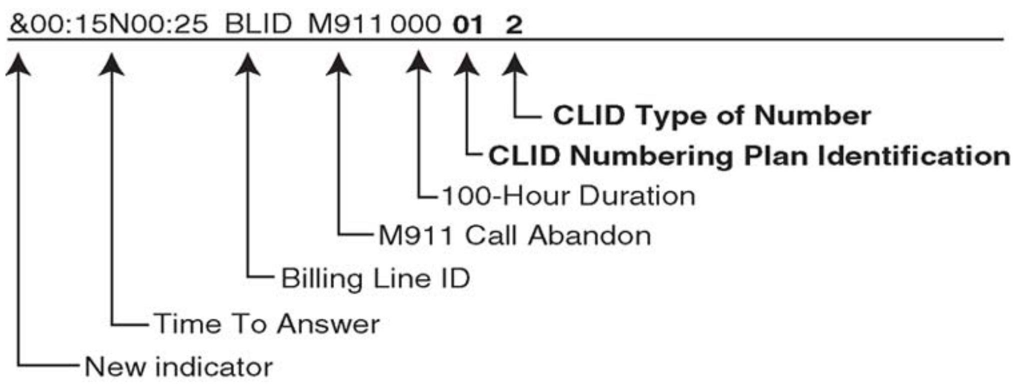


Figure 117: Third line format for CDR ticket

Table 476: Third line contents of CDRs on page 1046 describes the contents of the third line of CDRs.

Table 476: Third line contents of CDRs

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 118: Scenario involving a call over a Meridian Customer Defined Network on page 1047 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.

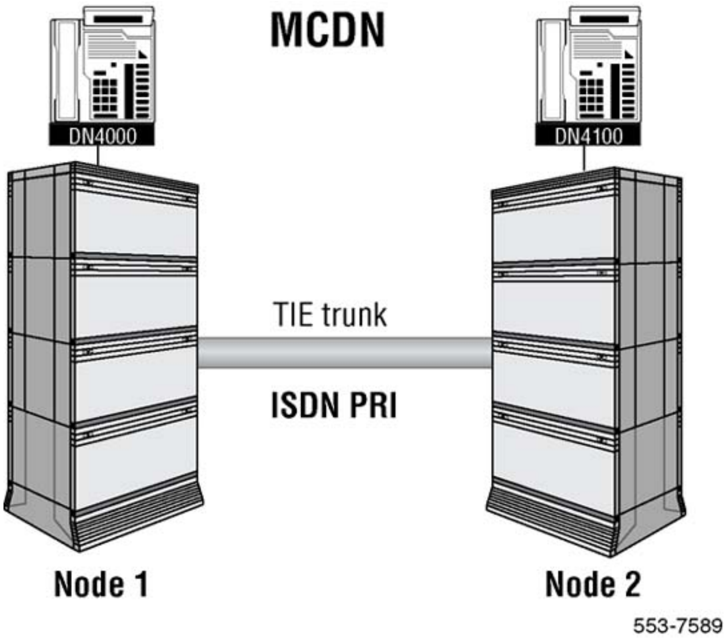


Figure 118: Scenario involving a call over a Meridian Customer Defined Network

The CDR N ticket produced in the described 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 Table 477: NPI information printed in the CDR ticket for an MCDN incoming trunk on page 1048 and Table 478: TON information printed in the CDR ticket for an MCDN incoming trunk on page 1048 for the NPI and TON information for an MCDN incoming trunk.

When an incoming call arrives on the system, NPI and TON are sent in the calling party Information Element (IE) and are mapped into internal values. The correspondence between the system values and the values given in the specifications are described in Table 477: NPI information printed in the CDR ticket for an MCDN incoming trunk on page 1048 to Table 486: TON information printed in the CDR ticket for an NI-2 incoming trunk on page 1052.

Table 477: NPI information printed in the CDR ticket for an MCDN incoming trunk on page 1048 to Table 486: TON information printed in the CDR ticket for an NI-2 incoming trunk on page 1052 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 118: Scenario involving a call over a Meridian Customer Defined Network on page 1047, Table 477: NPI information printed in the CDR ticket for an MCDN incoming trunk on page 1048 and Table 478: TON information printed in the CDR ticket for an MCDN incoming trunk on page 1048 show the NPI and TON information for an MCDN incoming trunk.

Table 477: 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 478: TON information printed in the CDR ticket for an MCDN incoming trunk

	Corresponding value of TON in specification	
TON code in CDR	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

Table 479: NPI information printed in the CDR ticket for a EuroISDN incoming trunk on page 1049 and Table 480: TON information printed in the CDR ticket for a EuroISDN incoming trunk on page 1049 show the NPI and TON information for a EuroISDN incoming trunk.

Table 479: 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 480: TON information printed in the CDR ticket for a EuroISDN incoming trunk

	Corresponding value of TON in specification	
TON code in CDR	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

Table 481: NPI information printed in the CDR ticket for a QSIG incoming trunk on page 1049 and Table 482: TON information printed in the CDR ticket for a QSIG incoming trunk on page 1050 show the NPI and TON information for a QSIG incoming trunk.

Table 481: 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)

NPI code in CDR	Corresponding value of NPI in specification
08	1000 - national standard numbering plan
09	1001 - private numbering plan

Note:

QSIG refers to ISO QSIG and ETSI QSIG.

Table 482: TON information printed in the CDR ticket for a QSIG incoming trunk

	Corresponding value of TON in specification	
TON code in CDR	**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.

Table 483: NPI information printed in the CDR ticket for a non-UIPE and non-MCDN incoming trunk on page 1050 and Table 484: TON information printed in the CDR ticket for a non-UIPE and non-MCDN incoming trunk on page 1051 show the NPI and TON information for a non-UIPE and non-MCDN incoming trunk.

Table 483: 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

NPI code in CDR	Corresponding value of NPI in specification
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 484: TON information printed in the CDR ticket for a non-UIPE and non-MCDN incoming trunk

	Corresponding value of TON in specification	
TON code in CDR	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.		

Table 485: NPI information printed in the CDR ticket for an NI-2 incoming trunk on page 1051 and Table 486: TON information printed in the CDR ticket for an NI-2 incoming trunk on page 1052 show the NPI and TON information for an NI-2 incoming trunk.

Table 485: 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

NPI code in CDR	Corresponding value of NPI in specification
09	1001 - private numbering plan

Table 486: TON information printed in the CDR ticket for an NI-2 incoming trunk

	Corresponding value of TON in specification	
TON code in CDR	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 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 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 feature does not have any specific interactions with other features.

