ISDN Primary Rate Interface Features Fundamentals — Book 1 of 3
Avaya Communication Server 1000

7.5
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Contents

Chapter 1: New in this release

The following sections detail what's new in this document for Avaya Communication Server 1000 Release 7.5.

Features

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

Other

There are no other changes.

Revision History

January 2012	Standard 05.05. This document is up-issued to support the removal of End of Life (EoL) and Manufacture Discontinued (MD) hardware content and associated diagrams.
September 2011	Standard 05.04. This document is up-issued to support the removal of content for outdated features, hardware, and system types.
May 2011	Standard 05.03. This document is up-issued to support Avaya Communication Server 1000 Release 7.5.
November 2010	Standard 05.02. This document is up-issued to support Avaya Communication Server 1000 Release 7.5. The information on the patch MPLR24744 is added.
June 2010	Standard 04.01. This document is up-issued to support Avaya Communication Server 1000 Release 7.0.
March 2010	Standard 03.05. This document is up-issued to support Communication Server 1000 Release 6.0.
February 2010	Standard 03.04. This document is up-issued to support Communication Server 1000 Release 6.0.

January 2010	Standard 03.03. This document is up-issued to support Communication Server 1000 Release 6.0.
May 2009	Standard 03.02. This document is up-issued to support Communication Server 1000 Release 6.0.
May 2009	Standard 03.01. This document is up-issued to support Communication Server 1000 Release 6.0.
December 2007	Standard 02.04. This document is up-issued to support Communication Server Release 5.5.
July 2007	Standard 01.02. This document is up-issued to reflect the addition of appendix, Call scenarios for name display.
May 2007	Standard 01.01. This document is issued to support Communication Server 1000 Release 5.0.
July 2006	Standard 5.00. This document is up-issued to reflect changes in content: • addition of Feature Packaging in the Network and Distinctive Ringing chapter • addition to Trunk Route Optimization chapter • addition of Table 137 to Engineering and Configuration Guidelines chapter
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 multiple legacy technical document. This new document consolidates information previously contained in the legacy document, now retired, *International ISDN Primary Rate Interface: Feature description and administration, 553-2901-301*.

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.

Legacy products and releases

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

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Applicable systems

This document applies to the following systems:

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

System migration

When you upgrade a Meridian 1 system to run Avaya Communication Server 1000 software, and configure the system to include a Signaling Server, it becomes an Avaya CS 1000 system. Table 1: Meridian 1 systems to CS 1000 systems on page 19 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 11C Chassis	CS 1000E
Meridian 1 PBX 11C Cabinet	CS 1000E
Meridian 1 PBX 61C	CS 1000M Single Group

This Meridian 1 system	Maps to this CS 1000 system
Meridian 1 PBX 81C	CS 1000M Multi Group

For more information, see one or more of the following technical publications:

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

Intended audience

This document is intended for individuals responsible for administering Avaya Communication Server 1000 (Avaya CS 1000) and Meridian 1 systems.

Conventions

Terminology

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

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

Related information

This section lists information sources that relate to this document.

Technical Documentation

The following technical publications are referenced in this document:

- *Avaya Network Routing Service Fundamentals, NN43001-130*
- *Avaya SIP Line Fundamentals, NN43001-508*
- *Avaya Co-resident Call Server and Signaling Server Fundamentals, NN43001-509*

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Chapter 4: ISDN product overview

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.

Integrated Services Digital Network

Integrated Services Digital Network (ISDN) is a recommended standard for digital communication. ISDN provides standard digital interfaces between phones, terminals, and telecommunication networks.

ISDN uses a common signaling protocol transmitted over a dedicated data channel called the D-channel. The D-channel carries call set-up and feature activation information to the call destination. This allows users to have access to network-wide features.

ISDN services are categorized into two types of interfaces: Primary Rate and Basic Rate.

ISDN Primary Rate Interface (PRI), provides 23B+D (T-1 carriers) or 30B+D (E-1 carriers) digital connectivity between systems and the following interfaces:

- host computers
- SL-100
- DMS-100, DMS-250
- QSIG

ISDN PRI 30B+D access is provided to the following Central Office connectivities:

- Australia ETSI
- AXE-10 (Sweden and Australia)
- SwissNet
- NEAX-61
- SYS-12
- Numeris VN3
- 1TR6
- NET-3 and ETS 300 403 (EuroISDN)
- Asia Pacific

ISDN protocol overview

ISDN protocols govern the format, timing and sequencing used to exchange data and control information between two terminal stations connected through an ISDN network.

These protocols are based on a model containing seven layers of protocol developed by the International Standards Organization (ISO). This seven-layer model, called the Open Systems Interconnection (OSI) model, has been adopted by the International Telegraph and Telephony Consultative Committee (CCITT). It is the basis for building protocol structures for ISDN service. ISDN currently uses four of the seven layers:

- Layer 1—Physical Layer
- Layer 2—Link Layer
- Layer 3—Network Layer
- Layer 7—Application Layer

Each layer uses the series of services provided by the layer beneath it, and builds on these services to perform communication functions for the layer above. For example, layer two builds on the services from layer one and provides the combined services to layer three. This layered approach splits the complex protocols into a series of easily managed blocks, each of which can be modified without affecting protocols in another layer.

Layers one through three control the set-up of connections. These layers also supervise the transmission of information between terminals and the packet-switched and circuit-switched networks.

The PRA protocol layers are implemented as follows:

- Layer 1 (the Physical Layer) is handled by the Primary Rate Interface (PRI) card.
- Layer 2 (the Link Layer) is handled by the D-channel Interface.
- Layer 3 (the Network Layer) is handled by the system software.
- Layer 7 (the Application Layer) is also handled by system software.

The Application Layer uses the Transaction Capability Application Part (TCAP) and the Remote Operation Service Element (ROSE) to process applications. The transport of applications has three main parts:

- a simple interface allowing applications to send and receive data
- a non-call-associated supplementary service to handle TCAP remote operation
- a call-associated supplementary service to handle ROSE remote operation

Note:

ROSE messaging is supported for call-associated messages; TCAP messaging is supported for non-call-associated messages.

ISDN Primary Rate Interface

ISDN PRI provides the interface between a customer's equipment and the public and private network, and allows basic call services and network business services capabilities across the public and private networks.

The characteristics of PRI provide a standard digital interface that supports the Q.931 protocol, as recommended by the International Telegraph and Telephony Consultative Committee (CCITT). This protocol is a message-oriented out-of-band signaling protocol that provides telephony, data and supplementary services. The PRI architecture allows continued growth in operations, maintenance and network business services.

Primary Rate Interface structure

The Primary Rate Interface is structured as a collection of digital, 64 Kbit/s channels. One channel is required for D-channel signaling information. The other channels are for user voice and data transport and are referred to as Bearer Channels or B-channels.

The physical (layer 1) specification for the Primary Rate Interface supports the standard electrical characteristics and frame structures of the 1.5 Mbit T-1 or 2.048 Mbit E-1 digital carrier. Therefore, the PRI can have up to 23 or 30 B-channels and one D-channel.

nB+D Primary Rate Interface

Although the PRI layer 1 specifies the protocol for 23B+D or 30B+D interface structures, layer 3 supports signaling for a "larger" PRI, in the form of the nB+D Primary Rate Interface.

In this configuration, one active D-channel can provide signaling support for all the B-channels contained on a maximum of 16 digital carriers (384 B-Channels for T-1 carriers or 480 B-channels with E-1 carriers). The following notes pertain to the nB+D configuration.

Note:

nB+D PRI is only provided on the interfaces that support it, as described in the feature modules throughout this document. Also, the maximum number of digital carriers can be constrained by the switching device on the far end of the link. To determine the constraints in your market, consult your local Avaya representative. Also note that the maximum number "n" in nB+D configurations is subject to trunk route limitations.

Note:

For nB+D PRI, the actual D-channel configuration can depend on market-specific requirements. Also, the backup D-channel can reside on the first or second carrier in some markets.

Note:

If the maximum number of carriers is configured for nB+D (that is, 16), then it is assumed that the configuration includes a backup D-channel. In configurations with less than 16 carriers, a backup D-channel is not required, but is recommended.

ISDN Signaling Link Interface

The CCITT currently limits the Layer 1 ISDN protocols to digital facilities only. Some customer applications can be met more effectively with analog facilities. Systems Networking offers customers this flexibility in the form of the ISDN Signaling Link (ISL) interface.

The ISL interface is a configuration unique to Systems Networking for system to system.

Note:

The ISL interface is not supported on any other private exchange, and no public exchange at all, due to the lack of standardization for ISL.

It extends the advantages of ISDN signaling to locations served by analog or digital facilities.

The ISL interface is structured as a collection of analog and/or digital trunks, and can operate under two basic modes of operation: shared and dedicated. Shared mode requires a PRI (either 30B+D, or nB+D) between originating and terminating switches. The ISDN D-channel is used to provide out-of-band signaling for both the ISDN and non-ISDN trunks. Dedicated mode is appropriate when no PRI exists between originating and terminating switches or when it is not desirable to share an ISDN D-channel as described above. In this mode of operation, a dedicated D-channel is established between originating and terminating switches. The signaling information for the selected, non-ISDN trunks is transported through this link.

The signaling connection is a data circuit which can be established over a leased line, multiplexed facilities, or an existing trunk circuit using standard data communication equipment such as modems, multiplexers, or system data adapters.

In the case of a failure on this link, signaling operation reverts to conventional inband signaling. This is a major advantage of the ISL interface.

Reverting to conventional trunk signaling

This feature handles ISL trunk calls by reverting to conventional trunk signaling when the primary and backup D-channels become inactive.

When a primary and backup D-channel go down:

- Established ISL calls remain established, regardless of the signaling method used.
- Transient ISL calls that are set up using conventional trunk signaling are not disturbed.
- Transient ISL calls that are set up by D-channel signalling are dropped. The user must re-initiate the call. Then, conventional trunk signaling is used if the D-channel remains inactive.
- ISL channels are not marked "maintenance busy".

There are two scenarios that can occur when a D-channel is re-established, one with backup D-channel and one without.

When a D-channel with backup is re-established:

There is no impact. The primary D-channel simply recovers. ISL calls, still using D-channel signaling (in existing software), can bypass the restart procedure.

When a D-channel without backup is re-established:

- Transient and established ISL calls that are set up using conventional trunk signaling are not disturbed.
- Established ISL calls that are set up using D-channels are disconnected.
- To disconnect an established ISL call, the system uses the same signaling method with which the call was setup.

Backup D-channel

In situations where the reliability of the D-channel signaling is critical, each of the PRI, nB+D PRI, and ISL interfaces can be configured so that there is one active and one backup D-channel. If the active D-channel fails, then D-channel processing switches over to the backup D-channel.

When dealing with standard 30B+D PRI structures, it is necessary to have at least two PRIs in order to provide a backup D-channel. The backup D-channel is installed and configured the same way as the primary D-channel. Note, however, that when configuring primary and backup D-channels on a system, the backup D-channels must be programmed the same as the primary D-channels and must be connected to the same card type.

Chapter 5: Connection parameters

Contents

This section contains information on the following topics:

Connection parameters for 2.0 Mbit PRI

This section describes the major parameters that must be coordinated between the system and the far-end facility, over a 2.0 Mbit ISDN PRI connectivity. These parameters are as follows:

- Frame formats for the 2.0 Mbit data stream are:
  - Alternate frame format
  - CRC 4 multiframe format
- Line encoding for the 2.0 Mbit data stream is:

  HDB3 coding, a modified form of Alternate Mark Inversion (AMI), for zero code suppression

Note:

(Line coding is bipolar for 2Mbit/s line transmission.)

- Error detection on the 2.0 Mbit data stream consists of:
  - Remote alarm detection
  - Bit error rate monitoring
  - Frame alignment monitoring
  - Frame slip detection
- Data rate parameters:

  64 Kbit/s clear
- Channel parameters:
  - B-channels
  - D-channels
- Interface protocols for system to system connections:

  slave to master

Frame formats

The CEPT basic format consists of 32 8-bit bytes with one byte for each channel. This makes a total of 256 bits for each frame. The nominal bit rate of the signal is 2.0 Mbit and the sampling rate for each channel is 8000 Hz.

Alternate frame format

The alternate frame format includes the following framing/control bit patterns in timeslot 0:

- FAS: Bits at position 2 to 8 in even frames constitute the FAS. This is in the form 0011011.
- Si bit: The Si bit is always 1 in transmission, and is ignored in reception.
- A bit: This bit is used for Remote Alarm Indication (RAI).

Table 2: Alternate frame formats

Bit number								
Alternate frame	1	2	3	4	5	6	7	8
Even frame	Si	0	0	1	1	0	1	1
Odd frame	Si	1	A	Sa_4	Sa_5	Sa_6	Sa_7	Sa_8

CRC 4 Multiframe Format

The CRC4 multiframe format includes the following framing/control bit patterns in timeslot 0:

- Multiframe Alignment Signal (MAS): Bits 1 in frames 1, 3, 5, 7, 9, and 11 constitute MAS. This signal is in the form of 001011.
- CRC bits: C1, C2, C3, and C4 constitute the CRC bits.
- Si bits: The NT8D72AA card can be configured in software to use the Si bit in one of two ways. The Si bit can be left as 1 during all transmission, or it can be set to 0 when a CRC-4 error arises during transmission.
- Sa bits, A bit, and FAS: The use of these bits is the same as in the Alternate Frame.

Table 3: CRC 4 Multiframe Format table

Sub-multiframe	Frame number	Bit number							
		1	2	3	4	5	6	7	8
I	0	C1	0	0	1	1	0	1	1
	1	0	1	A	Sa_4	Sa_5	Sa_6	Sa_7	Sa_8
	2	C2	0	0	1	1	0	1	1
	3	0	1	A	Sa_4	Sa_5	Sa_6	Sa_7	Sa_8
	4	C3	0	0	1	1	0	1	1
	5	1	1	A	Sa_4	Sa_5	Sa_6	Sa_7	Sa_8
	6	C4	0	0	1	1	0	1	1
	7	0	1	A	Sa_4	Sa_5	Sa_6	Sa_7	Sa_8
	8	C1	0	0	1	1	0	1	1
	9	1	1	A	Sa_4	Sa_5	Sa_6	Sa_7	Sa_8
	10	C2	0	0	1	1	0	1	1
	11	1	1	A	Sa_4	Sa_5	Sa_6	Sa_7	Sa_8
II	12	C3	0	0	1	1	0	1	1
	13	Si1	1	A	Sa_4	Sa_5	Sa_6	Sa_7	Sa_8
	14	C4	0	0	1	1	0	1	1
	15	Si_2	1	A	Sa_4	Sa_5	Sa_6	Sa_7	Sa_8

Line encoding

Line coding for CEPT is bipolar. A modified form of Alternate Mark Inversion (AMI) is used, and is called High Density Bipolar 3 (HDB3) coding (see below). The general requirements for CEPT code suppression is a maximum of 3 consecutive zero binary bits.

To meet the maximum and minimum requirements, PRI provides HDB3 zero-code substitution techniques as options.

HDB3 coding

This is a coding scheme used for bipolar alternate-mark-inversion digital transmission which replaces any sequence of four consecutive zeros with a unique code containing a bipolar violation (BPV). The unique sequence is (X,0,0,BPV) where X is set to either a one or a zero to ensure that the bipolar violation is of opposite polarity to the previous BPV. This maintains the DC balance of the line.

Error detection

Remote alarm

Remote alarm transmission

A remote alarm indicates that the far end (the remote switch) is not ready. If the PRI is receiving the remote-alarm pattern, it indicates that there is in fact a 2.0 Mbit digital line connection. (That is, transmission integrity is good, and the problem exists at the far end.)

When the PRI receives a remote-alarm signal from the far end, all 30 B-channels are disabled.

Remote alarm method

The A bit is used for Remote Alarm Indication (RAI). When the system software informs the card about an alarm indication, the card sets the A bit to one during transmission. This is a signal to the remote end of an alarm condition. If there is no alarm, the 2.0 Mbit PRI sets the A bit to zero during transmission.

Upon reception of an A bit equal to one, the PRI informs the system that an RAI has been received.

Channel restoration

When the PRI stops receiving the remote alarm, the 30 B-channels are placed into the idle state and made available for calls.

Each time a remote alarm is generated, a counter is incremented. The remote alarm is cleared immediately upon the reception of a cleared message from the card.

Bit error rate

Bit error-rate monitoring detects errors in transmission. There are two methods of bit error-rate monitoring, bipolar violation tracking (BPV) and cyclic redundancy check (CRC).

Bipolar Violations (BPV)

In a bipolar pulse stream, transmitted pulses alternate in polarity. A bipolar violation has occurred if two pulses of the same polarity are received in succession (this could be caused by an electrical disturbance, such as noise). See Figure 1: Bipolar violations on page 33.

Note:

Some bipolar violations are normal when using the HDB3 coding.

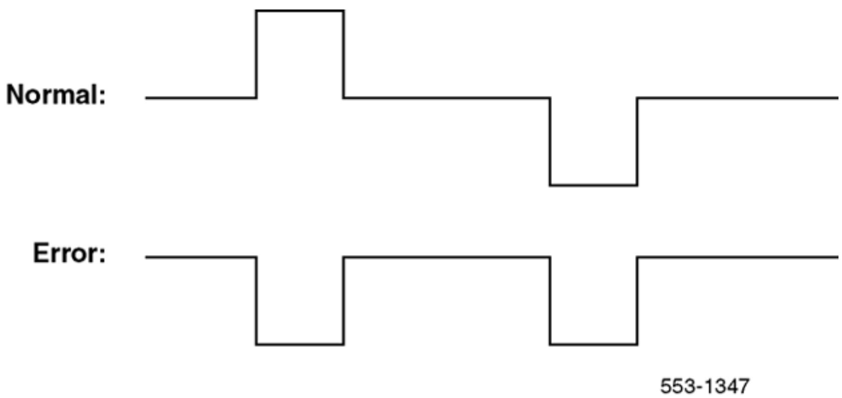

Figure 1: Bipolar violations

Cyclic redundancy check (CRC)

The CRC 4 Multiframe Frame format contains a checksum of all the data in the frame. The receiving side uses the checksum to verify that the data is error free.

The primary functional difference between BPV and CRC is that BPV indicates physical errors limited to the local span, while CRC indicates errors on an end-to-end span. For example, a satellite link BPV only detects errors in the span between the system and the satellite connection. Since CRC traverses the entire span, it detects errors from the system through to the far end switch, indicating an end-to-end bit error rate.

Bit error rate thresholds

PRI hardware detects BPV and CRC errors. Running the midnight routines, prints the number of overflows and clears the counter. The printout shows a total error count for each of the error types.

These are the bit error rate thresholds set in LD 73. The error counters are printed by way of LD 60.

PRI2 LOOP L

MNT	NNDC	NNC	OOS

BPV-	xxx	xxx	xxx	xxx
FAP-	xxx	xxx	xxx	xxx
SLP-	xxx	xxx	xxx	xxx
CRC-	xxx	xxx	xxx	xxx
G2 -	xxx	xxx	xxx	xxx

MAINT	NONEWCALL	UNAVAIL	SEVERE
TOTAL 24HR BPV- xxxxxxxxx	xxxxxxxxx	xxxxxxxxx	xxxxxxxxx
TOTAL 24HR CRC- xxxxxxxxx	xxxxxxxxx	xxxxxxxxx	xxxxxxxxx
TOTAL 24HR FAP- xxxxxxxxx	xxxxxxxxx	xxxxxxxxx	xxxxxxxxx
TOTAL 24HR SLPREP- xxxxxxxxx			
TOTAL 24HR SLPDEL- xxxxxxxxx			
TOTAL 24 HOUR G2 AIS - xxxxxxxxx			
TOTAL 24 HOUR G2 LFAS - xxxxxxxxx			
TOTAL 24 HOUR G2 LMAS - xxxxxxxxx			
TOTAL 24 HOUR G2 RAI - xxxxxxxxx			
TOTAL 24 HOUR G2 LOS - xxxxxxxxx			

Frame alignment

Loss-of-frame-alignment monitoring detects out-of-frame conditions on the 2.0 Mbit stream.

Loss of frame alignment thresholds

PRI hardware detects out-of-frame conditions. Running the midnight routines, prints the number of loss-of-frame-alignment occurrences and clears the counters.

If a loss-of-frame-alignment condition persists for three seconds, the affected PRI loop is taken out of service and a local alarm is raised. If the loss-of-frame-alignment condition clears for at least 15 seconds, the PRI is automatically restored to service. Three frame-alignment thresholds are set in LD 73. Setting FAP to zero enables automatic recovery.

FAP	aa bb

- aa is maintenance threshold range 1-(28)-255
- bb is out-of-service threshold range (1)-255

(Default values for these parameters are shown in brackets.)

Frame slip

Digital signals must have accurate clock synchronization for data to be interleaved into or extracted from the appropriate timeslot during multiplexing and demultiplexing operations. Frame-slip monitoring detects frame-deletion and repetition errors in clock synchronization.

Clock synchronization can track on either a primary or a secondary reference clock, or it can operate in free-run (non-tracking) mode. In LD 73 (prompts PREF and SREF), one PRI can be defined as the primary clock reference. Another can be defined as the secondary clock reference. All others are defined as free running.

Thresholds

PRI hardware detects frame slips in a tracking reference clock, or in the free-run mode. Running the midnight routines, prints the number of overflows and clears the counters.

There are two slip thresholds set in LD 73.

SLPaa   bbX cc ddX

- aa is maintenance threshold slip count; range 1-(5)-255
- bbX is maintenance threshold time (see below)
- cc is out-of-service threshold slip count; range 1-(30)-255
- ddX is out-of-service threshold time (see below)

Threshold times can be entered in milliseconds (X=T), seconds (X=S), minutes (X=M) or hours (X=H). For milliseconds, bb or dd are multiples of 20 ms, in the range 1 to 5000 (effective time range of 20 to 5000 ms). For seconds, bb or dd are in increments of seconds, in the range 1 to 240. For minutes, bb or dd are in increments of minutes, in the range 1 to 240. For hours, bb or dd are in hour increments, in the range 1 to 24.

Channel parameters

B-channels

To minimize glare situations, the system allocates B-channels from logical channel 30 down to logical channel 1.

Outgoing trunk hunting on B-channels should be set for the round-robin searching method, rather than the linear method. This way, when the switch is looking for an outgoing idle trunk, it will look for the next lower available trunk member, rather than the last member that was used.

The B-channel network loop, 0-159, and the PRI channel number, 1-30, are defined in LD 14.

D-channels

The D-channel location must be coordinated with the far end.

To establish the PRA link, the D-channel interface port number and PRI loop numbers are associated in LD 17. The DCHI port number must be an odd number, 1 through 15. PRI loop numbers can be between 0 and 159.

For information on the implementation of the Multi-purpose Serial Data Link (MSDL) DCH Handler feature, which allows a system to support up to 64 D-channels, refer to *Multi-purpose Serial Data Link*.

Interface protocols

The interface protocol between a system and another system, is a user-to-user protocol. Call processing uses a master-slave relationship for glare resolution.

One system must be designated as the master and the other as the slave (in LD 17). If the master side of the interface sends a SETUP message, as the slave initiates an outgoing call, priority is given to the call sent from the master switch. The outgoing call on that channel, the call initiated by the slave, will be dropped and another virtual B-channel will be selected for call origination.

Connection parameters for 1.5 Mbit PRI

This section describes the major parameters that must be coordinated between the system and the far end facility, over a 1.5 Mbit ISDN PRI connectivity. These parameters are as follows:

- Frame format
    - Superframe format
    - Extended superframe format
- Line encoding
    - B7 coding
    - B8ZS coding
- Error detection
    - Yellow alarm (remote alarm)
    - Bit error rate
    - Frame alignment
    - Frame slip

- Data rate parameters
    - 56 Kbit/s inverted
    - 64 Kbit/s clear and restricted
- Channel parameters
    - B-channels
    - D-channels
- Interface protocols
    - User to network
    - Symmetric

Frame formats

The DS-1 basic format consists of 24 8-bit bytes with one byte for each channel and one framing bit, or F-bit. This makes a total of 193 bits for each frame. The nominal bit rate of the DS-1 signal is 1.544 Mbit/s and the sampling rate for each channel is 8000 Hz.

Superframe format

A superframe format, the standard format, consists of 12 DS-1 frames (see Table 4: Superframe format table on page 37). It is consistent with the channel bank formats D2, D3, and D4. The signaling bit is time-shared to identify both the channel and the signaling frame. The framing pattern is the repeated sequence 100011011100.

Channel framing identifies the location of timeslot 1. The signaling frame identifies those frames in which two signaling states, A and B, are transmitted on a time-shared basis. The assignments of the F-bit and the A and B bits in the superframe format are shown in Table 4: Superframe format table on page 37.

Table 4: Superframe format table

F-Bit					
Frame number	**Terminal framing**	**Signaling framing**	**PCM coding bits**	**Signaling bit**	**Signal channel**
1	1	—	1–8		
2	—	0	1–8		
3	0	—	1–8		
4	—	0	1–8		
5	1	—	1–8		
6	—	1	1–7	8	A

F-Bit					
Frame number	Terminal framing	Signaling framing	PCM coding bits	Signaling bit	Signal channel
7	0	—	1–8		
8	—	1	1–8		
9	1	—	1–8		
10	—	1	1–8		
11	0	—	1–8		
12	—	0	1–7	8	B
Note: The most significant bit is defined as bit 1 and the least significant bit as bit 8.					

Extended superframe format

The Extended superframe format (ESF) consists of 24 frames. The 8 Kbit/s F-bit channel is divided into three separate channels.

Framing Pattern Sequence (FPS)

Beginning with frame 4 or ESF bit 579, the framing bit of every fourth frame forms FPS 001011, which is used to determine the mainframe, superframe, and robbed bit signaling synchronization. This sequence is a 2 Kbit/s channel.

Facility Data Link (FDL)

This is a 4 Kbit/s channel, used to turn on a yellow alarm. The system software uses FDL to convey yellow alarm (remote alarm) information or to transmit all 1s, as selected in service change.

Cyclic Redundancy Check (CRC)

The CRC sequence is a 2 Kbit/s channel. CRC indicates one or more bit errors in a block, or bits from the received bit stream. CRC can be used as an end-to-end bit error rate indicator.

The assignments of the F-bit and the A, B, C, and D bits in ESF are shown in Table 5: Extended superframe format table on page 38.

Table 5: Extended superframe format table

Frame number	F-Bit				Robbed bit signaling
	Bit number	Assignments			
		FPS	FDL	CRC	
1	0	—	m	—	
2	193	—	—	CB1	

Frame number	F-Bit				Robbed bit signaling
	Bit number	Assignments			
		FPS	FDL	CRC	
3	386	—	m	—	
4	579	0	—	—	
5	772	—	m	—	
6	965	—	—	CB2	A
7	1158	—	m	—	
8	1351	0	—	—	
9	1544	—	m	—	
10	1737	—	—	CB3	
11	1930	—	m	—	
12	2123	1	—	—	B
13	2316	—	m	—	
14	2509	—	—	CB4	
15	2702	—	m	—	
16	2895	0	—	—	
17	3088	—	m	—	
18	3281	—	—	CB5	C
19	3474	—	m	—	
20	3667	1	—	—	
21	3860	—	m	—	
22	4053	—	—	CB6	
23	4246	—	m	—	
24	4439	1	—	—	D

Line encoding

Line coding for DS-1 is bipolar, Alternate Mark Inversion (AMI). The general requirements for DS-1 code suppression are:

- a maximum of 15 consecutive zero binary bits
- a minimum average of 12.5 percent density of one binary bit over any 192 consecutive bits

To meet the maximum and minimum requirements, PRI provides B7 and B8ZS zero code substitution techniques as options.

B7 coding

B7 coding restricts the D-channel operating modes to 56 Kbit/s or 64 Kbit/s inverted (64 KI).

When all eight PCM bits in a channel are 0 and the eighth bit is not a signaling bit in state one, the seventh bit is substituted by a 1. This means zero code suppression is done on a single byte basis.

Note:
Do not invoke the seventh bit substitution when digital data is being transmitted as this causes data corruption.

B8ZS coding

The B8ZS coding format supports 64 Kbit/s clear channel (64 KC) or 64 Kbit/s inverted HDLC (64 KI).

When eight consecutive 0s appear on a channel and the last bit transmitted is positive, the eight bits are substituted by the following pattern:

Substituted word 0 0 0 +1 –1 0 –1 +1

If the last bit was negative, the polarity is reversed. This results in the following substituted word:

Reverse polarity 0 0 0 –1 +1 0 +1 –1

Bipolar violations occur in the fourth and seventh bit positions of the inserted code. Therefore, B8ZS coding can be used only when the receiving end is capable of recognizing that these are not bipolar violations or bit errors.

Error detection

This section describes the ISDN error detection. There are four types of error detection:

- yellow (remote) alarm
- bit error rate
- frame alignment
- frame slip

Yellow alarm (remote alarm)

A yellow alarm signal (received by the near end) indicates that the far end (the remote end) is not ready. If the PRI is receiving the yellow alarm pattern, it indicates that there is a T1 connection. When the PRI receives a yellow alarm signal from the far end, all 24 channels are disabled.

The yellow alarm method used depends on the framing format (D2, D3, D4, or ESF) selected. If D2, D3, or D4 framing formats are chosen, Digit 2 yellow alarm is automatically selected by software. If the ESF framing format is chosen, the yellow alarm method must be set through service change.

- Digit 2 (DG2) yellow alarm signaling is provided by external circuitry. This alarm is detected when each digit 2 in 63 contiguous channels is logic zero. Use DG2 yellow alarm signaling with D2, D3, and D4 frame formats in Canada and the U.S. Also use DG2 yellow alarm signaling with the ESF frame format in Canada, in compliance with Canadian standard CS03.
- Facility Data Link (FDL) yellow alarm signaling is a 4 Kbit/s channel. In the U.S., use FDL yellow alarm signaling when the ESF frame format is selected.

When the PRI stops receiving the yellow alarm, channels are placed into the idle state and made available for calls. (In comparison, TIE trunks using A&B bit signaling are made to match the state of the far end, as presented by the T1 port.)

Each time a yellow alarm is generated, a counter is incremented. When the remote alarm 24-hour threshold (RALM prompt in LD 73) is reached, the PRI must be restored to service manually.

Bit error rate

Bit error rate monitoring detects errors in transmission. There are two methods of bit error rate monitoring: bipolar violation tracking and cyclic redundancy check (CRC). If the D2, D3, or D4 framing format is selected in LD 17, bipolar violation tracking is implemented. If the extended superframe format (ESF) is selected, CRC is implemented.

Bipolar violations (BPV)

In a bipolar pulse stream, pulses alternate in polarity. A bipolar violation has occurred if, after transmission, two pulses of the same polarity are received in succession (this could be caused by an electrical disturbance, such as noise). Figure 1: Bipolar violations on page 33

Note:

Some bipolar violations are normal when using the B8ZS coding.

Cyclic redundancy check (CRC)

The extended superframe (ESF) format contains a checksum of all the data in the frame. The receiving side uses the checksum to verify that the data is correct.

The primary difference between BPV and CRC is that bipolar violations indicate errors on the local span, while CRC indicates errors on an end-to-end span. For example, on a satellite link, BPV only detects errors in the span between the system and the satellite connection. Since CRC traverses the entire span, it detects errors from the system to the satellite connection, then to the far end connection, indicating an end-to-end bit error rate.

Bit error rate thresholds

PRI hardware detects BPV or CRC errors. It sends an overflow (OVFL) message to the system CPU each time 1024 BPV or CRC errors are detected. Running the midnight routines prints the number of overflows and clears the counter.

There are three bit error rate thresholds set in LD 73. Setting BIPC to zero enables automatic recovery.

BIPV	1–(3)–4	maintenance threshold
	1–(2)–4	out-of-service threshold
BIPC	0–(2)–128	maximum number of times a DTI/PRI loop can be taken out of service in 24 hours

Note:
The BIPV values determine the sensitivity of the loop to errors, where BIPV = 1 is the least tolerant to errors, and BIPV = 4 is the most tolerant.

Frame alignment

Loss of frame alignment monitoring detects out-of-frame conditions on the DS-1 bit stream.

Loss of frame alignment thresholds

PRI hardware detects out-of-frame conditions. Running the midnight routines prints the number of loss of frame alignment occurrences and clears the counters.

If a loss of frame alignment condition persists for 3 seconds, the affected PRI loop is taken out of service and a red alarm (local alarm) is raised. If the loss of frame alignment condition clears for at least 15 seconds, the PRI is automatically restored to service. Three frame alignment thresholds are set in LD 73. Setting LFAC to 0 enables automatic recovery.

LFAL	1–(17)–10240	maintenance threshold
	1–(511)–10240	out-of-service threshold
LFAC	0–(3)–128	24–hour out of service limit

Frame slip

Digital signals must have accurate clock synchronization for data to be interleaved into or extracted from the appropriate timeslot during multiplexing and demultiplexing operations. Frame slip monitoring detects frame deletion and repetition errors in clock synchronization.

Clock synchronization can either track on a primary or secondary reference clock, or operate in free run (non-tracking) mode. In LD 73 (prompts PREF and SREF), one PRI can be defined as the primary clock reference. Another can be defined as the secondary clock reference.

Tracking mode

PRI hardware detects frame slips in a tracking reference clock. Running the midnight routines prints the number of overflows and clears the counters.

There are two thresholds set in LD 73.

SRTK	1–(5)–24	maintenance threshold (elapsed time in hours between frame slips)
	1–(30)–3600	out-of-service threshold (number of slips for each hour)

Automatic recovery

After the tracking mode (SRTK) or non-tracking mode (SRNT) out-of-service thresholds (the second value for these prompts) are exceeded, the slip rate is monitored for improvement. When the slip rate has improved, the trunks are returned to service.

There are two parameters set in LD 73:

SRIM	(1)–127	improvement timer in minutes
SRMM	1– (2)–127	improvement criteria

If the non-tracking mode maintenance threshold (the first value for SRNT) does not exceed SRMM in the duration of SRIM, then the trunks are returned to service. If not, the timer is restarted and monitoring continues.

Frame slippage is considered less important than alarms for loss of frame alignment persisting for 3 seconds, remote alarm, and bipolar violations exceeding the out-of-service threshold. If any of these alarms are reported while the slip rate is being monitored for improvement, then the monitoring stops. The trunks are returned to service only when the more serious alarm clears.

Free run (non-tracking) mode

PRI hardware detects frame slips in the free run mode. Running the midnight routines prints the number of frame deletions and repetitions and clears the counters.

Data rate parameters

ISDN uses three types of data rates 56 Kbit/s inverted, 64 Kbit/s clear, and 64 Kbit/s restricted.

56 Kbit/s inverted

A 56 Kbit/s channel is specified with the Bearer Capability Information Element (IE).

- The information transfer capability is set to restricted digital information.
- The information rate is set to 56 Kbit/s.
- The layer and protocol identification in octet 5 is set to user information layer 1 protocol, rate adaptation, and the rate is encoded as 56 Kbit/s.

64 Kbit/s clear and restricted

The 64 Kbit/s restricted (inverted HDLC), or 64 Kbit/s clear (64C) is specified in the Bearer Capability IE in the SETUP message.

The 64 Kbit/s restricted switched connections are supported by the System-to-DMS-100 protocol and the Succession1000-to-DMS-100 protocol.

The system-to-system protocol supports 64 Kbit/s clear transmission.

Channel parameters

There are two types of ISDN channels B-channels and D-channels.

B-channels

To minimize glare situations, the system allocates B-channels from channel 23 down to channel 1; DMS-100 begins at channel 1 and goes up to channel 23.

Set outgoing trunk hunting on B-channels for the round robin searching method, rather than the linear method. Thus, when the system looks for an outgoing idle trunk, it looks for the next lower available trunk member, rather than the last member that was used.

The B-channel network loop, 0–511, and the PRI channel number, 1–23, are defined in LD 14.

D-channels

The D-channel location must be coordinated with the far end. See the correlation tables in this document for specific information.

To establish the PRI link, the D-channel interface port number and PRI loop numbers are associated in LD 17. The DCHI port number must be an odd number between 1 and 15. PRI loop numbers can be between 0 and 511.

The D-channel can operate at 56 or 64 Kbit/s data rate. It can be 64 Kbit/s clear, 64 Kbit/s restricted (inverted HDLC), or 56 Kbit/s. The selection of data rate is on a PRI basis and is determined by service change. The B8ZS zero code suppression method is used to achieve 64 Kbit/s clear channel for the D-channel.

For incoming PRI messages, the printout of the D-channel monitor message differs from the actual message received by the system. This is due to the fact that layer 2 preprocesses the message before sending it to layer 3. Outgoing PRI messages appear exactly as sent.

Interface protocols

User-to-network

The interface protocol between the system and the central office (CO) PRI equipment is a user-to-network protocol. If the far end is identified as a CO in service change (LD 17), the system is automatically designated as the "user."

The user-to-network protocol does not employ the same call states and state transactions at each end, and does not always send the same response to a given protocol message. In addition, the user-to-network protocol has an implicit master-slave relationship, relinquishing control to the network in cases such as glare resolution.

Symmetric

The interface protocol between a system and another system is a symmetric protocol. Call processing uses a master-slave relationship for glare resolution.

One PBX must be designated as the network (master), the other as the user (slave) (in LD 17). If the master side of the interface sends a SETUP message as the slave initiates an outgoing call, priority is given to the call sent from the master switch. The outgoing call on that channel, the call initiated by the slave, is dropped and another virtual B-channel is selected for call origination.

Chapter 6: System correlation tables

Contents

This section contains information on the following topics:

System-to-system correlation tables over 2.0 Mbit PRI

The correlation tables that follow are used to coordinate the software features between two switches over 2.0 Mbit ISDN PRI. These tables describe how to coordinate the software features between two switches. The first set of tables indicates which prompts in which overlays are to be given the same responses on each system. (The possible responses to these prompts are discussed elsewhere in this technical document.) The second table shows the prompt which requires a different response at each system, and provides the response to be entered.

Table 6: PRI database correlation (protocol layer 1): Overlay prompts to be matched

Overlay	Prompts/Commands	Description
17	PRI2	Digital connection type, on a loop basis
60	DISY L	Inhibit alarm transmission
73	BIPV	Bit error rate maintenance threshold Bit error rate out-of-service threshold
	BIPC	Bit error rate 24-hour threshold
	FAP	Frame alignment maintenance threshold Frame alignment out-of-service threshold

Table 7: DCH database correlation (protocol layer 2): Overlay prompts to be matched

Overlay	Prompts/Commands	Description
17	DCHI DCHL	Associate D-channel with PRI2
	USR	DCHI mode
	RCVP	D-channel recovery

Table 8: Facility database correlation (protocol layer 3): overlay prompts to be matched

Overlay	Prompts/Commands	Description
16	SRCH MODE	B-channel selection DCHI mode route
14	TN	B-channels defined
17	IFC	Interface type

Table 9: Database correlation: Prompt requiring different responses at each system

Description	system	system
USER-USER INTERFACE	Overlay:17 Prompt:SIDE Response:MAS	Overlay:17 Prompt:SIDE Response:SLAV

System-to-system correlation tables over 1.5 Mbit PRI

The correlation tables that follow are used to coordinate the software features between two switches over 1.5 Mbit ISDN PRI.

The following correlation tables are contained in this module:

- system switch to another system switch
- system switch to DMS-100 switch
- system switch to DMS-250 switch

Note:
The system to SL-100 datafill is the same as the system to DMS-100 datafill.

Note:
Due to proprietary constraints, the system to AT&T 4ESS and AT&T 5ESS datafill is not available for publication in this document.

The following tables are provided for system to system:

- Table 10: PRI database correlation (protocol layer 1) for system-to-system on page 50 PRI database correlation (protocol layer 1)
- Table 11: DCH database correlation (protocol layer 2) for system-to-system on page 51 DCH database correlation (protocol layer 2)
- Table 12: Facility database correlation (protocol layer 3) for system-to-system on page 51 Facility database correlation (protocol layer 3)

The following tables are provided for system to DMS-100:

- Table 13: PRI database correlation (protocol layer 1) for system to DMS-100 on page 52 PRI database correlation (protocol layer 1)
- Table 14: DCH database correlation (protocol layer 2) for system to DMS-100 on page 54 DCH database correlation (protocol layer 2)
- Table 15: Facility database correlation (protocol layer 3) for system to DMS-100 on page 54 Facility database correlation (protocol layer 3)

The following tables are provided for system to DMS-250:

- Table 16: PRI database correlation (protocol layer 1) for System to DMS-250 on page 55 PRI database correlation (protocol layer 1)
- Table 17: DCH database correlation (protocol layer 2) for system to DMS-250 on page 56 DCH database correlation (protocol layer 2)
- Table 18: Facility database correlation (protocol layer 3) for system to DMS-250 on page 57 Facility database correlation (protocol layer 3)

System-to-system correlation tables

Table 10: PRI database correlation (protocol layer 1) for system-to-system on page 50 and Table 11: DCH database correlation (protocol layer 2) for system-to-system on page 51 describe how to coordinate the software features between two system switches. The tables consist of three columns. The description column lists the software feature to be coordinated. The first system column lists the system software prompts and the proper responses for a corresponding feature. The second system column lists the software tables and the correct values for the fields in these tables. The system information also corresponds to a particular feature.

Each table corresponds to one of three protocol layers.

Note:

Both the near and the far ends must match the parameters. For example, LCMT, and YALM responses must be the same for both ends. This applies to all the following tables for system to system configuration.

Table 10: PRI database correlation (protocol layer 1) for system-to-system

Description	System		System	
CARD TYPE	Program: Prompt: Response:	LD 17 MODE PRI	Program: Prompt: Response:	LD 17 MODE PRI
Superframe Extended Superframe (ESF)		ESF		ESF
LINE ENCODING	Program: Prompt: Response:	LD 17 LCMT AMI	Program: Prompt: Response:	LD 17 LCMT AMI
Zero code suppression Bit 8 zero suppression		B8S		B8S
BIT ERROR RATE BASE	Program: Prompt:	LD 73 n/a (preset to four classes of error rates)	Program: Prompt:	LD 73 n/a (preset)
Bipolar violations CRC				
DATA LINK (yellow alarm method)	Program: Prompt: Response:	LD 17 YALM DG2 FDL	Program: Prompt: Response:	LD 17 YALM DG2 FDL
INHIBIT ALARM TRANSMISSION	Program: Prompt:	LD 60 DISY L (disable yellow alarm for loop L)	Program: Prompt:	LD 60 DISY L (disable yellow alarm for loop L)
BIT ERROR RATE maintenance threshold	Program: Prompt: Response:	LD 73 BIPV 1–(3)–4	Program: Prompt: Response:	LD 73 BIPV 1–(3)–4
BIT ERROR RATE out of service threshold	Program: Prompt: Response:	LD 73 BIPC 1–(2)–4	Program: Prompt: Response:	LD 73 BIPC 1–(2)–4
BIT ERROR RATE 24-hour threshold (error second threshold)	Program: Prompt: Response:	LD 73 BIP 0–(3)–128	Program: Prompt: Response:	LD 73 BIP 0–(3)–128
FRAME ALIGNMENT maintenance threshold	Program: Prompt: Response:	LD 73 LFAL 1–(17)–10240	Program: Prompt: Response:	LD 73 LFAL 1–(17)–10240
FRAME ALIGNMENT out of service threshold	Program: Prompt: Response:	LD 73 LFAL 1–(511)–10240	Program: Prompt: Response:	LD 73 LFAL 1–(511)–10240
FRAME SLIP maintenance threshold	Program: Prompt: Response:	LD 73 SRNT 1–(15)–10240	Program: Prompt: Response:	LD 73 SRNT 1–(15)–10240

Description	System		System	
FRAME SLIP out of service threshold	Program: Prompt: Response:	LD 73 SRNT 1–(3)–10240	Program: Prompt: Response:	LD 73 SRNT 1–(3)–10240

Table 11: DCH database correlation (protocol layer 2) for system-to-system

Description	System		System	
Associate D-channel with PRI	Program:	LD 17	Program:	LD 17
	Prompt and Response:		Prompt and Response:	
	ADAN DCHL	DCH xx PRI loop	ADAN DCHL	DCH xx PRI loop
Associate backup D-channel with PRI	Program:	LD 17	Program:	LD 17
	Prompt and Response:		Prompt and Response:	
	ADAN	BDCH x is the backup D-channel number	ADAN	BDCH x is the backup D-channel number
	BCHL	xx is the associated PRI card (or PRI loop)	BCHL	xx is the associated PRI card (or PRI loop)
	RCVP	Yes requests recovery to the primary D-channel	RCVP	Yes requests recovery to the primary D-channel
Data rate of D-channel	Program:	LD 17	Program:	LD 17
	Prompt and Response:		Prompt and Response:	
	DRAT	64KC, 56KI	DRAT	64KC, 56KI
DCH mode	Program:	LD 17	Program:	LD 17
	Prompt and Response:		Prompt and Response:	
	USR	PRI, SHA, ISLD	USR	PRI, SHA, ISLD

Table 12: Facility database correlation (protocol layer 3) for system-to-system

Description	System		System	
B-channel selection	Program: Prompt: Response:	LD 16 SRCH RRB	Program: Prompt: Response:	LD 16 SRCH RRB

Description	System		System	
Loss and level		preset		preset
User-user interface	Program: Prompt: Response:	LD 17 SIDE NET	Program: Prompt: Response:	LD 17 SIDE NET
B-channels defined	Program: Prompt: Response:	LD 14 TN network loop and channel	Program: Prompt: Response:	LD 14 TN network loop and channel
Interface type	Program: Prompt: Response:	LD 17 IFC SL1	Program: Prompt: Response:	LD 17 IFC SL1
DCH mode route	Program: Prompt: Response:	LD 16 MODE PRI, ISLD	Program: Prompt: Response:	LD 16 MODE PRI, ISLD

System-to-DMS-100 correlation tables

Table 13: PRI database correlation (protocol layer 1) for system to DMS-100 on page 52 through Table 15: Facility database correlation (protocol layer 3) for system to DMS-100 on page 54 describe how to coordinate the software features between a system switch and a DMS-100 switch. The tables consist of three columns. The description column lists the software feature to be coordinated. The system column lists the software prompts and the proper responses for a corresponding feature. The DMS-100 column lists the software tables and the correct values for the fields in these tables. The DMS-100 information also corresponds to a particular feature.

Each table corresponds to one of three protocol layers.

Table 13: PRI database correlation (protocol layer 1) for system to DMS-100

Description	System		DMS-100	
Card type	Program: Prompt: Response:	LD 17 MODE PRI	Table: Field: Value:	CARRMTC CARD NT6X50AA NT6X50AB
Frame format	Program: Prompt:	LD 17 DLOP (field ff)	Table: Field:	CARRMTC FF
Superframe Extended Superframe	Response:	D3 ESF	Value:	SF ESF
Line encoding	Program: Prompt:	LD 17 LCMT	Table: Field:	CARRMTC ZLG

Description	System		DMS-100	
Zero code suppression Bit 8 zero suppression	Response:	AMI B8S	Value:	ZCS B8ZS
Bit error rate base	Program: Prompt:	LD 73 n/a (preset to four classes of error rates)	Table: Field:	CARRMTC BERB
Bipolar violations CRC			Value:	BPV CRC
Data link (yellow alarm method) No data link	Program: Prompt: Response:	LD 17 YALM DG2 (Note) FDL	Table: Field: Value:	CARRMTC DLK NILDL

Note:
When the DMS-100 CARRMTC table has Field = FF and Value = SF, configure the system with DG2. When the DMS-100 CARRMTC table has Field = FF and Value = ESF, configure the system with FDL.

Description	System		DMS-100	
Inhibit alarm transmission	Program: Prompt:	LD 60 DISY L (disable yellow alarm for loop L)	Table: Field: Value:	CARRMTC IAT Y N
Bit error rate maintenance threshold	Program: Prompt: Response:	LD 73 BIPV 1–(3)–4	Table: Field: Value:	CARRMTC BERML 6
Bit error rate out of service threshold	Program: Prompt: Response:	LD 73 BIPV 1–(2)–4	Table: Field: Value:	CARRMTC BERML 3 (exponent)
Bit error rate 24-hour threshold (error second threshold)	Program: Prompt: Response:	LD 73 BIPC 0–(3)–128	Table: Field: Value:	CARRMTC ES 864
Frame alignment maintenance threshold	Program: Prompt: Response:	LD 73 LFAL 1–(17)–10240	Table: Field:	CARRMTC FRAMEML
Frame alignment out of service threshold	Program: Prompt: Response:	LD 73 LFAL 1–(511)–10240	Table: Field: Value:	CARRMTC FRAMEOL 511 (exponent)
Frame slip maintenance threshold	Program: Prompt: Response:	LD 73 SRNT 1–(15)–1024	Table: Field: Value:	CARRMTC SLIPML 4
Frame slip out of service threshold	Program: Prompt: Response:	LD 73 SRNT 1–(3)–1024	Table: Field: Value:	CARRMTC SLIPOL 255 (exponent)

Table 14: DCH database correlation (protocol layer 2) for system to DMS-100

Description	System		System	
Associate D-channel with PRI	Program:	LD 17	Table:	TRKSGRP
	Prompt and Response:		Field:	DCHNL
	ADAN DCHL	DCH xx PRI loop	Value:	same as DS1 end point in table SPECCONN
Associate backup D-channel with PRI	Program:	LD 17	Table:	TRKSGRP
	Prompt and Response:		Field:	DCHBCKUP
	ADAN	BDCH x is the backup D-channel number	Value:	Same as DS1 end point in table SPECCONN
	BCHL	xx is the associated PRI card (or PRI loop)		
	RCVP	Yes requests recovery to the primary D-channel		
Data rate of D-channel	Program:	LD 17	Table:	STINV
	Prompt and Response:		Field:	CONTYPE
	DRAT	64KC, 56KI	Value: Field: Value:	PRIBAUD BAUD 64 Kbit/s 56 Kbit/s

Table 15: Facility database correlation (protocol layer 3) for system to DMS-100

Description	System 1		DMS-100	
Q.931 Interface identifier (used in CID IE)	n/a		Table: Field: Value:	IACPSINV IID 0
Q.931 Call Reference Value Length	n/a		Table: Field: Value:	TRKSGRP CRLENGTH 2
B-channel selection	Program: Prompt: Response:	LD 16 SRCH RRB LIN	Table: Field: Value:	TRKGRP SELSEQ MIDL ASEQ
Billing at Primary Rate Interface (PRI)	n/a		Table: Field: Value:	TRKGRP BILLDN N

Description	System 1		DMS-100	
loss and level	preset		Table: Field: Value:	TRKGRP PADGRP PRAC
User-network interface	Program: Prompt: Response:	LD 17 IFC D100 (sets user)	Table: Field: Value:	TRKSGRP IFCLASS NETWORK
Q.931 progress indicator location	n/a		Table: Field: Value:	TRKSGRP LOCATION USER
B-channels defined	Program: Prompt: Response:	LD 14 TN 0–159 = network loop 1–23 = channel	Table: Field: Value:	TRKMEM EXTTRKMEM IACCKTTS

System-to-DMS-250 correlation tables

Table 16: PRI database correlation (protocol layer 1) for System to DMS-250 on page 55 through Table 18: Facility database correlation (protocol layer 3) for system to DMS-250 on page 57 describe how to coordinate the software features between a system switch and a DMS-250 switch. The tables consist of three columns. The description column lists the software feature to be coordinated. The system column lists the system software prompts and the proper responses for a corresponding feature. The DMS-250 column lists the software tables and the correct values for the fields in these tables. The DMS-250 information also corresponds to a particular feature.

Each of the tables corresponds to one of three protocol layers.

Table 16: PRI database correlation (protocol layer 1) for System to DMS-250

Description	System		DMS-250	
Card type	Program: Prompt: Response:	LD 17 MODE PRI	Table: Field: Value:	CARRMTC CARD NT6X50AA NT6X50AB
Line encoding	Program: Prompt: Response:	LD 17 LCMT AMI B8S	Table: Field: Value:	CARRMTC ZLG ZCS B8ZS
Zero code suppression Bit 8 zero suppression				

Description	System		DMS-250	
Bit error rate base	Program: Prompt:	LD 73 n/a (preset to four classes of error rates)	Table: Field: Value:	CARRMTC BERB BPV CRC
Bipolar violations CRC				
Data link (yellow alarm method) No data link	Program: Prompt: Response:	LD 17 YALM n/a DG2 FDL	Table: Field: Value:	CARRMTC DLK NILDL FDL1 FDL2
Inhibit alarm transmission	Program: Prompt:	LD 60 DISL/X (disable yellow alarm for loop L)	Table: Field: Value:	CARRMTC IAT YN
Bit error rate maintenance threshold	Program: Prompt: Response:	LD 73 BIPV 1–(3)–4	Table: Field: Value:	CARRMTC BERML 6
Bit error rate out of service threshold	Program: Prompt: Response:	LD 73 BIPV 1–(2)–4	Table: Field: Value:	CARRMTC BEROL 3 (exponent)
Bit error rate 24-hour threshold (error second threshold)	Program: Prompt: Response:	LD 73 BIPC 0–(3)–128	Table: Field: Value:	CARRMTC ES 864
Frame alignment maintenance threshold	Program: Prompt: Response:	LD 73 LFAL 1–(17)–10240	Table: Field: Value:	CARRMTC FRAMEML 17
Frame alignment out of service threshold	Program: Prompt: Response:	LD 73 LFAL 1–(511)–10240	Table: Field: Value:	CARRMTC FRAMEOL 511 (exponent)
Frame slip maintenance threshold	Program: Prompt: Response:	LD 73 SRNT 1–(15)–1024	Table: Field: Value:	CARRMTC SLIPML 4
Frame slip out of service threshold	Program: Prompt: Response:	LD 73 SRNT 1–(3)–1024	Table: Field: Value:	CARRMTC SLIPOL 255 (exponent)

Table 17: DCH database correlation (protocol layer 2) for system to DMS-250

Description	System		DMS-250	
Associate D-channel with PRI	Program:	LD 17	Table: Field: Value: Field: Value: Field:	TRKSGRP PMTYPEDTCI DTCINO Nil DTCICKTNO 16 DTCICKTTS 24
	Prompt and Response:			
	ADAN DCHL	DCH xx PRI loop		

Description	System	DMS-250
		Value: Field: Value:
Data rate of D-channel	Program: LD 17 Prompt and Response: DRAT 64KC, 56KI	Table: TRKSGRP Field: DCHRATE 56 Kbit/s Value: 64 Kbit/s Value:

Table 18: Facility database correlation (protocol layer 3) for system to DMS-250

Description	System	DMS-250
Q.931 Interface identifier (used in CID IE)	n/a	Table: LTCTSINV Field: PSLNKTAB O/ Value: DSIPRA/ Default/N/Nil
Q.931 Call Reference Value Length	n/a	Table: TRKSGRP Field: CRLENGTH 2 Value:
B-channel selection	Program: LD 16 SRCH Prompt: RRB/LIN Response:	Table: TRKGRP Field: SELSEQ Value: MIDLASEQ
Billing at PRI interface	n/a	Table: TRKGRP Field: BILLDN N Value:
Loss and level	preset	Table: TRKGRP Field: PADGRP PRAC Value:
User-network interface	Program: LD 17 IFC D250 Prompt: (sets user) Response:	Table: TRKSGRP Field: IFCLASS Value: NETWORK
Q.931 progress indicator location	n/a	Table: TRKSGRP Field: LOCATION Value: USER
Backup D-channels defined	Program: LD 14 TN 0–159 Prompt: = network loop Response: 1–23 = channel	Table: TRKMEM Field: PMTYPE DTCI Value: DTCINO Nil Field: DTCICKTNO 16 Value: DTCICKTTS 5 Field: Value: Field: Value:

Chapter 7: Data administration

Contents

This section contains information on the following topics for Avaya Communication Server 1000 (Avaya CS 1000):

PRI implementation

Task summary list

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

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2. Table 20: LD 73 - Change existing thresholds or change tracking modes. on page 61
3. Table 21: LD 17 - Configure a PRI interface on the system PBX. on page 63
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6. Table 24: LD 17 - Configure nB+D on the system. on page 66
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8. Table 26: LD 16 - Configure a PRI route. on page 68
9. Table 27: LD 14 - Configure PRI trunks. on page 70

For the following LD 73, set error detection thresholds and clock synchronization control. This configuration is required only when configuring PRI for the first time, and is optional for subsequent configurations.

Table 19: LD 73 - Set error detection thresholds and clock synchronization control.

Prompt	Response	Description
REQ	NEW	Create a PRI data block
TYPE	DDB	Digital data block
PREF	xx	Primary reference source for clock controller.
	<cr>	Free run mode No primary or secondary reference source assigned X preceding the number deletes existing primary reference source
SREF	xx	Secondary reference source for clock controller prompted only if PREF is not free run
	<cr>	Free run mode No primary or secondary reference source assigned X preceding the number deletes existing secondary reference source
TRSH	0–99	Create or change a PRI threshold set Enter this number in LD 17 when defining the PRI loop
		X preceding number deletes threshold set
RALM	1–(3)–128	Yellow alarm (remote alarm) 24-hour threshold Number of remote alarm clear signals received in 24 hours
		If the threshold is reached, the PRI must be restored to service manually
BIPC	0–(2)–128	24-hour bit rate violation threshold If zero is entered, trunks are restored to service automatically.
		With D2, D3, or D4 framing format, bipolar violation threshold With ESF, Cyclic Redundancy Check (CRC) threshold
LFAC	0–(3)–128	24-hour loss of frame alignment threshold If zero is entered, trunks are restored to service automatically.
BIPV		Bit rate (bipolar violation and CRC) monitoring limits
	1–(3)–4	Maintenance threshold, the minimum time, in hours, between slips
	1–(2)–4	Out-of-service threshold, the maximum number of slips per hour
SRTK		Frame slip tracking monitoring limits
	1–(5)–24	Maintenance threshold

Prompt	Response	Description
	1–(30)–3600	Out-of-service threshold
SRNT		Frame slip free run (non-tracking) monitoring limits
	1–(15)–1024	Maintenance threshold, the minimum time, in seconds, between 10 consecutive slips
	1–(3)–1024	Out-of-service threshold, the minimum time, in seconds, between 10 consecutive slips.
LFAL		Loss of frame alignment monitoring limits
	1–(17)–10240 1–(511)–10240	Maintenance threshold Out-of-service threshold
SARR	YES–(NO)	Automatic recovery allowed after out-of-service condition
SRIM	(1)-127	Slip Rate Improvement Time, in minutes.
SRMM	1–(2)–127	Slip rate exceeded maintenance limit

Table 20: LD 73 - Change existing thresholds or change tracking modes.