Feature packaging

The NPI and TON in CDR feature requires the following packages:

- Call Detail Recording (CDR) package 4
- Call Detail Recording on Teletype Machine (CTY) package 5
- Calling Line Identification in Call Detail Recording (CCDR) package 118
- New Format Call Detail Recording (FCDR) package 234

Feature implementation

Task summary list

The only task for this section is Table 487: LD 17 - Configure the NPI and TON fields. on page 1053

Table 487: LD 17 - Configure the NPI and TON fields.

Prompt	Response	Description
REQ	CHG	Change existing data.
TYPE	PARM	System parameters.
...		
FCDR	NEW	New Format CDR. OLD = Old CDR format (default).
...		
CLID	YES	TON and NPI fields, in addition to CLID, are included.

Note:

For the NPI and TON in CDR feature, existing CDR implementation procedures must be performed.

Note:

CLID must be configured for the NPI and TON in CDR feature. Refer to the "Calling Line Identification" and "ISDN Calling Line Identification Enhancements" feature modules in this document.

Feature operation

No specific operating procedures are required to use this feature.

Chapter 82: Overlap Signaling on ISDN Networks

Contents

This section contains information on the following topics:

Feature description

On ISDN, dialed digits are sent out and received in the following modes:

- Enbloc
- Overlap

Enbloc Signaling

In Enbloc mode, the switch waits for all the dialed digits from the user and then sends all the digits in the SETUP message.

Overlap Signaling

In Overlap mode, the digits are sent out as they are dialed from the user, instead of waiting for an interdigit timer to timeout. This improves the call setup time. The Overlap Signaling method is useful when a system cannot determine the completion of all the digits unless the originator terminates dialing with an octothorpe "#". Examples of this are when a caller dials:

- international numbers
- private numbers where all the sub-DN digits are not known across the network

Overlap Sending and Overlap Receiving are optional on each D-channel interface for ISDN Trunks.

Overlap Sending

Overlap Sending is applicable to the outgoing leg of the ISDN interface and if enabled on a ISDN interface, it is assumed that the far end supports Overlap Signaling and Overlap Receiving.

Overlap Receiving

Overlap Receiving is applicable to the incoming leg of the ISDN interface. If the ISDN interface does not support Overlap Receiving, then INFO messages received from the originating switch are ignored and the system operates only on the digits received in the SETUP message as if the originating switch used ENBLOC dialing.

Table 488: ISDN Overlap support on page 1056 lists the ISDN interfaces on the system and whether they support or do not support Overlap Sending and Overlap Receiving.

Table 488: ISDN Overlap support

ISDN interface	Overlap Sending	Overlap Receiving	Notes
Australia ETSI	Y	N	
AXE-10 (Sweden and Australia)	Y	N	Overlap Receiving is possible only if the AXE-10 conforms to the guidelines contained in CCITT's preliminary Q.931 section 5.0
Swiss Net (Switzerland)	Y	N	
NEAX-61 (New Zealand) (non-Asia Pacific ISDN Connectivity)	Y	N	
SYS-12 (Norway)	Y	N	
Numeris VN3 (France)	Y	N	
1TR6 (Germany)	Y	N	

ISDN interface	Overlap Sending	Overlap Receiving	Notes
Japan D70 (non-Asia Pacific ISDN Connectivity)	N	N	
Euro ISDN ETS 300-102 basic protocol	Y	Y	
Austria	Y	Y	
Denmark	Y	N	
Finland	Y	Y	
Germany	Y	Y	
Italy	Y	Y	
Norway	Y	N	
Portugal	Y	N	
Sweden	Y	N	
Ireland	Y	Y	
Holland	Y	Y	
Switzerland	Y	Y	
Belgium	Y	Y	
Spain	Y	Y	
United Kingdom	Y	Y	
France	Y	Y	
Commonwealth of Independent States (Russia and the Ukraine).	Y	Y	
Asia Pacific			
Australia	Y	N	
China	Y	Y	
Hong Kong	Y	N	
India	Y	Y	
Indonesia	Y	Y	
Japan	N	N	
Malaysia	Y	Y	
Philippines	N	N	
Singapore	Y	N	
Taiwan	Y	N	

ISDN interface	Overlap Sending	Overlap Receiving	Notes
New Zealand	Y	N	
Thailand	Y	Y	
QSIG	Y	Y	
JTTC (Japan QSIG)	N	N	
BRI	Y	Y	
MCDN (only SL-1)	Y	Y	
National ISDN NI-1, NI-2, NI-3(North America CO interface)	N	N	
ISL	Y	Y	
VNS	N	N	Call establishment over the VNS D-channel does not use Overlap Sending/ Overlap Receiving even if it is configured. Enbloc sending is always used on the VNS D-channel. Overlap sending/receiving is supported on the Bearer trunks, if that capability already exists.

Operating parameters

Overlap Signaling is configured in LD17 at the Overlap Receiving (OVLR), Overlap Timer (OVLT), Direct Inward Dialing Delete (DIDD) and Overlap Sending (OVLS) prompts and in LD86 at the Overlap Length (OVLL) prompt.

The OVLT prompt is provided only when OVLS = YES. The response to OVLT indicates the time the MSL-1 system waits to accumulate digits to send in a INFORMATION message. If response to OVLT is zero, then during Overlap Sending state, each dialed digit generates an INFORMATION message.

The DIDD prompt indicates the number of leading digits to delete when receiving digits from a DID trunk.

OVLL is defined on a Route List Block basis. If this value is less than Flexible Length (FLEN), it is the minimum number of digits the user must dial before a SETUP is sent on an ISDN interface, or before outpulsing begins on a non-ISDN trunk. If the response to OVLL is zero, then FLEN must be used to determine how many digits to dial before sending out any Called

Party information. If FLEN is also zero, no Overlap Sending is attempted; ENBLOC dialing is used instead.

Feature interactions

Call Back Queuing and Off-hook Queuing

This feature does not support Overlap Signaling.

Flexible Hotline

This feature does not support Overlap Signaling.

VNS

Calls established over the VNS D-channel do not use Overlap Sending/Overlap Receiving, even if configured. The VNS D-channel always uses Enbloc Sending. The Bearer trunks support Overlap Sending and Overlap Receiving, if the capability already exists.

Feature packaging

This feature requires the following packages:

- Basic Routing (BRTE) package 14
- Basic Queuing (BQUE) package 28
- Network Class of Service (NCOS) package 32
- 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

Feature implementation

Task summary list

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

1. Table 489: LD 17 - Configure Overlap Receiving, Direct Inward Dialing Delete, Overlay Sending, and Overlap Timer. on page 1061

   Note:

   If the interfacing switch supports Overlap Receiving, set the value for the OVLR prompt as YES.

2. Table 489: LD 17 - Configure Overlap Receiving, Direct Inward Dialing Delete, Overlay Sending, and Overlap Timer. on page 1061

   Note:

   The recommended value for the DIDD prompt is 0. But incase, some digits need to be deleted, then this value has to be agreed upon with the interfacing switch.

3. Table 489: LD 17 - Configure Overlap Receiving, Direct Inward Dialing Delete, Overlay Sending, and Overlap Timer. on page 1061

   Note:

   If the interfacing switch supports Overlap Sending, set the value for the OVLS prompt as YES.

4. Table 489: LD 17 - Configure Overlap Receiving, Direct Inward Dialing Delete, Overlay Sending, and Overlap Timer. on page 1061

   Note:

   A recommended value for OVLT is 1. This means that the digits dialed by the user will be collected for 1 second before being sent in INFORMATION messages. If a user dials at a rate of 1 digit every.5 seconds, there will be two digits for each INFO message. If the messaging traffic on the D-Channel is of major concern, a higher value of OVLT can be used. The greater the value of OVLT, the lower the volume of messages generated by the particular call. OVLT must always be less than the values of T302 and T304 for the given interface.

5. Table 490: LD 86 - Configure the Overlap Digit Length. on page 1061

Note:

OVLL is flexible. The smaller the value, the fewer the number of digits that will be included in the SETUP message. For non-ISDN routes, the smaller the value, the faster a non-ISDN route will begin outpulsing. It is recommended that OVLL be at least the average number of digits in the ESN or CDP steering code that will be using the particular Route List Block. If the average number of digits in an ESN or CDP steering code is three then a recommended value for OVLL would be 5. It is also recommended that OVLL be less than FLEN.

6. Table 491: LD 90 - Configure the Flexible Digits. on page 1061

Note:

LD 90 - prompt FLEN: It is recommended that FLEN be the maximum number of digits that can be used for this ESN or CDP steering code. It is also recommended that FLEN be greater than OVLL.

Table 489: LD 17 - Configure Overlap Receiving, Direct Inward Dialing Delete, Overlay Sending, and Overlap Timer.

Prompt	Response	Description
REQ	NEW CHG	Add new data or change existing data
TYPE	CFN	Configuration Record
- OVLR	YES	Allow Overlap Receiving.
- - DIDD	(0)-15	Number of leading digits to delete from DID trunks.
- 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.

Table 490: LD 86 - Configure the Overlap Digit Length.

Prompt	Response	Description
REQ	NEW CHG	Add new data or change existing data
TYPE	CFN	Configuration Record
- OVLL	(0)-24	Enter the minimum Overlap Length, pertaining to Overlap Sending. If 0 (the default) is entered, then the Flexible Digit Number Length (FLEN) determines whether Overlap Sending takes place.

Table 491: LD 90 - Configure the Flexible Digits.

Prompt	Response	Description
REQ	NEW CHG	Add new data or change existing data

Prompt	Response	Description
TYPE	CFN	Configuration Record
- 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).

Feature operation

No specific operating procedures are required to use this feature.

Chapter 83: QSIG Message Waiting Indication Supplementary Service

Contents

This section contains information on the following topics:

Feature description

The QSIG Message Waiting Indication Supplementary Service (QSIG MWI SS) feature allows the transport of an MWI message between system networks and multi-vendor networks, using the industry standard QSIG protocol.

With this feature, the Served User (the person receiving the message) receives an MWI appropriate for their system when they have at least one new VoiceMail message. For the system user, the red LED is lit, the Message Waiting key lamp flashes or a stutter tone is heard to indicate a new VoiceMail message. When the system user retrieves the message, the MWI is deactivated.

The MWI is transported in the following manner:

1. A Meridian Message Center (for example, Avaya CallPilot) sends an MWI message to the Meridian Message Center Private Integrated Services Network Exchange (PINX) that an unheard VoiceMail message exists for a user. The user is located in a different multi-vendor Private Integrated Services Network (PISN). See Figure

119: MWI message transport between a system network and a multi-vendor network on page 1064.

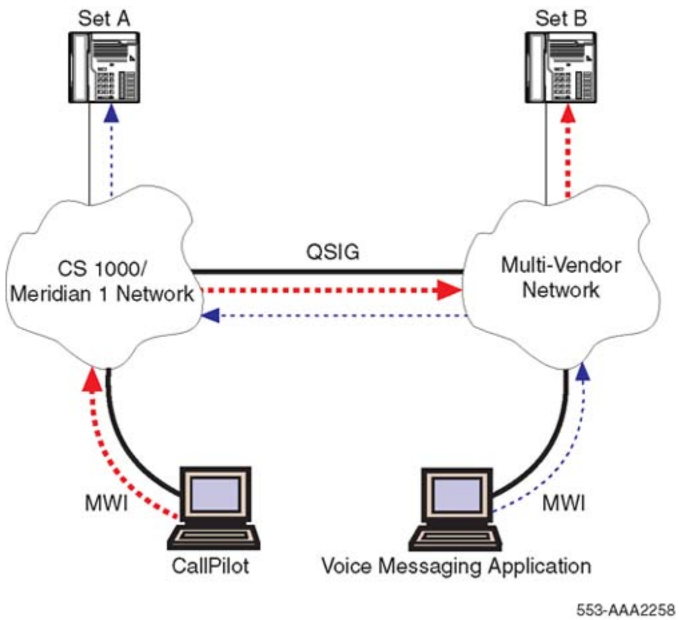


Figure 119: MWI message transport between a system network and a multi-vendor network

2. The Message Center PINX phones up a connection to the multi-vendor Message Center PINX, possibly through one or more transit PINXs.
3. At the MCDN-QSIG gateway of the system network, the MWI message is translated to a QSIG MWI GF Facility message and transported over QSIG to the multi-vendor PINX. See Figure 120: MWI message transport over QSIG in a PISN on page 1065.