Prompt	Response	Description
REQ	CHG	Change a PRI data block
TYPE	DDB	Digital data block
PREF	xx	Primary reference source for clock controller
	<cr>	Free run mode No primary or secondary reference source assigned X preceding the number deletes existing primary reference source
SREF	xx	Secondary reference source for clock controller Prompted only if PREF is not free run
	<cr>	Free run mode No primary or secondary reference source assigned X preceding the number deletes existing secondary reference source
TRSH	0–99	Create or change a PRI threshold set Enter this number in LD 17 when defining the PRI loop
		X preceding number deletes threshold set.
RALM	1–(3)–128	Yellow alarm (remote alarm) 24-hour threshold Number of remote alarm clear signals received in 24 hours
		If the threshold is reached, the PRI must be restored to service manually
BIPC	0–(2)–128	24-hour bit rate violation threshold If zero is entered, trunks are restored to service automatically

Prompt	Response	Description
		With D2, D3, or D4 framing format, bipolar violation threshold With ESF, Cyclic Redundancy Check (CRC) threshold
LFAC	0–(3)–128	24-hour loss of frame alignment threshold If zero is entered, trunks are restored to service automatically
BIPV		Bit rate (bipolar violation and CRC) monitoring limits
	1–(3)–4	Maintenance threshold
	1–(2)–4	Out-of-service threshold
SRTK		Frame slip tracking monitoring limits
	1–(5)–24	Maintenance threshold, the minimum time, in hours, between slips
	1–(30)–3600	Out-of-service threshold, the maximum number of slips per hour
SRNT		Frame slip free run (non-tracking) monitoring limits
	1–(15)–1024	Maintenance threshold, the minimum time, in seconds, between 10 consecutive slips
	1–(3)–1024	Out-of-service threshold, the minimum time, in seconds, between 10 consecutive slips. See "Coordinating PRI parameters" and *Avaya ISDN Primary Rate Interface Maintenance, NN43001-717* for a description of the automatic recovery sequence.
LFAL		Loss of frame alignment monitoring limits
	1–(17)–10240	Maintenance threshold
	1–(511)–10240	Out-of-service threshold
SRIM	(1)–127	Slip Rate Improvement Time in minutes
SRMM	1–(2)–127	Slip Rate Exceeded Maintenance limit
BIPV		Bit rate (bipolar violation and CRC) monitoring limits
	1–(3)–4	Maintenance threshold
	1–(2)–4	Out-of-service threshold
SRTK		Frame slip tracking monitoring limits
	1–(5)–24	Maintenance threshold, the minimum time, in hours, between slips
	1–(30)–3600	Out-of-service threshold, the maximum number of slips per hour
SRNT		Frame slip free run (non-tracking) monitoring limits

Prompt	Response	Description
	1–(15)–1024	Maintenance threshold, the minimum time, in seconds, between 10 consecutive slips
	1–(3)–1024	Out-of-service threshold, the minimum time, in seconds, between 10 consecutive slips.
LFAL		Loss of frame alignment monitoring limits
	1–(17)–10240	Maintenance threshold
	1–(511)–10240	Out-of-service threshold
SARR	YES–(NO)	Automatic recovery allowed after out-of-service condition
SRIM	(1)-127	Slip Rate Improvement Time, in minutes.
SRMM	1–(2)–127	Slip rate exceeded maintenance limit

Table 21: LD 17 - Configure a PRI interface on the system PBX.

Prompt	Response	Description
REQ	CHG	Change existing data.
TYPE	CEQU	Make changes to Common Equipment parameters.
...		
- PRI	xx	PRI loop number for Large Systems.

Table 22: LD 17 - Configure the Primary 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 logical port 0-63, for Large Systems.
- CTYP	MSDL TMDI	Card type where: MSDL = NT6D80 Multi-purpose Serial Data Link, or the NTBK51AA/NTBK51CA Downloadable D-Channel Daughterboard, for Large Systems.
- GRP	0-4	Network group number For Large Systems For Multi groups Fiber Network Fabric
- DNUM	0-15	Device number for I/O ports. All ports on the MSDL card share the same DNUM. The MSDL card address settings must match the DNUM value.
- PORT	0-3	Port number on the NT6D80 MSDL card, if the MSDL is used for D-Channel handling on Large Systems.

Prompt	Response	Description
	0-1	Port number of the NTBK51AA/NTBK51CA, if the NTBK51AA/NTBK51CA is used for D-Channel handling on Large Systems. Port 0 of the NTBK51AA/NTBK51CA can only be defined to work with Loop 0 of the NT5D97AA DDP2 card, and Port 1 of the NTBK51AA/NTBK51CA can only be defined to work with Loop 1 of the NT5D97AA. This relationship must be reflected in the DCHL prompt, which follows later (either 0 or 1 must be entered when specifying the loop number used by the D-Channel).
- DES	aaa...a	Designator. DES is used to identify the link and can be up to 16 alphanumeric characters: 0-9, and upper case A-Z. Characters "*" and "#" are not allowed.
- USR	PRI	This D-channel is used for Primary Rate only.
- IFC	xx	Interface type.
- - CNTY	xx	Country of connectivity associated with IFC type.
- - DCHL	0-159	PRI loop number for the D-channel, for Large Systems. If the NTBK51AA/NTBK51CA is used for D-Channel handling, only loop 0 or 1 can be configured.
- PRI	0-159 (0)-15 1-9	Secondary PRI loops and sequence, for nB+D configuration.
- OTBF	0-(32)-127	Output Request Buffers.
- SIDE	(USR) NET	The system is network side.
- RLS	xx	Software release of the far end switch.
- RCAP	aaa	Remote capabilities. Enter <cr> when finished entering values.
- OVLR	(NO) YES	Allow Overlap Receiving.
- OVLS	(NO) YES	Allow Overlap Sending.
- MBGA	(NO) YES	Allow Multi-location Business Group.
- NASA	(NO) YES	Allow Network Attendant Service.
- TIMR	(NO) YES	Change Protocol Timer values.
- LAPD	(NO) YES	Change Link Access Protocol for D-Channel parameters.

Table 23: LD 17 - Configure a Back-up D-channel. This step is optional and is performed only if a Back-up D-Channel is required.

Prompt	Response	Description
REQ	CHG	Change existing data.
TYPE	ADAN	Action device and number.
- ADAN	NEW BDCH xx	Add a Back-up D-channel on logical port 0-63 for Large Systems.
- PDCH	0-63	0-63 = Primary D-Channel associated with the Backup D-Channel for Large Systems.
- CTYP		Card type where: MSDL = NT6D80 Multi-purpose Serial Data Link, or the NTBK51AA/NTBK51CA Downloadable D-Channel Daughterboard for Large Systems.
- GRP	0-4	Network group number for Meridian 1 PBX 81Csystems.
- DNUM	0-15	Device number for I/O ports. All ports on the MSDL card share the same DNUM. The MSDL card address settings must match the DNUM value.
- PORT	0-3	Port number on the NT6D80 MSDL card, if the MSDL is used for D-Channel handling on Large Systems.
	0-1	Port number of the NTBK51AA/NTBK51CA, if the NTBK51AA/NTBK51CA is used for D-Channel handling on Large Systems. Port 0 of the NTBK51AA/NTBK51CA can only be defined to work with Loop 0 of the NT5D97AA DDP2 card, and Port 1 of the NTBK51AA/NTBK51CA can only be defined to work with Loop 1 of the NT5D97AA. This relationship must be reflected in the DCHL prompt, which follows later (either 0 or 1 must be entered when specifying the loop number used by the D-Channel).
- RCVP	YES	Auto-recovery to primary D-Channel.
- BCHL	0-159	PRI/PRI2 loop number for the Back-up D-channel, for Large Systems. If the NTBK51AA/NTBK51CA is used for D-Channel handling, only loop 0 or 1 can be configured.

For the following table LD 17, configure nB+D on the system. This step is optional and requires the International nB+d (INBD) package 255 to be equipped for the following interfaces only: D70, JAPN, HKNG, TCNZ, and MSIA.

Table 24: LD 17 - Configure nB+D on the system.

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 for Large Systems.
- CTYP		Card type where: MSDL = NT6D80 Multi-purpose Serial Data Link or the NTBK51AA/NTBK51CA Downloadable D-Channel Daughterboard for Large Systems.
- GRP	0-4	Network group number for Meridian 1 PBX 81Csystems.
- DNUM	0-15	Device number for I/O ports. All ports on the MSDL card share the same DNUM. The MSDL card address settings must match the DNUM value.
- PORT	0-3	Port number on the NT6D80 MSDL card, if the MSDL is used for D-Channel handling on Large Systems.
	0-1	Port number of the NTBK51AA/NTBK51CA, if the NTBK51AA/NTBK51CA is used for D-Channel handling on Large Systems. Port 0 of the NTBK51AA/NTBK51CA can only be defined to work with Loop 0 of the NT5D97AA DDP2 card, and Port 1 of the NTBK51AA/NTBK51CA can only be defined to work with Loop 1 of the NT5D97AA. This relationship must be reflected in the DCHL prompt, which follows later (either 0 or 1 must be entered when specifying the loop number used by the D-Channel).
- DES	aaa...a	Designator. DES is used to identify the link and can be up to 16 alphanumeric characters: 0-9, and upper case A-Z. Characters "*" and "#" are not allowed.
- USR	PRI	This D-channel is used for Primary Rate only.
- IFC	xx	Interface type that supports nB+D.
- - CNTY	xx	Country of connectivity associated with IFC type.
- - DCHL	LOOP ID 0-159 (0)-15	Primary PRI loop number and interface identifier, for the D-channel. When INBD package 255 is enabled values for both the PRI loop number (LOOP) and the D-channel interface identifier (ID) must be entered. For Large Systems.
- PRI	0-159 1-126	Secondary PRI loop number and interface identifier, for nB+D (prompted if INBD package 255 is enabled).

Prompt	Response	Description
	(LOOP) (ID)	The values entered must be different than those entered for the loop number and interface identifier at the DCHL prompt for Large Systems. The PRI prompt is generated until <CR> is entered.
PRI	<CR>	End configuration.

Table 25: LD 73 - Define the digital pad tables by country requirement.

Prompt	Response	Description
REQ	NEW	NEW = Add new data.
	CHG	CHG = Change existing data.
	OUT	OUT = Remove existing data.
	END	END = Exit LD 73.
	PRT	PRT = Print specified data.
TYPE	PRI	Primary Rate Interface.
FEAT	PAD	Feature is digital pad.
PDCA	(1)-16	Pad category table number.
TNLS	YES NO	Print TN list (if REQ = PRT).
DFLT	(1)-16	Default table (when REQ = NEW).
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 0-15. The pad values (in decibels) relating to these codes are shown after this table.		
ONP	r t	On-premises extension.
DSET	r t	Meridian Digital phone (prompted only if the 1.5/2.0 Mbit/s Gateway feature is available and equipped).
OPX	r t	Off-premises extension.
DTT	r t	Digital TIE trunks.
NTC	r t	Non-transmission compensated.
TRC	r t	Transmission compensated.
DTR		
DCO	x y	digital COT, FEX, WAT, and DID trunks.
VNL	r t	VIA NET LOSS.
ACO	r t	Analog CO or WATS trunks.
AFX	r t	Analog FEX trunks.
ADD	r t	Analog DID trunks.

Prompt	Response	Description
PRI	r t	1.5 Mbit/s PRI/DTI trunk (prompted only if the 1.5/2.0 Mbit/s Gateway feature is available and equipped and TYPE=PRI2).
PRI2	r t	2.0 Mbit/s PRI/DTI trunk (prompted only if the 1.5/2.0 Mbit/s Gateway feature is available and equipped and TYPE=PRI).
XUT	r t	Analog CO trunk
XEM	r t	Analog TIE trunk

Table 26: LD 16 - Configure a PRI 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, Media Gateway 1000B, and CS 1000E system.
DES	PRI	Designator field for trunk.
TKTP	TIE	TIE trunk type (this is the only type allowed).
...		
DTRK	YES	Digital trunk route.
- DGTP	PRI	PRI digital trunk.
ISDN	YES	Integrated Services Digital Network.
- MODE	PRA	ISDN PRI route.
- IFC	xx	Interface type.
- - CNTY	xx	Country of connectivity associated with IFC type.
- - PNI	(0)-32700	Private Network identifier.
- NCNA	YES	Network Calling Name allowed.
- NCRD	YES	Network Call Redirection allowed.
...		
ICOG	IAO	Incoming and Outgoing trunks.
SRCH	(LIN) RRB	Search method for outgoing trunk member.

Prompt	Response	Description
		LIN = Linear Hunt search method. RRB = Round Robin search method.
...		
ACOD	xxx..x	One-seven-digit access code for the trunk route.
TARG	0-(1)-31	Trunk Access Restriction Group Number.
SIGO	aaaa	Signaling Arrangement. aaaa = (STD), ESN2, ESN3, ESN5, ETN, EN19.
...		
DTRK	(NO) YES	Digital trunk route. YES=digital, NO=analog.
DGTP	PRI2	Select a digital trunk type of 2.048 Mb/s PRI.
ISDN	(NO) YES	ISDN option.
MODE	PRA ISL	ISA route for ISDN PRA. ISA route for ISL application. There is no default.
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). When interworking with different networks, the Customer Data Block PNI is the PNI of your switch. The Route Data Block PNI is the PNI of the target (remote) switch.
CTYP	DCHI MSDL	The card type on which the I/O device is to be configured. There is no default. You must choose one of the available responses.
PTYP		Port type at far end: Analog TIE trunk routes:
	(ATT) AST	Analog TIE trunk Analog satellite TIE trunk or ESN satellite TIE trunk
	AOT	Analog TIE trunk, used instead of ATT whenever the system has one or more digital satellite trunk routes (DST) to any digital satellite system which includes OPX sets
	(DTT) DCT DST	Digital TIE trunk routes: Digital TIE trunk Combination digital TIE trunk Digital satellite system TIE trunk
AUTO	(NO)YES	Auto-terminate must be NO if response to DSEL is VOD
ICOG	IAO ICT OGT	Incoming and outgoing trunk Incoming trunk Outgoing trunk
SRCH	(LIN) RRB	Linear search. round-robin search—use for outgoing trunks.

Prompt	Response	Description
ACOD	xxxx	Trunk route access code.
TARG	1-15	Trunk access restriction group for routes.
OABS	0-9	Outgoing digits to be absorbed.
INST	(0)-999	Digits to be inserted.
CNTL	(NO)YES	Changes to controls or timers.

Table 27: LD 14 - Configure PRI trunks.

Prompt	Response	Description
REQ	NEW	Add new data.
	CHG	Change existing data.
TYPE	TIE	TIE trunk data block.
TN		Terminal number
	l s c u	Format for Large System, Media Gateway 1000B, and CS 1000E system, where l = loop, s = shelf, c = card, u = unit.
DES	PRI	Designator field for trunk.
PDCA	(1)-16	Pad category table number.
PCML		Pulse Code Modulation Law. Enter the appropriate value, based on which companding law is being used on the system.
	A MU	A = A-Law. MU = u-Law.
CUST	xx	Customer number, as defined in LD 15
NCOS	(0)-99	Network Class of Service group.
RTMB		Route number and Member Number
	0-511 1-4000	Range for Large System and CS 1000E system.
MNDN	xxx..x	One-seven digit manual Directory Number.
TGAR	0 - (1) - 31	Trunk Group Access Restriction. The default of 1 automatically blocks direct access.
CLS	aaaa	Class of Service.

DTI implementation

Task summary list

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

Table 28: LD 17 - Configure a DTI interface on the PBX.

Prompt	Response	Description
REQ	CHG	Change existing data.
TYPE	CEQU	Make changes to Common Equipment parameters.
...		
- DTI	xx	DTI loop number

Table 29: LD 16 - Configure a DTI route.

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 , Media Gateway 1000B, and CS 1000E system.
DES	DTI	Designator field for trunk.
TKTP	TIE	TIE trunk type (this is the only type allowed).
...		

Prompt	Response	Description
DTRK	YES	Digital trunk route.
- DGTP	DTI/DTI2	Digital trunk.
ISDN	YES	Integrated Services Digital Network.
...		
ICOG	IAO	Incoming and Outgoing trunks.
SRCH	(LIN) RRB	Search method for outgoing trunk member. Linear Hunt search method. Round Robin search method.
...		
ACOD	xxx..x	One-seven-digit access code for the trunk route.
TARG	0-(1)-31	Trunk Access Restriction Group Number.
SIGO	aaaa	Signaling Arrangement. aaaa = (STD), ESN2, ESN3, ESN5, ETN, EN19.

Table 30: LD 14 - Configure DTI trunks.

Prompt	Response	Description
REQ	NEW	Add new data.
	CHG	Change existing data.
TYPE	TIE	TIE trunk 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.
DES	DTI/DTI2	Designator field for trunk.
PDCA	(1)-16	Pad category table number.
PCML		Pulse Code Modulation Law. Enter the appropriate value, based on which companding law is being used on the system.
	A MU	A = A-Law. MU = u-Law.
CUST	xx	Customer number, as defined in LD 15
NCOS	(0)-99	Network Class of Service group.
RTMB		Route number and Member Number
	0-511 1-4000	Range for Large System, Media Gateway 1000B, and CS 1000E system.
MNDN	xxx..x	One-seven digit manual Directory Number.

Promp t	Response	Description
TGAR	0 - (1) - 31	Trunk Group Access Restriction. The default of 1 automatically blocks direct access.
CLS	aaaa	Class of Service.

Table 31: LD 73 - Define the digital pad tables for each country requirement.

Prompt	Response	Description
REQ	NEW	Add new data.
	CHG	Change existing data.
	OUT	Remove existing data.
	END	Exit LD 73.
	PRT	Print specified data.
TYPE	DTI DTI2	Digital Trunk Interface.
FEAT	PAD	Feature is digital pad.
PDCA	(1)-16	Pad category table number.
TNLS	YES NO	Print TN list (if REQ = PRT).
DFLT	(1)-16	Default table (when REQ = NEW).
For the following prompts, x = Rx code (receive) and y = Tx code (transmit). You can assign receive and transmit dB values to the prompts by entering a code which corresponds to a dB value. **Note:** For North America, use the following values.		
ONP	Rx Tx	On Premise Extension.
ONP	Rx Tx	On Premise Extension. Rx = 8 Tx = -4
DSET	Rx Tx	Meridian Digital Set. Rx = 8 Tx = -4
OPX	Rx Tx	Off Premise Extension. Rx = 8 Tx = -4
DTT	Rx Tx	Digital TIE trunks. Rx = 0 Tx = 0
SDTT	Rx Tx	Digital Satellite Tie trunks. Rx = 6 Tx = 0
DCO	Rx Tx	1.5 Mbit DTI/PRI Digital COT, FEX, WAT, DID trunks. Rx = 3 Tx = 3
VNL	Rx Tx	Via Net Loss Analog TIE trunk. Rx = 3 Tx = 0
SATT	Rx Tx	Analog Satellite TIE trunk. Rx = 6 Tx = 0
ACO	Rx Tx	Analog CO trunk. Rx = 6 Tx = -3

Prompt	Response	Description
PRI	Tx Rx	1.5 Mbit PRI trunk. Applicable when Port Type at far end is PRI for the route (PTYP prompt = PRI in LD 16). Rx = 0 Tx = 0
PRI2	TxRx	2.0 Mbit DI/PRI trunk. Rx = 0 Tx = 0
XUT	Rx Tx	Extended Peripheral Equipment Universal trunk. Rx = 6 Tx = -3
XEM	Rx Tx	Extended Peripheral Equipment E&M trunk. Rx = 6 Tx = 0

Table 32: LD 73 - Configure an ABCD (signaling category table). ABCD tables apply only to DTI/DTI2 interfaces. This step is optional.

Prompt	Response	Comment
REQ	aaa	Request (aaa = CHG, END, NEW, OUT, or PRT).
TYPE	aaa	Type of data block.
FEAT	ABCD	Feature = ABCD.
SICA	2-16	Signaling Category.
TNLS	(NO) YES	Terminal Number List.
DFLT	(1)-16	Default signaling category to be used for Default values.
Prompts for Incoming/Outgoing Calls		
IDLE (S)	ABCD	Idle.
IDLE (R)	ABCD	Idle.
FALT (S)	ABCD	Fault (DTI out-of-service).
FALT (R)	ABCD	Fault (DTI out-of-service).
P RRC (S)	ABCD	Register Recall.
- TIME	10-(100)-630	Time of RRC (S) in milliseconds.
TIME	(0)-1920	Persistence Time required before signal is accepted.
Prompts for Incoming Calls		
SEZ (R)	ABCD	Seize for voice or data calls from a non-SL-1.
E SEZ (R)	ABCD	Seize for voice or data calls from a non-SL-1.
- TIME	16-(56)-1000 16-(296)-1000	
		Minimum and maximum acceptable pulse duration.
SEZD (R)	ABCD	Seize for data calls between SL-1s.
- SEZV (R)	ABCD	Seize for voice calls.

Prompt	Response	Comment
P CALL (R)	ABCD	Signal sent during seize by an incoming CO trunk.
- TIME	1-(2)-15 1-(8)-15	
		Pulse on time, pulse off time.
SEZA (S)	ABCD	Seize Acknowledgment.
- TIME	50-80-90	Time delay prior to sending SEZA.
PRCS (S)	ABCD	PRCS.
WNKS (S)	ABCD	Wink Start.
P WNKS (S)	ABCD	Wink Start.
- TIME	10-(220)-630	Time for P WNKS (S).
P DIGT (R)	ABCD	Decadic pulses.
NRCV (S)	ABCD	Number Received.
P EOSF (S)	ABCD	End of Selection Free.
- TIME	(100)-150	Time for EOSF (S).
- P EOSB (S)	ABCD	End of Selection Busy.
- - TIME	(100)-150	Time for EOSB (S).
P OPC (R)	ABCD	Operator Calling.
- TIME	64-(128)-192	Time of OPCA (R) pulse.
- TIME	16-(96)-1000 16-(160)-1000	
		Minimum and maximum acceptable pulse duration.
- REPT	(1)-5	Number of OPCA (R) pulses.
CONN (S)	ABCD	Connect.
E CON (S)	ABCD	Connect.
- TIME	10-(150) 630	Time of pulse length in 10 ms increments.
CONN (R)	ABCD	Connect.
P BURS (S)	ABCD	Bring Up Receiver for L1 networking.
P BURS (R)	ABCD	Bring Up Receiver for L1 networking.
- TIME	64-(128)-192	Time for BURS (R) pulse.
CLRB (S)	ABCD	Clear Back.
C CLRB (S)	ABCD	Clear Back.
- TIME	10-(600)-2000	Time of pulse length in 10 ms increments.

Prompt	Response	Comment
- P RCT (S)	ABCD	Release Control.
- - TIME	100-(150) 300	Time value is stored in 10 ms increments.
- - P RCOD (S)	ABCD	Release Control Originating party Disconnect.
- - - TIME	150	Timer value in milliseconds is fixed.
P OPRS (R)	ABCD	Operator manual recall.
- TIME	xxxx yyyy	Minimum and maximum time range for OPRS (R).
P NXFR (S)	ABCD	Network Transfer.
P ESNW (S)	ABCD	ESN Wink.
P CAS (S)	ABCD	Centralized Attendant.
CLRF (R)	ABCD	Clear Forward.
- SOS	ABCD	Special Operator Signal.
P BRLS (S)	ABCD	Backward Release.
- TIME	10-(600)-2000	Time of pulse length in 10 ms increments.
P FRLS (R)	ABCD	Forward Release.
- TIME	16-(296)-2000 16-(960)-2000	
		Minimum and maximum acceptable pulse duration.
Prompts for Outgoing Calls		
SEZ (S)	ABCD	Seize for voice or data calls to a non-SL-1.
E SEZ (S)	ABCD	Seize for voice or data calls to a non-SL-1.
- TIME	10-(150)-630	Time of pulse length in 10 ms increments.
SEZD (S)	ABCD	Seize for Data calls.
- SEZV (S)	ABCD	Seize for Voice calls.
SEZA (R)	ABCD	Seize Acknowledgment.
- TIME	xxx	Delay time for the SEZA signal (xxx = 50, 80, 90, (150), or 800).
WNKS (R)	ABCD	Wink Start.
- TIME	20-(140)-500 20-(290)-500	
		Minimum and maximum length of WNKS (R) pulse.
P WNKS (R)	ABCD	Wink Start.
- TIME	16-(136)-504 16-(288)-504	

Prompt	Response	Comment
		Minimum and maximum length of P WNKS (R) pulse.
P EOS (R)	ABCD	End of Selection.
- TIME	(64)-320 64-(256)-320	
		Length of EOS (R) pulse.
CONN (S)	ABCD	Connect.
CONN (R)	ABCD	Connect.
E CONN (R)	ABCD	Connect.
- TIME	16-(56)-1000 16-(296)-1000	
		Time of pulse length in 8 ms increments.
P OPRC (R)	ABCD	Operator Recall for special services.
P BURS (S)	ABCD	Bring Up Receiver for L1 networking.
P BURS (R)	ABCD	Bring Up Receiver for L1 networking.
- TIME	64-(128)-192	Time for BURS (R) pulse.
CLRB (R)	ABCD	Clear Back.
C CLRB (R)	ABCD	Clear Back.
- TIME	16-(296)-2000 16-(960)-2000	
		Time of pulse length in 8 ms increments.
- P RCTL (R)	ABCD	Release Control.
- - TIME	96-(128)-320 96-(256)-320	
		Time stored in 8 ms increments.
P NXFR (R)	ABCD	Network Transfer.
P ESNW (R)	ABCD	ESN Wink.
P CAS (R)	ABCD	Centralized Attendant Service.
CLRF (S)	ABCD	Clear Forward.
- TIME	(0)-800	Time in milliseconds.
- SOS	ABCD	Special Operator Signal.
P FRLS (S)	ABCD	Forward Release.
- TIME	10-(600)-2000	Only prompted for pulsed signals.

Prompt	Response	Comment
P BRLS (R)	ABCD	Backward Release.
- TIME	16-(296)-2000 16-(960)-2000	
		Time of pulse length in 8 ms increments.
C SUPO (S)	ABCD	Complex Supervision to Operator Signal used for KD3 signaling. Note that the input for a must be C.

Chapter 8: 1.5/2.0 Mbit/s Gateway

Contents

This section contains information on the following topics for Avaya Communication Server 1000 (Avaya CS 1000):

Feature description

For digital connectivity, through both Digital Trunk Interface (DTI) and Primary Rate Interface (PRI), North America uses 1.5 Mbit/s carriers programmed as Pulse Code Modulation Companding u-Law. Internationally, in places like Europe, 2.0 Mbit/s carriers programmed as Pulse Code Modulation Companding A-Law are used. To interconnect these two types of switches, gateways are used. A gateway switch performs conversion from A-Law to u-Law, and u-Law to A-Law.

The North America 1.5/2.0 Mbit/s Gateway feature provides support for the 1.5/2.0 Mbit/s gateway functionality in the North American market by introducing software and hardware into North America that is already being deployed in International markets. This feature allows a PBX to act as a gateway between a 1.5 Mbit/s system or Central Office switch programmed as u-Law, and a 2.0 Mbit/s system or Central Office switch programmed as A-Law. The system performs the conversion from A-Law to u-Law, and u-Law to A-Law.

Note:

This feature performs conversion from A-Law to u-Law, and u-Law to A-Law. It does not provide interworking between the T-1 and E-1 protocols. Only those features that are equipped on either side of the gateway will be supported transparently across the gateway.

The 2.0 Mbit (E-1) in North America capability allows an added bandwidth capability to current PBX customers, by introducing software and hardware into North America that is already being deployed in International markets. The Bearer channel (B-Channel) capability for PRI voice and data is increased from 23 to 30, and the A/B-channels for DTI voice and data is increased from 24 to 30. For an nB+D configuration (nB+D is only supported in an MCDN environment, and not supported by QSIG), the B-Channel capability is 480.

The 2.0 Mbit (E-1) in North America capability only applies to "closed campus" environments, supporting only system to system connectivity over the following interface types:

- SL1 (MCDN)
- ETSI (European Telecommunication Standard Institute)
- ETSI QSIG (European Telecommunication Standard Institute QSIG)
- ISIG QSIG (International Standards Organization QSIG)

Back-up D-Channel and nB+D are only supported in an MCDN environment (IFC type of SL1).

The North America 1.5/2.0 Mbit/s Gateway feature supports the following interfaces:

- SL1
- SL100
- D100
- D250
- ISIG
- ESIG
- ISGF
- ESGF

This feature is supported on Large Systems.

2.0 Mbit hardware introduced into North America

To support the North America 1.5/2.0 Mbit/s Gateway feature, the following existing 2.0 Mbit/s cards have been introduced in North America.

For PRI2 on Large Systems:

- NT5D97AA Dual-port DTI2/PRI2 card (in PRI2 mode)
- NTBK51AA/NTBK51CA Downloadable D-Channel Daughterboard

Note:

Please note the vintage requirement of the NTBK51 card. NTBK51AA/NTBK51CA can be used with the NT5D97 or the NT5D12. The NTBK51BA version has only 30+30 pin connectors (instead of 40+30 pins in the AA version). The missing 10 pins in the BA version prohibits the use of port 0 on the NT5D97 or NT5D12 card.

For DTI2 on Large Systems:

NT5D97AA Dual-port DTI2/PRI2 card (in DTI2 mode)

The tables that follow summarize the PRI2 and DTI2 hardware requirements and compatibility.

Note:
When configuring a Clock Controller for large systems, either a Stratum 3 or a Stratum 4 Clock Controller is supported, depending on specific country requirements.

Table 33: Large Systems PRI2 hardware compatibility

PRI2 card	D-Channel Handling	Clock Controller*
NT5D97AA in PRI2 mode	NT6D80 MSDL NT6D11AB/BA/AF external DCHI card NTBK51AA/NTBK51CA Downloadable D-Channel Daughterboard	QPC775 card (Stratum 4)

Table 34: Large Systems DTI2 hardware compatibility

DTI2 card	D-Channel Handling	Clock Controller*
NT5D97AA in DTI2 mode	Not Applicable	QPC775 card (Stratum 4)

Hardware installation

Refer to *Avaya ISDN Primary Rate Interface Installation and Commissioning, NN43001-301* for information on installing the following hardware:

- NT5D97AA Dual-port DTI2/PRI2 card (Large Systems)
- NTBK51AA/NTBK51CA DDCH card (Large Systems)
- Clocking (Large Systems)

2.0 Mbit software introduced into North America

To support the North America 1.5/2.0 Mbit/s Gateway feature, the following existing 2.0 Mbit software packages have been introduced in North America.

- 2.0 Mbit Digital Trunk Interface (DTI2) package 129
- 2.0 Mbit Primary Rate Interface (PRI2) package 154
- International ISDN Supplementary Features (ISDN INTL SUPP) package 161

- International Gateway (GPRI) package 167
- International Primary Rate Access (IPRA) package 202

The following packages are required as prerequisites:

- Digit Display (DDSP) package 19
- Digital Trunk Interface (DTI) package 75
- Integrated Services Digital Network (ISDN) package 145

Conversion scenarios

Scenario 1 - Gateway u-Law system connected to A-Law system/CO over an A-Law 1.5 Mbit/s connection

A North America (1.5 Mbit/s) PBX, programmed as u-Law, is connected to an International (2.0 Mbit/s) PBX, programmed as A-Law. The connection is over an A-Law 1.5 Mbit/s DTI/PRI link.

In this scenario, the conversion from A-Law to u-Law, and u-Law to A-Law, is done at the system programmed as u-Law. It is this PBX that acts as the 1.5/2.0 Mbit/s gateway.

Scenario 2 - Gateway u-Law system connected to A-Law Meridiansystem/CO over an A-Law 2.0 Mbit/s connection

A North America (1.5 Mbit/s) PBX, programmed as u-Law, is connected to an International (2.0 Mbit/s) PBX, programmed as A-Law. The connection is over an A-Law 2.0 Mbit/s DTI/PRI link.

In this scenario, the conversion from A-Law to u-Law, and u-Law to A-Law, is done at the system programmed as u-Law. It is this PBX that acts as the 1.5/2.0 Mbit/s gateway.

Scenario 3 - Gateway A-Law system connected to u-Law system/CO over an u-Law 1.5 Mbit/s connection

An International (2.0 Mbit/s) PBX, programmed as A-Law, is connected to a North America (1.5 Mbit/s) PBX, programmed as u-Law. The connection is over a u-Law 1.5 Mbit/s DTI/PRI link.

In this scenario, the conversion from A-Law to u-Law, and u-Law to A-Law, is done at the system programmed as A-Law. It is this PBX that acts as the 1.5/2.0 Mbit/s gateway.

Scenario 4 - Gateway A-Law system connected to u-Law system/CO over an u-Law 2.0 Mbit/s connection

An International (2.0 Mbit/s) PBX, programmed as A-Law, is connected to a North America (1.5 Mbit/s) PBX, programmed as u-Law. The connection is over a u-Law 2.0 Mbit/s DTI/PRI link.

In this scenario, the conversion from A-Law to u-Law, and u-Law to A-Law, is done at the system programmed as A-Law. It is this PBX that acts as the 1.5/2.0 Mbit/s gateway.

Operating parameters

The 1.5 Mbit/s and 2.0 Mbit/s Digital Trunk Interface (DTI2) cards do not support auto-recovery capability. In order to support auto-recovery, a dual-port DTI card with pack NT5D12AH must be used.

Feature interactions

There are no interactions associated with this feature.

Note:

ISDN networking features will function transparently across the gateway if the dialing plans are consistent on both sides of the gateway.

Feature packaging

This feature is included in base Meridian 1 PBX 61CSystem Software.

Feature implementation

This section contains procedures on how to configure DTI2 or PRI2 on a system, as required for the implementation of the North America 1.5/2.0 Mbit/s Gateway feature functionality.

Task summary list

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

Table 35:

1 Table 36: LD 15 - Configure the Pulse Code Modulation Conversion value on the PBX. on page 85

2 Table 37: LD 17 - Configure a PRI2 interface on the PBX. on page 85

3 Table 38: LD 17 - Configure the Pulse Code Modulation Companding Law. on page 86

Perform the following additional steps when using a Meridian M2317 phone on a system switch configured as A-Law:

4 Table 39: LD 17 - Configure the CODEC coding law for the phone. on page 86

5 Table 40: LD 97 - Configure the Multifrequency Sender parameters. on page 86

6 Table 41: LD 97 - Configure the system parameters for Intelligent Peripheral Equipment. on page 86

Initialize the system switch to download all parameters to the line cards. This initialization is only performed once.

For the system programmed as U-Law use the default values:

- LD 17, when configuring the CODEC coding law, use CODE = 0
- LD 97, when configuring the Multifrequency Sender parameters, use DTMF = 14
- LD 97, when configuring the system parameters for Intelligent Peripheral Equipment, use INTN = NO

7 Table 42: LD 17 - Configure the Primary D-channel. on page 87

8 Table 43: LD 17 - Configure a Back-up D-channel. on page 88

This step is optional, and performed only if a Back-up D-Channel is required. Note that Back-up D-Channel is only supported in an MCDN environment (IFC must be set to SL1).

9 Table 44: LD 17 - Configure nB+D. on page 89

This step is optional and requires the International nB+d (INBD) package 255 to be equipped for the following interfaces only: D70, JAPN, HKNG, TCNZ, and MSIA. Also, note that nB+D is only supported in an MCDN environment (IFC must be set to SL1).

10 Table 45: LD 73 - Define the digital pad tables by country requirement. on page 90

11 Table 46: LD 16 - Configure a PRI2 route. on page 92

12 Table 47: LD 14 - Configure PRI2 trunks. on page 93

Perform the following steps for the DTI2 implementation:

13 Table 48: LD 15 - Configure the Pulse Code Modulation Conversion value on the PBX. on page 93

14 Table 49: LD 17 - Configure a DTI2 interface on the PBX. on page 94

15 Table 50: LD 16 - Configure a DTI2 route. on page 94

16 Table 51: LD 14 - Configure DTI2 trunks. on page 95

17 Table 52: LD 73 - Define the digital pad tables for each country requirement. on page 96

18 Table 53: LD 73 - Configure an ABCD (signaling category table). on page 97

ABCD tables apply only to DTI/DTI2 interfaces. This step is optional.

PRI2 implementation

Table 36: LD 15 - Configure the Pulse Code Modulation Conversion value on the PBX.

Prompt	Response	Description
REQ	CHG	Change existing data.
TYPE	NET	Make changes to networking data.
CUST		Customer number
	0-99	Range for Large System, Media Gateway 1000B, and CS 1000E system.
...		
ISDN	YES	Integrated Services Digital Network.
...		
- RCNT	0-(5)	Redirection Count for ISDN calls.
- - PCMC	0-(15)-31	The number of Pulse Code Modulation Conversions allowed, from u-Law to A-Law or A-Law to u-Law, in a network connection.

Table 37: LD 17 - Configure a PRI2 interface on the PBX.

Prompt	Response	Description
REQ	CHG	Change existing data.
TYPE	CEQU	Make changes to Common Equipment parameters.

Prompt	Response	Description
...		
- PRI2	0-255	PRI2 loop number.

Table 38: LD 17 - Configure the Pulse Code Modulation Companding Law.

Prompt	Response	Description
REQ	CHG	Change existing data.
TYPE	PARM	Make changes to system parameters.
...		
- PCML		Pulse Code Modulation Law. Enter the appropriate value, based on which companding law is being used.
	(MU) A	MU = u-Law. A = A-Law.

Table 39: LD 17 - Configure the CODEC coding law for the phone.

Prompt	Response	Description
REQ	CHG	Change existing data.
TYPE	ATRN	ARIES (M2317) Transmission.
- CODE	2	CODEC coding law.

Table 40: LD 97 - Configure the Multifrequency Sender parameters.

Prompt	Response	Description
REQ	CHG	Change existing data.
TYPE	XCTP	Conference/TDS/Multifrequency Sender parameters.
...		
DTMF	138	Dual Tone Multifrequency.

Table 41: LD 97 - Configure the system parameters for Intelligent Peripheral Equipment.

Prompt	Response	Description
REQ	CHG	Change existing data.
TYPE	SYSP	System parameters for Intelligent Peripheral Equipment.
INTN	YES	International companding law.
...		

Prompt	Response	Description
FDLC	ALL	Fast Download parameters to all line cards.

Table 42: LD 17 - Configure the Primary 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 logical port 0-63 for Large Systems.
- CTYP	MSDL	Card type where: MSDL = NT6D80 Multi-purpose Serial Data Link or the NTBK51AA/NTBK51CA Downloadable D-Channel Daughterboard for Large Systems.
- GRP	0-7	Network group number for Meridian 1 PBX 81C.
- DNUM	0-15	Device number for I/O ports. All ports on the MSDL card share the same DNUM. The MSDL card address settings must match the DNUM value.
- PORT	0-3	Port number on the NT6D80 MSDL card, if the MSDL is used for D-Channel handling on Large Systems.
	0-1	Port number of the NTBK51AA/NTBK51CA, if the NTBK51AA/NTBK51CA is used for D-Channel handling on Large Systems. Port 0 of the NTBK51AA/NTBK51CA can only be defined to work with Loop 0 of the NT5D97AA DDP2 card, and Port 1 of the NTBK51AA/NTBK51CA can only be defined to work with Loop 1 of the NT5D97AA. This relationship must be reflected in the DCHL prompt, which follows later (either 0 or 1 must be entered when specifying the loop number used by the D-Channel).
- DES	aaa...a	Designator. DES is used to identify the link and can be up to 16 alphanumeric characters: 0-9, and upper case A-Z. Characters "*" and "#" are not allowed.
- USR	PRI	This D-channel is used for Primary Rate only.
- IFC		Interface type.
	SL1 SL100 D100 D250 ETSI ESIG ISIG	MCDN SL100 DMS-100 DMS-250 European Telecommunication Standard Institute ETSI QSIG International Standards Organization QSIG
- - DCHL	0-255	PRI2 loop number for the D-channel, for large systems.

Prompt	Response	Description
		If the NTBK51AA/NTBK51CA is used for D-Channel handling, only loop 0 or 1 can be configured.
- PRI2		Secondary PRI2 loops and sequence for nB+D configuration.
	0-255 2-15 1-9 (0)-15 11-19 21-29 31-39 41-49	For Large Systems.
		Enter 0, the default.
- OTBF	0-(32)-127	Output Request Buffers. Enter 32, the default.
- SIDE	(USR) NET	The system is network side.
- RLS	xx	Software release of the far end switch. For system to system connectivity, both switches must be running Release 24 software.
- RCAP	aaa	Remote capabilities. Enter <cr> when finished entering values.
- OVLR	(NO) YES	Allow Overlap Receiving. Enter NO, the default.
- OVLS	(NO) YES	Allow Overlap Sending. Enter NO, the default.
- MBGA	(NO) YES	Allow Multilocation Business Group. Enter NO, the default.
- NASA	(NO) YES	Allow Network Attendant Service. Enter NO, the default.
- TIMR	(NO) YES	Change Protocol Timer values. Enter NO, the default.
- LAPD	(NO) YES	Change Link Access Protocol for D-Channel parameters. Enter NO, the default.

Table 43: LD 17 - Configure a Back-up D-channel.

Prompt	Response	Description
REQ	CHG	Change existing data.
TYPE	ADAN	Action device and number.
- ADAN	NEW BDCH xx	Add a Back-up D-channel on logical port 0-63 for Large Systems.
- PDCH	0-63	0-63 = Primary D-Channel associated with the Backup D-Channel for Large Systems.

Prompt	Response	Description
- CTYP	MSDL	Card type where: MSDL = NT6D80 Multi-purpose Serial Data Link or the NTBK51AA/NTBK51CA Downloadable D-Channel Daughterboard for Large Systems.
- GRP	0-7	Network group number for the Meridian 1 PBX 81C.
- DNUM	0-15	Device number for I/O ports. All ports on the MSDL card share the same DNUM. The MSDL card address settings must match the DNUM value.
- PORT	0-3	Port number on the NT6D80 MSDL card, if the MSDL is used for D-Channel handling on large systems.
	0-1	Port number of the NTBK51AA/NTBK51CA, if the NTBK51AA/NTBK51CA is used for D-Channel handling on large systems. Port 0 of the NTBK51AA/NTBK51CA can only be defined to work with Loop 0 of the NT5D97AA DDP2 card, and Port 1 of the NTBK51AA/NTBK51CA can only be defined to work with Loop 1 of the NT5D97AA. This relationship must be reflected in the DCHL prompt, which follows later (either 0 or 1 must be entered when specifying the loop number used by the D-Channel).
- RCVP	YES	Auto-recovery to primary D-Channel.
- BCHL	0-255	PRI2 loop number for the Back-up D-channel, for large systems. If the NTBK51AA/NTBK51CA is used for D-Channel handling, only loop 0 or 1 can be configured.

Table 44: LD 17 - Configure nB+D.

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 for Large Systems.
- CTYP	MSDL	Card type where: MSDL = NT6D80 Multi-purpose Serial Data Link, or the NTBK51AA/NTBK51CA Downloadable D-Channel Daughterboard, for large systems.
- GRP	0-7	Network group number for Meridian 1 PBX 81C.
- DNUM	0-15	Device number for I/O ports. All ports on the MSDL card share the same DNUM. The MSDL card address settings must match the DNUM value.

Prompt	Response	Description
- PORT	0-3	Port number on the NT6D80 MSDL card, if the MSDL is used for D-Channel handling on large systems.
	0-1	Port number of the NTBK51AA/NTBK51CA, if the NTBK51AA/NTBK51CA is used for D-Channel handling on large systems. Port 0 of the NTBK51AA/NTBK51CA can only be defined to work with Loop 0 of the NT5D97AA DDP2 card, and Port 1 of the NTBK51AA/NTBK51CA can only be defined to work with Loop 1 of the NT5D97AA. This relationship must be reflected in the DCHL prompt, which follows later (either 0 or 1 must be entered when specifying the loop number used by the D-Channel).
- DES	aaa...a	Designator. DES is used to identify the link and can be up to 16 alphanumeric characters: 0-9, and upper case A-Z. Characters "*" and "#" are not allowed.
- USR	PRI	This D-channel is used for Primary Rate only.
- IFC	SL1	MCDN interface (nB+D is only supported in an MCDN environment).
- - DCHL	LOOP ID 0-255 (0)-15	Primary PRI2 loop number and interface identifier, for the D-channel. When INBD package 255 is enabled values for both the PRI loop number (LOOP) and the D-channel interface identifier (ID) must be entered. For Large Systems.
	LOOP ID 1-9 (0)-15 0-255 2-15 1-9 (0)-15 11-19 21-29 31-39 41-49	Primary PRI2 loop number and interface identifier for the D-channel. When INBD package 255 is enabled values for both the PRI loop number (LOOP) and the D-channel interface identifier (ID) must be entered.
- PRI	0-255 1-126 (LOOP) (ID)	Secondary PRI loop number and interface identifier, for nB+D (prompted if INBD package 255 is enabled). The values entered must be different than those entered for the loop number and interface identifier at the DCHL prompt. For large systems. The PRI prompt is generated until <CR> is entered.
PRI	<CR>	End configuration.

Table 45: LD 73 - Define the digital pad tables by country requirement.

Prompt	Response	Description
REQ	NEW	Add new data.
	CHG	Change existing data.

Prompt	Response	Description
	OUT	Remove existing data.
	END	Exit LD 73.
	PRT	Print specified data.
TYPE	PRI2	2.0 Mbit Primary Rate Interface.
FEAT	PAD	Feature is digital pad.
PDCA	(1)-16	Pad category table number.
TNLS	YES NO	Print TN list (if REQ = PRT).
DFLT	(1)-16	Default table (when REQ = NEW).
For the following prompts, x = Rx code (receive) and y = Tx code (transmit). You can assign receive and transmit dB values to the prompts by entering a code which corresponds to a dB value. **Note:** For North America, use the following values.		
ONP	Rx Tx	On Premise Extension. Rx = 8 Tx = -4
DSET	Rx Tx	Meridian Digital Set. Rx = 8 Tx = -4
OPX	Rx Tx	Off Premise Extension. Rx = 8 Tx = -4
DTT	Rx Tx	Digital TIE trunks. Rx = 0 Tx = 0
SDTT	Rx Tx	Digital Satellite Tie trunks. Rx = 6 Tx = 0
DCO	Rx Tx	1.5 Mbit DTI/PRI Digital COT, FEX, WAT, DID trunks. Rx = 3 Tx = 3
DTO	X Y	1.5 Mbit/s DTM/PRI Digital TOLL Office trunks
VNL	Rx Tx	Via Net Loss Analog TIE trunk. Rx = 3 Tx = 0
SATT	Rx Tx	Analog Satellite TIE trunk. Rx = 6 Tx = 0
ACO	Rx Tx	Analog CO trunk. Rx = 6 Tx = -3
ATO	X Y	Analog TOLL Office trunks
PRI	Tx Rx	1.5 Mbit PRI trunk. Applicable when Port Type at far end is PRI for the route (PTYP prompt = PRI in LD 16). Rx = 0 Tx = 0
PRI2	Tx Rx	2.0 Mbit DI/PRI trunk. Rx = 0 Tx = 0
XUT	Rx Tx	Extended Peripheral Equipment Universal trunk. Rx = 6 Tx = -3
XEM	Rx Tx	Extended Peripheral Equipment E&M trunk. Rx = 6 Tx = 0

Table 46: LD 16 - Configure a PRI2 route.

Prompt	Response	Description
REQ	NEW	Add new data.
	CHG	Change existing data.
TYPE	RDB	Route Data Block.
CUST		Customer number
	0-99	Range for Large System, Media Gateway 1000B, and CS 1000E system.
ROUT		Route number
	0-511	Range for Large System, Media Gateway 1000B, and CS 1000E system.
DES	PRI2	Designator field for trunk.
TKTP	TIE	TIE trunk type (this is the only type allowed).
...		
DTRK	YES	Digital trunk route.
- DGTP	PRI2	2.0 Mbit PRI digital trunk.
ISDN	YES	Integrated Services Digital Network.
- MODE	PRA	ISDN PRI route.
- IFC	SL1	SL-1 interface (system to system connectivity).
- - PNI	(0)-32700	Private Network identifier.
...		
- NCNA	YES	Network Calling Name allowed.
- NCRD	YES	Network Call Redirection allowed.
...		
ICOG	IAO	Incoming and Outgoing trunks.
...		
SRCH		Search method for outgoing trunk member.
	(LIN) RRB	Linear Hunt search method. Round Robin search method.
...		
ACOD	xxx..x	One-seven-digit access code for the trunk route.
...		
SIGO	aaaa	Signaling Arrangement. aaaa = (STD), ESN2, ESN3, ESN5, ETN, EN19.

Table 47: LD 14 - Configure PRI2 trunks.

Prompt	Response	Description
REQ	NEW	Add new data.
	CHG	Change existing data.
TYPE	TIE	TIE trunk data block.
TN		Loop and channel for digital trunks.
	l ch	Loop and channel for PRI2 trunks, where: l = previously defined PRI2 loops and ch = channel 1-30. Format for Large System, Media Gateway 1000B, and CS 1000E system.
DES	PRI2	Designator field for trunk.
PDCA	(1)-16	Pad category table number.
PCML		Pulse Code Modulation Law. Enter the appropriate value, based on which companding law is being used.
	A MU	A = A-Law. MU = u-Law.
CUST		Customer number
	0-99	Range for Large System and CS 1000E system.
NCOS	(0)-99	Network Class of Service group.
RTMB		Route number and Member Number
	0-511 1-4000	Range for Large System, Media Gateway 1000B, and CS 1000E system.
MNDN	xxx..x	One-seven digit manual Directory Number.
TGAR	0 - (1) - 31	Trunk Group Access Restriction. The default of 1 automatically blocks direct access.
CLS	aaaa	Class of Service.

DTI2 implementation

Table 48: LD 15 - Configure the Pulse Code Modulation Conversion value on the PBX.

Prompt	Response	Description
REQ	CHG	Change existing data.
TYPE	NET	Make changes to networking data.

Prompt	Response	Description
CUST		Customer number
	0-99	Range for Large System, Media Gateway 1000B, and CS 1000E system.
...		
ISDN	YES	Integrated Services Digital Network.
...		
- RCNT	0-(5)	Redirection Count for ISDN calls.
-PSTN	YES	
- - PCMC	0-(15)-31	The number of Pulse Code Modulation Conversions allowed, from u-Law to A-Law or A-Law to u-Law, in a network connection.

Table 49: LD 17 - Configure a DTI2 interface on the PBX.

Prompt	Response	Description
REQ	CHG	Change existing data.
TYPE	CEQU	Make changes to Common Equipment parameters.
...		
- DTI2	0-255	DTI2 loop number.

Table 50: LD 16 - Configure a DTI2 route.

Prompt	Response	Description
REQ	NEW	Add new data.
TYPE	RDB	Route Data Block.
CUST		Customer number
	0-99	Range for Large System, Media Gateway 1000B, and CS 1000E system.
ROUT		Route number
	0-511	Range for Large System, Media Gateway 1000B, and CS 1000E system.
DES	DTI2	Designator field for trunk.
TKTP	TIE	TIE trunk type (this is the only type allowed).
...		
DTRK	YES	Digital trunk route.
- DGTP	DTI2	2.0 Mbit digital trunk.

Prompt	Response	Description
ISDN	YES	Integrated Services Digital Network.
...		
ICOG	IAO	Incoming and Outgoing trunks.
SRCH		Search method for outgoing trunk member.
	(LIN) RRB	Linear Hunt search method. Round Robin search method.
...		
ACOD	xxx..x	One-seven-digit access code for the trunk route.
...		
SIGO	aaaa	Signaling Arrangement. aaaa = (STD), ESN2, ESN3, ESN5, ETN, EN19.

Table 51: LD 14 - Configure DTI2 trunks.

Prompt	Response	Description
REQ	NEW	Add new data.
	CHG	Change existing data.
TYPE	TIE	TIE trunk data block.
TN		Loop and channel for digital trunks.
	l ch	Loop and channel for PRI2 trunks, where: l = previously defined PRI2 loops and ch = channel 1-30. Format for Large System, Media Gateway 1000B, and CS 1000E system.
DES	DTI2	Designator field for trunk.
PDCA	(1)-16	Pad category table number.
PCML		Pulse Code Modulation Law. Enter the appropriate value, based on which companding law is being used.
		A = A-Law.
	A MU	MU = u-Law.
CUST		Customer number
	0-99	Range for Large System, Media Gateway 1000B, and CS 1000E system.
NCOS	(0)-99	Network Class of Service group.
RTMB		Route number and Member Number

Prompt	Response	Description
	0-511 1-4000	Range for Large System, Media Gateway 1000B, and CS 1000E system.
MNDN	xxx..x	One-seven digit manual Directory Number.
TGAR	0 - (1) - 31	Trunk Group Access Restriction. The default of 1 automatically blocks direct access.
CLS	aaaa	Class of Service.

Table 52: LD 73 - Define the digital pad tables for each country requirement.

Prompt	Response	Description
REQ	NEW	Add new data.
	CHG	Change existing data.
	OUT	Remove existing data.
	END	Exit LD 73.
	PRT	Print specified data.
TYPE	DTI2	2.0 Mbit Digital Trunk Interface.
FEAT	PAD	Feature is digital pad.
PDCA	(1)-16	Pad category table number.
TNLS	YES NO	Print TN list (if REQ = PRT).
DFLT	(1)-16	Default table (when REQ = NEW).
For the following prompts, x = Rx code (receive) and y = Tx code (transmit). You can assign, receive and transmit dB values to the prompts by entering a code which corresponds to a dB value. **Note:** For North America, use the following values.		
ONP	Rx Tx	On Premise Extension.
ONP	Rx Tx	On Premise Extension. Rx = 8 Tx = -4
DSET	Rx Tx	Meridian Digital Set. Rx = 8 Tx = -4
OPX	Rx Tx	Off Premise Extension. Rx = 8 Tx = -4
DTT	Rx Tx	Digital TIE trunks. Rx = 0 Tx = 0
SDTT	Rx Tx	Digital Satellite Tie trunks. Rx = 6 Tx = 0
DCO	Rx Tx	1.5 Mbit DTI/PRI Digital COT, FEX, WAT, DID trunks. Rx = 3 Tx = 3
VNL	Rx Tx	Via Net Loss Analog TIE trunk. Rx = 3 Tx = 0

Prompt	Response	Description
SATT	Rx Tx	Analog Satellite TIE trunk. Rx = 6 Tx = 0
ACO	Rx Tx	Analog CO trunk. Rx = 6 Tx = -3
PRI	Tx Rx	1.5 Mbit PRI trunk. Applicable when Port Type at far end is PRI for the route (PTYP prompt = PRI in LD 16). Rx = 0 Tx = 0
PRI2	Tx Rx	2.0 Mbit DI/PRI trunk. Rx = 0 Tx = 0
XUT	Rx Tx	Extended Peripheral Equipment Universal trunk. Rx = 6 Tx = -3
XEM	Rx Tx	Extended Peripheral Equipment E&M trunk. Rx = 6 Tx = 0

Before configuring the ABCD tables, refer to the section Signaling category (ABCD tables) assignment and modification on page 103, which explains ABCD table assignment and modification.

Refer to the following tables when configuring the ABCD table values:

- Table 54: Default values for Signaling Categories 1 and 16 (In/Out Calls) on page 101.
- Table 55: Default values for Signaling Categories 1 and 16 (Incoming Calls) on page 101.
- Table 56: Default values for Signaling Categories 1 and 16 (Outgoing Calls) on page 102.

Table 53: LD 73 - Configure an ABCD (signaling category table).

Prompt	Response	Description
REQ	aaa	Request (aaa = CHG, END, NEW, OUT, or PRT).
TYPE	aaa	Type of data block.
FEAT	ABCD	Feature = ABCD.
SICA	2-16	Signaling Category.
TNLS	(NO) YES	Terminal Number List.
DFLT	(1)-16	Default signaling category to be used for Default values.
Prompts for Incoming/Outgoing Calls		
IDLE (S)	abcd	Idle.
IDLE (R)	abcd	Idle.
FALT (S)	abcd	Fault (DTI out-of-service).
FALT (R)	abcd	Fault (DTI out-of-service).
P RRC (S)	abcd	Register Recall.

Prompt	Response	Description
- TIME	10 -(100)-630	Time of RRC (S) in milliseconds.
TIME	(0)-1920	Persistence Time required before signal is accepted.
Prompts for Incoming Calls		
E SEZ (R)	abcd	Seize for voice or data calls from a non-SL-1.
- TIME	16-(56)-1000 16-(296)-1000.	
		Minimum and maximum acceptable pulse duration.
SEZD (R)	abcd	Seize for data calls between SL-1s.
- SEZV (R)	abcd	Seize for voice calls.
P CALL (R)	abcd	Signal sent during seize by an incoming CO trunk.
- TIME	1-(2)-15 1-(8)-15	
		Pulse on time, pulse off time.
SEZA (S)	abcd	Seize Acknowledgment.
- TIME	50-80-90	Time delay prior to sending SEZA.
PRCS (S)	abcd	PRCS.
WNKS (S)	abcd	Wink Start.
P WNKS (S)	abcd	Wink Start.
- TIME	10-(220)-630	Time for P WNKS (S).
P DIGT (R)	abcd	Decadic pulses.
NRCV (S)	abcd	Number Received.
P EOSF (S)	abcd	End of Selection Free.
- TIME	(100)-150	Time for EOSF (S).
- P EOSB (S)	abcd	End of Selection Busy.
- - TIME	(100)-150	Time for EOSB (S).
P OPC (R)	abcd	Operator Calling.
- TIME	64-(128)-192	Time of OPCA (R) pulse.
- TIME	16-(96)-1000 16-(160)-1000	
		Minimum and maximum acceptable pulse duration.
- REPT	(1)-5	Number of OPCA (R) pulses.
CONN (S)	abcd	Connect.
E CON (S)	abcd	Connect.

Prompt	Response	Description
- TIME	10-(150) 630	Time of pulse length in 10 ms increments.
CONN (R)	abcd	Connect.
P BURS (S)	abcd	Bring Up Receiver for L1 networking.
P BURS (R)	abcd	Bring Up Receiver for L1 networking.
- TIME	64-(128)-192	Time for BURS (R) pulse.
CLRB (S)	abcd	Clear Back.
C CLRB (S)	abcd	Clear Back.
- TIME	10-(600)-2000	Time of pulse length in 10 ms increments.
- P RCT (S)	abcd	Release Control.
- - TIME	100-(150) 300	Time value is stored in 10 ms increments.
- - P RCOD (S)	abcd	Release Control Originating party Disconnect.
- - - TIME	150	Timer value in milliseconds is fixed.
P OPRS (R)	abcd	Operator manual recall.
- TIME	xxxx yyyy	Minimum and maximum time range for OPRS (R).
P NXFR (S)	abcd	Network Transfer.
P ESNW (S)	abcd	ESN Wink.
P CAS (S)	abcd	Centralized Attendant.
CLRF (R)	abcd	Clear Forward.
- SOS	abcd	Special Operator Signal.
P BRLS (S)	abcd	Backward Release.
- TIME	10-(600)-2000	Time of pulse length in 10 ms increments.
P FRLS (R)	abcd	Forward Release.
- TIME	16-(296)-2000 16-(960)-2000	
		Minimum and maximum acceptable pulse duration.
Prompts for Outgoing Calls		
E SEZ (S)	abcd	Seize for voice or data calls to a non-SL-1.
- TIME	10-(150)-630	Time of pulse length in 10 ms increments.
SEZD (S)	abcd	Seize for Data calls.
- SEZV (S)	abcd	Seize for Voice calls.
SEZA (R)	abcd	Seize Acknowledgment.

Prompt	Response	Description
- TIME	xxx	Delay time for the SEZA signal (xxx = 50, 80, 90, (150), or 800).
WNKS (R)	abcd	Wink Start.
- TIME	20-(140)-500 20-(290)-500	
		Minimum and maximum length of WNKS (R) pulse.
P WNKS (R)	abcd	Wink Start.
- TIME	16-(136)-504 16-(288)-504	
		Minimum and maximum length of P WNKS (R) pulse.
P EOS (R)	abcd	End of Selection.
- TIME	(64)-320 64-(256)-320	
		Length of EOS (R) pulse.
CONN (S)	abcd	Connect.
CONN (R)	abcd	Connect.
E CONN (R)	abcd	Connect.
- TIME	16-(56)-1000 16-(296)-1000	
		Time of pulse length in 8 ms increments.
P OPRC (R)	abcd	Operator Recall for special services.
P BURS (S)	abcd	Bring Up Receiver for L1 networking.
P BURS (R)	abcd	Bring Up Receiver for L1 networking.
- TIME	64-(128)-192	Time for BURS (R) pulse.
CLRB (R)	abcd	Clear Back.
C CLRB (R)	abcd	Clear Back.
- TIME	16-(296)-2000 16-(960)-2000	
		Time of pulse length in 8 ms increments.
- P RCTL (R)	abcd	Release Control.
- - TIME	96-(128)-320 96-(256)-320	
		Time stored in 8 ms increments.
P NXFR (R)	abcd	Network Transfer.