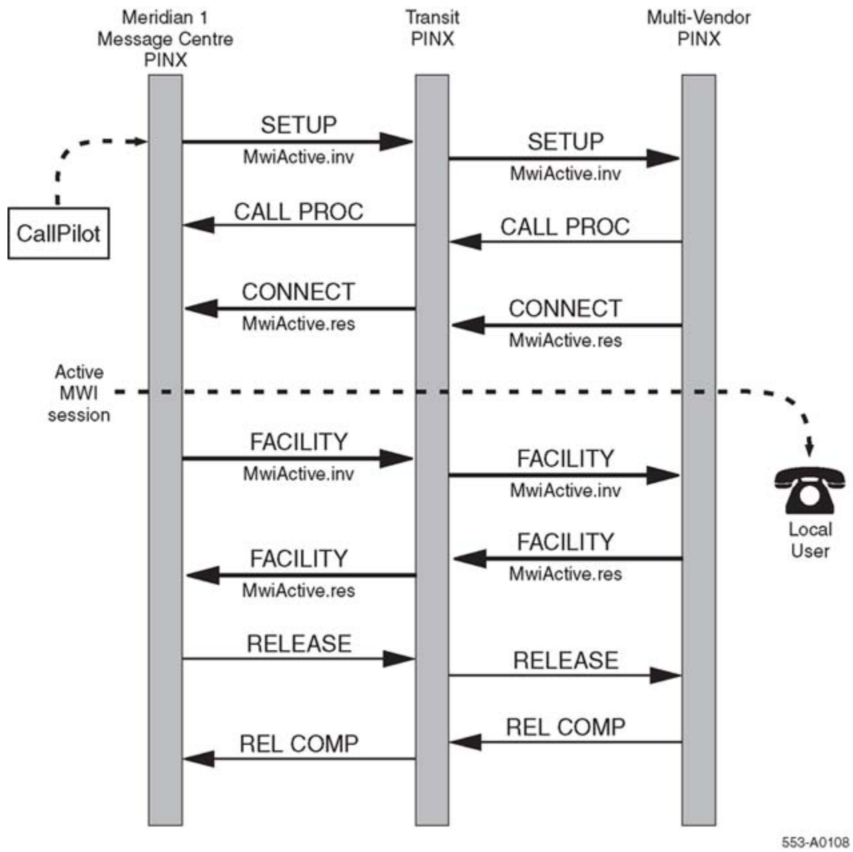


Figure 120: MWI message transport over QSIG in a PISN

4. The multi-vendor's PINX notifies its Message Center that there is an MWI message for a local user.
5. That Message Center activates the local user's VoiceMail message MWI. When the VoiceMail message has been heard, the MWI is de-activated.

For example, if the system network CallPilot receives an MWI message from Phone A for a user in a multi-vendor network (Phone B), the MCDN-QSIG gateway software sends a MWI GF Facility message telling Phone B that there is a new VoiceMail message in their VoiceMail box.

If the foreign network's voice messaging application sends an MWI message from Phone B to Phone A, the MCDN-QSIG gateway software receives the MWI GF Facility message. The system software interprets the message and performs the same actions as if the MWI message had been received from the local CallPilot. The system software activates Phone A's Message Waiting Indication key LED.

When an MWI message is received for a DN assigned to multiple phones, the Message Waiting Indication is activated on each of the phones.When all new messages are heard, the system cancels the MWI on all phones.

QSIG-MCDN Gateways

QSIG MWI Facility messages at QSIG-MCDN Gateway nodes are translated to MCDN MWI Facility messages. MCDN MWI Facility messages at QSIG-MCDN Gateway nodes are translated to QSIG MWI Facility messages.

Operating parameters

The QSIG Message Waiting Indication Supplementary Service feature supports all Meridian phones that support Message Waiting Indication.

QSIG MWI SS requires a Coordinated Dialing Plan (CDP) or a Uniform Dialing Plan (UDP) between the networks using QSIG MWI SS.

ESIG and ISIG networks do not support QSIG MWI SS.

QSIG MWI SS does not support the Remote Call Sender feature.

The QSIG MWI SS feature does not support the QSIG-DPNSS Gateway.

The QSIG-MCDN interface does not support the transport of the MWI Interrogate facility message; however, it tandems MWI Interrogate facility messages to other multi-vendor switches that do support it.

On any system, either the Server User node or the Message Center node must have the QSIG Supplementary Services (QSIG-SS) package 316 equipped in order to use the MWI service.

The QMWI feature is only supported with CallPilot as the local message centers hosted off a system running Succession 3.0 software or later. It is not supported with CallPilot MINI or any other systems (for example, Octel VM) that do not use a link for integration. QMWI cannot be used with phone integration or MIK MCK keys.

Feature interactions

Meridian End-to-End Transparency

The system sends the proprietary QSIG MWI message between system switches over a QSIG interface, using Meridian End-to-End Transparency (MEET). MEET requires a Remote Capability (RCAP) of MCDN QSIG Conversion as a Remote Capability (MQC) on the D-channel.

The QSIG MWI SS sends the QSIG MWI message between system switches and multi-vendor switches, using the industry-standard QSIG protocol. QSIG MWI SS requires a RCAP of QSIG Message Waiting Indication using Integer Value (QMWI) or QSIG Message Waiting Indication using Object Identifier (QMWO).

Remove RCAP MQC, if implemented on a D-channel, before implementing RCAP QMWI/ QMWO.

Feature packaging

QSIG Message Waiting Indication Supplementary Service (QSIG MWI SS) is included in QSIG Supplementary Services (QSIG-SS) package 316.

Feature implementation

Task summary list

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

1. Table 492: LD 15 - Configure a PINX DN for the customer. on page 1067
2. Table 493: LD 16 - Configure QSIG MWI for QSIG BRI trunks. on page 1068
3. Table 494: LD 17 - Configure QSIG MWI for PRI trunks. on page 1068

Table 492: LD 15 - Configure a PINX DN for the customer.

Prompt	Response	Description
REQ:	CHG	Change existing data.
TYPE:	NET	Networking data.
CUST		Customer number
	0-99	Range for Large System and Avaya Communication Server 1000E (Avaya CS 1000E) system.
	0-31	Range for Media Gateway 1000 B (Avaya MG 1000B).
PINX_DN	xx....x	Private Integrated Services Network Exchange DN Node DN (up to seven digits).

Table 493: LD 16 - Configure QSIG MWI for QSIG BRI trunks.