Prompt	Response	Description
P ESNW (R)	abcd	ESN Wink.
P CAS (R)	abcd	Centralized Attendant Service.
CLRF (S)	abcd	Clear Forward.
- TIME	(0)-800	Time in milliseconds.
- SOS	abcd	Special Operator Signal.
P FRLS (S)	abcd	Forward Release.
- TIME	10-(600)-2000	Only prompted for pulsed signals.
P BRLS (R)	abcd	Backward Release.
- TIME	16-(296)-2000 16-(960)-2000	
		Time of pulse length in 8 ms increments.
C SUPO (S)	abcd	Complex Supervision to Operator Signal used for KD3 signaling. Note that the input for a must be C.

Table 54: Default values for Signaling Categories 1 and 16 (In/Out Calls)

In/Out Calls	SICA 1	SICA 16
IDLE (S)	1001	1101
IDLE (R)	1001	1101
FALT (S)	1101	0101
FALT (R)	1101	0101
TIME	0	0
P RRC (S)	UNUSED	UNUSED

Table 55: Default values for Signaling Categories 1 and 16 (Incoming Calls)

Incoming Calls	SICA 1	SICA 16	Incoming Calls	SICA 1	SICA 16
E SEZ (R)	0001	0101	CONN (R)	0001	0101
SEZD (R)	UNUSED	UNUSED	P BURS (S)	UNUSED	UNUSED
SEZV (R)	UNUSED	UNUSED	P BURS (R)	UNUSED	UNUSED
P CALL (R)	UNUSED	UNUSED	C CLRB (S)	1101	1101
SEZA (S)	1101	UNUSED	P RCTL (S)	UNUSED	UNUSED
TIME	150		P RCOD (S)	UNUSED	UNUSED
PRCS (S)	UNUSED	UNUSED	P OPRS (R)	UNUSED	UNUSED

Incoming Calls	SICA 1	SICA 16	Incoming Calls	SICA 1	SICA 16
P WNKS (S)	UNUSED	PXXX	P NXFR (S)	UNUSED	UNUSED
TIME		220	P ESNW (S)	UNUSED	UNUSED
P DIGT (R)	UNUSED	PXXX	P CAS (S)	UNUSED	UNUSED
NRCV (S)	UNUSED	UNUSED	CLRF (R)	UNUSED	UNUSED
P EOSF (S)	UNUSED	UNUSED	SOS (R)	UNUSED	UNUSED
P EOSB (S)	UNUSED	UNUSED	P BRLS (S)	UNUSED	UNUSED
P OPCA (R)	UNUSED	UNUSED	P FRLS (R)	UNUSED	UNUSED
E CONN (S)	0101	0101			

Table 56: Default values for Signaling Categories 1 and 16 (Outgoing Calls)

Outgoing Calls	SICA 1	SICA 16
E SEZ (S)	0001	0101
SEZD (S)	UNUSED	UNUSED
SEZV (S)	UNUSED	UNUSED
SEZA (R)	1101	UNUSED
P WNKS (R)	UNUSED	PXXX
TIME		136 288
P EOS (R)	UNUSED	UNUSED
CONN (S)	0001	0101
E CONN (R)	0101	0101
P OPRC (R)	UNUSED	UNUSED
P BURS (S)	UNUSED	UNUSED
P BURS (R)	UNUSED	UNUSED
C CLRB (R)	1101	1101
P RCTL (R)	UNUSED	UNUSED
P NXFR (R)	UNUSED	UNUSED
P ESNW (R)	UNUSED	UNUSED
P CAS (R)	UNUSED	UNUSED
CLRF (S)	UNUSED	UNUSED
SOS (R)	UNUSED	UNUSED
P FRLS (S)	UNUSED	UNUSED

Outgoing Calls	SICA 1	SICA 16
P BRLS (R)	UNUSED	UNUSED

Signaling category (ABCD tables) assignment and modification

ABCD responses

Prompts which show the response abcd, such as IDLE (S), require a four field response to indicate the status of four bits: a, b, c and d. The abcd response represents a trunk supervisory message. The bit states within the message are determined by using the appropriate input. Allowable inputs for each bit are: 0, 1, C, P, U, X, N. These input options are explained as follows:

- 0 - Bit is a steady state 0 (LOW) e.g. 0000 bits abcd are all steady state 0.
- 1 - Bit is a steady state 1 (HIGH) e.g. 0101 bits b and d are steady state 1 while bits a and c are steady state 0.
- C - Bit is pulsed and present continuously (Continuous pulsing of two or more bits is not allowed.).
    - "C" can only be entered for signals that have "C" in front of them when the signal is prompted; the signals are: "C CLRB (S), C CLRB (R) and C SUPO (S) UNUSED"
    - "C" cannot be mixed with 0 or 1 or P in the ABCD pattern. Therefore, the entry must look like CXXX, XCXX, etc.
    - "C" can only be entered once in the ABCD pattern
- C cannot be entered for the CLRB (R) or CLRB (S) prompts if the pulsed E&M package (232) PEMD is equipped - Bit is pulsed. e.g. PC10 bit a is pulsed, bit b is pulsed and sent continuously, bit c is steady state 1 and bit d is steady state 0.
- U - Bit is a don't-care bit (for received signals only) e.g. U10U bits a and d are don't-care bits, bit b is steady state 1 and bit c is steady state 0.
- X - Bit is not to be changed (used in conjunction with Pulsed or Continuously pulsed bit) e.g. XPXX bits a, c and d are unchanged, bit b is set to steady state 1 and bit c is set to steady state 0.

Another input to the signal name prompt is allowed. The other allowable input is: N - The signal is not required.

Signs that the signal is pulsing, pulsed or steady

The signal type is identified by a single character followed by a blank space preceding the signal name. For example, the prompt E SEZ(R) indicates that the Seize signal can be either Pulsed or steady state. The signal type identifiers are:

- C - Continuous Pulsing, Pulsed or steady state
- E - Pulsed or steady state
- P - Pulsed (single pulse unless otherwise indicated)
- No preceding character indicates the signal is steady state only

Pulsed signals output the TIME prompt. This prompt is described for each of the signals that can prompt it.

How to determine signal direction

The direction of the signal is indicated by a single character in brackets at the end of the signal name:

- (R) - Indicates that the signal is to be received by the switch
- (S) - Indicates that the signal is to be sent by the switch

For example, E SEZ (R) indicates that the Seize signal can be either Pulsed or steady state and that the signal is to be received by the switch.

How to tell if the prompt is incoming or outgoing, or both

ABCD prompts correspond to incoming calls, outgoing calls or both incoming and outgoing calls. Prompts IDLE (S) to P RRC correspond to incoming/outgoing calls. Prompts E SEZ (R) to P FRLS (R) correspond to incoming calls. Prompts E SEZ (S) to C SUPO (S) correspond to outgoing calls.

Feature operation

No specific operating procedures are required to use this feature.

Chapter 9: 510 Trunk Route Member Expansion

Contents

This section contains information on the following topics for Avaya Communication Server 1000 (Avaya CS 1000):

Feature description

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

The need for this expansion was particularly evident for a system to DMS Central Office connectivity, where the full potential of a T-1 nB+D configuration (384 B-Channels) or an E-1 nB+D configuration (480 B-Channels) was not able to be realized. The DMS supports one D-Channel for each route. With a limit of 254 route members for each D-Channel, this meant that the maximum offered by nB+D could not be utilized.

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

For a system to system connectivity, a user is now able to configure, on one trunk route, one D-Channel with 382 (T1) or 480 (E1) B-Channels, a Backup D-Channel, and a second D-Channel with another 128 (T1) or 30 (E1) B-Channels. If desired, another Backup D-Channel can be configured. Refer to Figure 2: 510 trunk members expansion configured for a system to system connectivity (T1) on page 106 for a T1 representation, and to Figure 3: 510 trunk members expansion configured for a system to system connectivity (E1) on page 106 for an E1 representation.

For a system to DMS connectivity, a user can now configure the full complement of nB+D, which is one D-Channel with 382 (T1) or 480 (E1) B-Channels, and a Backup D-Channel. Refer to Figure 4: 510 trunk members expansion configured for a system to DMS connectivity (T1) on page 107 for a T1 representation, and to Figure 5: 510 trunk members expansion configured for a system to DMS connectivity (E1) on page 107 for an E1 representation.

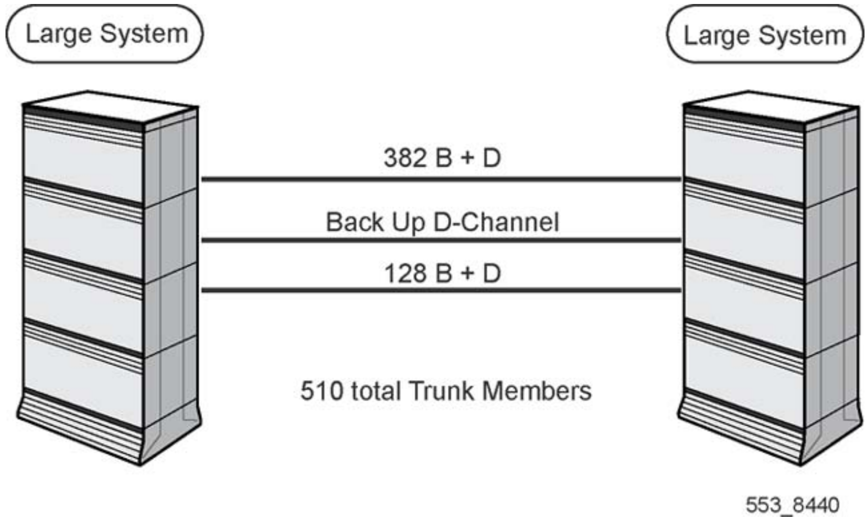

Figure 2: 510 trunk members expansion configured for a system to system connectivity (T1)

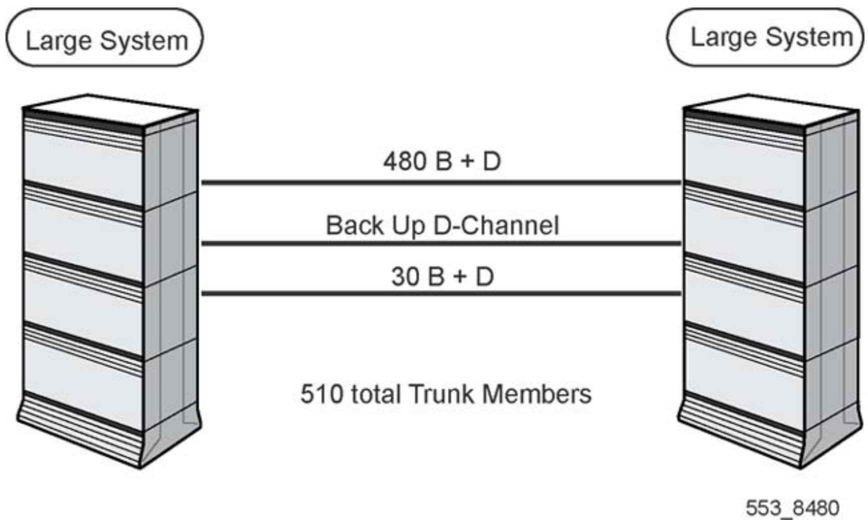

Figure 3: 510 trunk members expansion configured for a system to system connectivity (E1)

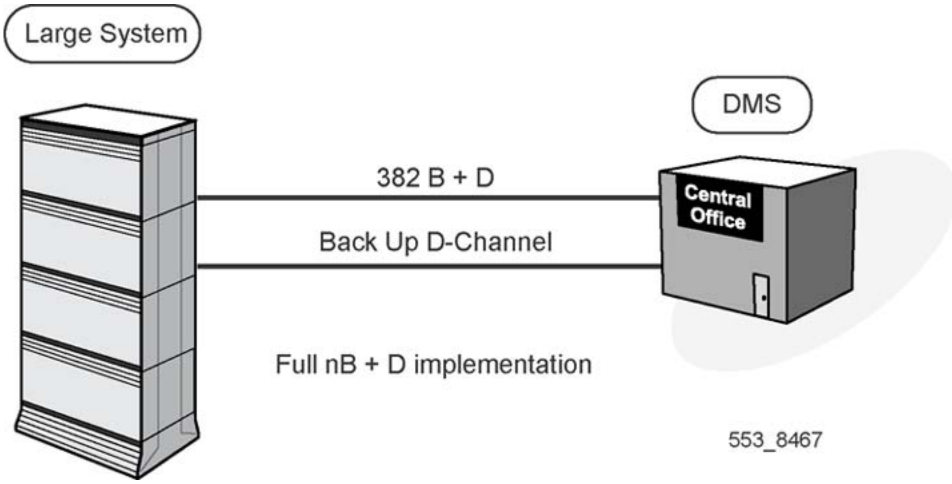

Figure 4: 510 trunk members expansion configured for a system to DMS connectivity (T1)

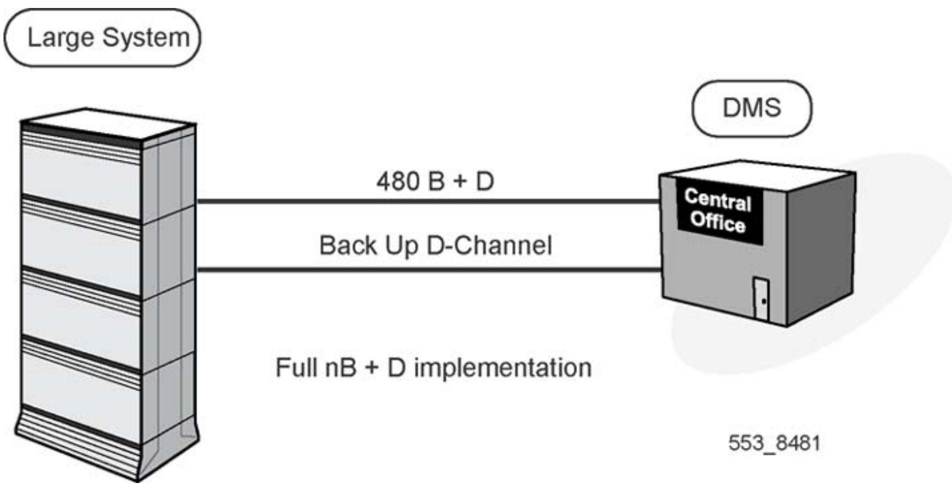

Figure 5: 510 trunk members expansion configured for a system to DMS connectivity (E1)

This feature applies to the following trunk mode configurations:

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

Note:
ISDN BRI trunking is not supported in North America.

The following trunk types are supported:

- Integrated Services Access (ISA)
- NI-2 Call By Call Service Selection (CBCT)

Note:
Integrated Services Access and NI-2 Call By Call Service Selection are only supported in North America.

- FGDT (Feature Group D)
- COT (Central Office)
- TIE
- DID (Direct Inward Dial)
- FEX (Foreign Exchange)
- WATS (Wide Area Telecommunication Service)

The overlay programs that are affected are:

- Trunk Data Block (LD 14)
- Route Data Block (LD 16)
- ISDN BRI trunking (LD 27)

Note:
ISDN BRI trunking is not supported in North America.

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

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

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

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

Operating parameters

There are no operating parameters associated with this feature.

Feature interactions

Integrated Services Access

The ISA minimum (MIN) and maximum (MAX) fields for service routes can be configured in LD 16 at a value from 0-510.

Meridian MAX

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

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

NI-2 Call By Call Service Selection

The NI-2 Call By Call Service Selection maximum (MAX) field for service routes can be configured in LD 16 at a value from 0-510.

Preference Trunk Usage

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

Feature packaging

This feature requires the following packages:

- Basic (BASIC) package 0
- Basic Rate Interface Trunk (BRIT) package 233 for BRI trunking applications

Feature implementation

Task summary list

The following is a summary of the tasks in this section. For supported trunks other than ISA and NI-2 CBC trunks:

For supported trunks other than ISA and NI-2 CBC trunks

Table 57: LD 16 - Define 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 , Media Gateway 1000B, and CS 1000E system.
DES	x...x	Designator field for trunk (0-16 alphanumeric characters).
TKTP		Supported trunk type (other than ISA or CBC).
	COT	Central Office.
	TIE	TIE
	DID	Direct Inward Dial.
	FEX	Foreign Exchange.
	WAT	Wide Area Telecommunication Service.
	FGDT	Feature Group D.
...		
DTRK	(NO) YES	Define whether or not the trunk route is digital. Enter NO for analog, YES for digital.
- DGTP	PRI	For digital trunks, enter PRI as the digital trunk type. This prompt appears only if DTRK = YES.
...		
ISDN		Integrated Services Digital Network
	(NO) YES	Enter NO for analog trunks or YES for digital trunks.
- MODE	aaa	Mode of operation, where aaa can be: APN = Analog Private Network. ISLD = ISDN Signaling Link. PRA = Primary Rate Access (ISDN must be YES).
- IFC	aaa	Interface type, where aaa can be: D100 = DMS-100. D250 = DMS-250. CS 1000M(the default value).
...		
NSF	YES	Network Service Facility option. This option is used to indicate to the system whether it is to expect a Network Service Facility (NSF) Information Element (IE) from the DMS. The NSF information dictates which service route the system will use to terminate a call. Enter YES. The default value is NO.
...		

Prompt	Response	Description
ICOG	aaa	Incoming or outgoing trunk type, where aaa can be: ICT = Incoming only. OGT = Outgoing only. IAO = Both incoming and outgoing.

Table 58: LD 14 - Define the trunk type, and the route member number.

Prompt	Response	Description
REQ	NEW	Add new data.
	CHG	Change existing data.
TYPE		Trunk type
	COT	Central Office.
	TIE	TIE
	DID	Direct Inward Dial.
	FEX	Foreign Exchange.
	WAT	Wide Area Telecommunication Service.
	FGDT	Feature Group D.
TN		Terminal number
	l s c u	Format for Large System , Media Gateway 1000B, and CS 1000E system, where l = loop, s = shelf, c = card, u = unit.
CUST	xx	Customer number, as defined in LD 15
...		
RTMB		Route number and Member Number
	0-511 1-4000	Range for Large System , Media Gateway 1000B, and CS 1000E system.

Table 59: LD 27 - Define the trunk type as BRIT, and the route member number on a Digital Subscriber Loop (DSL).

Prompt	Response	Description
REQ	NEW	Add new data.
	CHG	Change existing data.
TYPE	DSL	Administer the DSL data block.
DSL		Digital Subscriber Loop.

Prompt	Response	Description
	l s c dsl	For Large System , Media Gateway 1000B, and CS 1000E system.
DES	d...d	DSL designator (1 to 6 alphanumeric characters).
APPL	aaaa	Basic Rate Interface application (BRIT or BRIE).
...		
B1	(NO) YES	(Do not) change B-Channel 1 configuration.
- MEMB	1-510	Member number of BRI route.
...		
B2	(NO) YES	(Do not) change B-Channel 2 configuration.
- MEMB	1-510	Member number of BRI route.

For ISA trunks

Table 60: LD 16 - Define the ISA master route.

Prompt	Response	Description
REQ	NEW	Add a 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 , Media Gateway 1000B, and CS 1000E system.
TKTP	ISA	Create an ISA master route.
...		
DTRK	YES	The trunk route is digital.
- DGTP	PRI	For digital trunks, enter PRI as the digital trunk type. This prompt appears only if DTRK = YES.
...		
ISDN	YES	Integrated Services Digital Network

Promp t	Response	Description
- MODE	PRA	Mode of operation is Primary Rate Access.
- IFC	aaaa	Interface type, where aaaa can be: D100 = DMS-100. D250 = DMS-250. CS 1000M(the default value).
...		
ICOG	aaa	Incoming or outgoing trunk type, where aaa can be: ICT = Incoming only. OGT = Outgoing only. IAO = Both incoming and outgoing.

Table 61: LD 14 - Define the trunk type as ISA and the route member number.

Promp t	Response	Description
REQ	NEW xxx	Add new data. Follow NEW with xxx = 1-510, to create that number of consecutive trunks.
TYPE	ISA	ISA trunk type.
TN		Terminal number
	l s c u	Format for Large System , Media Gateway 1000B, and CS 1000E system, where l = loop, s = shelf, c = card, u = unit.
CUST	xx	Customer number, as defined in LD 15
...		
RTMB		Route number and Member Number
	0-511 1-4000	Range for Large System , Media Gateway 1000B, and CS 1000E system.

Table 62: LD 16 - Define the maximum and minimum number of channels for ISA service route.

Promp t	Response	Description
REQ	NEW	Add a 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 , Media Gateway 1000B, and CS 1000E system.

Prompt	Response	Description
TKTP	aaaa	Type of ISA service route.
...		
DTRK	YES	The trunk route is digital.
- DGTP	PRI	For digital trunks, enter PRI as the digital trunk type. This prompt appears only if DTRK = YES.
...		
ISDN	YES	Integrated Services Digital Network
- MODE	PRA	Mode of operation is Primary Rate Access.
- IFC	aaaa	Interface type, where aaaa can be: D100 = DMS-100. D250 = DMS-250. CS 1000M(the default value).
...		
- ISAR	YES	ISA service route.
- - MIN	0-510	Minimum number of channels allowed on an ISA service route.
- - MAX	0-510	Maximum number of channels allowed on an ISA service route.
...		
ICOG	aaa	Incoming or outgoing trunk type, where aaa can be: ICT = Incoming only. OGT = Outgoing only. IAO = Both incoming and outgoing.

For NI-2 CBC trunks

Table 63: LD 16 - Define the NI-2 CBC master route.

Prompt	Response	Description
REQ	NEW	Add a 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 , Media Gateway 1000B, and CS 1000E system.

Prompt	Response	Description
TKTP	CBCT	Trunk type is NI-2 Call By Call.
...		
DTRK	YES	Define whether or not the trunk route is digital. Enter YES for digital.
- DGTP	PRI	For digital trunks, enter PRI as the digital trunk type. This prompt appears only if DTRK = YES.
...		
ISDN	YES	Integrated Services Digital Network
- MODE	PRA	Mode of operation is Primary Rate Access.
- IFC	NI2	Interface type is NI-2.
- IPUB		Service route to be used for incoming network call
	0-511	For Large Systems
...		
ICOG	aaa	Incoming or outgoing trunk type, where aaa can be: ICT = Incoming only. OGT = Outgoing only. IAO = Both incoming and outgoing.

Table 64: LD 14 - Define the trunk type as NI-2 CBC and the route member number.

Prompt	Response	Description
REQ	NEW xxx	Add new data. Follow NEW with xxx = 1-510, to create that number of consecutive trunks.
TYPE	CBCT	NI-2 Call By Call trunk type.
TN		Terminal number
	l s c u	Format for Large System , Media Gateway 1000B, and CS 1000E system, where l = loop, s = shelf, c = card, u = unit.
...		
CUST	xx	Customer number, as defined in LD 15
...		
RTMB		Route number and Member Number
	0-511 1-4000	Range for Large System , Media Gateway 1000B, and CS 1000E system.

Table 65: LD 16 - Define the maximum number of channels for each NI-2 service 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 , Media Gateway 1000B, and CS 1000E system.
TKTP		CBCT service route trunk type.
	COT	Central Office.
	TIE	TIE
	DID	Direct Inward Dial.
	FEX	Foreign Exchange.
	WAT	Wide Area Telecommunication Service.
...		
ISDN	YES	ISDN route.
- IFC	NI2	Interface type is NI-2.
- CBCR	YES	NI-2 Cal By Call Service route indicator.
- - RTN		Master route number
	0-511	For Large Systems
- - SRVC	(0)-31	Decimal value identifying the type of service provisioned for the NI-2 service route.
...		
- - MAX	0-510	Maximum number of channels allowed on an CBCT service route.

Feature operation

No specific operating procedures are required to use this feature.

Chapter 10: Advice of Charge

Contents

This section contains information on the following topics for Avaya Communication Server 1000 (Avaya CS 1000):

Feature description

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

Introduction

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

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

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

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

AOC is summarized below for the following connectivities:

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

Storing charging information

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

- One meter is assigned to each phone, through Metered Assigned (MRA) Class of Service, on a TN basis. The charge unit count is stored in the meter of the station TN from which the call is made
- One meter, classified as "ATTN", is provided to collect charges for all metered calls made by attendant consoles within the same customer.
- One meter, classified as "ROUTE", is provided for each route.

- At each node, one meter classified as "CUST" is provided for each customer to accumulate any charges that cannot be charged to any other meter for the customer.
- One temporary meter, classified as TRK, is provided for each trunk; this temporary meter is used to accumulate charges for each call. At the end of the call, the charge is added to the permanent meter associated with the caller.

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

The charging information can also be recorded in the Call Detail Recording (CDR) record, as provided by the CDR Enhancement capability (for more information on the CDR Enhancement, refer to *Avaya Call Detail Recording Fundamentals, NN43001-550*). This information is printed at the end of call, and appears in the same location of the CDR ticket as the PPM call charge information would appear, immediately following the PPM counts.

Reading and changing meters

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

Displaying charging information

Charging information can be displayed on display-equipped Meridian digital telephones. This display capability is provided by the Charge Display at End of Call feature. This feature appends the charge information to existing information on the display and retains the information displayed for 10 seconds. Depending on the CO interface, the information can be displayed at call set-up, during a call, and/or at end of a call (for more information on the Charge Display at End of Call feature, refer to *Avaya Features and Services Fundamentals, NN43001-106*).

Printing charging information

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

Printing charging information on the TTY

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

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

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

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

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

Handling overflow charging information

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

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

Note:

When the meter overflows, the meter can still have stored charge information equal to less than the maximum. Therefore, it sometimes appears to a user that the charge is small for the length of time spent on the phone. Therefore, it is important to have a background terminal in order not to lose all of the charge information related to a call. To get the total charge for a call, it is necessary to add the information in the CDR record to the background terminal overflow messages.

Operating parameters

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

- There are limitations with respect to the size of the charge that can be displayed in either the CDR record or on the phone display. In addition, currency symbols and decimal places cannot be displayed on phones or CDR records.
- Central Offices can send the charge with an accuracy of up to one decimal place. Smaller amounts are rounded up to the nearest integer, and thus rounding errors can have a minor affect on charging accuracy.
- The Advice of Charge information is not transported between a system to system connection. Therefore, if a tandem call is made from one system to another system, and then to a Central Office, the originator's display is not updated with the charge information. However, the charge is printed in the CDR record at the switch that made the connection to the Central Office.
- Charging information received on incoming calls, such as reverse charging, is not supported. The existing error indication is given if an incoming call is received. This does not happen in Australia because the AOC information must be requested for each call and the system does not support requesting AOC information for incoming calls.

Feature interactions

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

Attendant Calls

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

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

Attendant operation of metered calls

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

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

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

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

Automatic Call Distribution

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

Barge-in/Busy Verification

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

Break-in

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

BRI Terminal

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

BRI Trunk Access for Japan

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

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

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

Call Park

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

Call Pickup

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

Call Transfer

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

Call Waiting

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

Conference

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

Consultation

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

Data Calls

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

Hospital Management

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

Hunt

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

In-band Tones and Announcements

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

Malicious Call Trace (Enhanced)

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

Meridian Mail

The same interaction applies as for Call Transfer.

Message Registration

Due to the change in the packaging for MR/PPM, anyone using MR must ensure that the prompt in LD 17 is set to MR before setting up the associated data. This is the default.

Multiple Appearance DN

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

Multiple Party Operation

The same interaction applies as for Call Transfer.

Network Call Transfer

Advice of Charge is not supported for Network Call Transfers.

Periodic Pulse Metering (PPM)

The following PPM capabilities and treatments are supported by AOC:

- Multiple Appearance DN
- Attendant Recall
- Automatic Call Distribution
- Barge-in
- Break-in
- Busy Verification
- Recording the accumulated call charging information for each call on the CDR record (if CDR is equipped)
- Calculating the total charge for each call, based on the assigned unit cost and the accumulated call charging information
- Allowing the attendant to mark a call as being metered, and to display the charge for the call, and
- Allowing a digital phone equipped with an MRK key and digit display to access meters.

Radio Paging

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

Recovery of Misoperation of Call Transfer

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

Advice of Charge for Central Office Connectivity

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

AXE-10 Australia (end of call)

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

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

Feature packaging

This feature requires the following packages:

- Call Detail Recording (CDR) package 4
- Call Detail Recording Teletype Terminal (CTY) package 5
- Controlled Class of Service (CCOS) package 81
- Background Terminal (BGD) package 99
- Periodic Pulse Metering/Message Registration (MR) package 101
- Integrated Services Digital Network (ISDN) package 145
- 2.0 Mbit Primary Rate Interface (PRI2) package 154
- 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 66: LD 17 - Select PPM functionality and CDR format. on page 130
2. Table 67: LD 15 - Allow CDR Charge. on page 130
3. Table 68: LD 15 - Allow Charge Display. on page 130
4. Table 69: LD 16 - Allow AOC on the route. on page 130
5. Table 70: LD 10/11 - Assign meters to phones. on page 131
6. Table 71: LD 12 - Assign a Meter Recall key to the Attendant Console. on page 131

Assume an ISDN Interface has been setup.

Table 66: LD 17 - Select PPM functionality and CDR format.

Prompt	Response	Description
REQ	CHG	Change existing data.
TYPE	CFN	Configuration record.
…		
PARM	YES	System parameters.
MTRO	PPM	Periodic Pulse Metering.

Table 67: LD 15 - Allow CDR Charge.

Prompt	Response	Description
REQ	CHG	Change existing data.
TYPE	PPM	Periodic Pulse Metering.
CUST		Customer number
	0-99	Range for Large System , Media Gateway 1000B, and CS 1000E system.
…		
- UCST	x	Unit cost, x = (0)-9999.

Table 68: LD 15 - Allow Charge Display.

Prompt	Response	Description
REQ	CHG	Change existing data.
TYPE	FTR	Customer features and options.
CUST		Customer number
	0-99	Range for Large System , Media Gateway 1000B, and CS 1000E system.
…		
OPT	CHDA	Charge display allowed.

Table 69: LD 16 - Allow AOC on the route.

Prompt	Response	Description
REQ	CHG	Change existing data.

Prompt	Response	Description
TYPE	RDB	Route data block.
…		
ISDN	YES	Configure ISDN data.
IFC	AXEA	ISDN Interface for Australia. Must be configured in LD 17 for the applicable D-channel.
…		
CDR	YES	Call Detail Recording.
…		
OTL	YES	CDR on outgoing toll calls.
…		
OAN	YES	CDR on all answered outgoing calls.
…		
MR	ENDC	AOC End of Call allowed for international ISDN interfaces.
RUCS	1	Route unit cost value received is treated as charge.
RUCF	1 0	Route unit conversion factor, no conversion required.

Table 70: LD 10/11 - Assign meters to phones.

Prompt	Response	Description
REQ	NEW	Add new data.
	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 , Media Gateway 1000B, and CS 1000E system, where l = loop, s = shelf, c = card, u = unit.
…		
CLS	MRA	Message Registration Allowed.

Table 71: LD 12 - Assign a Meter Recall key to the Attendant Console.

Prompt	Response	Description
REQ	NEW	Add new data.
	CHG	Change existing data.

Prompt	Response	Description
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 , Media Gateway 1000B, and CS 1000E system, where l = loop, s = shelf, c = card, u = unit.
…		
KEY	xx MTR	Assign a Meter recall key on the attendant console.

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

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

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

Operating parameters

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

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

Feature packaging

This feature requires the following packages:

- EuroISDN (EURO) package 261
- Call Detail Recording (CDR) package 4
- Call Detail Recording Teletype Terminal (CTY) package 5
- Controlled Class of Service (CCOS) package 81
- Background Terminal (BGD) package 99
- Periodic Pulse Metering/Message Registration (MR) package 101
- International Supplementary Features (SUPP) package 131

For PRI connectivity, the following packages are also required:

- Integrated Services Digital Network (ISDN) package 145
- 2.0 Mbit Primary Rate Access (PRI2) package 154
- ISDN Supplementary Features (ISDNS) package 161
- 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 78: LD 17 - Select PPM functionality and CDR format. on page 138
2. Table 73: LD 15 - Allow CDR Charge. on page 134
3. Table 74: LD 15 - Allow Charge Display. on page 134
4. Table 75: LD 16 - Allow AOC on the route. on page 135
5. Table 76: LD 10/11 - Assign meters to phones. on page 136
6. Table 85: LD 12 - Assign a Meter recall key on the Attendant Console. on page 141

Assume an ISDN Interface has been setup.

Table 72: LD 17 - Select PPM functionality and CDR format.

Prompt	Response	Description
REQ	CHG	Change existing data.
TYPE	ADAN	Change I/O devices.
- ADAN	CHG DCH XX	Change D-channel, xx = 0-15.
…		
- IFC	EURO	EuroISDN interface.
- CNTY		Enter country pertaining to EuroISDN interface.
	AUS	Austria
	DEN	Denmark
	(ETSI)	ETS 300-102 basic protocol
	FIN	Finland
	GER	Germany
	ITA	Italy

Promp t	Response	Description
	NOR	Norway
	POR	Portugal
	SWE	Sweden
	EIR	Ireland
	DUT	Holland
	SWI	Switzerland
	BEL	Belgium
	ESP	Spain
	UK	United Kingdom
	FRA	France
	CIS	Commonwealth of Independent States (Russia and the Ukraine).

Table 73: LD 15 - Allow CDR Charge.

Promp t	Response	Description
REQ	CHG	Change existing data.
TYPE	PPM	Periodic Pulse Metering.
CUST		Customer number
	0-99	Range for Large System, Media Gateway 1000B, and CS 1000E system.
…		
- UCST	x	Unit cost, x = (0)-9999.

Table 74: LD 15 - Allow Charge Display.

Promp t	Response	Description
REQ	CHG	Change existing data.
TYPE	FTR	Customer features and options.
CUST	xx	Customer number, as defined in LD 15
…		
OPT	CHDA	Charge display allowed.

Table 75: LD 16 - Allow AOC on the 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, Media Gateway 1000B, and CS 1000E system.
…		
ISDN	YES	Configure ISDN data.
…	…	Other ISDN sub-prompts.
-IFC	EURO	EuroISDN interface.
- CNTY		Enter country pertaining to EuroISDN interface.
	AUS	Austria
	DEN	Denmark
	(ETSI)	ETS 300-102 basic protocol
	FIN	Finland
	GER	Germany
	ITA	Italy
	NOR	Norway
	POR	Portugal
	SWE	Sweden
	EIR	Ireland
	DUT	Holland
	SWI	Switzerland
	BEL	Belgium
	ESP	Spain
	UK	United Kingdom
	FRA	France
	CIS	Commonwealth of Independent States (Russia and the Ukraine).
MR		Selected AOC subservice. Not printed for Denmark and Sweden.
	(NO)	No AOC service.

Prompt	Response	Description
	ENDC	AOC-E subservice activated.
	DURC	AOC-D subservice activated.
	STAC	AOC-S subservice activated.
RUCS	0-9999	Route unit cost. Not printed for Denmark and Sweden
RURC	0-9999 (0)-3	Route unit reference cost. Note that the formula for the route unit reference cost is: X*10 (-Y), where X=0-9999, and Y=0-3. The default value for X is identical to the previously entered RUCS value. Not printed for Denmark and Sweden.
RUCF	0-(1)-9999 (0)-3	Route unit conversion factor. Note that the formula for the route unit reference cost is: X*10 (-Y),where X=0-9999, and Y=0-3. The default value for X is identical to the previously entered RUCS value. Not printed for Denmark and Sweden.

Table 76: LD 10/11 - Assign meters to phones.

Prompt	Response	Description
REQ	NEW	Add new data.
	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, Media Gateway 1000B, and CS 1000E system, where l = loop, s = shelf, c = card, u = unit.
CLS	MRA	Message Registration allowed.

Table 77: LD 12 - Assign a Meter Recall key to the Attendant Console.

Prompt	Response	Description
REQ	NEW	Add new data.
	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 , Media Gateway 1000B, and CS 1000E system, where l = loop, s = shelf, c = card, u = unit.
…		

Prompt	Response	Description
KEY	xx MTR	Assign a Meter recall key on the attendant console.

INS1500 (Japan D70) (end of call)

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

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

Special handling for call charges that exceed normal capacity

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

Feature packaging

This feature requires the following packages:

- Call Detail Recording (CDR) package 4
- Call Detail Recording Teletype Terminal (CTY) package 5
- 1.5 Mbit Digital Trunk Interface (PBXI) package 75
- Controlled Class of Service (CCOS) package 81
- Background Terminal (BGD) package 99
- Periodic Pulse Metering/Message Registration (MR) package 101
- Integrated Services Digital Network (ISDN) package 145
- 1.5 Mbit Primary Rate Access (PRA) package 146
- International Primary Rate Access (IPRA) package 202

Feature implementation

Task summary list

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

The following steps assume an ISDN Interface has already been set up.

Table 78: LD 17 - Select PPM functionality and CDR format.

Prompt	Response	Description
REQ	CHG	Change existing data.
TYPE	CFN	Configuration record.
PARM	YES	Parameters for Interface and transmission mode.
…		
FCDR	NEW	Use New CDR format (recommended for Japan).
…		
MTRO	PPM	Periodic Pulse Metering.

Table 79: LD 15 - Allow CDR Charge.

Prompt	Response	Description
REQ	CHG	Change existing data.
TYPE	PPM	Periodic Pulse Metering.
CUST		Customer number
	0-99	Range for Large System , Media Gateway 1000B, and CS 1000E system.
…		
- UCST	x	Unit cost, x = (0)-9999.

Table 80: LD 15 - Allow Charge Display.

Prompt	Response	Description
REQ	CHG	Change existing data.
TYPE	FTR	Customer features and options.
CUST		Customer number
	0-99	Range for Large System , Media Gateway 1000B, and CS 1000E system.
…		
OPT	CHDA	Charge display allowed.

Table 81: LD 16 - Allow AOC on the route.

Prompt	Response	Description
REQ	CHG	Change existing data.
TYPE	RDB	Route data block.
…		
ISDN	YES	Configure ISDN data.
IFC	D70	ISDN Interface for Japan.
…		
CDR	YES	Call Detail Recording.
…		
OTL	YES	CDR on outgoing toll calls.
…		
OAN	YES	CDR on all answered outgoing calls.
…		
MR	ENDC	AOC End of Call allowed for international ISDN interfaces.
RUCS	1	Route unit cost value received is treated as charge.
RUCF	1 0	Route unit conversion factor, no conversion required.

Table 82: LD 10/11 - Assign meters to phones.

Prompt	Response	Description
REQ	NEW	Add new data.

Prompt	Response	Description
	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 , Media Gateway 1000B, and CS 1000E system, where l = loop, s = shelf, c = card, u = unit.
…		
CLS	MRA	Message Registration Allowed.

Table 83: LD 12 - Assign a Meter Recall key to the Attendant Console.

Prompt	Response	Description
REQ	NEW	Add new data.
	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 , Media Gateway 1000B, and CS 1000E system, where l = loop, s = shelf, c = card, u = unit.
…		
KEY	xx MTR	Assign a Meter recall key on the Attendant Console.

Numeris (end of call)

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

Feature packaging

This feature requires the following packages:

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

Feature implementation

Task summary list

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

Table 84: LD 17 - Change Configuration Record to allow PPM.

Prompt	Response	Description
REQ	CHG	Change existing data.
TYPE	CFN	Configuration record.
PARM	YES	Parameters for Interface and transmission mode.
…		
MTRO	PPM	Periodic Pulse Metering.

Table 85: LD 12 - Assign a Meter recall key on the Attendant Console.

Prompt	Response	Description
REQ	CHG	Change existing data.
TYPE	aaaa	Type of console, where aaaa = 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 , Media Gateway 1000B, and CS 1000E system, where l = loop, s = shelf, c = card, u = unit.
…		
KEY	xx MTR	key number, Meter key.

Table 86: LD 15 - Configure the Customer Data Block for AOC.

Prompt	Response	Description
REQ	CHG	Change existing data.

Promp t	Response	Description
TYPE	FTR	Features and options.
CUST		Customer number
	0-99	Range for Large System , Media Gateway 1000B, and CS 1000E system.
…		
ICI	xx MTR	ICI number, Meter Recall.

Table 87: LD 15 - Configure Periodic Pulse Metering.

Promp t	Response	Description
REQ	CHG	Change existing data.
TYPE	PPM	Periodic Pulse Metering.
CUST		Customer number
	0-99	Range for Large System , Media Gateway 1000B, and CS 1000E system.
HMTL	(YES) NO	Hotel/Motel environment.
PCDL	YES	PPM output on CDR Link. Additional three words added to tape record.
UCST	x	Unit cost for Periodic Pulse Metering, x=(0)-9999.
ATCH	(NO) YES	Attendant display of call charge.

Table 88: LD 16 - Modify the Trunk Route for AOC.

Promp t	Response	Description
REQ	NEW	Add new data.
	CHG	Change existing data.
TYPE	aaaa	Type of data block.
CUST	xx	Customer number, as defined in LD 15
…		
OAL	YES	CDR on Outgoing calls.
- OTL	YES	CDR on Outgoing toll calls.
…		
CCO	YES	Printing CDR records for no PPM or AOC count.

Promp t	Response	Description
…		
MR	ENDC	The AOC information is decoded at the end of the call.
…		
RUCF	x y	Route Unit Conversion Factor.

Numeris and SwissNet (during call)

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

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

Advice of Charge is provided as follows:

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

Operating parameters

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

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

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

Feature packaging

This feature requires the following packages:

Periodic Pulse Metering/Message Registration (MR) package 101

For PRI connectivity, the following packages are required:

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

Feature implementation

Task summary list

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

1. Table 89: LD 17 - Select PPM functionality and CDR format. on page 144
2. Table 90: LD 15 - Allow CDR Charge. on page 144
3. Table 91: LD 15 - Allow Charge Display. on page 145
4. Table 92: LD 16 - Allow AOC on the route. on page 145
5. Table 93: LD 10/11 - Assign meters to phones. on page 146
6. Table 95: LD 12 - Assign a Meter recall key on the Attendant Console. on page 147

Table 89: LD 17 - Select PPM functionality and CDR format.

Prompt	Response	Description
REQ	CHG	Change existing data.
TYPE	CFN	Configuration record.
PARM	YES	Parameters for Interface and transmission mode.
…		
MTRO	PPM	Periodic Pulse Metering.

Table 90: LD 15 - Allow CDR Charge.

Prompt	Response	Description
REQ	CHG	Change existing data.

Prompt	Response	Description
TYPE	PPM	Periodic Pulse Metering.
CUST		Customer number
	0-99	Range for Large System , Media Gateway 1000B, and CS 1000E system.
…		
- UCST	x	Unit cost, x = (0)-9999.

Table 91: LD 15 - Allow Charge Display.

Prompt	Response	Description
REQ	CHG	Change existing data.
TYPE	FTR	Customer features and options.
CUST		Customer number
	0-99	Range for Large System , Media Gateway 1000B, and CS 1000E system.
…		
OPT	CHDA	Charge display allowed.
…		

Table 92: LD 16 - Allow AOC on the 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 , Media Gateway 1000B, and CS 1000E system.
…		
ISDN	YES	Configure ISDN data.
IFC	NUME SWIS	ISDN Interface for Numeris (France). ISDN Interface for Swis (Switzerland).
CDR	YES	Call Detail Recording.

Prompt	Response	Description
OTL	YES	CDR on outgoing toll calls.
OAN	YES	CDR on all answered outgoing calls.
MR	DURC	AOC during a call.
CCO	TES	Disable the printing of CDR N records.
RUCS	1	Route unit cost value received is treated as charge.
RUCF	1 0	Route unit conversion factor, no conversion required.

Table 93: LD 10/11 - Assign meters to phones.

Prompt	Response	Description
REQ	NEW	Add new data.
	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 , Media Gateway 1000B, and CS 1000E system, where l = loop, s = shelf, c = card, u = unit.
…		
CLS	MRA	Message Registration Allowed.

Table 94: LD 12 - Assign a Meter Recall key to the Attendant Console.

Prompt	Response	Description
REQ	NEW	Add new data.
	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 , Media Gateway 1000B, and CS 1000E system, where l = loop, s = shelf, c = card, u = unit.
…		
KEY	xx MTR	Assign a Meter key on the attendant console.

1TR6 (end of call)

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

Feature packaging

This feature requires the following packages:

Periodic Pulse Metering/Message Registration (MR) package 101

For PRI connectivity, the following packages are required:

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

Feature implementation

Task summary list

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

Table 95: LD 12 - Assign a Meter recall key on the Attendant Console.

Prompt	Response	Description
REQ	CHG	Change existing data.

Prompt	Response	Description
TYPE	2250	Attendant Console type.
TN		Terminal number
	l s c u	Format for Large System , Media Gateway 1000B, and CS 1000E system, where l = loop, s = shelf, c = card, u = unit.
…		
KEY	x MTR x NULL	Key number, Meter key. Remove a Meter key.

Table 96: LD 15 - Configure the Customer Data Block for AOC.

Prompt	Response	Description
REQ	CHG	Change existing data.
TYPE	NET	ISDN and ESN Networking options.
CUST		Customer number
	0-99	Range for Large System , Media Gateway 1000B, and CS 1000E system.
…		
ISDN	YES	Integrated Services Digital Network allowed for customer.
…		
- PFX1	xxxx	Prefix 1.
- PFX2	xxxx	Prefix 2.

Table 97: LD 15 - Configure call charging for nodes with 1TR6 connections.

Prompt	Response	Description
REQ	CHG	Change existing data.
TYPE	ATT	Attendant Console options.
CUST		Customer number
	0-99	Range for Large System , Media Gateway 1000B, and CS 1000E system.
…		
ICI	x MTR	ICI number, Meter Recall.

Table 98: LD 15 - Configure charge recording for nodes with 1TR6 connections.

Prompt	Response	Description
REQ	CHG	Change existing data.
TYPE	PPM	Periodic Pulse Metering.
CUST		Customer number
	0-99	Range for Large System , Media Gateway 1000B, and CS 1000E system.
HMTL	(YES) NO	Hotel/Motel environment.
PCDL	YES	PPM output on CDR Link. Additional three words added to tape record.
UCST	(0) - 9999	Unit Cost for Periodic Pulse Metering.
ATCH	(NO) YES	Attendant display of call Charge.
SCDL	(0) 1 2 3	Schedule for printing Message Registration and PPM data. No scheduled printing. Daily printout. Weekly printout. Monthly printout.
- WKDY	1-7	Week Day for weekly printout. Where 1 = Sunday.
- DAY	0-28	Day of month for printout.
- HOUR	hh or hh hh	Hour of day for printout.
- MCLR	(NO) YES	Meter Clear after printing.
- PTTY	(0)-15	PPM TTY number for printing meters, one per switch.

Table 99: LD 17 - Allow for AOC.

Prompt	Response	Description
REQ	CHG	Change existing data.
TYPE	ADAN	All input/output devices.
ADAN	CHG DCH x	Action device and number, x = 0-15.
…		
ISDN	YES	Integrated Services Digital Network.
…		
- IFC	1TR6	1 TR6 for Germany.
- CNEG	(1) 2	Channel Negotiation option.
- LAPD	YES	Link Access Protocol for D-channel. Change LAPD parameters.

Prompt	Response	Description
- - T203	2-(10)-40	Maximum Time allowed without frames being exchanged in seconds.

Table 100: LD 16 - Change Trunk Route for AOC.

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 , Media Gateway 1000B, and CS 1000E system.
…		
ISDN	YES	Integrated Services Digital Network
- IFC	1TR6	1TR6 for Germany

Table 101: LD 16 - Configure call-charge metering and printing for nodes with 1TR6 connections.

Prompt	Response	Description
REQ	NEW	Add new data.
	CHG	Change existing data.
TYPE	RDB	Route data block.
CUST		Customer number
	0-99	Range for Large System , Media Gateway 1000B, and CS 1000E system.
ROUT		Route number
	0-511	Range for Large System , Media Gateway 1000B, and CS 1000E system.
…		
CDR	YES	Call Detail Recording.
- OAL	YES	CDR on outgoing calls.
- - OTL	YES	CDR on Outgoing Toll calls.

Prompt	Response	Description
CCO	(NO) YES	Printing of CDR records for no PPM or AOC count
RUCS	0-9999	Route Unit Cost.

Feature operation

No specific operating procedures are required to use this feature.

Chapter 11: Alternative Call Routing for NBWM

Alternative Call Routing for Network Bandwidth Management (NBWM) allows a station-to-station call (that is, a call that does not use a trunk) between a branch office and main office to overflow to traditional routes. This can occur if there is insufficient inter-zone bandwidth available to carry the call or the Quality of Service (QoS) has degraded to unacceptable levels. The feature also applies to station-to-station calls from one branch office to another branch office, provided both stations are registered to the same main office.

For more information about this feature, refer to *Avaya Branch Office Installation and Commissioning, NN43001-314*.

Chapter 12: Analog Semi-Permanent Connections

Contents

This section contains information on the following topics for Avaya Communication Server 1000 (Avaya CS 1000):

Feature description

The Analog Semi-Permanent Connections (ASPC) feature provides the capability to automatically re-establish any disconnected ASPC call that the system detects. The user configures the ASPC feature for a trunk and sets a re-connection timer defined at the customer level.

The XFEM card in a system controls incoming and outgoing manual trunks. Each trunk is dedicated to a single piece of equipment. When equipment linked to an incoming trunk wants to establish a communication with equipment on an outgoing manual trunk, both trunks have to be linked in a software manner using the Manual Directory Number (MNDN) prompt. The originating equipment has to connect the E-lead of the trunk on its side to the ground. This connection is seen as a seize message on the PBX side; both trunks are then linked together. The configuration can also involve calls across intermediate TIE trunks.

If communication is broken after an ASPC call is established between two systems, the incoming trunk is left in the busy state. A message is then printed along with the date, time, manual incoming trunk TN and its corresponding MNDN on the TTY. This message indicates that the call has been disconnected. Re-connection is attempted at regular intervals based on the ASPC timer. The call is re-established once the connection is made again. The call can also be re-established when the IPE manual incoming E&M TIE trunk is disabled and enabled in LD 36. Another message is printed on the TTY when the call is re-established.

When the system detects a disconnected ASPC call, the ASPC re-connection Timer (ASPCT) starts. When the ASPCT time expires, re-connection of the call is attempted. If the call is not established, the ASPCT time is reset and the process repeats until the call is re-established.

To enable/disable this feature, the Analog Semi-Permanent Connections Allowed (SPCA) or Analog Semi-Permanent Connections Denied (SPCD) Class of Service must be defined in LD 14 for incoming manual trunks.

Operating parameters

The ASPC feature is only applicable to IPE 2/4 E&M manual incoming trunks. It does not apply to trunks that are Incoming and Outgoing (IAO). Set up ASPC over IPE 2/4 wire E&M analog trunks with the Manual Incoming Service Allowed (MIA) Class of Service on XFEM trunk cards.

ASPC trunks must have ASPC Class of Service on each switch on which incoming manual trunks are terminating. Therefore, such switches must be C machines only. However, Omega machines can be present on any other node of the network, including tandem nodes and outgoing nodes.

If system initialization occurs during reconnection, the ASPC reconnection mechanism fails and the call is not re-established.

The incoming manual trunk, when seized at the far end, is automatically terminated on the MNDN. The manual outgoing trunk service dials the outgoing route access code to complete an outgoing call after ringing the trunk. Therefore, an incoming seizure from the manual trunk (performed as soon as one piece of equipment connects the E-lead to the ground), configured to terminate on a second piece of equipment, using the outgoing manual route access code, ends up as an established tandem connection between the two pieces of equipment.

The MNDN of the incoming trunk can provide access to a private network route, and then to another manual outgoing E&M route. In this event, the MNDN cannot be the collation of the two access codes, because of the resulting conflict between the first access code and the MNDN. One solution is to define a Trunk Steering Code with Digit Manipulation. The digits related with the MNDN are defined to give access to the private route, then deleted to be replaced by the access code of the E&M route.

Feature interactions

There are no feature interactions associated with this feature.

Feature packaging

The Analog Semi-Permanent Connection (ASPC) 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 102: LD 15 - Configure ASPC reconnection timer. on page 157
2. Table 103: LD 16 - Configure Outgoing Manual Route. on page 158
3. Table 104: LD 16 - Configure Incoming Manual Route. on page 158
4. Table 105: LD 14 - Configure Incoming Manual Trunk. on page 159

Table 102: LD 15 - Configure ASPC reconnection timer.

Prompt	Response	Description
REQ	CHG	Change existing data.
TYPE	FTR	Customer features and options.
CUST		Customer number
	0-99	Range for Large System , Media Gateway 1000B, and CS 1000E system.
....	....	
BSFE	(NO) YES	Boss Secretary Filtering Enhancement

Prompt	Response	Description
ASPCT	(10)–180	ASPC reconnection Timer (in seconds).

Table 103: LD 16 - Configure Outgoing Manual Route.

Prompt	Response	Description
REQ	CHG	Change route data.
TYPE	RDB	Route data block.
CUST		Customer number
	0-99	Range for Large System , Media Gateway 1000B, and CS 1000E system.
....	....	
TKTP	a..a	Outgoing manual trunk type
...		
ICOG	OGT	Outgoing route.
ACOD	xxx	Access code for the trunk route.
NEDC	ETH	Near End Disconnect Control. If the far end goes on-hook for either incoming or outgoing calls, the on-hook condition is recognized and the call is disconnected.
FEDC	ETH	Far End Disconnect Control. Conditions at the near end are recognized for both incoming and outgoing calls.
MANO	YES	Manual Outgoing trunk route.

Table 104: LD 16 - Configure Incoming Manual Route.

Prompt	Response	Description
REQ	CHG	Change route data.
TYPE	RDB	Route data block.
CUST		Customer number
	0-99	Range for Large System , Media Gateway 1000B, and CS 1000E system.
....	....	
TKTP	a..a	Outgoing manual trunk type
....	....	

Promp t	Response	Description
ICOG	ICT	Incoming route.
NEDC	ETH	Near End Disconnect Control on either side.
FEDC	ETH	Far End Disconnect Control on either side.

Table 105: LD 14 - Configure Incoming Manual Trunk.

Promp t	Response	Description
REQ	CHG	Change existing data.
TYPE	TIE	TIE trunk data block.
TN		Terminal number
	l s c u	Format for Large System , Media Gateway 1000B, and CS 1000E system, where l = loop, s = shelf, c = card, u = unit.
...		
CUST		Customer number
	0-99	Range for Large System , Media Gateway 1000B, and CS 1000E system.
....	....	
RTMB		Route number and Member Number
	0-511 1-4000	Range for Large System , Media Gateway 1000B, and CS 1000E system.
...		
MNDN	xx...x	Manual Directory Number xx...x is the access code of the outgoing manual route or is the DN of the terminating phone.
CLS	MIA	Manual Incoming Allowed.
	SPCA	ASPC allowed.
	SPCD	ASPC denied (default).

Feature operation

No specific operating procedures are required to use this feature.

Chapter 13: Attendant and Network Wide Remote Call Forward

Contents

This section contains information on the following topics for Avaya Communication Server 1000 (Avaya CS 1000):

Feature description

In the past, the Remote Call Forward (RCFW) feature allowed a user to administer Call Forward from a remote phone within the system or from outside the system through the Direct Inward System Access (DISA) number. The RCFW feature was not available on a network wide basis, nor was it applicable to Attendant Consoles. This enhancement introduces the RCFW feature across the Meridian Customer Defined Network (MCDN), while also providing the attendant with RCFW capabilities.

The feature capabilities of the phone-based (FFC activated) network wide application of the RCFW feature match those of the current stand-alone RCFW feature.

The attendant RCFW functionality is controlled by a flexible Attendant key (RFW). The attendant has the capability to view the current call forward number and determine the Call Forward status of any station. It is also possible for an attendant to activate or deactivate call forward for a particular station. This functionality is applicable both locally within the system and network wide. ISDN FACILITY messages are used to facilitate the RCFW Feature operation in an MCDN.

Attendant RCFW does not require a station password to activate call forward for a station. An optional, customer-based, password can be configured for attendant operation of the RCFW feature.

Operating parameters

The network wide application of this feature is only applicable to nodes in an MCDN. No other Central Office (CO) or PBX type is supported for this Feature operation.

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

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

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

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

Feature interactions

Outpulsing of Asterisk and Octothorpe (OAPO)

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

BRI

Since ISDN BRI phones do not support Flexible Feature Codes (FCCs), Remote Call Forward is not supported on BRI phones.

Multiple Appearance DNs

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

Network operation of the RCFW feature simply provides network access to the stand-alone RCFW feature, therefore the requirement that only one prime DN per Multiple Appearance DN also applies to the network-based RCFW feature.

However, with the stand-alone phone-based RCFW feature, there is no code in place to explicitly prohibit the configuration of Multiple Appearance DNs with the same prime DN and Station Controlled Passwords (SCPWs) on different stations. No code is added for the network or attendant implementation of this feature. As such, in the event that multiple stations are configured with the same prime DN, the phone-based network RCFW Feature operation will be the same as that for stand-alone RCFW Feature operation.

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

If none of the stations having the DN and SCPW assigned are configured as the MARP TN for that DN, the RCFA and RCFD will apply to all stations matching the DN and SCPW. RCFW will apply to the station with MADN call presentation priority (i.e., the station with the last service change is placed at the end of the list).

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

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

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

Preventing Reciprocal Call Forward

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

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

Phantom TN

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

The Phantom TN feature allows users to configure the CFW DN of Phantom TNs to their current location. The Phantom TN feature uses the RCFW feature to configure and activate/deactivate the CFW DN on the Phantom TNs.

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

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

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

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

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

Traffic Measurements

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

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

Feature packaging

The following package is required for Attendant and Network Wide Remote Call Forward:

The Attendant Remote Call Forward (ARFW) package 253

For phone-based RCFW, the following packages are required:

- Optional Features (OPFT) package 1
- Controlled Class of Service (CCOS) package 81
- Flexible Feature Codes (FCC) package 139

For implementation on PBX phones, the following software packages are required:

- Special Service for 2500 phones (SS25) package 18
- 500 Set Dial Access to Features (SS5) package 73

For network operation, the following software packages are required:

- Integrated Services Digital Network (ISDN) package 145
- Network Alternate Route Selection (NARS) package 58 and/or
- Coordinated Dialing Plan (CDP) package 59

Feature implementation

Task summary list

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

Phone-based Configuration:

1. Table 106: LD 15 - Set the Station Control Password Length. on page 166
2. Table 107: LD 15 - Configure a Special Prefix Number (SPRE) for the customer. on page 166
3. Table 108: LD 15 - Map the NARS/BARS access code to the incoming call types. on page 167
4. Table 109: LD 10 - Set the Station Control Password and allow Call Forward. on page 167

5 Table 110: LD 11 - Set the Station Control Password and allow Call Forward. on page 168

6 Table 111: LD 16 - Configure the route data block. on page 168

7 Table 112: LD 57 - Define Remote Call Forward FFCs and set FFCT. on page 169

Attendant-based Configuration:

8 Table 113: LD 12 - Configure the Flexible Attendant feature key, RFW. on page 169

9 Table 114: LD 15 - Configure the Attendant RCFW password. on page 170

10 Table 115: LD 15 - Map the NARS/BARS access code to the incoming call types. on page 170

Phone-based Configuration

Table 106: LD 15 - Set the Station Control Password Length.

Prompt	Response	Description
REQ	CHG	Change existing data.
TYPE	FFC	Flexible Feature Codes.
CUST		Customer number
	0-99	Range for Large System , Media Gateway 1000B, and CS 1000E system.
...		
- SCPL	0-8	Station Control Password Length (must be consistent network wide).

Table 107: LD 15 - Configure a Special Prefix Number (SPRE) for the customer.

Prompt	Response	Description
REQ	CHG	Change existing data.
TYPE	FTR	Customer features and options.
CUST		Customer number
	0-99	Range for Large System , Media Gateway 1000B, and CS 1000E system.
...		
- SPRE	xxxx	Special Prefix Number.

Table 108: LD 15 - Map the NARS/BARS access code to the incoming call types.

Prompt	Response	Description
REQ	CHG	Change existing data.
TYPE	NET	Networking data.
CUST		Customer number
	0-99	Range for Large System , Media Gateway 1000B, and CS 1000E system.
...		
AC2	aaaa	Access code 2, as defined in LD 86. aaaa = NPA, NXX, INTL, SPN, LOC.
ISDN	YES	ISDN capabilities.
- PNI	1-327000	Private Network Identifier.
...		
- - - HLOC	0-9999999 X	Home location code (ESN), 1-7 digits. X = delete digits.

Table 109: LD 10 - Set the Station Control Password and allow Call Forward.

Prompt	Response	Description
REQ	CHG	Change existing data.
TYPE	500	Phone type.
TN		Terminal number
	l s c u	Format for Large System , Media Gateway 1000B, and CS 1000E system, where l = loop, s = shelf, c = card, u = unit.
...		
SCPW	xxxxxxxx	Station Control Password (0 to 8 digits, defined in LD 15).
...		
CLS	CFXA	Call Forward to External DN allowed.
...		
FTR	aaaa	Feature configuration.

Table 110: LD 11 - Set the Station Control Password and allow Call Forward.

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 , Media Gateway 1000B, and CS 1000E system, where l = loop, s = shelf, c = card, u = unit.
...		
SCPW	xxxxxxxx	Station Control Password (0 to 8 digits, defined in LD 15).
...		
CLS	CFXA	Call Forward to External DN allowed.
...		
KEY	xx CFW 4-(16)-23	Assign Call Forward key (xx) and set the forwarding DN length.

Table 111: LD 16 - Configure the route data block.

Prompt	Response	Description
REQ	CHG	Change existing data.
TYPE	RDB	Route data block.
CUST		Customer number
	0-99	Range for Large System , Media Gateway 1000B, and CS 1000E system.
...		
ISDN	YES	ISDN configuration.
...		
INAC	YES	Insert ESN access codes to incoming private network calls.
...		
- PNI	1-327000	Private Network Identifier.

Table 112: LD 57 - Define Remote Call Forward FFCs and set FFCT.

Prompt	Response	Description
REQ	CHG	Change existing data.
TYPE	FFC	Flexible Feature Codes.
CUST		Customer number
	0-99	Range for Large System , Media Gateway 1000B, and CS 1000E system.
FFCT	(NO) YES	Confirmation tone is (is not) to be given after an FFC.
CODE	RCFA	Remote Call Forward Activate.
RCFA	xx	xx = RCFA code.
CODE	RCFD	Remote Call Forward Deactivate.
RCFD	xx	xx = RCFD code.
CODE	RCFV	Remote Call Forward Verify.
RCFV	xx	xx = RCFV code.

Attendant-based Configuration

Configuration of the key on the Attendant Console is required to allow attendant access to the RCFW feature. Configuration of the RFW key is only allowed if the ARFW package is equipped.

Table 113: LD 12 - Configure the Flexible Attendant feature key, RFW.

Prompt	Response	Description
REQ	NEW	Add new data.
	CHG	Change existing data.
TYPE	2250	Attendant Console type.
TN		Terminal number
	l s c u	Format for Large System , Media Gateway 1000B, and CS 1000E system, where l = loop, s = shelf, c = card, u = unit.
...		
KEY	xx RFW	Key number assigned as Attendant Remote Call Forward key.

Table 114: LD 15 - Configure the Attendant RCFW password.

Prompt	Response	Description
REQ	CHG	Change existing data.
TYPE	ATT	Attendant console.
CUST		Customer number
	0-99	Range for Large System , Media Gateway 1000B, and CS 1000E system.
...		
- IRFR	(NO) YES	Internal Remote Call Forward Password required.
-- IRFP	xxxxxxxx	Internal RCFW Password (only prompted if the response to IRFR is YES). The password length is one to eight digits; the password is numeric only.
- XRFR	(NO) YES	External Remote Call Forward Password required.
-- XRFP	xxxxxxxx	External RCFW password (only prompted if the response to XRFR is YES). The password length is one to eight digits; the password is numeric only.

Table 115: LD 15 - Map the NARS/BARS access code to the incoming call types.

Prompt	Response	Description
REQ	CHG	Change existing data.
TYPE	NET	Networking data.
CUST		Customer number
	0-99	Range for Large System , Media Gateway 1000B, and CS 1000E system.
...		
AC2	aaaa	Access code 2, as defined in LD 86. aaaa = NPA, NXX, INTL, SPN, LOC.
ISDN	YES	ISDN capabilities.
- PNI	1-327000	Private Network Identifier.
...		
- - - HLOC	0-9999999	Home location code (ESN), 1-7 digits.
	X	X = delete digits.

Feature operation

Network Wide Phone-based Remote Call Forward

From the remote phone dial:

1 FFC RCFA code.

2 SCPW for the phone to be forwarded.

3 The complete DN of the phone to be forwarded. This DN is the full DN required to call the phone to be forwarded from the user's present location.

Expected Result: Provided everything is correct, confirmation tone is delivered to the user.

From the remote phone continue dialing:

4 The CFW DN to be activated followed by the end of dial indicator (#).

Expected Result: Confirmation tone is delivered to the user.

Network Wide Attendant-based Remote Call Forward

From the Attendant Console, perform the following:

1 Press an idle loop key followed by the RFW key.

Expected Result: The RFW key is flashing and the Loop key is steady lit.

2 Dial the DN of the phone to be forwarded.

Expected Result: If a password is required, the RFW key is winking, and the console display shows "PWD –". If the console does not support alpha characters, the display will be blank.

If a password is not required, the console display will show the DN of the phone to be forwarded followed by the CFW DN stored on that phone. The RFW key lamp will display the status of the CFW DN. If the RFW lamp is flashing, CFW is not active; if the RFW lamp is steadily lit, CFW is active. Proceed to Step 4.

3 Dial the password followed by #.

Expected Result: The console display will show the DN of the phone to be forwarded followed by the CFW DN stored on that phone. The RFW key lamp will display the status of the CFW DN. If the RFW lamp is flashing, CFW is not active; if the RFW lamp is steadily lit, CFW is active.

4 The user can now enter a new CFW DN or press the RFW key to activate or deactivate the stored CFW DN.

Expected Result: The console display will show the DN of the phone to be forwarded followed by the CFW DN. If the RFW lamp is flashing, CFW is not active; if the RFW lamp is steadily lit, CFW is active.

5 When RCFW operation is in this state, the user has the following three options:

a Press the Release or Release Source key to terminate the RCFW operation.

b Press the RFW key to reverse the CFW status.

c Enter a new CFW DN to begin the task of changing the CFW DN programmed. The new CFW DN is not active until the RFW key is pressed again.

Chapter 14: Attendant Blocking of Directory Number

Contents

This section contains information on the following topics for Avaya Communication Server 1000 (Avaya CS 1000):

Feature description

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

When the attendant is ready to make the external (long distance) call, the Attendant Blocking of DN feature provides the attendant with the ability to block the DN while the external call request is being processed. The line appears busy to any caller attempting to contact the blocked DN. The blocked DN cannot be used to originate a call and will be connected to the attendant if it goes off hook.

When the attendant has completed the external call, the blocked DN can be rung and the call extended. The attendant is guaranteed that the requesting DN is not busy and is available to take the call when the processing has been completed. This feature works in both stand-alone and Meridian Customer Defined Network (MCDN) environments.

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

Operating parameters

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

The attendant has the ability to use the Attendant Blocking of DN feature only for the following types of DNs: phones with ordinary Single Call Arrangement No Ringing key (SCN)/Single Call Arrangement Ringing key (SCR) DNs and PBX phones. HOT DNs, MCN/MCR DNs, ACD DNs, PLDNs, any trunk access code, FFCs, BRI and all other types of extensions are considered to be invalid for the Attendant Blocking of DN feature.

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

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

No new hardware is required for this feature.

Feature interactions

ACD

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

Advice of Charge for EuroISDN

For Advice of Charge at start of the call (AOC-S) and during the call (AOC-D), charging information is assigned respectively to the Attendant and the phone's Message Registration (MR) meters for the charge incurred before and after the call transfer completion by an attendant. Advice of Charge at end of the call (AOC-E) charging information is assigned to the phone's MR meter.

Attendant Hold

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

Automatic Redial

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

Break-in

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

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

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

Busy Lamp Field/Enhanced Busy Lamp Field

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

Busy Verify

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

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

Call Detail Recording Time to Answer

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

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

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

Call Forward No Answer

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

Call Park

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

Call Waiting

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

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

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

Camp-on

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

CDR Time to Answer

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

Directory Number Delayed Ringing

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

Do Not Disturb

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

FFC Boss Secretary Filtering

The FFC Boss Secretary Filtering feature will be overridden. If an Attendant Blocking of DN attempt is made for a phone that has the Boss Secretary Filtering feature active, the dialed DN will be blocked if idle. If it is busy, busy tone will be heard.

Flexible Feature Codes

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

Group Hunting

It is not possible to activate the Attendant Blocking of DN feature for a Pilot DN (PLDN). If an attempt is made to block a PLDN, the attempt will be canceled and overflow tone will be returned. If a DN that is a member in a Group Hunt (or Hunt) list is blocked by the Attendant Blocking of DN feature, the DN is considered to be busy.

Hunting

If Attendant Blocking of DN is attempted on a busy DN having the Hunting feature active, busy tone will be returned (overriding the Hunting feature).

Idle Extension Notification

The Attendant Blocking of DN feature and the Idle Extension feature both use the SACP key for feature activation on the source side of the Attendant Console. The difference is that Attendant Blocking of DN only is valid when dialing a DN, whereas Idle Extension Notification only is valid when a busy DN is reached. If an Attendant Blocking of DN attempt is made for a DN that is busy, the Attendant Blocking of DN is canceled and it is possible to activate the Idle Extension Notification feature for the busy DN.

Intercept Computer Interface (ICP)

The Attendant Blocking of DN feature will override the ICP Call Forward feature. If the dialed DN of the phone that has the ICP Call Forward feature active is idle, the DN will be blocked and if the DN is busy, busy tone will be heard.

ISDN Basic Rate Interface (BRI) Trunk Access

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

Line Lockout

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

Make Set Busy

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

Multiple Appearance Multiple Call Arrangement

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

Multiple Appearance Single Call Arrangement

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

New Flexible Code Restriction

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

Phantom TN

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

Radio Paging

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

Ring Again

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

Ringing Change Key

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

Ring/Hold LED Status

When a DN is blocked, the status of the DN lamp will be according to the Ring/Hold LED status for ringing calls.

Semi-automatic Camp-on

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

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

Signal Source

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

Single Call No Ringing DN

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

Source Included when Attendant Dials

The Attendant Blocking of DN feature will follow the current Source Included when Attendant Dialing handling, depending on what is configured.

Vacant Number Routing

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

Feature packaging

The following package is required for Attendant Blocking of Directory Number:

Semi-automatic Camp-on (SACP) package 181

For an ISDN network environment, the following package is required:

Network Attendant Services (NAS) package 159

Feature implementation

Task summary list

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

1. Table 116: LD 12 - Configure the SACP key on the Attendant Console. on page 182
2. Table 117: LD 15 - Modify the Customer Data Block. on page 182

Table 116: LD 12 - Configure the SACP key on the Attendant Console.

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 , Media Gateway 1000B, and CS 1000E system, where l = loop, s = shelf, c = card, u = unit.
...		
KEY	xx SACP	Semi-automatic Camp-on key.

Table 117: LD 15 - Modify the Customer Data Block.

Prompt	Response	Description
REQ	CHG	Change existing data.
TYPE	ATT	Attendant Console options.
CUST		Customer number
	0-99	Range for Large System , Media Gateway 1000B, and CS 1000E system.
...		
ATT	YES	Change Attendant Console options.
...		
- SACP	(NO)	Semi-automatic Camp-on not allowed.
	ALL	Semi-automatic Camp-on for all Camp-on occurrences.
	SNGL	Semi-automatic Camp-on an a per call basis.
- ABDN	(NO) YES	Activation of the Attendant Blocking of DN feature is (not) allowed. The ABDN prompt only appears when the SACP package is equipped.

Feature operation

To block a DN

1. The attendant presses an idle Loop key.
2. The attendant presses the Semi-automatic Camp-on (SACP) key.
3. The SACP key lamp lights.
4. The attendant dials the source DN that is to be blocked. If the dialed DN is idle, the DN lamp will have the same state as if it were ringing, but the DN will not ring. If the DN is busy, the attendant hears busy tone and the SACP key lamp darkens.
5. If the dialed DN is idle, it is blocked. The DN lamp indicates a ringing state, although the DN will not ring (for PBX phones, there is no indication that the DN is blocked.) On the Attendant Console, the SACP key lamp remains lit and the Source key lamp begins blinking.

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

To place an outgoing call for the blocked DN

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

To release a blocked DN

The attendant presses either the SACP key or Signal Source key to ring the DN –or– The attendant presses either the Release key or the Release Source key to release the source DN, which then becomes idle.

To notify a blocked DN of an established call

The attendant presses the SACP or Signal Source key.

Chapter 15: Attendant Through Dialing Networkwide

Contents

This section contains information on the following topics for Avaya Communication Server 1000 (Avaya CS 1000):

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.

Figure 6: Attendant Through Dialing Networkwide on page 186 illustrates Attendant Through Dialing Networkwide.

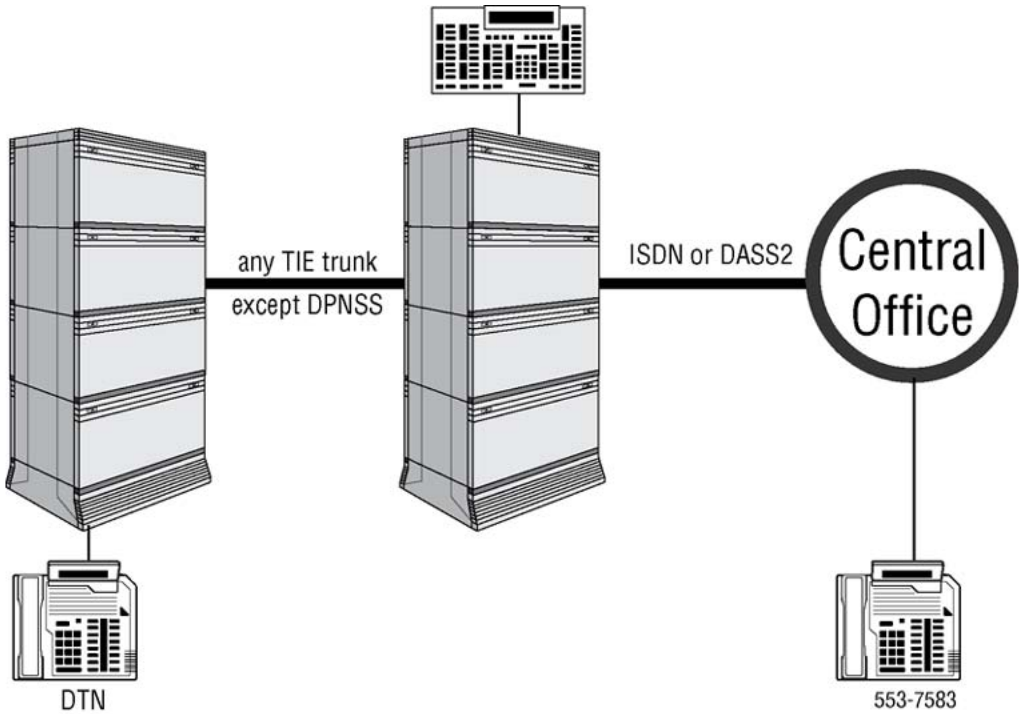

Figure 6: Attendant Through Dialing Networkwide

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 118: Numbering Plans and Attendant Release of external trunk on page 186 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 118: 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 119: LD 15 - Allow Attendant Through Dialing Networkwide. on page 190
2. Table 120: LD 15 - Configure Improved End-to-End Signaling. on page 191
3. Table 121: LD 17 - Allow Calling Line Identification (CLID) field in Call Detail Recording (CDR) records. on page 191

Table 119: 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 , Media Gateway 1000B, and CS 1000E system.
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 120: 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 , Media Gateway 1000B, and CS 1000E system.
...		
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 121: 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 16: Australia ETSI

Contents

This section contains information on the following topics for Avaya Communication Server 1000 (Avaya CS 1000):

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 Large Systems and 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:

Primary Rate Configuration

Table 122: 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 123: 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 for Large Systems.
- CTYP	MSDL	Multi-purpose Serial Data Link card or Downloadable D-Channel Daughterboard Large Systems.
- 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.
- 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.

Prompt	Response	Description
- - CNTY	EAUS	Australia ETSI.
- - PINX_CUST	0-99	The customer number to be used for the DN address translation associated with call independent connection messages received on the D-Channel.
...		
- - DCHL	0-159	PRI2 loop number for D-channel.
- 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.

Prompt	Response	Description
- - 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 124: 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		Customer number
	0-99	Range for Large System , Media Gateway 1000B, and CS 1000E system.
ROUT		Route number
	0-511	Range for Large System , Media Gateway 1000B, and CS 1000E system.
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.

Prompt	Response	Description
	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 125: LD 14 - Configure the Australia ETSI ISDN PRI2 trunks.

Prompt	Response	Description
REQ	NEW	Add new data.
	CHG	Change existing data.
TYPE		**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
	c u	Format for Media Gateway 1000B, where c = card and u = unit.
CUST		Customer number
	0-99	Range for Large System , Media Gateway 1000B, and CS 1000E system.

Prompt	Response	Description
RTMB		Route number and Member Number
	0-511 1-4000	Range for Large System , Media Gateway 1000B, and CS 1000E system.
...		
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 126: 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 127: LD 15 - Allow Charge Display and CDR Charge.

Prompt	Response	Description
REQ	CHG	Change existing data.
TYPE	PPM	Periodic Pulse Data.
CUST		Customer number
	0-99	Range for Large System , Media Gateway 1000B, and CS 1000E system.
OPT	CHDA	Charge Display Allowed.
...		
UCST	(0)-9999	Unit cost for PPM.

Table 128: LD 10 - Assign meters to analog (500/2500-type) phones.

Prompt	Response	Description
REQ	NEW	Add new data.

Prompt	Response	Description
	CHG	Change existing data.
TYPE	500	Analog phone.
TN		Terminal number
	l s c u	Format for Large System , Media Gateway 1000B, and CS 1000E system, where l = loop, s = shelf, c = card, u = unit.
DES	x...x	ODAS Station Designator.
CUST		Customer number
	0-99	Range for Large System , Media Gateway 1000B, and CS 1000E system.
...		
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 129: LD 11 - Assign meters to the system proprietary phones.

Prompt	Response	Description
REQ	NEW	Add new data.
	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 , Media Gateway 1000B, and CS 1000E system, where l = loop, s = shelf, c = card, u = unit.
DES	x...x	ODAS Station Designator.
CUST		Customer number
	0-99	Range for Large System , Media Gateway 1000B, and CS 1000E system.
...		
TGAR	0 - (1) - 30	Trunk Group Access Restriction The default of 1 automatically blocks direct access.
...		

Prompt	Response	Description
CLS	MRA	Message registration Allowed.
...		
KEY	xx aaa yyy	Phone function key assignments.

Table 130: 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		Customer number
	0-99	Range for Large System , Media Gateway 1000B, and CS 1000E system.
ROUT		Route number
	0-511	Range for Large System , Media Gateway 1000B, and CS 1000E system.
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.

Prompt	Response	Description
...		
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 131: 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	(NO) YES	LAPD parameters. (NO) = Does not prompt the LAPD parameters and assigns the default values shown in ( ) to these parameters. YES = Define or modify the LAPD parameters.
USER	(NO) YES	(Do not) print groups selected at PGN prompt.
- T200	(2)-40	Retransmission timer specifies the time delay before the system retransmits the information. Delay is in increments of 0.5 seconds.
- T203	4-(20)-80	Maximum time between transmission frames Delay is in increments of 0.5 seconds.

Prompt	Response	Description
- 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 132: 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		Customer number
	0-99	Range for Large System , Media Gateway 1000B, and CS 1000E system.
ROUT		Route number
	0-511	Range for Large System , Media Gateway 1000B, and CS 1000E system.
TKTP	TIE	TIE trunk type.
	COT	COT Central Office Trunk trunk type.
	DID	Direct Inward Dialing trunk type.
...		
DTRK	YES	BRI Digital Trunk Route
BRIP	NO	ISDN BRI packet handler route (NO is entered, since packet data is not required).
- DGTP	BRI	Digital trunk type.
ISDN	YES	Integrated Services Digital Network.
...		
- IFC	EURO	EuroISDN interface.
- - CNTY	EAUS	Australia ETSI.
- CNEG	(NO) YES	(Do not) allow Channel Negotiation.
...		
OVLS	(NO) YES	(Do not) allow Overlap Sending.

Prompt	Response	Description
- 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.
...		
- 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 133: 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.
APPL		Application type.
	BRIE	Enter BRIE for Australia ETSI.
APPL	<CR>	To end configuration procedure.

Table 134: 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.
	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
MISP		Must be an even loop number that has already been configured.
	0-158	MISP loop number for Large 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 135: 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.

Prompt	Response	Description
TYPE	DSL	Digital Subscriber Loop data block.
DSL		Digital Subscriber Loop.
	lll s cc dsl#	For Large System , Media Gateway 1000B, and CS 1000E system, where: 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
APPL	BRIE	BRI trunk application for Australia ETSI.
CUST		Customer number
	0-99	Range for Large System , Media Gateway 1000B, and CS 1000E system.
CTYP	SILC	Interface card type is SILC.
MISP	0-158	MISP loop number for Large Systems
MODE	TE	Enter TE (user side) as the mode for Australia ETSI.
- 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 , Media Gateway 1000B, and CS 1000E system.
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.

Prompt	Response	Description
- 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 136: LD 27 - Assign meters to a DSL.

Prompt	Response	Description
REQ	NEW	Add an ISDN protocol group.
TYPE	DSL	LAPD Protocol group.
DSL		Digital Subscriber Loop.
	lll s cc dsl#	For Large System , Media Gateway 1000B, and CS 1000E system, where: 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
...		
CLS	MRA	Allow Message Registration on the DSL.

Table 137: 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		Customer number
	0-99	Range for Large System , Media Gateway 1000B, and CS 1000E system.
ROUT		Route number
	0-511	Range for Large System , Media Gateway 1000B, and CS 1000E system.
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.

Prompt	Response	Description
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 17: Backup D-channel

Contents

This section contains information on the following topics:

Feature description

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

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

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

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

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

Note:

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

Operating parameters

There are no operating parameters associated with this feature.

Feature interactions

ISDN QSIG Basic Call

Backup D-channel is not supported on the QSIG interface.

Feature packaging

This feature requires the following packages:

- Integrated Services Digital Network (ISDN) package 145
- 1.5 Mbit Primary Rate Access (PRA) package 146 or
- ISDN Signaling Link (ISL) package 147 or
- 2.0 Mbit Primary Rate Interface (PRI2) package 154

Feature implementation

Note:

Basic PRI or ISL administration must be performed before the backup D-channel is defined. Also, the PRI loop must already be defined in LD 17.

Table 138: LD 17: Change the Configuration Record to define the Backup D-channel.

Prompt	Response	Description
REQ	CHG	Change existing data.
TYPE	CFN	Configuration Record.
ADAN	NEW BDCH x	Add a backup D-channel (also CHG, MOV, and OUT BDCH). x = 0-63
PDCH	0-63	Primary D-channel.
CTYP	DCHI	D-channel interface card.
DNUM	0-15	Device number: physical port (odd) for D-channel on DCH, physical card address for MSDL.
- PORT	0-3	Port number on MSDL card.
RCVP	(NO) YES	Auto-recovery to primary D-channel option.
BCHL	0-159	PRI loop number for back-up D-channel.

Table 139: Recovery to primary D-channel RCVP prompt responses

RCVP = YES	RCVP = NO
primary D-channel up—active backup D-channel up	primary D-channel up—active backup D-channel up
primary D-channel down backup D-channel up—active (see below)	
primary D-channel up—active backup D-channel up	primary D-channel up backup D-channel up—active

When RCVP is YES, the primary D-channel is down, and the backup D-channel is up, and the following occurs. First the switch tries to re-establish the primary D-channel connection. If this cannot be done successfully, then the backup D-channel is switched in. When the primary D-channel is brought up again, the primary D-channel becomes the active D-channel.

If RCVP is NO and the primary D-channel is down, the backup D-channel remains active when the primary D-channel is brought up. It is important to note that the backup D-channel remains the active D-channel.

Feature operation

No specific operating procedures are required to use this feature.

Chapter 18: Basic Call Service

Contents

This section contains information on the following topics:

Feature description

ISDN Basic Call Service provides for the transmission of ISDN calls. Basic call service consists of call-progress signaling, and voice and data transmission.

ISDN PRI Basic Call Service is supported for system to system, and system to Central Offices that support AXE-10, Numerous VN2, 1TR6, Japan D70, SwissNet, NEAX-61, SYS-12, Asia Pacific, Australia ETSI, or EuroISDN protocols.

Call progress signaling

ISDN PRI supports 64 Kbit/s out-of-band signaling (on the D-channel) to effect:

- call setup
- call tear down
- feature activation
- local-busy and reorder tones (overflow tone is supplied locally)

Both out-of-band signaling messages and in-band audible tones are provided for ringback.

Voice and data transmission

Voice and high-speed data are transmitted over B-channels. Call connections are assigned to these B-channels on a per-call basis. The following modes of voice and data transport are available:

- 64 Kbit/s circuit-switched voice and data transmission
- 64 Kbit/s packet data transmission (or interfaces that support it)

Numbering plans

Numbering plans supported on ISDN PRI are the following:

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

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

The following variations apply in the United Kingdom:

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

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
- 1.5 Mbit Primary Rate Access (PRA) package 146 or
- ISDN Signaling Link (ISL) package 147 or
- 2.0 Mbit Primary Rate Interface (PRI2) package 154

Feature implementation

There are no specific implementation procedures for this feature.

Feature operation

No specific operating procedures are required to use this feature

Chapter 19: B-channel Overload Control

Contents

This section contains information on the following topics:

Applicable regions

This feature is only available in North America. Contact your system supplier or your Avaya representative to verify support of this product in your area.

Feature description

The B-channel Overload Control feature provides a solution to the high rate of incoming calls from a Central Office over ISDN PRI trunks to busy destinations on the system.

For example, a telemarketing firm advertises a product by giving an 1-800 number. In response to the advertisement, the company receives a burst of incoming calls in a short period of time. Many of the calls can receive busy treatment. In a busy situation, the system rejects the calls.

The feature delays the release of an ISDN PRI call by using a configurable timer (BCOT) when a call encounters a busy condition. The delay in releasing the seized B-channel prevents a new call from being presented on the same B-channel. This delay cumulatively results in decreasing the incoming call rate, thereby avoiding degradation of real-time response. This delay is applied using ISDN protocol compliant messaging. The delay is in milliseconds, so that it is virtually transparent to the caller.

The value for the B-channel Control timer is configured in LD 16, on a per-route basis. Although a value from 0-4000 milliseconds is accepted, a value of 256 is recommended. The entered value is rounded down to multiples of 128. After a value has been entered and rounded down (if necessary), the value is printed on the screen before the next prompt is displayed. For example, if a value of 400 is entered, the system rounds this value down to 384 and prints 384 on the display. The next prompt is then displayed.

After the BCOT timer expires, the normal disconnect sequence takes place and a new call can be presented.

Figure 7: Before the B-channel Overload Control Timer has been applied on page 223 and Figure 8: After the B-channel Overload Control Timer has been applied on page 224 represent a "before-and-after" representation of the B-channel Overload Control feature application.

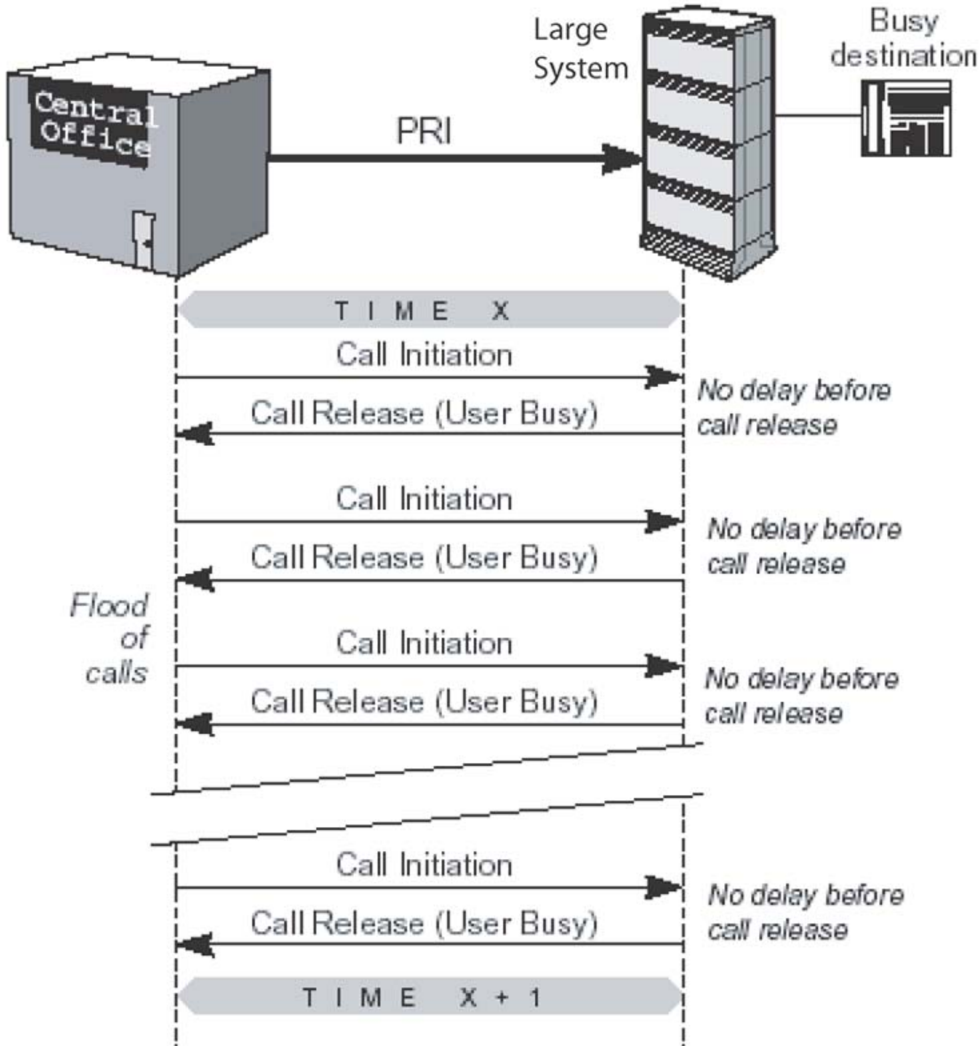

Figure 7: Before the B-channel Overload Control Timer has been applied

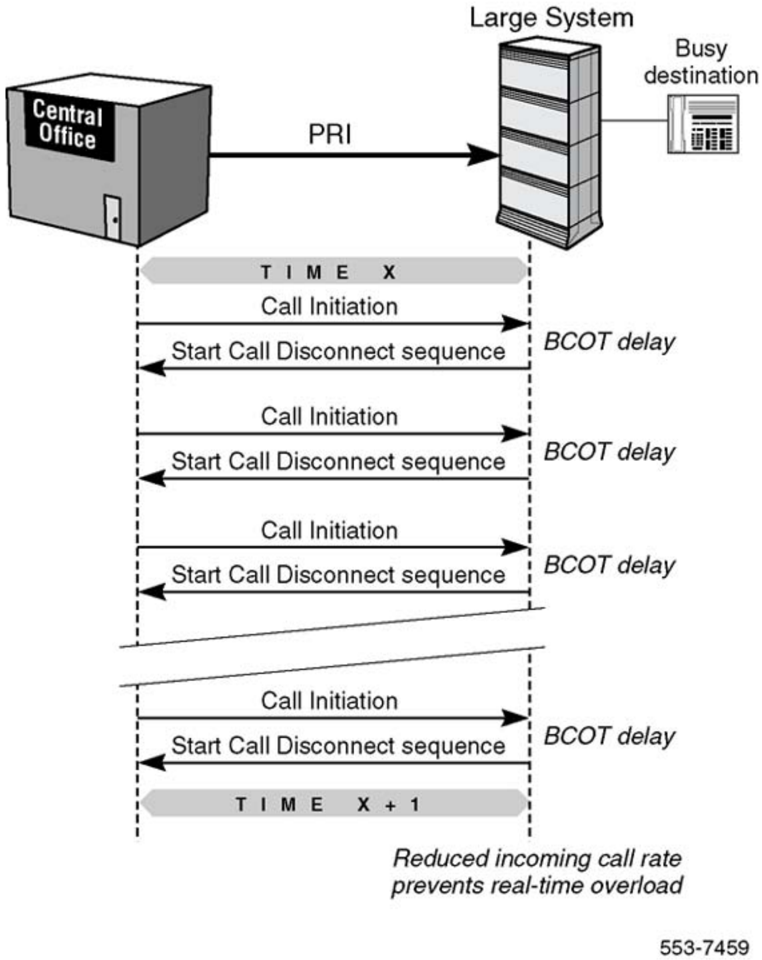

Figure 8: After the B-channel Overload Control Timer has been applied

B-channel Overload Control Timer (BCOT) considerations

Consider the following pertaining to the application of the BCOT:

- When configuring the BCOT in LD 16, the recommended value to be used is 256 milliseconds (most problem scenarios can be solved with this value). Even though the maximum value of 4 seconds can be entered, it is suggested that this value not be used (since callers do not receive any audio feedback while the timer is running, they would receive only silence for four seconds).
- After a value is entered in response to the BCOT prompt in LD 16, the system rounds this value down by a multiple of 128 milliseconds, if necessary (the actual time delay is in the range of +0 to -128 milliseconds from the entered value). This value is printed before the next prompt is displayed.
- When a system is upgraded for the first time to a software release containing the BCOC feature, all BCOT timers are initialized to 0.

- By default, the BCOC feature is disabled on all routes (BCOT = 0). To activate the BCOC feature for a route, a service change on the BCOT timer is required using LD 16.
- A new Peg counts field for the "Total number of activations of BCOC activations for this route" is added to the TFC002 traffic report. This count will apply to all system routes that interface with DMS, Lucent, and NI-2 TR-1268 switches. For the Integrated Services Access (ISA) Call By Call Type and the NI-2 Call By Call Service Selection features, the peg counts are done against the service routes, and not a master route. The peg count for a master route will always be zero.

Refer to Table 141: TFC002 format for an ISA service route on page 225 for a sample traffic measurement report output.

Traffic measurement output

For non-ISA and non-NI-2 Call By Call service routes, the TFC002 traffic measurement report has been updated for the BCOC feature as shown by Table 140: TFC002 format for a non-ISA and non-NI-2 Call By Call routes on page 225.

Table 140: TFC002 format for a non-ISA and non-NI-2 Call By Call routes

System ID	TFC002
...	
Route Number	Route Type
Total number of trunks configured	Total number of trunks working
...	Total number of incoming calls
...	Total number of outgoing calls
...	...
Total number of activations of BCOC for this route	

For ISA and NI-2 Call By Call service routes, the TFC002 traffic measurement report has been updated for the BCOC feature as shown by Table 141: TFC002 format for an ISA service route on page 225.

Table 141: TFC002 format for an ISA service route

System ID	TFC002
...	
Route Number	Route Type
Total number of trunks configured	Total number of trunks working
...	Total number of incoming calls
...	Total number of outgoing calls

Total incoming calls on the service route	Total outgoing calls on the service route
Total number of activations of BCOC for this route	

Operating parameters

This feature applies only to the following ISDN PRI to North American Central Office connectivities:

- DMS-100
- DMS-250
- SL-100
- Lucent 4ESS
- Lucent 5ESS and
- National ISDN-2 (NI-2) TR-1268

This feature supports both circuit switched voice and data calls.

This feature does not distinguish between normal busy conditions and overload busy conditions, since its functionality depends on the BCOT value configured at the route level. All busy calls on the route receive the same BCOT treatment.

This feature does not support Virtual Network Services trunks.

This feature is activated for the following call types:

- incoming calls presented on a busy phone on a node interfacing directly to a Central Office (see Figure 9: Busy phone is a on node interfacing directly to a Central Office on page 227 on Figure 9: Busy phone is a on node interfacing directly to a Central Office on page 227), or incoming calls from a Central Office being tandemed to a private network and presented on a busy phone (see Figure 10: Busy phone is a on node tandemed to a Central Office on page 227 on Figure 10: Busy phone is a on node tandemed to a Central Office on page 227)
- incoming calls presented on an ACD DN whose ACD queue has reached its maximum limit
- incoming calls presented on a Controlled Directory Number (CDN), and being released by Customer Controlled Routing (CCR) due to a "User Busy" cause
- incoming calls that are tandemed to a route whose trunk members are all busy

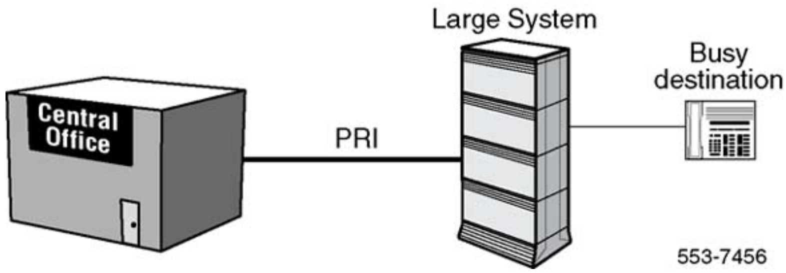

Figure 9: Busy phone is a on node interfacing directly to a Central Office

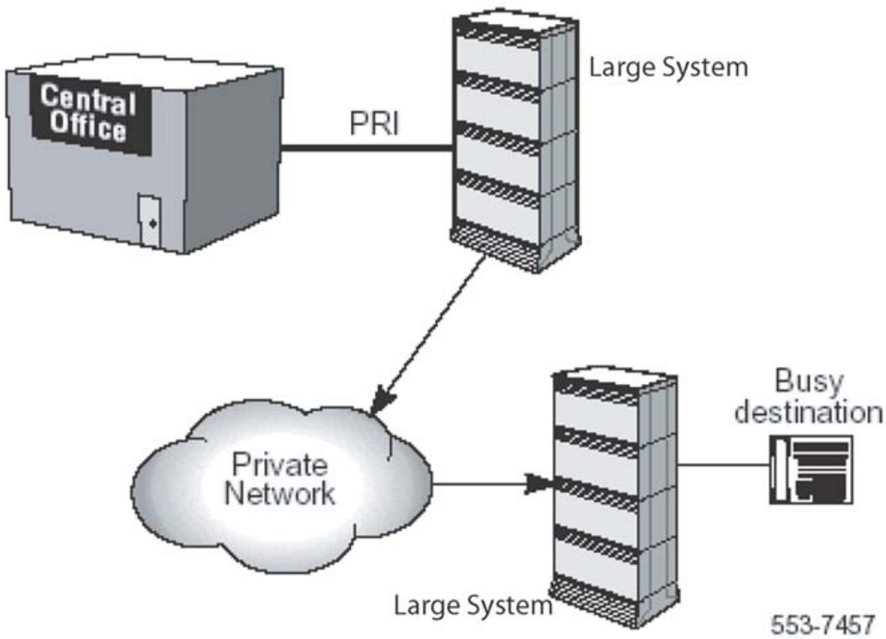

Figure 10: Busy phone is a on node tandemed to a Central Office

Feature interactions

Automatic Call Distribution

ACD allows a large volume of calls to be made to the same DN (called the ACD DN), and answered by a group of phones (called ACD agent positions). Incoming calls are distributed so that the agent that has been idle the longest receives the first call.

Whenever an incoming ACD call gets released immediately due to a "User Busy" cause, the system starts the B-channel Overload Control timer and sends a "Call Proceeding" message rather than releasing the call immediately (the amount of time, in milliseconds, that the B-channel remains seized depends on the value entered for the timer in LD 16). After the timer expires, the normal disconnect sequence will take place.

Call Interflow

Call Interflow allows an ACD supervisor to redirect excess traffic to an Interflow DN. All Interflow calls that are to be given busytone will be affected by the B-channel Overload Control feature, in that a delay will be imposed on these calls before the caller receives the busytone.

Enhanced ACD Routing

Enhanced ACD Routing allows different delay treatments to be given to ACD calls from different sources but to the same ACDN DN. Enhanced ACD Routing uses a Control DN, which does not have agent positions but rather specifies a destination default ACD DN to which incoming calls are directed. Multiple Control DNs can place calls into the same ACD queue, so different treatment can be given to these calls (the treatment given to a call is determined by the parameters defined for the Control DN, and not the ACD queue).

The Control DN has a limit to the number of unanswered calls that it can have at its default DN. Once the limit is exceeded, new calls are given busytone and Central Office calls are placed in the ACD queue. These calls that are to receive busytone will be affected by the B-channel Overload Control feature, in that a delay will be imposed on these calls before the caller receives the busytone.

Secondary DN Call Blocking

Secondary DN Call Blocking blocks new incoming ACD calls to the Secondary DN of an ACD agent, so that the agent can handle a current call without interruption. The calls to the Secondary DN receive busytone. These calls that are to receive busytone will be affected by the B-channel Overload Control feature, in that a delay will be imposed on these calls before the caller receives the busytone.

Supervisor Control of Queue Size

Supervisor Control of Queue Size allows ACD DNs busytone to be given to selected call types. These calls that are to receive busytone will be affected by the B-channel Overload Control feature, in that a delay will be imposed on these calls before the caller receives the busytone.

Integrated Services Access

Integrated Services Access (ISA) allows multiple routes to share the same the common pool of B-channels for connectivity between a system and a Central Office PBX such as the DMS-100 and the DMS-250. Unlike dedicated routes, which require each service route to have its own trunks of the same trunk type, ISA trunks are shared among many service routes, which can carry calls of different types that can change on a per call basis.

The B-channel Overload Control feature interacts with ISA in that all incoming calls over an ISA route that are released because of a "User Busy" cause, will be delayed as defined by the B-channel Overload Control timer.

NI-2 Call by Call Service Selection

NI-2 Call By Call Service Selection allows multiple services to share the same the common pool of B-channels for an NI-2 TR-1268 PRI interface. Dedicated routes are not required. The service types can be assigned on a per call basis.

Intercept Treatment

All calls that are released by Intercept treatment due to a "User Busy" cause will be delayed as defined by the B-channel Overload Control timer.

Remote Virtual Queuing

Since Remote Virtual Queuing interworks over a system to DMS interface, the B-channel Overload Control feature imposes a delay on the automatic retry capability of RVQ, which is used when congestion is encountered due to no idle trunks being available.

Auxiliary Products

All calls that interact with the auxiliary products Customer Controlled Routing (CCR), Integrated Call Center Manager (ICCM), and Meridian Link, an that are being released due to "User Busy" cause, are delayed by B-channel Overload Control until its timer expires.

Attendant Blocking of DN

This feature enables the attendant to block calls from being made to a DN while an external call request from that DN is being processed. The blocked calls receive busytone.

The B-channel Overload Control feature interacts with Attendant Blocking of DN in a system to Central Office interface, by imposing a delay on any call coming in from a CO that terminates on a blocked DN and receives busytone treatment due to "User Busy" cause.

Call Connection Restriction

This feature imposes restrictions on a caller's access to the public network, private network, and services and features. If any restriction is detected when a call is attempted, the call is denied and intercept treatment as defined in the Customer Data Block is applied. If the intercept treatment results in the call being released due to a "User Busy" cause, the B-channel Overload Control feature imposes a delay on the release.

Meridian Network Services Drop Back Busy and Off Hook Queue

The B-channel Overload Control feature does not affect the operation of the Drop Back Busy capability. If Off Hook Queuing is active, B-channel Overload Control will not be activated.

Network Individual Do Not Disturb

This feature allows extends the functionality of the Individual Do Not Disturb feature to a network environment. If a DN is in the Do Not Disturb mode, calls can be made from it, but incoming calls to it would receive Intercept treatment. If the treatment is busytone, the B-channel Overload Control feature imposes a delay on the release of these calls.

Trunk Barring

This feature allows customers the option of denying certain types of trunk-to-trunk connections. Attempted calls over these trunks would receive Intercept treatment. If the treatment is busytone, the B-channel Overload Control feature imposes a delay on the release of these calls.

Feature packaging

This feature requires the following packages:

- Integrated Services Digital Network (ISDN) package 145
- 1.5 Mbit Primary Rate Access (PRA) package 146

Feature implementation

Table 142: LD 16: Configure the B-channel Overload Control timer in response to the BCOT prompt.

Prompt	Response	Description
REQ	NEW	Add new data.
	CHG	Change existing data.
TYPE	RDB	Route Data Block.
...		
ISDN	YES	Integrated Services Digital Network.
...		
- IFC		Supported interface types.
	D100 D250 S100 ESS4 ESS5 NI2	D100 = DMS-100 D250 = DMS-250 S100 = SL100 ESS4 = Lucent 4ESS ESS5 = Lucent E5SS NI2 = National ISDN-2
...		
- - BCOT	(0)-4000	The value for the B-channel Control timer, in milliseconds. This value indicates the delay that the system imposes on a B-channel before starting the disconnect sequence. A value of 256 is recommended. The entered value is rounded down to multiples of 128. After a value has been entered and rounded down (if necessary), the (rounded down) value is autoprinted on the screen before the next prompt is displayed. Refer to B-channel Overload Control Timer (BCOT) considerations on page 224 for more information. **Note:** In the case of Integrated Services Access (ISA) Call By Call Type and NI-2 Call By Call Service Selection, BCOT is prompted only for Service Routes.

Note:

You can print the BCOT timer value that was entered in LD 16 by using LD 21.

Feature operation

No specific operating procedures are required to use this feature.

Chapter 20: Break-in features

Contents

This section contains information on the following topics for Avaya Communication Server 1000 (Avaya CS 1000):

Feature description

The Break-in (BKI) feature allows an attendant to interrupt an established call in order to connect an important incoming call to one of the established parties. For a description of the basic Break-in feature, refer to *Avaya Features and Services Fundamentals, NN43001-106*.

The following sections describe Break-in related features, as applied to an ISDN PRI environment.

Break-in busy indication and prevention

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

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

Break-in busy indication

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

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

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

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

Note:
The BBIN and EBIN options work only when the busy party is a PBX phone or a Digital phone.

Break-in prevention

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

Break in with secrecy

The Break In With Secrecy (BKIS) feature enhances the operation of the Break In (BKI) feature. With BKIS, when the break-in conference is established between the attendant and the desired and undesired parties of a call, the attendant can press the BKI key again to exclude the undesired party and talk directly to the desired party. Once the undesired party is excluded, intrusion tone is no longer provided.

BKIS applies to both predial and post-dial Break-in operations. In a post-dial situation, the attendant dials the desired party before pressing the BKI key. Whereas in a predial case, the attendant presses the BKI key prior to dialing the digits of the desired party.

BKIS operates in both stand-alone and within an Meridian Customer Defined Network (MCDN) Integrated Services Digital Network (ISDN) environment. In a MCDN ISDN environment BKIS is an enhancement of Network Attendant Service (NAS) Break-in (BKI), described in the NAS description found later in this document.

Operating parameters

The operating parameters which apply to the Break-In feature and the Network Attendant Service Break-In feature also apply to this feature.

Indication of station status on a call to a busy station is available only on Attendant Consoles.

Break-in with secrecy

The same feature requirements apply as for the Break-in feature. Within an ISDN environment:

- All conditions for NAS Break-in must be met.
- In order for this feature to operate correctly over the network, all nodes connected to the attendant must have Break-in software equipped.

Feature interactions

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

With the Break-In to Line Lockout Set product improvement, the appropriate busy indication is supported. If the Break-In to Line Lockout Set Denied option (option BLD) was set for the customer, attempts at break-in results in Temporarily Denied—1 status. If the Break-In to Line Lockout Set Allowed option (option BLA) was set then break-in proceeds normally. If the Attendant Busy Display Allowed option of the First-Second Degree Busy Indication feature is chosen in LD 15 (option ABDA), display of digits could be changed. When the attendant dials a second-degree busy station, the attendant digit display shows —0. In combination with the Break In Indication and Prevention feature, the display can show —0——— or —0 followed by another dash and a mode digit.

Break-in with secrecy

Other than the interactions described below, the Feature interactions are the same as for the Break-in and NAS Break-in features.

Break-in to Enquiry Calls

Break-in with Secrecy interacts with Break-in to Enquiry Calls (BIEC) when the desired party has gone on-hook leaving an undesired party off-hook and excluded. BIEC has enhanced the existing BKI feature by giving overflow tone to the undesired party if it is a 500 type phone (irrespective of whether the undesired party was involved in an enquiry call or not). BKIS does not change this operation for non-BKIS calls.

BKIS has a choice of options to be given to the undesired party if the desired party goes on-hook while the undesired party is excluded. These are taken from the AOCS options in the Customer Data Block. These options are not given to the undesired party if the undesired party has a call on hold, this only applies to analog (500/2500 type) phones. The BIEC treatment of giving overflow tone is done instead so that the undesired party can be reconnected to the held party.

Therefore, it is quite possible for PBX-type phones and trunks to get different treatment depending on the circumstances.

The following is a list of treatments for different circumstances:

- Existing BKI BIEC disconnects undesired parties when the desired party goes on-hook, except for analog (500/2500 type) phones where overflow is given. Therefore Meridian digital phones and trunks are disconnected.
- BKIS will give either overflow, transfer to attendant, or disconnect treatment to analog (500/2500 type) phones or trunks. Meridian digital phones are disconnected.

Multi-Party Operation

For Multi-Party Operation (MPO), the operation of features, such as going on-hook and releasing from a call, during the BKIS conference between the attendant and the desired party, takes precedence over MPO operations for those cases where the treatment differs from that defined by the customer.

All network nodes must have MPO software, with identical Multiple-party Operation (MPO) options. Otherwise, MPO options in the desired party's node have precedence.

Pertaining to MPO options, if the undesired party is not located on the same node as the desired party, the undesired party is considered as an external party on the desired party node.

Network Attendant Service (NAS)

The BKIS feature operates in a networking environment with regard to the NAS Break-in Feature operations and limitations.

Secrecy

The source and destination parties cannot be joined together on the attendants conference bridge if BKIS is active. This is consistent with the existing Break-in feature.

Music

During secrecy, if there is only one undesired party in the conference, music is not provided to this party when excluded. However, intrusion tone is given to this party.

Display

In all cases, when displays are equipped, the information displayed is consistent with current operation (i.e., when connected to only one party, the display shows the number and name, if equipped and configured, of that party, and when connected to more than one party, the display is blank).

Feature packaging

This feature requires the Attendant Break-in/Trunk Offer (BKI) package 127.

For an ISDN network environment, the following software packages are required:

- Integrated Services Digital Network (ISDN) package 145
- Network Attendant Service (NAS) package 159 and
- Integrated Services Digital Network Supplementary Features (ISDNS) package 161

Break-in with secrecy

Break-in with Secrecy requires Attendant Break-in (BKI) package 127.

For an MCDN ISDN environment, the following packages are required:

- ISDN basic (ISDN) package 145
- Network Attendant Service (NAS) package 159 and
- ISDN Supplementary Features (ISDNS) package 161

Multi-Party Operations (MPO) package 141 is optional, but if used in an MCDN ISDN environment all nodes must be equipped with the MPO package.

Feature implementation

Task summary list

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

1. Table 143: LD 15: Configure Customer Data Block for Break-In features. on page 238
2. Table 144: LD 12: Modify or create Attendant Console data blocks. on page 238
3. Table 145: LD 15: Modify Multi-Party Operations data in Customer Data Block, if MPO package 141 is equipped. on page 239
4. Table 146: LD 20: When the Attendant Console data block is printed the BKI key information is output. on page 239
5. Table 147: LD 21: When the Customer data block is printed all Multi-Party Operations options are output. on page 240

Table 143: LD 15: Configure Customer Data Block for Break-In features.

Prompt	Response	Description
REQ	CHG	Change existing data.
TYPE	FTR	Features and options.
CUST		Customer number
	0-99	Range for Large System , Media Gateway 1000B, and CS 1000E system.
...		
OPT	(BIXA)	Break-In to external call Allowed.
	BIXD	Break-In to external call Denied.
	(BIND)	Break-In Indication Denied.
	BBIN	Basic Break-In Indication.
	EBIN	Extended Break-In Indication.

Break-in with secrecy

Table 144: LD 12: Modify or create Attendant Console data blocks.

Prompt	Response	Description
REQ	NEW	Add new data.
	CHG	Change existing data.
TYPE	aaaa	Type of console
...		
AADN	...	Attendant Alternate Directory Number
KEY	0-19 BKI	Key number assigned to Break-in.

Table 145: LD 15: Modify Multi-Party Operations data in Customer Data Block, if MPO package 141 is equipped.

Prompt	Response	Description
REQ	NEW	Add new data.
	CHG	Change existing data.
TYPE	MPO	Multi-Party Operations.
CUST		Customer number
	0-99	Range for Large System , Media Gateway 1000B, and CS 1000E system.
...		
- FMOP	(NO) YES	Flexible Misoperation Options.
- - RGNA	xxx yyy	Ringing No Answer treatment
- - AOCS	xxxyyy AAR AAR ATN(ATN) DAR DAR (DIS) DIS OVF OVF STD STD	All Other Cases where: the first field (xxx) defines treatment for internal calls and the second field (yyy) defines the treatment for external calls.
		AAR – The transferring station is re-rung. If the transferring station fails to answer, the transferred station is routed to the attendant. ATN – Attendant DAR – The transferring station is re-rung. If the transferring station fails to answer, the transferred station is disconnected. DIS – Disconnect OVF – Overflow STD – Standard
- - RCY1	1-(6)-15	Number of Cycles of Re-ringing before forwarding or disconnecting
- - RCY2	1-(4)-15	Number of Cycles of Ringing before forwarding to transferring station
- - ACNS	(NO) ALL EXT	Attendant Clearing during Night Service
- - RALL	(NO) YES	Mandatory recall is required prior to dialing control digits
- - CDTO	2-(14)	Control digit timeout; in multiples of two seconds

Table 146: LD 20: When the Attendant Console data block is printed the BKI key information is output.

Prompt	Response	Description
REQ	PRT	Print data block.

Prompt	Response	Description
TYPE	xxxx	Type of data block.

Table 147: LD 21: When the Customer data block is printed all Multi-Party Operations options are output.

Prompt	Response	Description
REQ	PRT	Print data block.
TYPE	CDB	Type of data block: Customer Data Block.
CUST		Customer number
	0-99	Range for Large System , Media Gateway 1000B, and CS 1000E system.

Feature operation

No specific operating procedures are required to use this feature.

Break-in with secrecy

Break-in to two-party connection

The following sections describe a post-dial Break-in. For predial Break-in, Break-in is done on the Source of the attendant and there is no party A calling the attendant, but the BKIS operation is identical.

The scenario is as follows:

Party A calls the attendant, the attendant calls party B who is talking to party C. The attendant presses the BKI key to intrude into the conversation. At this point, the attendant and both parties B and C are in conversation with intrusion tone provided, while party A is on HOLD (with music if EMUS, package 119, is equipped).

Break-in Allowed

This situation will arise when party A is an external call and Camp-on or Call Waiting is possible at the wanted station B. At this point, the BKI, Exclude Source (EXCL SRC) and Exclude Destination (EXCL DEST) indicators are active (lamps are lit or Liquid Crystal Display [LCD] is on), and the following actions can occur:

Request the unwanted party to terminate

The attendant can request the unwanted party to terminate. A positive response will terminate the conference that included the attendant and intrusion tone. This is a current BKI operation.

Request the wanted party to terminate

The attendant can request the wanted party to terminate the call. The party disconnects, terminating the BKI conference. This is a current BKI operation.

Attendant presses Release Destination key

The attendant can press the RLS DEST key to release the call. This action terminates the conference and the original call is reestablished as it was prior to Break-in. The Source party A is connected to the Attendant. This is a current BKI operation.

Attendant presses Exclude Destination key

The attendant can press the EXCL DEST key to return to the incoming call. The intruded parties keep receiving the intrusion tone. This is a current BKI operation.

Attendant presses Release key

The attendant can press the Release (RLS) key to apply Camp-on. This is a current BKI operation.

Attendant presses Break-in key again

The BKIS feature allows the attendant to press the BKI key again in order to exclude the undesired party C (who continues to hear intrusion tone) and to talk directly to the desired party B without intrusion tone. The BKI indicator, which was active, flashes at 60 impulses per minute (ipm).

Note:

Pressing of the BKI key a second time with the Break-in conference excluded will not activate secrecy (i.e., if the Break-in conference is on the destination but the attendant is talking on the source, secrecy cannot be activated).

From this point, the following attendant operations can occur:

Attendant actions

Break-in

The attendant presses the flashing BKI key. In this case, party C, which was excluded, is brought back into conversation with the attendant, party B, and intrusion tone. The BKI indicator reverts to an active state. The situation reverts to a normal BKI conference with intrusion tone.

In other words, the lit BKI key can be used to exclude the unwanted party from the BKI conference and the flashing BKI key can be used to reestablish the BKI conference (with intrusion tone).

Exclude Destination

The attendant presses the EXCL DEST key to return to the incoming call. The attendant is connected to the source party. The unwanted party B and the wanted party C are reconnected with intrusion tone. The EXCL SRC indicator is now off and the EXCL DEST lamp and the BKI indicators are active. The operation of the EXCL DEST key has the same effect as for a normal BKI conference situation, as described previously.

Release

The attendant presses the RLS key to apply Camp-on. If Camp-on or Call Waiting is available, parties B and C are reconnected and party A is released and either Camp-on or Call Waiting is applied to the wanted party A. The BKI indicator is off. If Camp-on or Call Waiting is not available, the operation of the RLS key causes secrecy to be turned off and the situation to go back to the Break-in conference with intrusion tone. The loop can only be released by pressing the RLS DEST key, leaving the source connected to the attendant. The operation of the RLS key has the same effect as for a normal BKI conference situation, as described previously.

Release Destination

The attendant presses the RLS DEST key. The BKI, EXCL SRC, and EXCL DEST indicators are off and party A is connected to the attendant. Party B (desired) and party C (excluded party) are reconnected.

Undesired party action

Party C (undesired party) goes on-hook and is disconnected. Then the BKI indicator goes off and the attendant treats the call as a normal two-party connection. The attendant is talking directly to party B (desired party) and can press the RLS key to extend the call.

Desired party action

At this point, if party B (controlling party) goes on-hook, the treatment depends upon the Customer Data Block (LD 15) Multi-party Operations (MPO) Flexible Misoperation Options (FMOP) All Other Cases (AOCS) settings if the undesired party is a trunk or 500-type phone and MPO package 141 is equipped. If the MPO package is not equipped, internal calls will be disconnected, while external calls will be rerouted to the attendant.