Prompt	Response	Description
REQ	NEW	Add new data.
	CHG	Change existing data.
TYPE	RDB	Route Data Block.
...	...	
DTRK	YES	Digital Trunk Route.
DGTP	BRI	Basic Rate Interface (BRI) Digital Trunk Type.
...	...	
IFC		Interface type for this route.
	ESGF ISGF EGF4	ESIG interface with GF platform. ISIG interface with GF platform. Q Reference Signaling Point interface.
...	...	
RCAP		Remote Capabilities.
	QMWI	Add Message Waiting Indication as a remote capability. The encoding method uses Integer Value.
	QMWO	Add Message Waiting Indication as a remote capability. The encoding method uses Object Identifier. Do not configure QMWI and QMWO on the same link at the same time. XQMW = Remove Message Waiting Indication as a remote capability.

Table 494: LD 17 - Configure QSIG MWI for PRI trunks.

Prompt	Response	Description
REQ	CHG	Change existing data.
TYPE	ADAN	Action Device and Number.
ADAN	NEW DCH xx CHG DCH xx	New D-channel number. Change D-channel number where: xx = 0 - 63.
...	...	
IFC		Interface type for this route.
	ESGF ISGF EGF4	ESIG interface with GF platform. ISIG interface with GF platform. Q Reference Signaling Point interface.
RCAP		Remote Capabilities.

Prompt	Response	Description
	QMWI	Add Message Waiting Indication as a remote capability. The encoding method uses Integer Value.
	QMWO	Add Message Waiting Indication as a remote capability. The encoding method uses Object Identifier. Do not configure QMWI and QMWO on the same link at the same time.

Feature operation

No specific operating procedures are required to use this feature.

Chapter 84: Radio Paging Improvement

Contents

This section contains information on the following topics:

Feature description

A Radio Paging system is a communication tool used to contact mobile parties by means of radio signals. A caller can use their phone to page a mobile party who has a mobile portable receiving device.

Prior to the improvement, the Radio Paging (RPA) feature supported attendant recall in stand-alone 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 can be located anywhere within an ISDN Meridian Customer Defined Network (MCDN) configured with Network Attendant Services (NAS).

The improvement also enables the upgrade to the attendant's display with paged name to display paged name instead of answering name on the paging party when answered. It also allows Network Radio Paging to show the same display information as in the stand-alone operation.

Operating parameters

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 phone on the paging node and is redirected to paging by CFNA, the calling name cannot be retrieved and updated on the answering phone when the paging call is answered. This happens only if the phone on the paging node has CPND defined. If the phone does not have CPND defined, the calling name can be updated on the answering party's phone.

Feature interactions

Call Detail Recording Enhancement

With this 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 (for example, 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 module 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 phones that determine whether or not their DN and CPND information will be displayed on other phones. No CPND or DN information is displayed on phones 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 the existing operation will be followed.

With NAS configured, if an 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

This feature requires the following packages:

- Radio Paging (RPA) package 187
- To gain access to RPA, Flexible Feature Codes (FFC) package 139

- For the Radio Paging Network Recall operation, Network Attendant Service (NAS) package 159
- 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
- Integrated Services Digital Network (ISDN) package 145 and its dependencies for operation in an MCDN ISDN network

Feature implementation

Task summary list

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

1. Table 495: LD 15 - Activate the Recall to Same Attendant (RTSA) prompt on the originating node for this feature to operate network-wide. on page 1074
2. Table 496: LD 87 - Set up remote Radio Paging on originating node. on page 1075

Table 495: LD 15 - Activate the Recall to Same Attendant (RTSA) prompt on the originating node for this feature to operate network-wide.

Prompt	Response	Description
REQ:	NEW CHG	Add new data, or change existing data.
TYPE:	ATT	Attendant Consoles data.
CUST		Customer number
	0-99	Range for Large System and Avaya Communication Server 1000E (Avaya CS 1000E) system.
	0-31	Media Gateway 1000 B (Avaya MG 1000B).
- OPT	aaa	Options.
...		
RTSA	RSAA	Recall to Same Attendant Allowed.

Table 496: LD 87 - Set up remote Radio Paging on originating node.

Prompt	Response	Description
REQ	NEW	Add new data.
	CHG	Change existing data.
TYPE	TSC DSC	Trunk Steering Code Distant Steering Code (enter RPAX FFC defined on paging node).
RRPA	(NO) YES	Remote Radio Paging option.
RLI	xxx	Route List Index of Route List Block used to route to paging node.

Feature operation

No specific operating procedures are required to use this feature.

Chapter 85: Recall with Priority during Night Service Network Wide

Contents

This section contains information on the following topics:

Feature description

This feature adds the Recall with Priority during Night Service (RPNS) functionality to a network-wide application. For this network application, RPNS is activated on a customer basis from an Attendant Console, according to the NITE service specifications defined in the Network Attendant Service feature description.

If NAS is not active, then conventional attendant service and Night Service are in effect.

Service is provided as follows when NAS is activated:

- If Night Service is not active (the NITE key is dark), calls are routed to a Local or Remote attendant.
- If Night Service is active at a local node (the NITE key is lit), the call is directed to a Remote attendant, queued to a route, or presented to a Night Station. If the NAS schedule does not define an alternative attendant location for this period:
    - If the call is local, night service is given or

- If the call is from a Remote node, it is routed to its originating node if Drop Back is allowed. If Drop Back is not allowed, Night Service is given at the node receiving the call.

Operating parameters

This feature applies only to incoming external calls.

Feature interactions

Night Service Improvement or Enhanced Night Service feature

If Recall with Priority during Night Service is equipped along with either the Night Service Improvement or Enhanced Night Service feature, calls are processed according to priority.

Feature packaging

International Supplementary Services (SUPP) package 131 is required for Recall with Priority during Night Service.

Since Network-wide LDN requires Network Attendant Service routing, the following software packages must also be provisioned:

- Basic Routing (BRTE) package 14
- Network Class of Service (NCOS) package 32
- Network Alternate Route Selection (NARS) package 58
- Network Attendant Service (NAS) package 159
- Applicable ISDN options depending upon customer requirements

Feature implementation

Task summary list

The only task in this section is Table 497: LD 15 - Enable the network capability of Recall with Priority during Night Service in the Customer Data Block. on page 1079

Table 497: LD 15 - Enable the network capability of Recall with Priority during Night Service in the Customer Data Block.