The following shows what happens to 500-type phones or trunks depending on the AOCS options:

AOCS set to AAR for party C

If AOCS is set to AAR for party C, then party C is routed to the attendant and party B is re-rung by the attendant. BKI indicator goes off and a simple call is set up between attendant and party B when B answers.

AOCS set to ATN for party C

If AOCS is set to ATN for party C, then party C is routed to the attendant while B is re-rung by the attendant. The BKI indicator goes off and the attendant hears ring back and the DEST indicator winks at 30 ipm. The attendant can extend the call as normal.

AOCS set to DAR for party C

If AOCS is set to DAR for party C, then party C is disconnected and party B is re-rung by the attendant. The BKI indicator goes off and when B answers a simple call exists between the attendant and party B.

AOCS set to DIS for party C

If AOCS is set to DIS for party C, then C is disconnected and party B is re-rung by the attendant. The BKI indicator goes off and the attendant hears ringback and the DEST indicator winks at 30 ipm. The attendant can then extend the call as normal.

AOCS set to OVF for party C

If AOCS is set to OVF then overflow tone is given to party C and party B is re-rung by the attendant. The BKI indicator goes off, the attendant hears ringback, and the DEST indicator winks at 30 ipm. The attendant can then extend the call as normal.

AOCS set to STD for party C

If AOCS is set to STD for party C, the treatment is the same as default for the AOCS option. If party C is internal, then DIS option applies to party C, and if party C is external, then ATN option applies to party C.

Break-in ÔConsultation Only

This console state indicates that the attendant has been allowed to Break-in to the desired party's call; however, the attendant will not be able to extend the originating call. This situation occurs under the following conditions:

1. An internal call is on the source port of the Attendant Console.
2. The attendant originated the call. In this case, the source indicator will be used instead of the destination indicator to provide status information (predial situation).
3. An external call is on the source and neither Camp-on nor Call Waiting is possible at the wanted station (i.e., Camp-on or Call Waiting not possible or the station already has a call camped on).
4. The desired station is busy with Call Forward active and the attendant initiated a predial Break-in.

The BKI and the EXCL SRC indicators are active, the DEST indicator is flashing. At this point, the attendant is not allowed to press the RLS key to extend the originating call, party A. The operation of the RLS key is ignored. This is a current BKI operation.

The attendant can press the BKI key to exclude party C and talk directly to party B, as described under the Attendant actions section. The BKI and DEST indicators are flashing. While in this state, the attendant is not allowed to press the RLS key to extend the originating call, party A. The operation of the RLS key causes the secrecy to be turned off and the situation to revert to a Break-in conference. The other operations described in the Attendant actions section are available.

Break-in to a conference

Party A (either internal or external) calls the attendant, the attendant calls party B who is involved in a conference call with parties C and D. The attendant presses the BKI key to intrude into the conversation. At this point, the attendant, party B and all the original conferees are in conversation with intrusion tone provided, while party A is on HOLD. The BKI and EXCL SRC indicators are active. The DEST indicator is flashing and the BKI status is ÔConsultation Only'.

At this point, the attendant can press the BKI key to talk directly to party B without intrusion tone. The Break-in indicator flashes at 60 ipm. The original conference is excluded from party

B (the other parties in the conference remain connected without intrusion tone). Party A is still excluded on the attendant loop and the attendant is talking directly to party B without intrusion tone.

While in this state, the following situations can occur:

Attendant actions

Break-in

The attendant can press the flashing BKI key. The original conference is reestablished with intrusion tone. The BKI indicator reverts to active.

Exclude Destination

The attendant can press the EXCL DEST key to return to the incoming call. The original conference is reestablished and party A is connected to the attendant.

Release

The attendant is not allowed to extend the original call to the wanted party B by pressing the RLS key. The operation of the RLS key causes the secrecy to be turned off and the situation reverts to a Break-in conference.

Release Destination

The attendant can press the RLS DEST key. The BKI, EXCL SRC and EXCL DEST indicators are off and party A is reconnected to the attendant. The original conference (B, C, and D) is reestablished.

Undesired party action

All but one of the conferees (C or D) go on-hook. The last undesired party will start getting the intrusion tone once again. The situation reverts to the previously described operation (See Undesired party action on page 242).

Desired party action

At this point, if party B goes on-hook, party B is re-rung by the attendant and the conferees are left in conference without party B and without intrusion tone. The BKI indicator goes off, the attendant hears ringback tone, and the DEST indicator winks at 30 ipm. The attendant can extend the call as normal.

Table 148: Summary of possible Break-in situations and indications

State	Operation	SRC or DEST Indicator	Break-in Indicator	Tone
Allowed	a) post-dial	ACTIVE	ACTIVE	intrusion
	predial	ACTIVE	ACTIVE	busy
	b) post-dial	ACTIVE	OFF	none
	predial	ACTIVE	ACTIVE->OFF	override
Consultation Only	a) post-dial	FLASH	ACTIVE	intrusion
	b) predial	FLASH	ACTIVE	busy

State	Operation	SRC or DEST Indicator	Break-in Indicator	Tone
Temporarily Denied 1		FLASH	FLASH	busy override if override is involved
Temporarily Denied 2	a) post-dial only	FLASH	WINK	overflow
	b) predial	FLASH	WINK	busy or ring back
	(then post-dial)	FLASH	WINK	intrusion
Denied		FLASH	OFF	overflow
Break-in	a) post-dial	WINK	OFF	ringback
Ignored station is rung	b) Predial	WINK	OFF	ringback
Invalid	post-dial or predial	OFF	OFF	overflow
Break-in with Secrecy	after post-dial or predial, active BKI key is pressed	ACTIVE or FLASH	FLASH	no tone

Table 149: Summary of possible Break-in situations and actions

Condition of called DN	Action
Established call, Call Waiting or Camp-on allowed, Multiple Appearance DN. Lockout (if not denied).	Break-in allowed, connection established. Connection is made.
Attendant dialing on SRC, internal call on SRC, CWT or Camp-on not available, desired party in conference, Call Forward active on phone.	Connection is made for the attendant only.
Tones, ringing, dialing, blocking, Override, Camp-on, Hold, talking to another attendant, Call Transfer, WTD on undesired party.	Release DEST, wait and repeat.
Make Set Busy, Do not disturb.	Predialing operation possible.
Warning tone denied on desired party, maintenance busy.	Break-in impossible.
Station is idle.	Station is rung, station not affected.
Invalid numbers.	Break-in impossible.

Condition of called DN	Action
The previous status was 'Allowed' or 'Consultation Only'. SRC or DEST indicator was active ('Allowed') or flashing ('Consultation Only').	Undesired party is excluded and the attendant is talking to the wanted party.

Chapter 21: 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
- 4ESS and 5ESS 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.

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:

Table 150: LD 17: Configure far-end BRI support.

Prompt	Response	Description
REQ	CHG	Change existing data.
TYPE	CFN	Configuration record

Prompt	Response	Description
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 151: 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 22: Business Network Express/ EuroISDN Call Diversion

Contents

This section contains information on the following topics for Avaya Communication Server 1000 (Avaya CS 1000):

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.

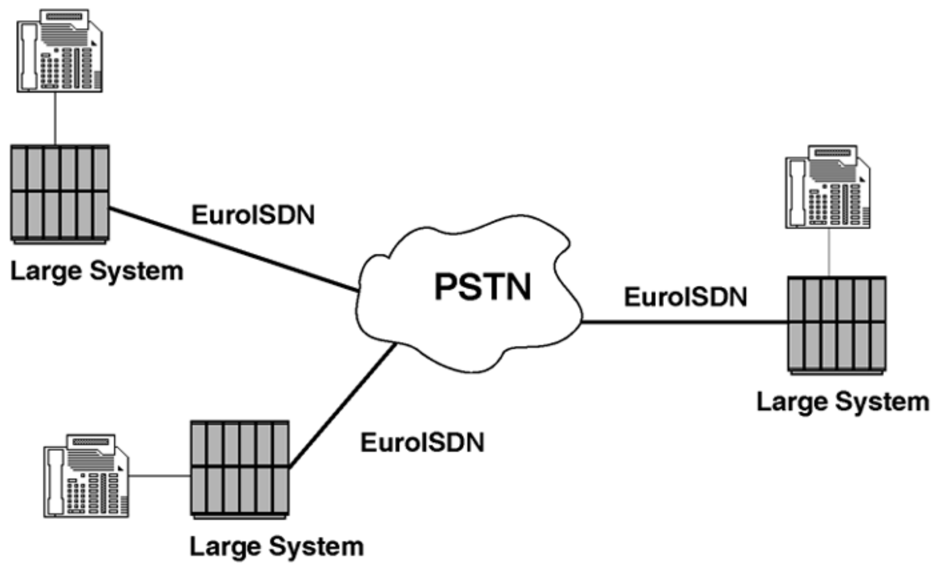

Figure 11: Example of a network where BNE is useful

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 155: Correspondence between the ETSI reason for diversion names and the system features on page 269 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 12: Call Diversion environment on page 254 shows the component parts and terms used in the Call Diversion environment.

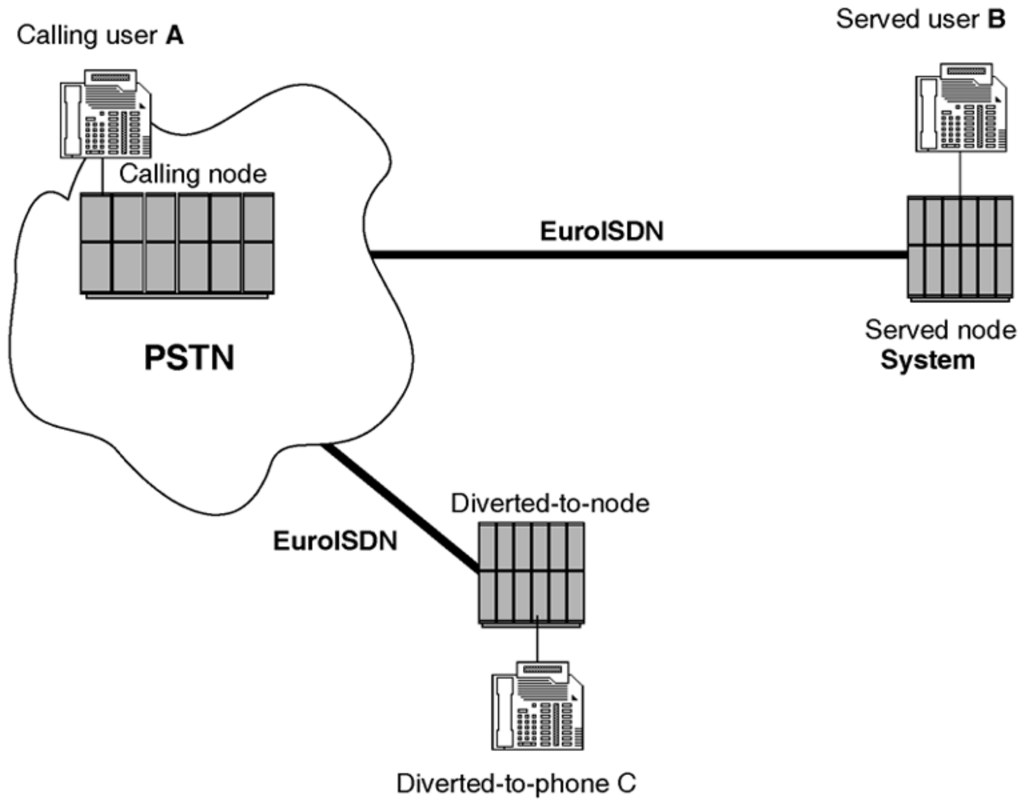

Figure 12: Call Diversion environment

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 152: Relationship between the served users Class of Service and the calling users notification on page 255 and Table 154: Relationship between the served users Class of Service and the notification on the final destination (diverted-to) phone on page 256 shows a summary of the relationships.

Table 152: Relationship between the served users Class of Service and the calling users notification

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 153: Examples of calling users display related to the served users Class of Service on page 256 provides examples of the display on the calling user's phone under different conditions.

Table 153: Examples of calling users display related to the served users Class of Service

Calling user's 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 154: Relationship between the served users 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.

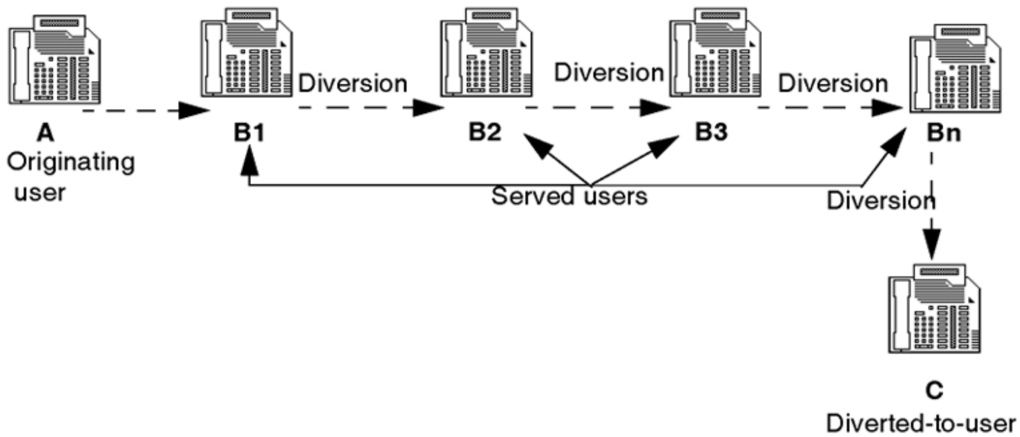

Figure 13: Example of multiple diversions

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 14: Rerouting takes place in the public network on page 258 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.

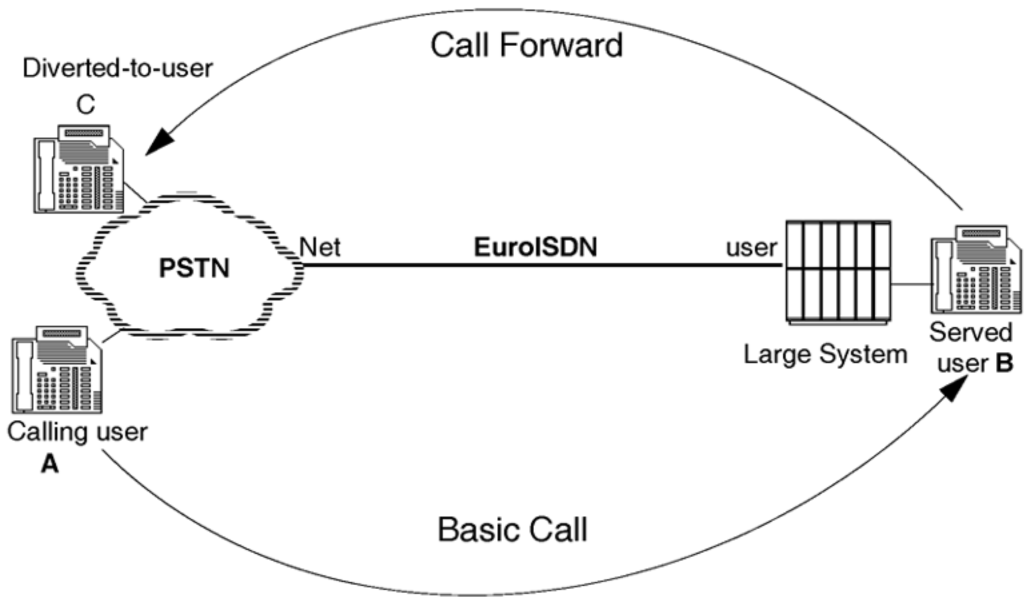

Figure 14: Rerouting takes place in the public network

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 152: Relationship between the served users Class of Service and the calling users notification on page 255).

A call from the public ISDN is diverted within the system

Figure 15: Call from the public ISDN is diverted within the system on page 259 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.

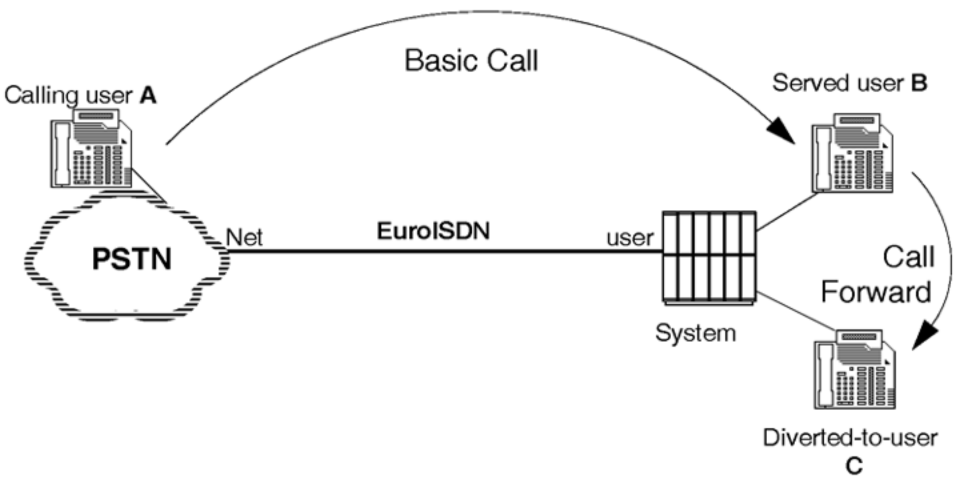

Figure 15: Call from the public ISDN is diverted within the system

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 152: Relationship between the served users Class of Service and the calling users notification on page 255)
- 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 16: The calling and served users are in the public network and the diverted-to-user is on the system on page 260 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.

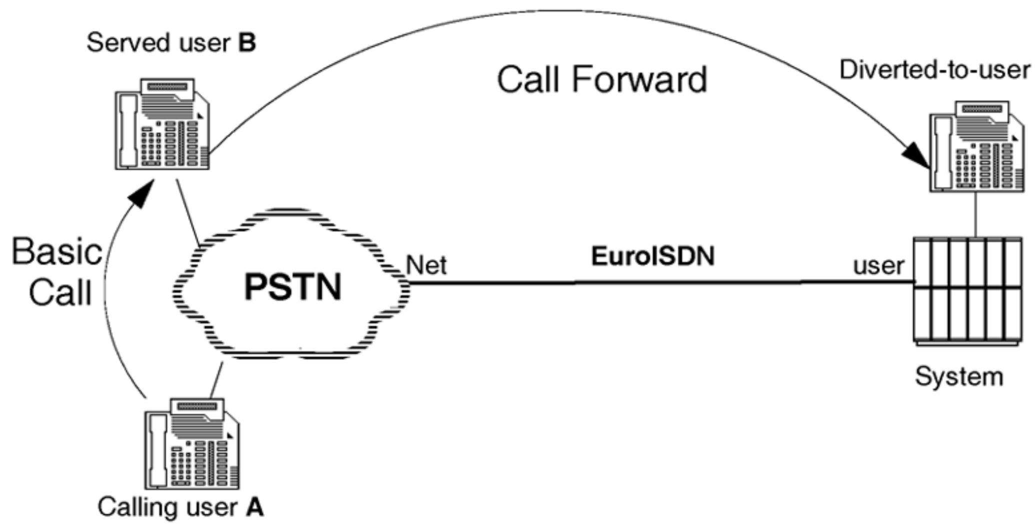

Figure 16: The calling and served users are in the public network and the diverted-to-user is on the system

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 17: The calling and served users are on the system and the diverted-to-user is in the public network on page 261 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.

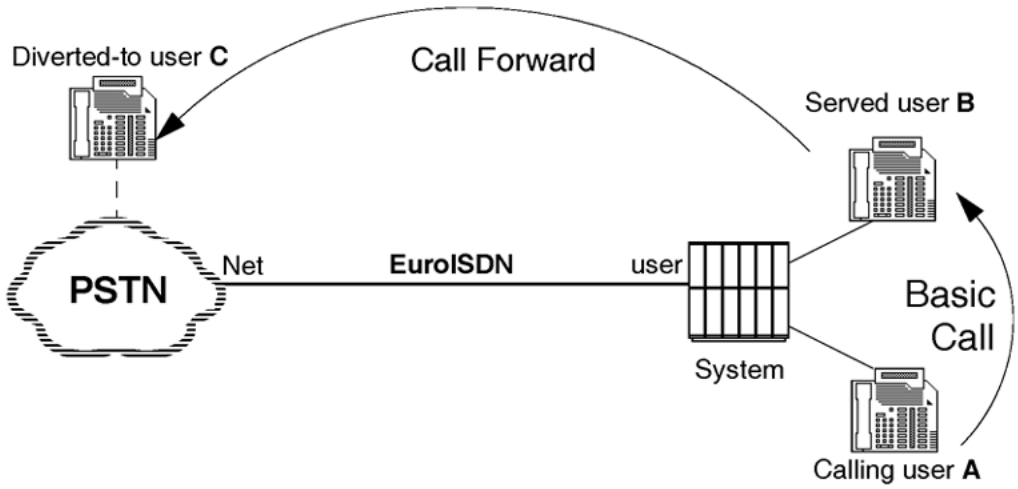

Figure 17: The calling and served users are on the system and the diverted-to-user is in the public network

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 154: Relationship between the served users Class of Service and the notification on the final destination (diverted-to) phone on page 256)
- originalCalledNr—the ISDN number of the first served user (depending on the served user's Class of Service; see Table 154: Relationship between the served users Class of Service and the notification on the final destination (diverted-to) phone on page 256)

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 18: The calling user is on the system and the served user and the diverted-to-users are in the public network on page 262 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.

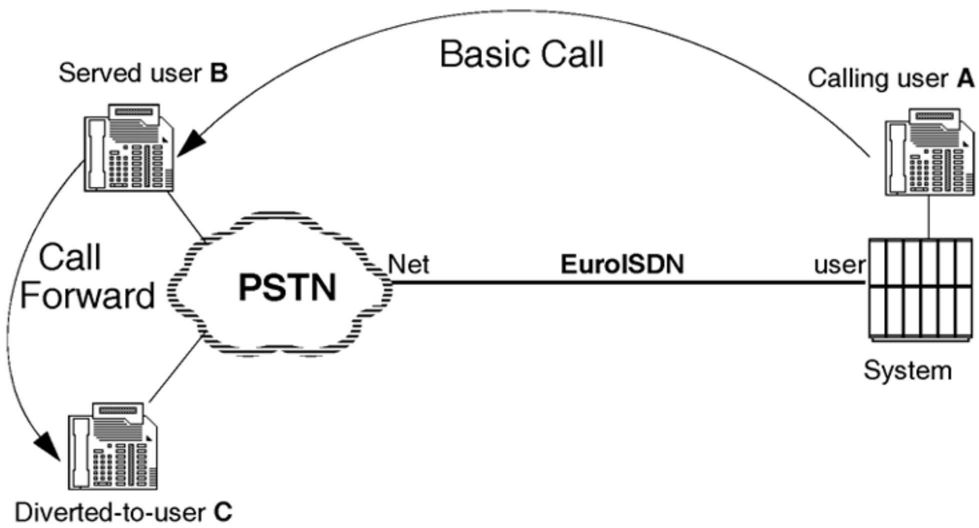

Figure 18: The calling user is on the system and the served user and the diverted-to-users are in the public network

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 19: Interworking with EuroISDN, QSIG, MCDN, or DPNSS on page 263):

- 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.

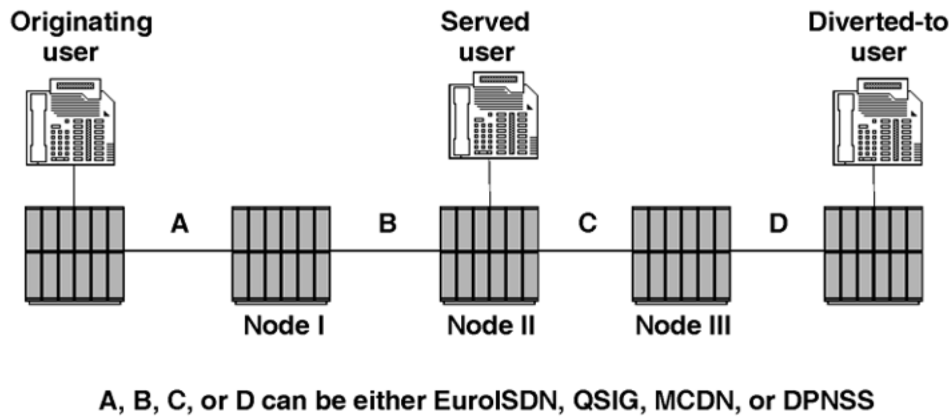

Figure 19: Interworking with EuroISDN, QSIG, MCDN, or DPNSS

Node I Gateways (A/B)

The following gateways can exist between the Originating user node and the Served user node (refer to Figure 19: Interworking with EuroISDN, QSIG, MCDN, or DPNSS on page 263):

- 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 19: Interworking with EuroISDN, QSIG, MCDN, or DPNSS on page 263):

- 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 19: Interworking with EuroISDN, QSIG, MCDN, or DPNSS on page 263):

- 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 20: Rerouting request received at the originating node. on page 267, Figure 21: Call rerouting request sent to the CO. on page 268, and Figure 22: Call rerouting request processed at the gateway node. on page 268).

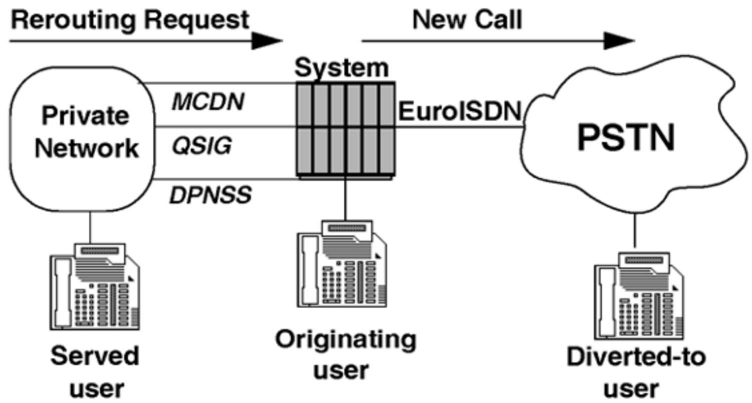

Figure 20: Rerouting request received at the originating node.

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.

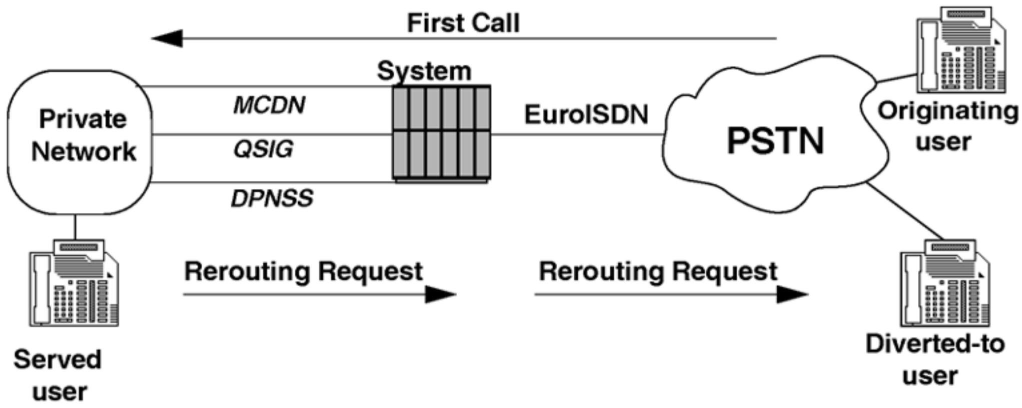

Figure 21: Call rerouting request sent to the CO.

If the diverted-to user is located within the private network, the request is processed by the gateway node.

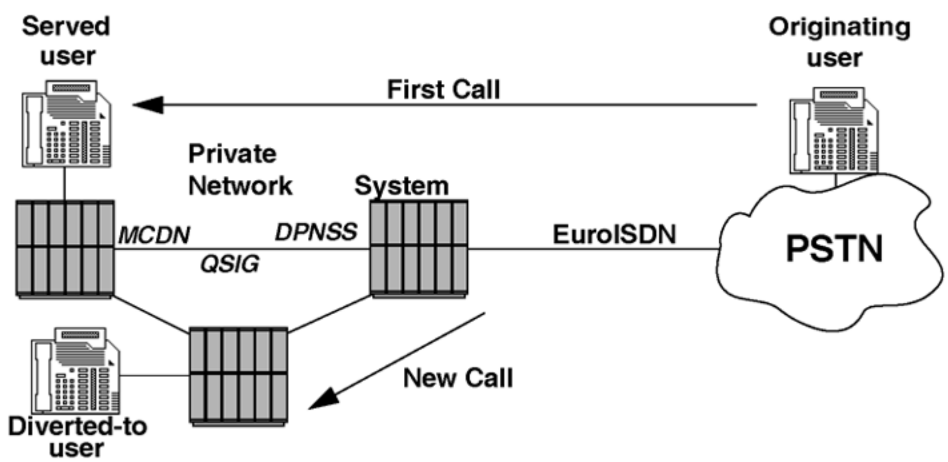

Figure 22: Call rerouting request processed at the gateway node.

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 155: Correspondence between the ETSI reason for diversion names and the system features on page 269 summarizes the correspondence between the ETSI reason for diversion supported by this feature, and the system equivalent features.

Table 155: 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 156: Relationship between the calling users display and the served users Class of Service on page 270 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 156: Relationship between the calling users display and the served users Class of Service

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 157: Relationship between the display on the diverted-to-phone and the served users Class of Service. on page 271 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 157: Relationship between the display on the diverted-to-phone and the served users Class of Service.

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 (CWTR) 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 Meridian Mail, 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.

Meridian Mail

A call diverted to Meridian Mail through EuroISDN can access Meridian Mail functionalities (such as message reception and mailbox interrogation) in the same way as a simple call to the mailbox.

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 158: 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 , Media Gateway 1000B, and CS 1000E system, where l = loop, s = shelf, c = card, u = unit.
DN	x..x	Directory Number.
...	...	
HUNT	x..x	Hunt DN.

Prompt	Response	Description
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.
	CFXA	Call Forward to external DN allowed.
	HTA	Hunting allowed.
	FNA	Call Forward No Answer allowed.
...	...	
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 159: 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 , Media Gateway 1000B, and CS 1000E system, where l = loop, s = shelf, c = card, 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.

Prompt	Response	Description
	(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.
	FNA	Call Forward No Answer allowed.
	CFTA	Call Forward by Call Type allowed.
...	...	
EFD	x..x	External Flexible Call Forward No Answer DN.
HUNT	x..x	Hunt DN.
EHT	x..x	External Hunt DN.

Table 160: 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		Digital Subscriber Loop.
	l s c dsl	Format for Large System , Media Gateway 1000B, and CS 1000E system.
...	...	
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 161: LD 27: Configure BRI phone for EuroISDN Call Diversion notification.

Prompt	Response	Description
REQ	CHG	Change

Prompt	Response	Description
TYPE	TSP	Terminal Service Profile.
DSL		Digital Subscriber Loop.
	l s c dsl	Format for Large System , Media Gateway 1000B, and CS 1000E system.
...	...	
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: no served user's number notification (when available) to the diverted-to-user.
	FNA	Call forward No Answer allowed.
SSRV _ ETSI		ETSI Supplemetary 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 162: 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.
...	...	

Prompt	Response	Description
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.
HUNT	aaaa (B)	Mnemonic for Hunting/Call Forward Busy display.

Table 163: 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 164: 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.

Prompt	Response	Description
...	...	
RCAP	DV3I	EuroISDN Call Diversion.

Feature operation

No specific operating procedures are required to use this feature.

Chapter 23: Business Network Express/ EuroISDN Explicit Call Transfer

Contents

This section contains information on the following topics for Avaya Communication Server 1000 (Avaya CS 1000):

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 23: Example of a BNE solution on page 284 for an example.

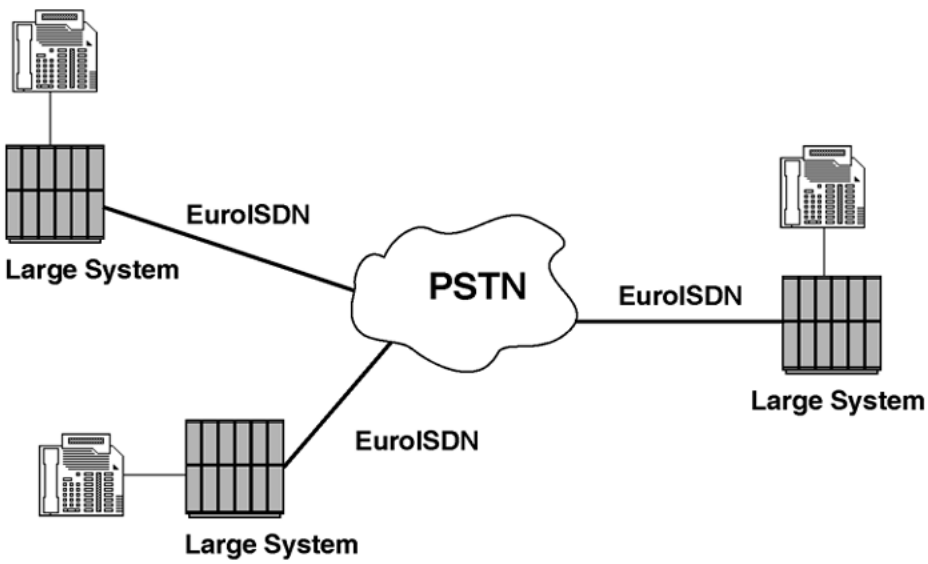

Figure 23: Example of a BNE solution

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 24: Served User A is the only one in the system private network (or stand alone system) on page 286 and Figure 25: Served user A is the only one in thesystem private network on page 286.

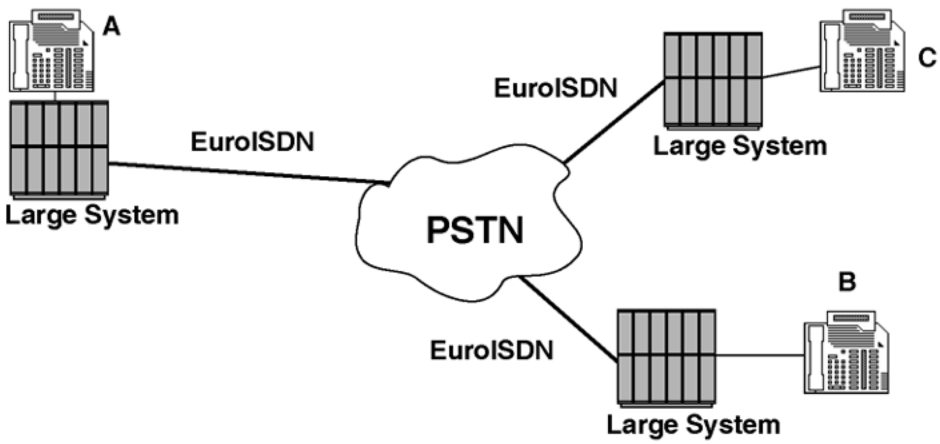

Figure 24: Served User A is the only one in the system private network (or stand alone system)

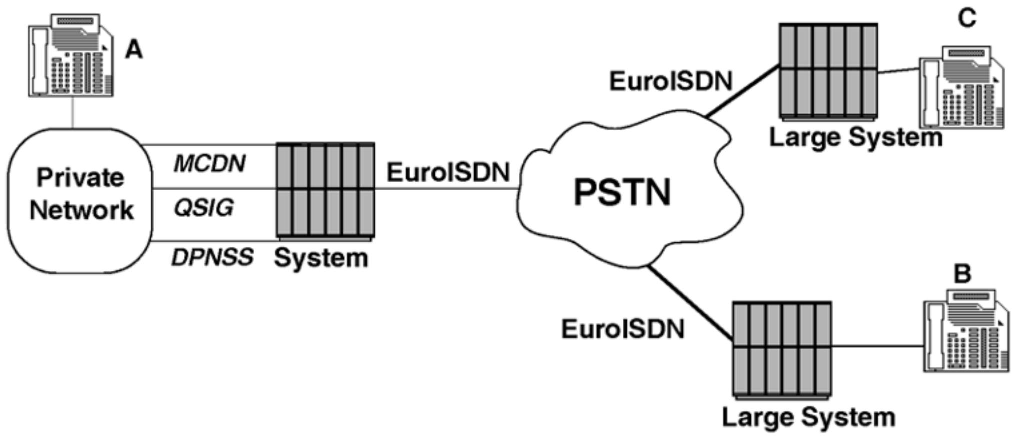

Figure 25: Served user A is the only one in thesystem private network

- 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 26: User A and user B or C are in the system private network on page 287 to Figure 28: User A and user B or C are in the system private network. on page 287. In a gateway connection, the system can be the gateway node (Figure 27: User A and user B or C are in the system private network on page 287), or the gateway node and the served node together (Figure 28: User A and user B or C are in the system private network. on page 287).

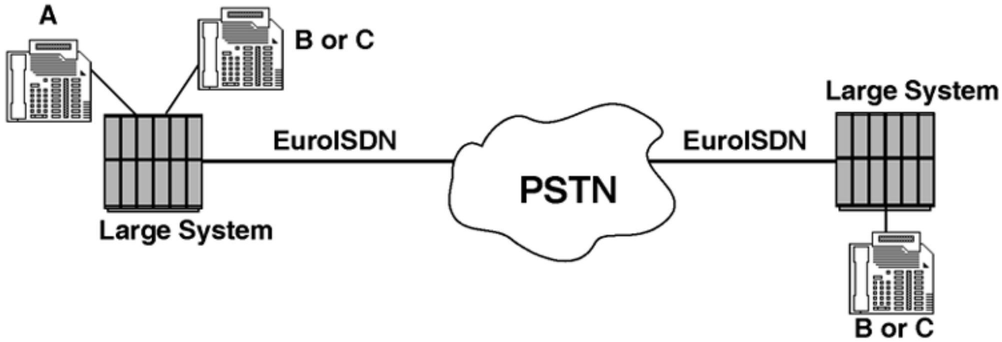

Figure 26: User A and user B or C are in the system private network

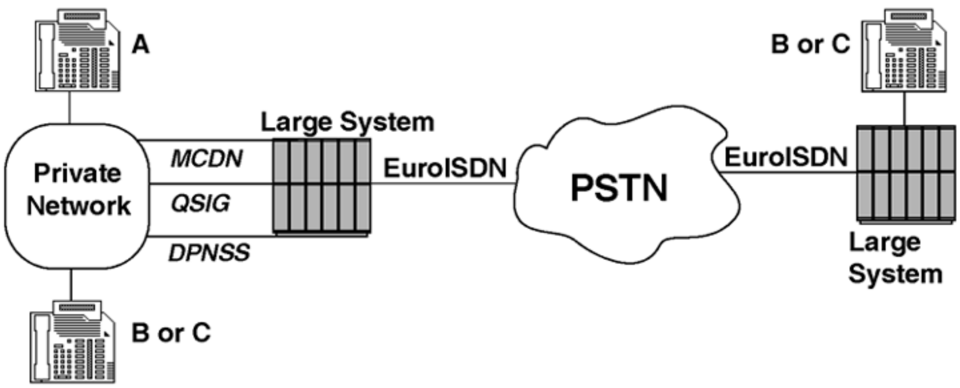

Figure 27: User A and user B or C are in the system private network

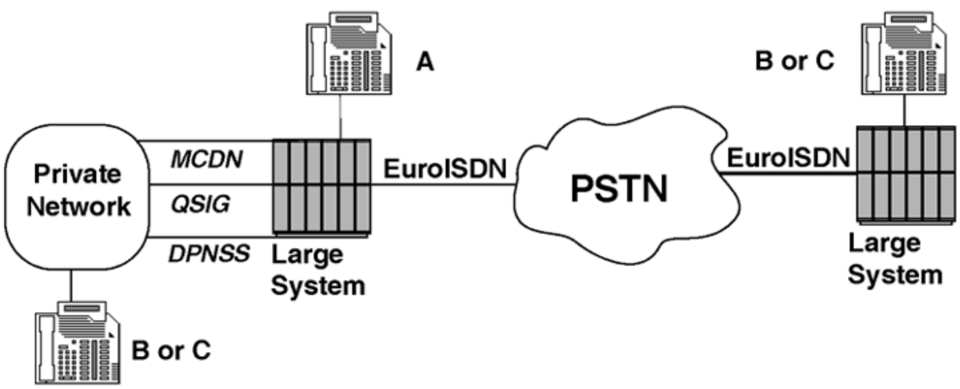

Figure 28: User A and user B or C are in the system private network.

- 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 29: Remote user B or C is in a system private network on page 288 and Figure 30: Remote user B or C is in the system private network on page 288.

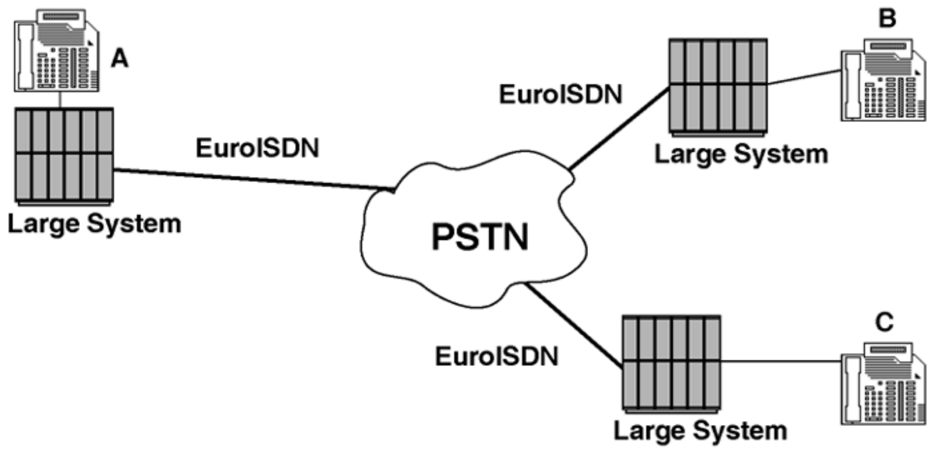

Figure 29: Remote user B or C is in a system private network

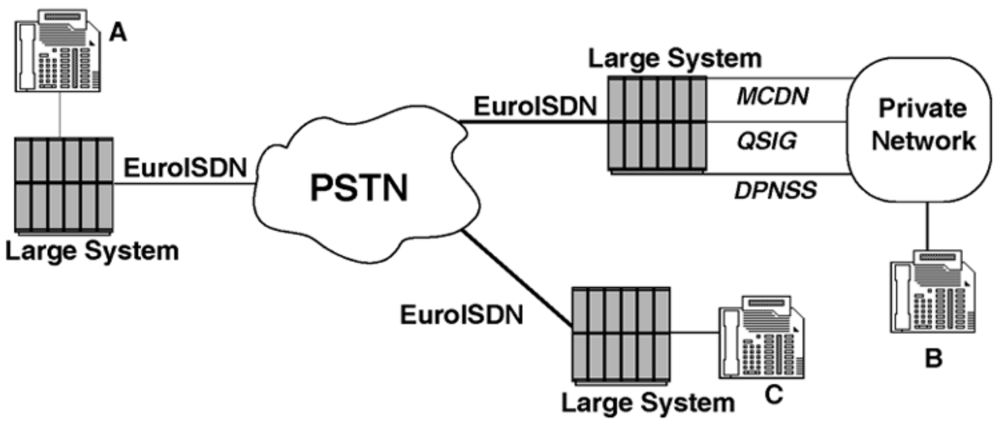

Figure 30: Remote user B or C is in the system private network

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 24: Served User A is the only one in the system private network (or stand alone system) on page 286).
- 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 31: User B calls user A. User A transfers call to user C. on page 291, Figure 32: Display of

established call between user A and user B on page 291, Figure 33: User A presses Call Transfer key and calls user C on page 291, and Figure 34: User A presses Call Transfer key a second time to transfer the call to user C on page 292):

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).

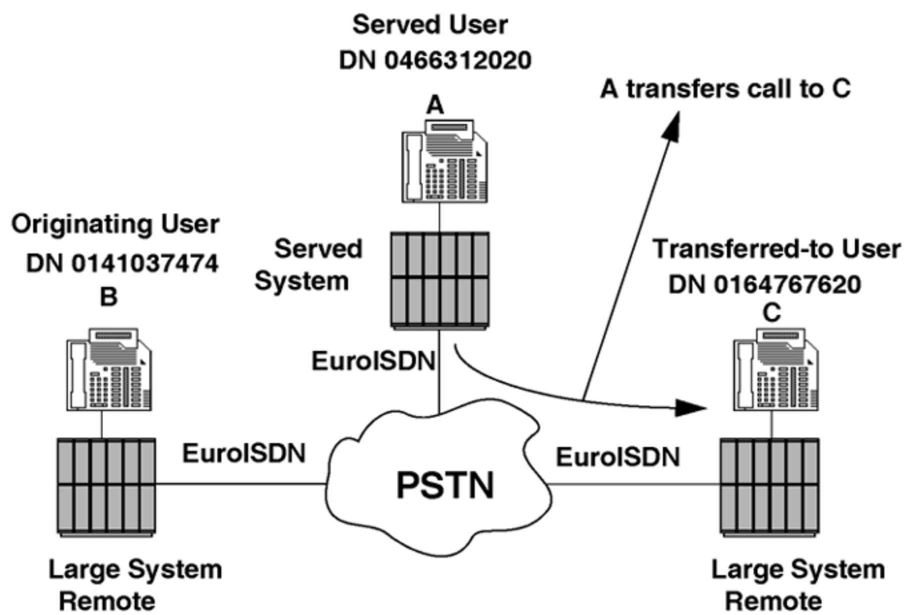

Figure 31: User B calls user A. User A transfers call to user C.

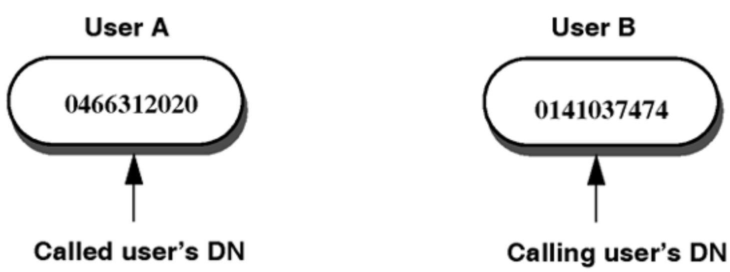

Figure 32: Display of established call between user A and user B

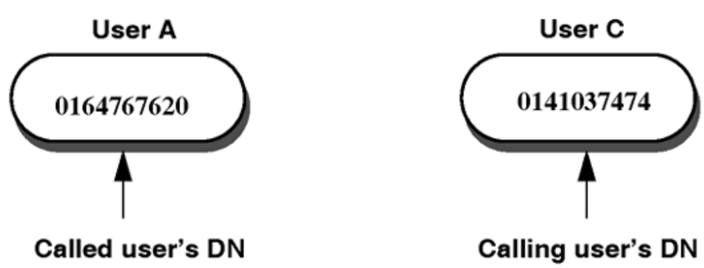

Figure 33: User A presses Call Transfer key and calls user C

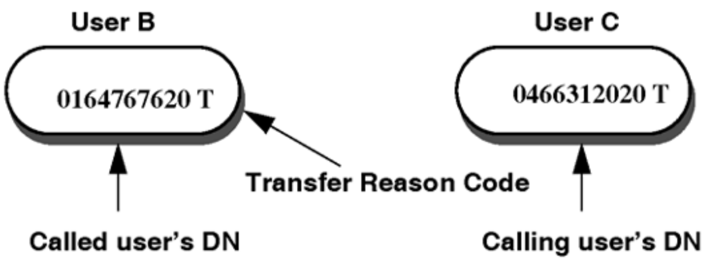

Figure 34: User A presses Call Transfer key a second time to transfer the call to user C

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 users 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 165: Originating and Transferred-to users notification in a system environment on page 293 identifies the originating user's notification according to the originating and transferred-to users' configuration options.

Table 165: 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
    - The Dual PRI pack
    - 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

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.

Meridian Mail

A caller transferred to Meridian Mail through EuroISDN can access Meridian Mail functionalities such as message reception and mailbox interrogation.

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 166: LD 10: Configure an analog (500/2500-type) phone for EuroISDN Call Transfer. on page 296
2. Table 167: LD 11: Configure a Meridian 1proprietary phone for EuroISDN Call Transfer. on page 297
3. Table 168: LD 95: Configure the Call Transfer Reason for Redirection Code. on page 297
4. Table 169: LD 17: Configure EuroISDN Call Transfer on PRI2. on page 298
5. Table 170: LD 16: Configure EuroISDN Call Transfer on BRI trunk. on page 298

Table 166: 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 , Media Gateway 1000B, and CS 1000E system, where l = loop, s = shelf, c = card, u = unit.
DN	x..x	Directory Number.
...	...	
CLS		Class of Service.

Prompt	Response	Description
	XFA	Call Transfer Allowed
	DDGA	DN Display on the other phone allowed (default) DDGD = DN Display on the other phone denied

Table 167: LD 11: Configure a Meridian 1proprietary 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 , Media Gateway 1000B, and CS 1000E system, where l = loop, s = shelf, c = card, 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 168: 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 169: 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 170: 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.
...	...	
CNTY	xxxx	Country.

Prompt	Response	Description
		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 24: Business Network Express/ Name and Private Number Display

Contents

This section contains information on the following topics for Avaya Communication Server 1000 (Avaya CS 1000):

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.

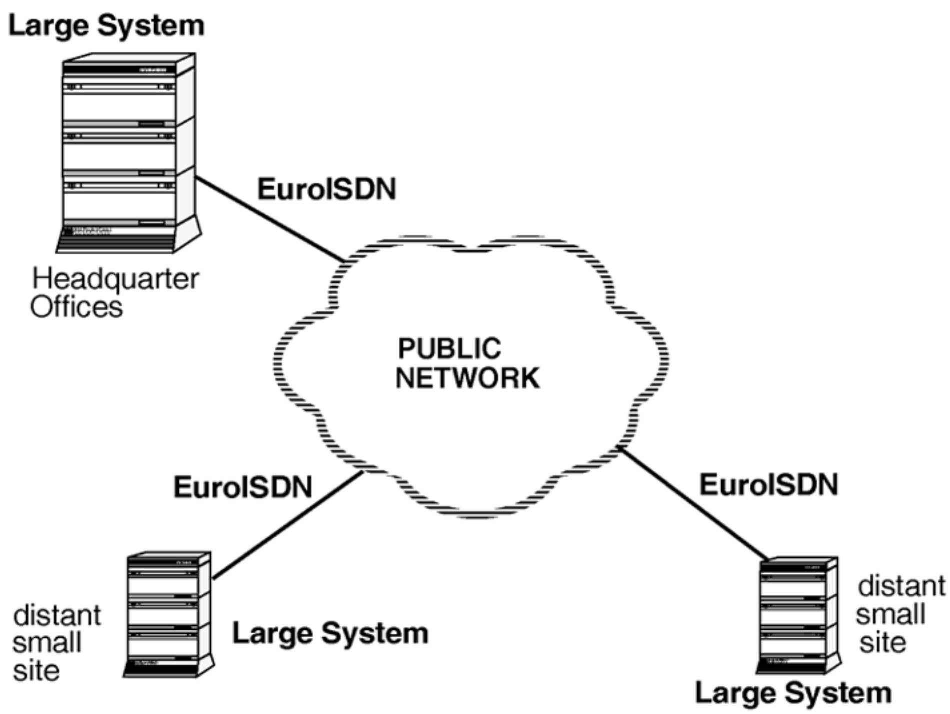

Figure 35: Example of a network where BNE is useful

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 171: 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.

ÔH' 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 172: Reception of COLP on BRI phones on page 310 summarizes the possibilities:

Table 172: 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 36: Call Pickup in a EuroISDN network on page 313. 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.

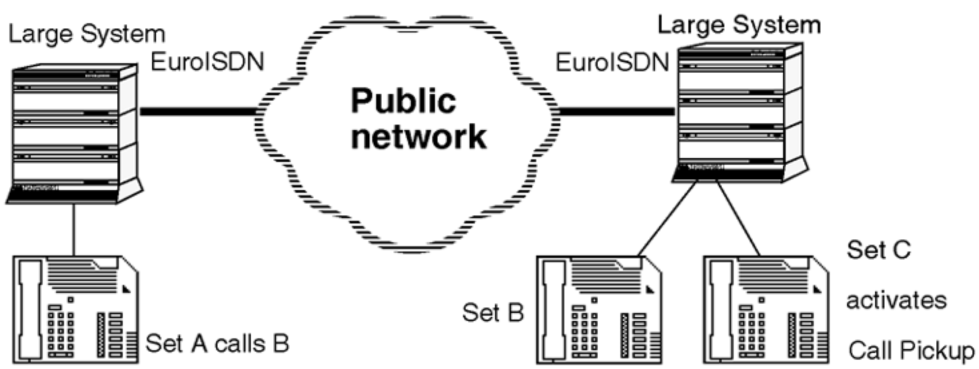

Figure 36: Call Pickup in a EuroISDN network

Call Transfer

Refer to Figure 37: Local Call Transfer on page 314 for an illustration of a local Call Transfer. Refer to for an illustration of an external Call Transfer. Note that in these illustrations, Explicit Call Transfer is not activated.

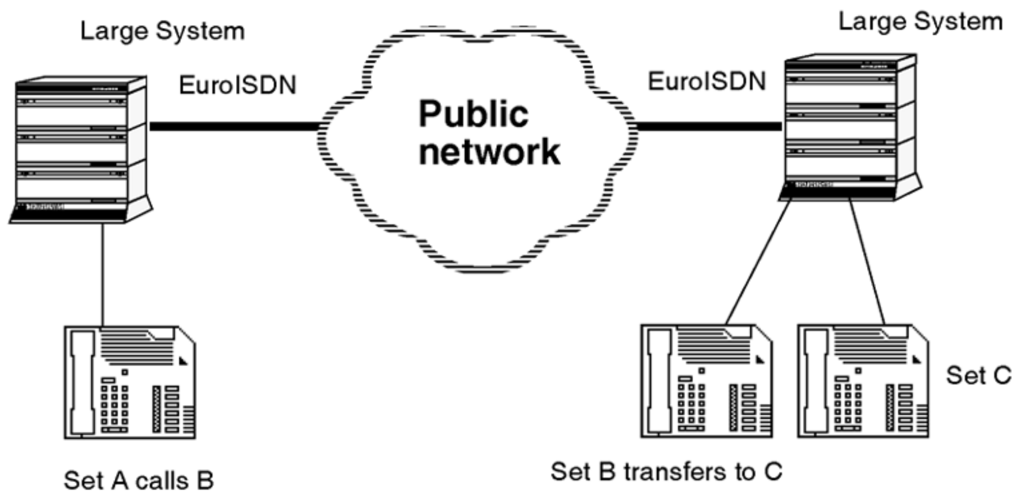

Figure 37: Local Call Transfer

Transfer on ringing (internal)

Figure 37: Local Call Transfer on page 314 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 37: Local Call Transfer on page 314 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.

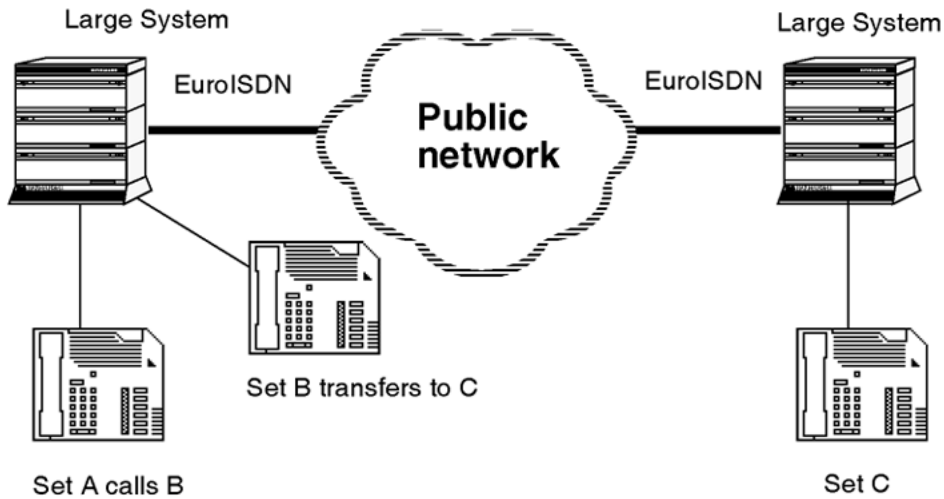

Figure 38: External Call Transfer

Transfer on ringing (external)

Figure 38: External Call Transfer on page 315illustrates 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 38: External Call Transfer on page 315 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 39: Local Conference on page 316 illustrates a conference call involving parties connected through a EuroISDN network.

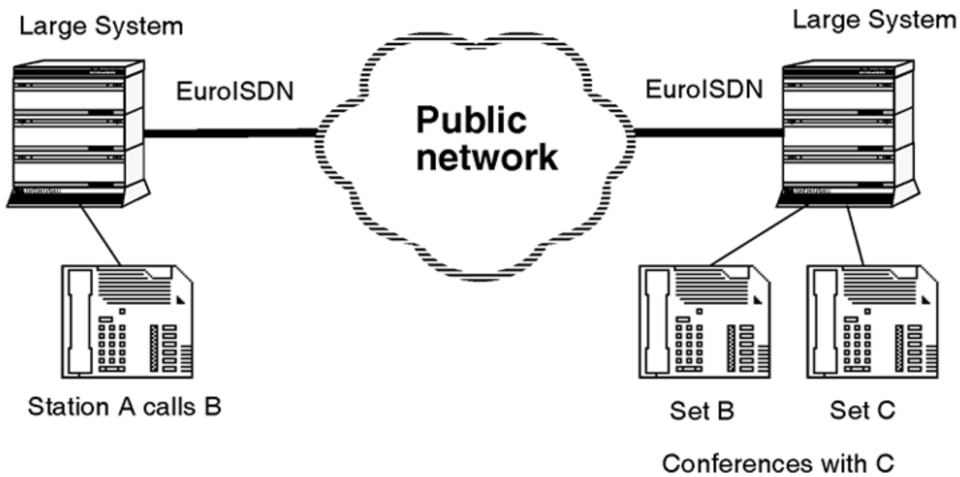

Figure 39: Local Conference

Figure 39: Local Conference on page 316 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 40: Local Call Forward on page 317 illustrates a local Call Forward situation involving parties connected through a EuroISDN network. Figure 41: External Call Forward on page 318 illustrates an external Call Forward situation involving parties connected through a EuroISDN network. Note that in this illustration, Explicit Call Transfer is not activated.

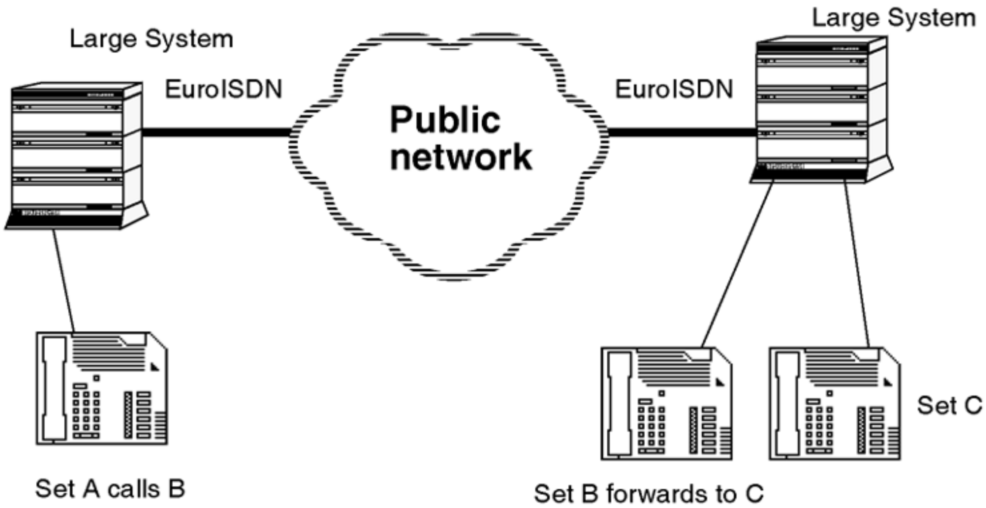

Figure 40: Local Call Forward

Call Forward All Calls (internal)

Figure 40: Local Call Forward on page 317 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 40: Local Call Forward on page 317 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.

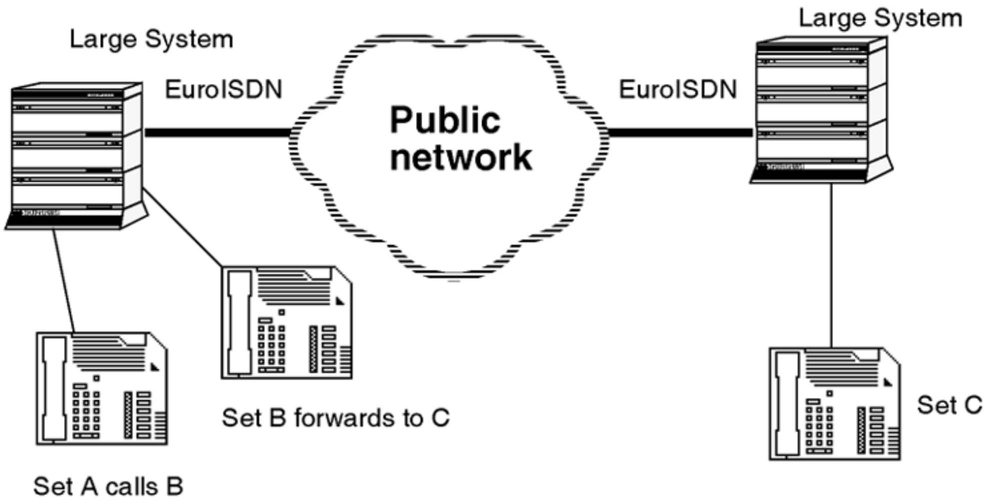

Figure 41: External Call Forward

Call Forward All Calls (external)

A local call can forward over a EuroISDN network, as shown in Figure 41: External Call Forward on page 318. 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 41: External Call Forward on page 318. 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 42: Local Hunting on page 319 illustrates a local Hunting situation.

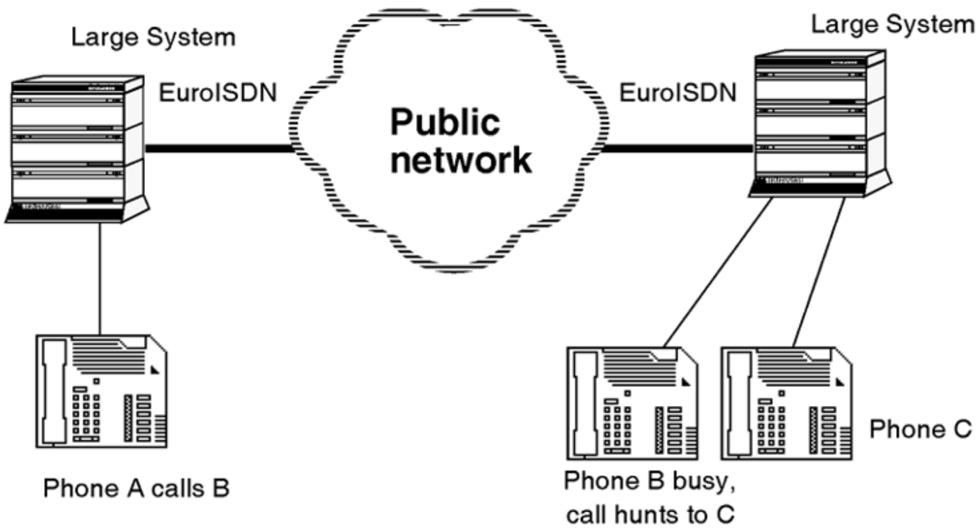

Figure 42: Local Hunting

Figure 42: Local Hunting on page 319 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 43: AOC and Charge Display on page 319 illustrates an example of a situation involving the Advice of Charge feature.

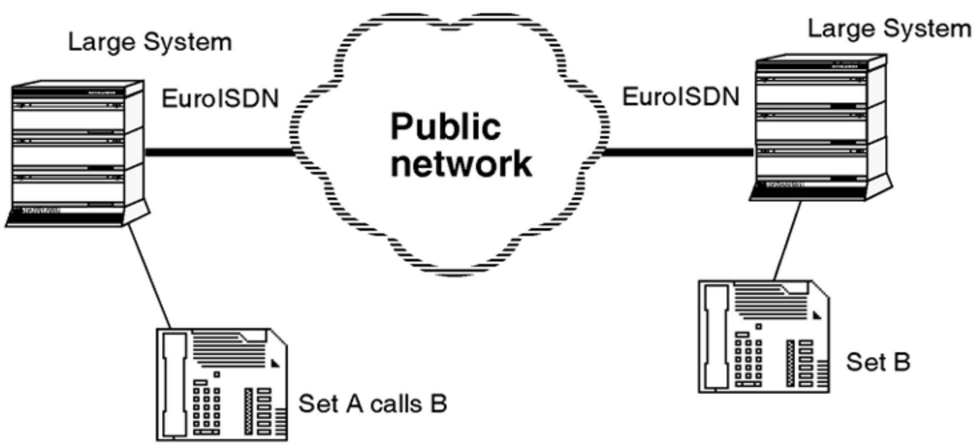

Figure 43: AOC and Charge Display

An outgoing EuroISDN call, carrying BNE signaling, as shown in Figure 43: AOC and Charge Display on page 319, 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 44: Call Diversion in networking on page 321 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 44: Call Diversion in networking on page 321 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.

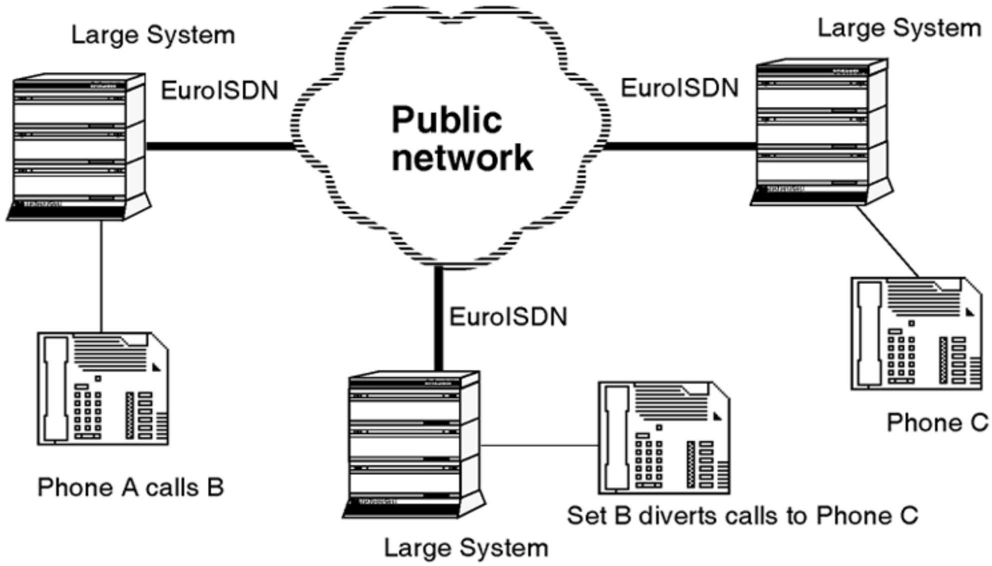

Figure 44: Call Diversion in networking

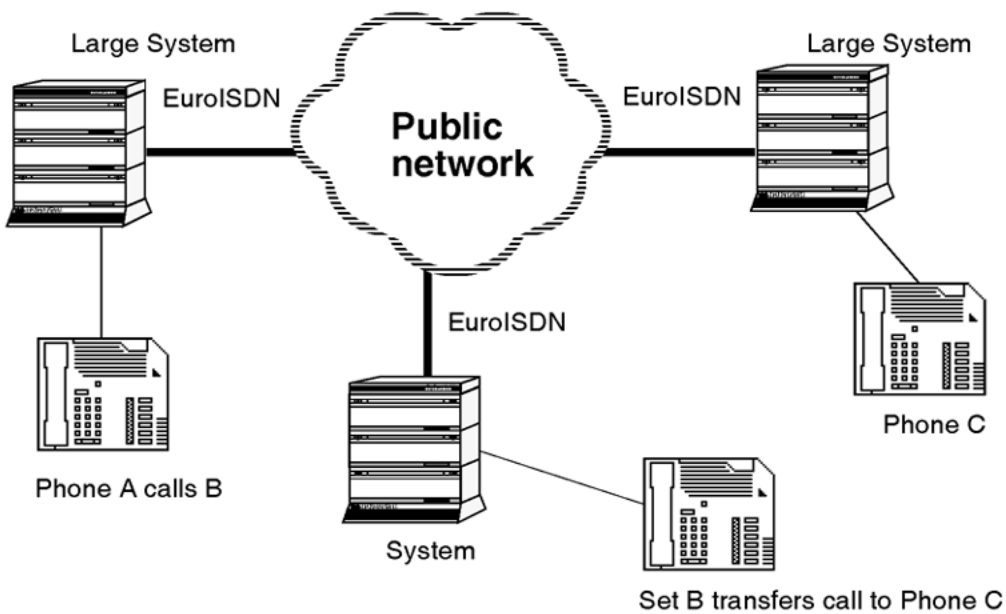

Figure 45: Call Transfer across a network

Explicit Call Transfer (Call Transfer notification sent)

Figure 45: Call Transfer across a network on page 321 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:

1. Table 173: LD 95: Create a CPND data block. on page 323
2. Table 174: LD 95: Create a new name string. on page 324
3. Table 175: LD 10: Allow or deny name and digit display on analog (500/2500- type) phone. on page 324
4. Table 176: LD 11: Allow or deny name and digit display on Meridian 1proprietary phones. on page 325
5. Table 177: LD 12: Allow or deny name display on 2250 consoles. on page 325
6. Table 178: LD 27: Configure the BRI Digital Subscriber Loop. on page 325
7. Table 179: LD 57: Assign Flexible Feature Code for Name Display. on page 326
8. Table 180: LD 16: Configure D-channel for User-to-User service 1 (BRI). on page 326
9. Table 181: LD 17: Configure D-channel for User-to-User service 1 (PRI). on page 327
10. Table 182: LD 86: Configure Route List Index for BNE feature. on page 327

Table 173: LD 95: Create a CPND data block.

Prompt	Response	Description
REQ	NEW	New.
TYPE	CPND	CPND data block.
CUST		Customer number

Prompt	Response	Description
	0-99	Range for Large System , Media Gateway 1000B, and CS 1000E system.

Table 174: LD 95: Create a new name string.

Prompt	Response	Description
REQ	NEW	New.
TYPE	NAME	Create a new name string.
CUST		Customer number
	0-99	Range for Large System , Media Gateway 1000B, and CS 1000E system.
...	...	
DN	xxxx	DN linked with the name string.

Table 175: LD 10: Allow or deny name and digit display on analog (500/2500- type) phone.

Prompt	Response	Description
REQ	NEW	Add new data.
	CHG	Change existing data.
TYPE	500	Type of phone.
CUST		Customer number
	0-99	Range for Large System , Media Gateway 1000B, and CS 1000E system.
...	...	
FTR	CPND	Allow CPND name assignment on this phone (not required if CPND is programmed in LD 95).
CLS	CNDA (CNDD) NAMA (NAMD) DDGA (DDGD) CLBA (CLBD)	Allow (deny) user names to be displayed on this phone. Allow (deny) name display on the far end. Allow (deny) digit display on the far end. Allow (deny) calling number and name per-call blocking.

Table 176: LD 11: Allow or deny name and digit display on Meridian 1proprietary phones.

Prompt	Response	Description
REQ	NEW	Add new data.
	CHG	Change existing data.
TYPE	xxxx	Phone type
CUST		Customer number
	0-99	Range for Large System , Media Gateway 1000B, and CS 1000E system.
...	...	
CLS	CNDA (CNDD) DNDA (DNDD) NAMA (NAMD) DDGA (DDGD) CLBA (CLBD)	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. 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 177: 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		Customer number
	0-99	Range for Large System , Media Gateway 1000B, and CS 1000E system.
...	...	
CPND	CNDA (CNDD)	Allow (deny) user names to be displayed on this console.

Table 178: LD 27: Configure the BRI Digital Subscriber Loop.

Prompt	Response	Description
REQ	NEW	New
TYPE	TSP	Terminal Service Profile.
DSL		Digital Subscriber Loop.

Prompt	Response	Description
	l s c dsl	Format for Large System , Media Gateway 1000B, and CS 1000E system.
...	...	
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 179: LD 57: Assign Flexible Feature Code for Name Display.

Prompt	Response	Description
REQ	NEW	New.
TYPE	FFC	Flexible Feature Code.
CUST		Customer number
	0-99	Range for Large System , Media Gateway 1000B, and CS 1000E system.
...	...	
CPP	xxxx	Calling Party Privacy feature access code. Four digit maximum. Prompted only if CPP software package is equipped.

Table 180: LD 16: Configure D-channel for User-to-User service 1 (BRI).

Prompt	Response	Description
REQ	CHG	Change
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 181: LD 17: Configure D-channel for User-to-User service 1 (PRI).

Prompt	Response	Description
REQ	CHG	Change
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 182: 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.
CUST		Customer number
	0-99	Range for Large System , Media Gateway 1000B, and CS 1000E system.
FEAT	RLB	Route List Data Block.
RLI	xxx	Route List Index.
ENTR	xx	Entry number.
...	...	
FSNI		
BNE	YES (NO)	Business Network Express/Name Display, Private CLID and COLP allowed. 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 25: 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 digital 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 digital phone 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 26: Call Connection Restriction

Contents

This section contains information on the following topics for Avaya Communication Server 1000 (Avaya CS 1000):

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.

- ?/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, ?/A-Law conversions and network call redirections.

Feature packaging

This feature is included in base System Software.

Feature implementation

Table 183: 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 , Media Gateway 1000B, and CS 1000E system.
…		
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 27: Call Forward All Calls/No Answer

Contents

This section contains information on the following topics:

Feature description

Call Forward All Calls and Call Forward No Answer enable callers to manually forward or forward on a no answer to any other station on the ISDN network. The receiving location is provided with the dialed number, the calling number, and if CPND is optioned, the caller's name plus the reason for the redirection. The caller's display is also updated to show the name and number of the person the call was redirected to plus the reason for the redirection.

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

- M2250 Attendant Console
- Meridian Phones equipped with digit display:
  - M2317
  - M2008
  - M2616
  - M2216

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:

- Digit Display (DDSP) package 19
- Integrated Services Digital Network (ISDN) package 145
- International Primary Rate Access (PRI) package 146 or
- ISDN signaling Link (ISL) package 147
- Calling Party Name Display (CPND) package 95

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

Contents

This section contains information on the following topics for Avaya Communication Server 1000 (Avaya CS 1000):

Feature description

The Call Forward, Break-In and Hunt Internal or External Network Wide feature determines whether a call is treated as internal or external on a network wide basis. A call is treated as internal if it terminates and originates within a private network. A call is treated as external if it terminates or originates outside a private network. These definitions of internal/external call are applied to incoming calls for the Call Forward by Call Type feature and to outgoing calls for the Break-In feature.

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

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

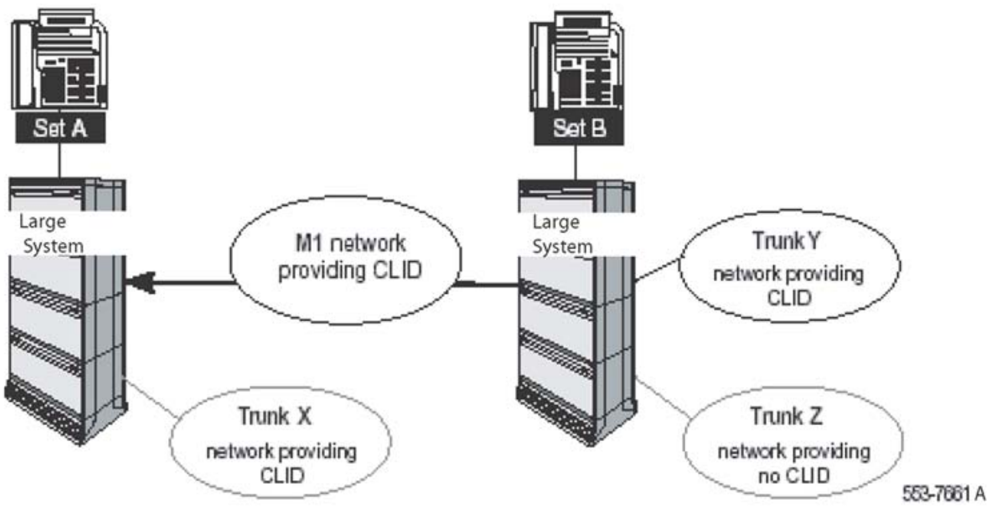

Figure 46: Example of Calling Line Identification for Incoming Calls

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

Figure 46: Example of Calling Line Identification for Incoming Calls on page 338 illustrates several examples of incoming call treatments. When Phone B dials Phone A and receives no answer the Call Forward by Call Type features is activated from Phone A's node. This type of call is treated as an internal call because the Number Plan Identification (NPI) is private in the Calling Line Identification (CLID).

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

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

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

Operating parameters

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

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

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

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

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

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

Feature interactions

Attendant Break-In

Attendant Break-In Indication and Prevention

Call Forward by Call Type

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

Call Forward, Internal

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

Call Transfer

Network Call Transfer

Network Call Redirection

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

Network Attendant Service

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

DPNSS1

QSIG

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

Feature packaging

This feature requires the following packages:

- Call Forward, Break-In and Hunt Internal or External Network Wide is included in base system software
- NAS messaging requires Network Attendant Services (NAS) package 159
- DPNSS requires Integrated Digital Access (IDA) package 122
- Digital Private Network Signaling System No. 1 (DPNSS) package 123
- 2Mbit Primary Rate Interface (PRI2) package 154
- QSIG requires (QSIG) package 263

Feature implementation

Task summary list

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

1. Table 184: LD 15: Define the network wide definition of Internal/External calls in the Customer Data Block. on page 342
2. Table 185: LD 16: Modify Route Data Block to define Internal/External calls. on page 342

Table 184: LD 15: Define the network wide definition of Internal/External calls in the Customer Data Block.

Prompt	Response	Description
REQ	CHG	Change existing data.
TYPE	FTR	Customer Features and options.
CUST		Customer number
	0-99	Range for Large System , Media Gateway 1000B, and CS 1000E system.
- IDEF	YES	Network-wide internal or external definition for Call Forward/Hunt by Call Type, Internal Call Forward and Break-In Indication Prevention. Calls will be treated as internal or external according to the network-wide definition. NO = call will be treated as internal or external as it was previously programmed.

Table 185: LD 16: Modify Route Data Block to define Internal/External calls.

Prompt	Response	Description
REQ	CHG	Change existing data.
TYPE	RDB	Route data block.
CUST		Customer number
	0-99	Range for Large System , Media Gateway 1000B, and CS 1000E system.
ROUT		Route number
	0-511	Range for Large System , Media Gateway 1000B, and CS 1000E system.
...		
RCLS	(EXT) INT	Route Class marked as External or Internal.
IDEF	(NET) LOC	Internal/External Definition. If NET is entered, any call over the selected route will receive network treatment according to available network information. If LOC is entered, the route class of the selected route will supercede any other information. A call over this route will receive internal treatment if the route class is set to internal, otherwise it will receive external.

Note:

The prompt IDEF will output only if IDEF = YES in the Customer Data Block. If IDEF = NO, any information that was entered previously at the IDEF prompt will not influence the treatment received by a call.

Feature operation

No specific operating procedures are required to use this feature.

Chapter 29: 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 186: LD 57: Define FFC for Call Forward/Hunt Override analog (500/2500-type). on page 352
2. Table 187: LD 10: Set Class of Service for the Forward/Hunt Override Via FFC. on page 352
3. Table 188: LD 11: Set Class of Service for the Forward/Hunt Override through FFC for Meridian 1 proprietary telephones. on page 353
4. Table 189: LD 18: Configure ABCD key for the Forward/Hunt Override Via FCC. on page 353

Table 186: 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.
CFHO	nnnn	Call Forward/Hunt FFC.

Table 187: 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 188: LD 11: Set Class of Service for the Forward/Hunt Override through FFC for Meridian 1 proprietary telephones.

Prompt	Response	Description
REQ	NEW	Add new data.
	CHG	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 189: 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.
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 30: Call Page Network Wide

Contents

This section contains information on the following topics for Avaya Communication Server 1000 (Avaya CS 1000):

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:

1. Table 190: LD 16: Configure Paging Route. on page 358
2. Table 191: LD 16: Configure BRI Trunk Route. on page 359
3. Table 192: LD 14: Configure Page Trunk. on page 359
4. Table 193: LD 10: Assign Class of Service to analog (500/2500-type) phones. on page 359
5. Table 194: LD 11: Assign Class of Service to Meridian 1 Proprietary Phones. on page 360
6. Table 195: LD 27: Assign Class of Service to ISDN BRI phones. on page 360
7. Table 196: LD 17: D-channel Message Configuration. on page 360

Table 190: LD 16: Configure Paging Route.

Prompt	Response	Description
REQ	NEW	New.
TYPE	RDB	Route data block.
CUST		Customer number
	0-99	Range for Large System , Media Gateway 1000B, and CS 1000E system.
…		
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.)

Prompt	Response	Description
ICOG	OGT	Outgoing trunk.
TARG	1-15	Trunk Access Restriction Group.

Table 191: LD 16: Configure BRI Trunk Route.

Prompt	Response	Description
REQ	NEW	New.
TYPE	RDB	Route data block.
CUST		Customer number
	0-99	Range for Large System , Media Gateway 1000B, and CS 1000E system.
…		
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 192: 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 , Media Gateway 1000B, and CS 1000E system, where l = loop, s = shelf, c = card, u = unit.
RTMB		Route number and Member Number
	0-511 1-4000	Range for Large System , Media Gateway 1000B, and CS 1000E system.

Table 193: LD 10: Assign Class of Service to analog (500/2500-type) phones.

Prompt	Response	Description
REQ	NEW	Add new data.
	CHG	Change existing data.

Promp t	Response	Description
TYPE	500	Type of phone.
…		
CLS	PGNA	Call Page Network Wide Allowed. (PGND) = Call Page Network Wide Denied (default).

Table 194: LD 11: Assign Class of Service to Meridian 1 Proprietary Phones.

Promp t	Response	Description
REQ	NEW	Add new data.
	CHG	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 195: LD 27: Assign Class of Service to ISDN BRI phones.

Promp t	Response	Description
REQ	NEW	Add new data.
	CHG	Change existing data.
TYPE	DSL	Type of data block.
DSL		Digital Subscriber Loop.
	l s c dsl	Format for Large System , Media Gateway 1000B, and CS 1000E system.
CLS	PGNA	Call Page Network Wide Allowed. (PGND) = Call Page Network Wide Denied (default).

Table 196: LD 17: D-channel Message Configuration.

Promp t	Response	Description
REQ	CHG	Change.
TYPE	ADAN	Input or Output Devices.
ADAN	CHG DCH x	Change D-channel. x = 0-63.

Prompt	Response	Description
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 31: Call Park Network Wide

Contents

This section contains information on the following topics for Avaya Communication Server 1000 (Avaya CS 1000):

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 197: 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 , Media Gateway 1000B, and CS 1000E system.
- OPT	CPN	Enable Call Park Network Wide. CPA = Enables Call Park. CPD = Disables Call Park (default).

Table 198: 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 199: 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		Customer number
	0-99	Range for Large System , Media Gateway 1000B, and CS 1000E system.
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.

Promp t	Response	Description
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 200: LD 10 - Enable Call Park for analog (500/2500-type) phones.

Promp t	Response	Description
REQ	NEW	Add new data.
	CHG	Change existing data.
TYPE	500	Type of phone.
TN		Terminal number
	l s c u	Format for Large System , Media Gateway 1000B, and CS 1000E system, where l = loop, s = shelf, c = card, u = unit.
CLS	XFA	Call Transfer Allowed. XFD = Call Transfer Denied.

Table 201: LD 11 - Add/Change Call Park Key on Meridian 1 proprietary phones.

Promp t	Response	Description
REQ	NEW	Add new data.
	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 , Media Gateway 1000B, and CS 1000E system, where l = loop, s = shelf, c = card, 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 202: LD 12 - Add/Change Call Park Key on Attendant Consoles.

Promp t	Response	Description
REQ	NEW	Add new data.
	CHG	Change existing data.

Prompt	Response	Description
TYPE	xxxx	Console type.
TN		Terminal number
	l s c u	Format for Large System , Media Gateway 1000B, and CS 1000E system, where l = loop, s = shelf, c = card, u = unit.
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 32: Call Pickup Network Wide

Contents

This section contains information on the following topics for Avaya Communication Server 1000 (Avaya CS 1000):

Feature description

Large installations exceeding the capacity of a single private branch exchange, or requiring network solutions to security requirements (as in the case of hospitals, chemical companies, nuclear power plants, etc.) have presented the possible circumstance of their employees working in close proximity to each other, but not sharing the same phone switch. This raises the need to make available network-wide those features that were previously only required locally; Call Pickup is one such feature.

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

- Ringing Number Pickup
- Directed Call Pickup by Group Number
- Directed Call Pickup by DN
- Display Call Pickup

With the exception of Display Call Pickup, user operation of the above features remains unchanged. To display Call Pickup, press the Display key, followed by the Call Pickup key.

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

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

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

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

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

Operating parameters

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

A Speed Call List must be configured with the PINX DNs of the remote switch. The local PINX DN must be configured in the Customer Data Block (LD 15).

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

Feature interactions

Access Restrictions

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

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

Attendant-Extended Calls

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

Automatic Call Distribution (ACD)

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

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

Call Back Queuing

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

Call Collision (Glare)

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

Call Park

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

Called Party Name Display

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

Name displayed on the requesting party

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

Call Redirection

Network Call Redirection

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

Call Transfer

Network Call Transfer

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

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

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

Call Waiting

Call Waiting calls cannot be picked up.

Call Waiting Redirection

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

Calling Party Number Privacy

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

Conference

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

Data Calls

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

Dial Intercom

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

Display of Calling Party Denied

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

Group Call

The Group Call feature does not allow having a remote party in a Group Call list. Therefore, a Group Call cannot be picked up by a remote station. If during the network scanning a Group Call is found, it will be ignored and the network scanning will continue.

ISDN Basic Rate Interface (BRI)

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

ISDN QSIG Name Display

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.

Multi-Tenant Service

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

Network HOT Type D Intercom

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

Network Routing Restrictions

NARS Anti-Tromboning

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

Network Attendant Service/ISDN Call Connection Limitations

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

Night DN

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

Off-Hook Queuing

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

Path Optimization

Network Attendant Services (NAS)

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

Trunk Route Optimization Before Answer

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

Secondary DN

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

Vacant Number Routing

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

Virtual Network Services

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

Feature packaging

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

The following packages are also required:

- Directed Call Pickup (DCP) package 115
- Advanced ISDN Network Services (NTWK) package 148
- 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 203: LD 10: Assign a Ringing Pickup Group number to an analog (500/2500-type) phone. on page 383
2. Table 204: LD 11: Assign a pickup group to a Meridian 1 proprietary phone. on page 384
3. Table 205: LD 15: Define the customers Private Integrated Network Identifier (PNI) and Private Integrated Services Network Exchange (PINX) DN. on page 385

4. Table 206: LD 57 - Define the Flexible Feature Codes (FFCs) used to activate Call Pickup and Directed Call Pickup from an analog (500/2500-type) phone. on page 385
5. Table 207: LD 18: Configure a Speed Call List. on page 386
6. Table 208: LD 18 - Define the Ringing Number Pickup Groups that are network wide by linking them to one of the previous defined Speed Call Lists. on page 387
7. Table 209: LD 17: Define the software Release ID at the far end of the D-channel. on page 387
8. Table 210: LD 15: Define the special prefix code (SPRE) to be able to activate the Call Pickup features by dialing SPRE + xx. on page 387
9. Table 211: LD 16: Configure the PNI on the route. This PNI must correspond to the one configured in LD 15 on the target (remote) node. on page 388

Note:
Define the Class of Service for Call Pickup features that are allowed to be activated from this station.

Table 203: LD 10: Assign a Ringing Pickup Group number to an analog (500/2500-type) phone.

Prompt	Response	Description
REQ	CHG	Change.
TYPE	500	Type of phone.
TN		Terminal number
	l s c u	Format for Large System , Media Gateway 1000B, and CS 1000E system, where l = loop, s = shelf, c = card, u = unit.
...		
RNPG	(0)-4095	Ringing Number Pickup Group (RNPG). If the RNPG is set to 0 (the default) on a station, it is not possible to pick up any call ringing this station from any other phone. Enter 0 to remove a station from the RNPG. **Note:** If the RPNG is two or more digits in length, it must be configured in the Customer Data Block (LD 15), so that it can use the Group Pickup key. See the procedure which follows LD 11.
...		
CLS	(PUD) PUA	Ringing Number Pickup (denied) allowed.
	(DPUD) DPUA	DN pickup (denied) allowed.

Prompt	Response	Description
	(GPUD) GPUA	Group pickup (denied) allowed.

Define with the Class of Service which of the Call Pickup features are allowed to be activated from this station.

The configuring of different pickup keys is optional, since the Call Pickup and Directed Call Pickup features can be activated by dialing the SPRE + xx or by dialing a Flexible Feature Code. The Digit Display key is needed for the Display Call Pickup feature.

Table 204: LD 11: Assign a pickup group to a Meridian 1 proprietary phone.

Prompt	Response	Description
REQ	CHG	Change.
TYPE	a...a	Telephone type. Type **?** for a list of possible responses.
TN		Terminal number
	l s c u	Format for Large System , Media Gateway 1000B, and CS 1000E system, where l = loop, s = shelf, c = card, u = unit.
...		
RNPG	(0)-4095	Ringing Number Pickup Group (RNPG). If the RNPG is set to 0 (the default) on a station, it is not possible to pick up any call ringing this station from any other phone. Enter 0 to remove a station from the RNPG. **Note:** If the RPNG is two or more digits in length, it must be configured in the Customer Data Block (LD 15), so that it uses the Group Pickup key. See the procedure which follows.
...		
CLS	(PUD)PUA	Ringing Number Pickup (denied) allowed.
	(DPUD)DPUA	DN pickup (denied) allowed.
	(GPUD)GPUA	Group pickup (denied) allowed.
...		
KEY	xx RNP yyyy	Key number, Ringing Number Pickup, Pickup Group number (optional). If the Group number is not entered, the key will pick up calls in the group assigned to the station. If the Group number is entered, the key will pick up calls in the specified group yyyy.
KEY	xx GPU	Key number, Group Number Pickup.

Prompt	Response	Description
KEY	xx DPU	Key number, DN Pickup.
...		
KEY	xx DSP	Key number, Digit Display.

Table 205: LD 15: Define the customers Private Integrated Network Identifier (PNI) and Private Integrated Services Network Exchange (PINX) DN.

Prompt	Response	Description
REQ	CHG	Change.
TYPE	NET	Networking data.
CUST		Customer number
	0-99	Range for Large System , Media Gateway 1000B, and CS 1000E system.
...		
ISDN	YES	Allow ISDN for this customer.
- PNI	1-32700	Define the Private Network Identifier.
- PINX_DN	xx...x X	Node DN, up to seven digits with DN Expansion package 150. Enter X to remove.

For the following table LD 57, define the Flexible Feature Codes (FFCs) used to activate Call Pickup and Directed Call Pickup from a an analog (500/2500-type) phone. The FFCs can also be used on Meridian 1 Proprietary Phones.

Table 206: LD 57 - Define the Flexible Feature Codes (FFCs) used to activate Call Pickup and Directed Call Pickup from an analog (500/2500-type) phone.

Prompt	Response	Description
REQ	NEW	Add new data.
	CHG	Change existing data.
TYPE	FFC	Flexible Feature Code.
CUST		Customer number
	0-99	Range for Large System , Media Gateway 1000B, and CS 1000E system.
FFCT	(NO) YES	Flexible Feature Confirmation tone.
CODE	mmmm	Specific FFC type.
...		

Prompt	Response	Description
- PUDN	PUDN xxxx	Pickup DN code. Enter the Flexible Feature Code.
- PUGR	PUGR xxxx	Pickup Group code. Enter the Flexible Feature Code.
- PURN	PURN xxxx	Pickup Ringing Number code. Enter the Flexible Feature Code.

This Speed Call List is used by a Ringing Number Pickup Group (RNPG) or a set of RNPGs as a search list to scan the MCDN network.

The Speed Call List entries should contain digits which can be used to route a network Call Pickup request to a remote node (e.g., mainly the ISDN PINX DNs of the remote nodes which will be scanned after a network Call Pickup request).There must be no gaps in the Speed Call List (i.e., each Speed Call List entry should be present).

Due to the time it takes to scan the remote node, it is strongly recommended to configure less than six entries in the Speed Call List.

Table 207: LD 18: Configure a Speed Call List.

Prompt	Response	Description
REQ	NEW	Add new data.
	CHG	Change existing data.
TYPE	SCL	Speed Call List.
LSNO	0-8190	Speed Call List number
...		
DNSZ	4-(16)-31	Maximum size of DNs allowed for Speed Call list.
SIZE	1-1000	Maximum number of DNs allowed in Speed Call list. **Note:** The size cannot be greater than the value entered against the STOR prompt below.
...		
STOR	000-999 xxxx	Speed Call List entry number and digits (PINX DN) stored against it. **Note:** The STOR entry cannot be blank.

For the following table LD 18, define the Ringing Number Pickup Groups that are network wide by linking them to one of the previous defined Speed Call Lists. Different RNPGs can be linked to different Speed Call Lists.

Table 208: LD 18 - Define the Ringing Number Pickup Groups that are network wide by linking them to one of the previous defined Speed Call Lists.

Prompt	Response	Description
REQ	NEW	Add new data.
	CHG	Change existing data.
	OUT	Remove existing data.
TYPE	CPNW	Call Pickup Network Wide data.
CUST		Customer number
	0-99	Range for Large System , Media Gateway 1000B, and CS 1000E system.
LSNO	0-8190	A Speed Call List associated with Call Pickup network wide groups.
- GRP	0-4095	Ringing Number Pickup Group (RNPG) using this Speed Call List. Repeat for all groups sharing the same list. Enter <CR> to reprompt LSNO.
	X0-4095	Enter X to remove an RNPG.

Table 209: LD 17: Define the software Release ID at the far end of the D-channel.

Prompt	Response	Description
REQ	CHG	Change I/O device.
TYPE	ADAN	Action Device and Number.
ADAN	CHG DCH 0-63	Change D-channel information
...		
- RLS	xx	Release ID of the switch at the far end of the D-channel.

Table 210: LD 15: Define the special prefix code (SPRE) to be able to activate the Call Pickup features by dialing SPRE + xx.

Prompt	Response	Description
REQ	CHG	Change existing data.
TYPE	FTR	Customer features and options
CUST		Customer number
	0-99	Range for Large System , Media Gateway 1000B, and CS 1000E system.
...		

Prompt	Response	Description
- SPRE	xxxx	Special Prefix number for this customer.

Table 211: LD 16: Configure the PNI on the route. This PNI must correspond to the one configured in LD 15 on the target (remote) node.

Prompt	Response	Description
REQ	CHG	Change existing data.
TYPE	RDB	Route data block.
ROUT		Route number
	0-511	Range for Large System , Media Gateway 1000B, and CS 1000E system.
...		
ISDN	YES	ISDN is allowed on this route.
- PNI	1-32700	Define the Private Network Identifier.

Feature operation

Display Call Pickup

This feature is only supported on Meridian digital telephones (with the exception of the M2317).

Display Call Pickup

1. Press the DSP key.

   The DSP key is lit.
2. Press the RNP key.

   The RNP key is flashing.
3. The display will show: Hbbbb1 Haaaa2

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

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

The RNP key is winking for five seconds.

4. Press the DN key.

   The DN key is lit.

5. Press the RNP key. If the RNP key is pressed during the five seconds it is winking, Directed Call Pickup by DN is attempted. If the RNP key is dark when pressed, normal Ringing Number Pickup treatment takes place.

   The RNP key is flashing.

6. You are connected to the call.

   - The RNP key becomes dark.
   - For phones with display the following is shown: "Name"1 Hcccc2 Hbbbb3 "P"4

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

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

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

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

Ringing Number Pickup

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

Ringing Number Pickup (from a Meridian Digital Phone)

1. Lift the handset, or press a DN key.

   The DN key is lit (if pressed).

2. Press the RNP key or dial SPRE + 3 or dial PURN FFC.

   The RNP key is flashing.

3. You are connected to the call.

   - The RNP key becomes dark.
   - For phones with display, refer to step 3 under "Display Call Pickup".

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

Ringing Number Pickup (from an analog 500/2500 type phone)

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

Directed Call Pickup by Group Number (Group Pickup)

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

Directed Call Pickup by Group Number (Group Pickup from a Meridian Digital Phone)

1. Lift the handset, or press a DN key.

   The DN key is lit.
2. Press the GPU key or dial SPRE + 94 or dial PUGR FFC.

   The GPU key is lit (if pressed).
3. Dial the pickup group number.

   Once the group number is dialed completely, the GPU key is flashing to indicate that a search or scan is in progress.
4. You are connected to the call.
   - The GPU key becomes dark.
   - For phones with display, refer to Step 3 under "Display Call Pickup".

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

Directed Call Pickup by Group Number (Group Pickup from an analog 500/2500 type phone)

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

Directed Call Pickup by Group Number (Group Pickup)

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

Directed Call Pickup by Group Number (Group Pickup from a Meridian Digital Phone)

1. Lift the handset, or press a DN key.

   The DN key is lit.

2. Press the DPU key or dial SPRE + 95 or dial PUDN FFC.

   The DPU key is lit (if pressed).

3. Dial the extension number.

   Once the group number is dialed completely, the DPU key is flashing to indicate that a search or scan is in progress.

4. You are connected to the call.

   - The DPU key becomes dark.
   - For phones with display, refer to Step 3 under "Display Call Pickup".

To answer a call on a specified DN from an analog 500/2500 type phone:

Directed Call Pickup by Group Number (Group Pickup from an analog 500/2500 type phone)

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

You are connected to the call.

Chapter 33: Calling Line Identification in CDR

Contents

This section contains information on the following topics for Avaya Communication Server 1000 (Avaya CS 1000):

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 212: CLID number in the TTY output

Rec Type	Rec No	Cust No	OrigID	TerID	AuxID lll.s.cc.uu	Date mm/dd	Time hh:mm	Duration hh:mm:s	Digits
N	001	00	DN499	A001000	027.1.02.1	06/28	10:14	00:00:20	95559124
N	002	00	T00201	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 47: CDR in Multi-site Configuration on page 395 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.

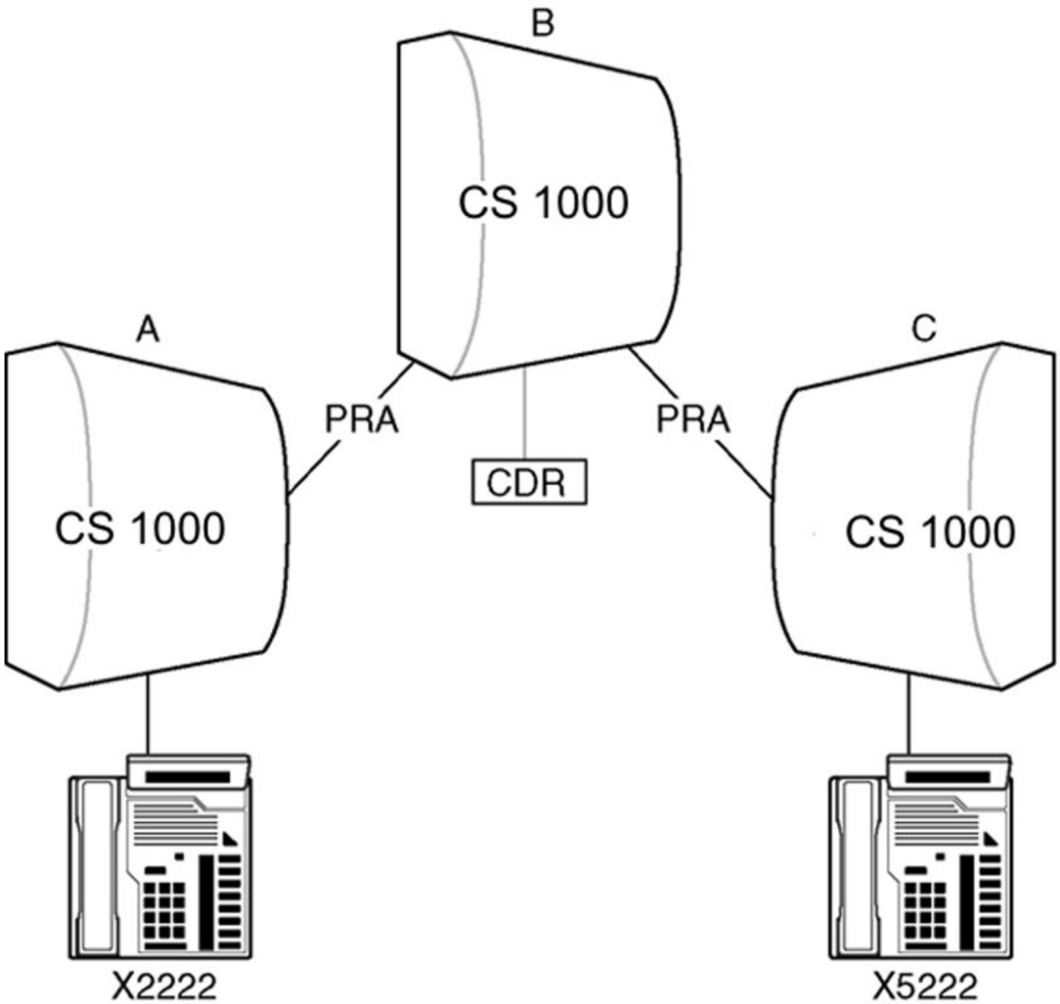

Figure 47: CDR in Multi-site Configuration

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 213: LD 17: Change the Configuration Record to enable CLID. on page 397

   It is assumed that ISDN PRI is configured for the customer.
2. Table 214: LD 15: Change the Customer Data Block to configure CLID. on page 397

3. Table 216: LD 16: Allow CDR records in the Trunk Route Data Block. on page 398
4. Table 215: LD 17: Allow CDR to be printed on the TTY terminal. on page 398
5. Table 216: LD 16: Allow CDR records in the Trunk Route Data Block. on page 398

Table 213: LD 17: Change the Configuration Record to enable CLID.

Prompt	Response	Description
REQ	CHG	Change existing data
TYPE	PARM	Change parameters.
...		
CLID	YES	Enable CLID.

Table 214: 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 , Media Gateway 1000B, and CS 1000E system.
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.

Prompt	Response	Description
- 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 215: 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 216: 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		Customer number
	0-99	Range for Large System , Media Gateway 1000B, and CS 1000E system.
...		
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 34: Calling Party Privacy

Contents

This section contains information on the following topics for Avaya Communication Server 1000 (Avaya CS 1000):

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 for incoming calls received from the North American public ISDN network.

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) change from restricted or denied to allowed.

If the CLID and CPND information is available, it appears on the terminating phone, and the CLID passes to the Auxiliary processor. Avaya recommends that you configure the PII prompt 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.

With CS 1000 Release 6.0 (or later), the CPPE feature (PII prompt) is applicable for incoming routes for all ISDN interfaces, and signaling mechanisms using ISDN between the call server and an additional server (such as the signaling server).

The CPPE feature also introduces a new route option AUXP (for Auxiliary processor applications), in overlay 16 (RDB – Route Data Block) to enhance the ability of the system to honor or ignore the Privacy indicator for a Calling Party Privacy call, on a per incoming route basis. If the AUXP is set to YES, the CLID Presentation Indicator and the CPND Indicator (if it exists) in an incoming SETUP message change from restricted or denied to allowed for auxiliary applications such as Contact Center Manager (CCM). If AUXP is NO, then there is no change to the CLID Presentation Indicator.

If the PII option is set to YES, then the AUXP prompt is automatically set to YES. As long as the PII prompt remains YES, the AUXP prompt will be printed as YES and you cannot modify it. When PII is set as NO, you can modify the AUXP prompt.

The default values for both the prompts (PII and AUXP) is NO.

Example: PII and AUXP settings

Consider a scenario in which an originating node, with blocked CLID, calls Node A (Set A and Aux A) and Node B (Set B and Aux B). Node A is a tandem node. Table 217: PII and AUXP settings on page 401 displays the results of PII and AUXP settings.

Table 217: PII and AUXP settings

Settings at Tandem Node (Node A)	Settings at Node B		
	PII - NO AUXP - NO	PII - NO AUXP - YES	PII - YES AUXP - YES
PII - NO AUXP - NO Call to Node A	Set A: No presentation Aux A: No presentation Set B: No impact		
PII - NO AUXP - NO Call to Node B	Set A: No impact Set B: No presentation Aux B: No presentation	Set A: No impact Set B: No presentation Aux B: Presented	Set A: No impact Set B: Presented Aux B: Presented
PII - NO AUXP - YES Call to Node A	Set A: No presentation Aux A: Presented Set B: No impact		
PII - NO AUXP - YES Call to Node B	Set A: No impact Set B: No presentation Aux B: No presentation	Set A: No impact Set B: No presentation Aux B: Presented	Set A: No impact Set B: Presented Aux B: Presented
PII - YES AUXP - YES Call to Node A	Set A: Presented Aux A: Presented Set B: No impact		
PII - YES AUXP - YES Call to Node B	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. Therefore, when PII is YES at Node A, all calls are treated as		

Settings at Tandem Node (Node A)	Settings at Node B	
	unrestricted at Node B.	

Routes configuration for PII and AUXP prompts

You can use the Routes Configuration page in Element Manager to configure the PII and AUXP prompts for the following ISDN interfaces: 1TR6 APAC AXEA, AXES D70, D100, D250 EURO, E403 EGF4, ESIG, ESGF, ESS4, ESS5 ISGF, ISIG JTTC NUME S100 SS12 SWIS TCNZ

The AUXP prompt is dependent on PII prompt. If the PII prompt is checked, then the AUXP prompt is automatically checked and cannot be modified. When PII is unchecked, AUXP can be modified.

You can enable or disable the PII and AUXP prompts by checking or unchecking them, respectively. By default, both PII and AUXP prompts are unchecked.

Important:

Access rights for Element Manager

You must log in to Element Manager with sufficient security access rights to configure the PII and AUXP prompts, and add a new route. For more information on using Element Manager, see *Avaya Element Manager System Reference - Administration, NN43001-632*.

Adding a new route using Element Manager

1. Navigate to 'Routes and Trunks' page of Element Manager.
2. Click "Add Route".

   The Routes Configuration page appears on the screen as seen in Figure 48: New Route Configuration in Element Manager on page 403.

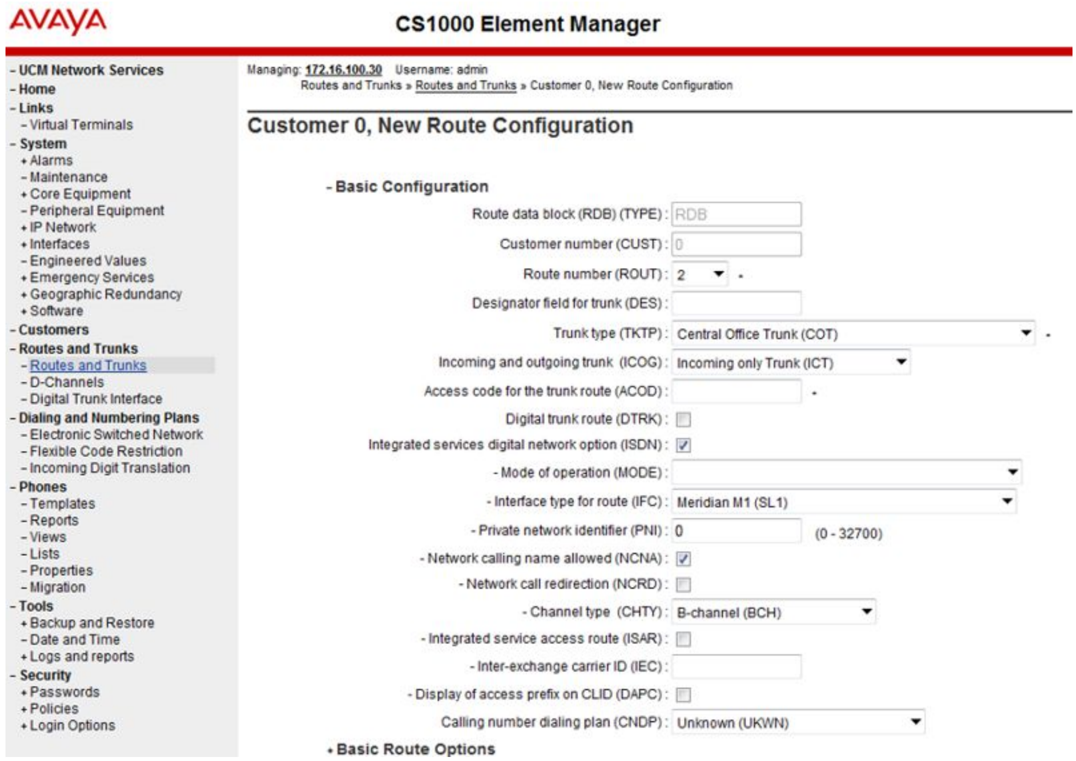

Figure 48: New Route Configuration in Element Manager

3. In the ‘Routes Configuration’ page, navigate to “Trunk Type (TKTP)” drop box menu under “Basic Configuration”.
4. Select any option from ‘COT’, ‘FEX’, ‘WAT’, ‘TIE’ or ‘DID’.
5. Navigate to “Incoming and Outgoing trunk (ICOG)” drop box menu and select any option except ‘OGT’.
6. Check the “ISDN” checkbox.

   The PII and AUXP prompts are displayed under Advanced Configurations.
7. Configure the “PII” and/or “AUXP” prompts as enabled or disabled by checking or unchecking the prompt checkboxes, respectively.

   Provide valid values for the other required fields on the same page.

8. Click “Save”.

   A new route is added successfully.

Software upgrades

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

When you upgrade the database of an older release to CS 1000 Release 6.0 or later, the PII and AUXP options are set to the value NO by default. When you upgrade M1 Release software with a pre-release database (where the route configuration has PII option and PII is set to YES), the PII and AUXP fields in route data blocks for the new release will be set to the value YES.

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.

The trunk route types that can outpulse the Privacy Indicator for an outgoing non-ISDN call are as follows: - Central Office Trunk (COT) - Foreign Exchange (FEX) - Wide Area Telephone Service (WATS) - Direct Inward Dialing (DID) - Internal private network trunks

The CPPE 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.

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 (#) are 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 (for example, *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.

The following parameters apply to the PII and AUXP route options:

- The PII and AUXP prompts apply to all ISDN interfaces.
- The PII and AUXP route options apply to the CLID Presentation Indicator in the Calling Number Information Element (IE) and the CPND Indicator in the Display IE in incoming SETUP messages only.
- The PII and AUXP route options do not apply to the CLID Presentation Indicator in the Redirecting Number IE of a SETUP message for call redirection.
- The PII and AUXP route options do not apply to the CLID Presentation Indicator in the Connected Number IE of a NOTIFY message for call modification.
- The AUXP route option does not apply to the CLID Presentation Indicator in the Original Called Number in the SETUP message.

For an outgoing ISDN call to the Central Office (CO), 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. Thus, if PII is YES at the incoming route, it cannot unblock the name because the Display IE is not present in the incoming message.

For call modifications, certain interfaces such as EURO APAC NI1 NI2 do not support UPDATE NOTIFY/FACILITY messages. In such instances the presentation indicator (PI) from the SETUP message is retained, and if PI received in SETUP message is modified by this feature, the updated PI is retained.

The QSIG MCDN interface supports UPDATE NOTIFY/FACILITY messages.

Appropriate error messages are displayed if a wrong input is given to AUXP while configuration. The error messages are listed below:

- SCH0387: Unable to match input with stored mnemonics – this error message appears when the system is not able to match input entered with the existing (expected) inputs.
- SCH0560: Wrong number of inputs – this error message appears when more than one input is given while configuring AUXP.

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 (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.

Display Calling Party Denied

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

EuroISDN Trunk - Network Side

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

Feature Group D

If an incoming Feature Group D (FGD) call terminates at a 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 Mail

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

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

Meridian MAX

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

Meridian 911

An incoming 911 call with ANI information will always display the ANI digits on the terminating 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.

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.

Note:

The asterisk (*) used to introduce a pause while outpulsing digits is supported on analog and DTI trunks, but not supported on ISDN trunks. On ISDN trunks, if the OPAO feature is enabled, the asterisk (*) is outpulsed as a called party digit.

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 218: LD 57- Define the FFC for CPP feature. on page 414
2. Table 219: LD 16 - Define Privacy Indicators. on page 415
3. Table 220: LD 10/11 - Activate Calling Party per-line blocking. on page 417

Table 218: 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		Customer number
	0-99	Range for Large System , Media Gateway 1000B, and CS 1000E system.
FFCT	(NO) YES	Flexible Feature Confirmation Tone.
...		
CODE	CPP	FFC type to be altered.

Promp t	Response	Description
		<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 telephones, 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 219: LD 16 - Define Privacy Indicators.

Promp t	Response	Description
REQ	NEW	Add new data.
	CHG	Change existing data.
TYPE	RDB	Route Data Block.
CUST		Customer number
	0-99	Range for Large System , Media Gateway 1000B, and CS 1000E system.
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

Prompt	Response	Description
		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.
- 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 prompted only when you install the CPP package; the trunk route type is COT, DID, FEX, TIE, or WAT; the ISDN option is YES; the route is Incoming and Outgoing (IAO) or Incoming Only Trunk (ICT); and on all ISDN interfaces.
- AUXP	(NO) YES	Calling Party Privacy Indicator is honored for auxiliary applications. Calling Party Privacy Indicator is ignored for auxiliary applications. **Note:** AUXP is prompted only when you install the CPP package; the trunk route type is COT, DID, FEX, TIE, or WAT; the ISDN option is YES; the route is Incoming and Outgoing (IAO) or Incoming Only Trunk (ICT); and on all ISDN interfaces. AUXP is configurable only when PII is NO.

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 220: LD 10/11 - Activate Calling Party per-line blocking.

Prompt	Response	Description
REQ	NEW	Add new data.
	CHG	Change existing data.
TYPE	nnnn	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.
...		
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 Code is 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.

Calling Line Restriction Override Feature

With Calling Line Restriction Override feature, calling party information can be unblocked on Set and CDN basis.

When the CLS on the terminating telephone (analog CLASS or digital) is configured to Calling Line Restriction override allowed (CROA), the Calling Line Identification (CLID) Presentation Indicator changes from restricted or denied to allowed. In the case of CDN this is done by configuring the CLRO prompt to YES. This applies to all local and trunk calls.

The following example illustrates the Calling Line Restriction Override Feature functionality on a telephone (digital or analog class) basis:

Example 1

1. Telephone B is configured with a Class of Service value of CROA.
2. Telephone A is a Meridian 1 proprietary telephone with the number and name restricted
3. Telephone A dials Telephone B's DN
4. The call rings on Telephone B
5. The number and name of Telephone A are presented on the display screen of Telephone B

Example 2

1. Telephone B is configured with a Class of Service value ofCROD
2. Telephone A is a Meridian 1 proprietary telephone with the number and name restricted
3. Telephone A dials Telephone B's DN
4. The call rings on Telephone B
5. The number and name of Telephone A are not displayed on Telephone B

The following example illustrates the Calling Line Restriction Override Feature functionality on a CDN basis.

Example 1

1. In the CDN data block, the CLRO prompt is configured to YES
2. Telephone A is a Meridian 1 proprietary telephone with number and name restricted
3. Telephone A dials CDN
4. The calling number is added in the message sent to the CCR

Example 2

1. In the CDN data block, the CLRO prompt is configured to NO
2. Telephone A is a Meridian 1 proprietary telephone with the number and name restricted

3. Telephone A dials CDN
4. The calling number is not present in the message sent to the CCR

Feature Interactions: Automatic Call distribution

When there is an incoming call to a CDN whose CLRO prompt is configured to YES, then the calling number is added in the message which is sent to the CCR application. This is applicable to all local and trunk calls.

Call Detail Recording

The current Call Detail Recording (CDR) record continues to include the Calling Party Number even if the Calling Line Restriction Override Feature is activated.

Call Party Name Display (CPND)

An incoming ISDN call with the Call Party Name Display (CPND) Indicator field configured to "Presentation Denied" still displays the Calling Party Name if CROA is configured at the terminating telephone. If CROD is configured at the terminating telephone then the Calling Party name will not be displayed if Call Party Name Display (CPND) Indicator field is configured to "Presentation Denied".

Calling Line Identification Restriction (CLIR)

When an incoming call is received on a telephone with Class of Service configured to CROA, the Presentation Indicator field value becomes "Presentation Allowed" and the calling DN and name is displayed on the terminating telephone, even if the originating telephone has the Calling Line Identification Restriction feature activated. This applies to all local and trunk calls.

Display of Calling Party Denied

When an incoming call is received on a telephone with Class of Service configured to CROA, the Presentation Indicator field value becomes "Presentation Allowed" and the calling DN and name is displayed on the terminating telephone, even if the originating telephone has Display of Calling Party Denied feature activated. This applies to all local and trunk calls.

Feature Implementation:

Task summary list

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

- Table 221: LD 10: Define CLS for CLRO feature on page 420
- Table 222: LD 11: Define CLS for CLRO feature on page 420
- Table 223: LD 23: Define CLRO prompt for CLRO feature for CDN on page 421

Table 221: LD 10: Define CLS for CLRO feature

Prompt	Response	Description
REQ	NEW	Add new data.
	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 , Media Gateway 1000B, and CS 1000E system, where l = loop, s = shelf, c = card, u = unit.
DES	D…d	Designator
CUST	Xx	Customer number, as defined in LD 15
CLS	(CROD) CROA	(Deny) Allow Calling Line Identification Restriction Override

Table 222: LD 11: Define CLS for CLRO feature

Prompt	Response	Description
REQ	NEW	Add new data
	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 , Media Gateway 1000B, and CS 1000E system, where l = loop, s = shelf, c = card, u = unit.
DES	D…d	Designator

CUST	Xx	Customer number, as defined in LD 15
CLS(CROD) CROA	(CROD) CROA	(Deny) Allow Calling Line Identification Restriction Override

Table 223: LD 23: Define CLRO prompt for CLRO feature for CDN

Prompt	Response	Description
REQ	NEW	Add new data.
	CHG	Change existing data.
TYPE	CDN	Control DN data block
CLRO	(NO) YES	(Deny) Allow Calling Line Identification Restriction

Chapter 35: Calling Party Privacy Override

Contents

This section contains information on the following topics for Avaya Communication Server 1000 (Avaya CS 1000):

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.