Prompt	Response	Description
REQ:	CHG	Change existing data.
TYPE:	NIT	Night Service data
CUST		Customer number
	0-99	Range for Large System and Avaya Communication Server 1000E (Avaya CS 1000E) system.
	0-31	Range for Media Gateway 1000 B (Avaya MG 1000B).
RPNS	(NO) YES	Recall with Priority during Night Service.

Feature operation

No specific operating procedures are required to use this feature.

Chapter 86: Redirecting Name Display Enhancement for QSIG Call Rerouting

Contents

This section contains information on the following topics:

Feature description

Dialed Name Display Allowed and Dialed Name Display Denied (DNDA/DNDD) functionality is now supported for Call Rerouting when both the originating and the diverted-to user are on the same node.

Call Diversion notification provides, for QSIG generic functional protocol (GF) interfaces, the capability of displaying the diverted-to user's calling line identification (CLID) on the calling user's phone when the diverted-to user's phone is rung during QSIG call diversion.

When both the originating and the diverted-to user are on the same system node and call diversion is performed by the Rerouting method, the originating user's subscription options have no impact on the diverted-to user's notification. If the diverted-to user has Class of Service DNDA, then the diverted-to phone displays the redirecting name.

Operating parameters

There are no operating parameters associated with this feature.

Feature interactions

There are no feature interactions associated with this feature.

Feature packaging

The Redirecting Name Display Enhancement for QSIG Call Rerouting feature does not introduce any new packages.

This feature requires the following packages:

- Call Party Name Display (CPND) package 95
- Integrated Services Digital Network (ISDN) package 145
- 2 Mbit/s Primary Rate Interface (PRI2) package 154
- International Primary Rate Access (IPRA) package 202
- Basic Rate Interface (BRI) package 216
- Multi-purpose Serial Data Link (MSDL) package 222
- ISDN BRI Trunk Access (BRIT) package 233
- BRI Line Application (BRIL) package 235
- Q Reference Signalling Point Interface (QSIG) package 263
- QSIG Generic Functional protocol (QsigGF) package 305
- QSIG Supplementary Service (QSIG-SS) package 316

Feature implementation

There are no specific implementation procedures for this feature.

Feature operation

No specific operating procedures are required to use this feature.

Chapter 87: Recorded Announcement for Calls Diverted to External Trunks

Contents

This section contains information on the following topics:

Feature description

Recorded Announcement for Calls Diverted to External Trunks (RANX) provides an optional recorded announcement when the call is being forwarded to the external Public Exchange through DTI, DTI2, PRI2, PRI, analog, and BRI trunks connected to AXE-10 or EuroISDN routes. The announcement notifies the calling party that call forwarding is taking place and the call set-up can require more time than usual. The delay depends upon the signaling that is required to reach the destination party.

Note:

ISDN BRI trunking is not available in North America.

The message is given if the outgoing route is supported by the RANX feature. The calling party receives RAN treatment until either the message is finished or until an answer is received from the external Public Exchange trunk.

This feature operates on a route basis and is controlled by a the RANX prompt in LD 16, for calls diverted to external trunks.

Operating parameters

The RANX feature is supported only for external CO routes. The corresponding RANX prompt in LD 16 appears when the trunk type is:

- A Central Office Trunk (COT) and is not configured as a Radio Paging (RPA) trunk. The trunk has to be configured as an outgoing or outgoing/incoming trunk. The feature is not applicable for trunks configured solely for data traffic.
- A Direct Inward Dialing (DID) trunk configured as an outgoing or outgoing/incoming trunk. The feature is not applicable for trunks configured solely for data traffic.

The RANX feature requires a Recorded Announcement (RAN) machine in the same node as the outgoing route.

If the RAN trunk is configured with supervision, the calling party who is connected to the RAN trunk will be charged from the time the answer signal is sent.

This feature is supported network wide in a Meridian Customer Defined Network (MCDN) environment; no other private network protocols are supported.

Since Digital Private Network Signaling System (DPNSS1) does not support any information concerning redirection, this feature is not supported in a DPNSS1 network.

If a phone is forwarded to a trunk configured with the RANX feature and the party that calls the extension is supposed to dial additional digits, this is not always possible. If the calling party dials digits when being provided with RAN, those digits will be lost since they are not buffered.

If the outgoing trunk, or the phone from which the call originated, times out during the RAN, it is not possible to dial additional digits after the RAN is terminated.

If a DISA call is forwarded to a route with the RANX feature configured, and the RAN message that is provided to the calling party is longer than the duration of the EOD timer, the RAN will be interrupted when the EOD timer expires. When the EOD timer expires, the call is considered as established.

Feature interactions

Call Forward All Calls/Busy/No Answer/Hunt

RANX is activated if the call is forwarded to an outgoing external CO trunk with the RANX feature active.

Network Call Forward

The RANX feature supports call forward to an outgoing external CO route if the route has RANX configured and is located in a node with a RAN trunk. The originating party and the forwarded phone can be in different nodes in the MCDN network.

Internal Call Forward

The RANX feature supports call forward to an outgoing external CO trunk if the route has RANX configured and is located in a node with a RAN trunk.

Phantom TN

If a Phantom TN is forwarded to an external outgoing CO route and the RANX feature is configured for this route, the calling party that is forwarded due to the Phantom TN feature will be provided with a recorded announcement.

Expensive Route Warning Tone

If the calling party is being forwarded to a route with the RANX feature and the Expensive Route Warning Tone feature configured, the Expensive Route Warning Tone will be heard prior to the recorded announcement.

Feature packaging

This feature requires the following packages:

- Recorded Announcement (RAN) package 7
- Intercept Treatment (INTR) package 11

Note:

The use of this feature is not recommended if the outgoing external Central Office (CO) route is not configured with answer supervision.

Feature implementation

Task summary list

The only task in this section is Table 498: LD 16 - Configure Recorded Announcement for Calls Diverted to External Trunks in the Route Data Block. on page 1088

Table 498: LD 16 - Configure Recorded Announcement for Calls Diverted to External Trunks in the Route Data Block.

Prompt	Response	Description
REQ	NEW	Add new data.
	CHG	Change existing data.
TYPE	RDB	Route Data Block.
DLDN	YES	YES if no CPG configured.
...		
FORM	aaa	Signaling format.
ICOG	IAO	Incoming and/or outgoing trunk.
RANX	(NO) YES	(RAN not requested when a call is forwarded to this route), RAN is requested when a call is forwarded to this route.
RANR		RAN route number for "Authcode Last" prompt (NAUT)

Prompt	Response	Description
	0-511	Range for Large System and Avaya Communication Server 1000E (Avaya CS 1000E) system.
	0-127	Range for Media Gateway 1000 B (Avaya MG 1000B).
	x	Response for Avaya Communication Server 1000M (Avaya CS 1000M).