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.

Note:

The asterisk (*) used to introduce a pause while outpulsing digits is supported on analog and DTI trunks, but not supported on ISDN trunks. On ISDN trunks, if the OPAO feature is enabled, the asterisk (*) is outpulsed as a called party digit.

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 Mail

When an incoming ISDN call with the CLID Presentation Indicator set to "Allowed" terminates on Meridian Mail, the CLID passed to Meridian Mail is recorded. The call is treated by Meridian Mail as an external call.

Calls placed by means of Through Dial can request Calling Party Privacy Override. These calls involve the person accessing Meridian Mail (mailbox user or incoming caller) dialing 0 followed by any phone number. The caller is able to dial a CPPO Flexible Feature Code plus the normal dialing sequence, following the 0. The asterisk (*) or octothorpe (#), as part of the CPPO Flexible Feature Code, are rejected by Meridian Mail. Therefore, the CPPO Flexible Feature Code can only consist of seven digits (0-9).

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 Server

As per existing operation, an incoming CPPO call routed to Symposium Call Center Server 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.

Note:
The asterisk (*) used to introduce a pause while outpulsing digits is supported on analog and DTI trunks, but not supported on ISDN trunks. On ISDN trunks, if the OPAO feature is enabled, the asterisk (*) is outpulsed as a called party digit.

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 224: LD 16 - Configure Privacy Override Indicators for a Non-ISDN route. on page 438
2. Table 225: LD 16 - Set the TCPP flag in RDB to tandem non-ISDN calls on an ISDN trunk route. on page 438
3. Table 226: LD 57 - Define the Flexible Feature Code for the Calling Party Privacy Override feature. on page 439
4. Table 227: LD 10/11 - Activate Calling Party Number and Name per-line blocking. on page 440

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 224: 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		Customer number
	0-99	Range for Large System , Media Gateway 1000B, and CS 1000E system.
ROUT		Route number
	0-511	Range for Large System , Media Gateway 1000B, and CS 1000E system.
…		
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 225: 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.

Prompt	Response	Description
TYPE	RDB	Route Data Block.
CUST		Customer number
	0-99	Range for Large System , Media Gateway 1000B, and CS 1000E system.
ROUT		Route number **Note:** All ISDN trunk routes are CPPO configurable.
	0-511	Range for Large System , Media Gateway 1000B, and CS 1000E system.
…		
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 226: 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		Customer number
	0-99	Range for Large System , Media Gateway 1000B, and CS 1000E system.
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.

Prompt	Response	Description
CODE	CPPO	CPPO Flexible Feature Code
- 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 227: LD 10/11 - Activate Calling Party Number and Name per-line blocking.

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 , Media Gateway 1000B, and CS 1000E system, where l = loop, s = shelf, c = card, u = unit.
DES	d…d	Designator The response d...d represents an Office Data Administration System (ODAS) Station Designator of 1-6 alphanumeric characters.
CUST		Customer number
	0-99	Range for Large System , Media Gateway 1000B, and CS 1000E system.
…		
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 36: 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, M3902, M3903, M3904, M3905).
- Applications: Avaya CallPilot, Customer Controlled routing, Meridian Mail, 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 37: CLID Redirecting Number Enhancement

Contents

This section contains information on the following topics for Avaya Communication Server 1000 (Avaya CS 1000):

Description

The Calling Line Identification (CLID) Redirecting Number Enhancement feature allows network administrators to select the redirecting number that displays for calls which redirect a number of times across a network.

The main benefits of this enhancement are:

- the appropriate redirecting number displays on the terminating telephone
- the call redirects to the appropriate voicemail box, if applicable

The desired number is selected from the incoming call redirection information based on the following parameters.

Incoming diverted call with multiple internal and/or external diversions

The configurable option "ARDN" in the Route Data Block (LD 16) operates in the following ways in the situations listed below:

1. When the ARDN prompt is set to NO (default), the originally called number (OCN) is displayed or the voicemail box associated with that number is utilized, if applicable.
2. When the ARDN prompt is set to YES, the last redirecting number is displayed or the voicemail box associated with that number is utilized, if applicable.
3. When the ARDN prompt is set to RPO (Redirecting number for Public OCN), the system checks the originally called number to see if it is a Private or Public number (that is, whether the call is first forwarded from a Private network or the Public network).
    - If the OCN is Private, the OCN is selected.
    - If the OCN is Public, the last redirecting number is selected.

Set the response to the ARDN prompt on the incoming trunk route at the terminating node.

Incoming calls to CDN acquired by SCCS

For calls routed by Symposium Call Center Server (SCCS), when both the Automatic Call Distribution (ACD) agent and the SCCS-controlled CDN have a voicemail box configured, the mail box routing decision is based on the prompt "CMB" in the ACD Data Block (LD 23).

- If the CMB prompt is set to YES, the call is connected to the CDN's mailbox.
- If the CMB prompt is set to NO (default), the call is connected to the agent's mailbox.

This feature applies to the display of the terminating telephone. The display on the originating telephone is not affected.

CLID Redirecting Number Enhancement supports the following redirection features:

- Call Forward All Calls (on-site or networkwide)
- Call Forward No Answer
- Call Forward Busy
- Call Waiting Redirection

- Hunting
- Call Pickup

Operation with ARDN = NO

Examples of call diversions across a network with the ARDN prompt defined as NO are illustrated in Figure 49: Call is diverted more than once, ending with a diversion to an external destination on page 449, , and Figure 50: Call is diverted more than once, ending with a diversion to an internal destination on page 450.

Diversion ending with an external destination

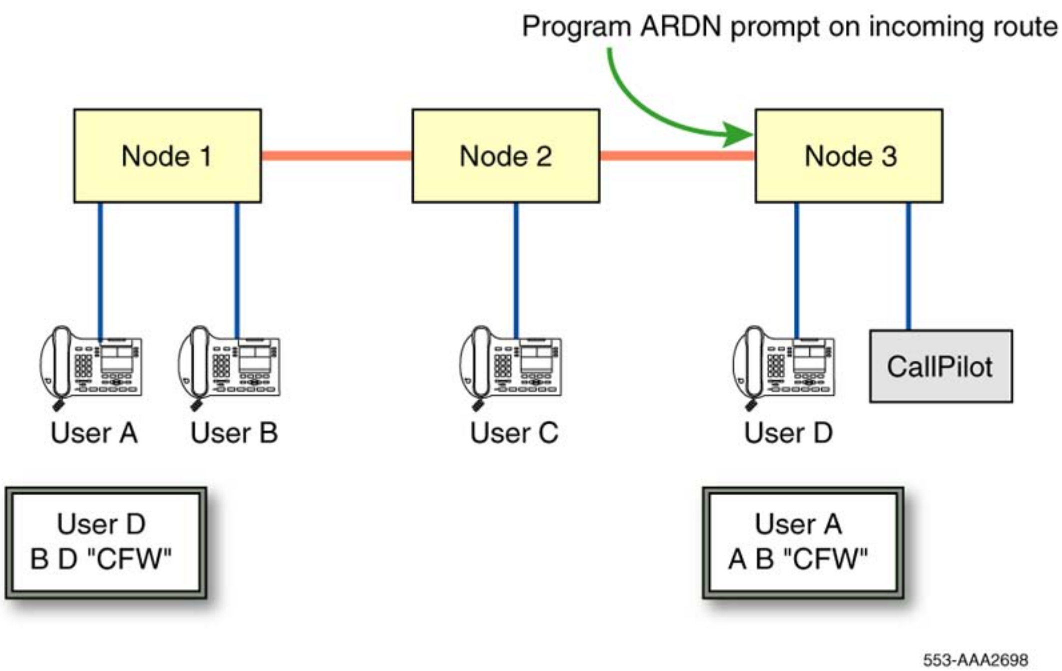

Figure 49: Call is diverted more than once, ending with a diversion to an external destination

Note:

The information presented to the displays of the originating user's telephone and the terminating user's telephone is shown in the boxes at the bottom of the diagram. The Reason for Redirection Code for the Call Forward All Calls feature ("CFW") is used in the example. Reason for Redirection Codes for the other supported redirection features can also appear on the display.

Sequence of events

1. User A calls User B.
2. User B's calls are redirected to User C.

3. User C's calls are redirected to User D or voicemail.
4. When the call is presented to:
    a. telephone D, the display shows the DN of telephone A followed by the DN of telephone B and the Reason for Redirection Code associated with the original call diversion.
    b. a voicemail system, it enters User B's mailbox. The greeting indicates the reason for the original diversion of User B's calls.

The redirection of calls from User B's telephone can be over TDM trunks (Private or Public) or IPT/virtual trunks.

Diversion ending with an internal destination

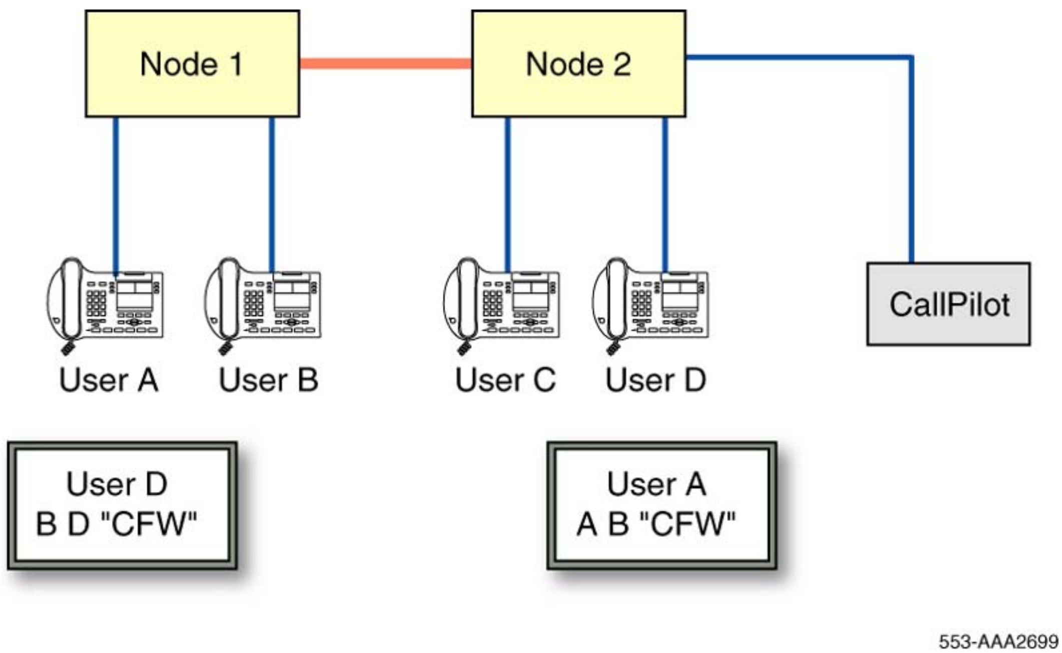

Figure 50: Call is diverted more than once, ending with a diversion to an internal destination

Note:
The information presented to the displays of the originating user's telephone and the terminating user's telephone is shown in the boxes at the bottom of the diagram. The Reason for Redirection Code for the Call Forward All Calls feature ("CFW") is used in the example. Reason for Redirection Codes for the other supported redirection features can also appear on the display.

Sequence of events

1. User A calls User B.
2. User B's calls are redirected to User C.

3. User C's calls are redirected to User D or voicemail.
4. When the call is presented to:
    a. telephone D, the display shows the DN of telephone A followed by the DN of telephone B and the Reason for Redirection Code associated with the original call diversion.
    b. a voicemail system, it enters User B's mailbox. The greeting indicates the reason for the original diversion of User B's calls.

Diversion starting with an internal destination and ending with an external destination

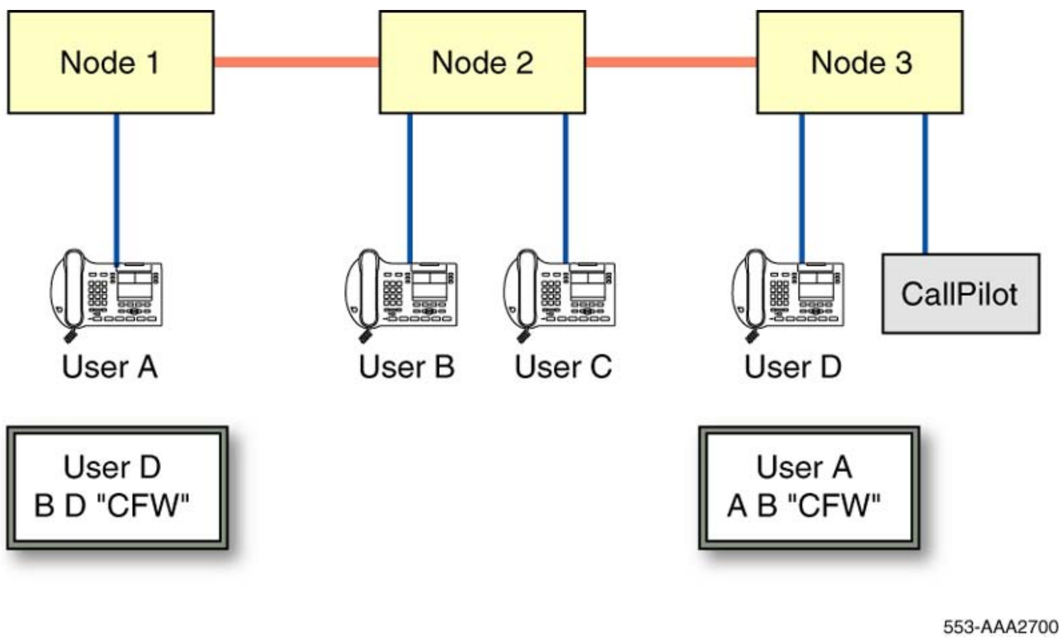

Figure 51: Call is diverted more than once, starting with an internal diversion and ending with a diversion to an external destination

Note:

The information presented to the displays of the originating user's telephone and the terminating user's telephone is shown in the boxes at the bottom of the diagram. The Reason for Redirection Code for the Call Forward All Calls feature ("CFW") is used in the example. Reason for Redirection Codes for the other supported redirection features can also appear on the display.

Sequence of events

1. User A calls User B.
2. User B's calls are redirected to User C.

3. User C's calls are redirected to User D or voicemail.
4. When the call is presented to:
    a. telephone D, the display shows the DN of telephone A followed by the DN of telephone B and the Reason for Redirection Code associated with the original call diversion.
    b. a voicemail system, it enters User B's mailbox. The greeting indicates the reason for the original diversion of User B's calls.

Operation with ARDN = YES

Examples of call diversions across a network with the ARDN prompt defined as YES are illustrated in Figure 52: Call is diverted more than once, ending with a diversion to an external destination on page 453, Figure 53: A call is diverted more than once, ending with a diversion to an internal destination on page 454, and Figure 54: A call is diverted more than once, starting with an internal diversion, ending with a diversion to an external destination on page 455.

Important:

If there are multiple call diversions on the system called Node 2 in our examples, the behavior of the feature is the same when ARDN = NO or YES, except that the Reason for Redirection Code depends on the setting. The code associated with the final diversion is displayed if ARDN = YES. The code associated with the original diversion is displayed if ARDN = NO.

Diversion ending with an external destination

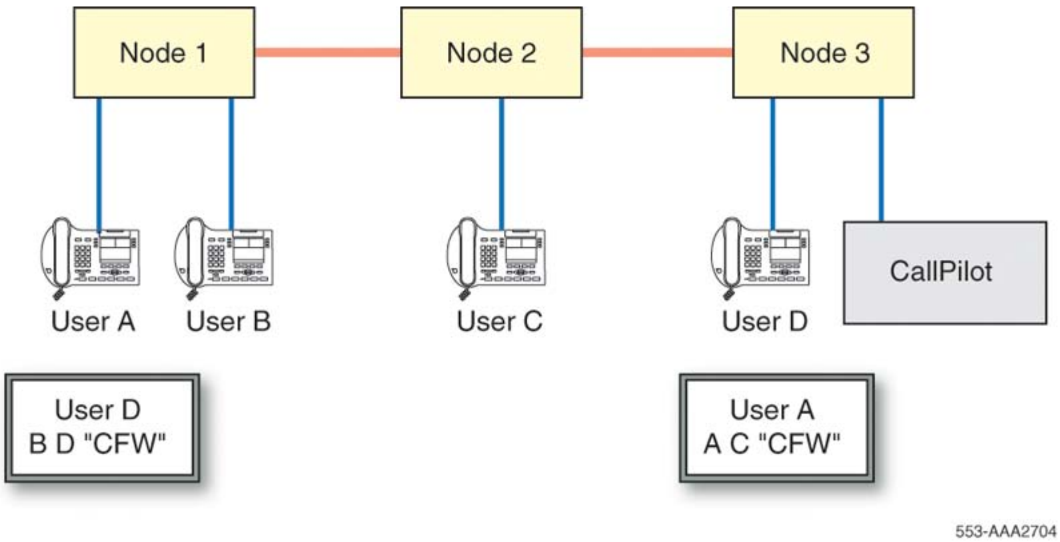

Figure 52: Call is diverted more than once, ending with a diversion to an external destination

Note:

The information presented to the displays of the originating user's telephone and the terminating user's telephone is shown in the boxes at the bottom of the diagram. The Reason for Redirection Code for the Call Forward All Calls feature ("CFW") is used in the example. Reason for Redirection Codes for the other supported redirection features can also appear on the display.

Sequence of events

1. User A calls User B.
2. User B's calls are redirected to User C.
3. User C's calls are redirected to User D or voicemail.
4. When the call is presented to:
    a. telephone D, the display shows the DN of telephone A followed by the DN of telephone C and the Reason for Redirection Code associated with the final call diversion.
    b. a voicemail system, it enters User C's mailbox. The greeting indicates the reason for the final diversion of User B's calls.

The redirection of calls from User B's telephone can be over TDM trunks (Private or Public) or IPT/virtual trunks.

Diversion ending with an internal destination

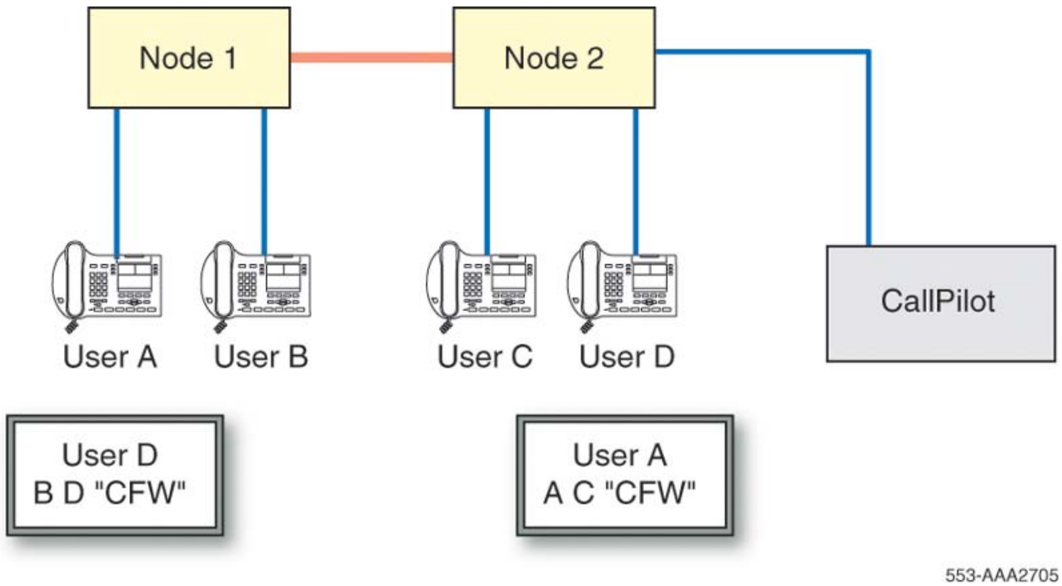

Figure 53: A call is diverted more than once, ending with a diversion to an internal destination

Note:
The information presented to the displays of the originating user's telephone and the terminating user's telephone is shown in the boxes at the bottom of the diagram. The Reason for Redirection Code for the Call Forward All Calls feature ("CFW") is used in the example. Reason for Redirection Codes for the other supported redirection features can also appear on the display.

Sequence of events

1. User A calls User B.
2. User B's calls are redirected to User C.
3. User C's calls are redirected to User D or voicemail.
4. When the call is presented to:
    a. telephone D, the display shows the DN of telephone A followed by the DN of telephone C and the Reason for Redirection Code associated with the final call diversion.
    b. a voicemail system, it enters User C's mailbox. The greeting indicates the reason for the final diversion of User B's calls.

Diversion starting with an internal destination, ending with an external destination

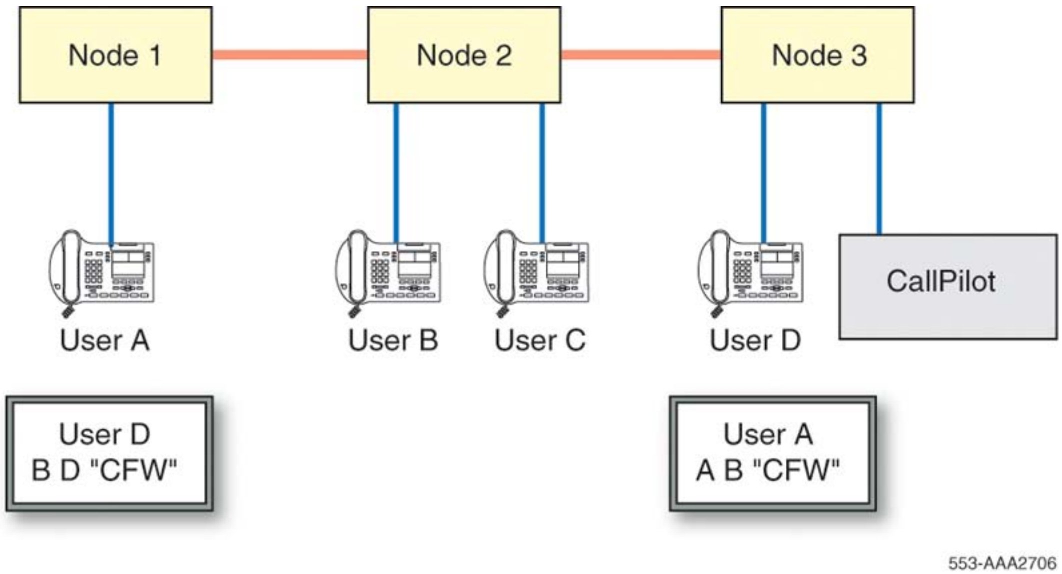

Figure 54: A call is diverted more than once, starting with an internal diversion, ending with a diversion to an external destination

Note:

The information presented to the displays of the originating user's telephone and the terminating user's telephone is shown in the boxes at the bottom of the diagram. The Reason for Redirection Code for the Call Forward All Calls feature ("CFW") is used in the example. Reason for Redirection Codes for the other supported redirection features can also appear on the display.

Sequence of events

1. User A calls User B.
2. User B's calls are redirected to User C.
3. User C's calls are redirected to User D or voicemail.
4. When the call is presented to:
    a. telephone D, the display shows the DN of telephone A followed by the DN of telephone B and the Reason for Redirection Code associated with the final call diversion.
    b. a voicemail system, it enters User B's mailbox. The greeting indicates the reason for the final diversion of User B's calls.

Operation with ARDN = RPO

For ARDN = RPO, the information in the Type of Number (TON) field associated with the call is used to identify whether the OCN is Public or Private.

If the OCN is Public and the redirecting party is in the Public network, a voicemail greeting is given based on the final redirecting number.

If OCN is Private and the redirecting party is in the Private network, the voicemail greeting given is based on the original or final diversion, depending on the setting for the ARDN prompt.

If both the Type of Number (TON) and the numbering plan for the OCN are UNKNOWN, then the system decides whether to treat the call as Public or Private in the following ways. If:

- the last in the series of trunks used for the call is a PSTN trunk (DID, FEX, WAT, COT), the number is considered to be Public
- the call is a Network Attendant Service (NAS) call, it is treated as a Public call

An example of a call diversion across a network with the ARDN prompt defined as RPO is illustrated in Figure 55: Feature operation with ARDN = RPO on page 456.

Originally called number is in the Public network

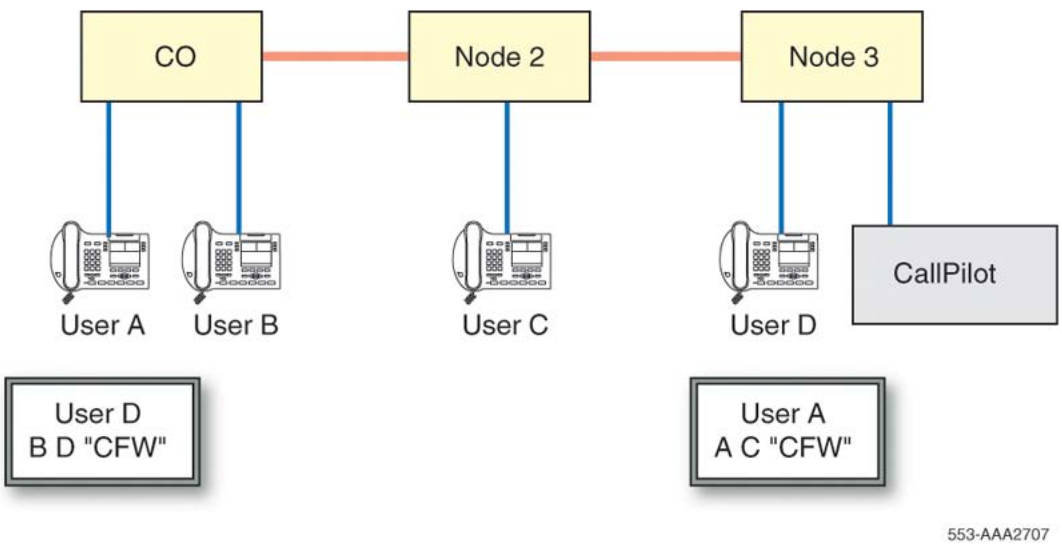

Figure 55: Feature operation with ARDN = RPO

Note:
The information presented to the displays of the originating user's telephone and the terminating user's telephone is shown in the boxes at the bottom of the diagram. The Reason for Redirection Code for the Call Forward All Calls feature ("CFW") is used in the example.

Reason for Redirection Codes for the other supported redirection features can also appear on the display.

Sequence of events

1. User A calls User B.
2. User B's calls are redirected to User C.
3. User C's calls are redirected to User D or voicemail.
4. When the call is presented to:
    a. telephone D, since the originally called number (OCN) in this case is Public, the display shows the DN of telephone A followed by the DN of telephone C and the Reason for Redirection Code associated with the final call diversion.
    b. a voicemail system, it enters User C's mailbox. User B does not have a voice mailbox on the Private network. The greeting indicates the reason for the final diversion of User C's calls.

Originally called number is in the Private network

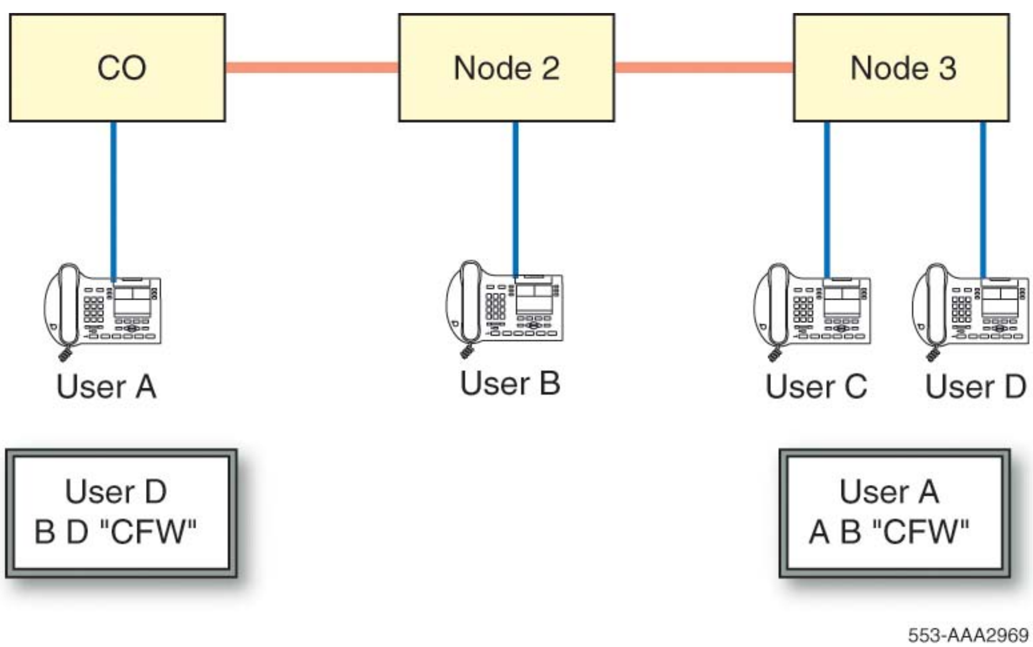

Figure 56: Feature operation with ARDN = RPO

Note:
The information presented to the displays of the originating user's telephone and the terminating user's telephone is shown in the boxes at the bottom of the diagram. The Reason for Redirection Code for the Call Forward All Calls feature ("CFW") is used in the example. Reason for Redirection Codes for the other supported redirection features can also appear on the display.

Sequence of events

1. User A calls User B.
2. User B's calls are redirected to User C.
3. User C's calls are redirected to User D or voicemail.
4. When the call is presented to:
    a. telephone D, since the originally called number (OCN) in this case is Private, the display shows the DN of telephone A followed by the DN of telephone B and the Reason for Redirection Code associated with the original call diversion.
    b. a voicemail system, it enters User B's mailbox. The greeting indicates the reason for the original diversion of User B's calls.

Summary of ARDN = NO, YES, and RPO

Figure 57: Example of a call diverted more than once on page 459 illustrates a call involving multiple diversions, The following three tables illustrate the results that occur when variables are set in different ways and various parameters are, or are not, available.

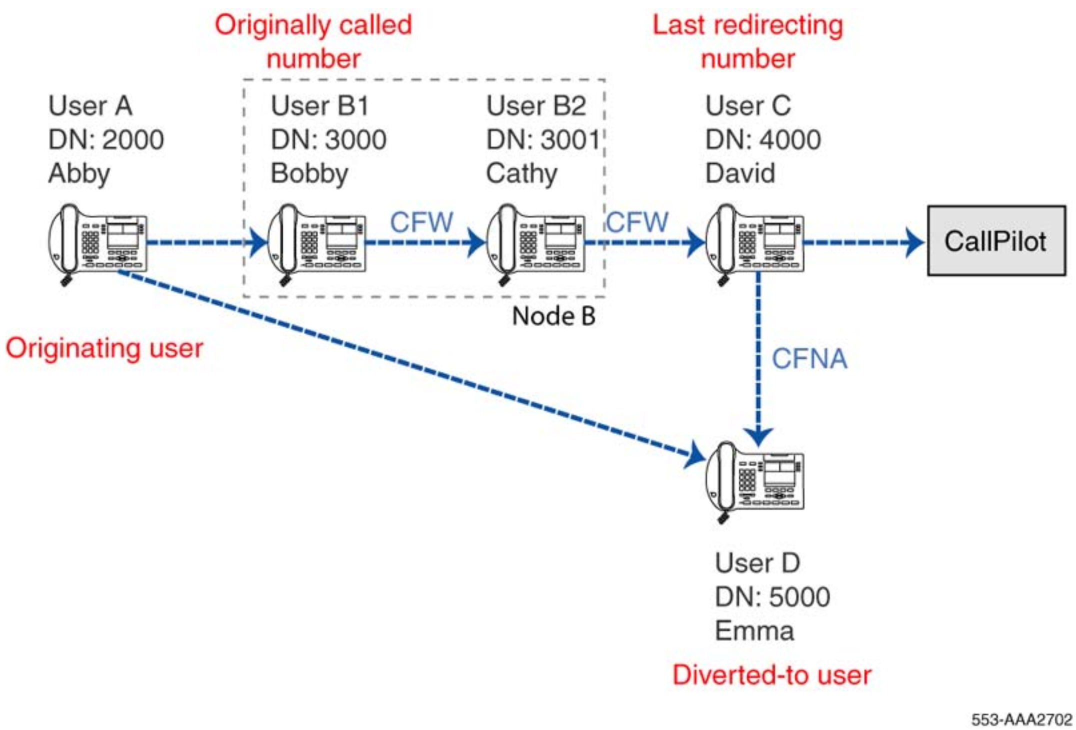

Figure 57: Example of a call diverted more than once

Table 228: Results with Route Data Block programmed with ARDN = NO

OCN available?	LRN available?	TON of OCN	Display on endpoint telephone	Voicemail greeting given belongs to
Yes	Yes	*	Abby H2000 H3000 "CFW"	User B1 (DN 3000)
Yes	No	*	Abby H2000 H3000 "CFW"	User B1 (DN 3000)
No	Yes	*	Abby H2000 H4000 "FNA"	User C (DN 4000)
No	No	*	Abby H2000	User A (DN 2000) (considers the call a direct call to voicemail)
OCN = Originally Called Number; LRN = Last Redirecting Number; TON of OCN = Type of Number of Originally Called Number * This field is not a factor in this situation.				

Table 229: Results with Route Data Block programmed with ARDN = YES

OCN available?	LRN available?	TON of OCN	Display on endpoint telephone	Voicemail greeting given belongs to
Yes	Yes	*	Abby H2000 H4000 "FNA"	User C (DN 4000)
No	Yes	*	Abby H2000 H4000 "FNA"	User C (DN 4000)
Yes	No	*	Abby H2000 H3000 "CFW"	User B1 (DN 3000)
No	No	*	Abby H2000	User A (DN 2000) (considers the call a direct call to voicemail)
OCN = Originally Called Number; LRN = Last Redirecting Number; TON of OCN = Type of Number of Originally Called Number * This field is not a factor in this situation.				

Table 230: Results with Route Data Block programmed with ARDN = RPO

OCN available?	LRN available?	TON of OCN	Display on endpoint telephone	Voicemail greeting given belongs to
Yes	Yes	Private	Abby H2000 H3000 "CFW"	User B1 (DN 3000)
Yes	Yes	Public	Abby H2000 H4000 "FNA"	User C (DN 4000)
No	Yes	*	Abby H2000 H4000 "FNA"	User C (DN 4000)
Yes	No	*	Abby H2000 H3000 "CFW"	User B1 (DN 3000)
No	No		Abby H2000	User A (DN 2000) (considers the call a direct call to voicemail)
OCN = Originally Called Number; LRN = Last Redirecting Number; TON of OCN = Type of Number of Originally Called Number * This field is not a factor in this situation.				

Symposium Call Center Server (SCCS)

An example of a network with voicemail is illustrated in Figure 58: Feature operation for SCCS routed call on page 461.

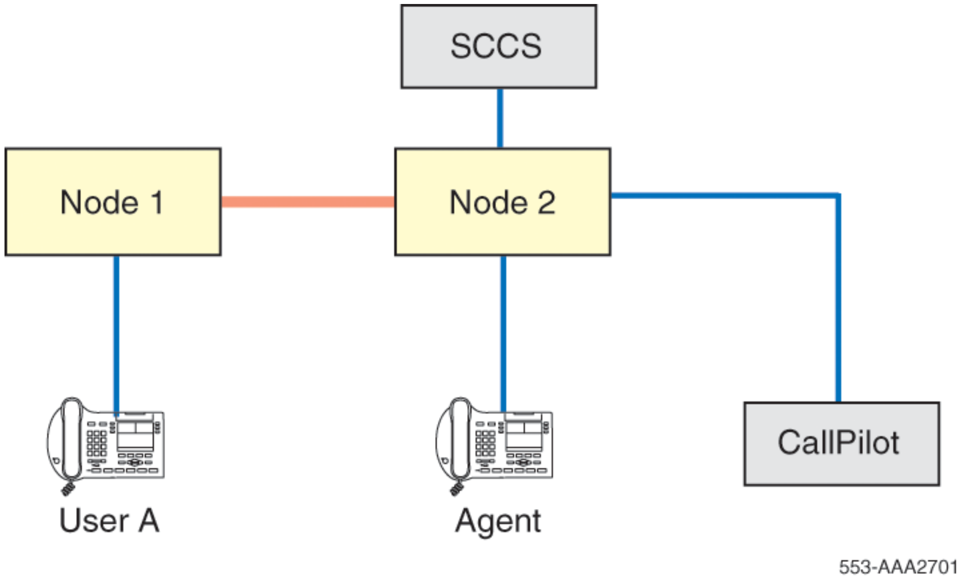

Figure 58: Feature operation for SCCS routed call

Operation without the CLID Redirecting Number Enhancement

As illustrated in Figure 58: Feature operation for SCCS routed call on page 461, if User A calls a Control Directory Number (CDN) acquired by SCCS, SCCS routes the call to an agent. If the agent's telephone has a redirection feature configured, or activated, and the call redirects to Avaya CallPilot or Meridian Mail, the call is always routed to the SCCS agent's mailbox. This is true even if the redirected call enters the voicemail system over a TDM (MCDN or QSIG) link or over an IP Peer Virtual Trunk.

Operation with the CLID Redirecting Number Enhancement

The CLID Redirecting Number Enhancement introduces the CDN Mailbox (CMB) prompt in the ACD Block (LD 23). When both the agent and the SCCS-controlled CDN have a voice mailbox configured for redirected calls, the mailbox routing decision is based on the response to the CMB prompt.If a call comes in to a CDN acquired by SCCS and the agent has redirection configured to voicemail, if CMB = NO, the call is routed to the agent's mailbox. If CMB = YES, the call is routed to the CDN's mailbox which must be configured.

Important:

The Control DN mailbox (CMB) setting in LD 23 determines whether the call routes to the agent's mailbox or the SCCS controlled CDN mailbox. The CMB prompt cannot be used to route an SCCS CDN call to a CallPilot CND mailbox.

Operating parameters

Calls can be redirected over TDM trunks (Private or Public) or over IP Peer Virtual Trunks.

CLID Redirecting Number Enhancement is supported on all telephones and consoles that have display capability and currently display CLID and redirection information. It does not modify the existing display functionality.

If the terminating telephone is an Integration telephone (a digital telephone used by third-party voicemail systems), then the redirection information presented to the telephone is used for voice mailbox routing decisions and greetings.

If redirection to voice mail is desired, proper configuration of voice mail and Network Message Services (NMS) is required.

For Meridian Companion and MDECT telephones, the Reason for Redirection Code does not display due to the size limitation of the display.

This feature does not apply to CLASS telephones. CLASS telephones do not support redirection information.

The operation of this feature is only effective if the originally called number, the redirecting number, and the corresponding Reason for Redirection codes are available at the terminating node.

Feature interactions

Attendant Alternative Answering

Attendant Alternative Answering is not considered to be a valid network redirection. When ARDN is YES, the last redirecting number before the call was presented to the attendant, and the corresponding Reason for Redirection Code is displayed on the telephone that has the Attendant Alternative Answering DN.

Call Pickup

If a call that has been redirected more than once is picked up by a user with the Call Pickup feature, the answering telephone shows the calling number, the DN of the ringing telephone, and the Reason for Redirection Code for Call Pickup, if ARDN is YES.

CallPilot

When the call terminates on a CallPilot mailbox, the accuracy of the routing decision is based on the Present Incoming Call (PCI) message content.

With CS 1000 Release 4.5 and later, the PCI message is populated with the called party DN. The ARDN setting for network call scenarios determines mailbox routing. The Control DN Mailbox (CMB) setting in LD 23 determines whether the call routes to the agent's mailbox or the CDN's mailbox.

Call Transfer

If an incoming call is transferred, the terminating telephone displays the calling DN. If an incoming diverted call is transferred, the terminating telephone does not display redirection information. This feature does not affect these situations.

If a transferred call undergoes further redirections and the ARDN prompt is set to YES, then instead of the OCN, the last redirecting DN is displayed, along with the CLID of the calling party.

Conference

The redirection information is not displayed on the terminating telephone if the incoming call at the terminating switch is part of a Conference. This feature does not affect the operation of Conference as it relates to network-wide redirections.

Dialed Name Display

The operation of the Dialed Name Display (DND) Class of Service is not affected by this feature. If the terminating telephone has DNDA programmed in the Class of Service, the originally called user's name displays as the diverted call is presented.

DPNSS-MCDN gateway

An example of a call diversion across a network with the ARDN prompt defined as NO or RPO is illustrated in Figure 59: Example of DPNSS-MCDN gateway interaction with ARDN = RPO or NO on page 464. The boxes at the bottom left and right of the figure illustrate the information that displays on the calling and called telephones respectively.

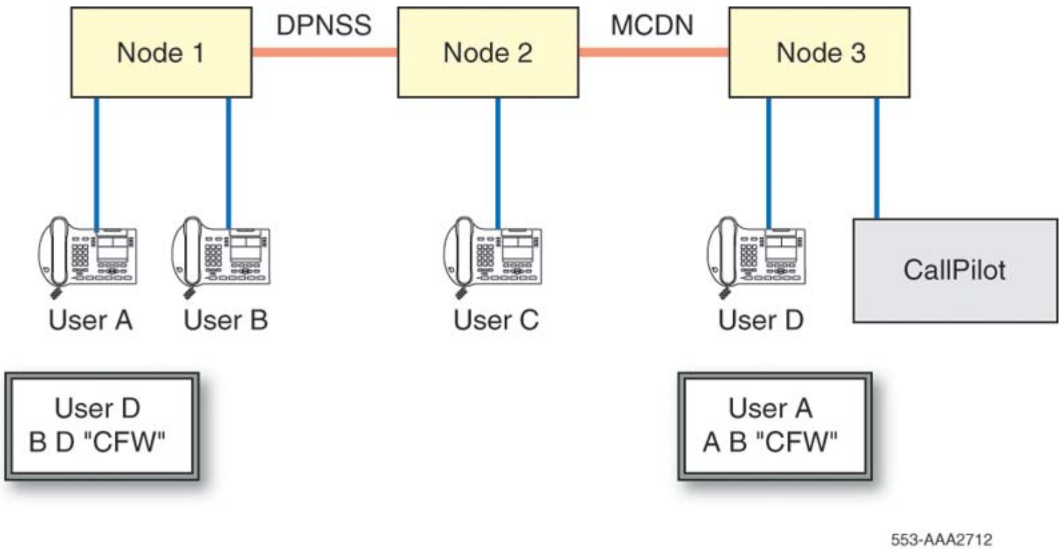

Figure 59: Example of DPNSS-MCDN gateway interaction with ARDN = RPO or NO

Sequence of events

1. User A calls User B. User B's calls are redirected to User C (using Call Forward All Calls, Call Forward No Answer, or Call Forward Busy).
2. User C's calls are redirected to User D.
3. If ARDN = YES
    a. the display on User D's telephone shows the DN of User A, followed by the DN of User C and the Reason for Redirection Code associated with the final call diversion.
    b. and if the call terminates on a voicemail system, the message is left in User C's mailbox. The Reason for Redirection of the final redirection is given in the greeting.
4. If ARDN = RPO or NO
    a. the display on User D's telephone shows the DN of User A, followed by the DN of User B and the Reason for Redirection Code associated with the original call diversion.
    b. and if the call terminates on a voicemail system, the message is left in User B's mailbox. The Reason for Redirection of the original redirection is given in the greeting.

EuroISDN

When ARDN = NO, the originally called number displays on the terminating telephone. However, with EuroISDN trunks, the originally called number is not displayed when the call is diverted externally by the Call Forward All Calls feature or transferred using Explicit Call Transfer. These features are part of the Business Network Express EuroISDN Call Diversion and EuroISDN Explicit Call Transfer features. The operation of the presentation indicator is not changed by the CLID Redirecting Number Enhancement feature.

Night Call Forward

The default operation (ARDN = NO) displays the calling DN followed by the OCN. If the ARDN prompt is YES, then the last redirecting number displays instead of the OCN.

QSIG Path Replacement

ISDN QSIG Path Replacement allows an active connection through an ISDN QSIG private network to be replaced with a more efficient connection. Path replacement service is invoked by triggers such as QSIG Call Diversion. (QSIG Call Diversion redirects calls to another telephone over a QSIG network using Call Forwarding Busy, Call Forwarding No Reply and Call Forwarding Unconditional features.)

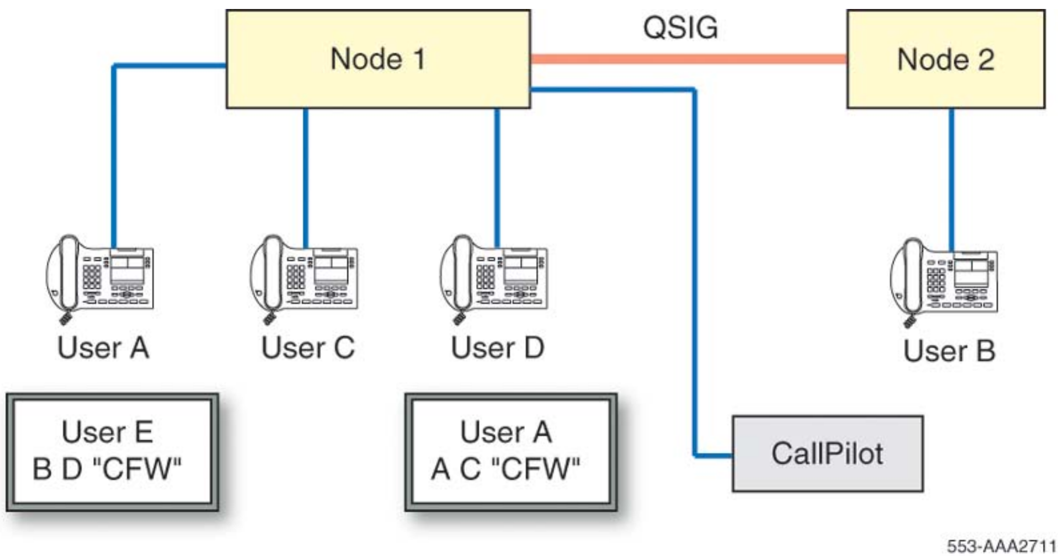

Figure 60: Example of QSIG Path Replacement interaction with ARDN= YES

Note:

The information presented to the displays of the originating user's telephone and the terminating user's telephone is shown in the boxes at the bottom of the diagram. The Reason for Redirection Code for the Call Forward All Calls feature ("CFW") is used in the example. Reason for Redirection Codes for the other supported redirection features can also appear on the display.

Sequence of events

1. User A calls User B. User B's calls are redirected to User C (using Call Forward All Calls, Call Forward No Answer, or Call Forward Busy).
2. User C's calls are redirected to User D.

3. If ARDN = YES

    a. the display on User D's telephone shows the DN of User A, followed by the DN of User C and the Reason for Redirection Code associated with the final call diversion.

    b. and if the call terminates on a voicemail system, the message is left in User C's mailbox. The Reason for Redirection of the final redirection is given in the greeting.

4. If ARDN = RPO or NO

    a. the display on User D's telephone shows the DN of User A, followed by the DN of User B and the Reason for Redirection Code associated with the original call diversion.

    b. and if the call terminates on a voicemail system, the message is left in User B's mailbox. The Reason for Redirection of the original redirection is given in the greeting.

Trunk Anti-Tromboning (TAT)

Tromboning occurs when a call goes from one node to another node and is transferred back to the originating switch on a second trunk, tying up two trunks. This can happen to forwarded or transferred calls. Trunk Anti-Tromboning optimizes (releases) the tromboned trunks for calls that are redirected or modified, after they are answered.

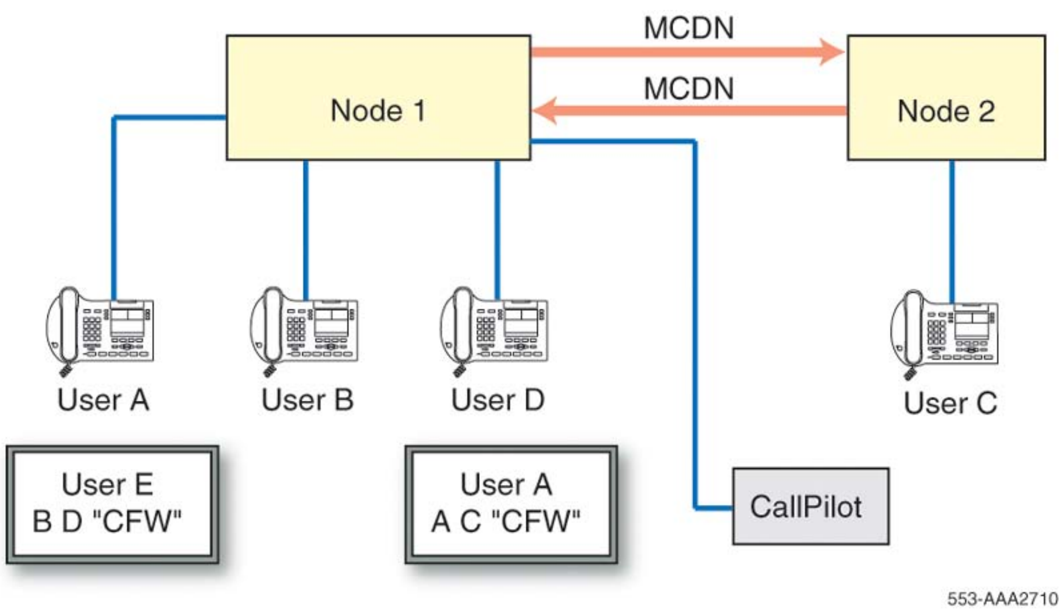

Figure 61: Example of Trunk Anti-Tromboning in operation and ARDN = YES

Note:

The information presented to the displays of the originating user's telephone and the terminating user's telephone is shown in the boxes at the bottom of the diagram. The Reason for Redirection Code for the Call Forward All Calls feature ("CFW") is used in the example.

Reason for Redirection Codes for the other supported redirection features can also appear on the display.

Sequence of events

1. User A calls User B. User B's calls are redirected to User C (using Call Forward All Calls, Call Forward No Answer, or Call Forward Busy).
2. User C's calls are redirected to User D.
3. If ARDN = YES
    a. the display on User D's telephone shows the DN of User A, followed by the DN of User C and the Reason for Redirection Code associated with the final call diversion.
    b. and if the call terminates on a voicemail system, the message is left in User C's mailbox. The Reason for Redirection of the final redirection is given in the greeting.
4. If ARDN = RPO or NO
    a. the display on User D's telephone shows the DN of User A, followed by the DN of User B and the Reason for Redirection Code associated with the original call diversion.
    b. and if the call terminates on a voicemail system, the message is left in User B's mailbox. The Reason for Redirection of the original redirection is given in the greeting.

Trunk Route Optimization (TRO)

Trunk Route Optimization (TRO) enhances routing on PRI and ISL routes for redirected calls (for example, calls redirected by Call Forward All Calls, Call Forward Busy, Call Forward No Answer, and Hunting). TRO occurs when a direct call is set up between the originating station and the final destination station. The ARDN prompt must be set on the alternate incoming MCDN route that is used after the call is optimized. Refer to Figure 62: Example of Trunk Route Optimization in operation and ARDN = YES on page 468.

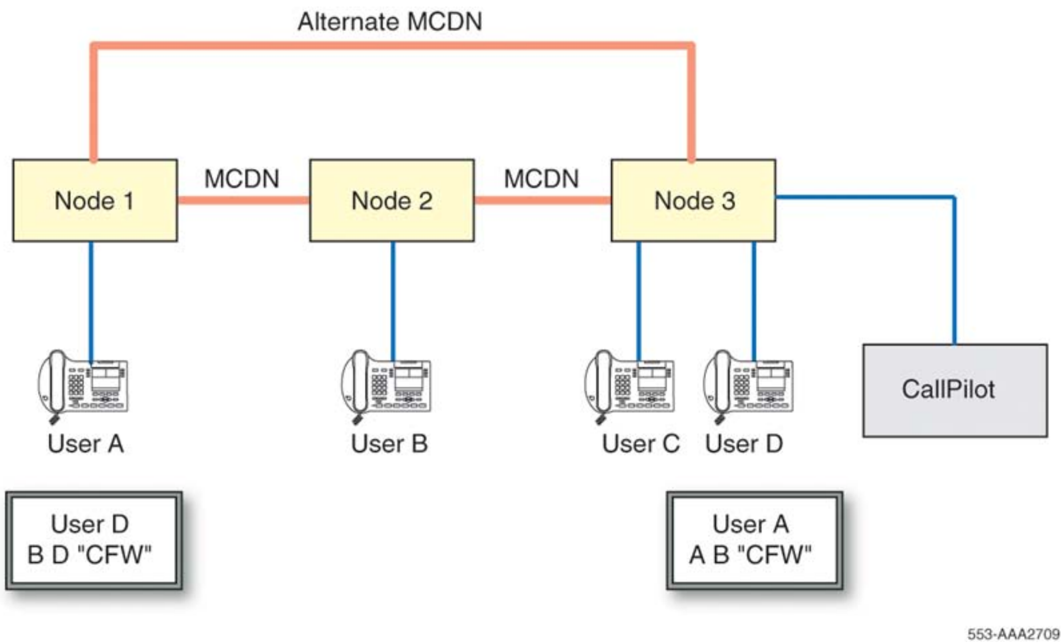

Figure 62: Example of Trunk Route Optimization in operation and ARDN = YES

Sequence of events

1. User A calls User B
2. User B has redirection1 configured to User C.
3. User C has redirection configured to User D/Call Pilot.
4. If ARDN = YES
    a. the display on User D's telephone shows the DN of User A, followed by DN of User C and the Reason for Redirection Code associated with the final call diversion.
    b. and the call terminates on a voicemail system, the message is left in User C's mailbox. The Reason for Redirection of the final redirection is given in the greeting.
5. If ARDN = RPO or NO
    a. the display on User D's telephone shows the DN of User A, followed by the DN of User B and the Reason for Redirection Code associated with the original call diversion
    b. and if the call terminates on a voicemail system, the message is left in User B's mailbox. The Reason for Redirection of the original redirection is given in the greeting.

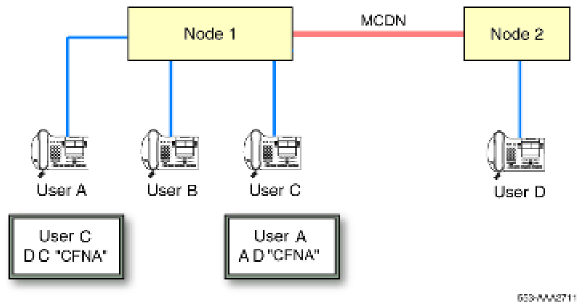

Figure 63: Example of TRO-BA in operation and ARDN = YES or NO

Sequence of events

1. User A calls User D.
2. User D Call Forward No Answer (CFNA) to User B.
3. User B CFW to User C.
4. TRO = YES on MCDN routes.
5. If ARDN = YES or NO and CLS of all users are CNDA, NAMA and DDGA.

    The display on User C's telephone is DN of user A, followed by DN of user D and the Reason for Redirection associated with the original redirection (CFNA) instead of DN of user A , followed by DN of User B and the Reason for Redirection associated with the final redirection (CFW).

Note:

The ARDN feature is not effective when all the trunks are dropped due to TRO. So the terminating User displays the originally called number (User D's DN) and not the last redirecting number (User B's DN).

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 231: LD 16 - Configure the ARDN prompt on the incoming route at the terminating switch. on page 470
2. Table 232: LD 23 - Enable redirection to SCCS CDN or agent mailbox. on page 470

Table 231: LD 16 - Configure the ARDN prompt on the incoming route at the terminating switch.

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 , Media Gateway 1000B, and CS 1000E system.

NCRD	YES	NCRD = YES, supplied message provides information for the CLID display. NCRD = NO (default) The call is redirected without the CLID redirection information, if CLID is enabled.

CCBA	(NO) YES	
ARDN		Allow last redirecting number, where:
	(NO) YES RPO	Treatment for originally called number (default). Enables treatment for last redirecting number. Enables treatment for last redirecting number, if OCN is Public.

Table 232: LD 23 - Enable redirection to SCCS CDN or agent mailbox.

Prompt	Response	Description
REQ	CHG	Change existing data.
TYPE	CDN	

NAME	(NO) YES	Display CDN name for redirected calls.
CMB	(NO) YES	Deny or Allow redirection to Control DN mailbox.

Feature operation

No specific operating procedures are required to use this feature.

Chapter 38: 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 39: D-Channel Expansion

Contents

This section contains information on the following topics for Avaya Communication Server 1000 (Avaya CS 1000):

Feature description

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

Table 233: Maximum physical I/O addresses in a Meridian 1 PBX 81C on page 477 shows a summary of the physical I/O addresses permitted in a system.

Table 233: Maximum physical I/O addresses in a Meridian 1 PBX 81C

Card Type	Application		
	TTY	AML	DCH Only
MSDL (NT6D80)	0-15 per system	0-15 per system	0-15 per group
DDCH (NTBK51AA/ NTBK51CA)	Does not apply	Does not apply	0-15 per group

Card Type	Application		
Non-MSDL DCH devices DCHI (QPC757)	Does not apply	Does not apply	0-15 per system
Any non-MSDL I/O device, such as SDI (QPC139)	0-15 per system	Does not apply	Does not apply

Operating parameters

Although all systems support the D-Channel Expansion feature, it applies only to multiple group systems. For single group systems, the maximum number of D-channels in the system remains at 64.

D-Channel Expansion allows physical I/O addresses, or device numbers (DNUM), to be duplicated in separate network groups. The duplicate DNUM must be a Multi-purpose Serial Data Link (MSDL) or Downloadable D-Channel Handler (DDCH) card with only D-Channel applications configured.

If a non-MSDL device uses a physical I/O address, then this:

- I/O address is available for use by DDCH cards, or MSDL cards running DCH applications only, in another group.
- I/O address is no longer available for other non-MSDL cards, or MDSL cards running non-DCH applications.

If an MSDL card uses a physical I/O address, and a non-DCH application is configured one of it's ports, then this:

- I/O address is available for use by DDCH cards, or MSDL cards running DCH applications only, in another network group.
- I/O address is no longer available for other non-MSDL cards, or MSDL cards running non-DCH applications.

If a DDCH card or an MSDL card, with only DCH applications configured, uses a physical I/O address, then this:

- I/O address is available for use by DDCH cards, or MSDL cards running DCH applications only, in another network group.
- I/O address is also available for one other non-MSDL card, or an MSDL card running non-DCH applications, in one other network group.

Note:

You cannot configure a duplicate physical I/O address within the same network group.

Table 234: Use of the same physical I/O address in a multi-group system

Device/application	MSDL (DCH only) DNUM x GROUP z	MSDL (non-DCH) DNUM x GROUP z	Non-MSDL DNUM x GROUP z
MSDL (DCH only) DNUM x GROUP z	valid	valid	valid Adjacent Devices on page 479
MSDL (non-DCH) DNUM x GROUP z	valid	not valid	not valid
Non-MSDL DNUM x GROUP z	valid **Note:** Adjacent Devices on page 479	not valid	not valid
• x = I/O device number • y = Group number • z = Alternate group number			

Adjacent Devices

Non-MSDL I/O devices can appropriate one or more pairs of physical device numbers. Switch settings on the hardware define the device numbers. The adjacent device is the second device number of the pair. Quad SDIs (QPC841, NT8D41BA) can have two separate pairs of adjacent device numbers.