Feature operation

No specific operating procedures are required to use this feature.

Chapter 88: Ring Again on No Answer

Contents

This section contains information on the following topics:

Feature description

This feature extends the capabilities of Network Ring Again for ISDN applications.

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. Ring Again on Busy gives a caller the opportunity, after encountering a busy Directory Number (DN), to ring the DN again when it becomes free. If a dialed DN is busy, or if all the trunks are busy, pressing the Ring Again key asks the system to monitor the dialed DN or trunk. When it becomes available, the system notifies the caller. The call is automatically dialed again when the caller presses the Ring Again key a second time.

Ring Again on No Answer is applied to the originally dialed DN only.

Operating parameters

Ring Again on No Answer cannot be applied:

- if the dialed DN is a Pilot DN
- to Attendant Consoles
- to a station which has been intercepted to the attendant
- to a station which is queued for an attendant
- to a station which has been recalled to an attendant due to misoperation
- to Automatic Call Distribution (ACD) stations
- to a station with Radio Paging active
- to trunks

Meridian digital telephones must be equipped with a Ring Again (RGA) key/lamp combination.

Ring Again on No Answer is applied to the originally dialed DN only.

Feature interactions

Call Forward (All Calls)

Call Forward No Answer

Automatic Call Forward

If an unanswered call is forwarded to another station by any of these features, RANA is applied to the originally dialed station.

Hunt

If RANA has been applied to a station going through a Hunt sequence, Ring Again is applied to that station and not the ringing station.

ISDN QSIG/EuroISDN Call Completion

Analog (500/2500 type) phones can have only one Call Completion to Busy Subscriber request at a given time. Meridian 1 Proprietary Phones can make Ring Again requests based on the number of Ring Again keys programmed on a phone.

Feature packaging

This feature requires the following package:

- Advanced ISDN Network Services (NTWK) package 148.

Feature implementation

Task summary list

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

Table 499: LD 15 - Allow or deny Ring Again on No Answer operation.

Prompt	Response	Description
REQ:	CHG	Change
TYPE:	FTR	Features and options data.
CUST		Customer number

Prompt	Response	Description
	0-99	Range for Large System and Avaya Communication Server 1000E (Avaya CS 1000E) system.
	0-31	Range for Media Gateway 1000 B (Avaya MG 1000B).
OPT	(RND) RNA	Ring Again No Answer (Denied) Allowed.

Table 500: LD 11 - Define RGA keys against M2317 phones.

Prompt	Response	Description
REQ:	NEW	Add new data.
	CHG	Change existing data.
TYPE:	aaaa	Type of phone.
TN		Terminal number
	l s c u	Format for Large System and CS 1000E system, where l = loop, s = shelf, c = card, u = unit.
KEY	xx RGA	Key number, Ring Again Must be key 10 on LOGIVOX phones. RANA can be activated if OPT = RNA in LD 15. When OPT = RND in LD 15, all phones with the RGA key will only be able to activate Ring Again Busy.

Feature operation

No specific operating procedures are required to use this feature.

Chapter 89: SDID Number as CLID for EuroISDN Trunks

Contents

This section contains information on the following topics:

Applicable regions

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

Feature description

The following is a description of the SDID number as CLID for EuroISDN Trunks feature. SDID capability is available for EuroISDN calls that have CLID OPT1 enabled. For BRIE interfaces, the OPT1 option is configured at the CLID prompt in LD 16. For PRI2 interfaces, the OPT1 option is configured at the CLID prompt in LD 17.

The Send DID Number (SDID) feature sends the Direct Inward Dial (DID) number of a specific DN as Calling Line Identification (CLID) on an outgoing trunk call. SDID replaces the internal

DN of the phone with the DID (external DN) of the phone. The DID number is obtained from the Incoming Digit Conversion (IDC) table.

The IDC table converts the following:

- the internal DN of the phone to the external DN of the phone
- the external DN of the phone to the internal DN of the phone

Table 501: Example of an IDC table

Incoming Digits (IDGT)	Converted Digits (CDGT)
4322 (external DN)	726 (internal DN)
8741 (external DN)	12 (internal DN)

Phone A with an internal DN of 726 calls Phone B. The IDC table converts Phone A's internal DN 726 to its external DN 4322. The CLID of Phone A is sent to Phone B as 4322. If the internal DN is not entered in the IDC table, the internal DN is sent as the CLID.

Phone C calls Phone D by dialing Phone D's external DN 8741. The IDC table converts Phone D's external DN 8741 (external DN) to its internal DN 12. Phone D's external DN is sent to Phone C as the Connected Number (CONN).

Type Of Number of Calling Party Number

The SDID feature allows the Type of Number (TON) of the calling party number to be changed in the Route Data Block (RDB). The TON is changed when the calling party number has an ISDN numbering plan.

Connected Number Identification

The SDID DN (the external DN) is sent to the CO as the Connected Number for an incoming call.

Operating parameters

This feature is only available on EuroISDN routes.

The EuroISDN route must have an IDC table associated with it when SDID is enabled.

This feature replaces the internal DN with the DID DN for the following:

- analog (500/250- type) phones
- Meridian Digital phones

- Basic Rate Interface (BRI) Line phones
- Attendant Consoles

This feature does not apply to trunks as the originator.

Feature interactions

Automatic Call Distribution

The SDID number as CLID for EuroISDN trunks feature is not applicable to Automatic Call Distribution (ACD), as calls cannot originate from an ACD key.

If the ACD phone is equipped with an active Single Call Ringing (SCR) key, the DN is obtained from the active key. If the DN has been entered in the IDC table, the external DID number is used. See "ISDN Calling Line Identification Enhancement".

Business Network Express

Even though the CLID is changed to the SDID DN, the private CLID or name is not changed.

Call Detail Recording

Call Detail Recording (CDR) is not affected by the SDID feature. The record's Originating ID (ORIGID) and Terminating ID (TERID) remain as the internal DN.

Call Forward

If a forwarding DN on a EuroISDN trunk is used as CLID and is found in the IDC table, the SDID DN is sent as the CLID.

Calling Party Privacy

The Calling Party Privacy (CPP) feature is not affected by SDID.