When one device number of the pair is configured, the adjacent device number is reserved for the same device type. Therefore, both adjacent device numbers are considered used, even if only one is configured. This is consistent with existing operation. The adjacency rule can cause exceptions to Table 234: Use of the same physical I/O address in a multi-group system on page 479.

For example, a system has an MSDL, non-DCH (any single port not configured as a DCH), configured as DNUM 4 in GROUP 0. The system also has an MSDL (DCH only) configured as DNUM 5 in GROUP 0. To configure DCHI (Non-MSDL) DNUM 5 in GROUP 1 is not valid. This design appears valid in Table 234: Use of the same physical I/O address in a multi-group system on page 479, however DCHI 5 has an adjacent DNUM 4, which is a non-MSDL. Because another device (MSDL non-DCH) uses DNUM 4, the operation is not valid.

Feature interactions

License

The maximum number of D-Channels in a system is one of the License limits in the system. The keycode file defines the License limits in an IODU/C based system. The DCH limit is set in the keycode generation process. If the DCH limit is 64, the Keycode Generation group can change the DCH limit to a maximum of 255.

Network Capacity Expansion

The D-Channel expansion feature increases the number of physical I/O addresses for DCH to 16 per network group. The limit of physical I/O addresses in a multiple group system depends on the number of groups in the system. The Network Capacity Expansion feature increases the maximum number of network groups allowed in a system to eight.

Feature packaging

This feature requires the following packages:

- Integrated Services Digital Network (ISDN) package 145
- Multi-purpose Serial Data Link (MSDL) package 222

This feature requires at least one of the following packages:

- Primary Rate Access (PRA) package 146
- ISDN Signaling Link (ISL) package 147
- 2.0 Mbit/s Primary Rate Interface (PRI2) package 154

Feature implementation

Table 235: LD 17 - Define D-channels with D-channel Expansion feature.

Prompt	Response	Description
REQ	CHG	Change existing data.
TYPE	ADAN	All input/output devices (includes D-channels).
CUST		Customer number
	0-99	Range for Large System , Media Gateway 1000B, and CS 1000E system.
ADAN		Action device and number.
	aaa	aaa = NEW, CHG, MOV, or OUT
	bbb	bbb = I/O device type
	x	x = port number (0-254)
CTYP	MSDL	Multi-purpose Serial Data Link card.
GRP	0-7	Network group number.
DNUM	0-15	Device number for I/O ports. **Note:** This limit applies to each group.
PORT	x	Port number. x = 0-3 for MSDL cards.

Note:
You can define a DDCH card as an MSDL card, but the card will have two D-channel ports instead of four.

Example of D-channel configuration in LD 17

Table 236: Example of D-channel configuration in LD 17 on page 482 shows an example of how you can configure D-channels in LD 17. In this example, you can define a TTY device using I/O address 0 (device number) in group 0. You can also define an MSDL card using I/O address 0, but it must be in a different network group. You can define additional MSDL cards using the same I/O address, depending on the number of groups in the system.

Table 236: Example of D-channel configuration in LD 17

Prompt	Response	Description
REQ	CHG	Change existing data.
TYPE	ADAN	All input/output devices (includes D-channels).
CUST		Customer number
	0-99	Range for Large System , Media Gateway 1000B, and CS 1000E system.
ADAN	NEW TTY 0	Define new TTY device on port 0.
CTYP	SDI	Single port SDI card.
GRP	0	Network group 0.
DNUM	0	Device number 0.
....	....	
ADAN	NEW DCH 100	Define new DCH device on port 100.
CTYP	MSDL	Multi-purpose Serial Data Link card.
GRP	1	Network group 1.
DNUM	0	Device number 0.
....	....	
ADAN	NEW DCH 200	Define new DCH device on port 200.
CTYP	MSDL	Multi-purpose Serial Data Link card.
GRP	2	Network group 2.
DNUM	0	Device number 0.

Note:
The MSDL cards must run DCH applications only.

Feature operation

No specific operating procedures are required to use this feature.

Chapter 40: Data Packet Network access

Contents

This section contains information on the following topics:

Feature description

PRI connections to DMS-100 allow users to access Data Packet Networks (DPNs) connected to the Central Office. The steps are listed below. Equipment configuration is shown in Figure 64: System PRI to DMS-100 Data Packet Network on page 484.

1. In coordination with DMS-100 maintenance personnel, a system 1 DN associated with a High Speed Data Module (HSDM) is specified as the port for DPN access.
2. The system software initiates a hot line call through the specified HSDM to a High Speed Data Unit (HSDU) connected to the DMS-100. The HSDU communicates with the system HSDM through the T-Link version 2 protocol. This requires the QPC720 PRI card.
3. The data is sent from the PRI through the B-channel; and the call is set up using standard ISDN D-channel messaging.
4. The HSDU and HSDM go through T-Link protocol exchange.
5. The DN of a Packet Assembler/Disassembler (PAD) output port is associated with the HSDM. The user accesses the Data Packet Network by dialing the DN of the PAD.

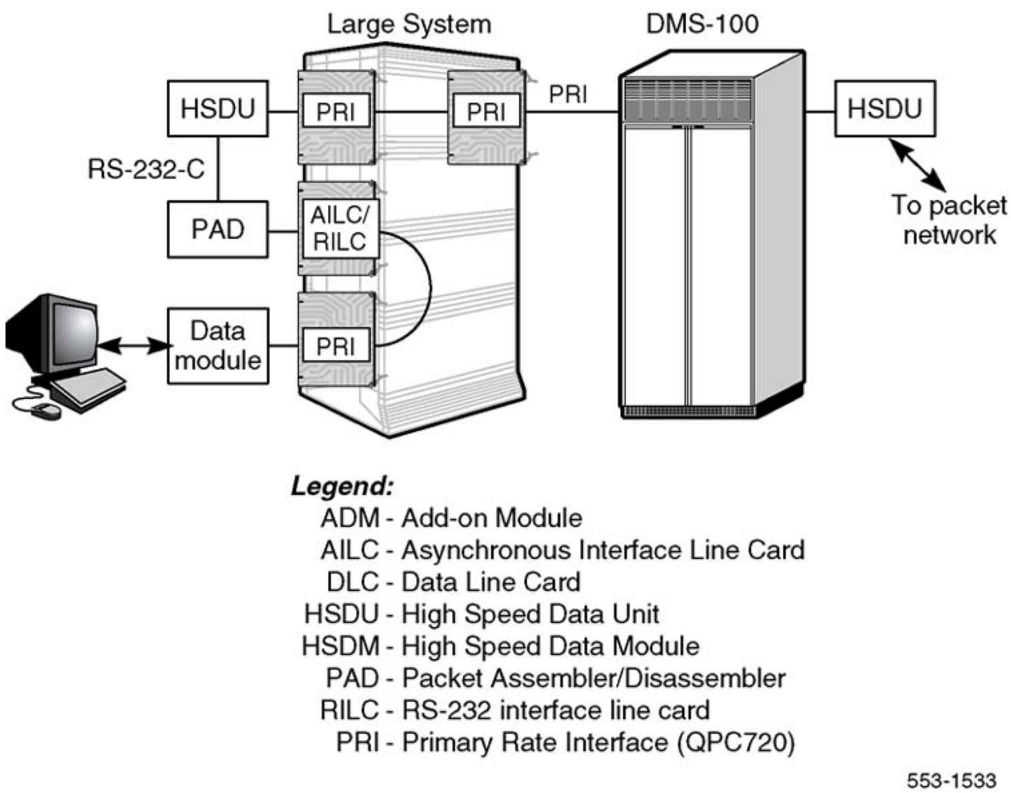

Figure 64: System PRI to DMS-100 Data Packet Network

Data Packet Network Access X.25

PRI trunks can be configured to access a Public Data Packet Network (DPN - X25) through the DMS-100 ISDN node, provided that this service is made available by the serving Central Office. On a single call basis, any B-channel can be used to access the packet 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 requires the following packages:

- Integrated Services Digital Network (ISDN) package 145
- Primary Rate Interface (PRI2) package 146

Feature implementation

There are no specific implementation procedures for this feature.

Feature operation

No specific operating procedures are required to use this feature.

Chapter 41: 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 42: 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 Meridian Mail functions at the console. It allows attendants to enter command digits during certain call states. These digits are sent to Meridian Mail over the ISDN/AP link.

These functions allow attendants to help callers operate the features offered by Meridian Mail (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 Meridian Mail
- during direct calls to Meridian Mail

The digits are sent to Meridian Mail by way of ISDN/AP KEY messages (and not by way of End to End signaling). Dialed digits are not saved by the system and are not displayed at the Attendant Console.

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

Extending source calls to Meridian Mail

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

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

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

Direct calls to Meridian Mail

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

- No call is present on the DEST key of the active loop.
- A call is established on the SRC key to the Meridian Mail agent (Class of Service of VMA).
- The SRC call to Meridian Mail is not a conference call.

The attendant cannot extend the call to a destination party using dialed digits. (Key pad input is treated as Digit Key signaling and not as dialing digits.)

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 43: Digital Trunk Interface and Primary Rate Interface Time Slot Reuse

Contents

This section contains information on the following topics:

Feature description

This feature eliminates call blocking due to unavailable time slots. This feature allows a time slot reserved for a primary function to be reused for any subsequent requirements during call processing. Therefore, all channels on Digital Trunk Interface (DTI) or Primary Rate Interface (PRI) loops are available for use.

Time slots reserved for a particular DTI or PRI channel are stored in an unprotected data structure allowing easy access, such as a call register. Each time a new path is required for the same DTI or PRI channel, the data structure can be accessed to determine if there is a reserved time slot that can be reused. If the new path is intra-group (single-group switch), the time slot can be reused. If the new path requirement is inter-group (multi-group switch), the reserved time slot can be reused if a matching junctor slot is available.

A count is kept of the number of times that a time slot has been reused. The time slot is not idled until all reservations have been canceled.

This feature applies to the following loop types on single-group switches:

- 1.5 Mbit 24-channel DTI
- 2.0 Mbit 30-channel DTI
- 1.5 Mbit 23 B+D PRI
- 2 Mbit 30 B+D PRI
- 2 Mbit Japan Digital Multiplex Interface (DMI)

Operating parameters

This feature applies only to network enhanced machines.

Feature interactions

There are no interactions with other features.

Feature packaging

No packaging requirements are specified for this feature.

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: Display of Access Prefix on CLID

Contents

This section contains information on the following topics for Avaya Communication Server 1000 (Avaya CS 1000):

Feature description

The Display of Access Prefix on Calling Line Identification (CLID) feature enhances the phone display by adding the Local, National or International prefix to the CLID display.

The four digit access prefix is a combination of the access code and the National or International prefix. The access code defines the best outgoing route for an external call and the prefix code defines National or International calls. If the trunk requires the International prefix of two digits, the access code (ACOD) is a maximum of two digits.

The Access Prefix Display on CLID feature supports:

- Digital proprietary phones: M2008, M2216, M2016, and M2317
- M2250 Attendant Consoles

The Display of Access Prefix on CLID feature supports all Integrated Services Digital Networks (ISDN) interfaces, including:

- Meridian Customer Defined Network (MCDN)
- Q-reference Signaling Point (QSIG)
- Japan Telecommunication Technology Committee (JTTC)
- Euro-ISDN
- NI2
- Asia-Pacific ISDN

Operating parameters

If CLID is not available, the trunk access code and the prefix number are displayed according to the existing feature operation.

The Display of Access Prefix on CLID feature supports the Coordinated Dialing Plan (CDP) or the Uniform Dialing Plan (UDP).

The Display of Access Prefix on CLID feature does not support D100 routes configured for ISA or NI2 routes configured for CBC.

Feature interactions

Selectable Conferee Display and Disconnect

The Selectable Conferee Display and Disconnect feature displays and allows the disconnect of conferenced phones.

Feature packaging

The Display of Access Prefix on CLID 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:

Table 237: LD 11 - Enable the Display of Access Prefix on CLID feature for digital phones.

Prompt	Response	Description
REQ	NEW	Add new data.
	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 , Media Gateway 1000B, and CS 1000E system, where l = loop, s = shelf, c = card, u = unit.
DES	x...x	Office Data Administration System Designator

Table 238: LD 12 - Enable the Display of Access Prefix on CLID for attendant consoles.

Prompt	Response	Description
REQ	NEW	Add new data.
	CHG	Change existing data.
TYPE	2250	Attendant Console type.
TN		Terminal number

Prompt	Response	Description
	l s c u	Format for Large System , Media Gateway 1000B, and CS 1000E system, where l = loop, s = shelf, c = card, u = unit.
SETN		Second Terminal Number
	l s c u	Format for Large System , Media Gateway 1000B, and CS 1000E system, where l = loop, s = shelf, c = card, u = unit.
CUST		Customer number
	0-99	Range for Large System , Media Gateway 1000B, and CS 1000E system.
ANUM	1-63	Attendant Number.
...	...	
DAPC	DAPA	Allow Display of Access Prefix on CLID. (DAPD) = Deny Display of Access Prefix on CLID.

Table 239: LD 15 - Configure the access prefix table.

Prompt	Response	Description
REQ	CHG	Change existing data.
TYPE	FTR	Customer Features and options.
CUST		Customer number
	0-99	Range for Large System , Media Gateway 1000B, and CS 1000E system.
...	...	
DAPC	YES	Configure Display of Access Prefix table entry option. (NO) is the default.
TBL	1-15	Table number. Table 0 is non configurable. Precede table number with "X" to remove.
-NPI	aaaa	Numbering Plan Identification. Valid entries for aaaa include: UNKN - Unknown. E164 - Numbering Plan based on E164. PRIV - Private. E163 - Numbering Plan based on E163. TELX - Telex. X121 - Data X121. NATL - National.
--TON	aaaa	Type of number. Valid entries for aaaa include: UNKN - Unknown. INTL - International. NATL - National. ESPN - ESN_SPN. LOCL - Local. ELOC - ESN_LOC. ECDP - ESN_CDP.

Prompt	Response	Description
---PREF	0-9999	Up to four digit Access Prefix for a unique NPI/TON combination in the table. Carriage return is taken as NIL Access Prefix value.
	#	Wild Character for replacement of any digit. The entry of "#" for wild card character is stored and displayed as "*". **Note:** The entry of "*" would be misinterpreted by the overlay supervisor for the standard overlay operation.)
	X	Reset the access prefix value to NIL.
--TON	...	Repeat for every value of TON for the particular NPI. Entry of Carriage return prompt is NPI.
-NPI	...	Repeat for every value of NPI.

Table 240: LD 16 - Enable the Display of Access Prefix on CLID feature.

Prompt	Response	Description
REQ	NEW	Add new data.
	CHG	Change existing data.
TYPE	RDB	Route Data Block.
CUST		Customer number
	0-99	Range for Large System , Media Gateway 1000B, and CS 1000E system.
ROUT		Route number
	0-511	Range for Large System , Media Gateway 1000B, and CS 1000E system.
...	...	
ISDN	YES	Integrated Services Digital Network route.
...	...	
DAPC	YES	Enable feature at the route data block Level. (NO) = Disable feature at the route data block level.
- TBL	1-15	Prefix table number as defined in LD 15. Prompted only when ADDP is answered as YES.

Feature operation

No specific operating procedures are required to use this feature.

Chapter 45: Display of Calling Party Denied

Contents

This section contains information on the following topics for Avaya Communication Server 1000 (Avaya CS 1000):

Feature description

Display of Calling Party Denied (DPD) permits Analog (500/2500 type) phones and Meridian 1 Proprietary Phones to either allow or deny associated name and number from being displayed on other phones when involved in a call. This feature is supported on internal calls (same node) and calls placed over a Meridian Customer Defined Network (MCDN) Integrated Services Digital Network (ISDN). Display of Calling Party Denied Class of Service options are programmed on a per phone basis.

DPD uses the following Class of Service options in LD 10 and LD 11: Display Digits Allowed/Denied (DDGA/DDGD) and Name Display Allowed/Denied (NAMA/NAMD).

Display of Digits Denied (DDGD) Class of Service restricts digits from being displayed on another phone when involved in a call. However, on the calling party's phone, the dialed digits are always displayed regardless of the Class of Service assigned to the called party's phone. Display of Digits Allowed (DDGA) allows the number associated with a phone to be displayed when that phone is involved in a call.

Display of Name Denied (NAMD) Class of Service restricts the name associated with a phone from being displayed on another phone when involved in a call. Display of Name Allowed

(NAMA) allows the name associated with a phone to be displayed when that phone is involved in a call.

Table 241: Display Scenarios on Calling and Called Partys phones on page 502 outlines the possible configurations and the resulting display on both the calling and called parties' displays.

Table 241: Display Scenarios on Calling and Called Partys phones

CALLING PARTY'S CLASS of SERVICE	CALLED PARTY'S CLASS of SERVICE	DISPLAY OF CALLING PARTY	DISPLAY OF CALLED PARTY
NAMA DDGA	NAMA DDGA	Name of Called Party Called Party's Number	Name of Calling Party Calling Party's Number
NAMA DDGA	NAMD DDGD	X X X X Called Party's Number	Name of Calling Party Calling Party's Number
NAMD DDGD	NAMA DDGA	Called Party's Name Called Party's Number	X X X X – – – –
NAMD DDGD	NAMD DDGD	X X X X Called Party's Number	X X X X – – – –
NAMA DDGD	NAMD DDGA	X X X X Called Party's Number	Name of Calling Party – – – –
NAMD DDGA	NAMA DDGD	Called Party's Name Called Party's Number	X X X X Calling Party's Number

Figure 65: Calling Party and Called Party on the Same Node on page 503 illustrates the functionality of Display of Calling Party Denied (DPD) on a nodal basis when both the calling and called parties have enabled DPD Class of Service options. The calling party has configured a Digit Display Denied (DDGD) and Name Display Denied (NAMD) Class of Service. The called party has the same Class of Service options programmed. The display of the calling party's associated name and number is restricted on the called party's phone. The calling party's number is replaced by a string of dashes (– – – –), where each "-" represents a suppressed digit. The associated name is replaced by a string of four Xs (XXXX).

On the calling party's phone, the name associated with the called party is replaced by a string of four Xs (XXXX). However, the dialed digits remain displayed on the calling phone.

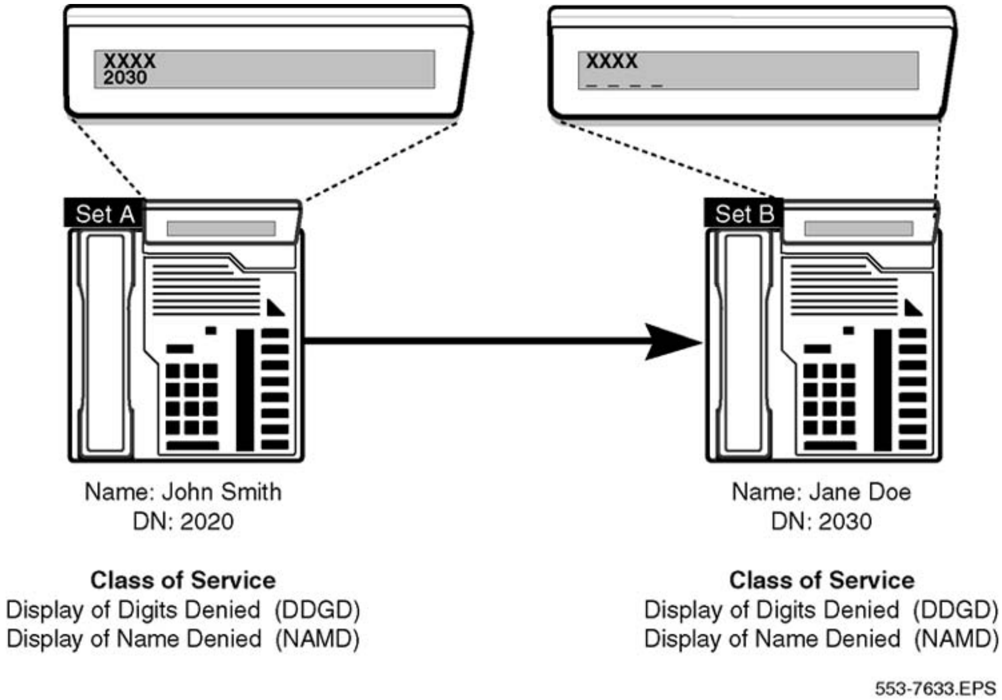

Figure 65: Calling Party and Called Party on the Same Node

As illustrated in Figure 66: ISDN Network Calls on page 504, for calls placed over a MCDN ISDN network, Display Digits Denied (DDGD) Class of Service on the calling party's phone prevents the Calling Line Identification (CLID) from being displayed on the called party's phone. In this case, the calling party's CLID is replaced by the ISDN route access code and route member number. If the calling party has configured Name Display Denied (NAMD), the calling party's associated name is not displayed on the called party's phone. The associated name is replaced by a string of four Xs (X X X X).

On the calling party's phone, the name associated with the called party is replaced by a string of four Xs (X X X X). However, the dialed digits remain displayed on the calling phone.

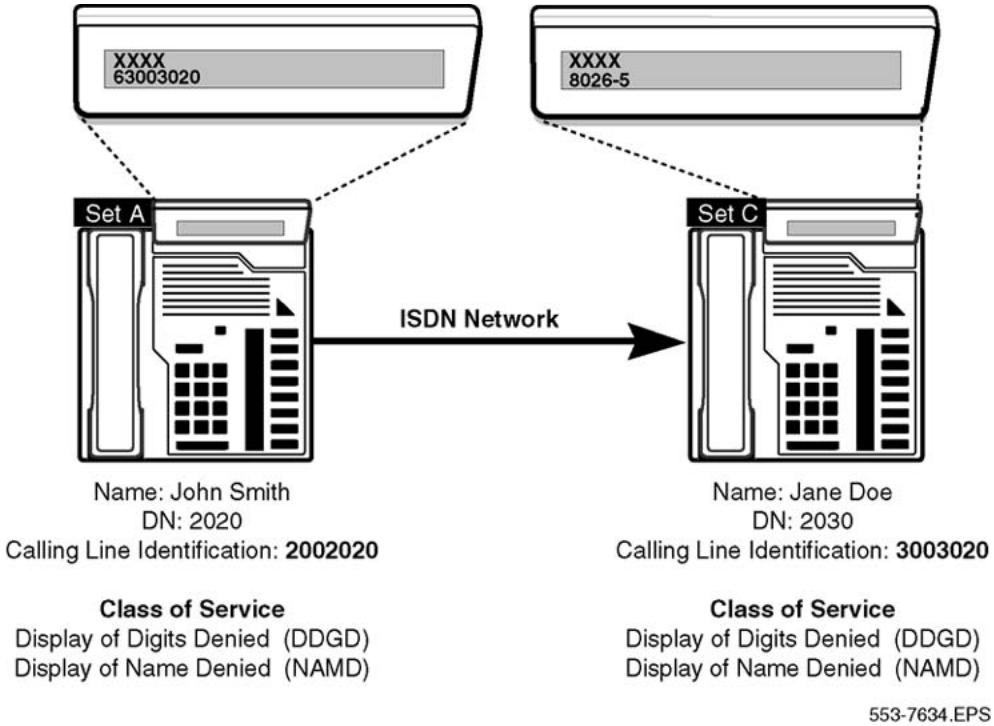

Figure 66: ISDN Network Calls

Operating parameters

Display of Calling Party Denied (DPD) is only supported on internal calls, Meridian Customer Defined Network (MCDN) ISDN networks and the interface to the public network.

This feature does not support calls over DASS/DPNSS1 trunks.

Display of Calling Party Denied is not supported on ISDN BRI phones.

The dialed digits always appear on the display of the calling party's phone. This occurs whether or not the called party has a Digit Display Denied (DDGD) Class of Service. However, if the dialed digits represent a Multiple Appearance Directory Number (MADN), the calling party's display reflects the Class of Service of the answering phone.

Feature interactions

Attendant Console

A local attendant console's display is not impacted by this feature. Display is provided to an attendant regardless of the Class of Service configured on a local phone. In order to allow calls placed over an ISDN network to be displayed on the Attendant Console when a remote phone has a denied Class of Service (DDGD/NAMD), Network Attendant Service (NAS) must be configured. Then the denied Class of Service is ignored.

Call Detail Recording

For internal calls, calling and called party's Directory Numbers are included in Call Detail Recording (CDR) records regardless of the phone's Class of Service. For MCDN ISDN network calls, the calling party's or connected party's number is included in the CDR regardless of the phone's Class of Service.

Call Forward All Type

When a phone activates any of the call forwarding features, the displays given on the calling phone and the terminating phone are in accordance with the Class of Service of the phones involved in the call.

If the terminating phone has Dialed Name Display Denied (DNDD), the display on the terminating phone reflects the name and number of the calling party and the name and the number of the forwarding phone.

If the terminating phone has Dialed Name Display Allowed (DNDA), the display on the terminating phone reflects the number of the calling party and the name and number of the forwarding phone. In both cases, the terminating phone's display is in accordance with the DPD Class of Service options of the calling and forwarding phones.

For a MCDN ISDN call, the calling party's Calling Line Identification (CLID) is replaced with the ISDN route access code (ACOD) and the route member number, and the calling party's name is replaced by a string of four Xs (X X X X).

The display given on the calling phone of an internal call, which has been forwarded to a phone within the same switch, includes the name and number of the terminating phone along with the number of the forwarding phone. If the DPD Class of Service options, which are specified for the terminating phone, indicate that the display of the name and number of the terminating phone be denied, then on the calling phone, the name of the terminating phone is replaced by a string of four Xs (X X X X). The number is replaced by dashes (- - - -). If the number of the terminating phone is blocked from being displayed on the calling phone, the number of the forwarding phone is also blocked from being displayed on the calling phone, regardless of the DPD Class of Service options of the forwarding phone. Conversely, if the display of the terminating phone's number is allowed in the calling phone, then the number of the forwarding phone is also displayed on the calling phone, irrespective of the DPD Class of Service options of the forwarding phone.

Call Hold

When a call is retrieved from hold, the calling and called parties' displays reflect their individual DPD Class of Service options.

Call Park

When the Call Park timer expires on a parked call, a phone's display reflects the Directory Number the call is parked against. The display does not include the name and DN of the calling party. When a parked call is retrieved by another phone, display information is based on the DPD Class of Service of the individual phones.

Call Pickup

When a call is picked up from another phone, the terminating phone's display is in accordance with the Class of Service of the dialed and calling phones. The calling party's display includes the dialed DN, the terminating DN and the name of the terminated phone. However, if the terminating phone has Digit Display Denied (DDGD), then both the dialed and terminating phones' DNs are blocked from the calling party's display. The same occurs when Digit Display Allowed (DDGA) is configured on the terminating phone. Both the dialed and terminating phones' DNs are displayed on the calling party's phone, regardless of the Class of Service of the dialed phone.

Call Transfer

When a phone transfers a call, display information is updated according to the Class of Service of the respective phones. This occurs for both internal and ISDN network calls.

If an unsupervised call transfer occurs on an internal call, the DN of the terminating phone is displayed to the calling party regardless of the DPD Class of Service options that are configured on the terminating phone.

Calling Party Name Display

If Calling Party Name Display Denied (CNDD) is configured for a phone, then the name associated with that phone is not displayed when it is involved in a call. This is so regardless of the DPD Class of Service Options of that phone.

Calling Party Privacy

Calling Party Privacy takes precedence over Display of Calling Party Denied on outgoing calls over a MCDN ISDN network. This precedence pertains to the display of the calling party's name and number on the called party's phone.

Calling Party Privacy does not affect internal calls using Display of Calling Party Denied.

Conference

For internal conference calls, display information is not provided on any of the conferee's phones. When setting up a conference call, by conferencing one phone at a time, the display on the conferee's phone is in accordance with the Class of Service of the individual phones.

For network conference calls, display is provided. This is in accordance with the DPD Class of Service options of the individual phones.

Dial Intercom

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

Group Call

The calling party's display shows the DN of the last phone to connect into the Group Call regardless of the Class of Service. The called phone displays the Group Number only.

Hot Line

Display information on phones in a Hot Line call is based on the individual Class of Service of each phone.

Meridian 911

An incoming M911 call with Automatic Number Identification (ANI) information always displays ANI digits on the terminating phone regardless of the calling party's DPD Class of Service.

Meridian Mail

A calling party's name and/or DN within Meridian Mail are not impacted by this feature. When a Digit Display Denied (DDGD) call is forwarded to Meridian Mail, the calling party's DN is provided to Meridian Mail and is available when reviewing messages.

Multiple Appearance Directory Number

When a Multiple Appearance Directory Number (MADN) is dialed and ringing, the calling party's display does not show name information, provided that at least one of the MADN TNs has Name Display Denied (NAMD) Class of Service. Once a call is answered, the calling party's phone reflects the DPD Class of Service options of the answering phone.

The called party's display reflects the DPD Class of Service options of the calling party's phone.

Multi Party Operations - Call Join

When three parties are joined using the Call Join capabilities of the Multi Party Operations feature, display information is not provided on any of the conferee's phones. When setting up a conference call, by conferencing one phone at a time, the display on the conferee's phone is in accordance with the individual phone's Class of Service. If one phone leaves a three party conference, display information on the remaining phones is based on the individual Class of Service of each phone.

Network Attendant Services

When a call is placed to an attendant over an ISDN network with Network Attendant Services (NAS) configured and Calling Party Name Display (CPND) equipped, the Attendant Console displays the calling party's DN and name. The same occurs when an attendant places a call over the network; however, an Attendant Console's display shows the connected party name and number.

If NAS is not configured, the name and calling/connected party's number is displayed to the attendant provided that the calling/connected party has the Display of Calling Party Denied (DPD) Class of Service options configured to Allowed. If Class of Service options are set to Denied, then this information is not displayed to the attendant.

No Hold Conference

Voice Call

Display information on phones involved in a No Hold Conference and Voice call is based on the individual Class of Service of each phone.

Feature packaging

For internal calls, Display of Calling Party Denied (DPD) requires Calling Party Name Display (CPND) package 95.

For ISDN Network calls the following packages and their dependencies are also required:

- Calling Party Name Display (CPND) package 95
- Integrated Service Digital Network (ISDN) package 145 and
- Network Attendant Services (NAS) package 159

Note:

The NAS package is required to display a calling party's name and DN on the attendant console, when using the Display of Calling Party Denied feature within an ISDN PRI private network. With NAS configured and Calling Party Name Display equipped, the attendant console will display the calling party's name and DN, regardless of the Display of Calling Party Denied Class of Service of the calling party's phone. Without NAS, if the Display of Calling Party Denied Class of Service of the calling party's phone is Denied, the name and DN of the calling party will not be displayed on the attendant console.

Feature implementation

Task summary list

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

1. Table 242: LD 10 - Deny Directory Number Display and Name Display for analog (500/2500-type) phones. on page 510
2. Table 243: LD 11 - Deny Digit Display and Name Display for digital proprietary phones. on page 510

Table 242: LD 10 - Deny Directory Number Display and Name Display for analog (500/2500-type) phones.

Prompt	Response	Description
REQ	CHG	Change existing data.
TYPE	500	500/2500-type phone data block.
TN		Terminal number
	l s c u	Format for Large System , Media Gateway 1000B, and CS 1000E system, where l = loop, s = shelf, c = card, u = unit.
...		
CLS	DDGD	Digit Display Denied DDGA = Digit Display Allowed (default).
CLS	NAMD	Name Display Denied NAMA = Name Display Allowed (default).

Table 243: LD 11 - Deny Digit Display and Name Display for digital 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 , Media Gateway 1000B, and CS 1000E system, where l = loop, s = shelf, c = card, u = unit.
...		
CLS	DDGD	Digit Display Denied DDGA = Digit Display Allowed (default).
CLS	NAMD	Name Display Denied NAMA = Name Display Allowed (default).

Feature operation

No specific operating procedures are required to use this feature.

Chapter 46: DPNSS1/DASS2 to Q.931 Gateway

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

This feature allows a system node to act as a gateway between DPNSS1, DASS2 and Q.931 (ISDN PRI) protocols.

This gateway supports these functionalities:

- Basic Call Service (circuit-switched voice calls)
- Calling Line Identification
- Connected Line Identification

- Call Diversion
- Coordinated Dialing Plan
- Network Ring Again, between DPNSS1 and Q.931 protocols

Operating parameters

This feature has this limitation. PSTN incoming trunks are not allowed access to PSTN outgoing trunks.

Feature interactions

There are no feature interactions associated with this feature.

Feature packaging

This feature is packaged as UK Program (UK) package 190 and it has the following dependencies:

- Integrated Digital Access (IDA) package 122
- Digital Private Network Signaling System (DPNSS) package 123
- Digital Access Signaling System 2 (DASS2) package 124
- Integrated Services Digital Network (ISDN) package 145
- 2.0 Mbit Primary Rate Interface (PRI2) package 154

Interworking of Network Ring Again across DPNSS and Q.931 requires the Advanced ISDN Network Services (NTWK) package 148.

Feature implementation

There are no specific implementation procedures for this feature.

Feature operation

No specific operating procedures are required to use this feature.

Chapter 47: DPNSS1 Route Optimization/ MCDN Trunk Anti-Tromboning Interworking

Contents list

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 Digital Private Networking Signalling System No.1 (DPNSS1) Route Optimisation (RO)/ Meridian Customer Defined Networking (MCDN) Trunk Anti-Tromboning (TAT) Interworking feature provides RO and TAT interworking at DPNSS1/MCDN gateway nodes.

Note:

For detailed information on the DPNSS1 Route Optimisation feature, refer to *Avaya DPNSS1 Fundamentals, NN43001-572*. For detailed information on the Trunk Anti-Tromboning feature, refer to the feature description in this book.

RO/TAT interworking scenarios

RO/TAT interworking within a DPNSS1 to MCDN gateway

The following example presents a case where RO/TAT interworking occurs within a DPNSS1 to MCDN gateway.

Note:

In this example, we have used the case where a call has been redirected due to Network Call Transfer. The same functionality would apply if the call had been redirected by Network Call Forward No Answer, and Network Hunting, or modified by Network Call Transfer or Attendant Call Transfer.

Referring to Figure 67: DPNSS1/MCDN scenario with Network Call Transfer, before RO/TAT optimisation on page 517, Station A, located at Node 1 on the DPNSS1 side of the DPNSS1/MCDN gateway, calls Station B located at Node 4 on the MCDN side of the gateway. It is to be assumed that the optimum DPNSS1 route has been selected at the originating node (the case where a non-optimum route is selected is discussed in the note following Figure 68: DPNSS1/MCDN RO/TAT Interworking scenario, after TAT has been applied on page 518.) Station B activates Network Call Transfer to Station C, located at Node 2 on the DPNSS1 side of the gateway.

Upon activation, the existing call is put on hold and a new call is originated to Station C. Station C Answers. Station B completes the call transfer, leaving A connected to C using two DPNSS1 trunks and two PRI trunks.

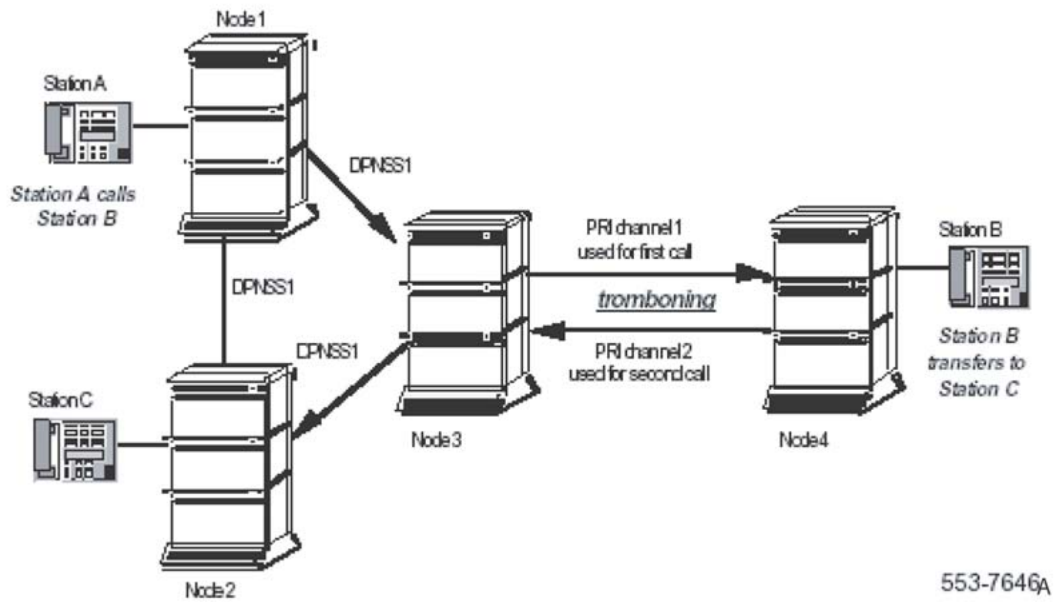

Figure 67: DPNSS1/MCDN scenario with Network Call Transfer, before RO/TAT optimisation

Note:

The Network Call Transfer/Three Party Service gateway is not supported at the gateway Node 3. Therefore, RO is not initiated at Node 1, and the non-optimised DPNSS1 trunks remain connected.

On the MCDN side, TAT is initiated at Node 4. The call between A and C is bridged, and the redundant PRI trunks are removed between Node 4 and Node 3. For the meantime, the non-optimised DPNSS1 trunks remain connected, as shown in Figure 68: DPNSS1/MCDN RO/TAT Interworking scenario, after TAT has been applied on page 518.

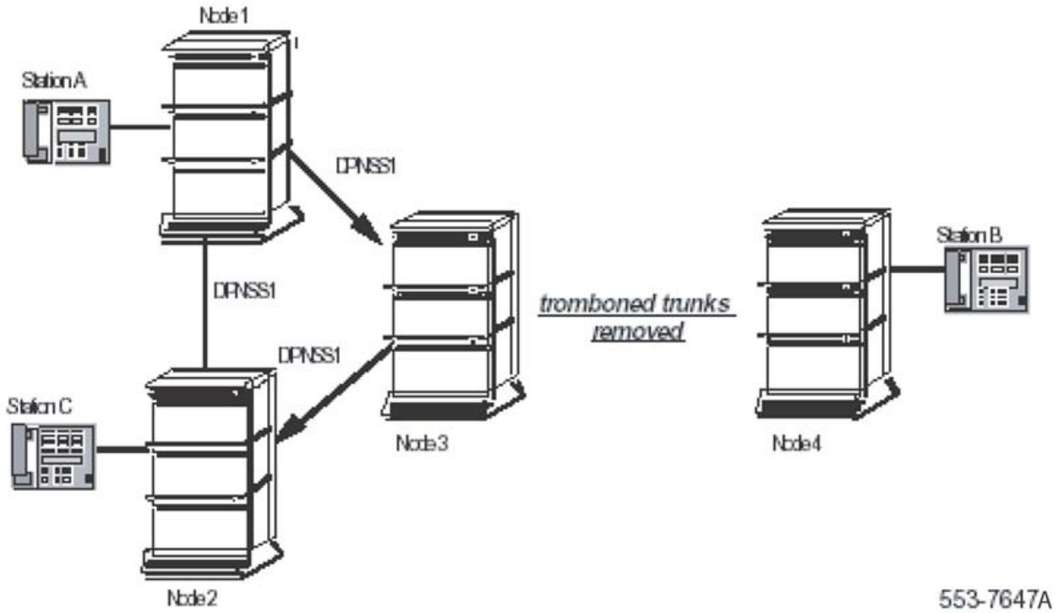

Figure 68: DPNSS1/MCDN RO/TAT Interworking scenario, after TAT has been applied

When TAT is completed on the MCDN side, The RO/TAT Interworking feature initiates RO on the DPNSS1 side by simulating a transfer at the gateway Node 3. The Three Party Service feature initiates signaling to update displays. Then, RO is initiated at Node 1, the originating node. The DPNSS1 trunks are dropped between Node 3 and 2 and Node 3 and Node 1, with Station A and Station C being connected over one DPNSS1 trunk. This is shown in Figure 69: DPNSS1/MCDN RO/TAT Interworking scenario, after RO has been applied on page 519.

Note:

If a non-optimum route is used at the originating node or at any transit node, Route Optimisation can start from Node 1 (the normal RO operation for the first call optimisation) or Node 3 (the normal RO operation for the second call optimisation), before TAT is completed. If TAT invocation is received on Node 3 while RO is being applied between Node 1 and Node 3 or Node 3 and Node 2, the completion of TAT is delayed until RO is totally finished.

Upon the completion of TAT on Node 3, a call transfer operation is simulated, and a new RO operation is initiated to remove any potential triangulation of routes.

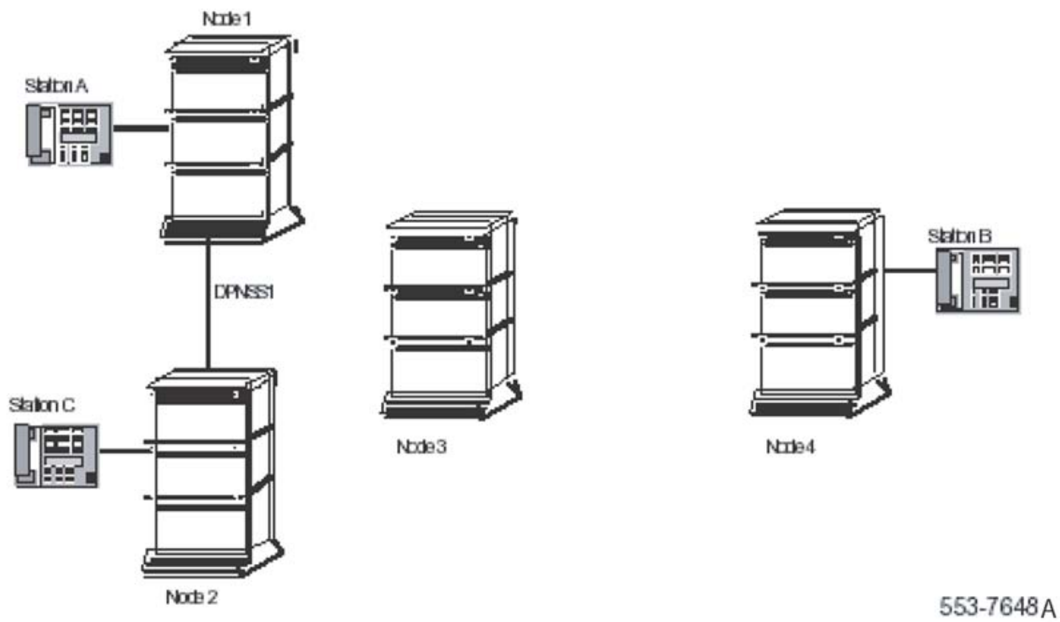

Figure 69: DPNSS1/MCDN RO/TAT Interworking scenario, after RO has been applied

Note:

If Station A is an attendant, TAT takes place on the MCDN side of the gateway but RO cannot take place on the DPNSS1 side. This is a RO limitation.

RO/TAT interworking within a DPNSS1 to MCDN gateway

The following example presents a case where RO/TAT interworking occurs within an MCDN to DPNSS1 gateway. Here, too, we are using the case of a call being transferred (using the DPNSS1 Three Party Service feature) across the gateway.

Referring to Figure 70: MCDN/DPNSS1 RO/TAT Interworking scenario, before RO has been applied on page 520, Station A, located at Node 1 on the MCDN side of the MCDN/DPNSS1 gateway, calls Station B located at Node 3 on the DPNSS1 side of the MCDN/DPNSS1 gateway. Station B transfers the call (using the Three Party Service feature) to Station C, also located at Node 1 on the MCDN side of the gateway.

Upon activation, the existing call is put on hold and a new call is originated to Station C. Station C Answers. Station B completes the call transfer, leaving A connected to C using three DPNSS1 trunks (in the example, the call is routed through Node 4) trunks and two PRI trunks.

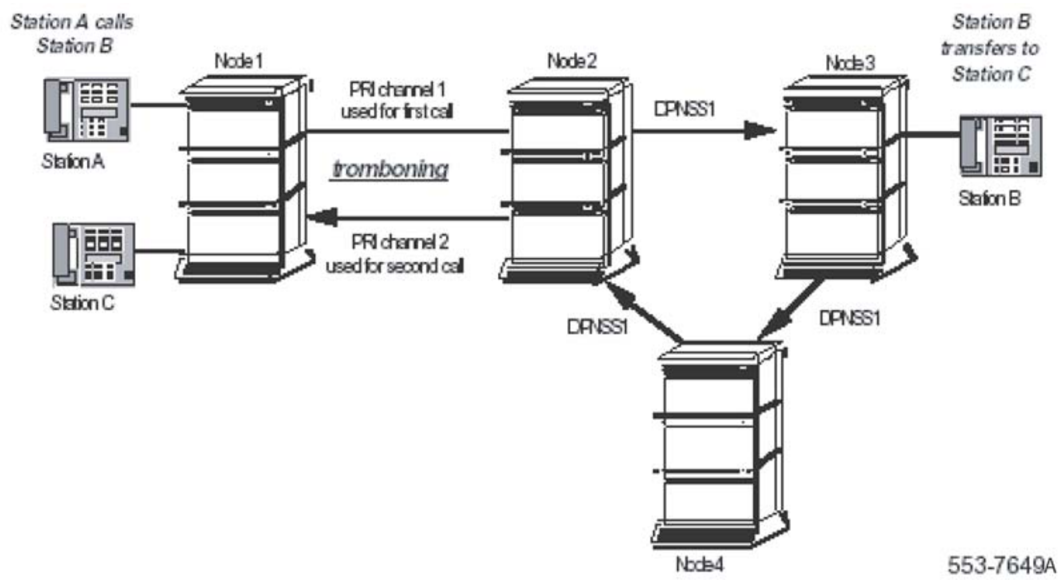

Figure 70: MCDN/DPNSS1 RO/TAT Interworking scenario, before RO has been applied

Once Three Party Service messaging has taken place, Node 2 initiates RO. The initial DPNSS1 routes are cleared. Node 2 becomes a MCDN/MCDN transit node, and the two tromboning PRI routes between Node 2 and Node 1 remain, as shown in Figure 71: MCDN/DPNSS1 RO/TAT Interworking scenario, after RO has been applied on page 520.

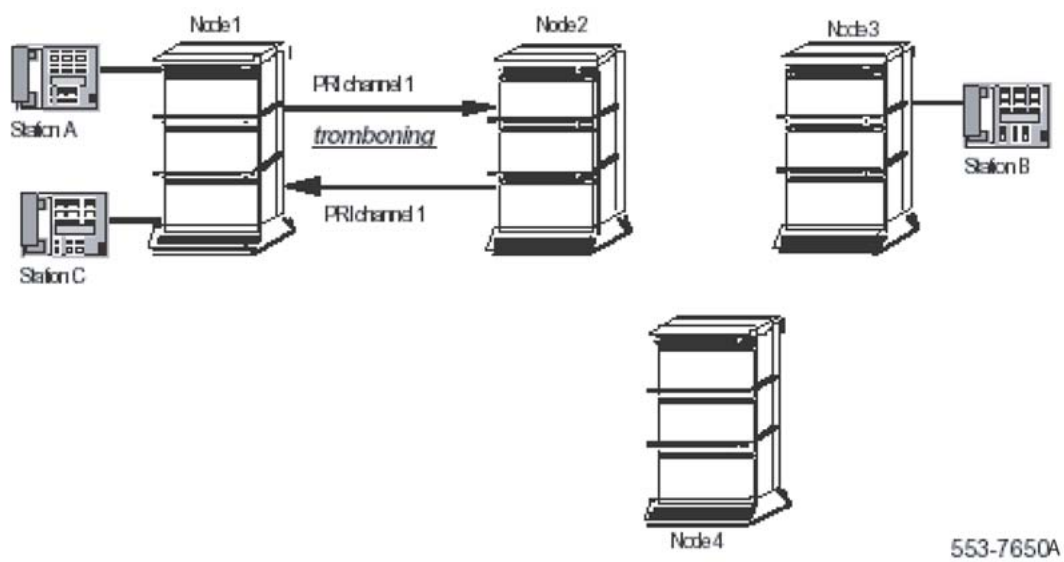

Figure 71: MCDN/DPNSS1 RO/TAT Interworking scenario, after RO has been applied

As soon as RO is completed, the RO/TAT initiates TAT at gateway Node 2. After TAT has been completed at Node 1, Node 2 simulates a transfer message to both Station A and Station C. This allows the Network Call Redirection feature to update the displays.

Note:

If the originating and terminating nodes are one and the same, and if this node is not a tandem node, as is the case for Node 1 in our example, the displays are updated without the notification from the Network Call Redirection feature.

TAT is then completed. The redundant routes are cleared, and Station A and Station C are bridged, as shown in Figure 72: MCDN/DPNSS1 RO/TAT Interworking scenario, after TAT has been applied on page 521.

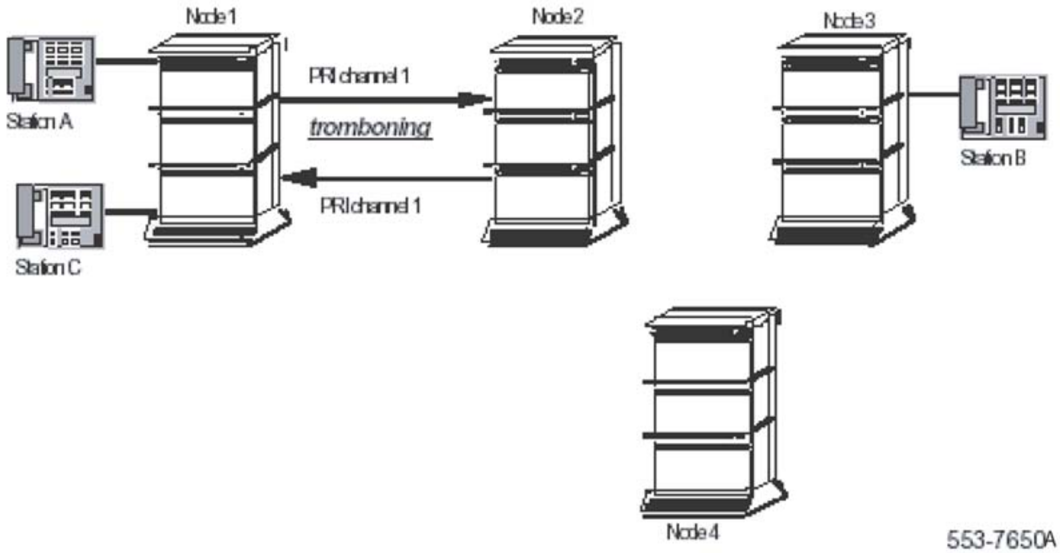

Figure 72: MCDN/DPNSS1 RO/TAT Interworking scenario, after TAT has been applied

Note:

If Station A is an attendant, and the Network Attendant Service feature is configured, Station B cannot transfer to Station C, and no optimisation can take place. If NAS is not configured, Station B can transfer to Station C, and optimisation takes place as described in this example.

Note:

In the case of call diversion on the DPNSS1 side (Diversion Immediate, Diversion on Busy, and Diversion on No Reply), there is no interaction with the RO/TAT Interworking feature (the interaction occurs between the Diversion and TAT features.) In the case of tromboning on the DPNSS1 side, the Diversion feature clears the DPNSS1 tromboning trunks before Station C answers the call. When C answers, TAT is applied transparently.

Note:

Node 1 cannot be a DMS switch for the RO/TAT Interworking feature to operate.

RO/TAT interworking within multiple MCDN/DPNSS1 gateways

A RO/TAT Interworking is supported within a multiple gateway scenario, as illustrated by the following example. Referring to Figure 73: RO/TAT Interworking within multiple DPNSS1/MCDN gateways, before RO/TAT on page 522, Station A on the originating node call Station B across the multiple gateway scenario over PRI and DPNSS1 trunks, as shown below. Station B then transfers to Station C, over different PRI/DPNSS1 trunks. When Station C has completed the call transfer, and Station C answers, TAT is first activated at the far end node, removing the two end PRI trunks. The RO/TAT Interworking feature then activates RO on the DPNSS1 portion of the gateway, removing the DPNSS1 trunks. Then, TAT is activated to remove the last two PRI trunks at the near end of the gateway, leaving Station C and Station A bridged, as shown in Figure 74: RO/TAT Interworking within multiple DPNSS1/MCDN gateways, after RO/TAT on page 522.

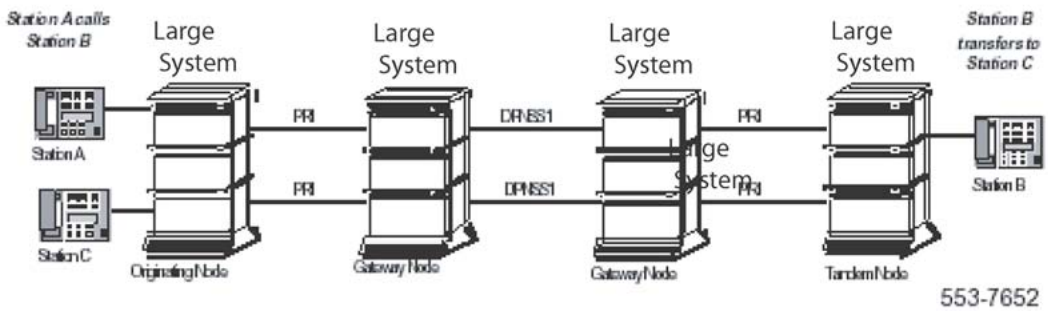

Figure 73: RO/TAT Interworking within multiple DPNSS1/MCDN gateways, before RO/TAT

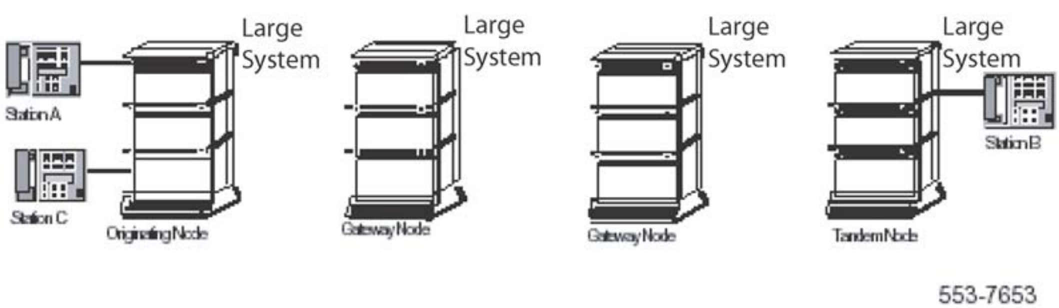

Figure 74: RO/TAT Interworking within multiple DPNSS1/MCDN gateways, after RO/TAT

Abnormal RO/TAT interworking scenarios

The following are scenarios where the RO/TAT Interworking feature can function abnormally.

1. RO fails or is not configured, and TAT is configured.

   In the case of a DPNSS1/MCDN gateway, TAT optimises the PRI trunks on the MCDN side, but the DPNSS1 trunks are not optimised on the DPNSS1 side.

In the case of an MCDN/DPNSS1 gateway, RO is not activated and the DPNSS1 side is not optimised. Since the DPNSS1 trunks remain, TAT is not invoked at the gateway node, even though it is equipped. Therefore, if RO is not activated, the RO/TAT Interworking functionality is not invoked.

2. TAT fails or is not configured, and RO is configured.

   In the case of an MCDN/DPNSS1 gateway, RO optimises the DPNSS1 trunks on the DPNSS1 side, but the MCDN trunks are not optimised on the MCDN side.

   In the case of a DPNSS1/MCDN gateway, TAT is not activated on the MCDN side and the tromboning PRI trunks remain. Since the PRI trunks remain, RO is not invoked at the gateway node, even though it is equipped, and DPNSS1 trunks are not optimised on the DPNSS1 side. Therefore, if TAT is not activated, the RO/TAT Interworking functionality is not invoked.

Operating parameters

Although Trunk Anti-Tromboning functions between a system switch and a DMS switch, no TAT messaging is initiated to a DMS switch after Route Optimisation is activated on the DPNSS1 side of an ISDN MCDN/DPNSS1 gateway.

As explained in Abnormal RO/TAT interworking scenarios on page 522, both RO and TAT must be activated in order for the RO/TAT Interworking functionality to operate.

The RO/TAT Interworking functionality is only activated after call connection.

RO/TAT Interworking functionality is not applied if the originating party of the first call or the terminating party of the second call is on a conference call.

RO/TAT Interworking functionality is not applied if the originating party of the first call is an attendant.

RO/TAT Interworking functionality is not applied to data calls.

Route Optimisation can be applied to any portion of a DPNSS1 network, as long as both the originating node and terminating nodes are equipped with the RO feature. This is because optimisation is performed by initializing a new call between the originating node and terminating node. However, for the same to apply to Trunk Anti-Tromboning within an MCDN network, every exchange along the network must be equipped with the TAT feature. This is because TAT releases trunks step by step.

Multiple hops across a gateway are supported separately by RO and TAT.

Feature interactions

Multiple Hops

Multiple hops are supported within every RO/TAT Interworking gateway scenario, since they are supported separately by RO and TAT.

Network Attendant Service

If tromboning trunks are removed on the MCDN side of a RO/TAT Interworking gateway scenario by the Network Attendant Service feature (since NAS has predence over TAT), the RO/TAT Interworking functionality is not invoked. The result is that, if NAS is equipped, attendant-extended calls that are in a tromboning state are optimised on the MCDN side, but DPNSS1 trunks are not optimised on the DPNSS1 side of the RO/TAT Interworking gateway scenario.

Network Call Pickup

If tromboning trunks are removed on the MCDN side of a gateway scenario by the Network Call Pickup feature (since Network Call Pickup has predence over TAT), TAT is invoked since the Network Call Pickup action is considered as a call forward action. RO/TAT functionality is invoked upon completion of the TAT operation.

Network Call Redirection

If Network Call Redirection is not configured in an DPNSS1/MCDN gateway, the displays are updated normally, since the RO/TAT Interworking feature is not affected.

If Network Call Redirection is not configured in an MCDN/DPNSS1 gateway, the displays are not updated on the bridged 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.

Three Party Service

DPNSS1 Three Party Service is required for every RO/TAT Interworking scenario.

Trunk Route Optimization before Answer

There is no interaction between the Trunk Route Optimization before Answer feature and the RO/TAT Interworking feature, since Trunk Route Optimization before Answer is activated before call completion, and the RO/TAT Interworking functionality is only activated after call connection.

Virtual Network Services

The RO/TAT Interworking feature is not supported over VNS trunks, since VNS uses only MCDN signaling (DPNSS1 is not supported.)

Feature packaging

For the software packages required to support the DPNSS1 Route Optimisation/MCDN Trunk Anti-Tromboning Interworking feature, consult the following documentation:

- For DPNSS1 network functionality, refer to *Avaya DPNSS1 Fundamentals, NN43001-572*.
- For MCDN Network Attendant Service interworking, refer to the NAS feature description module in this book.

Feature implementation

No new steps are required to configure the DPNSS1 Route Optimisation/MCDN Trunk Anti-Tromboning Interworking feature. However, the following basic configuration must be done:

- Configure MCDN Trunk Anti-Tromboning at the far-end switch, in LD 17 (TAT is configured on a D-Channel basis, and not on a route basis). Refer to the Trunk Route Optimization feature in this book.
- Configure DPNSS1 Route Optimisation options, in LD 15. Refer to *Avaya DPNSS1 Fundamentals, NN43001-572*.
- Configure call display transfer indication for DPNSS1 Three-Party Service in LD 95. Refer to *Avaya DPNSS1 Fundamentals, NN43001-572*.
- Optionally (to update terminal displays), configure Network Call Redirection using LD 15, LD 16, LD 95, LD 10, and LD 11. Refer to the Network Call Redirection feature in this book.

Feature operation

No specific operating procedures are required to use this feature.

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