Direct Inward System Access

Direct Inward System Access (DISA) numbers are not affected by the SDID feature.

EuroISDN Trunk - Network Side

The SDID number as CLID feature is supported on the network side of the EuroISDN trunk.

Feature packaging

This feature requires the following packages:

- Incoming Digit Conversion (IDC) package 113
- International Supplementary Features (SUPP) package 131
- Integrated Services Digital Network (ISDN) package 145

Feature implementation

Task summary list

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

1. LD 49 - Define the IDC table.
2. LD 16 - Define the Route Data Block (RDB) for ISDN trunks.

Prompt	Response	Description
REQ	NEW	Add new data.
	CHG	Change existing data.
TYPE	IDC	Incoming Digit Conversion.
CUST	xx	Customer number, as defined in LD 15.
DCNO	0 - 254	Day IDC tree number.
FDID	(NO) YES	Flexible DID.

Prompt	Response	Description
SDID	(NO) YES	Send DID number instead of internal DN.
IDGT	xxxx xxxx	Incoming Digit or range of digits where: xxxx = 0 - 9999.
xxxx	yyyy	Converted Digits (CDGT).
IDGT	xxxx xxxx	Incoming Digit or range of Digits where: xxxx = 0 - 9999.
xxxx	yyyy	Converted digits (CDGT).

Table 502: LD 16 - Define the Route Data Block (RDB) for ISDN trunks.

Prompt	Response	Description
REQ	NEW	Add new data.
	CHG	Change existing data.
TYPE	RDB	Type of data block = RDB (route data block).
ROUT		Route number
	0-511	Range for Large System and Avaya Communication Server 1000E (Avaya CS 1000E) system.
	0-127	Range for Media Gateway 1000 B (Avaya MG 1000B).
ISDN	(NO) YES	Integrated Services Digital Network option.
- SDID	YES	Send DID number instead of internal DN. (NO) = default.
-- CTON	(NCHG) UKWN INTL NATL LOCL	Call Type of Number. Call Type is not changed. Unknown Call Type. International Call Type. National Call Type. Subscriber Call Type.
....	....	
IDC	YES	Incoming DID Digit Conversion on this route.
- DCNO	xx	Day IDC tree number where: xx = (0) - 254.
- NDNO	xx	Night IDC tree number where: xx = 0 - 254.

Feature operation

No specific operating procedures are required to use this procedure.

Chapter 90: Singapore ISDN Restart Message Enhancement

Contents

This section contains information on the following topics:

Feature description

The Singapore ISDN Restart Message Enhancement allows systems with Asia Pacific-Singapore ISDN connectivity to recognize and process Restart Acknowledge messages sent from an Alternate Carrier's Nokia Central Office (CO) switch.

Note:

The Alternate Carrier's Nokia CO must be located in Singapore.

With the Singapore ISDN Restart Message Enhancement, the system accepts an Indicated Channels Restart Acknowledge message from the CO. The acknowledge message from the CO is in response to the Single Interface Restart Message.

Figure 121: Restart messaging sequence for the Singapore ISDN Restart Message Enhancement on page 1102 shows the Restart messaging sequence between a system and a Nokia CO switch over the Asia-Pacific Singapore ISDN interface.

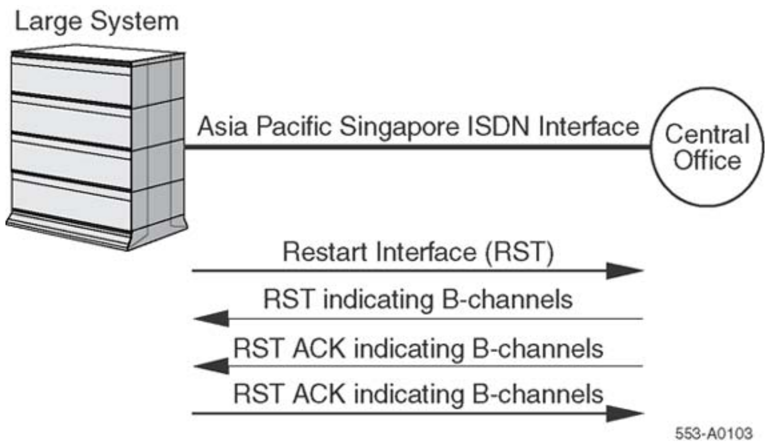


Figure 121: Restart messaging sequence for the Singapore ISDN Restart Message Enhancement

Operating parameters

There are no operating parameters associated with this feature.

Feature interactions

There are no feature interactions associated with this feature.

Feature packaging

There are no new packages associated with this feature.

Feature implementation

There are no specific implementation procedures associated with this feature.

Feature operation

No specific operating procedures are required to use this feature.

Chapter 91: Software Release ID

Contents

This section contains information on the following topics:

Feature description

Software Release ID uses the D-channel connection of your switch to identify the software release of an adjoining switch. This feature identifies the software release of the NRAG, NACD, NMS, and NCRD features. The software release ID can be requested for all direct connections to the system. However, the Software Release ID cannot be obtained for switches in a tandem configuration. The information provided by the Software Release ID depends on the interfacing switches. That is, a system switch provides a release number and a DMS-100 switch provides a BCS number.

Note:

If the interface is changed, the release ID is also changed. The release ID must then be reconfigured.

This feature prevents software incompatibility between two switches. Different applications are supported by different releases, and for most of the ISDN applications, operations are invoked by sending messages back and forth. To prevent software incompatibility, the following occurs. The release ID of the connecting D-channel is checked before data is sent through the ISDN interface. If the connecting switch does not have the software to handle the feature requested, an application message is not sent. Instead, an error message is printed.

Note:

The release ID information is required and supported for connection to Avaya equipment only.

Operating parameters

There are no operating parameters associated with this feature.

Feature interactions

There are no feature interactions associated with this feature.

Feature packaging

This feature is included in base System Software.

Feature implementation

Table 503: LD 17 - Configure the Software Release ID feature.

Prompt	Response	Description
REQ	CHG	Change
TYPE	CFN	Configuration Record
ADAN	NEW DCH xx	Add a primary D-channel
CTYP	DCHI MSDL	Card type
DNUM	xx	Device number: physical port (odd) for D-channel on DCH, physical card address for MSDL or DDCH.
- PORT	xx	Port number on MSDL or card
RLS	xx	Release ID of the switch at the far end of the D-channel.

Feature operation

No specific operating procedures are required to use this feature.